IPitomy IP PBX Administrator Guide

Version v4.0.1

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INTRODUCTION

About the IPitomy IP PBX

The IPitomy IP PBX is a powerful business communications platform. It is a pure IP PBX designed to use IP networks for voice calls. Engineered to support from 10 to 500 users, the system will work with analog lines and T1 /PRI lines for traditional Public Switched Telephone Network (PSTN) connectivity. In addition to traditional telephone lines, the IPitomy IP PBX can use VoIP SIP Trunks, replacing traditional PSTN lines with a broadband telephone service.

Benefits of VoIP Technology

The IPitomy IP PBX can support any or all of these connectivity methods simultaneously or in any combination. Customers not quite ready to depend on VoIP providers for all of their business communications can start at their own pace and gain a comfort level, shifting to VoIP broadband providers at their own pace.

Benefits of VoIP technology include:

- **One Wiring System** – The system uses a single wiring system for telephones and data—all data and voice are on Local Area Network (LAN) Category 5 wiring.
- **Web-based Administration** – System administration is performed on the network through a Web-based administration program. The Web-Based Administration can be used locally or remotely from anywhere.
- **Remote Users** – When calls are routed over the Internet, long distance charges can be avoided. In businesses with remote workers, these employees can stay logged into the office through a broadband connection at all times without incurring any additional charges. Remote users have all of the features of the local users. Remote users can be included in any ring groups, ACD (Automatic Call Distribution) Queues and other call routing schemes.
- **Centralized System Features** – Every extension that is logged into the system is capable of receiving and originating calls. The use of system features such as voicemail, automated attendant and email are all centralized simplifying all support and maintenance.
- **Reduced Costs** – VoIP system users can reduce cost in many areas of a business. VoIP telephony lowers the cost of support and maintenance costs, as well as, reducing telephony line costs by up to 50%.
- **Simplifies Administration** – Moves, additions and changes are simple. The IPitomy IP PBX provides enhanced capabilities for users to make changes without incurring a service call.
- **Investment Protection** – VoIP, and in particular, Session Initiation Protocol (SIP)-based VoIP products offer investment protection. The industry is rapidly moving toward Internet Protocol (IP) communications technologies. Older digital and analog technologies are becoming obsolete and are being replaced with IP-based products that will be around for a long time.

IPitomy IP PBX Features

Understanding the IPitomy IP PBX's architecture and how it works will make installing the system simple.

The IPitomy IP PBX is an all-in-one business communications system. This powerful system includes a complete suite of business communication applications in one appliance:

- Fully-featured Business Phone System
- Automated Attendant and Interactive Voice Response (IVR)
- Enhanced Call Distribution

- Enhanced Voice Messaging System with Unified Messaging
- Meet-me Conference Application
- Built-in Music on Hold
- Call Queuing for Inbound Calls
- Find Me/Follow Me
- Remote Extensions
- Browser-based Administration
- Branch Offices
- Automatic Call Distribution (ACD)

Figure 1 – IPitomy IP PBX System

The IPitomy IP PBX's administration menus are a series of Web pages accessible from a Web browser. To the left of the Menu is a navigation bar that allows users to click on and administer each section of the system. Administration of the IPitomy IP PBX is simple and intuitive. The system is designed with six primary areas of functionality:

- **System** – System setup consists of network configuration settings.
- **Providers** – Providers are sources of PSTN and VoIP connectivity. Providers are the lines that handle all incoming and outgoing calls. All VoIP and traditional telephone providers are setup here. DID numbers are also entered here.
- **Destinations** – Destinations are places where calls are routed in the system: extensions, groups of extensions, automated attendants, conferences, and voicemail.
- **Call Routing** – These settings route inbound calls to specific destinations within the system, and send outbound calls over specific local, long distance, international, and emergency routes.
- **PBX Setup** – These settings globally configure PBX timers, voice messaging, and other system features.
- **Reporting** – These reports display system usage, monitor activity, and provide diagnostic information.

Feature	Description
Extensions	Extensions are telephones. A telephone can be an IP (SIP) telephone or a Softphone. Calls are routed to an extension where people answer them. In the IPitomy IP PBX, an extension can be located in an office or outside the office where a broadband connection is used.
Groups	Groups are a set of extensions. Once a group is created, extensions can be designated as members of the group. This is accomplished by selecting group members from a drop-down list. Calls can be routed to groups via inbound routing.
Menus (Automated Attendant)	To create an automated attendant use the system's Menus feature. The Menus feature allows you to route calls to a destination in the system like a group, extension or another menu. Call Destinations are selected from a drop-down list for each corresponding key-pad digit a caller must select to get to their chosen destination. A Menu must have a Menu Prompt. This is a recording that identifies for callers the destinations they may choose. For example, a Menu Prompt might offer callers the option to press "1" for Sales, "2" for Accounts Receivable or other digits for another department.
DTMF Menu Administration	New in version: 3.4.1 is DTMF Menu Administration. This feature allows the user to administer the Menu (Auto Attendant) remotely using just a telephone (with DTMF dial capability).
Advanced Routing Functions	When building an automated attendant (Menus) all routable destinations in the system will appear in the drop-down list. In addition to the destinations that are created while configuring the system, there are several advanced functions that can be used from the drop-down list.
Voicemail and Unified Messaging	When an extension is created, a voicemail box for that extension is also created. A voicemail box allows a caller to leave a message if a person is not available at the extension. When dialing into a mailbox for the first time, a user should record their name and a mailbox greeting. The name is used in the company's dial-by-name directory when selected from the auto attendant (Menus). The greeting is played when they are not available to take a call and a caller reaches their mailbox. If an email address is included in the Extension page, you can configure Unified Messaging and a copy of the voicemail message will be emailed as a .Wav file to the users email account. This message can then be listened to on a PC.
Directory	The system has a dial-by-name directory. This option may be part of the automated-attendant. When this option is selected, a caller dials the first three letters of the last / first name of the party they would like to reach. Names that match these three letters are played and the caller selects the extension to which they want to be transferred. Names are stated in the directory as they have been recorded by users in their voicemail box, and spelt out if they have not recorded their name.
Direct Inward Dialing (DID) Numbers	A Direct Inward Dialed (DID) number is a telephone number assigned by a service provider (i.e., T1 line, PRI or VoIP). DIDs allow direct routing of a call to a destination within the system. You can route to any destination available on the PBX.
Conferencing (Meet Me)	A Meet-me Conference is an extension on the system used for conference calls. Participants can access a conference by dialing the designated Meet-me Conference extension. Routing callers to a Meet-me Conference can be accomplished by using a DID, a menu, or simply transferring callers to the conference extension.

FollowMe	This feature allows the PBX to try and find users who are not at their desk. It can be configured to call their cell phones, house phones, or other extensions in the PBX. Once answered, the user can accept the call, or refuse it. Unhandled calls return to the PBX to leave a message at the original extension's voicemail.
Forwarding Gateway	Mobility has become a part of everyday life for most people. System users need to be able to take calls anywhere. The IPitomy IP PBX has the ability to forward calls. Users can turn call forwarding "**on**" and "**off**" while in the office or away from the office by using a touch-tone key pad. This is set up in the edit Extensions page, but can be modified from any phone, including a cell phone. Modifying forward settings remotely requires the automated attendant (Menus) option to be programmed.
Cascading Message Notification	This feature works using the same methods as FollowMe, but pertains to voicemail messages. When configured, if an extension gets a new voicemail, you will be able to send the voicemail message to a variety of numbers (Destinations), define the order in which to send the message and can be set to make the system to notify you that a new message was received. Additionally, you can add or remove extensions to the list of recipients when a broadcast message is sent.
Voicemail Gateway	Using either a Menu or a DID, users can call in from any telephone and check messages. The voicemail gateway allows users to dial a pre-defined digit from a touch- tone key pad on any phone to retrieve their messages.
Branch Offices	Branch offices can be created to allow multiple PBXs to route calls to each other. Branch office extensions can be transferred to, placed in ring groups, or selected as menu destinations.

Table 1 – IP PBX Features

GETTING STARTED

Boot Up – Safe Power Down

Each IPitomy IP PBX is equipped with a Boot Up/Safe Power Down momentary switch at the front of the chassis. This switch is a primary element of controlling not only the hardware but the Operating System (OS) and PBX software. If it becomes necessary to use this switch – especially when powering down – it is imperative that you read and understand the functionality of this switch.

IMPORTANT:

Use of this switch is very convenient and VERY specific.

Improper use can cause very undesirable results.

Use this switch by pressing the elevated side and releasing immediately.

1100	1200	2000/5000
		 Behind Drive Access Door

The switch has three functions:

(Use 1 & 2 only – unless directed to use the 3rd function by a technical support representative). Repeatedly turning off the power with a hard boot can damage (corrupt) the data on the hard drive. In some instances, this can void the warranty.

Use	Condition	Operation
1.	While the PBX is **not** operating… (power cord connected but not yet powered up OR previously, properly powered down)	Press and release the switch to start PBX operation
2.	While the PBX **is** operating… Shut down can take as long as 5 minutes. **Please be patient.**	Press AND **Release** the switch to initiate the safe power-down sequence. The PBX will shut-down all applications running in the proper sequence – ensuring that no damage occurs to the database and applications during shut-down.
3.	While the **PBX** is **NON-Responsive** and all other methods to regain control of the OS and applications have failed.	Press AND **Hold** the switch to FORCE immediate shut-down. This is **NOT recommended** and should be avoided.

Connecting the System

Hardware Setup

The IPitomy IP PBX comes assembled and ready to install. The system requires some form of trunking, be it a connection to the PSTN for analog or T1 lines or a SIP trunk. It requires telephones to be connected to the local area network (LAN). Broadband access must be established for VoIP connectivity (allowing remote extensions , branch offices, remote management, and SIP trunks).

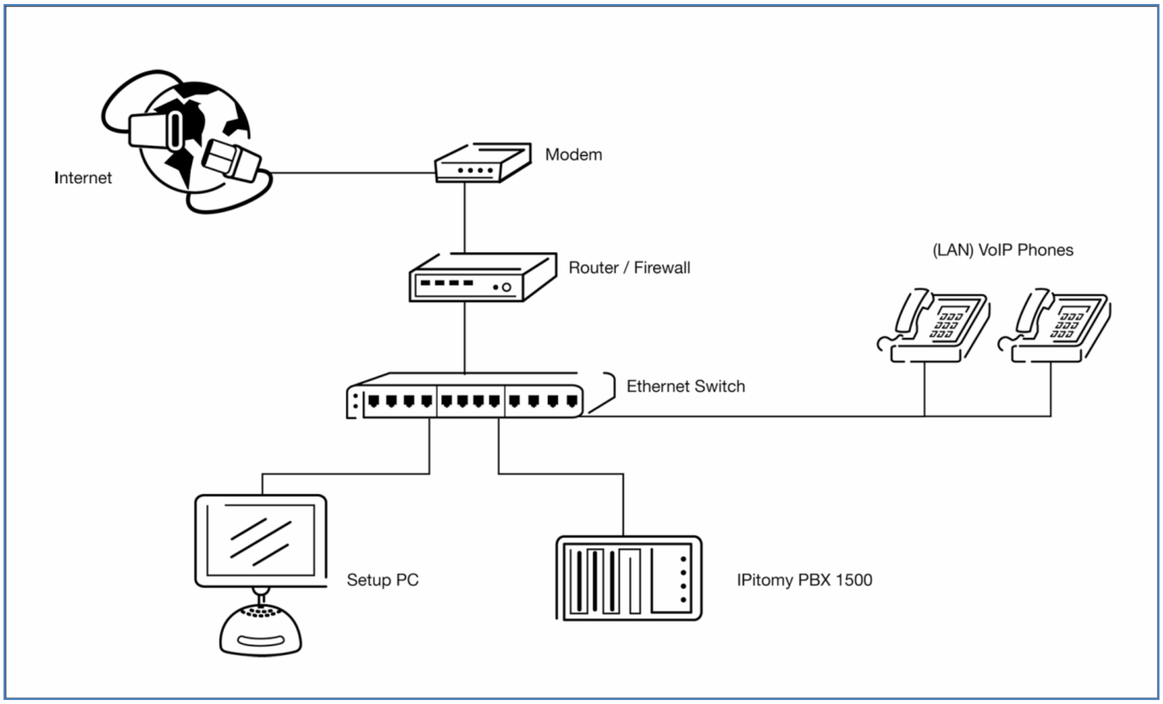


Figure 2 – Hardware Setup Diagram

Connecting the Phone Lines

The IPitomy IP PBX is equipped to support analog, digital, gateway or SIP connections. Analog lines or a T1/PRI are connected with internal hardware resources. A gateway connects analog telephone lines by registering itself as a SIP provider over the LAN. SIP providers create a direct connection to the system.

INTERNAL ANALOG LINE CARDS

The IPitomy IP PBX uses three basic analog devices for PSTN connectivity.

Digium TDM04B Analog Line Card (4 PSTN line connections) – This analog line card supporting these connections is already installed and completely configured. Simply connect the phone lines to the RJ11 jacks at the rear of the IPitomy IP PBX and start making calls.

Digium TDM808B Analog Line Card (8 PSTN line connections) – This analog line card supporting these connections is already installed and completely configured. Simply connect the phone lines to the RJ11 jacks at the rear of the IPitomy IP PBX and start making calls.

IMPORTANT: The 4 and 8 PSTN line connections are a single pair, one line per jack. It may be necessary to echo tune the system. Please contact an IPitomy Technical Support personnel if you notice an echo on the line.

Digium 2404B 16 and 2406B 24 Analog Line Cards (16 or 24 PSTN line connections) – These lines are plugged into an Amphenol connector and need to be terminated in a cross connect, break-out box or patch panel. The Amphenol connector uses a single pair connection to each phone line.

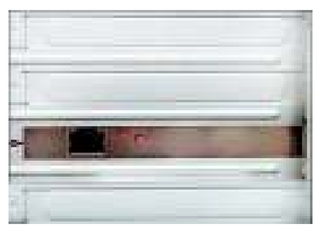
(Diagram 4)

INTERNAL DIGITAL T1 CARDS

The IPitomy IP PBX uses two devises for digital connectivity.

Digium TE122P T1 Card (a single T1/E1/J1 connection) – This card supports industry standard telephony and data protocols, including both RBS and Primary Rate ISDN (PRI) protocol families for voice traffic. The card has a default configuration for the system, but this configuration may be adjusted based on preference. Plug the RJ45 connector into the T1 TDM source supplied by the T1 provider.

Digium TE205P T1 Card (a dual T1/E1/J1 connection) – This card supports industry standard telephony and data protocols, including both RBS and Primary Rate ISDN (PRI) protocol families for voice traffic. The card has a default configuration for the system, but this configuration may be adjusted based on preference. Plug up to two RJ45 connectors into the T1 TDM sources supplied by the T1 provider.



(**Diagram 5)**

Connecting Using an External Gateway

Gateways connect legacy phone equipment (PBXs, ACDs, voicemail systems, etc.) to modern VoIP systems and services. IPitomy supports many different communications protocols from both the modern world of VoIP and from the legacy PSTN. This makes it a powerful tool for building gateways and protocol converters.

PSTN lines can be connected to a Gateway device. The gateway device is connected to the LAN. The Gateway is then registered as a SIP provider in the system.

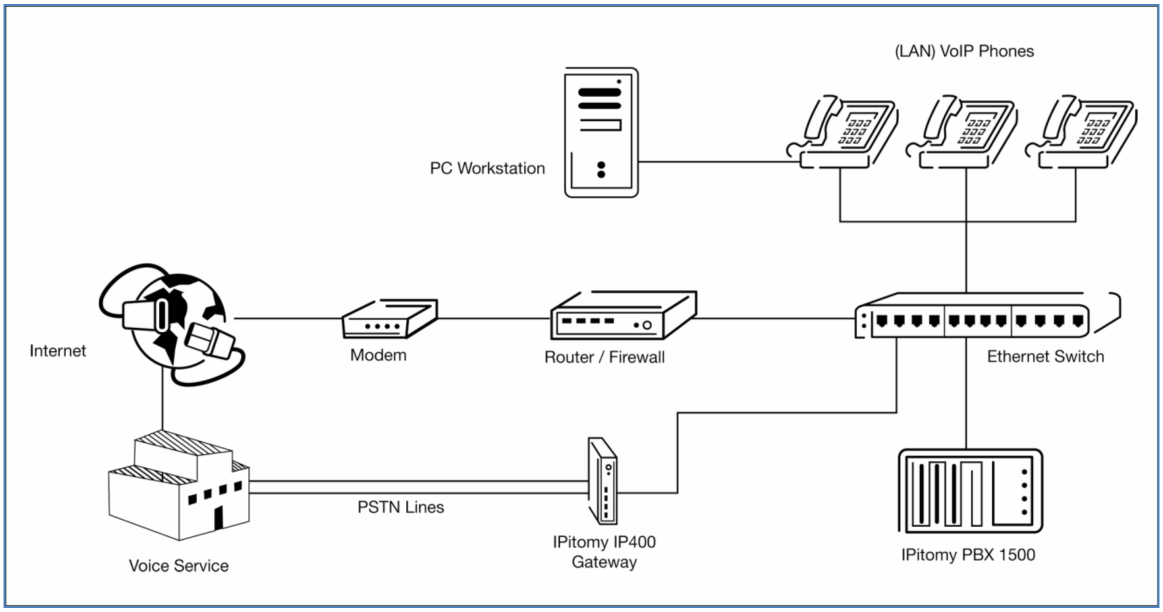


Figure 3 – External Gateway Connection Diagram

Connecting to a LAN

In order to be able to connect all of the devices (PCs, Phones, Gateways, etc) to their Local Area Network (LAN) you will most likely need to install at least one Switch.

Connecting Using SIP Providers

Once connected to the LAN, the LAN's broadband connection provides a pathway for SIP VoIP Providers. Use the SIP Provider pages to setup a connection.

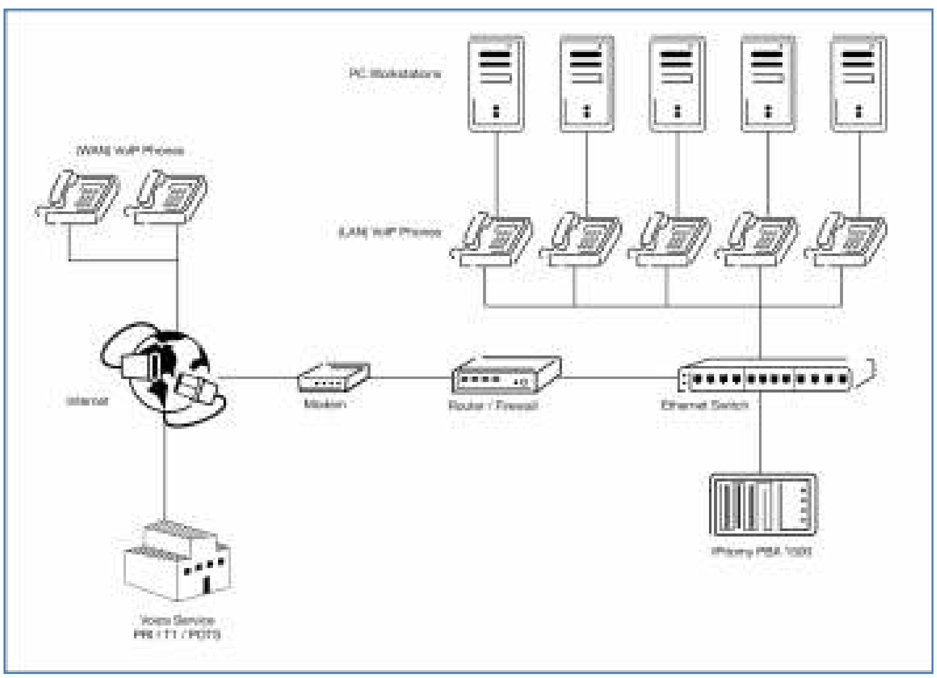


Figure 4 – SIP Provider Connection Diagram

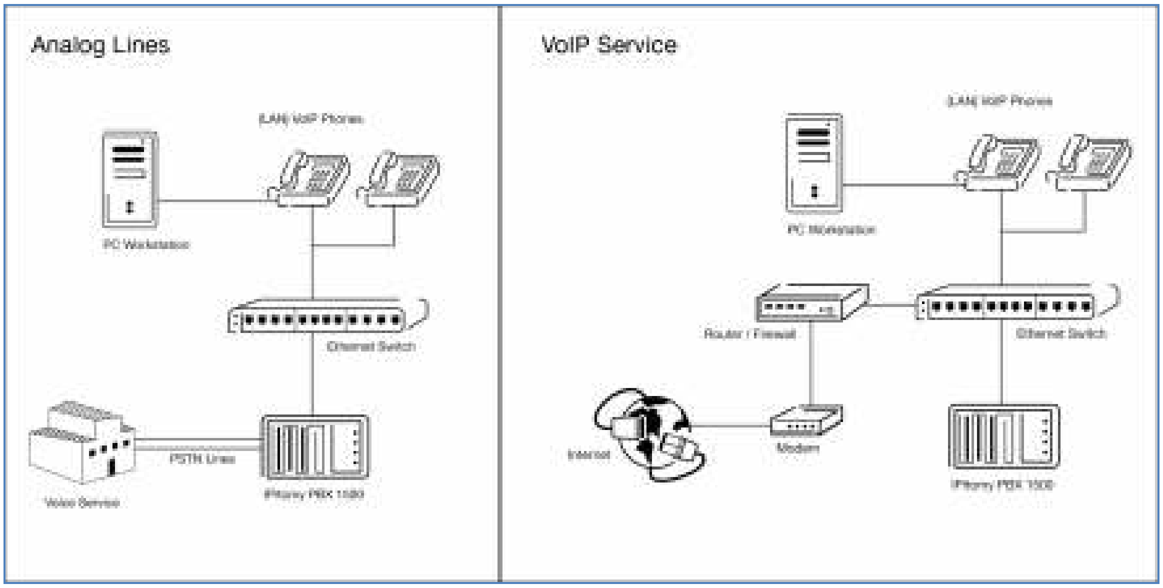


Figure 5 – LAN Connection Diagram

Connecting Telephones

The following diagrams indicate the port locations on the IP120 and IP1500.

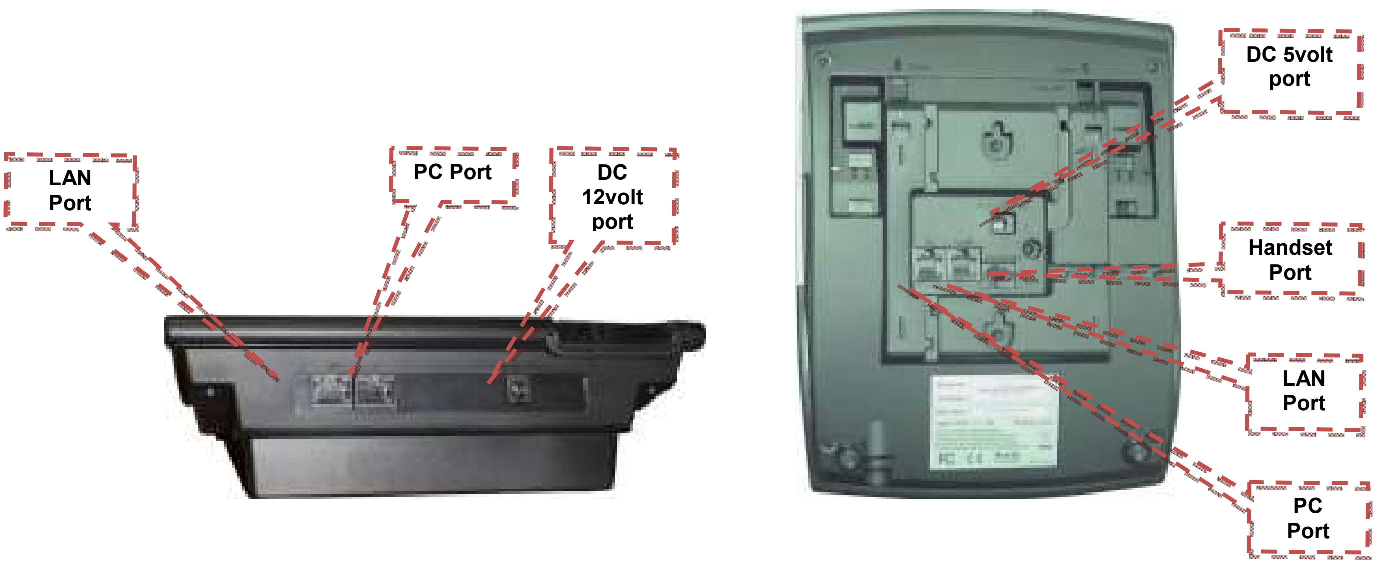


Figure 6 – IP120 and IP1500 Connector Ports

DATA AND NETWORK CONFIGURATION

Planning before getting started will make the setup and installation of the IPitomy IP PBX simple. IPitomy has created the **IPitomy Setup Worksheet** to assist in recording business and system information used in planning system setup and installation. This Checklist can be downloaded from IPitomy.com in the Dealer Section of the Website.

IMPORTANT: Please contact your IPitomy Sales Representative to obtain a login information for the Dealer site at (941) 306-2200.

Setup Worksheet

The **Setup Worksheet** is broken out into sections where required data can be entered. Once all the information in the setup worksheet is populated, it can then be imported to the IPitomy system. This will save a great amount a setup/configuration time.

IMPORTANT: To take full advantage of the functionality in the setup worksheet, you will need to make sure that your Excel macro security is set to Medium or lower (you will need to restart Excel after changing these settings). I

Welcome to the Ipitomy Setup worksheet. This worksheet is designed to help you organize the data required to set up an IPitomy IP PBX system.

By planning the application, gathering, and organizing the data in advance, installation is simple and can be completed rapidly. You may want to provide a copy of this document to your customer for them to fill in certain items like names and email addresses. To fully take advantage of the functionality of this worksheet, you will need to make sure that your Excel macro security is set to Medium or lower (you will need to restart Excel after changing these settings). If set to Medium you will need to click 'Enable Macros' when prompted upon opening the document, if set to Low macros will be automatically enabled.

Information required for the successful installation of an IPitomy IP PBX system:

Network Information	Prompts
You will need to know some basic information on the target network. The type of data facilities you will be using for access to the Internet will dictate how much bandwidth is available for phone calls if you are going to use IPitomy EXchange or any other VoIP service provider. You will need to have a fixed IP address if calls are to be sent to remote extensions via the Internet. The IP address of the PBX on the local subnet will need to be static.	Prompts for menus can be recorded using the sound recorder on your PC or through the prompt recording feature in IPitomy. If you use a sound recorder, you will need to upload the file to IPitomy. If you use the prompt recording feature in IPitomy, prompts are recorded using your telephone extension. Voice mail greeting are recorded using the mail box menu.
User Data	Telephone Lines
The required information for users will be extension number, name and email address. The email address is used to send voice mails to email. When an extension is created, a voice mail box is automatically created for that user. Voice mail only boxes may be created as well. You can print out a copy of this sheet and ask your customer to fill it in or provide them the spreadsheet to fill out.	If you know the telephone numbers for your telephone lines, it is good to have then all listed ahead of time. You may want to route calls based on a particular number or specific DID.
Groups	Music On Hold
Once you have all the users entered, the users can be put into logical groups for call routing. An example would be to put all of the sales people in the sales group. Calls from the automated attendant or from DID's can be routed directly to that group. Calls can be distributed to the members in the group in several ways by using the ring strategies in the group. Ring strategies are: ring all, round robin, fewest calls, least recent, fewest recent calls, round robin with memory or random.	When a caller is place on hold, park, or is in Queue for a group of extensions, music may be played for them while they wait. The music on hold files can be uploaded to the system from your browser. The music on hold files need to be in MP3 format for IP1200 and IP1500, and 8bit 8Khz mono .wav file for the IP1000.
Menus	Licensing
Menus are for creating automated attendanst and other audio menus. Once you have created groups and extension, calls can be routed to them from the automated attendant. Prompts can be recorded to provide instructions.	IPitomy comes with a license for 16 users/extensions. To activate the initial license, you must have a connection to the internet and a license key will be uploaded to the system. This is a security feature that protects the user for warranty coverage as well as against unauthorized use.
Unified Messaging	Branch Office
Unified Messaging allows the PBX to send your voicemail messages directly to your email inbox.	Branch Offices allow multiple sites with PBXs to interact as one large office. Unique extension numbers can be configured as branch extensions and dialed directly, while non-unique extension numbers can be dialed with the dial prefix + ext number; the same is true for Ring Groups and Menus. You can also configure at each PBX whether they will allow the other to use their trunks for calling.

Figure 7 – Sample Setup Worksheet

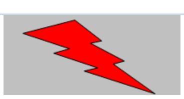
Data Type	Description/Recommended Settings
Network Information	You will need to know some basic information on the target network. Use this page to store all pertinent information regarding the sites LAN and WAN connection. • The type of data facilities you will be using for access to the Internet will dictate how much bandwidth is available for phone calls. • If you are going to use remote phones, SIP providers, or Branch Offices, you will need to have a fixed IP address.
User Data	Use this page to gather and store information for the extensions and stand alone mailboxes on the PBX. This page allows you to export the extension information to another worksheet. Save this worksheet as a .csv file and you can import it directly into the PBX, saving you time. • **Extension Numbers –** required • **Name –** required • **Email Address -** required if the end user desires to use Unified Messaging • **Device –** setting this in the worksheet will make your installation much easier when using autoprovisioning and IP550/IP120 phones • **MAC Address –** for the most part this field can be left blank. If using Aastra phones, it may be more useful. Voice mail only boxes may be created as well. You can print out a copy of this sheet and ask your customer to fill it in or provide them the spreadsheet to fill out.
Groups	Use this page to gather and store information about ring groups. This will allow you to build the groups in the PBX much quicker at the time of install because everything has already been decided upon in advance.
Menus	Building menus in the PBX will be much easier once you have completed the sections pertaining to Menus. In addition to the setting in the PBX, you will be able to have the end user write out the prompt scripts, ensuring that the key press destinations match the dialogue the callers will hear.
Unified Messaging	Unified Messaging allows the PBX to send your voicemail messages directly to your email inbox. Use this section to log the end users email server information.
Telephone Lines	If you know the telephone numbers for your telephone lines, it is good to have them all documented ahead of time. You may want to route calls based on a particular number or specific DID.
Music On Hold	When a caller is place on hold, park, or is in Queue for a group of extensions, music may be played for them while they wait. The music on hold files can be uploaded to the system from your browser. The music on hold files need to be in MP3 format for IP1100+ systems, and 8bit 8Khz mono .wav file for the IP1000.
Licensing	Most IPitomy systems come default with a license for 16 users/extensions. To activate the initial license, you must have a connection to the internet and a license key will be uploaded to the system. This is a security feature that protects the user for warranty coverage as well as against unauthorized use. Filling out the User Data page will automatically tell you how many extensions the system needs, at a minimum, to handle the desired users.
Branch Office	This page will allow you to more accurately plan out an install that will have multiple sites connected via Branch Office. Using unique numbering for each PBX will allow for easier calling between sites.
	The lightning bolt, found on the User Data section of the Setup Worksheet, accesses a macro that allows you to export the User Data information pertaining to extension to another worksheet. Once exported, you can save that worksheet as a .csv file, and then load it directly into the PBX.

Table 2 – Setup Worksheet Descriptions

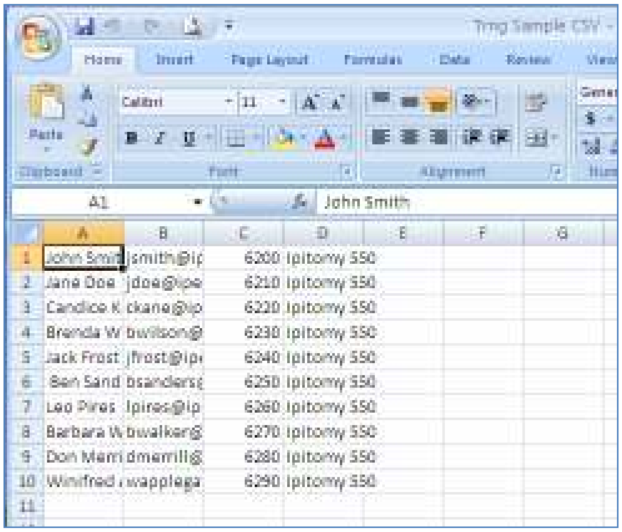


Table 3 – Sample Exported .CSV Worksheet

CSV Upload

Once all the extension information has been exported to a .CSV file, you can import that file into the PBX.

STEPS:

1. Navigate in the PBX to **Destinations ▶ Extensions.** The **Extensions** page appears.
2. Select the **BROWSE** button at the top of the page. The **File Upload** window appears. Locate and select the file that you want to import then click the **OPEN** button.

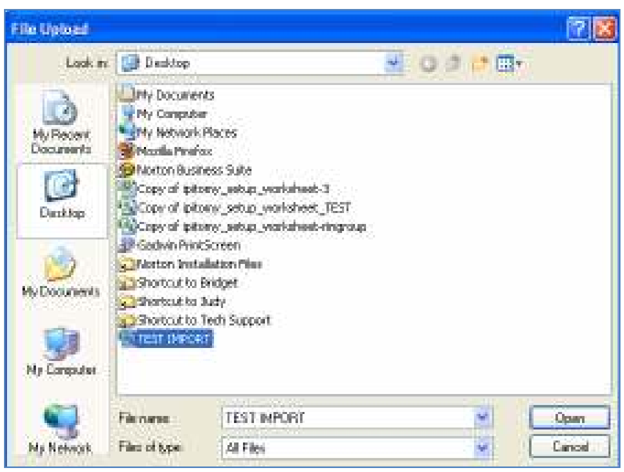


Figure 8 – View of Upload Location Window

3. The **CSV File** field is updated with the source directory information for the file to be imported.
4. Click the **IMPORT** button. The **Create Extensions** page appears displaying the information that is going to be imported.
5. Review the information then select the **CREATE** button when you are done.

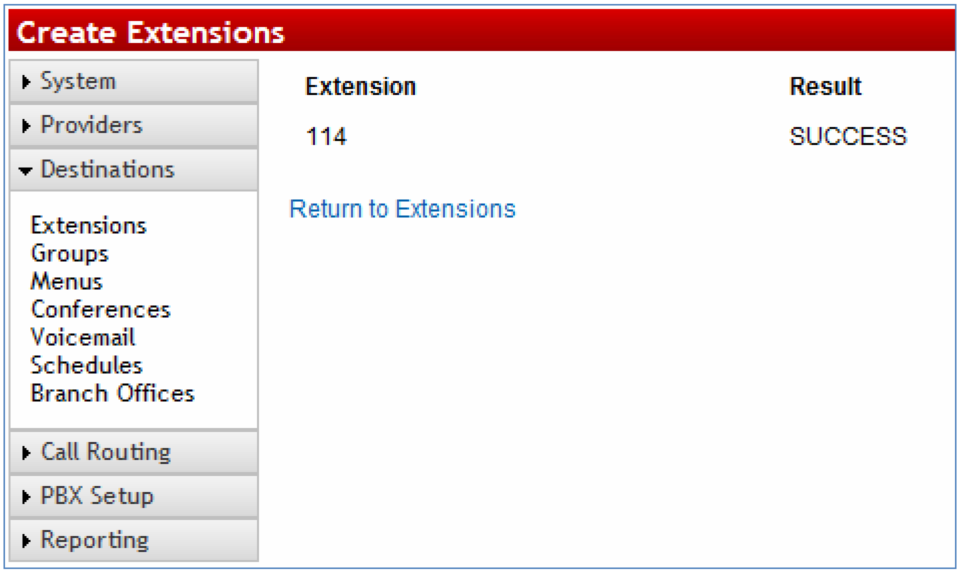


Figure 9 – Create Extension Results Page

6 The **Create Extensions** page will display the results of your import for each extension. You should see a “**SUCCESS**” message.

7 Click on the **Return to Extensions** link that appears on the page. This brings you back to the Create Extensions page. Review the Extensions List and find the extension (s) you just created.

8 Click on the **Apply Changes** link at the top of the page to make these changes live on the PBX. A message “Settings Applied” will appear.

9 You are now ready to update the various settings for the new extension. (*See the **Destination Extensions** section of this user guide for more details on Extensions)*.

Network Requirements

Making preparations for the network in advance will assure there are no surprises. If you are going to have remote extensions, you will need access to the router to setup a network address translation (NAT) and port forwarding.

A LAN with a broadband connection is required for operation of the system. It must be on a fast Ethernet (100baseT or better). The system must also use Ethernet data switches. The router can use DHCP or not, depending on preference. The IPitomy IP PBX requires a fixed IP on the router subnet. Several ports may need to be forwarded. Make sure your router has TCP/IP port range forwarding by checking the box or product guide. The router should have a fixed IP address with a public IP.

Pre-existing TFTP servers or DHCP servers broadcasting TFTP servers may cause conflicts for phone configuration. If you have a pre-existing TFTP server being broadcast by your DHCP server, please contact IPitomy support to determine the best solution.

Port Forwarding

The following table outlines the port forwarding information in the router that maps public IP addresses to internal IP addresses. Port forwarding must be configured to utilize features such as remote phones, SIP Providers, remote administration and branch office. IPitomy port forwarding requirements are specified below.

Single Port Forwarding

Application Name	Port	Protocol	To IP Address
Remote Management	80	TCP	PBX Internal IP
SSH Support	22	TCP	PBX Internal IP
SIP	5060	UDP	PBX Internal IP
Branch Office	4569	UDP	PBX Internal IP

Port Range Forwarding

Application Name	Port Range	Protocol	To IP Address
RTP	10000-20000	UDP	PBX Internal IP

Figure 10 – Port Forwarding Configuration Table

!	**IMPORTANT: Please contact an IPitomy Technical Support Representative for assistance or additional information regarding how to configure port forwarding for the IP PBX system.**

IP Addresses

To determine the IP address that can be used on the network, check the router or get this information from the Network Administrator. In many cases, this default IP address will be acceptable without any router configuration other than port forwarding (required for remote administration and remote telephones).

IMPORTANT: The IP PBX is required to have a fixed (static) IP address. The system comes with a default static IP address of 192.168.1.249/ippbx.

We recommend that you use this default IP address. Please contact IPitomy's Technical Support Group if you need further assistance by email at support@ipitomy.com or phone 941-306-2200 option2, or visit our FAQ page at our website at www.ipitomy.com.

Changing the IP Address

If the default IP address does not match the network schema for your install, the PBX has a simple method for changing this information. The Network Console allows you to view and change settings without requiring an external web browser or any network connection.

STEPS:

1 Plug a keyboard and monitor into the IP PBX.

2 From this keyboard, press CTRL-ALT-F7

3 Follow the instructions displayed on the page to configure the IPs for the PBX. The options used are as follows.

- **1**: Network Type – Static or DHCP
- **2**: IP address
- **3**: Subnet Mask
- **4**: Default Gateway
- **5**: Static DNS
- **6**: Static DNS 2
- **7**: Static DNS 3
- **R**: Refresh View
- **D**: Default Settings
- **S**: Save & Update

4 Once **all values have been configured to match the install network, choose S to save and update**.

Note: *For further details on using the* ***Net Console****, please refer to* ***Appendix 6.***

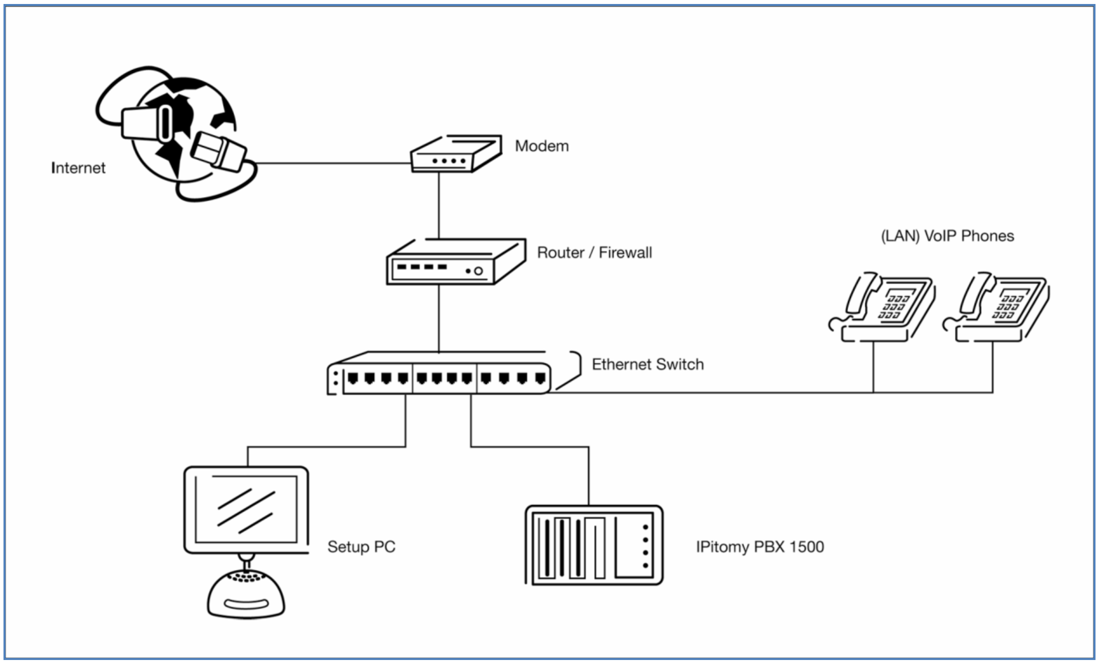


Figure 11 – Typical Network Configuration Diagram

Service Providers

In order to provision the IPitomy IP PBX it is necessary to know the type of Service Providers being used. Carrier and SIP are the most common service providers. Carriers provide plain old telephone lines (POTS), T1s and PRI lines. SIP Providers route voice calls over the internet. This is called voice over internet protocol or VoIP. As part of the installation it will be important to know the:

- Name of Providers
- Type of Service Provided (i.e., POTS, T1,or SIP)
- Phone Numbers Associated with the Service
- Password and Login Information for SIP Service

You should always consider any previous historical data that you have from your existing system. Any past reports or spreadsheets which relate to the components being designed, should be considered as possible input when configuring your IP PBX system.

We recommended that you record this information on the ***IPitomy Setup Worksheet.***

SYSTEM ADMINISTRATION

Administration Menu

IPitomy IP PBX's online administration makes it simple to meet the demands of a frequently changing business. It is also designed to be quick to setup and install. The **Administration Menu** is located in the **Navigation Bar** to the left of the page.

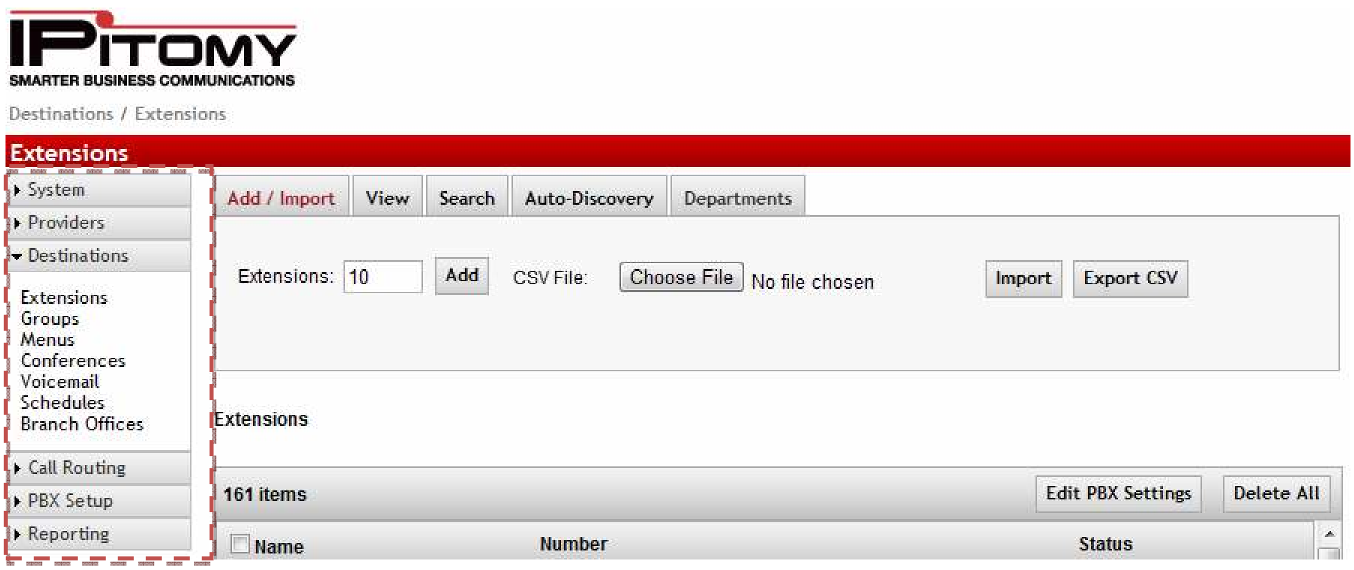
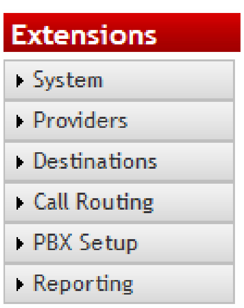


Figure 12 – Administration Menu Options

IP PBX Administration Options



This menu contains the administration pages used to configure the system. The Administration Menu is divided into six sections. To navigate to an **Administration Page** click on the menu section and page to be changed.

- **System** – System setup consists of network configuration settings.
- **Providers** – Providers are sources of PSTN and VoIP connectivity. Providers are the lines that handle all incoming and outgoing calls. All VoIP providers will be setup here. DID numbers are also entered here.
- **Destinations** – Destinations are extensions, groups of extensions, automated attendants (menus), conferences and voicemail. Destinations are places where calls get routed to in the system.
- **Call Routing** – Routing sends callers to specific inbound destinations within the system, and routing outbound callers over specific outbound routes like local, long distance, international and emergency.
- **PBX Setup** – System settings allow global configuration settings for system applications like PBX timers, voice messaging settings and music-on-hold.
- **Reporting** – The system displays usage reports, diagnostic information and monitors system activity.

Administration Page Layout

Each online Administration web page contains the standard layout. The following section describes the features or links you will have access to on each page.

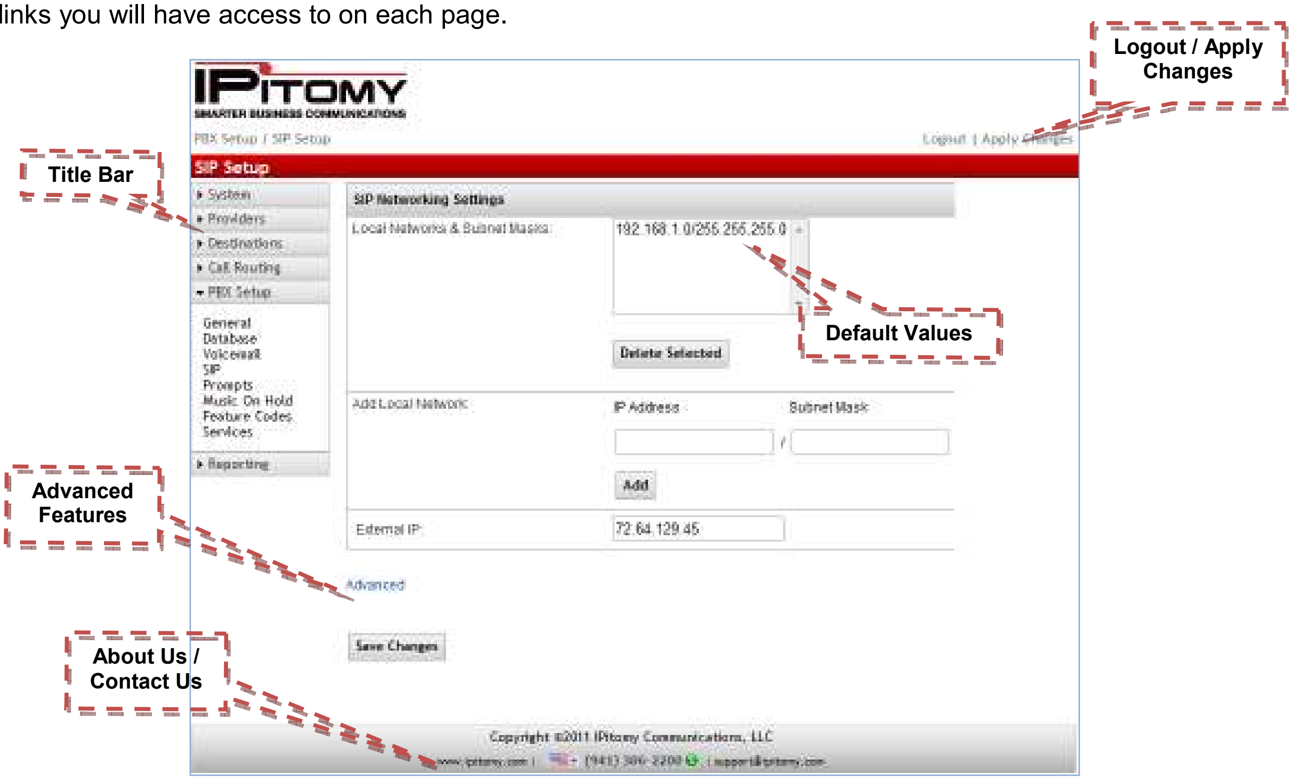


Figure 13 – Standard Page Layout and Features

Link	Description
Title Bar	The Title Bar at the top of each page displays the name of the section of the Administration Menu which is currently being edited.
Default Values	When the system is installed it automatically registers default values in many of the administration fields. This simplifies the implementation process.
Advanced Link	In several of the online administration features there is an **Advanced** link where the **most sophisticated capabilities of the IPitomy IP PBX** can be configured. If available (depending on applicable functions), the Advanced link will be located on the lower left side of each page.
About Us Link	The About Us link is located at the bottom left corner of each page This link provides additional information about IPitomy.
Contact Us Link	The Contact Us link is located at the bottom left corner of each page. This link will take you to the Contact Us page that provides you with important contact information for IPitomy Communications, LLC. The IPitomy team is never more than a call or email away. To **contact an IPitomy team member**, click on Contact Us in the lower left corner of the page.
Apply Changes	**To apply changes to the system you must click the Apply Changes button.** Located in the top right corner of the page, this link **globally applies changes** to the system which is different than the **Save Changes** button that only saves changes to the current page.
Logout	The **Logout** link located at the top right corner of the page allows you to logout of the Administration page and takes you back to the IP PBX Admin Login page**.**

Table 4 – Feature and Link Descriptions

Navigational Tools

The following section describes the navigation tools (buttons) that will allow you to perform various functions such as Add, Delete, Save or Update data within the Administration system. These buttons will appear where the functions are applicable to the type of data being created or modified.

Note: *Depending on the page that is being displayed, the buttons/icons may be referred to (labeled) differently. This list describes the more commonly seen and used buttons/icons.*

Button/Icons	Description
	This button or icon allows you to expand or open the function to view more options.
	This button or icon allows you to collapse or close the function.
	This button or icon allows you to view the items from a drop down list.
	This checkbox button or icon allows you to select or deselect the record for viewing or modifying, as well as to define if certain features are enabled.
	This radio button or icon allows you to select or deselect (toggle) between options.
	This pencil icon allows you to edit the current settings for a specific record (i.e. extensions or groups).
	This pencil with a phone icon allows you to edit an extension's current phone settings.
	This down arrow icon allows you to download a file off of the system.
	This "**X**" icon allows you to delete the item selected
	This curved arrow icon allows you to restore a file to the system or in some cases, clear out voice messages.
Add	The **Add** button allows you to create a new **destination, provider, route, schedule,** etc. For example, to add a new extension, click the **Add** button on the Extension Administration page. In some cases, the Add button will change (have a different label) depending on the function or page you are viewing or updating. For instance, the button may be displayed as Add Menu, if you are on the Menus page.
Save Changes	Located in the bottom left corner of the screen, the Save Changes button executes all the changes made to the current page. **This button must be pressed before leaving a page or changes will be lost**.

Table 5 – Navigational Buttons and Links

An easy way to get the information that is required for a specific parameter or field is to mouse over it. A callout box with a description of the requirements or recommended settings will appear.

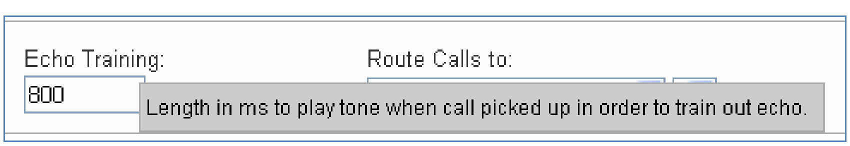


Figure 14 – View of Mouse Over Field Information

Login Page

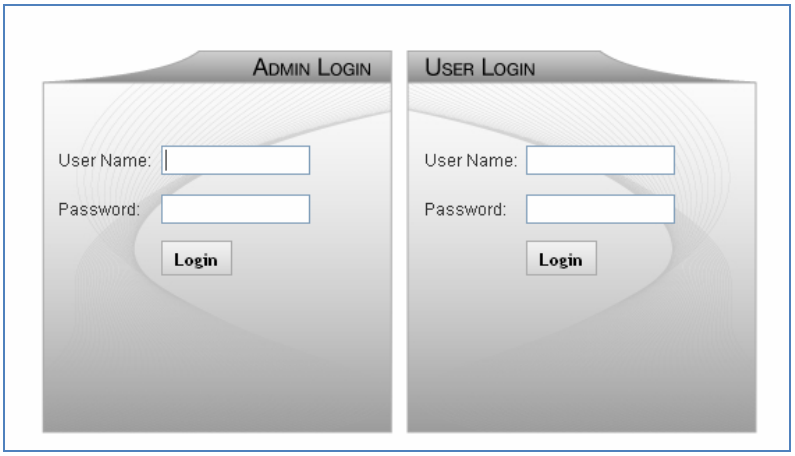


Figure 15 – IP PBX ADMIN Login Page

Logging In

STEPS:

1 From a web browser (ie. Firefox), enter the **IP Address** of the System Administration site.

2 Using the Admin Login (left side), enter the username and password of the PBX. The default login information is:

Username = pbxadmin
Password = ipitomy

IMPORTANT: Please contact IPitomy's Technical Support Group if you need further assistance by email at support@ipitomy.com or phone 941-306-2200 option 2. You can also visit our FAQ page at faq.ipitomy.com.

3 The **PBX Administration** screen appears. Now you can navigate around the PBX, configuring different features as needed.

Logging Out

STEPS:

1 You can logout of the **PBX Administration** page at anytime by clicking on the **Logout** link located on the top right corner of the screen.

2 The system logs out and returns you to the **PBX Administration Login** page.

SYSTEM NETWORKING

The IPitomy System Menu is for configuring network attributes. For example the IP address of the system and router information. The System Networking Setup Page allows you to define the Internet Setup for the system's hardware. The system must operate using a static IP address; DHCP should only be used on the IPitomy IP PBX if the router is configured to assign a specific static DHCP address to the system.

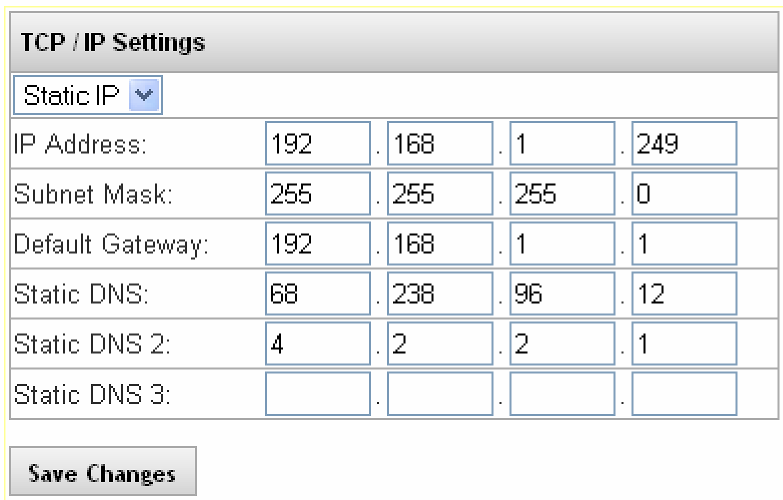


Figure 16 – Networking Setup Page

The following table describes the fields and recommended settings for Networking Setup for the IP PBX system:

Field	Recommended Settings
IP Address	**Use the default address (192.168.1.249)** of the IPitomy IP PBX or an address outside the range of existing IP addresses assigned by DHCP in the router.
Subnet Mask	**Leave the default setting for the Subnet Mask as (255.255.255.0)**. The subnet mask defines what traffic the PBX will listen and communicate to. A value of 255 means the octet in question needs to match exactly, while a value of 0 means the octet is not restricted at all. When the PBX is set to the default IP address, a subnet mask of 255.255.255.0 tells the system to communicate with any devices in the 192.168.1.xxx range.
Default Gateway	**The default gateway provided is 192.168.1.1**. Though this default is a common router IP, every network is different. Enter the IP address of the router handling their Internet connection here.
Static DNS	**Enter the DNS IP address being used on the network**. If a default DNS IP address is not provided by the router it can be obtained from the network's Internet Service Provider.
Static DNS2	**Enter the DNS IP address being used on the network**. If a default DNS IP address is not provided by the router it can be obtained from the network's Internet Service Provider.

Static DNS3	**Enter the DNS IP address being used on the network**. If a default DNS IP address is not provided by the router it can be obtained from the network's Internet Service Provider.

Table 6 – Network Setting Descriptions

TCP/IP Settings Section

Edit TCP/IP Default Settings

STEPS:

1 Navigate to **System ▸ Networking**. The **TCP/IP Settings** page appears displaying the default values for the following setting:

- **IP Address**
- **Subnet Mask**
- **Default Gateway**
- **Static DNS**

2 Click on the **IP Address** field. Enter the **IP address** for the Router. **Use the default address (192.168.1.249)** of the IPitomy IP PBX or an address outside the range of existing IP addresses assigned by DHCP in the router Enter the desired IP Address. *See Table above for recommended settings.*

3 Click on the **Subnet Mask** field. **Leave the default setting for the Subnet Mask as (255.255.255.0)**. *See Table above for recommended settings.*

4 Click on the **Default Gateway** field. Change the default Gateway value to the desired target network. *See Table above for recommended settings.*

5 Click on the **Static DNS** field. Change the default DNS value to the desired target network. *See Table above for recommended settings.*

6 Repeat **step 5** to set the remaining DNS values, if necessary.

7 Click on the Save Changes button

8 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Access Control (PBX Access)

The Access Control page is comprised of 3 sub-pages; Host Access, Web Server, and Access Control List. Each is accessible from the buttons at the top of the page and pertains a different method of controlling access to the PBX.

Host Access

This feature allows you to limit access to special services on the PBX. An "**allow from**" entry is a list of one or more host names, host addresses, patterns or wildcards that will be matched against the client host name or address. List elements should be separated by blanks and/or commas.

Note: *The parameter for the IP PBX Host Access is pre-configured per the manufacturer's specifications. We recommend that you* ***do not change*** *this configuration value.*

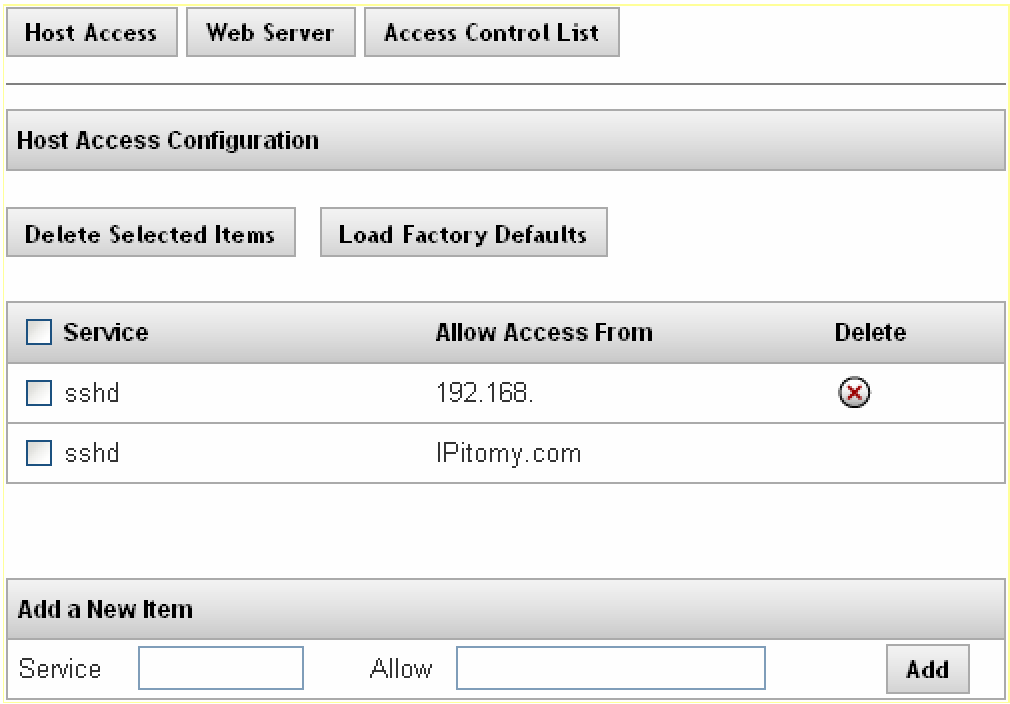


Figure 17 – PBX Host Access Page

The following table describes the features and functions available on the Host Access page:

Fields/Buttons	Description
Delete Selected Items	This button allows you to delete multiple services at a time.
Load Factory Defaults	This button will set the PBX back to the default Host Access settings.
Add a New Item	This section is where you would add new rules for accessing special services on the PBX

Table 7 – Network Features and Descriptions

IMPORTANT: Changes to the Host Access List are installed immediately. They are database independent so custom changes do not migrate from one box to another via a database backup file.

Please contact IPitomy's Technical Support Group if you think you need to modify these settings. Email via support@ipitomy.com or phone at 941-306-2200 option 2. You can also visit our FAQ page at faq.ipitomy.com.

Web Server Configuration

This feature allows you to define which IPs and/or domains can access the web server, as well as restart the Web Server. In order for changes to this list to take effect, you must Restart the Web Server.

*The parameter for the IP PBX Web Server is pre-configured per the manufacturer's specifications. We recommend that you **do not change** this configuration.*

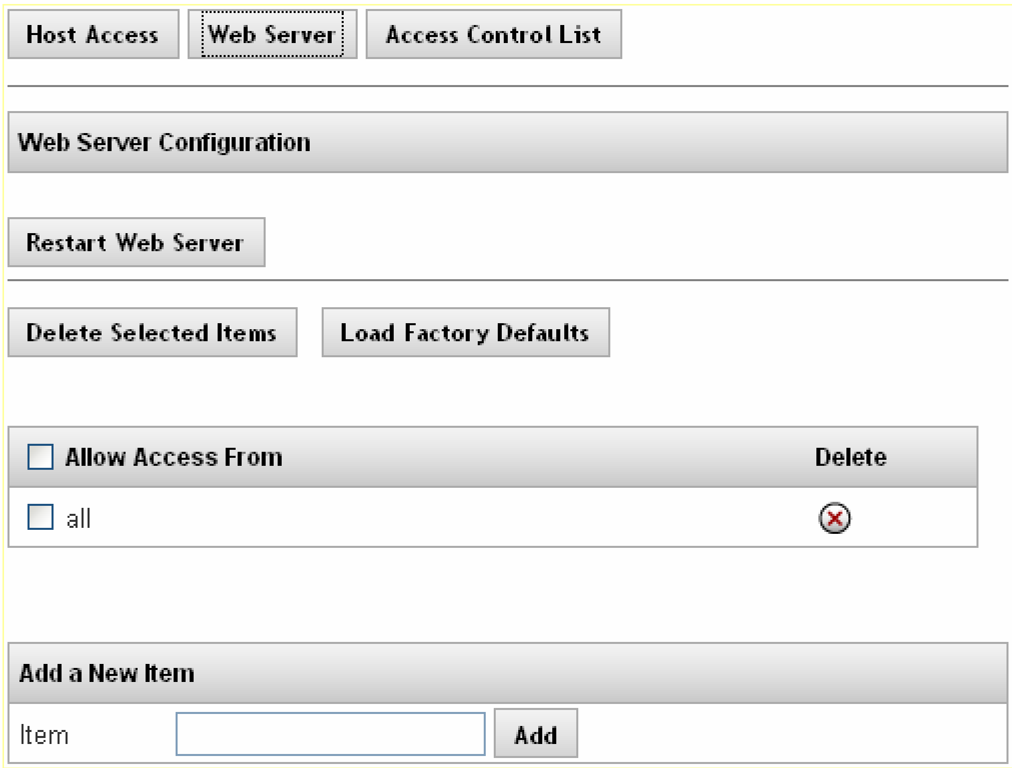


Figure 18 – Web Server Configuration Page

Feature	Description
Restart Web Server	This feature allows you to restart the web server so that changes made that impact server components can be applied. **Note: Restarting the server will not interrupt phone service. A reboot of the PBX system will also apply changes made to other attributes.**
Allow Access From	Defines the networks and/or domains that are allowed to access the PBX. The "Allow" format may be: • **Domain name** • **Full IP address** • **Partial IP address** • **Network / netmask pair** • **Network / CIDR specification**
Delete Selected Items	This button allows you to delete multiple services at a time.
Load Factory Defaults	This button will set the PBX back to the default Web Server settings.
Add a New Item	This section is where you would add new rules for accessing the Web Server

Table 8 – Web Server Features and Descriptions

!	**IMPORTANT: Changes to the Web Server Access List are preconfigured. They are database independent so custom changes do not migrate from one box to another via a database backup file.** **Please contact IPitomy's Technical Support Group if you think you need to modify these settings. Email via support@ipitomy.com or phone at 941-306-2200 option 2. You can also visit our FAQ page at faq.ipitomy.com.**

Add New Permission

STEPS:

1 Navigate to **System→ Access Control**

2 Click on the **Web Server** button located at the top of the page. The **Web Server Configuration** page appears.

3 In the **Add a New Item** section, enter the network or domain you want to allow to access the PBX web server

4 Click the **ADD** button.

5 The new permission rule will be displayed under Allow Access From window above

6 Click on the **Restart Web Server** button.

7 Click OK when prompted to confirm you wish to restart the Web Server

8 A "*Please Standby*" message appears. Once the reboot process is completed you will be returned to the **Web Server Configuration** page.

Load Factory Default

This feature allows you to restore the manufacturer's factory default settings. It will restore the settings to the factory recommended defaults.

STEPS:

1 From the **Web Server** page, click **Load Factory Default** button. This will return or restores the Allow Access From list back to the manufacturer's default setting.

2 Once the changes applied, you will need to restart the web server. Please refer to the *Restart Web Server* topic of this user guide for steps on how to restart the server.

3 Click on the **Restart Web Server** button.

4 Click OK when prompted to confirm you wish to restart the Web Server

5 A "*Please Standby*" message appears. Once the reboot process is completed you will be returned to the **Web Server Configuration** page.

Access Control List

The Access Control List defines what networks different PBX features are permitted to communicate with.

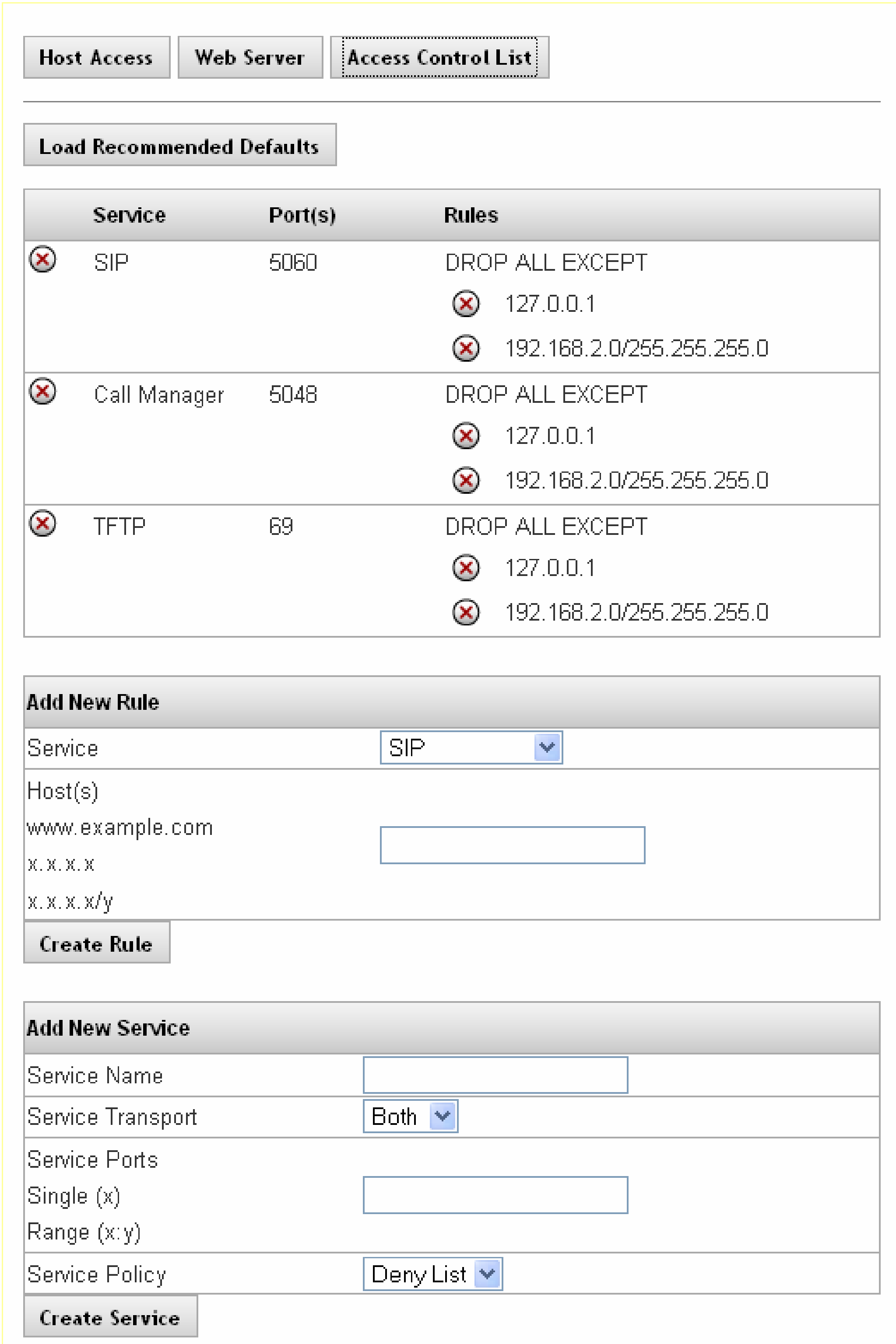


Figure 19 – Access Control List - Add New Rule Section

The following table outlines the parameters and descriptions for the Access Control List.

Feature	Description
Service	Displays the name of configured services. Typical services on the PBX are: SIP - Used for Calls Call Manager - Used for Desktop Call Manager TFTP – Used by phones to pull down config and firmware files
Ports	Displays the ports that were defined for a particular service. SIP – 5060 Call Manager – 5048 TFTP - 69
Rules	Displays the rules that were configured for a particular service. Deny List – Accepts all traffic, unless specifically defined Allow List – Denies all traffic, unless specifically defined

Table 9 – Access Control List Definitions

Load Recommended Default

This is the recommended method to set the Access Control List to the typically used settings.

STEPS:

1. Navigate to PBX Setup➔SIP
2. Set the LocalNet to match the network the PBX is installed on, Save, and Apply Changes
3. Navigate to the **Access Control List** page, click **Load Recommended Default** button. This will create default rules allowing the PBX to communicate to devices on the LocalNet in regards to SIP, Call Manager, and TFTP
4. Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Add New Service

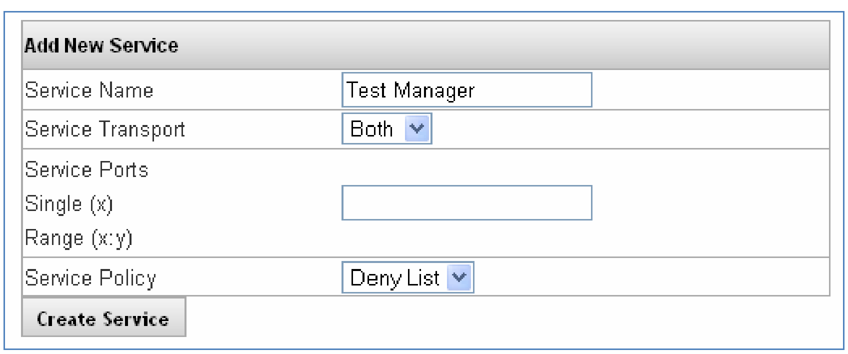


Figure 20 – Access Control List Page - Add New Service Section

The following table outlines the parameters and descriptions required for adding a new service.

Feature	Description
Service Name	This is the name of the new service and will populate the Service drop-down list in the Add New Rule section.
Service Transport	This is the service type that will be used to transport the message. The options are Both, TCP or UDP. SIP and RTP traffic both occur on UDP, TFTP traffic is UDP, and Call Manager traffic is TCP. Any other rules created would need to be configured for the protocol used by this service.
Service Ports	This is the port information that is associated with the host. You can enter a single or range of ports that will be used for this service. SIP uses 5060, Call Manager uses 5048, and TFTP uses 69. Other services must be configured to use the appropriate ports.
Service Policy	This is the umbrella rule for the service, which will be further defined under Add New Rules. The options are: **Deny List; ACCEPT ALL EXCEPT** rule will apply. This will allow all traffic on the defined port, allowing you to configure a list of Denied IP addresses. **Allow List: DROP ALL EXCEPT** rule will apply. This will block all traffic on the defined port, allowing you to configure a list of Allowed IP addresses.

Table 10 – Add New Service Settings and Descriptions

The following outlines the steps to add a new service in the PBX system.

STEPS:

1. Navigate to **System➔Access Control**
2. Click on the Access Control List button, The Access Control List page appears.
3. From the **Add New Service** section, enter a Name, and select the appropriate Transport Protocol, Ports, and Policy; then click the **Create Service** button.
4. The new service and its associated values will be displayed in the Service listing.
5. Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

The following table outlines the parameters and descriptions required for adding a new rule.

Feature	Description
Service	This drop-down list is populated when a new services is added. This is done in the Add New Service section.
Host(s)	This is the IP Address, Domain Name or URL of the host.

Table 11 – Add New Rule Settings and Descriptions

Add New Rule

The following outlines the steps to add a new rule for Services in the PBX system.

STEPS:

1 Navigate to **System→Access Control,** click on the Access Control List button, the Access Control List appears.

2 From the **Add New Rule** section, select the Service type from the drop-down list.

3 Enter the **Host/s** to be allowed/denied by the service

4 Click the **Create Rule** button.

5 The new rule is added and will be displayed in the rules list.

6 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database. .

Delete Rules or Services

The following outlines the steps to delete existing rules or services.

STEPS:

1 From the **Service** section of the **PBX Access – Access Control List** page, find the service or rule that you want to delete.

2 Click on ⊗ icon to the left of either the service or rule. The selected item is removed from the list.

3 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

PROVIDERS

Providers are telephone lines, VoIP providers and other telecommunication resources. This section of the system's online administration is where these provider resources are configured. The PBX system is equipped to handle two types of provider setting's Hardware Trunks and SIP Providers.

Hardware Trunks

Hardware trunks are associated with telephone lines that connect to the PSTN. These lines process inbound and outbound communication traffic that flows over communication channels. For example, a T1 can be a trunk resource that has multiple lines and multiple Direct Inward Dialed (DID) numbers. These individual numbers can be routed to different destinations within the system.

Connection Types

The IP PBX is equipped to support an assortment of hardware cards as discussed earlier. Depending on which card you have, the PBX can integrate with analog lines, T1, or PRI.

Provisioning a New Hardware Trunk Group

Provisioning a hardware trunk defines how the system works with the provider equipment, tells the system what phone numbers are associated with the trunk, and establishes rules for the system to follow when processing incoming and outgoing calls through this physical network connection.

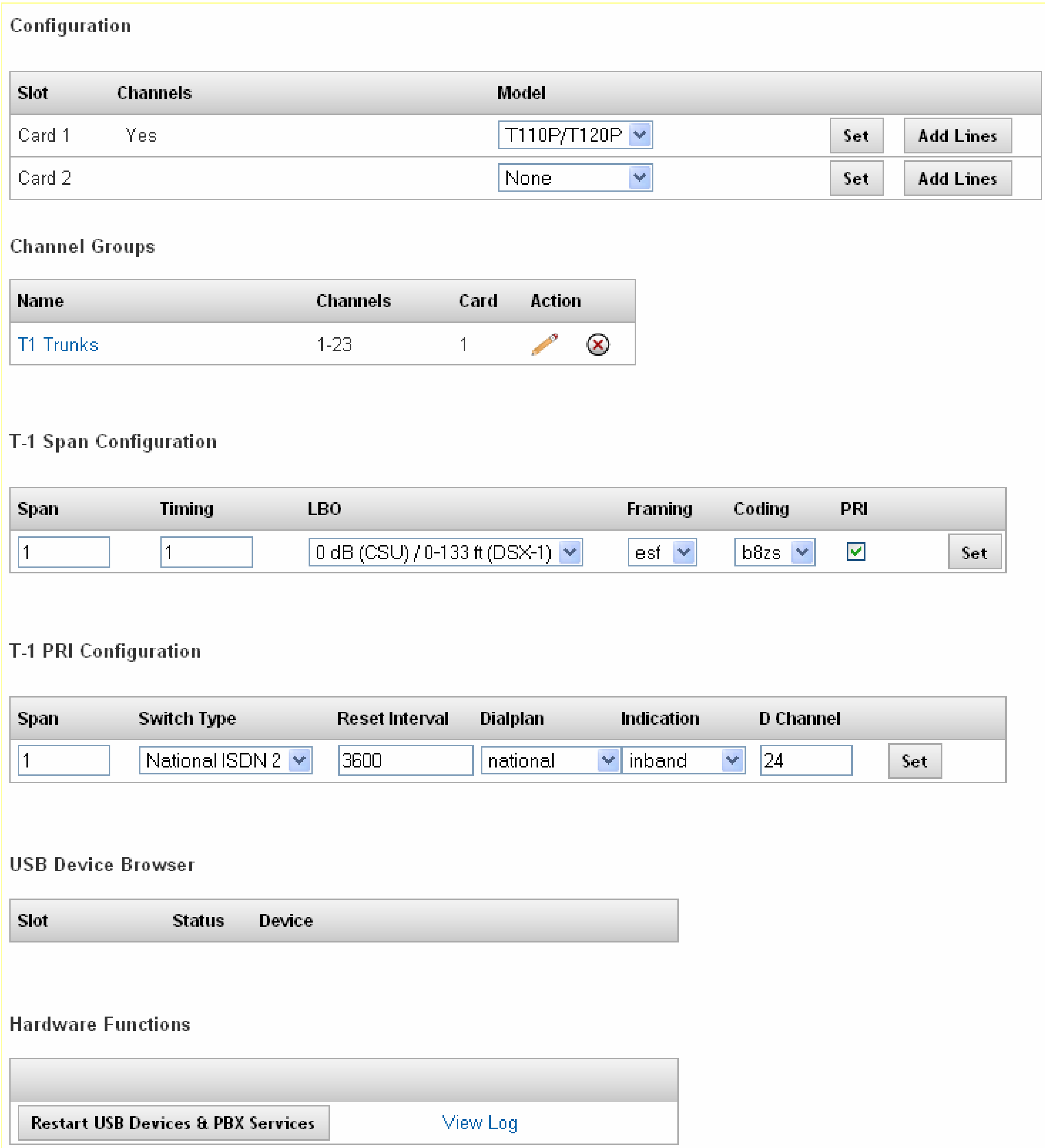


Figure 21 – Hardware Providers Trunk Setup Page

The following table describes the fields and functions available on the Hardware Trunks Provider Configuration page.

Section	Description
	Configuration Section
Slot	Defines which physical PCI slot the card is connected to.
Channels	This indicates whether the card has channels configured. Using the mouse to hover over this field will provide you with the number of channels at are set for this card.
Model	This is the model or type of card.
Set	This button changes the card set in the database for the adjacent slot. The change is immediate and will delete trunks associated with another card in the same slot.
Add Lines	Selecting this button will take you to the Channel Group Parameters Card Configuration page, where you can create a channel group.
	Channel Groups Section
Name	The name associated with this channel group.
Channels	The number of channels that have been assigned.
Card	The card number associated with the channel group.
Action	- Clicking on the pencil icon allows you edit the settings for the selected channel group. - Clicking on the "**X**" icon allows you to delete/remove the selected channel group. **NOTE: You must delete channels from outbound routes in order to delete them.**
	T-1 Span Configuration
Span	Should automatically set to the appropriate span in regards to the cards that are installed in the PBX.
Timing	Set per your providers instructions, "**0**" (zero) would be the PBX provides timing, "**1**" would be the timing is provided by the Provider.
LBO	Line Build Out, set per your providers instructions.
Framing	Set per your provider's instructions.
Coding	Set per your provider's instructions.
PRI	Check if the trunk is a PRI, leave blank if the trunk is a standard T-1.
	T-1 PRI Configuration
Span	Should automatically set to the appropriate span in regards to the cards that are installed in the PBX.
Switch Type	Defines what type of equipment the PBX is communicating with. Set per your provider's instructions.
Reset Interval	Set per your provider's instructions.
Dialplan	Set per your provider's instructions.

Indication	Set per your provider's instructions.
D Channel	Set to 24.
USB Device Browser	
Slot	Should automatically set to the appropriate slot that corresponds to the USB slot the device is connected to.
Hardware Functions	
Restart USB Devices & PBX Services	This button will restart USB devices and PBX Services.

Table 12 – Hardware Trunk Provider Settings and Descriptions

The following section outlines how to setup the parameters for hardware trunks.

Configuring Hardware Trunks

STEPS:

1. Navigate to **Providers➔Hardware Trunks**. The **Hardware Trunk (Providers)** page appears. The hardware trunks and their values that have already been provisioned will be displayed.
2. The system will allow configuring of up to 2 cards that interface with analog or T-1 devices. Select the card models installed in your IP PBX individually and click **Set** button for each card in order to tell the system which cards you have.
3. Once you have set the card, click the **Add Lines** button to the right of the corresponding card to define line/channel/trunk groups associated with that card. The **Edit Hardware Provider** page appears.

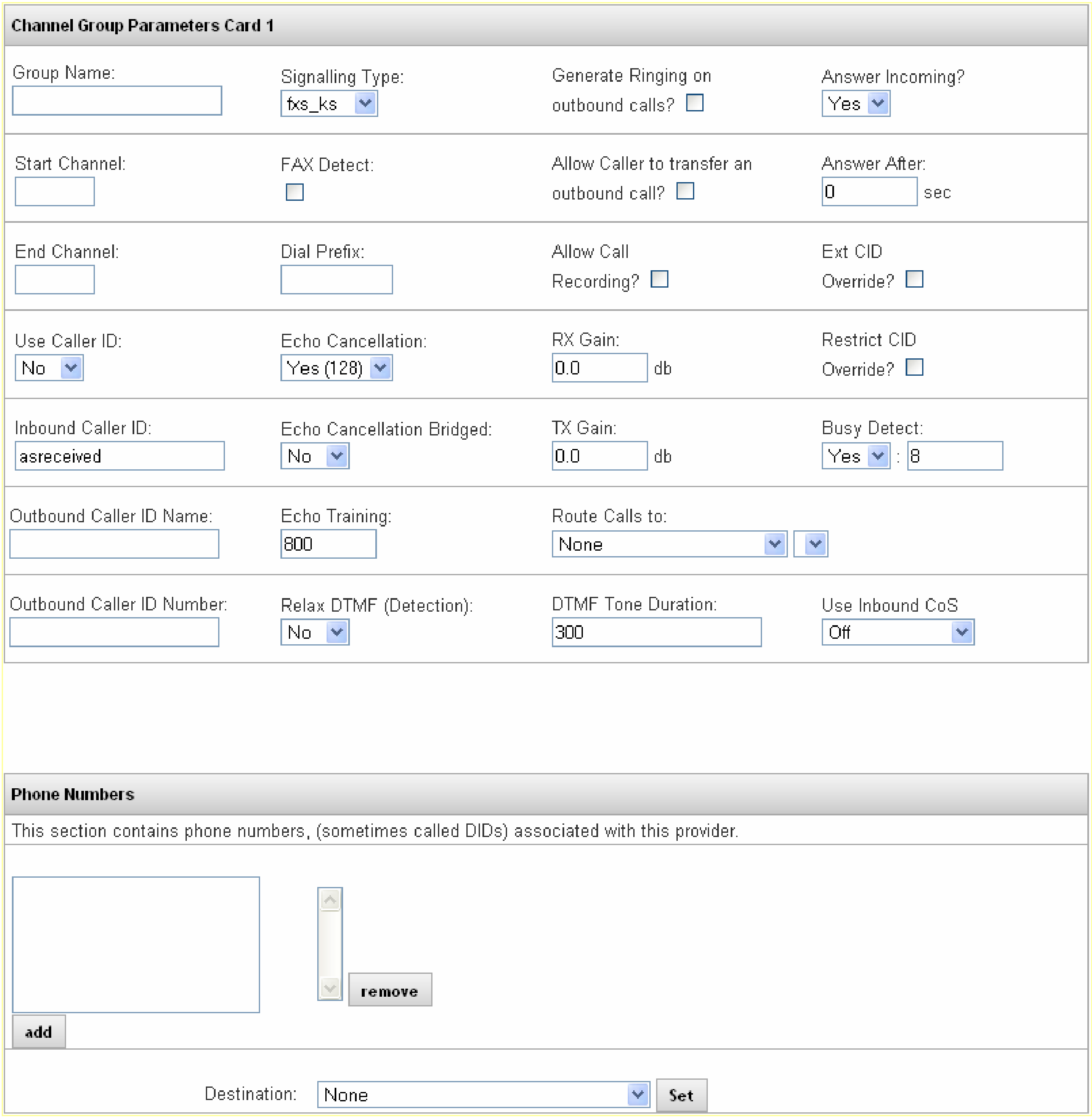


Figure 22 – Add/Edit Hardware Provider Configuration Page

4 Edit the necessary parameters. Required fields are: Group Name, Signalling Type, Start Channel, and End Channel. The rest of the fields can be configured as needed, or left at their defaults. (refer to the table below for descriptions and recommended settings)

5 Click on the Save Changes button.

6 Click the **Apply Changes** link located on the right hand corner of the page to commit the changes to the database.

7 Once the changes are applied, navigate to the **Hardware Providers** page and click the **Restart USB and PBX Services** button.

The following table describes the fields and parameters available on the **Add/Edit Hardware Provider Configuration** page:

Section/Field	Description
Channel Group Parameters Card 1	
Group Name	This is the name for the groups of channels you want to create. This Group Name will be associated with the line/channel group.
Signaling Type	This is the type of signaling used for these channels/lines. Recommendation and default settings are listed below: • **Analog** lines should be set to **fxs_ks**. • **T-1** lines should be set to **fxs_ls** or **em_w** • **T-1 PRI** lines should be set to **pri_cpe.**
Generate Ringing on Outbound Calls?	This should only be checked if the telephone company is not providing ringing. Default this is **disabled**.
Answer Incoming?	This indicates whether the system should answer incoming calls on these lines/channels. The default value is **Yes**.
Start & End Channel	Used to define the Start and End channel for this group on the associated card. If you would like a 1 channel group set the start and end channel to the same value (i.e. **Start=1 and End=1**). **IMPORTANT: If you wish to address analog lines individually you must build multiple single channel groups. T1 and PRI can be grouped with multiple channels and routed via DID.**
Allow Caller to transfer outbound call?	This indicates whether the system will allow user to transfer a call they orignginated that has been connected to the PSTN. Default this is **disabled**.
Answer After	If the system is configured to answer incoming calls, this defines how many seconds to wait before answering the call. At times this field needs to be tweaked to ensure consistent CID. Default this is set to **0**.
Dial Prefix	Digits defined here will be dialed out the trunk ahead of the digits dialed by the user. This would be required if your provider needs a 9 or some other digit to dial outbound. Analog trunks might need a one second pause which is denoted by a lowercase w. Default this is **blank**.
Allow Call Recording?	This setting toggles the ability to record calls on this trunk. Default this is **disabled**.
Ext CID Override?	This setting allows the users to set an alternate caller ID under their extension that will override the outgoing caller ID. If this is **enabled**, then this allows extension caller id override capabilities. Default this is **disabled**. **IMPORTANT: Contact your provider to determine whether CID override is allowed on your specific trunk**
Restrict CID Override?	This setting is used to limit an extension's ability to override Caller ID (CID). If **enabled**, this allows extension CID Override only for the phone numbers defined for this provider. Default this is **disabled**.
Use Caller ID	This setting indicates whether the system should attempt to detect caller id on the lines/channels in this group. Default this is set to **Yes**.

Echo Cancellation	This allows you to configure the Echo Cancel parameter. By default this is set to **Yes (128)** which will work in most scenarios. Adjustments can be made as needed.
RX Gain	This allows you to set or adjust the gain on sound received from these lines/channels. Typically these settings don't need to be changed from default settings, unless there are issues relating to the volume on the calls. (This setting is in decibels so 3.0 = 100% increases in volume). Default this is set to **0.0**.
TX Gain	This allows you to set or adjust the gain on sound transmitted to these lines/channels. Typically these settings don't need to be changed from default settings, unless there are issues with the volume on the calls. (This setting is in decibles so 3.0 = 100% increases in volume). Default this is set to **0.0**.
Inbound Caller ID	This is the name of the Caller ID of the inbound calls on this trunk. If set to the system default "**asreceived**', the calls will capture the Caller ID information sent over the incoming lines. Otherwise you can override the name and number that appears to those receiving calls on the corresponding channels. **IMPORTANT: If you need to change this parameter, we recommend that you contact IPitomy's Technical Support Group at (941) 306-2200 for the proper syntax or additional information.**
Echo Cancellation Bridged	This allows you to **enable or disable** the Echo Cancellation Bridge parameter on calls routed internally across the network. Set this parameter to"**YES**" (enable) if you want echo cancellation on calls that are bridged from one port to another on the network. Default this is set to **No**.
Busy Detect	By default, this is set to **YES** and **8**. This allows you to enable/disable and set the number of busy signals the system will detect before disconnecting from the call (hanging up). Set the parameter to "**YES**" if you want the PBX to hang up after it detects a busy signal. The number field is the number of busy tones the system must detect before it disconnects from the call (hangs up).
Outbound Caller ID Name	Any values entered here will display as the Outbound CID Name when making calls out this trunk. If this is left blank, the name displayed on outbound calls will match what is registered with the provider. **Blank** by default. **IMPORTANT: Contact your provider to determine whether CID override is allowed on your specific trunk**
Outbound Caller ID Number	Any numbers entered here will display as the Outbound CID number when making calls out this trunk. If this is left blank, the number displayed on outbound calls will match what is registered with the provider. **Blank** by default. **IMPORTANT: Contact your provider to determine whether CID override is allowed on your specific trunk**
Echo Training	This parameter is used to adjust the time spent training the echo cancellation algorithm before a call is picked up. The length of time is set in milliseconds (ms), and is set to **800** by default.
Route Calls to	This is the default destination for the trunk. A destination is where calls from this provider will be routed to, unless a specific Direct Inward Dial (DID) is indicated. A typical destination is defined as a Menu, Extension, Schedule, Voicemail, or a Ring Group. This can also be set under the Incoming Routing page.
Relax DTMF	This allows you to turn **On/Off** the Relax DTMF detection. Enabling this setting

(Detection)	allows the line/channels to be more permissive of tone lengths and result in over detecting. By default this is set to **No**. **IMPORTANT: We recommend that this parameter is set to NO unless you are having issues. Please contact IPitomy's Technical Support Group for assistance or more information.**
DTMF Tone Duration	This allows you to set the DTMF tone duration in milliseconds (ms) for generated tones generated by calls bridged to SIP devices. Default value of **300**.
Use Inbound CoS	This allows you to enable/disable inbound routing and provides the dial plan that corresponds to the outbound routes in this Class of Service (CoS). This is **Off** by default. **IMPORTANT: Use this only if you intend to provide PSTN connectivity to 3rd party equipment. Please contact IPitomy's Technical Support Group for assistance or more information.**
Phone Numbers	On a T1 or PRI, this is where you would configure any DID information. Only numbers or a + should be entered for a DID. Once configured, you can define a different inbound destination for each number, allowing you to route calls to separate places. Contact your provider to find out if they are sending 4, 7, or 10 digits inbound for the DID.

Table 13 – Hardware Provider Configuration Settings and Descriptions

Add Phone Numbers to Hardware Provider

Once all the destinations are created, they will appear in a drop-down list and can be selected for updating.

STEPS:

If a DID destination is set to NONE, it will route to the default location for the trunk itself.

1 From **Channel Groups** section of the **Hardware Provider** page, click the icon to the right of the name you want to modify. The **Edit Hardware Providers** page appears.

2 The hardware trunks and their values that have already been provisioned will be displayed. Scroll down the page to the **Phone Numbers** section. Enter the phone number that you want to add then click the **ADD** button. You can also copy and paste a list in. The list should be separated by newline characters.

3 The phone number entered will appear in the drop-down list to the right of the **ADD** button. The numbers will be listed in sequential order.

4 Click on the Save Changes button to save the changes to the system.

5 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Remove Phone Numbers from Hardware Provider

STEPS:

1 From **Channel Groups** section of the **Hardware Provider** page, click the icon to the right of the name you want to modify. The **Edit Hardware Providers** page appears.

2 Scroll down the page and click on the field to the left of the **REMOVE** button. Scroll through the drop-down list to find the phone number that you want to remove.

3 Click on the desired phone number. Click on the **REMOVE** button to the right of the list. The phone number is deleted and will not appear in the drop-down list.

Use the CTRL or SHIFT button to select multiple or a range of extensions from the list.

4 Click on the Save Changes button to save the changes to the system.

5 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Provisioning SIP Providers

SIP Providers are VoIP service provider accounts or other SIP-based devices that provide PSTN connectivity. SIP provider accounts can have multiple phone numbers or Direct Inward Dialing numbers (DIDs). The individual numbers can be routed to different destinations within the system as DIDs.

IMPORTANT: SIP Providers settings vary widely. To simplify the provisioning process, we recommend that you ask the provider for a sample Asterisk configuration. Please contact IPitomy's Technical Support Group for assistance.

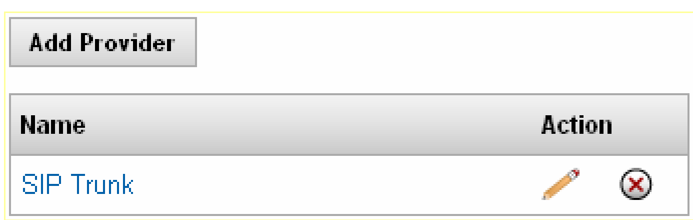


Figure 23 – SIP Providers Configuration Main Page

SIP Provider	
Name:	
User Type:	friend
DTMF Mode:	auto
RFC2833 Compensate:	No
Host:	
Port:	Default Custom
Register:	Yes No Custom
Authentication:	Yes No Custom
Auth User:	Default Custom
From User:	Default Custom
From Domain:	Default Custom
Realm:	Default Custom
Outbound Proxy:	Disabled Enabled
Username:	
Secret:	
Inbound Caller ID:	
Outbound Caller ID Name:	
Outbound Caller ID Number:	
Call Limit:	1
Qualify:	30000
Default Destination:	None
Dial Prefix:	
Area Code:	
Generate Ringing on outbound calls:	☐
Allow Outbound Caller to transfer:	☐
Allow Call Recording:	☐

Figure 24 – Add/Edit SIP Provider Configuration Page 1

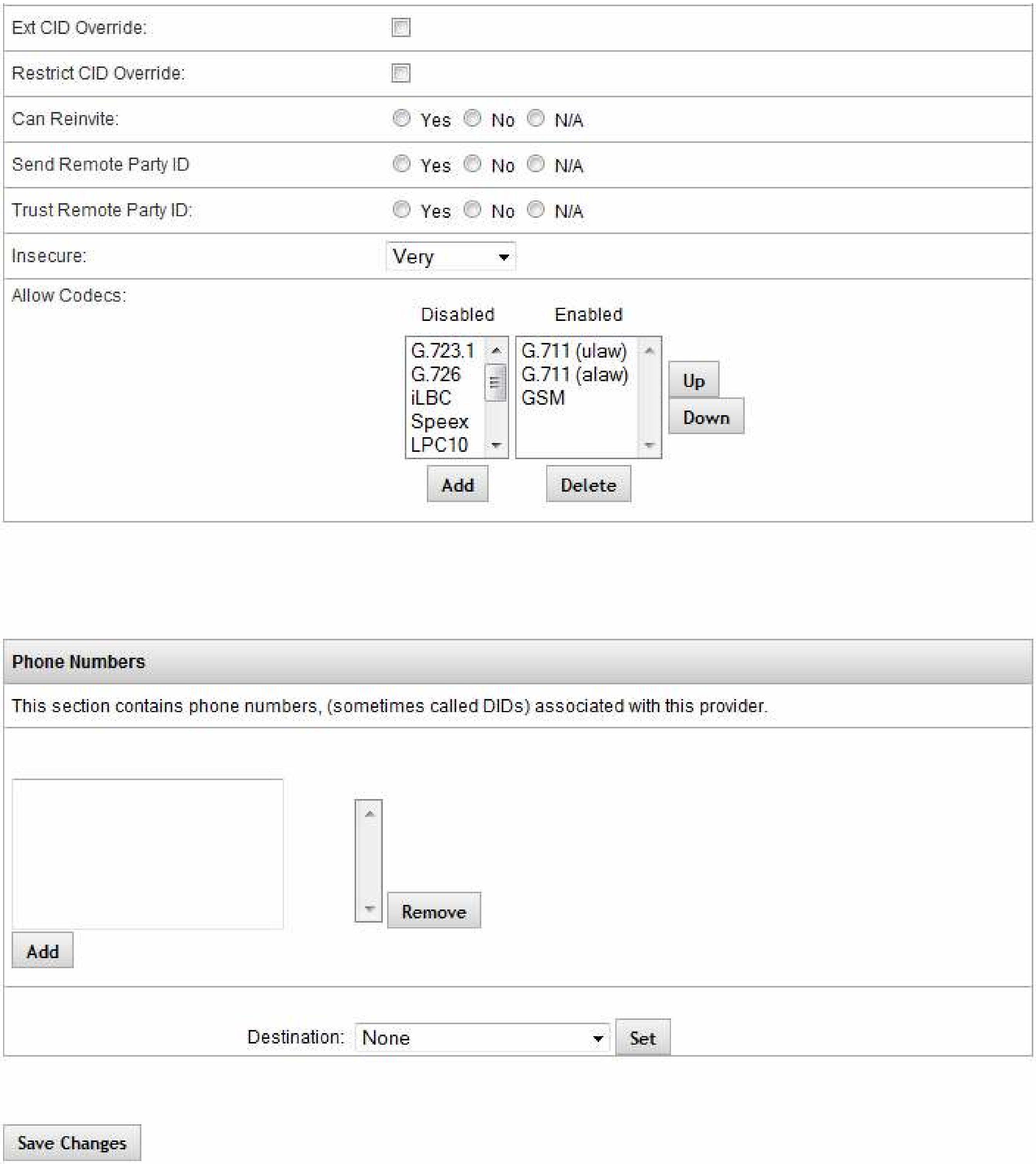


Figure 25 – Add/Edit SIP Provider Configuration Page 2

The following table describes the fields and functions available on the SIP Provider Configuration page:

Sections/Fields	Description
	SIP Provider Section
Name	Name assigned to the SIP Provider. If given a username, this needs to match that field.
User Type	User type of the associated SIP Provider. This can be User, Friend, or Peer.
DTMF Mode	DTMF tone for the trunk. This can be Inband, rfc2833, info, or auto.
RFC2833 Compensate	This advanced feature is sometimes necessary when using RFC2833 is problematic or if the SIP provider has indicated that it is required.
Host	The domain or IP address associated with the SIP host.
Port	If your SIP provider requires registration on a port other than the default 5060, set this field to Custom and enter that port value; otherwise set to Default. **NOTE: Be sure to forward this alternate port in your router or the traffic will not be passed through.**
Register	Registration option for the SIP Provider. • **YES** = Automatically generates based upon provided settings • **NO** = Doesn't require authentication from the SIP Provider • **CUSTOM** =Allows for any special authentication rules required from the SIP Provider
Authentication	Authentication (previously "Authorization") requirements for the SIP Provider. • **YES** = Automatically generates based upon provided settings • **NO** = Doesn't require authentication from the SIP Provider • **CUSTOM** =Allows for any special authentication rules required from the SIP Provider
Authorized User	Some providers require a different Authorization User Name than the Username provided. The sample asterisk configuration from your provider will let you know if this field needs to be set to **Custom**; otherwise set to **Default.**
From User	The sample asterisk configuration from your provider will let you know if this field needs to be set to **Custom**; otherwise set as **Default.**
From Domain	This parameter is rarely used and if needed, the SIP Domain information can be obtained from your SIP Provider
Realm	If your provider requires you to set the Realm to something other than asterisk, set to **Custom** and enter the realm given; otherwise this can be set to **Default**.
Outbound Proxy	**Enable** if the SIP Provider requires you to provide a different outbound IP address, otherwise leave as **Default**.
Username	Username used for Registration and Authentication.
Secret	The password used for SIP Registration and Authentication.

Inbound Caller ID	This parameter should be left blank to display incoming caller ID as received. Otherwise you can override the name and number that appears to those receiving calls on the corresponding channels. **IMPORTANT: If you need to configure the system in a different way, please contact IPitomy's Technical Support Group for assistance and the proper syntax needed.** **Contact us via phone at 941-306-2200 option 2 or via email at support@ipitomy.com. Additional information can be found at faq.ipitomy.com.**
Outbound Caller ID Name	Enter an outbound Caller ID name that will override the Caller ID name that is displayed on outgoing calls through this provider. **IMPORTANT: Contact your provider to determine whether CID override is allowed on your specific trunk**
Outbound Caller ID Number	Enter the Caller ID number that will override on outgoing calls through this provider.. **IMPORTANT: Contact your provider to determine whether CID override is allowed on your specific trunk**
Call Limit	Set to match the number of concurrent calls allowed by the SIP trunk. (Usually consistent with SIP provider subscription limit.)
Qualify	This is the number of milliseconds (ms) the system should wait before checking to see if the SIP provider is available. Check with your provider to see what value this should be set to. Typically this can be left at the default value of 30000.
Default Destination	This is where all inbound calls will be routed to, unless a different specific destination is designated for DIDs on an individual basis.
Dial Prefix	Set this parameter only if the SIP Provider requires it.
Area Code	This parameter is obsolete and will be removed in future releases.
Generate Ringing On Outbound Calls	This parameter should only be enabled (checked) if ringing is not provided by the SIP Provider.
Allow Outbound Caller to transfer	Enabling this parameter allows a user to transfer a call that they originated on this trunk. Default is **disabled.**
Allow Call Recording	If enabled (checked), this parameter allows calls to be recorded.
Ext CID Override	If enabled (checked), this parameter allow the users to set an alternate caller ID under their extension that will override the outgoing Caller ID. Default is **disabled.** **IMPORTANT: Contact your provider to determine whether CID override is allowed on your specific trunk**
Restrict CID Override?	This setting is used to limit an extension's ability to override Caller ID (CID). If **enabled**, this allows extension CID Override only for the phone numbers defined for this provider. Default this is **disabled**.

Can Reinvite	This parameter allows a mechanism to reconnecting calls midstream. • **YES** = if the phone type allows the re-invite feature • **NO** = if the phone type does not allow the re-invite feature • **N/A** = accepts the system wide default defined under PBX Setup➔**SIP**➔**Advanced**. If the default setting is acceptable and works within your business, we recommend leaving the parameter set to N/A.
Send Remote Party ID	Usually not required. Leave on "N/A" unless advised differently by tech support.
Trust Remote Party ID	Usually not required. Leave on "N/A" unless advised differently by tech support.
Insecure	This parameter allows you to specify how to handle connections with peers. Explanation of the different options available on the drop-down list are: • **PORT** = Ignore the port number where authentication came from. • **INVITE** = Do not require the initial invite to authenticate. • **PORT+INVITE** = Do not require initial invite to authenticate and ignore the port where the request came from. • **YES** – To match a peer based by IP Address only and not the port. • **VERY** – To allow registered hosts to call without re-authenticating.
Allow Codecs	Encodes a stream or signal for transmission. Select which codecs will be enabled for this provider. Can typically be left to defaults.
Phone Numbers Section	
Phone Numbers	This is where you would configure any DID information. Only numbers or a + should be entered for a DID. Once configured, you can define a different inbound destination for each number, allowing you to route calls to separate places. Contact your provider to find out if they are sending 4, 7, or 10 digits inbound for the DID.

Table 14 – SIP Provider Configuration Settings and Descriptions

Add New SIP Provider

The following section outlines the steps to add a new SIP Provider in the PBX system.

STEPS:

1 Click **Providers**➔**SIP Providers**. The **SIP Providers** page appears.

2 Click on the Add Provider button. The **Edit SIP Provider** page is displayed.

3 Configure the pertinent fields per your SIP Providers instructions. See the table above for descriptions.

4 Click on the Save Changes button.

5 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Add Phone Numbers

The following section outlines the steps to Add phone numbers to the SIP Provider.

STEPS:

1. Click **Providers➔SIP Providers**. The **SIP Providers** page appears.

2. From the SIP Providers page, select the SIP provider link (under the **Name** field) or the  icon to the left of the provider you want to modify. The **Edit SIP Providers** page appears.

3. Scroll down to the **Phone Numbers** section of the page. Enter the phone number in the box above the ADD button then click **ADD**. The number enter will appear in the list to the right of the box.

4. To add multiple phone numbers at one time, press the **ENTER** key to move to the next space on the list then enter the number. Click on the **ADD** button and all the numbers will appear in the list box (on the right). You can also copy and paste a list in. The list should be separated by newline characters.

5. Click on the 

button, once all the numbers have been added, to save the changes.

6. Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Remove Phone Numbers

The following section outlines the steps to remove phone numbers from the SIP Provider.

STEPS:

1. Click **Providers➔SIP Providers**. The **SIP Providers** page appears.

2. From the SIP Providers page, select the SIP provider link (under the **Name** field) or the  icon to the left of the provider you want to modify. The **Edit SIP Providers** page appears.

3. Scroll down to the **Phone Numbers** section of the page. Select the number that you want to remove from the list then click the **REMOVE** button. The number enter will be deleted from the list.

4. To remove multiple phone numbers, hold down the **SHIFT** key to select the numbers in sequence or the **CTRL** key to select in random order. Click the **REMOVE** button to delete the numbers selected.

5. Click on the 

button, to save the changes.

6. Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Set Destination

The following section outlines the steps to set a Destination for DID phone numbers.

STEPS:

1 Click **Providers→SIP Providers**. The **SIP Providers** page appears.

2 From the SIP Providers page, select the SIP provider link (under the **Name** field) or the icon to the left of the provider you want to modify. The **Edit SIP Providers** page appears.

3 Scroll down to the **Phone Numbers** section of the page. Select the number that you want to assign a destination. Select the desired Destination from the drop-down list then click the **SET** button.

4 To assign the same destination to multiple phone numbers, hold down the **SHIFT** key to select the numbers in sequence or the **CTRL** key to select in random order. Click the desired Destination from the drop-down list the click **SET** button. You can also set the destinations under Call Routing→**Outgoing.**

5 Click on the Save Changes button, to save the changes.

6 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Delete SIP Provider

The following section outlines the steps to DELETE a SIP Provider in the PBX system.

!	**IMPORTANT:** You must remove all instances of the SIP provider from any outbound dialing routes before you can delete it.

STEPS:

1 Click **Providers→SIP Providers**. The **SIP Providers** page appears.

2 From the **SIP Providers** page, icon to the left of the provider you want to delete. The SIP provider will be removed and will no longer appear on the list.

3 Click on the Save Changes button, to save the changes.

4 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

DESTINATIONS

Destinations are the various places that a call can be routed to within the system. The destinations drop-down list is populated as destinations are added to the system. During system implementation, if destinations are populated first it is easier to provision the hardware trunks because all of the destinations will be available in the drop down menu. These system destinations should be obtained in advance to simplify the setup process. In most cases, there will be a menu to setup for an automated attendant. There will also be business hours to setup. We recommend that you use the **IPitomy Setup Worksheet** to help you gather and organize the necessary provisioning information prior to installation (see the figure below). The Setup Worksheet can be found on IPitomy's website.

(Please refer to the "Installation Worksheet" section of this user guide for details on how to use the Setup Worksheet).

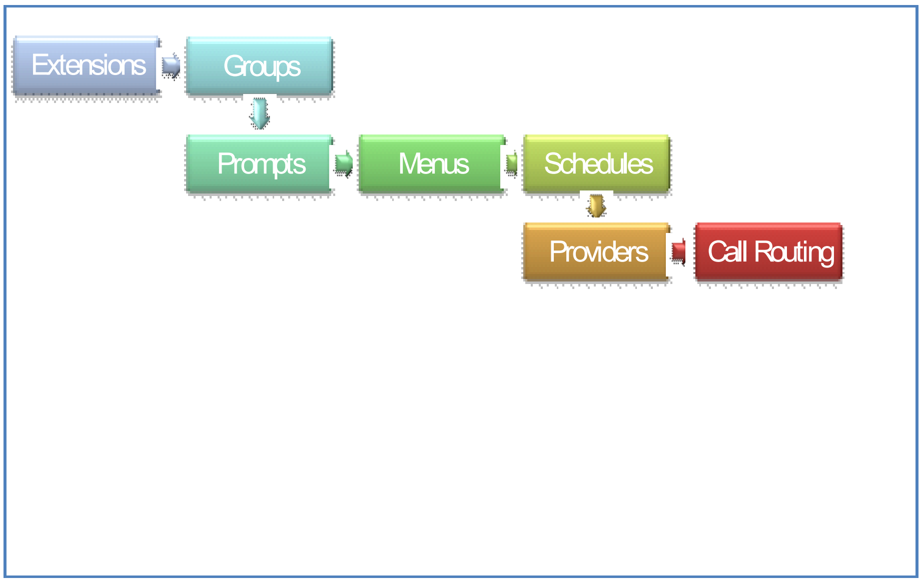


Figure 26 – IP PBX Data Components

- **Extensions:** are individual extensions assigned to a telephone. When an extension is created, a voicemail box is also created. This voicemail box is automatically the destination for an extension call which exceeds its ring time.
- **Groups:** are groups of extensions that can have different ringing strategies and can be routed from any trunk, another destination or dialed from an extension.
- **Menus:** are used for creating automated attendant menus to route callers to different destinations within the system. The voice prompt can also be used to play information like driving directions.
- **Conference Rooms:** allow multiple users to participate in the same call. Callers can be routed to a Conference via; direct dialing, through a DID, using an automated attendant or transferred by a person.
- **Voicemail Boxes:** are where callers leave a message when someone is not available at an extension. Voicemail boxes that are created separate from extensions can be used to route callers after hours or as an overflow destination. Note that in the current implementation Voicemail Boxes without an extension cannot be dialed with only the voicemail box number. You

can enter as one leaving a message by dialing ***+** box number or you can check the messages by going through the voicemail gateway or dialing 924.

- **Schedules:** route callers to different destinations in the organization during specified times and dates.
- **Branch Office:** connections provide broadband access to other branch locations by dialing a short access code followed by the extension number. Branch Extensions can also be defined as part of a branch office. These are remote extensions that can be dialed directly as though they were local extensions on your PBX. Branch Extensions also appear in drop down boxes for routing calls to specific destinations.

Extensions

Extensions define where specific people or departments can be reached in an organization. They should be setup first in the system. The following is a list of the various settings/parameters that will can be updated for each extension. The parameters you configure for the extensions will vary based on the customer's general business practices.

- **General Settings**
- **Forwarding Settings**
- **Advanced Network Settings**
- **Advanced Voicemail Settings**
- **Advanced Allow Codecs Settings**
- **Advanced Calling Permissions**
- **Advanced Follow-Me**

Add/Import Tab

The **Add/Import** tab allows you to create new extensions or edit existing extensions.

Use the Setup Worksheet to upload the CSV file and automatically create the Extension information. Please refer to the Setup Worksheet section of this guide for details.

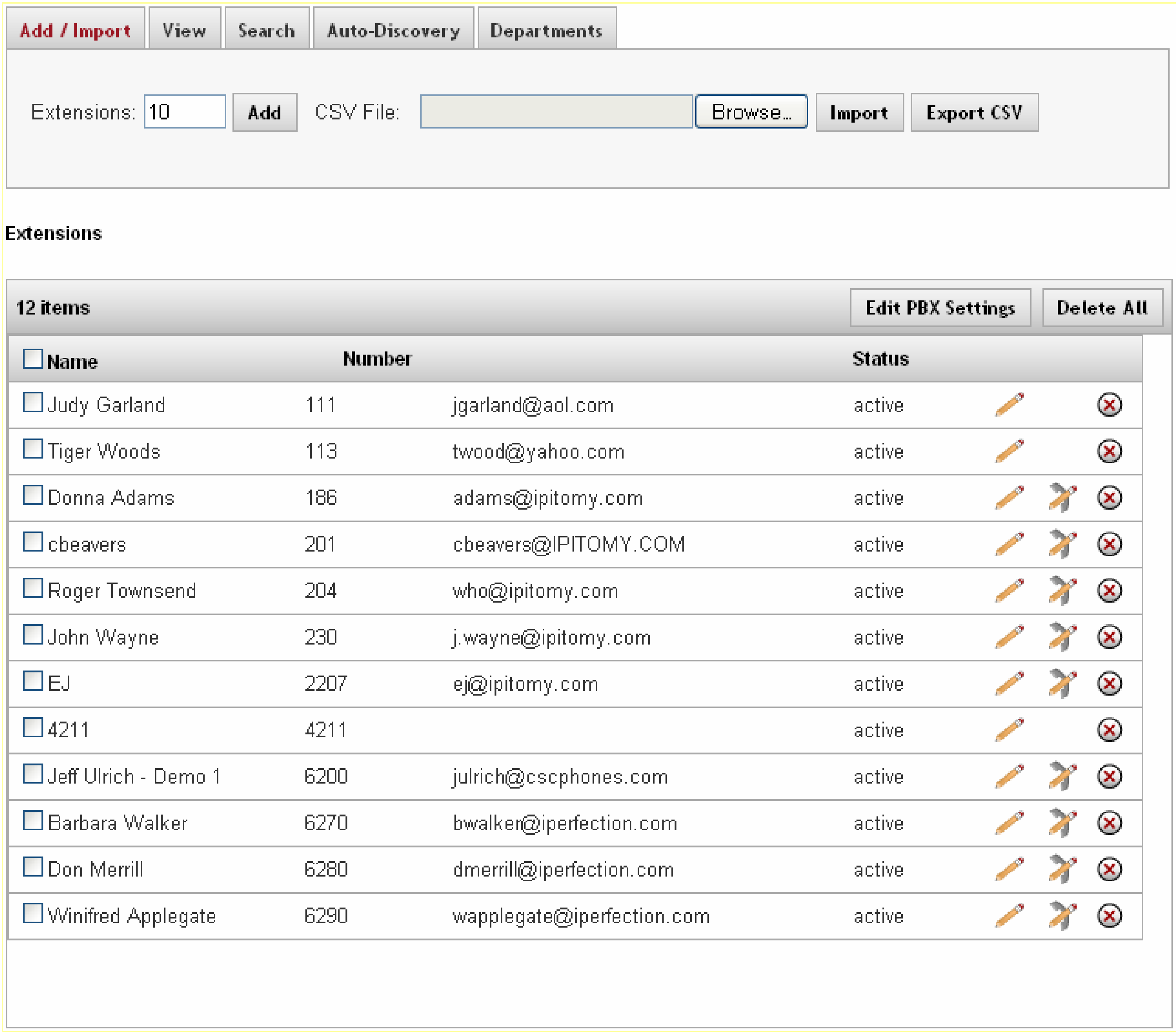


Figure 27 – Extensions Add/Import Page

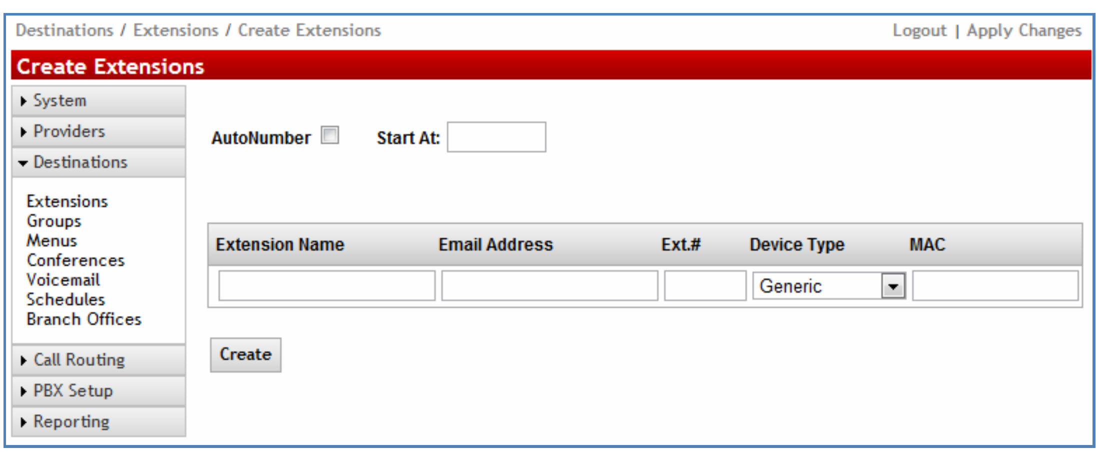


Figure 28 – Create Extensions Page

Sections/Fields	Description
Extension Name	This is the name of the person that the extension will be assigned to (using this device).
Email Address	This is the user's email address (optional)
Ext. #	The extension number that is assigned to this device.
Device Type	This is the type of device that will be using this extension (i.e. IP550).
MAC	This is the MAC ID of the device (optional).
AutoNumber/Start At	Selecting this option allows you to automatically number the extensions that you need to add. To use this feature, simply enter the extension number you want to start with then select (place a checkmark) the AutoNumber option.

Table 15 – Create Extension Fields and Descriptions

Add/Create Extensions

This section describes in detail how to create a new extension.

STEPS:

1 From any page of the PBX Admin system, click on the ⊞ button next to the **Destinations** link to expand the menu. The **Destination** menu opens and displays the options available.

2 Click on the **Extensions** link. The Extensions page opens and displays a listing of extensions (if ones already exists).

3 Click on the box to the left of the **Extensions** field. This value defaults to "**10**". Enter the number of extensions you want to add and then click the **ADD** button.

4 The **Create Extensions** page appears displaying the number of rows that was specified. Enter information for the extension in these fields. See table above for details.

*You can have the system automatically number the extensions you want to create by clicking the **AutoNumber** checkbox located on the top left hand corner of the **Create Extensions** page.*

5 Click on the **CREATE** button when all the extension information you want to create is entered. The system responds with a message indicating the results of adding the new extension(s). You should see a "**SUCCESS**" message.

6 If there is an error, you will see an "**ERROR**" indicated under the Results field. An error is typically due to an extension number that is being duplicated (already existing in the system). Make the necessary adjustments to correct the error then click the **CREATE** button.

7 Click on the [Save Changes] button, to save the changes.

8 Select the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Search Tab

This tab allows the user to search the extensions for keywords in the fields: Class of Service, Departments, Name, Numbers, CID Names, CID Numbers, Email, or Status .This section describes in detail how to search for existing extensions.

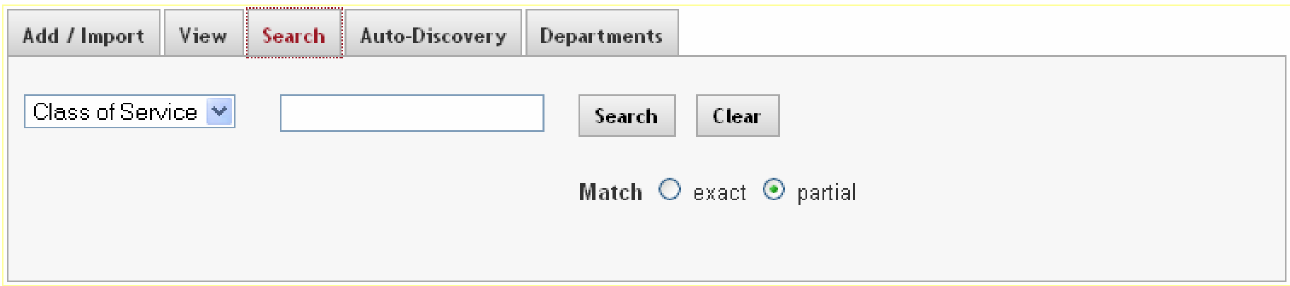


Figure 29 – Search Extension Page

The following table describes the types of search parameters the system will perform.

Sections/Fields	Description
Class of Service	This is the class of service that is assigned to the extension(s).
Departments	This is the department that is assigned to the extension(s).
Names	This is the name of the user assigned to the extension(s).
Numbers	This is the extension number.
CID Names	The caller ID name that is associated with the extension(s).
CID Numbers	This is the caller ID number that is associated with the extension(s).
Emails	Email address that is associated with the extension(s).
Status	This is the status of the extension(s).
Match Search Filter	**Exact** – indicates that you want the search to match exactly as the search criteria that is entered. **Partial** – indicates that you want to partially match the search criteria entered.

Table 16 – Search Extension Parameters and Description

Search Extension

This section describes how to search for extensions.

STEPS:

1 From the **Extensions** page, click the **Search Tab**. The list of extensions that are in the system appears with search options at the top section of the page.

2 Click the drop-down arrow icon next to the **Class of Service** list.

Select the desired **search criteria** then enter the parameters in the box to the right of the list then click the **Search** button. If the system finds any extensions matching your search parameter, it will display the information in the extensions window.

If the field is left blank, the system will bring all the extensions.

View Tab (Extensions)

This tab allows the user to sort the display of extensions by Phone Model. Once sorted, phone key settings can be mass edited for phones of the same model. This section describes in detail how to view existing extensions.

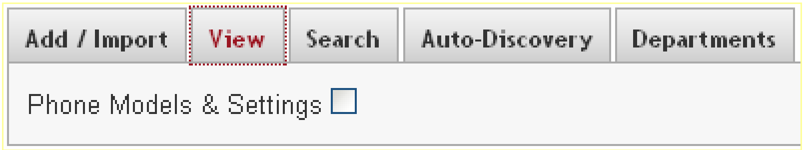


Figure 30 – Extensions View Page

Edit or View Extension

This section describes in detail how to view or edit extension details.

STEPS:

1 Navigate to the **Destinations➔Extensions** page.

2 Select the or the icon to the right of the extension name you want to view or edit. The pencil edits the PBX settings and the pencil with the handset behind it edits the Phone settings.

3 The **Edit Extensions** page displays with setting details for the extension.

4 Make the necessary changes to the extension.

5 Click on the 

button to save the changes.

6 Click the **Apply Changes** link located at the right hand corner of the top of the page, to commit the changes to the database.

You can edit multiple extensions by selecting (placing a checkmark in the box next to extension name). Only the fields being changed (that is common for all extensions selected) will be modified – i.e. Status or call group, etc.).

*Another shortcut that the system provides you is the **Previous** and **Next** button located on the top left corner of the **Edit Extension** page. Use these buttons to navigate backward or forward to find the extensions you want to view or modify.*

Mass Edit PBX Extension Settings

This section describes in detail how to view or edit extension details.

STEPS:

1 From the **Destinations➔Extensions** page, click on the **View** tab. A list of extensions appears.

*You can sort the list by phone model by selecting (placing a checkmark) the **Phone Model & Settings** option located in the top left hand corner of the screen.*

If you are using this method, you can also Mass Edit Phone Key settings. See the section below for steps.

2 Select (place a checkmark) in the box next to the extension name(s) you want to view or edit.

3 Click on the **Edit PBX Settings** button. The extension details page appears. On the top left corner of the screen, you will see the extension numbers that you are viewing or editing.

4 Make the necessary changes to the extension settings then click on the 

button to save the changes.

5 Click the **Apply Changes** link located at the right hand corner of the top of the page, to commit the changes to the database.

Mass Edit Phone Key Settings

This section of the Administration Guide describes how to mass edit the key settings for extensions using the same phone model.

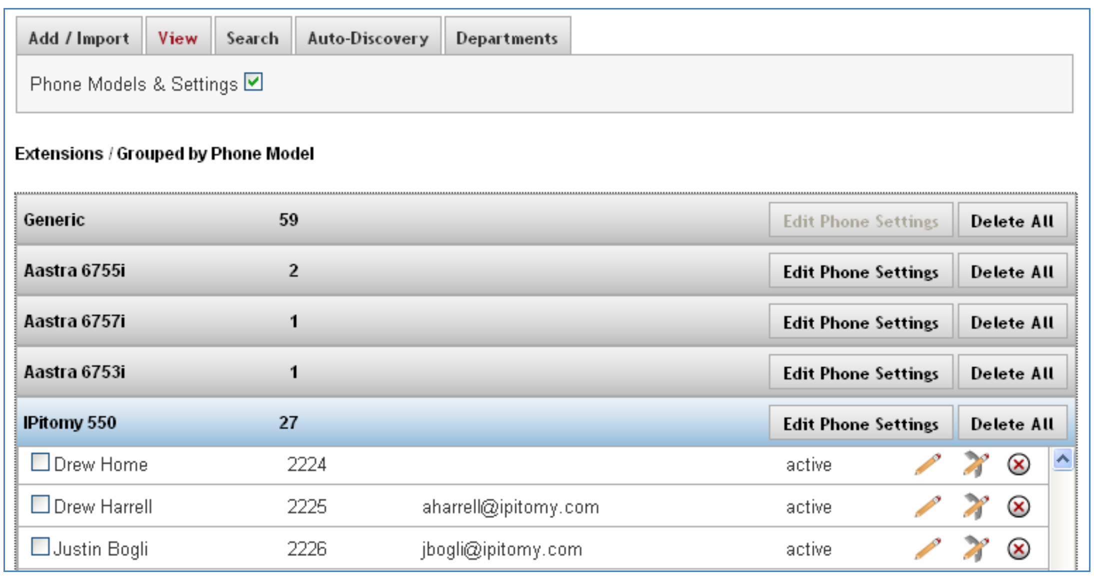


Figure 31 - Extensions View Tab (Mass Edit Feature)

STEPS:

1 From the **Destinations→Extensions** page, click on the **View** tab.

2 Select the **Phone Models and Settings** option at the top left hand corner of the list.

3 The system will display a list of the extensions in a grouping of phone types. Select the group of phones you want to edit. A listing of all the extensions in the phone group will appear.

4 Select the box next to the name of the extensions you want to update or click on box next to the **Name** field at the top of the column to select all the extensions.

5 Click the **Edit Phone Settings** button. The system will take you to the **Key Settings** page for the phones. At the top of the page you will see a list of all the extensions that you are updating.

6 By default, the **Only Save Changed Fields** option Is selected, de-select if needed.

7 Once all the changes are made, select the **Save and Restart Phone** button to save changes and reboot the phones so they can pull down their updated configuration file.

Delete Extension

This section describes in detail how to delete existing extensions.

STEPS:

1 Navigate to the **Destinations➔Extensions** page

2 Select the icon to the right of the extension name you want to Delete

3 The extension is deleted and a confirmation message will appear. Click **OK** on the message window.

4 The system returns you to the **Extensions** page. The extension that was just deleted will no longer appear on the list of extensions.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Delete Multiple Extension

This section describes in detail how to delete multiple extensions.

STEPS:

1 Navigate to the **Destinations➔Extensions** page

2 Select (place a checkmark) in the boxes next to the extension name(s) you want to delete.

3 Click on the Delete All button. The selected extension will be removed from the list and deleted from the database.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Extensions - General Settings Section

Once the extensions have been added to the database, you can edit the settings for each of the extensions.

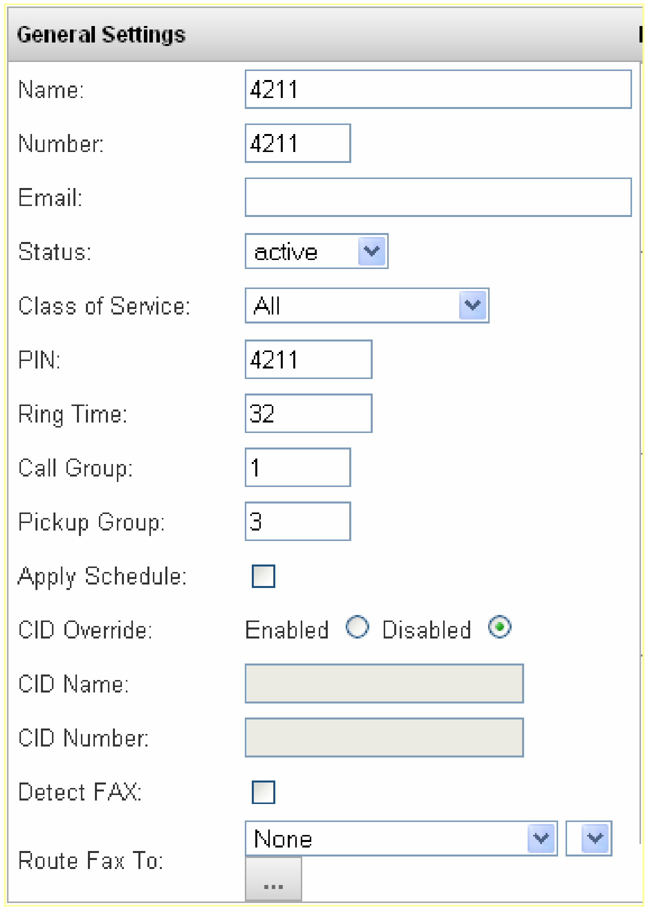


Figure 32 – Extensions General Settings Section

Sections/Fields	Description
Name	Name of the user associated with the extensions being created
Number	Extension number for this person or department. This must be 3 to 4 digits in length
Email	Email address for the person assigned to the extension. This will allow the system to forward email messages to the address of the person at the extension when properly configured.
Status	**Active** = currently in use **Disabled** = currently not in use
Class of Service	This is the service type for the extension. When initially created, the PBX will set this to the COS you have definined as the system default class of service on the PBX Setup➔General Page

PIN	This is the number used to access the extensions voicemail and can be between 1 and 6 digits long. The default setting is for the PIN to be the extension number. Be sure to instruct users to change the PIN to avoid unauthorized use.
Ring Time	This is the time in seconds that a call will ring before it is considered unanswered. Ring time must be between 1 and 100 seconds in length.
Call Group	This number assigns this extension to a group with a similar purpose (e.g., Sales or Customer Service). Multiple call groups can be assigned to each extension by putting a comma between the group numbers. The call groups also define which Pickup Groups can answer calls to this extension.
Pickup Group	This number should match any Call Group number entered on an extension. It defines the Call Group Numbers this extension can pickup remotely by pressing 99.
Apply Schedule	When an extension is created, a schedule destination is created automatically. This schedule is not activated until the Apply Schedule box is selected. When it is selected, all calls sent directly to this extension must first pass through the extension's schedule and will be routed accordingly. Extension schedules will appear with the name of the extension (e.g., Extension 123 would appear as "ext_123"). *(See the Schedules section of this guide for more information.)*
CID Override	If enabled, the user will be able to override the Caller ID settings. When the user places a call, the original assigned CID will be bypassed and the name and number will that is entered in the CID Name and Number fields will be sent instead. Always check with your provider that CID override is allowed before configuring.
CID Name	If the CID Override parameter is enabled, this is the Caller ID name that will be seen by the recipient when an outbound call is placed. Always check with your provider that CID override is allowed before configuring.
CID Number	If the CID Override parameter is enabled, this is the Caller ID number that will be seen by the recipient when an outbound call is placed. Always check with your provider that CID override is allowed before configuring.
Detect Fax	If checked, calls from an Analog or T1/PRI card that route direct to this extension will spend a period of time (Defined under PBX Setup➔General) checking if the call is a fax. During this time, the PBX holds the call; if fax tone is detected, the call will be passed along to the destination defined for Route Fax To, otherwise it will pass the call to the extension after the detection time has expired.
Route Fax To	Define where calls will be routed if fax tone is detected.

Table 17 – General Extension Settings and Descriptions

Extensions - Forward Settings Section

The extensions forwarding settings are made to be very user friendly. The settings may be modified from the Smart Personal Console, changed from your telephone extension, or changed remotely from any telephone (including cell phones), using the touch-tone key pad.

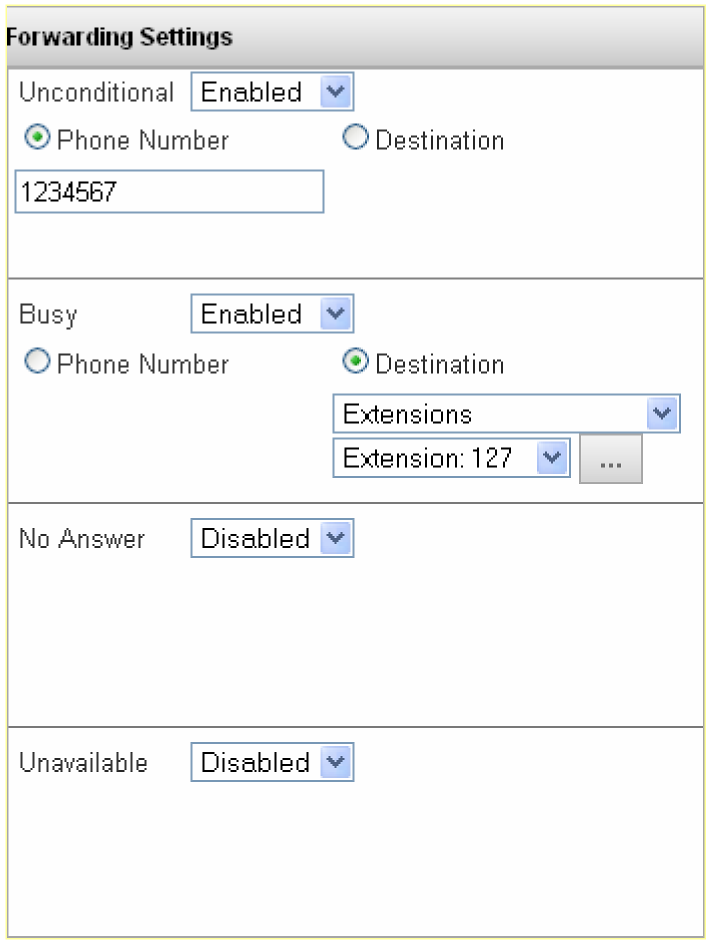


Figure 33 – Extensions Forward Settings Section

Forward settings routes calls to a different destination. These settings can be:

Sections/Fields	Description
Unconditional	Always route calls to a specific destination.
Busy	Route calls to a specific destination when the extension is in use or do not disturb is enabled.
No Answer	Route calls to a specific destination when a call is not answered in the defined Ring Time
Unavailable	Route calls to a specific destination when a phone is turned off, is not registered with the system, or has reached its call limit (as set in the IPitomy IP PBX).

Table 18 – Extension Forward Settings and Descriptions

Enable/Disable Forward Settings

The following outlines steps to enable or disable forward settings:

STEPS:

1. Pick the setting to be modified – Unconditional, Busy, No Answer or Unavailable.
2. Select **Enabled** or **Disabled**. Disabled turns the forward setting off. Enabled turns the forward setting on.
3. Select either Phone Number or Destination. Phone Number allows you to enter the digits you want dialed, like a PSTN number. Destination will bring up the standard dropdown list of destinations in the system; Extensions, Groups, etc.
4. Enter the Phone Number or select the Destination you would like the PBX to route to when meeting the forwarding requirements.
5. Click 

button to save the changes. The system returns you to the **Edit Extensions** page.
6. Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Change Unconditional Forwarding via Keypad

STEPS:

Only unconditional forwarding can be changed from a touch-tone keypad. Enter the following code to set the unconditional forwarding setting.

1. Dial ***90** to **disable** forwarding.
2. Dial ***91** to **enable** forwarding.
3. Dial ***92** to **set** the forwarding number.

Change Unconditional Forwarding via PC

The following outlines the steps for the end user to change the extension forwarding setting from a PC using the Smart Personal Console. The administrator will need to have enabled Allow User to control Forwarding under the extensions calling permissions.

STEPS:

1. Browse the internet to the **Smart Personal Console** page.
2. Login.
3. Click the 

link to access the forwarding page.
4. Enable/Disable the desired forwarding setting.
5. Select either Phone Number or Destination. Phone Number allows you to enter the digits you want dialed, like a PSTN number. Destination will bring up the standard dropdown list of destinations in the system; Extensions, Groups, etc.
6. Enter the Phone Number or select the Destination you would like the PBX to route to when meeting the forwarding requirements.
7. Click 

button to save the changes

Change Forwarding Number While Away from an Extension

Only unconditional forwarding can be changed from a touch-tone keypad.

STEPS:

1. Call into the **Automated Attendant** (menu).
2. Select the touch-tone digit that routes to the Forwarding Gateway.
3. The system will prompt for an **Extension Number** and **Password**.
4. The system will indicate if extension forwarding is **Enabled** or **Disabled**.
5. Pressing “**1**” toggles between Enabled and Disabled.
6. Pressing “**2**” allows the forwarding destination to be modified.

Extensions - Advanced Settings

Extensions - Network Settings Section

Network settings automatically register in the extension through the system. These settings represent registration and identification information. The system (extension) defaults should not be changed without advanced knowledge of the behaviors of the particular settings.

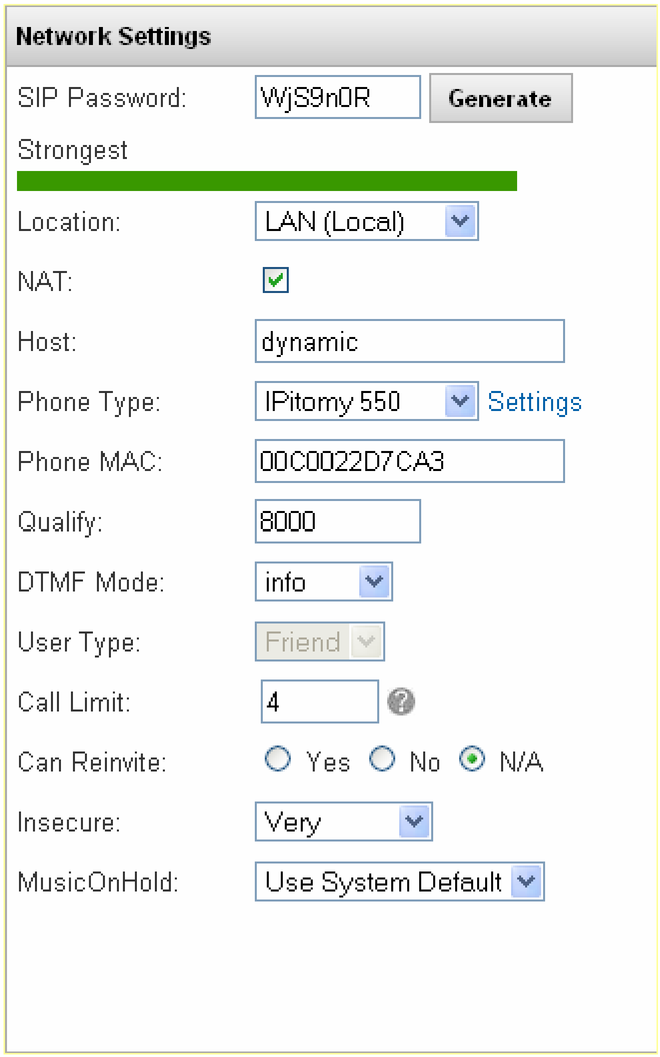
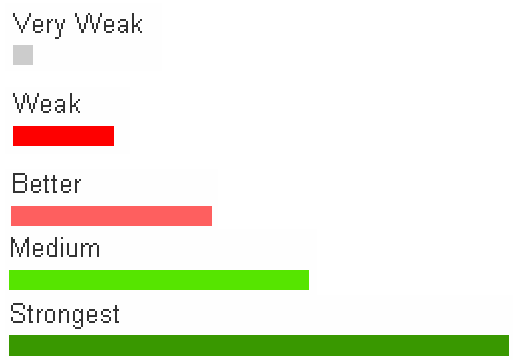


Figure 34 – Extensions Advanced Network Settings Page

Sections/Fields	Description
SIP Password	Password for the user to access IP PBX web-based administration system. Use a combination of uppercase letters, lowercase letters, and numbers.
Generate	If clicked, the system will automatically generate a password for the extension.
Password Strength	This color code bar indicates the **strength of the password** being assigned for the extension. The strengths are represented with the following colors: 
Location	This allows the user to tell the PBX whether to expect this extension to register as a local extension (LAN) or as a remote extension (WAN).
NAT	**IMPORTANT: This setting should be ENABLED (checked) unless otherwise instructed. Please contact an IPitomy's Technical Support Group for assistance or more information.**
HOST	**IMPORTANT: This should be set to DYNAMIC unless otherwise instructed. Please contact IPitomy's Technical Support Group for assistance.**
Phone Type	The phone type is a drop down list for selecting which IP phone hardware is being used on the extension. IPitomy supports Aastra® phones as well as our own IP550 and IP120 phones, and will be adding additional phone types in the future. When the phone type is selected, another configuration option is available to program the button mapping of each telephone model. The IPitomy IP PBX supports a variety of pre-programmed buttons like BLF, park, voicemail, as well as custom configurable speed dial buttons. Each phone can be configured for its own unique set of buttons.
Phone MAC	All of the IP phones have a MAC Address. The MAC ID identifies the piece of equipment for configuration. The auto configuration features of IPitomy rely on the MAC address to load the proper configuration files into the telephone when changes are made in the Web-based interface. The configuration files are stored on the IPitomy IP PBX and used when the phone powers back on after a power down cycle. If the configuration files have been updated when the phone powers back on, a new configuration is loaded into the phone. When the new configuration file is loaded, manual settings on the phone take priority and will be kept intact during the upgrade. Note that at this time only Aastra® phones require and utilize the MAC address in the phone settings.
Qualify	This is the interval of time between checking registration for the phone. Default is set to 8000 and should not be changed unless instructed by an IPitomy representative.

DTMF Mode	This is where you set what kind of DTMF signaling the extension will use. The dropdown lists options are: • rfc2833 • auto • info • inband Typically you would want the DTMF type to be **info** on both the PBX and in the phone itself.
User Type	This is defaulted to **Friend** and should not be changed unless instructed to do so by an IPitomy representative.
Call Limit	This is the number of concurrent calls allowable at an extension. The Call Limit selected must be between **1** and **99**. Default is set to **4**. It needs to be above **0** for BLF keys to function.
Can Reinvite	This parameter allows a device to reconnect calls midstream. • **YES** = if the phone type allows the re-invite feature • **NO** = if the phone type does not allow the re-invite feature • **N/A** = accepts the system wide default defined in the System Setup section of the Administration System. If the default setting is acceptable and works within your business, we recommend leaving the parameter set to N/A.
Insecure	This parameter allows you to specify how to handle connections with peers. Explanation of the different options available on the drop-down list are: • **PORT** = Ignore the port number where authentication came from. • **INVITE** = Do not require the initial invite to authenticate. • **PORT.INVITE** = Do not require initial invite to authenticate and ignore the port where the request came from. • **YES** – To match a peer based by IP Address only and not the port. • **VERY** – To allow registered hosts to call without re-authenticating. This is the default setting.
Music On Hold	This setting allows the user to select a different Music On Hold playlist for their extension then the system default playlist.

Table 19 – Extensions Advanced Networking Settings and Descriptions

Edit Extensions - Network Settings

STEPS:

1 From the **Edit Extensions** page, locate the extension that you want to update. Click the (edit extensions) icon to the right of the name.

*You can also **edit multiple** extensions by selecting (placing a checkmark) in the boxes to the left of the extensions you want to update. Click the Edit PBX Settings button located at the top right hand corner of the list.*

*You will see the extensions that are currently being updated. Make sure that the "**Only save the changed fields**" box is selected.*

2 The **Extension Details** page appears. Select the Advanced link to open the **Advanced Settings** page.

3 Make the necessary changes in the **Network Settings** section of the page.

4 Click Save Changes button to save the changes. The system returns you to the **Edit Extensions** page.

5 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Extensions - Voicemail Settings Section

These settings manage voicemail messaging and routing. Mailboxes created when the extension is built can either be edited on the Advanded section of the extension or on the Voicemail section under Destinations.

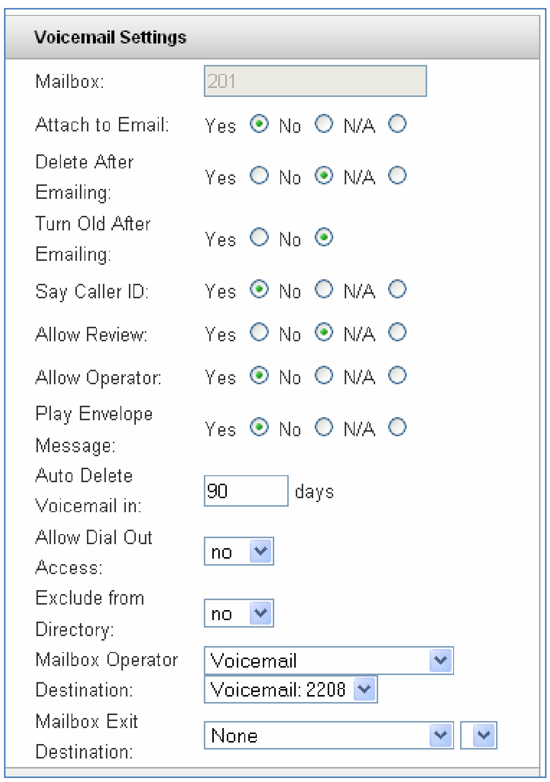


Figure 35 – Extensions Voicemail Settings Section

This table describes the voicemail setting options for each extension.

Sections/Fields	Description
Mailbox	This is the number associated with the extension.
Attach to Email	Select to enclose the message received in a notification e-mail as an attachment to the email address entered for the extension. An audio file (.Wav) will be the attachment. This requires for Unified Messaging to be configured on the PBX. **YES** = attach message to email. **NO** =do not attach message to email. **N/A** = accepts the system wide default defined under PBX Setup→Voicemail. If the default setting is acceptable and works within your business, we recommend leaving the parameter set to N/A.
Delete After Emailing	Delete the voicemail after it has been emailed to the email address provided for the extension in General Settings. **YES** = delete message after emailing. **NO** = do not delete message after emailing. **N/A** = accepts the system wide default defined under PBX Setup→Voicemail. If the default setting is acceptable and works within your business, we recommend leaving the parameter set to N/A. **Note: This option should not be enabled with turn old after emailing. If you enable both, the message will not be emailed but it will be deleted.**
Turn Old After Emailing	After emailing, the system moves the voicemail message to the Old folder. **YES** = will move message to Old messages folder after emailing. **NO** = messages will not be moved after emailing. **Note: This option should not be enabled with delete after emailing. If you enable both, the message will not be emailed but it will be deleted.**
Say Caller ID	State Caller ID prior to playback of the message. **YES** = play caller id prior to message content. **NO** = do not play caller id prior to message content. **N/A** = accepts the system wide default defined under PBX Setup→Voicemail. If the default setting is acceptable and works within your business, we recommend leaving the parameter set to N/A.
Allow Review	Allow callers to review a message after they have recorded it. **YES** = give callers leaving messages the option to review and rerecord the message they are leaving. **NO** = do not give callers the option to rerecord. **N/A** = accepts the system wide default defined under PBX Setup→Voicemail. If the default setting is acceptable and works within your business, we recommend leaving the parameter set to N/A.

Allow Operator	Allow pressing “0” during the voicemail greeting to reach the system-wide operator. **YES** = allow dialing 0 from mailbox. **NO** = disallow dialing 0 from mailbox. **N/A** = accepts the system wide default defined under PBX Setup➔Voicemail. If the default setting is acceptable and works within your business, we recommend leaving the parameter set to N/A.
Play Envelope Message	Play the time of call prior to the message. **YES** = enabled. **NO** = disabled. **N/A** = accepts the system wide default defined under PBX Setup➔Voicemail. If the default setting is acceptable and works within your business, we recommend leaving the parameter set to N/A.
Auto Delete Voicemail In	Define the number of days in which voicemail messages are to be automatically deleted from a mailbox. If this is set to “**0**” (zero) the voicemail message will never expire.
Allow Dial Out Access	This allows the ‘Dial Out’ feature when a user is listening to their voicemail.
Exclude from Directory	This indicates whether to exclude this extension from the directory.
Mailbox Operator	If a caller presses “**0**” (zero) while listening to your mailbox greeting, the caller will be routed to this destination. Also, this is where a user will be sent if he dials ”0” from this extension. **Set this to None to use the system default which is set under PBX Setup➔General.**
Mailbox Exit Destination	The Mailbox Exit Destination is where the system will route a caller who presses **#** when they finish leaving a voicemail message. **Setting this to None will use the system default, which is set under PBX Setup➔General.**

Table 20 – Voice Mail Settings and Descriptions

Edit Voicemail Settings

The following outlines the steps to set voicemail parameters.

STEPS:

1 From the **Edit Extensions** page, locate the extension that you want to update. Click the (edit extensions) icon to the right of the name.

*You can also **edit multiple** extensions by selecting (placing a checkmark) in the boxes to the left of the extensions you want to update. Click the Edit PBX Settings button located at the top right hand corner of the list.*

*You will see the extensions that are currently being updated. Make sure that the “**Only save the changed fields**” box is selected.*

2 The **Extension Detail** page appears. Select the Advanced link to open the **Advanced Settings** page.

3 Make the necessary changes in the **Voicemail Settings** section of the page.

4 Click Save Changes button to save the changes. The system returns you to the **Edit Extensions** page.

5 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Extensions - Allow CODECs Section

These transmission speeds are configured by the service provider and designed to automatically register in the extension through the system.

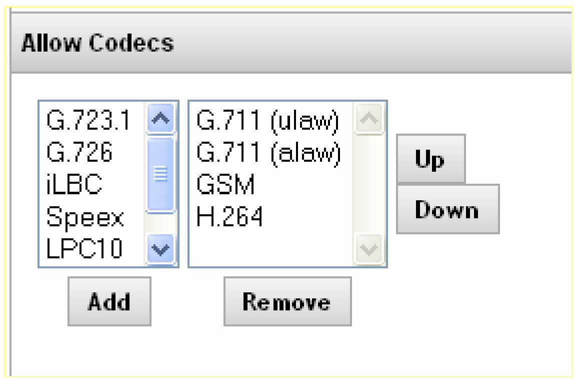


Figure 36 – Extensions CODECS Settings Page

Sections/Fields	Description
CODEC Permissions (Allow CODECs)	Allows the administrator to define which codec the extension should use, and specify a priority from top down. **IMPORTANT: Please contact IPitomy's Technical Support Group for assistance if you feel you need to change these settings.**

Table 21 – Extensions CODECS Settings and Recommendations

Edit CODEC Settings

The following outlines the steps to set CODECs parameters.

STEPS:

1 From the **Edit Extensions** page, locate the extension that you want to update. Click the (edit extensions) icon to the right of the name.

*You can also **edit multiple** extensions by selecting (placing a checkmark) in the boxes to the left of the extensions you want to update. Click the Edit PBX Settings button located at the top right hand corner of the list.*

*You will see the extensions that are currently being updated. Make sure that the "**Only save the changed fields**" box is selected.*

2 The **Extension Detail** page appears. Select the Advanced link to open the **Advanced Settings** page.

3 Make the necessary changes in the **Allow CODECs Settings** section of the page. Clicking Up or Down when highlighting a codec will allow you to raise or lower its usage priority.

4 Click Save Changes button to save the changes. The system returns you to the **Edit Extensions** page.

5 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Extensions Calling Permissions Section

Calling permissions define the types of calls that can be sent and received from an extension and the call actions this extension can take. For example, you may want to limit who has the ability to monitor another extension.

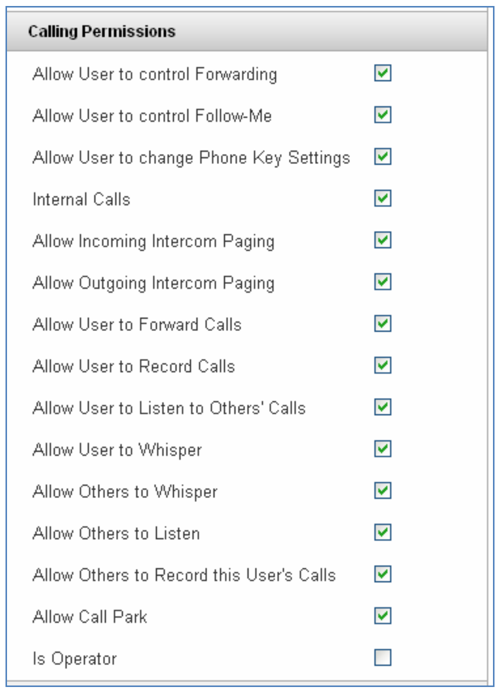


Figure 37 – Extensions Calling Permissions Section

The following table describes the settings for an extension's calling permissions.

Sections/Fields	Description
Allow User to Control Forwarding	If **Enabled** (checked), this allows the user to modify their forwarding and schedule settings when they log into the Smart Personal Console user interface.
Allow User to Control Follow-Me	If **Enabled** (checked), this allows the user to modify their Follow-Me settings when they log into the Smart Personal Console user interface.
Allow User to Control Phone Key Settings	If **Enabled** (checked), this allows the user to modify their Phone Key settings when they log into the Smart Personal Console user interface.
Internal Calls	If **Enabled** (checked), this permits calls made from internal extensions.

Allow Incoming Intercom Paging	If **Enabled** (checked), this permits a page to be heard through this extension.
Allow Outgoing Intercom Paging	If **Enabled** (checked), this permits a page to be made through this extension.
Allow User to Forward Calls	If **Enabled** (checked), this permits an extension to forward a call or voicemail message to another destination on the system.
Allow User to Record Calls	If **Enabled** (checked), this permits the extension to record phone conversations.
Allow user to Listen to Others' Calls	If **Enabled** (checked), this allows the user to listen to other user's phone conversations.
Allow User to Whisper	If **Enabled** (checked), this allows the user to whisper to another user during a phone conversation. Whisper is similar to Listen but you can coach and only the person at the extension can hear.
Allow Others to Whisper	If **Enabled** (checked), other extensions can Whisper to your extensions calls
Allow Others to Listen	If **Enabled** (checked), other extensions can Listen to your extensions calls
Allow Others to Record this User's Calls	If **Enabled** (checked), other extensions can Record calls at your extension
Allow Call Park	If **Enabled** (checked), this permits extension to park a call.
Is Operator	If **Enabled** (checked), the extension is designated as an operator. Being an operator allows the extension to control Day/Night mode overrides.

Table 22 – Calling Permission Settings and Descriptions

Add/Edit Calling Permissions

The following outlines the steps to set calling permissions for extensions.

STEPS:

1 From the **Edit Extensions** page, locate the extension that you want to update. Click the (edit extensions) icon to the right of the name.

*You can also **edit multiple** extensions by selecting (placing a checkmark) in the boxes to the left of the extensions you want to update. Click the Edit PBX Settings button located at the top right hand corner of the list.*

*You will see the extensions that are currently being updated. Make sure that the "**Only save the changed fields**" box is selected.*

2 The **Extension Detail** page appears. Select the Advanced link to open the **Advanced Settings** page.

3 Make the necessary changes in the **Calling Permissions** section of the page.

4 Click 

button to save the changes. The system returns you to the **Edit Extensions** page.

5 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Extensions - Follow-Me Section

The Follow Me feature allows the PBX to try and find a user by calling pre-configured numbers, simultaneously or in sequence of priority. Once answered, the called party is given the option to accept or reject the call. If the call is rejected, or not answered at all, the call will return to the PBX allowing the caller to leave a Voice Mail message.

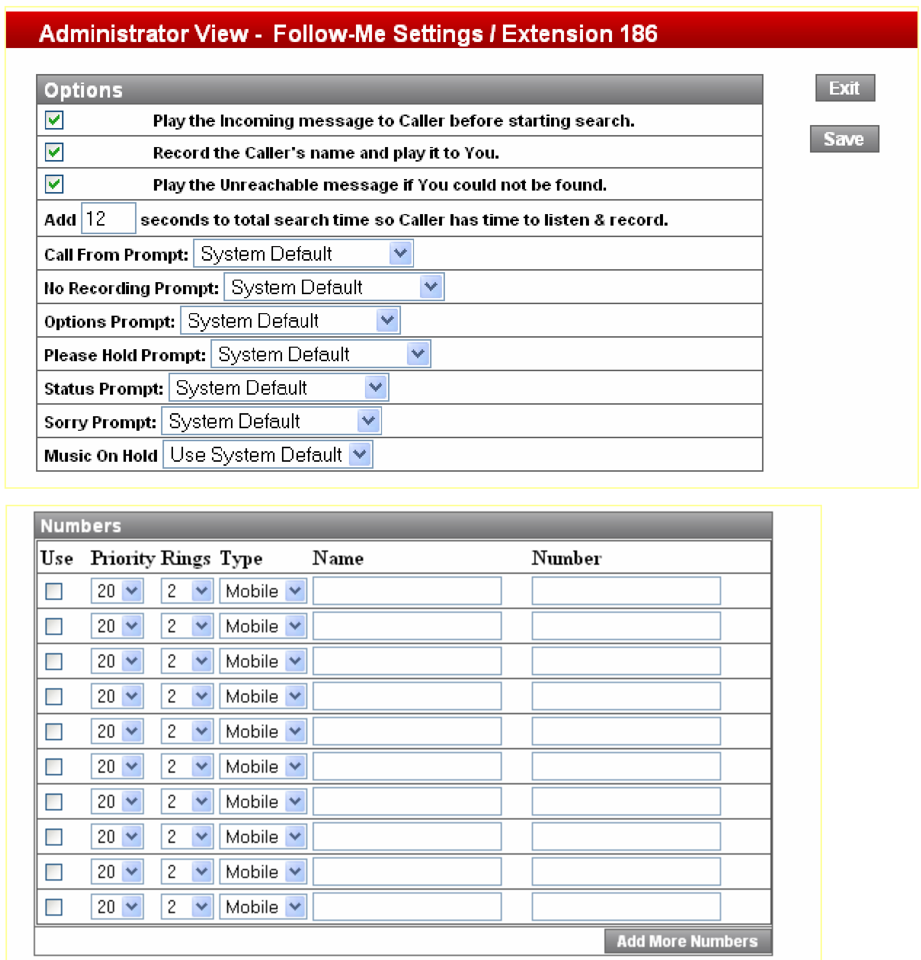


Figure 38 – Extensions Advanced Settings Follow Me Setup Page

From this window it is easy to configure numbers to be dialed by the system in order to find the user. You can adjust the Name, Number (Number to be dialed), Type of Device that is being called, Rings to wait for an answer, and you can *weight* the priority from 1-20 (20 being lowest 1 being highest) to define search order.

The Use radio box allows you to easily turn on and off numbers to be dialed. Since the SPC can be accessed via remote management, this allows the users to modify and create these lists from remote locations easily. The following table details the parameters and descriptions necessary to configure the Follow-Me feature.

The system also allows you to record your own Prompts (recordings) under **PBX Setup→Prompts**, and then use those prompts as custom messages for FollowMe feature.

Sections/Fields	Description
Play the Incoming Message to Caller before Starting Search	When **Enabled** (checked), the system plays the Status Prompt to the caller.
Record the Caller's Name and Play it to You	When **Enabled** (checked), the caller will be asked to record their name, and will announce that recorded name prior to prompting the called party to accept or reject the call.
Play the Unreachable Message if You Could Not Be Found	When **Enabled** (checked), this will play the Sorry Prompt if the call is not answered, otherwise it goes right to the voicemail greeting.
Number of Seconds to Total Search Time so Caller Has Time to Listen & Record	This allows you to configure how many seconds the system will spend searching for the called party. Default is 12 seconds.
Call From Prompt	This plays when **Record the Caller's Name and Play it to You** is **enabled** (checked). The system default message is **"Incoming Call From"** followed by the recording the caller made of their name.
No Recording Prompt	This plays when **Record the Caller's Name and Play it to You** is **disabled** (not checked). The system default message of ***"You have an incoming call"*** followed by the Options Prompt.
Options Prompt	This plays after you have answered the call and prompts you to press either "**1**" to **accept** the call or "**2**" to **reject** the call. The system default message can be changed, but the options remain the same.
Please Hold Prompt	This plays to the caller alerting them that the system is going to find the user they are trying to reach. The system default message of ***"Please hold while I try to locate the person you are calling"*** will play during the search process.
Status Prompt	This plays the system default message of ***"The person you are calling is not at their desk, I will try to locate them for you"***.
Sorry Prompt	This plays if the person could not be reached or they reject the call. The system default message of ***"I'm sorry, but I was unable to locate the person you were calling"***.
Music On Hold	This allows you to specify a particular Music On Hold playlist to play to the caller during the search process. The system default Music on Hold (set at PBX Setup➔Music on Hold) will play when this parameter is set to system default.
Numbers	If selected (checked), the Follow Me feature will try to either simultaneously or in sequence of priority to try and locate you by using this call list.

Table 23 – Extensions Follow Me Settings and Descriptions

Add/Edit Follow Me Settings

The following outlines the steps to set calling permissions for extensions.

STEPS:

1 From the **Edit Extensions** page, locate the extension that you want to update. Click the (edit extensions) icon to the right of the name.

2 The **Extension Detail** page appears. Select the Advanced link to open the **Advanced Settings** page.

3 Click on the **Numbers and Settings** button under the **Follow-Me** section of the **Advanced** page.

4 The **Administrator View – Follow-Me Settings/Extensions** window appears.

5 Make the necessary changes in the **Follow-Me** parameters. Click on the **SAVE** button to save the changes to the Follow-Me settings. A message confirming the changes will appear.

6 Click the **EXIT** button to close the window.

7 Click Save Changes button to save the changes. The system returns you to the **Edit Extensions** page.

8 Click the **Apply Changes** link located at the right hand corner of the top of the page, to commit the changes to the database.

Provisioning - Auto-Discovery Tab

This network discovery tool will automatically detect new devices that have been added to the network. This feature helps identify phones and other devices connected to the network and allow you to configure networked phones. The IP PBX system provides you with the option to either scan the network, or if a scan has been done recently, you could select No Scan to save time.

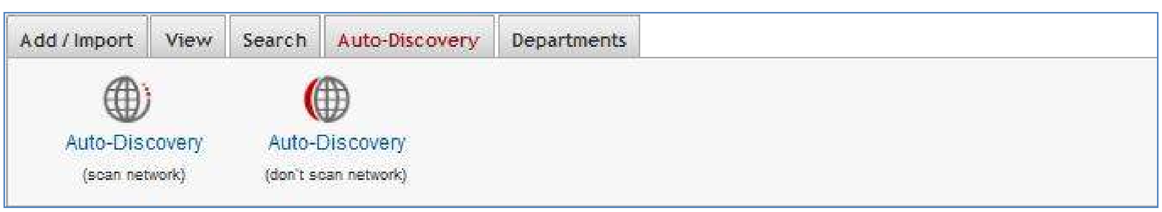


Figure 39 – Extensions Auto-Discovery Tab

Start Auto-Discovery Scan

This section describes how to start the Auto-Discovery scan for active devices on the network.

STEPS:

1 From the **Destination➔Extensions** page, click the **Auto-Discovery Tab**.

2 Select to either Scan or Not Scan the network and an Alert message indicating that the system is scanning the network for active devices appears.

3 Once the scanning process is complete, the system will display a list of active devices and assigned values such as IP Address, Device Type, Status, etc. Devices that appear

on this list may be selected for editing. By default, the PBX will only display IPitomy and Aastra devices. Filter options can be modified to change display other devices.

If you click the refresh-view more than once within a short time period, the network is not rescanning. The minimum time period allowed between scans is indicated under the Advanced Settings (Min Refresh Interval).

The network is only scanned for new information when you click REFRESH. Both REFRESH and FILTER ONLY apply whatever filters you have chosen to the displayed list.

List of Devices & Extensions

The list of discovered network devices comes from two sources:

- Network
- PBX Database

Assigned	Device	Status	MAC Address	IP
None	IPitomy	Online	00C0020FF3CE	192.168.2.104
None	IPitomy IPR20	Online	00C0022BAA2C	192.168.2.78
Extension: 2203	IPitomy 550	Online	00C0022D7C69	192.168.2.132
Extension: 2208	IPitomy 550	Online	00C0022D7C79	192.168.2.121
Extension: 201	IPitomy 550	Online	00C0022D7CA3	192.168.2.126
Extension: 2221	IPitomy 550	Online	00C0022D7CB5	192.168.2.100
Extension: 3207	IPitomy 550		00C0022D7CFF	
None	IPitomy 550	Online	00C0022D7D38	192.168.2.160
Extension: 2210	IPitomy 550	Online	00C0022D7D59	192.168.2.139
Extension: 2228	IPitomy 550	Online	00C0022D92FB	192.168.2.148
Extension: 2218	IPitomy 550	Online	00C0022D930E	192.168.2.108
Extension: 3007	IPitomy 550		00C0022E1321	

Color Legend

	Using Local PBX	Not Using Local PBX
Assigned ->		
Not Assigned ->		

Figure 40 – Auto-Discovery Device Color Legend

The following table describes the status of devices that will appear once an Auto-Discovery scan is completed.

Sections/ Fields	Description
Assigned	This field indicates that extension that is assigned to the device (phones). Clicking on the box to the left of this field allows you to select it for editing.
Device	The Device column lists info about the device. Database info is preferred, so if a device is set to a certain phone type in the database, this is listed instead of the actual device type.
Status	This is the status of the device. If the status field is blank, this indicates that the device was not found on the network. ***NOTE:*** *Phones that are in the process of restarting, turned-off, or operating on a different network during a network scan will not be in an "**Online**" status.*
	Hovering the mouse over this icon will provide device information. See **Figure 43** for an example of the data that can be viewed.
MAC Address	This is the MAC Address assigned to the device.
IP Address	This is the IP Address assigned to the device.

Table 24 – Auto-Discovery Scan Details

Device Information

To view detail information for a specific device, hover your cursor over the information icon to obtain details about the network device or phone. This can be an important source of information about which phones are available for live configuration.

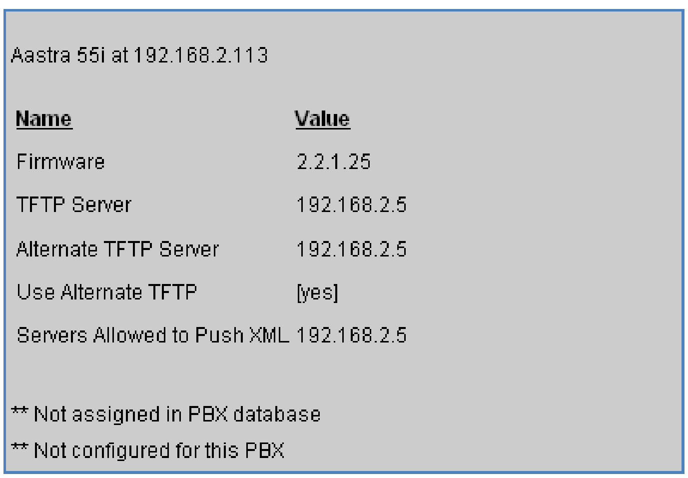


Figure 41 – Auto- Discovery Device Values

Edit Selected Tab

The IP PBX system offers many ways to configure devices. This section describes an optional method using the Auto-Discovery feature.

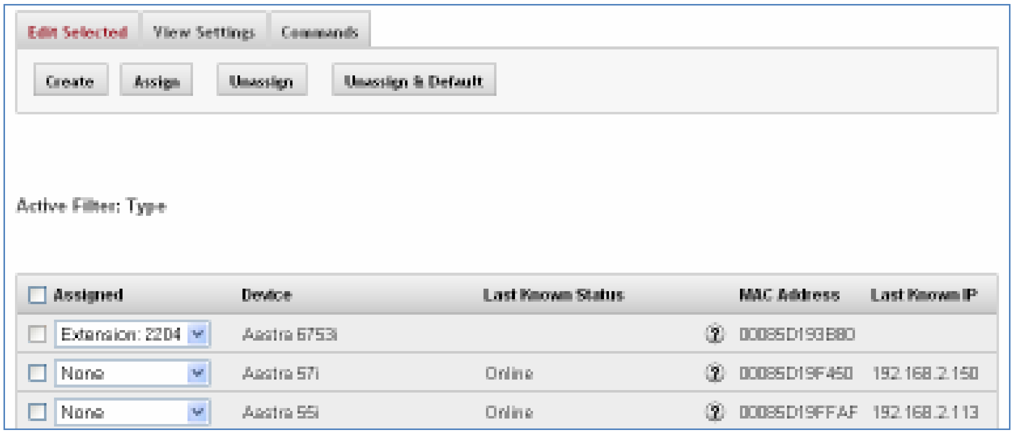


Figure 42 – Auto-Discovery Edit Selected Functions

The following table describes the fields and functions available on the Edit Selected Tab of the Auto-Discovery page:

Fields	Description	Commands Phone?
Create	Create new extensions for selected phones. You can upload extension details in a file or manually enter information.	NO
Assign	Assign existing extensions to selected phones.	NO
Unassign	Removes the configuration file for the phone and all association between the extension and the phone.	NO
Unassign & Default	Same as Unassign but also sends a Factory Default instruction to the phone.	YES

Table 25 – Auto-Discovery Edit Selected Tab Functions and Descriptions

Create, Assign and Configure Phone

The following outlines steps to create, assign and configure a device.

IMPORTANT: This assumes that the phones are in a FACTORY DEFAULTED state.

IPitomy recommends using the CSV upload file and the IP550 phone type which will help to simplify the setup and auto configuration process.

STEPS:

1 From the **Extensions➔Destinations➔Auto-Discovery** page, click the **Edit Selected** tab.

2 Place a check mark next to the phones you wish to use when creating the extensions

3 Click the **Create** button.

4 Enter the new extension information. The MAC addresses of the selected phones will be populated so you have a reference as to what phones you are creating the extensions for.

5 Click the “**Create**” button. If successful, Click “**Return to Extensions**”

6 Click the **Apply Changes** link located at the right hand corner of the top of the page, to commit the changes to the database.

7 Click the “**Auto-Discovery**” button and you will note the newly created extensions are now assigned to the phones that were checked.

8 Check the boxes next to the phones that you just created extensions for.

9 Click the Commands Tab.

10 Click on the **Configure & Restart** button to provision the phones.

View Settings Tab

Filtering does not change how the network scanning is performed. It only limits the list of items displayed. This means that if you only want to change filters it is not necessary to re-scan the network. All of the network information from the last network scan is retained and used by filters.

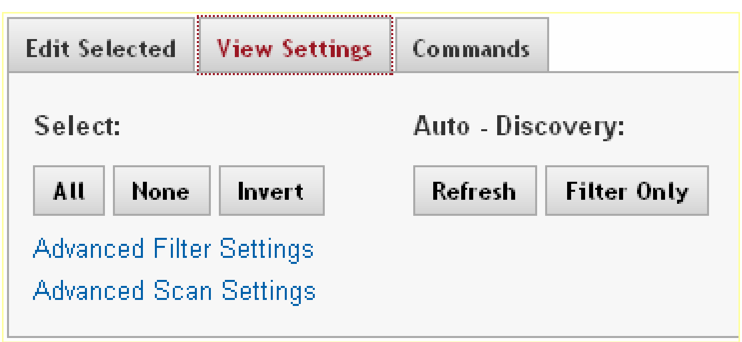


Figure 43 – Auto-Discovery View Settings Page

Sections/Fields	Description	Commands Phone
Select All, None, Invert	Check or uncheck multiple checkboxes with the click of a button.	NO
Refresh	Scan the network for devices. Scanning is done using the settings in Advanced Scan Settings. The scan results displayed depend on the active filters,controlled via Advanced Filter Settings.	NO
Filter Only	Displays results of the last network scan after applying the filters set in Advanced Filter Settings.	NO

Table 26 – Auto-Discovery Functions and Descriptions

Advanced Filter Settings

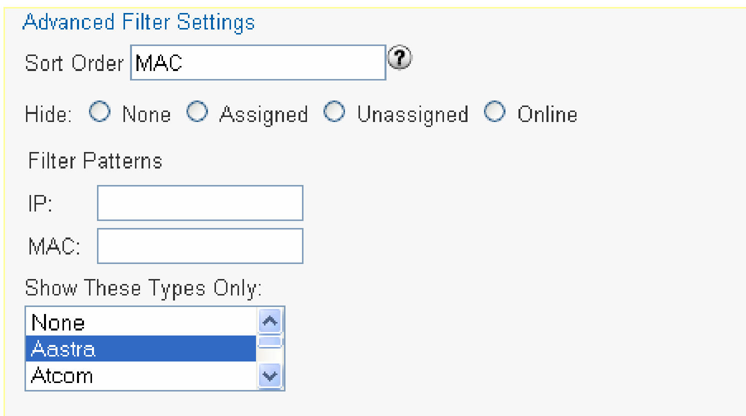


Figure 44 – Auto-Discovery Advanced Filter Settings

Sections/Fields	Description
Sort Order	This allows you to set what order items are listed. Hover your cursor over the help icon next to the Sort Order fields for options.
Hide	Allows you to hide different status types from view.
Filter Patterns	Allows you to either enter a partial MAC or IP here. The only system criterion is that the pattern must match from the first character onward. i.e. “000D” would match “000DE” but not “A000DE”. Full Regular Expression Matching is supported when you enclose your search pattern in double quotes. Some Examples are: • Pattern 1.2.3 = ^1\.2\.3 • Pattern “1\.2\.3.*” = 1\.2\.3.*
Filter by Type	You can select multiple device types to filter by. Auto-Discovery is aware of MAC address ranges used by many popular networked devices. Although this is not the only way Auto-Discovery identifies devices, this allows the tool to identify and filter devices by type. *Only IPitomy devices and Aastra phones are shown by default because they are set in the Filter by Type.*

Table 27 – Advanced Filter Settings and Descriptions

Advanced Scan Settings

Changes to Scan Settings are kept when you click the GUI 'Refresh' button. Clicking on your browser's refresh button will discard changes. For the most part these settings can remain unchanged and all will function as needed.

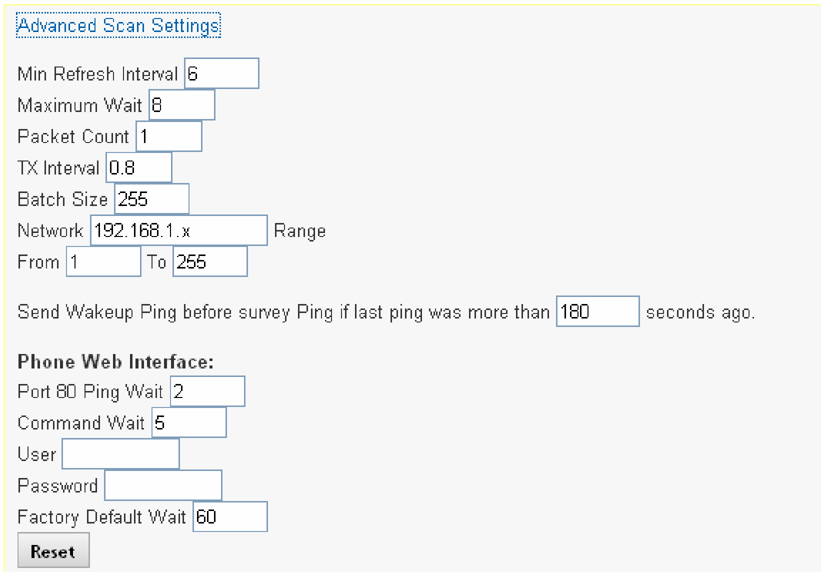


Figure 45 – Auto-Discovery Advanced Scan Settings Page

Sections/Fields	Description
Min Refresh Interval	Minimum number of seconds required between fresh network scans. If you attempt to scan the network more often than this you only get results stored from a previous scan.
Maximum wait	Maximum number of seconds to wait for all packets to return.
Packet Count	Number of packets to send
TX Interval	Transmission interval between packets. Only meaningful when Packet Count is greater than 1. Minimum allowed value is 0.2
Batch Size	Divide scanning range into batch jobs. Jobs are run sequentially. Each job will have at most <Batch Size> pings running in parallel at a time. The special value 255 means run all pings as one big job.
Network	Network address with last octet reserved for From and To range. Example: 192.168.1.x
From	Start number for last octet of network address.
To	End number for last octet of network address.

Send Wakeup ping before Survey Ping if…	Some devices are slow to respond to the first ping they receive but have faster response times afterwards.
Port 80 Ping Wait	Maximum seconds to wait for basic phone web interface to signal that it is alive. Rarely used.
Command Wait	Maximum seconds to wait for most commands.
User and Password	Phone's Web User Name & Password may have to be specified if the phone is not using factory default settings for these items. When these fields are blank Auto-Discovery uses default settings.
Factory Default Wait	Maximum seconds to wait for the phone to signal completion of a Factory Default.
Reset	The 'Reset' button will return all Scan Settings to original default values.

Table 28 – Auto-Discovery Advanced Scan Settings and Descriptions

Commands Tab

The IP PBX system offers many ways to configure devices. This section describes an optional method using the use Auto-Discovery feature.

Figure 46 – Auto-Discovery Command Functions Page

Sections/ Fields	Description	Commands Phone?
Factory Default	Sends a Factory Default and restarts the phone.	YES
Restart	Sends a Restart instruction to the phone.	YES
Configure & Restart	Instructs phone to set TFTP server to the Server IP specified under Advanced Settings. Then the phone is commanded to restart, which will result in pulling down configuration files and firmware files if there are updated ones present.	YES
Assign, Configure & Restart	Combines the functions of the Assign button with the Configure & Restart button.	YES

Table 29 – Auto-Discovery Command Settings and Descriptions

Factory Default Phone

STEPS:

1 From the **Extensions→Destinations** page, click the **Auto-Discovery Tab**.

2 Click the "**Auto-Discovery**" button, choosing to scan or not scan the network as needed.

3 Click (placing a checkmark) on the boxes next to phones you wish to default.

4 From the **Auto-Discovery** page, click on the **Command** tab. Click the "**Factory Default**" button. Wait long enough for the phones to default and restart. This could be anywhere from 30 seconds to 2 minutes for each phone, depending on the model. The phones will set themselves back to factory default and reboot during this period.

Troubleshooting Network Scanning Problems

Scan settings have been configured to work in a standard one subnet network environment. More complicated environments may require further tuning. The most useful setting is 'Maximum wait'. However noisy environments will require increasing the packet Count and TX interval.

STEPS:

To tune scanning for your network:

1 From the **Extensions→Destination→Auto-Discovery** page, click on **Advanced Scan Settings** link.

2 Increase the '**Maximum Wait**' setting by a few seconds.

3 Click the GUI '**Refresh**' button.

More specific tuning will require knowledge of your network and diagnostic tools like Ping.

*Phones that are in the process of restarting, turned-off, or operating on a different network during a network scan will not be in an "**Online**" status.*

Auto Provisioning Phone Settings from Actual Device

The remainder of the configuration is done via the Phone interface, and will only work with IPitomy phones.

STEPS:

1 Create an extension, setting the Device to either IP550 or IP120, depending on what IPitomy phone you intend to register to the extension.

!	**IMPORTANT: To configure the phone using the auto provisioning, make sure that the "Auto-Provisioning" parameter is RUNNING. This can be done in PBX Setup➔General option of the system.**

2 From your phone, select the Menu button to start Auto-Provisioning then select **Option 7: SIP Settings** (you can do this without using the scroll keys by just pressing Menu then 7).

3 It will prompt you for the **Admin** password. Enter the default password "**1234**" then press the Enter softkey (IP550) or the OK button (IP120)

4 Select **Option 8: Auto Prov**.

5 Select "**YES**" when the ***Autosearch PBX*** *prompt* appears.

6 **Enter** either the **extension** number of the extension that has been created.

7 Press the Done softkey (IP550) or the OK button (IP120) and the phone will run through the process of downloading its configuration file, updated firmware file, etc.

Edit Phone Settings

The settings for each individual device (phone) can be configured in the IP PBX system. The following outlines the steps to modify the phone settings from the IP PBX system.

View Phone Settings

Figure 47 – Extension Listing

Edit Phone Settings

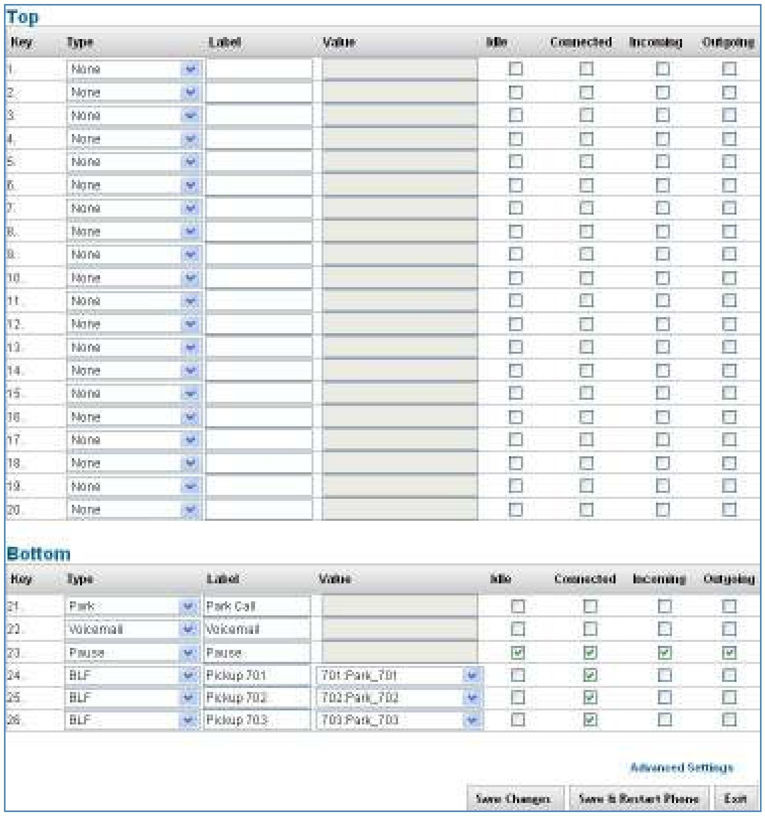


Figure 48 – Edit Phone Settings Page

Sections/ Fields	Description
Settings for Top Section of the Phone Setting	Many phone models have multiple areas where you can set keys. The TOP section typically refers to the keys around the LCD screen, or Soft keys.
Settings for Bottom Section of the Phone Setting	Many phone models have multiple areas where you can set keys. The BOTTOM section typically refers to the Programmable keys. See Appendix 1 for details on key types.
Type	Key type determines what function a key performs. A dropdown list shows which key types are available to the phone model you are working with.
Label	This is the label associated with this key. If configured for a key around the LCD screen, this is what will display.
Value	The value completes a key's function. It may contain a number to dial special codes or allow you to select an extension from a dropdown list. Most key types do not require you to enter a value. For example, the Voicemail key does not require a value because the System Administrator determines it for the end user.

Key State	Only necessary when configuring Soft Keys around the LCD screen. When checked, the key in question will display when the phone is in the states defined: **Idle –** No current call activity **Connected –** A call is connected **Outgoing –** A call is being placed from the extension **Incoming –** A call is ringing the extension

Table 30 – Edit Phone Settings and Descriptions

STEPS:

1 From the **Extensions➔Destination** page, find the extension you want to edit.

2 Click on the  icon to the right of the extensions name. The **Edit Phone Settings** page appears.

3 Make the necessary changes to the settings for the phone the click the **Save & Restart Phone** button.

4 The PBX will update the configuration file and reboot the phone so that it can download the updated information for the keys.

Edit Advance Phone Settings

The advanced settings page varies by the type of phone you are configuring. The following outlines the advance parameters for IPitomy phones, as well as Aastra phones.

Edit Advance Phone Settings – IPitomy Phones

Advanced Settings	
Custom Sip Settings	
Network	WAN (remote)
Server	72.64.129.45
Time Settings	
Network	Custom
Time Server:	us.ntp.org
Time Zone:	US/Eastern
Volume Controls	
Microphone Mute	☐
Auto-Answer Incoming Intercom	☑
Warn on Incoming Intercom	☑
Call Waiting Tone	☑
Stutter Tone	☐
Handset Gain Level	10
Headset Gain Level	10
Other Options	
Digit Timeout	4
Backlight	On
Backlight Time	10
Display Call Waiting	☑
Display Missed Calls	☑
Disable Phone Forwarding	☐
Display Softkey Dots	☐

Table 31 – Edit Advanced Phone Settings – IPitomy Phones

Sections/ Fields	Description
	Custom SIP Settings
Network	This defines whether the phone is communicating to the PBX over the LAN, WAN, or a Custom server address. **LAN** – Tells the phone to communicate to the **local** IP of the PBX **WAN** – Tell the phone to communicate to the **public** IP of the PBX, as entered under PBX Setup➔SIP **Custom** – Tells the phone to communicate to the PBX via an IP you enter manually
Server	Displays the IP of the PBX as configured from the previous setting. If **Network** is set to Custom, this is where you would manually set the IP.
	Time Settings
Network	Defines where the phone will get its time settings from. **PBX –** Sets the PBX local IP as the time server for the phone **Custom –** Allows you to manually enter a time server for the phone.
Time Server	Displays the address the phone is using for its time server. When set to **Custom** this field allows you to manually enter the time server address.
Time Zone	Allows you to set the time zone the phone will use. **System –** the phone follows the PBX time zone settings **Custom –** Select from the dropdown of a number of time zone settings, allowing the phone to display a different time then the PBX would
	Volume Controls
Microphone Mute	Default N; The IP550 phone will answer a page call and automatically engage Mute when this option is selected.
Auto-Answer Incoming Intercom	Default Y; If enabled, this will automatically answer incoming pages via speakerphone. If the user is on a call, the page will ring the extension instead.
Warn on Incoming Intercom	Default Y; If enabled the phone will play a tone alerting the user that their phone has answered a page with the speakerphone.
Call Waiting Tone	Default Y; If enabled the phone will play a tone while the extension is busy on a call alerting the user that another call is waiting.
Stutter Tone	Default N; Stutter Dial Tone is provided to alert a user that messages are waiting. It is presented to the user at the time the handset is taken off-hook or speakerphone button pressed from an idle state.
Handset Gain Level	Default 10; Gain can be loosely defined as volume. This setting allows the programmer to adjust the gain to the user's handset. DREW what is the range?
Headset Gain Level	Default 10; Gain can be loosely defined as volume. This setting allows the programmer to adjust the gain to the user's headset. DREW what is the range?

Other Options	
Digit Timeout	Default 4; This is a counter that determines how long the telephone will wait between digits dialed before sending digits dialed by the user to the PBX. The shorter the time the faster the call will be placed, BUT short times mean that a user has to enter all digits without delaying more than this time period or the call will be routed with fewer than required digits and may not route properly.
Backlight	Default N; This is the backlight of the LCD display. Setting this option to yes will force the LCD backlight to remain ON at all times.
Backlight Time	Default 10; When the Backlight option above is set to “N” (default) this timer determines how long the backlight will remain ON following the last activity.
Display Call Waiting	Default Y; The IP550 has the ability to display cal waiting information on the display of a busy extension. If this is desired set this to Yes.
Display Missed Calls	Default Y; The IP550 telephone has the ability to display the number of calls that rang but were not answered. If desired, select Yes.
Disable Phone Forwarding	Default N; This option is provided to disable telephone (phone-based) Call Forward. This is often desired since the user may invoke a mode of Call Forwarding not desired and conflicting with call forward conditions implemented in PBX programming. This is also beneficial in hospitality applications when a telephone is installed in a location where the user should not be given the ability to use cal forward.
Display Softkey Dots	Default N; The IP550 has been enhanced to provide an alignment “Dot” at the edge of the display for each of the Softkey function prompts. For users that find it difficult to associate Softkey prompts to Softkeys, enable this option.

Table 32 – Edit Advanced Phone Settings for IPitomy Phones

STEPS:

1. From the **Extensions➔Destination** page, find the extension you want to edit.

2. Click on the  icon to the right of the extensions name. The **Edit Phone Settings** page appears.

3. Click the **Advanced Settings** link at the bottom of the page.

4. Make the necessary changes to the settings for the phone the click the **Save & Restart Phone** button. This will update the configuration file and when the phone reboots, it will download the new information.

Edit Advance Phone Settings – Aastra Phones

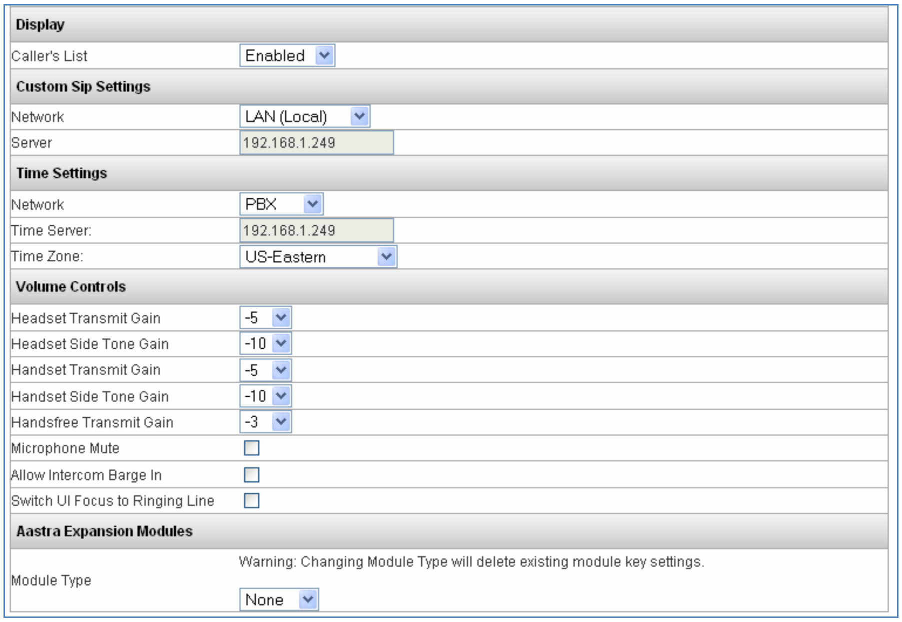


Figure 49 – Aastra Advanced Phone Settings Page

Sections/ Fields	Description
	Display Section
Caller's List	If Enabled, the Missed/Dialed/Answered calls on Aastra phones will display.
	Custom Sip Settings
Network	This defines whether the phone is communicating to the PBX over the LAN, WAN, or a Custom server address. **LAN** – Tells the phone to communicate to the local IP of the PBX **WAN** – Tell the phone to communicate to the public IP of the PBX, as entered under PBX Setup➔SIP **Custom** – Tells the phone to communicate to the PBX via an IP you enter manually
Server	Displays the IP of the PBX as configured from the previous setting. If **Network** is set to Custom, this is where you would manually set the IP.
	Time Settings
Network	Defines where the phone will get its time settings from. **PBX –** Sets the PBX local IP as the time server for the phone **Custom –** Allows you to manually enter a time server for the phone.
Time Server	Displays the address the phone is using for its time server. When set to **Custom** this field allows you to manually enter the time server address.

Time Zone	Allows you to set the time zone the phone will use. **System –** the phone follows the PBX time zone settings **Custom –** Select from the dropdown of a number of time zone settings, allowing the phone to display a different time then the PBX would
Volume Controls	
Headset Transmit Gain	Applies to Aastra Only - This allows you to set the gain on the transmission of calls made using a headset.
Headset Side Tone Gain	Applies to Aastra Only - This allows you to set the gain on for the headset side tone.
Handset Transmit Gain	Applies to Aastra Only - This allows you to set the gain on the transmission of calls made using the handset.
Handset Side Tone Gain	Applies to Aastra Only - This allows you to set the gain on the handset side tone.
Microphone Mute	This defines if the phone will answer a page in a muted state.
Allow Intercom Barge In	Applies to Aastra Only - If enabled, this allows a page to place your current active call on hold, and answers the page with the speaker phone.
Switch UI Focus to Ringing Line	Applies to Aastra Only - Aastra phones switch the UI Focus to the line ringing in.
Aastra Expansion Modules	
Module Type	This allows you to change the type of module for the phone. ! **IMPORTANT: Changing this parameter will delete the existing module key settings.**

Table 33 – Aastra Advanced Phone Settings and Descriptions

STEPS:

1. From the **Extensions➔Destination** page, find the extension you want to edit.

2. Click on the icon to the right of the extensions name. The **Edit Phone Settings** page appears.

3. Click on the **Advanced Settings** link. The Advanced Settings page for the phone appears.

4. Make the necessary changes to the settings for the phone the click the **Save & Restart Phone** button. This will update the configuration file and when the phone reboots, it will download the new information.

Groups

A Group is a logical grouping of extensions and their privileges. Each Group is associated with a set of PBX features and call routing strategies. It is advised to build the extensions in the PBX first, allowing the groups to be populated at the time of creation.

These Groups allow incoming calls to be distributed to a group of extensions rather than just one extension. Within the Groups, different distribution strategies may be selected based on the call coverage required by the application. Groups also provide off-hook paging functionality. By dialing the intercom button or code followed by the group number, extensions in the group will receive the page over the intercom. Groups are a set of extensions that are related either because they:

- Serve a similar business function.
- Work within the same department.
- Are located in proximity to each other.

For example, a business might create a few user Groups in an IP telephony network, e.g. Sales, Marketing, Administration, Accounting, Engineering, etc. Groups may also be created for people in similar locations like a plant floor, the north section of a building or the front office. The goal of a group is to ring telephones based on the incoming routing logic (e.g. DIDs, Auto Attendant selected by the incoming caller, or time of day). Note that calls routing to an extension from a ring group will ignore that phones PBX forwarding settings. Some phones have forwarding settings which are independent of the PBX (not set through the PBX Web GUI). These settings will still be applied.

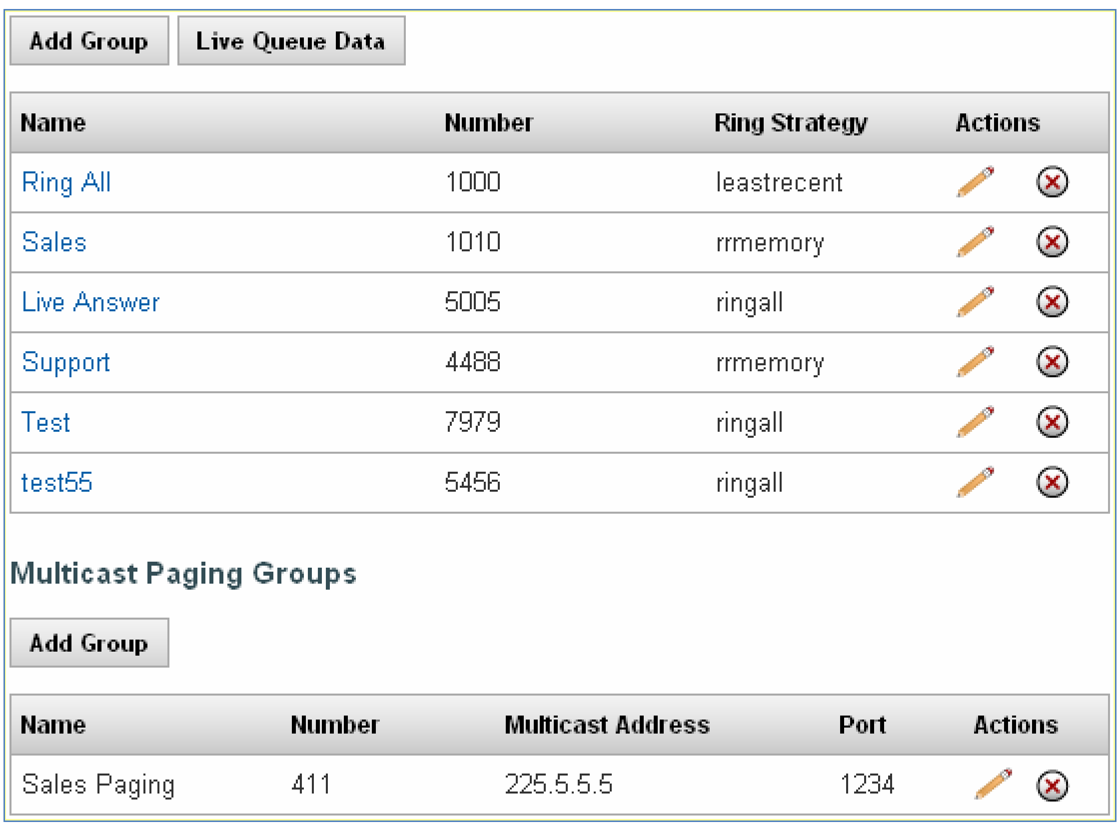


Figure 50 – Ring Groups Page

Ring Group Examples

Ring groups define a set of extensions (people) that answer calls. Ring groups can be created for departments (e.g. Sales, Engineering) or buisiness regions (e.g. north, south, etc), or areas of a business (e.g. a warehouse or plant floor). These ring groups can appear on an Automated Attendant (menu). When the group is selected from a Menu, the call is routed to the group. These calls will be distributed to the member extensions based upon the ring strategy for the group. The ring strategy for the group can be set from the drop-down list. Available call distribution options are:

- Ring All – Ring all extensions in the group simultaneously
- Round Robin – Take turns ringing each extension, starts at the first extension for each new call
- Round Robin (with memory) – Take turns ringing each extension, starts where the last call left off
- Least Recent – Ring extensions in order of which was least recently called by the queue
- Fewest Calls – Ring extension in order of which has the fewest completed calls from the queue
- Random – Randomly ring one extension at a time

Example Ring Group 1 – Departmental Grouping

Ring group definitions can be found in the **Add a New Ring Group** section of this guide. If a business has the following departments and people: This business can setup the following ring groups supporting business operations.

GROUP 1 - Sales	Ext.	GROUP 2 – Customer Services	Ext.
Cathy Caldwell	123	Gretchen Goodall	134
David Dawson	124	Peter Polk	135
Susan Smith	125		
Robert Reed	126		

Using this ring group scenario the **Menu would look like**:

- Sales (Destination - Group 1).
- Customer Service (Destination - Group 2).
- Office Manager (Destination - Ext. 113).

The menu prompt (message) for this menu and group arrangement would read as follows:

"Thank you for calling CompanyA, a leading manufacturer of cable assemblies and wiring harnesses. If you know the extension of the party you would like to reach you may dial it at any time."

- For Sales, press 1.
- For Customer Service, press 2.
- For Accounting, press 3.

Once a call is sent to Sales, member extension will be rang according to the **ring group strategy.**

Example Ring Group 2 – Regional Sales Grouping

GROUP 1 East Coast Sales	Ext.	GROUP 2 -West Coast Sales	Ext.	GROUP 3 Customer Service	Ext.

Cathy Caldwell	123	Susan Smith	125	Gretchen Goodall	134
David Dawson	124	Robert Reed	126	Peter Polk	135

Using this ring group scenario the **Menu would look like**:

- East Coast Sales (Destination - Group 1).
- West Coast Sales (Destination - Group 2).
- Customer Service (Destination - Group 3).
- Office Manager (Destination - Ext. 113).

The menu **prompt** for this menu and group arrangement would read as follows:

"Thank you for calling CompanyA, a leading manufacturer of cable assemblies and wiring harnesses. If you know the extension of the party you would like to reach you may dial it at any time. For"

- East Coast Sales, press 1.
- West Coast Sales, press 2.
- Customer Service, press 3.
- Accounting, press 4.

Calls sent to these groups might use different ring strategies. The East Coast Sales group might answer calls Round Robin, distributing the calls to each Sale Representative consecutively. If a sales person is missing from the West Coast team this group might set phones to Ring All in the group so that calls don't get missed. The Customer Service team may get high volumes of calls during a specified period of time, so this group may be set to ring Least Recent.

Add/Edit New Ring Group

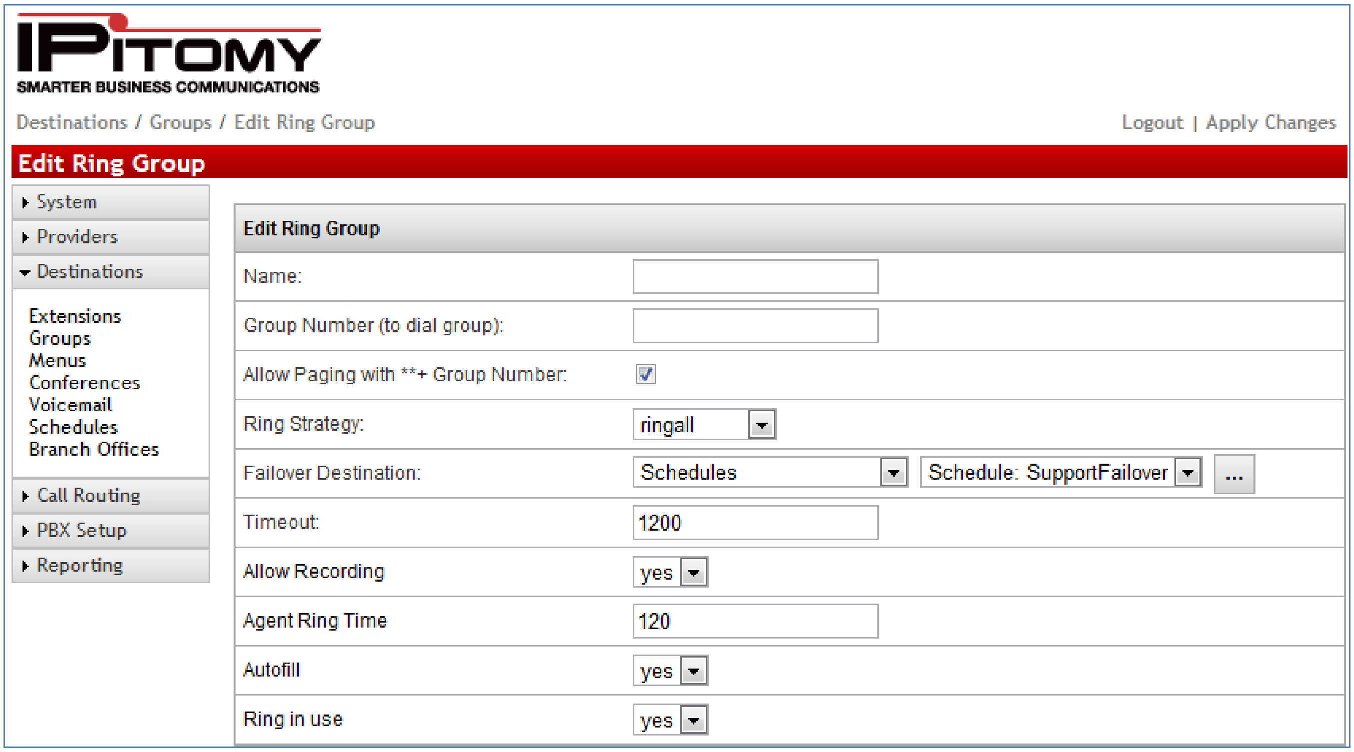


Figure 51 – Edit Ring Group Page

The following table describes the parameters (fields) on the Edit Ring Group page and their descriptions and recommended settings.

Sections/Fields	Description
Name	The name associated with this ring group.
Group Number (to dial group)	The number dialed to access this ring group. Must be 3 or 4 digits.
Allow Paging with **+Group Number	If enabled (checked), this will allow ****+** group number to page this ring group

Ring Strategy	This defines how calls are distributed to member extensions. Options are: **Ring All** – Ring all extensions in the group simultaneously **Round Robin** – Ring extensions consecutively. Delivers a new call to the first person in the group only after the last person in the group has taken a call. If an extension is busy the call will automatically be routed to the next extension in the group. **Round Robin (with memory)** – Ring extensions consecutively. Delivers a new call to the first person in the group only after the last person in the group has taken a call. Remembers where the last call was taken and distributes new calls to the next extension in the rotation. **Least Recent** – Ring extensions in order of which was least recently called by the queue. **Fewest Calls** – Ring extension in order of which has the fewest completed calls from the queue **Random** – Distributes calls randomly to the group.
Failover Destination	This defines where a call will be sent when it goes unanswered for a duration that matches the Timeout for the group.
Timeout	Defines, in seconds, the total time a call will spend in the ring group before being sent to the Failover Destination
Allow Recording	Determines if feature code recording is allowed for calls in this queue. If **YES** = enables the recording feature. If **NO** = disables the recording feature.
Agent Ring Time	Defines, in seconds, how long the queue will ring a particular agent. If using the Ring All strategy, this value should match the Timeout. For other strategies, this should be set to how long an extension should ring before moving on to the next.
Autofill	This deals with how the queue handles multiple concurrent calls. If set to **NO** the queue will wait until the first call is answered before ringing phones with the second call. If set to **YES** the queue will ring phones for all calls as they come in.
Ring In use	If **YES** = Distribute calls to group members if they are already on a call If **NO** = Do NOT distribute calls to group members if they are already on a call

Table 34 – Edit Ring Group Settings and Descriptions

The following outlines the steps to add/edit ring group settings.

STEPS:

1 From the **Destinations➔Group** page, click the **ADD Group** button. The **Edit Ring Group** page appears.

2 In the **Name** field, enter a name for the group.

3 In the **Group Number** field, enter a number for the group. This number must be three or four digits in length.

4 Check the **Allow Paging with** **+ **the Group Number** to enable the setting.

5 Select the desired **Ring Strategy** from the drop-down list.

6 Select the desired **Failover Destination** from the list of available options.

7 Enter the time for the **Timeout** parameter.

8 Select “**Yes**” if you want to enable the **Allow Recording** setting. Otherwise, set it to “**No**”.

9 Enter the time for the **Agent Ring Time** parameter.

10 Select “**Yes**” if you want to enable the **Autofill** setting. Otherwise, set it to “**No**”

11 Select “**Yes**” if you want to enable the **Ring in use** setting. Otherwise, set it to “**No**”

12 Click the 

button.

13 From the **Members** section of the **Edit Ring Group** page, select the extension number from the list of extensions that you want to add to the ring group. Click the **ADD** button.

14 A second list appears next to the Members list displaying the selected extension. Repeat step 13 until all extensions that you want to add are on the list. Additionally, Shift or Control click functionality works for adding multiple extensions at once.

15 To remove an item from the list of additions, select the extension number then click the **REMOVE** button.

16 Define **Priorities** for the extensions, if needed. The scale ranges 0 thru 9 (0 is the highest priority).

17 Click 

button to save the changes.

18 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Ring Group Advanced Settings

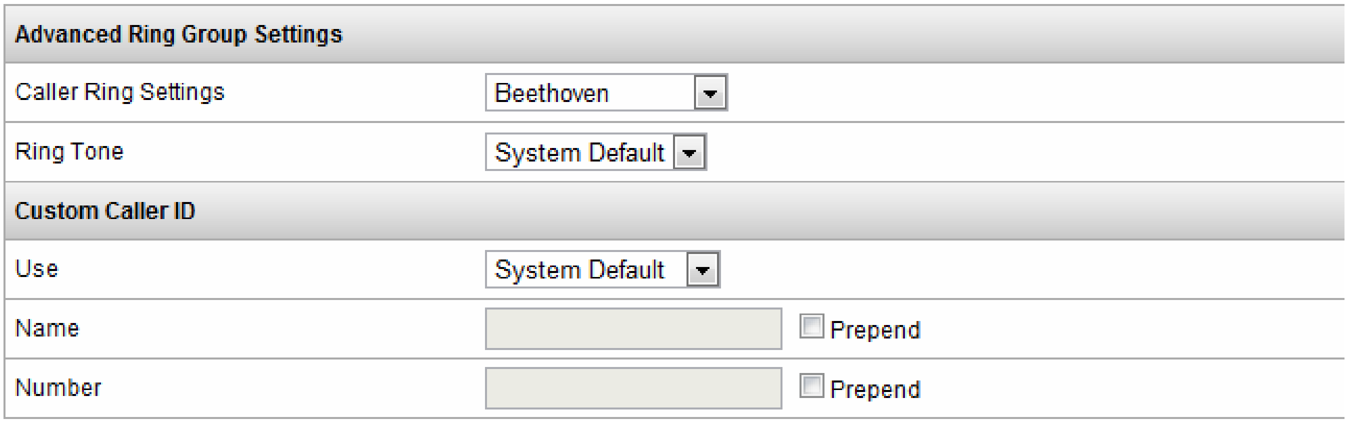


Figure 52 – Advanced Ring Group and Custom Caller ID Settings Section

The following table describes the parameters (fields) on the **Advanced Settings** for Ring Groups page and their descriptions and recommended settings.

Sections/Fields	Description
	Advanced Ring Group Settings
Caller Ring Settings	Defines what a caller will hear while they are waiting for someone to pick up a call from the call group. Callers can either hear: **Ringing** – The phone continues to ring while the caller is waiting. **Music on Hold** – The caller hears music while waiting for a group member to pick up the call. The dropdown list displays the MOH playlists that are on the PBX. Select a playlist and it will play the sound files found in the playlist.
Queue Dial String	**OBSOLETE** – and will be removed in a future release.
Ring Tone	When **Caller Ring Settings** are set to Ringing, use this to define the ring cadence.
	Custom Caller ID Settings
Use	Defines what method, if any, you wish to use to override Inbound CID. The options are: System Default, Name, Number, Name and Number
Name	This is the name of the caller that will be displayed when a call is received.
Number	This is the number of the caller that will be displayed when a call is received.
Prepend	Selecting Prepend will enter the text you put in the field, followed by the original CID. If not prepending, the override eliminates the original information.

Table 35 – Advanced Ring Group and Custom Caller ID Settings and Descriptions

Edit Advanced Ring Group Settings

STEPS:

1. Select **Destinations➔Groups**. The **Ring Groups** page appears.
2. Select the Ring Group you want to edit by either select the icon to the right of the group name or click on the blue Group Name link.
3. The **Edit Ring Group** page appears. Scroll down the page and click on the **Advanced** link. The **Advanced Ring Group Settings** section is displayed.
4. Make the necessary changes to the settings then click Save Changes button to save the changes.
5. Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Edit Custom Caller ID Settings

STEPS:

1. Select **Destinations➔Groups**. The **Ring Groups** page appears.
2. Select the Ring Group you want to edit by either select the icon to the right of the group name or click on the blue Group Name link.
3. The **Edit Ring Group** page appears. Scroll down the page and click on the **Advanced** link. **The Custom Caller ID** section is displayed.
4. Make the necessary changes to the settings then click Save Changes button to save the changes.
5. Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Edit Members/Agents

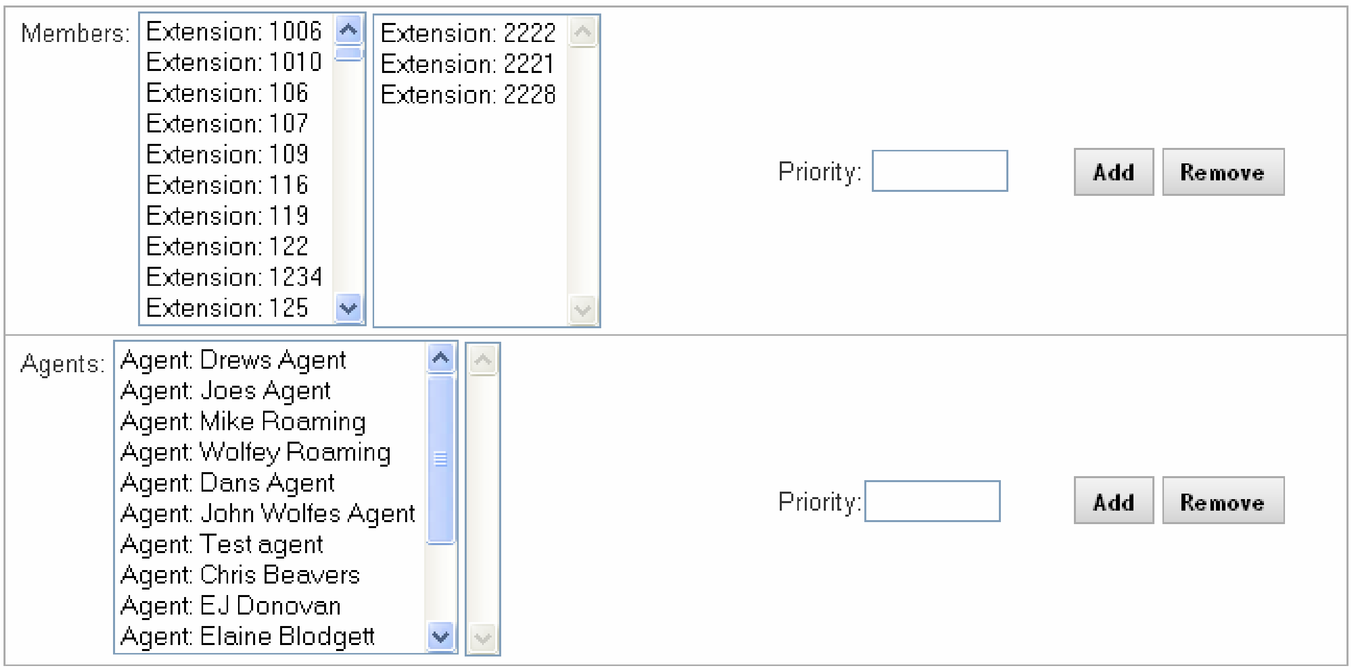


Figure 53 – Edit Members/Agents Section

Sections/Fields	Description
Members	This is the list of available extensions that are in the database.
Agents	This is the list of available agents that are in the database.
Priority	This field allows you to set the priority level for the members or agents. The lower the number, the higher the priority.
ADD	This button will add the select item on the list to the right of the available members or agents. Shift and Control click functionality works for adding more than one at a time.
REMOVE	This button will remove the item(s) that have been selected from the list of available members or agents. Shift and Control click functionality works for adding more than one at a time.

Table 36 – Edit Members/Agents Settings and Descriptions

Add Agents/Members to the Group

STEPS:

1 Navigate to the **Destinations➔Groups➔Edit Group** page.

2 Select from the left hand column which agents/members to add to the group.

3 Click the **Add** button.

4 Click 

button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Delete Agents/Members from the Group

STEPS:

1 Navigate to the **Destinations➔Groups➔Edit Group** page.

2 Select from the right hand column which agents/members to remove from the group.

3 Click the **Add** button.

4 Click 

button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Automatic Call Distribution (ACD)

Systems licensed to use ACD will have access to additional functionality pertaining to Ring Groups. ACD allows for more flexibility and control over the queuing in a Ring Group, as well as allowing the creation of Agents who can be added to the Ring Group.

ACD Settings - Agents			
Agent Retry Timer	1		
Weight	0		
Wrap-up Time	1		
Autopause	no		
Maximum # of people in queue	0		
Announce Frequency	5		
Periodic Announce Frequency	0		
Announce Holdtime	no		
Join Empty Queue	yes		
Leave Empty Queue	no		
Report Hold Time	no		
Member Delay before connect	0		
Timeout Restart	yes		
Service Level	10		
Exit Status: FULL	Ring Groups	Ring Group: Sales	...
Exit Status: JOINEMPTY	Voicemail	Voicemail: 5001	...
Exit Status: LEAVEEMPTY	Voicemail	Voicemail: 5001	...
Exit Status: JOINUNAVAIL	Voicemail	Voicemail: 5001	...
Exit Status: LEAVEUNAVAIL	Extensions	Extension: 5001	...
Intro Announcement	None		
Agent Announcement	None		
Periodic Announcement	None		
Exit Menu	none		

Advanced

Table 37 – Automatic Call Distribution (Agents) Page

Sections/Fields	Description
Agent Retry Timer	This defines, in seconds, how long the queue will wait before attempting to ring an agent again.
Weight	The priority of the queue in relation to other queues in the system.
Wrap-up Time	This defines, in seconds, how long the queue will wait after an agent has ended a call before flagging then as an active agent ready to accept more calls.
Autopause	This will place an agent in a pause state in relation to this queue when they ignore or answer a call. To become an active member of the queue again unpause using the 1* feature code a phone or 1*+agent# for an agent.
Maximum # of people in queue	Defines how many callers can join the queue before it is considered full.
Announce Frequency	Must be set for Announce Holdtime to function. This pertains to how often the caller will hear the hold time announcement, in seconds
Periodic Announce Frequency	This defines how much time passes between playing the recording that was set for this queue under Periodic Announcement.
Announce Holdtime	If set to **YES** this will announce a caller's status in the queue with regards to their expected hold time, pending the hold time is estimated to be over 1 minute. If the hold time is less than one minute, this will announce the caller's position in the queue. A period of time is needed for the algorithm to 'learn' and estimate hold times accurately. This message will not play for the first caller in the queue. Caller Ring Settings must be set to a MOH file in order for this message to play.
Join Empty Queue	Controls if a caller can join a queue. **YES** = Callers can join the queue with no members or whose members are busy. **NO** = Ignores agent and member status, they can be logged out as long as they exist in the configuration callers can join a queue. **STRICT** = Callers cannot join the queue if there are no members or if the members are all busy, paused, or logged off.
Leave Empty Queue	Controls if a caller will leave an empty queue. **YES** = Callers will leave the queue if there are no active members, or all members are busy. If all agents are logged out, the caller will not leave the queue. **NO** = Callers will not leave the queue even if there are no active members, or all members are busy. **STRICT** = Callers will leave the queue if there are no active members, but will remain in the queue if all members are busy. If agents are pause or logged off, the caller will leave the queue.
Report Hold Time	Set to **YES** if you want the person answering the queue call to hear a message indicating how long the call they just answered was in the queue

Member Delay before connect	Delay, in seconds, before a member is connected to the queue.
Timeout Restart	If set to **YES** the ring group timeout will reset after attempting to ring an agent/member. When set to **NO**, time taken to ring an agent/member will be subtracted from the ring group time, and when the timer ends the group will send the caller to the Failover Destination
Service Level	Set this to the time, in seconds, that defines the acceptable level of service for the queue. The Live Queue Data page will track the percentage of calls answered within the allotted time.
Exit Status: FULL ***Exit Status: JOINEMPTY*** ***Exit Status: LEAVEEMPTY*** ***Exit Status: JOINUNAVAIL*** ***Exit Status: LEAVEUNAVAIL***	These are the destinations where calls will go based on the exit strategy defined. If no one in the group accepts the call this is the destination to which the call will be sent. Destinations can be extensions, other ring groups, a menu or a voicemail box.
Intro Announcement	Define what prompt to play to the caller just before they enter the queue. (Previously "Announce Filename")
Agent Announcement	Defines what prompt to play to the agent/member who answers the queue call. This setting is set per Queue and therefore can be used to identify what Queue is being answered (for agents handling calls from multiple Queues.
Periodic Announcement	Defines what prompt to play to the caller at the Periodic Announce frequency interval. This will not play for the caller in the first position of the queue. Caller Ring Settings must be set to a MOH file in order for this message to play.
Exit Menu	Set this to a menu. Key presses while in this group will send people the corresponding destination from that menu.

Table 38 – Edit Ring Group ACD Agents Settings and Descriptions

Edit Automatic Call Distribution (ACD) Settings

STEPS:

1 From the **Destinations→Groups→Edit Group** page, scroll to the **ACD Settings** section.

2 Make the necessary updates to the ACD settings.

3 Click Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Add Automatic Call Distribution (ACD) Agents

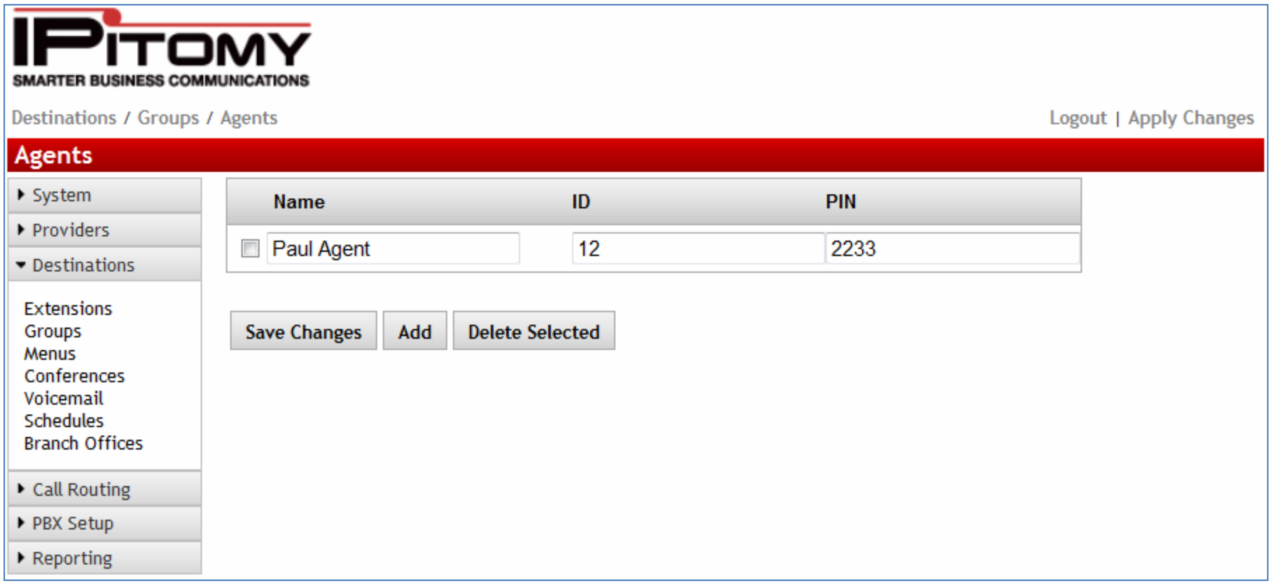


Figure 54 – ACD Edit Agents Page

Sections/Fields	Description
Name	Enter a name that will identify this agent.
ID	Enter a unique 3-4 digit number to identify this agent. This number is used to log in/out.
PIN	Enter a minimum of 4 digits for the PIN. This is the password used to log in/out with the agent.

Table 39 – Add/Edit Agent Features and Descriptions

STEPS:

1 From the **Destinations➔Groups➔Edit Group** page, scroll to the **ACD Settings** section.

2 Click on the blue **Agents** link to the right of the words **ACD Settings**. The Agents page appears.

3 Click on the **ADD** button to add a new agent. A box with the name, ID and PIN field appears. Enter the information for the new agent.

4 Click 

button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Edit Automatic Call Distribution (ACD) Agent

STEPS:

1 From the **Destinations➔Groups➔Edit Group** page, scroll to the **ACD Settings** section.

2 Click on the blue **Agents** link to the right of the words **ACD Settings**. The **Agents** page appears.

3 Update the necessary information for agent(s) then click the **SAVE CHANGES** button. The changes that you made will appear in the updated list.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Delete Automatic Call Distribution (ACD) Agent

STEPS:

1 From the **Destinations➔Groups➔Edit Group** page, scroll to the **ACD Settings** section.

2 Click on the blue **Agents** link to the right of the words **ACD Settings**. The **Agents** page appears.

3 From the list of existing agents, click on the box (placing a checkmark) to the left of the Agent(s) that you want to delete. Click the **DELETE SELECTED** button.

4 The agent selected is removed from the list.

5 Click 

button to save the changes.

6 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Multicast Paging Group

A Multicast Paging Group provides a way to page a large number of phones, while keeping the burden on the network and PBX to a minimum.

NOTE: ***IPitomy recommends using multicast paging for groups of 10 or more phones that are paged frequently.***

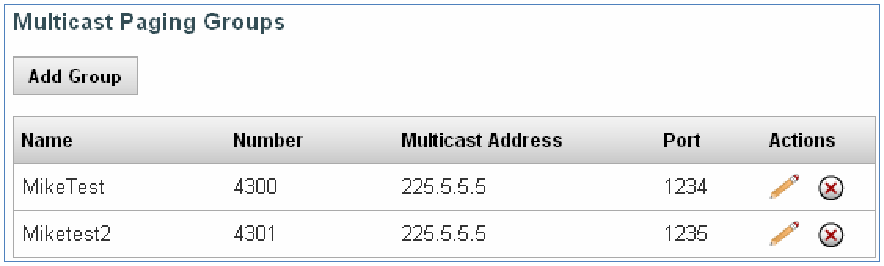


Figure 55 – Multicast Paging Groups Page

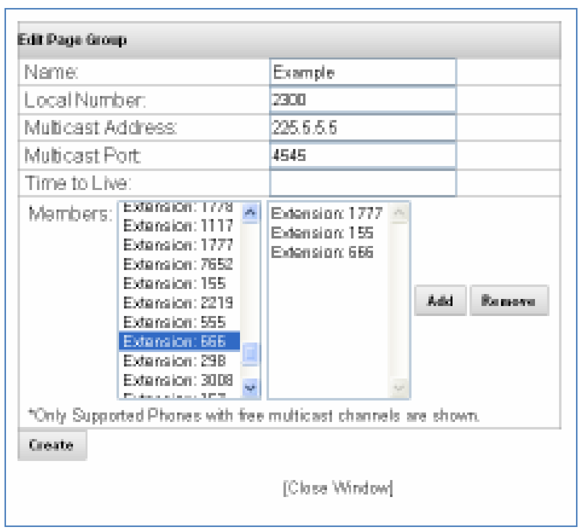


Figure 56 – Edit Page Group Window

The following table describes the fields (parameters) on the Multicast Edit Paging Group window and the recommended settings.

Sections/Fields	Description
Name	This is a name of the multicast paging group.
Local Number	This is the number that will be dialed to launch the multicast paging to the defined group. This is the unique 3-4 digit number.
Multicast Address	This should be a valid multicast IP for a Multicast Address. We recommend **225.5.5.5**. You can use the same IP for each group, so long as you assign them different ports.
Multicast Port	This must be a valid, unused port Multicast Port. ***NOTE: This must be a port below 6535.***
Time to Live	Time to Live is a counter that decreases by 1 each time a packet passes through a router. When the counter reaches 0, the packet is considered dead and will not pass to any other networks. This can be left blank or configured as needed for your particular install. ***NOTE: Leaving it blank the packet will not pass to another network.***
Members	The Members section allows you to add the extensions to be paged within the group when a page is sent.
Add/Remove	The Add and Remove button allows you to add or remove members (extensions) from the paging group.
Create/Update	Depending on the process that is being performed, this button will appear (be labeled) as either Create or Update. This will create a new member or process the changes made.
Close Window	This link will close the Edit Page Group window and return you to the Ring Groups page.

Table 40 – Multicast Paging Settings and Descriptions

The following section outlines the steps to add a new Multicast Paging Group.

Add Multicast Paging Group

STEPS:

1. Navigate to the **Destinations→Group** page
2. Locate to the **Multicast Paging Groups** section (at the bottom of the page).
3. Click on the **Add Groups** button. The **Edit Page Group** page appears.
4. Enter the necessary information for the new paging group.
5. From the **Members** section of the **Edit Page Group** window, select the members (extensions) that you want to include in the paging group.

Use the CTRL or SHIFT button to select multiple or a range of extensions from the list.

6 Click on the **ADD** button to add the selected members to the group.

7 Click the **CREATE** button to save the changes. The “Group Created” message appears. Close the window (click on the “X”). The system returns you to the **Ring Groups** page.

8 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

9 **Re-boot** the phones that correspond to extensions that were added to the **Multicast Paging Group**. Once all the phones have been re-booted, you will be able to utilize this feature by simply dialing the Local Number that was just created.

Edit Multicast Paging Group

STEPS:

1 Navigate to the **Destinations→Group** page

2 Locate the **Multicast Paging Groups** section (at the bottom of the page).

3 Click on icon to the right of the **Name** of the paging group you want to update. The **Edit Page Group** page appears.

4 Enter the necessary information for the paging group.

5 Click the **UPDATE** button to save the changes. Close the window (click on the “X”). The system returns you to the **Ring Groups** page.

6 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

7 **Re-boot** the phones for all the extensions that were involved in the Multicast Paging Group.

Delete Multicast Paging Group

STEPS:

1 Navigate to the **Destinations→Group** page

2 Locate the **Multicast Paging Groups** section (at the bottom of the page).

3 Click on icon to the right of the **Name** of the paging group you want to delete. The group is removed from the **Multicast Paging Group** page.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

5 **Re-boot** the phones for all the extensions that were involved in the Multicast Paging Group which was deleted.

Live Queue Data

The Live Queue Data page displays current call statistics for ACD queues. The page will refresh every 10 seconds, allowing a manager to monitor the activity in the queue as it happens.

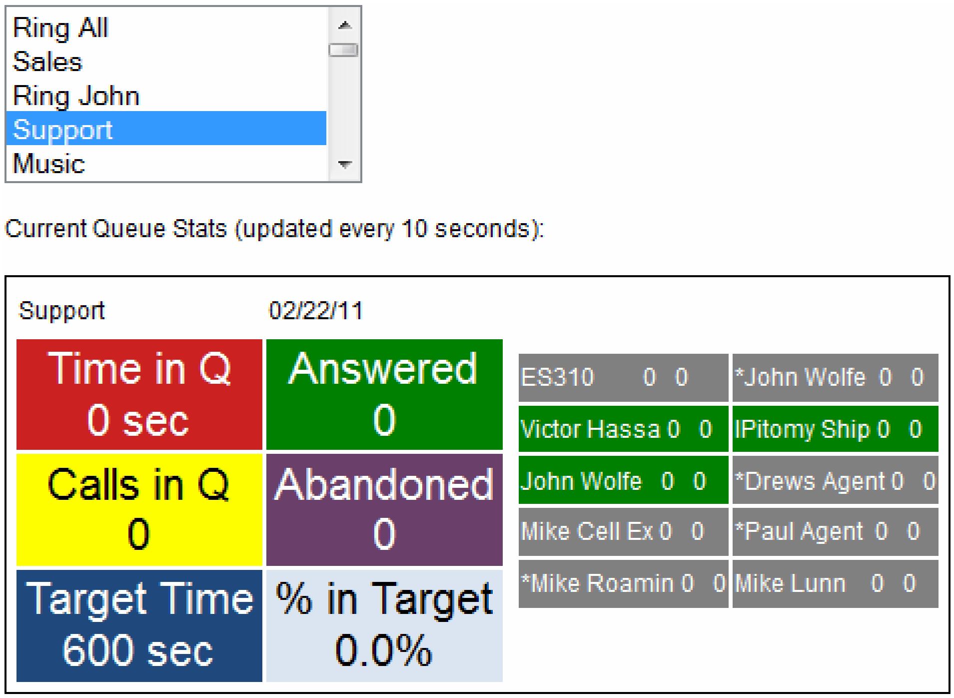


Figure 57 – Ring Group Live Queue Data

Sections/Fields	Description
<queue selection box>	In the queue selection box the Queue or Queues to be displayed can be selected. Use the Control key (Command on Mac) to select multiple queues for display.
<queue statistics box>	The Queue Statistics box has been reformatted in SW release 3.4.1 to allow its use on call center wall boards (large PC displays). Each queue is encapsulated with a thin-line box. Queue statistics are color-coded for quick orientation on wall boards. Since this data is presented in GUI format it can be easily resized to accommodate large displays. The top line of the queue display presents the Queue Name and the date. Statistics are a running total and will reset to zero whenever Apply Changes is clicked (as is the case of the screen capture above).
Time in Q	This is the average time that callers have waited in queue before being answered by a queue member/agent.
Calls in Q	This is the number of calls that are waiting in the queue.
Answered	This is the number of calls answered by members/agent of this queue.
Abandoned	This is the number of calls that have not been answered and the caller

	abandoned the queue.
Target Time	Displays the service level set for this ACD queue. The Service Level is the arbitrary time allocated by the queue administrator which is set as a gauge by which all calls should be answered.
% in Target	This is the percentage of answered calls measured against the service level. (The percentage of calls being answered within the set Target Time.)
<Member/Agent field>	At the right side of the Queue Statistics box the agents and queue members are listed. There are two maximum columns and the window will resize to accommodate all members and agents registered to this queue. Member/Agent status is available at a glance with color-codes as follows: **Red** = In Use **Green** = Idle (Logged In) **Yellow** = Ringing **Gray** = Unavailable (Not Logged In) **Orange** = Paused Also in the Member/Agent data are two numeral fields. The first number represents the Priority of this Member/Agent in this queue. The second number is the total calls taken by this Member/Agent.

Table 41 – Live Queue Fields and Descriptions

Live Queue Data – Wallboard

Software version 3.4.1 and above allows the use of call center wall boards with Live Call Queue. This function is facilitated using a special user name and password via Smart Personal Console.

Activate Live Call Queue Wallboard

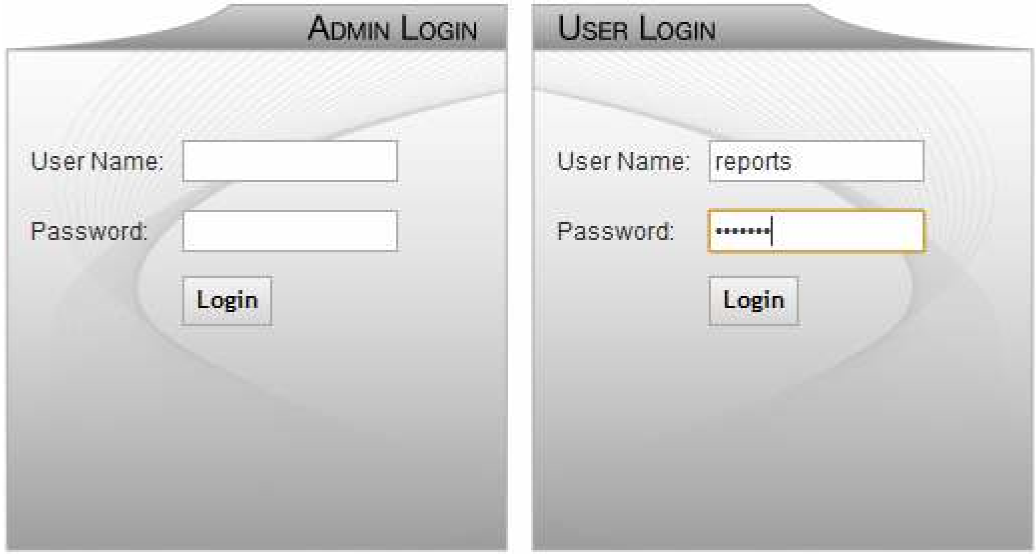
STEPS:

1. Navigate to the **PBX Setup→General** page
2. Locate to the **General Settings** section (second section from the top).
3. Locate the SPC Reports User Password field (bottom of section)

   SPC Reports User Password | display

4. It may be helpful to change this to a user-friendly password. If you change the password be sure to Save Changes and Apply Changes.
5. Make a note of this password.
6. On the PC that will be used for the wallboard display, open a web browser and navigate to the IPitomy PBX login screen

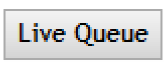
7 Using the right side – Smart Personal Console (User Login) input the word “reports” as the user name and the password saved in step 3.



8 Click on 

.

9 The CDR-Report page is opened. On this page is a button to select Live Queue.

10 Click on 

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11 The same display presented in PBX Administration is displayed. Simply select the group or groups that you wish to display on the wallboard.

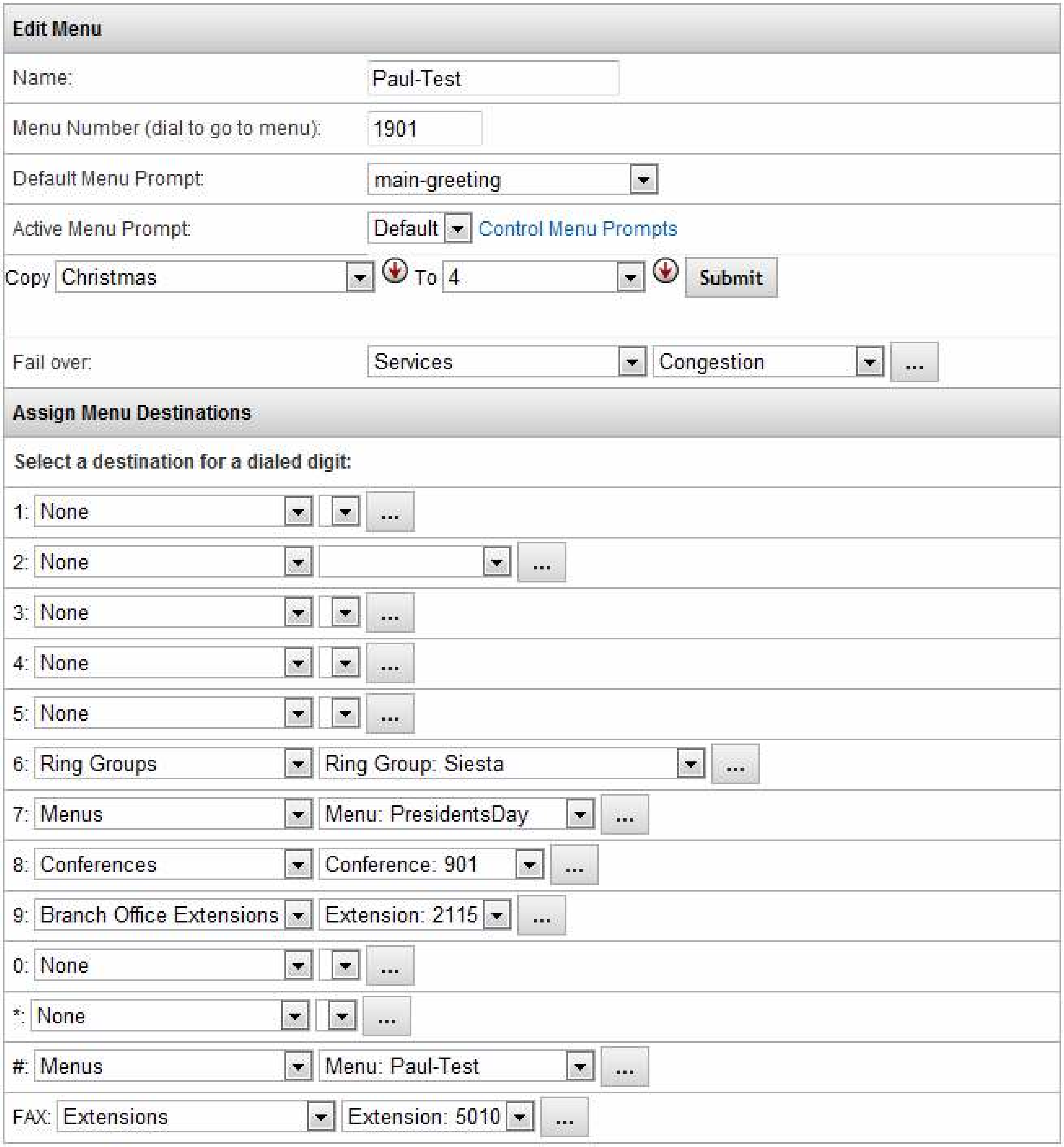
12 Size this web page data output for the monitor in use by pressing and holding the Cntrl key (PC, Command key on Mac) and press “+” or “-“ until the data is sized as you wish.

Menus

Menus direct callers to destinations within a business. A menu can route a caller to a destination once a number on a key-pad has been pressed. The menu prompt tells a caller what number to press to get to a desired destination. Menus can also be used to provide information to callers like driving directions, etc.

Figure 58 – Menus Page

Edit Menu



Edit Menu	
Name:	Paul-Test
Menu Number (dial to go to menu):	1901
Default Menu Prompt:	main-greeting
Active Menu Prompt:	Default Control Menu Prompts
Copy Christmas To 4 Submit	
Fail over:	Services / Congestion / ...
Assign Menu Destinations	
Select a destination for a dialed digit:	
1:	None ...
2:	None ...
3:	None ...
4:	None ...
5:	None ...
6:	Ring Groups / Ring Group: Siesta ...
7:	Menus / Menu: PresidentsDay ...
8:	Conferences / Conference: 901 ...
9:	Branch Office Extensions / Extension: 2115 ...
0:	None ...
*:	None ...
#:	Menus / Menu: Paul-Test ...
FAX:	Extensions / Extension: 5010 ...

Figure 59 – Edit Menus Page (with Control Menu Prompts open)

Sections/Fields	Description
Name	Enter a name for the menu.
Menu Number (dial to go to menu)	This is the number of the menu. This number is used to access the menu from extensions – if necessary. The number must be a unique 3 or 4 digit number.
Default Menu Prompt	This is the recording that is active for callers who reach this menu when the Menu Prompt is set to default.
Active Menu Prompt	This function is enhanced in version 3.4.1. (Previously "Menu Prompt".) As it indicates, this selection determines what prompt recording is heard by callers who reach this menu. Record this first under PBX Setup➔Prompts.
Control Menu Prompt: Copy … To …	The Copy… To… function allows you to place pre-recorded (and previously uploaded, see: PBX Setup➔Prompts) prompts into the available Prompts to be associated to this menu. Available "slots" are 1-5 where the user may then select 0-5, 0 being the default and 1-5 being those that are loaded. The user also has the ability to rerecord the prompts 1-5 via DTMF Administration access. Note: uploaded prompts are "copied" to the menu; therefore rerecorded prompts do not overwrite the original uploaded file.
Fail Over	This is the destination that the menu will fail over to if the user does not press a digit in the allotted Response Time.
Assign Menu Destinations	
Menu Destinations	This list represents the destinations callers are routed to when pressing 0-9,*, #, and FAX. If a destination is set for FAX, the PBX will detect if a call inbound over an Analog or T1/PRI card is a fax while the prompt is playing, and will route calls to this destination if fax tone is detected.

Table 42 – Add/Edit Settings and Descriptions

Add Menu Settings

STEPS:

1. From the **Destinations➔Menus** page, click the **ADD** menu button. The **Edit Menu** page appears.
2. Enter the information for the new Menu item and make necessary destination assignments.
3. Click 

button to save the changes.
4. Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Edit Menu Settings

STEPS:

1. From the **Destinations➔Menus** page, locate the name of the Menu item you want to edit.
2. Click on icon to the right of the **Name** of the menu you want to update. The **Edit Menu** page appears.
3. Make the necessary updates to the menu setting then click button.
4. Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Delete Menu Settings

STEPS:

1. From the **Destinations➔Menus** page, locate the name of the Menu item you want to edit.
2. Click on icon to the right of the **Name** of the menu you want to delete. The Menu name is removed from the Menu listing.
3. Click button.
4. Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Advanced Menu Settings

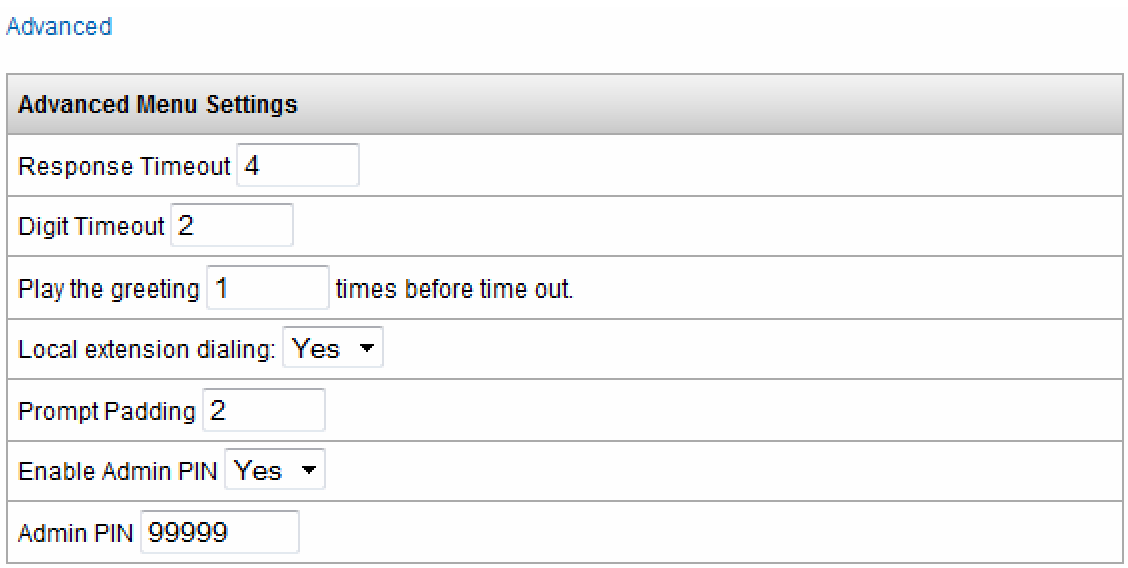


Figure 60 – Advanced Menu Settings Page

Sections/Fields	Description
Response Timeout	This is the number of seconds (0-60) the PBX will wait for digits to be dialed. If nothing is dialed in the allotted time, the menu will route the call to the Failover Destination. The minimum and default value is **4**.
Digit Timeout	While entering digits, the PBX will wait this many seconds (1-20) after the last digit was entered to take any actions. Single decimal digits are permitted. The default value is **2**.
Play the Greeting	This is the number of times the greeting will be played before the call is sent to the fail over destination. Default and minimum value is **1**.
Local extension dialing	This indicates whether you are able to dial local extensions from the menu. Default value is **YES.** Note: It is not necessary to designate leading digits for extensions. This setting alone indicates when digits collected should be compared to the available list of extensions.
Prompt Padding	This is the time in seconds that the PBX will wait when entering the menu before playing the prompt. The default is set to **2** seconds.
Enable Admin PIN	This enables the use of DTMF Menu Prompt Administration. When enabled the user may access the menu from a DTMF-enabled device and input the PIN to enter into Menu Prompt Administration Default value is **No**.
Admin PIN	This is the PIN used to access DTMF Menu Prompt Administration. Default value is: <empty>.

Table 43 – Advanced Settings and Description

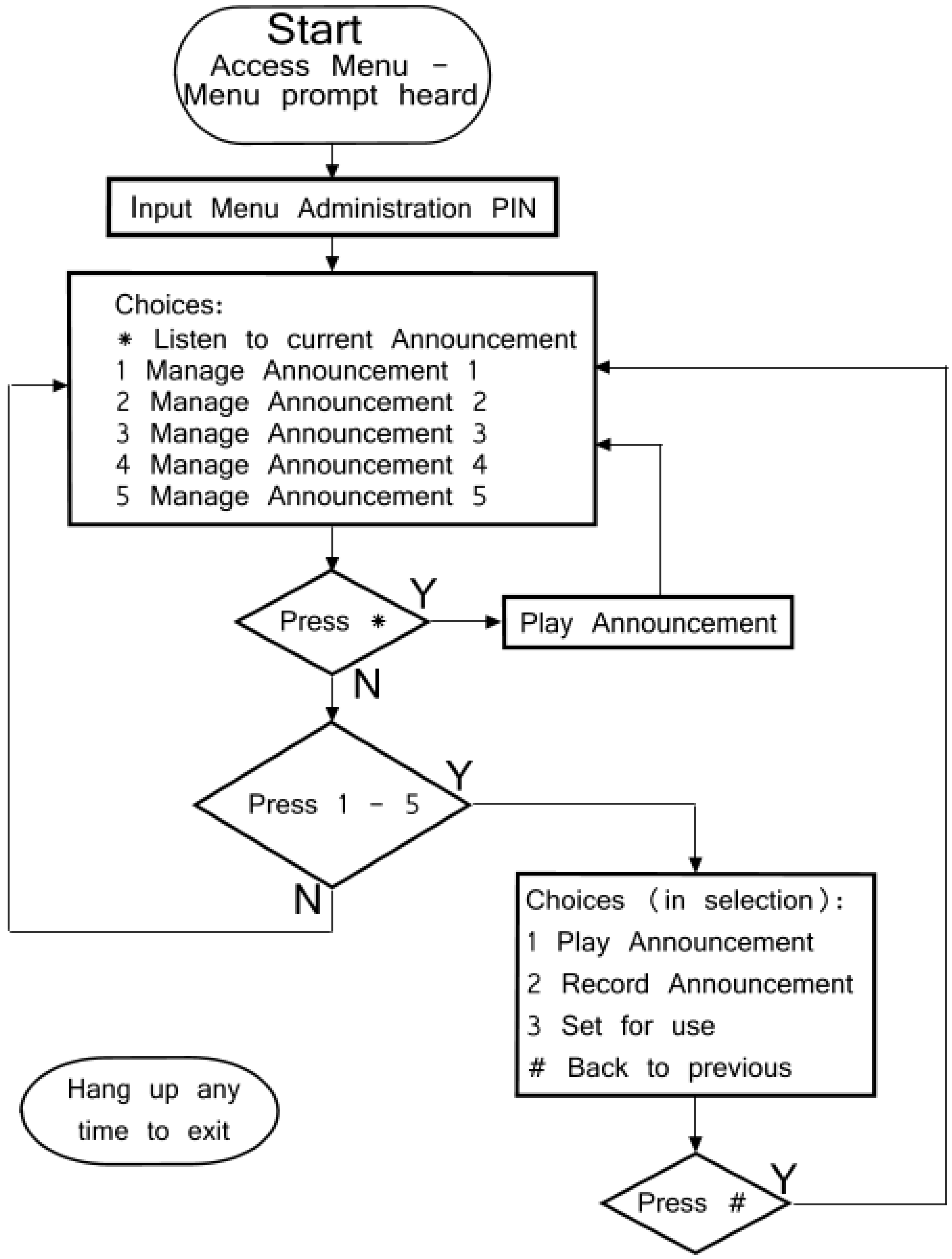


Figure 61 – Remote Menu Announcement DTMF Admin Flow

Conferences

The Conference Bridge allows groups of callers (large and small) to participate in the same call. An internal extension can dial the number direct to a conference; and outside callers can be routed to a conference by: being transferred from an existing call, being routed direct via a DID, or selecting an option from a menu. Each conference can be configured to require a PIN for entrance, or not.

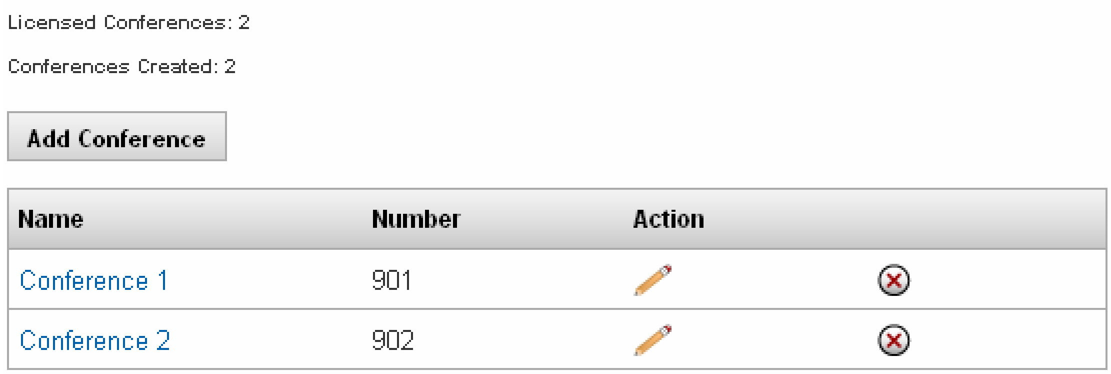


Figure 62 – Conferences Page

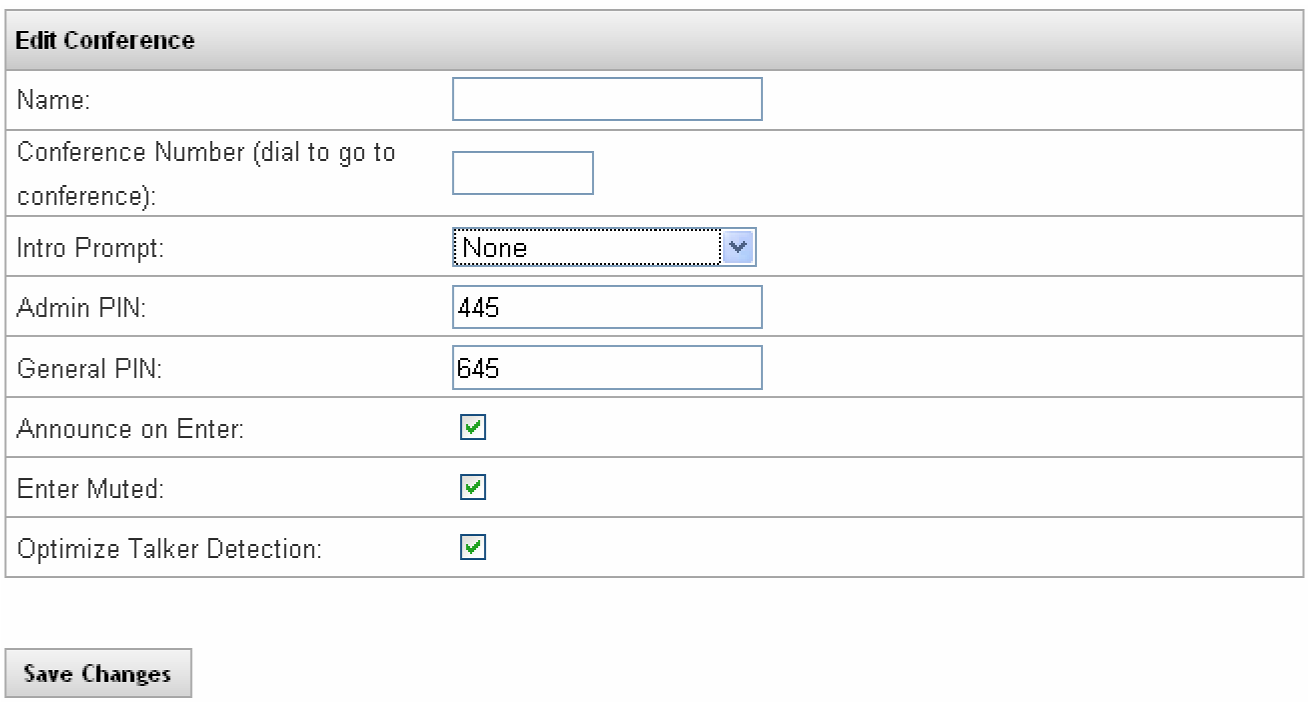


Figure 63 - Edit Conference

Sections/Fields	Description
Name	Enter a name to distinguish this conference from others.
Conference Number	Enter a unique 3-4 digits number for the conference.
Intro Prompt	If desired, a Prompt file can be played when entering the conference.
Admin PIN	This is the Administrator personal identification number (PIN). This is a **3-4 character numeric** field.
General PIN	This is the User personal identification number (PIN). This is a **3-4 character numeric** field.
Announce on Enter	If Enabled (checked), this prompts the user to record their name before entering the conference.
Enter Muted	If Enabled (checked), users enter the conference in muted mode. The user can dial *1 to un-mute the themselves.
Optimize Talker Detection	If the DCM speech detection is triggering when no one is talking you can check this box to lessen the effect of background noise.

Table 44 – Conference Settings and Descriptions

Add/Edit Conference

STEPS:

1 Navigate to **Destinations➔Conferences.**

2 Click the Add button to create a new Conference, or the button to edit an existing entry.

3 Build the conference room to match your end users desires.

4 Click Save Changes button.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Conference Feature Codes

The Feature Code section of the Conference page displays the key or key combinations that need to be pressed to activate or deactivate (toggling function) features for conferencing.

Feature	Code
Feature List:	*
Mute / Unmute:	*1
Lock / Unlock:	*2
Eject Last User:	*3
Decrease Conference Volume:	*4
Increase Conference Volume:	:*6
Decrease User Volume:	*7
Increase User Volume:	*9
Exit Menu:	*8

Figure 64 – Features Code Page

Sections/Fields	Description
Feature List	Pressing * will list all the features available while in the conference.
Mute/Unmute	Pressing *1 will allow you to toggle (switch) the mute state for yourself.
Lock/Unlock	Pressing *2 as the administrator will allow you to toggle (switch) whether new callers can join the conference.
Eject Last User	Pressing *3 as the administrator will allow you to eject the last caller who entered the conference.
Decrease Conference Volume	Pressing *4 will allow you to decrease the overall volume of the conference call.
Increase Conference Volume	Pressing *6 will allow you to increase the overall volume of the conference call.
Decrease User Volume	Pressing *7 will decrease your call volume during the conference call.
Increase User Volume	Pressing *9 will increase your call volume during the conference call.
Exit Menu	Pressing *8 will exit the features menu and return you to the conference call.

Table 45 – Features Code Settings and Descriptions

Voicemail

When an extension is created, a Voicemail box tied to that extension is automatically created. Voicemail boxes that do not have an extension associated with them are created and administered exclusively through **Destinations➔Voicemail.** There are three locations in the IP PBX that allow management of a voicemail box:

- **Destinations➔Extensons –** Under the Advanced Settings section, the basic settings for the mailbox can be configured.
- **Destinations➔Voicemail –** This area allows for total configuration of the voicemail box, including features like Broadcast and Cascading Message Notification.
- **Smart Personal Console –** Gives end users the ability to configure some mailbox settings.

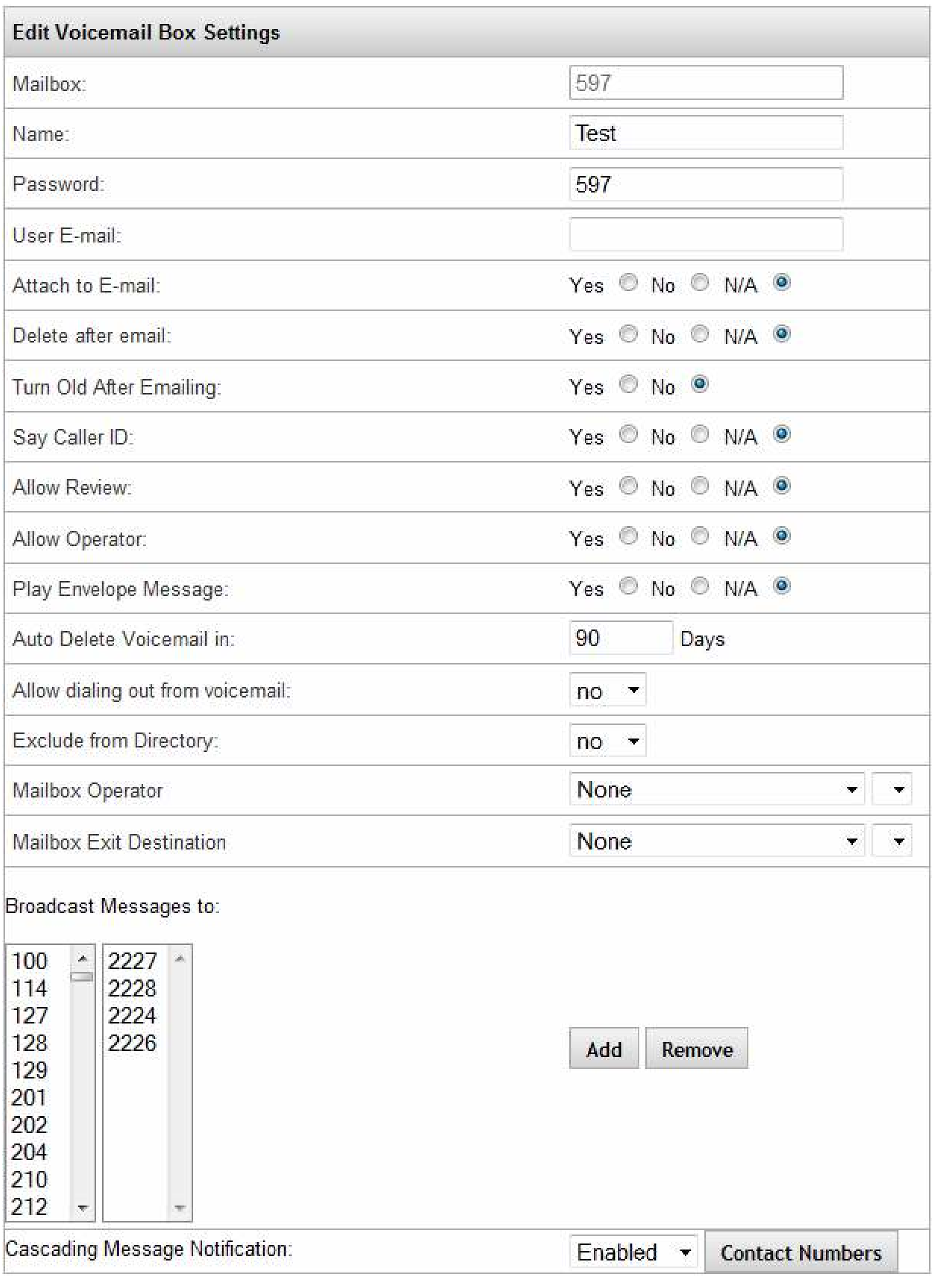


Figure 65 – Edit Voicemail Box Settings Page

Sections/Fields	Description
Mailbox	This is the mailbox number, which must be a unique **3 or 4** digit number.
Name	This is the name of the voicemail box.
Password	This is the password to access the voicemail messages. Only numbers are allowed.
User E-mail	This is the email address associated with this mailbox.
Attach to Email	Defines whether the PBX will send an email to the user informing them they have a new message. An audio file (.Wav) of the message will be attached. **YES** = send email with attachment. **NO** = do not send an email. **N/A** = use the system wide default defined under PBX Setup➔Voicemail.
Delete After Emailing	Defines if the voicemail message will be deleted from the PBX after the email has been sent. **YES** = delete message after it has been emailed. **NO** = do NOT delete message after emailing. **N/A** = use the system wide default defined under PBX Setup➔Voicemail. **Note: This feature should not be enable at the same time as Turn Old After Emailing.**
Turn Old After Emailing	Defines if the voicemail message will be sent from the NEW folder to the OLD folder after emailing. **YES** = move message to Old messages folder after emailing. **NO** = leave the message in the NEW messages folder after emailing. **Note: This feature should not be enable at the same time as Delete After Emailing.**
Say Caller ID	Defines if the Caller ID will be played prior to playback of the message. **YES** = state caller id prior to message content playback. **NO** = do NOT state caller id prior to message content playback. **N/A** = use the system wide default defined under PBX Setup➔Voicemail.
Allow Review	Defines if the caller is permitted to review and take action on the message they leave. **YES** = allow callers to review and re-record the message they left. **NO** = do not allow callers to re-record their message. **N/A** = use the system wide default defined under PBX Setup➔Voicemail.
Allow Operator	Defines if the caller is permitted to press “0” to reach the mailbox operator while leaving a voicemail message. **YES** = allow dial 0 while listening to mailbox greeting. **NO** = disallow dialing 0 while listening to mailbox greeting **N/A** = use the system wide default defined under PBX Setup➔Voicemail.

Play Envelope Message	Defines if the system will play caller ID and time of call prior to playback of the message. **YES** = play message details prior to playback of message contents **NO** = do not play message details prior to playback of contents. **N/A** = use the system wide default defined under PBX Setup➔Voicemail.
Auto Delete Voicemail In	Define the number of days in which voicemail messages are to be automatically deleted from a mailbox. If this is set to "0" (zero) the voicemail message will never expire.
Allow Dial Out Access	This permits the user to dial out from voicemail.
Exclude from Directory	This indicates whether to exclude the extension paired with this mailbox from the directory.
Mailbox Operator	Defines where to route a calller if they press "0" (zero) while listening to your mailbox greeting. Set this to None to use System Default from value defined at **PBXSetup➔General**.
Mailbox Exit Destination	If a caller remains on the line or presses # after leaving a message in this voice mailbox, this is where they will be routed. Set this to None to use System Default from **PBXSetup➔General**.
Broadcast Messages	If this setting is configured, messages that are directly left in this box are broadcasted to the boxes that are added to the list. Note that messages forwarded to this box are not broadcast.
Cascading Message Notification	If enabled, the Cascading Message Notification feature will be activated, allowing you to configure the system to call a single or group of phone numbers when a new voicemail message has been received.

Table 46 - Voice Mailbox Settings and Descriptions

Add Voicemail Settings

The following outlines the steps to create a stand alone voicemail box.

STEPS:

1. From the **Destination➔Voicemail** page, click the **Add Mailbox** button located at the top left corner of the page.

2. The **Edit Voicemail Box Settings** page is displayed. Using the table of parameters and descriptions above, configure the voicemail box to meet your business requirements.

3. Click Save Changes button to save the changes. The system returns you to the **Edit Voicemail Box** page.

4. Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Edit Voicemail Settings

The following outlines the steps to modify voicemail parameters.

STEPS:

1. From the **Voicemail Box Setup** page, locate the mailbox that you want to update.
2. Click the icon to the right of the **Mailbox Name**.
3. The **Edit Voicemail Box Settings** page is displayed. Make the necessary changes to the extension information.
4. Click Save Changes button to save the changes. The system returns you to the **Edit Voicemail Box** page.

Clear Voicemail Messages

The following outlines the steps to clear a voicemail box of all messages and greetings.

STEPS:

1. From the **Voicemail Box Setup** page, locate the mailbox that you want to clear messages from. Click the icon to the right of the **Mailbox Name**.
2. The messages from that mailbox will be deleted.
3. Click Save Changes button to save the changes. The system returns you to the **Edit Voicemail Box** page.
4. Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Delete Voicemail Box

STEPS:

1. From the **Voicemail Box Setup** page, locate the mailbox that you want delete.
2. Click on icon to the right of the **Mailbox Name** of the menu you want to delete.
3. The mailbox is removed from the Mailbox listing.
4. Click Save Changes button to save the changes. The system returns you to the **Edit Voicemail Box** page.
5. Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Broadcast Message – Add

STEPS:

1 From the **Edit Voicemail Box Settings** page is displayed. Scroll to the **Broadcast Messages** section.

2 Select the mailbox from the left list box then click the **Add** button. The item selected will populate the list on the right.

CTRL and Shift clicking is supported to select multiple mailboxes at once.

3 Click Save Changes button to save the changes. The system returns you to the **Edit Voicemail Box** page.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Broadcast Message – Remove

STEPS:

1 From the **Edit Voicemail Box Settings** page is displayed. Scroll to the **Broadcast Messages** section.

2 Select the mailbox from the right list box then click the **Remove** button. The item selected will be removed from the list.

CTRL and Shift clicking is supported to select multiple mailboxes at once.

3 Click Save Changes button to save the changes. The system returns you to the **Edit Voicemail Box** page.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

5 Click the **Apply Changes** link located on the right hand corner of the page, to commit the changes to the database.

Cascading Message Notification

Cascading Message Notification works along the same principle as FollowMe, but pertains to voicemail messages. When configured, if an extension gets a new VM, the PBX can call a variety of numbers, in the order defined, to inform the user of the message.

Note: *Once the call is answered, the user will have the ability to listen to their message from their phone, or to refuse the call. Until the voicemail box has its new messages cleared, the system will continue to call out when the Notification Interval is met.*

Set Cascading Messages

STEPS:

1 Navigate to the **Edit Voicemail Box Settings** page.

2 Scroll to the **Cascading Message Notification** section.

3 Select **Disable** to turn the notification feature **OFF**. Select **Enabled** to turn the notification feature **ON**.

4 Click 

button to save the changes. The system returns you to the **Edit Voicemail Box** page.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Setting Cascading Interval

STEPS:

1 Navigate to **PBX Setup➔General**.

2 Locate the **Cascading Voicemail Notification Interval**, set the value to the desired number of minutes, the PBX will check for new emails and make calls to notify when the interval passes. Default is 5 minutes.

3 Click the 

button to save the changes.

4 Cick the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Configure Contact Numbers for Cascading Message Notification

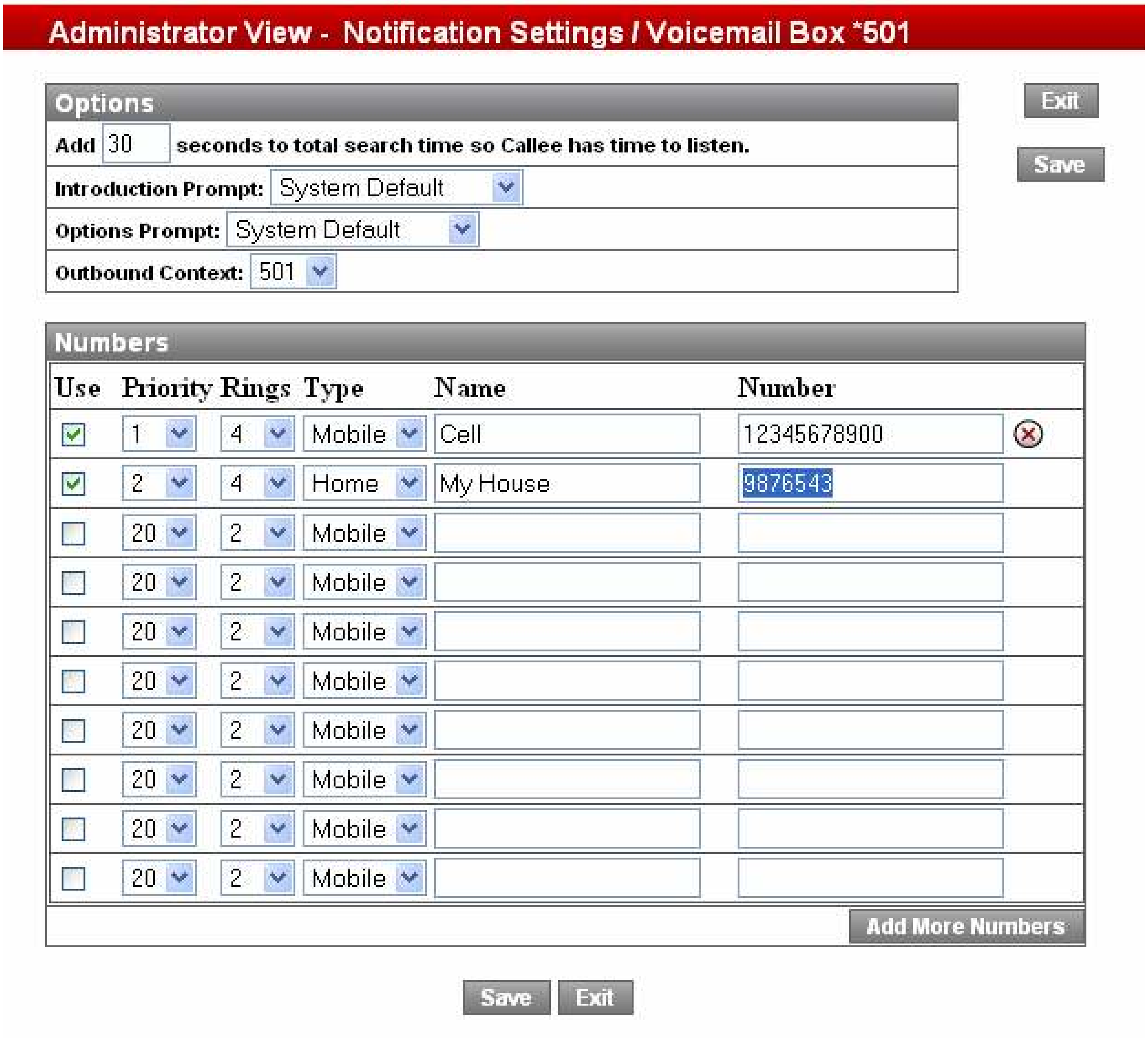


Figure 66 – Notification Settings / Voicemail Box Page

Sections/Fields	Description
Add X seconds to total search time	This allows you to configure how many seconds the system will spend searching for the called party. Default is 12 seconds.
Introduction Prompt	Introduction Prompt plays ***"You have new messages"*** after you answer the call. This can be set to any custom prompt you have recorded on the system.
Options Prompt	Options Prompt plays "***Please enter your voicemail password, or 2 to reject this call"*** after you answer the call. This can be set to any custom prompt you have recorded on the system.
Outbound Context	Outbound Context gets set to an extension number, and will use that extension's Class of Service when making outbound calls.
Numbers	Configure the numbers you wish to be called when a VM message is received. The calls will be made in order, with 1 being the highest priority and 20 the lowest. You must check the Use column for each number to be considered active. If you wish to delete a number, click the button then save and exit.

Table 47 – Notification and Voicemail Box Settings and Descriptions

STEPS:

1 From the **Destinations➔Voicemail** page, click the pencil of the mailbox you wish to edit.

2 Scroll to the **Cascading Message Notification** section and click the **Contact Numbers** button (Cascading Message Notification needs to be enabled)

3 The **Notification Settings / Voicemail Box** window will appear. Set the parameters based on the description of the values provided in the table above.

4 Click the Save Changes button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Schedules

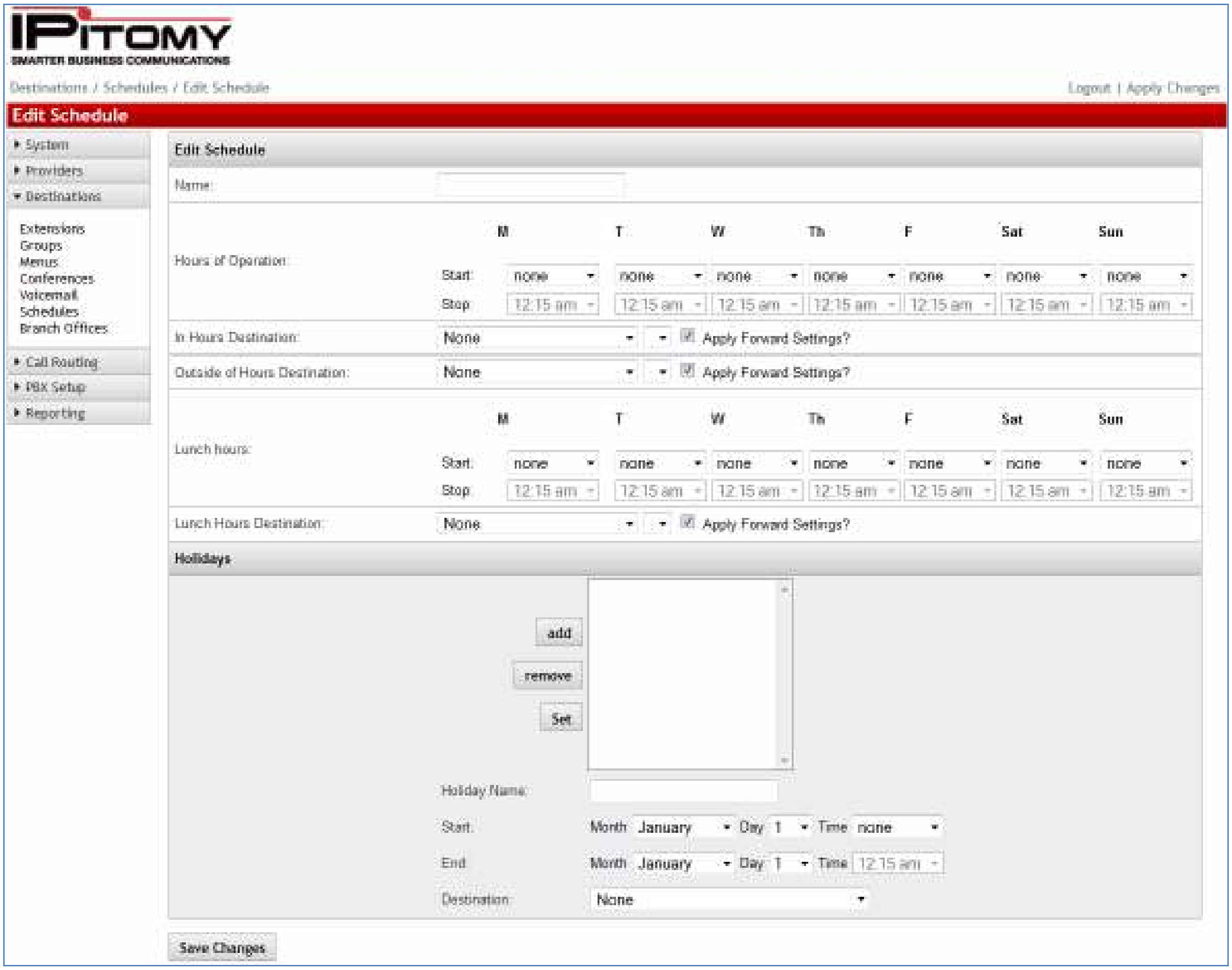


Figure 67 – Edit Schedule and Holidays Page

Sections/Fields	Description
Name	This is the name that is associated with the schedule.
Hours of Operation	Define the start and end time for the hours that will be considered Operational hours
In Hours Destination	This is the assigned Destination where the calls will go during the schedule's hours of operation. If "**Apply Forwarding Settings**" is enabled (checked), the destination's call forward settings. The items in the drop-down listing are the configured Destinations in your PBX system

Outside of Hours Destination	This is the assigned Destination where the calls will go when it is received outside the schedule's hours of operation. If "**Apply Forwarding Settings**" is enabled (checked), to use the destination's call forward settings. The items in the drop-down listing are the configured Destinations in your PBX system.
Lunch Hours	Define the hours allocated for lunch for a schedule. Calls received during this time will be sent to the Lunch Hours Destination.
Lunch Hours Destination	This is the assigned Destination where the calls will go when it is received during the allocated Lunch Hours. If "**Apply Forwarding Settings**" is enabled (checked), to use the destination's call forward settings. The items in the drop-down listing are the configured Destinations in your PBX system.
Holidays	**Holiday Name** – This is the name associated with this holiday schedule. **Start** – This is the start date and time for this holiday schedule. (The year is defaulted to the current year.) **End** – This is the end date and time (expiration) for this holiday schedule. (The year is defaulted to the current year.) **Destination** – This is the assigned Destination where calls will be routed when it is received during the holiday schedule.

Table 48 – Edit Schedule Settings and Descriptions

Add Schedule

STEPS:

1 From the **Destinations➔Schedules** page, click on the **ADD Schedule** button.

2 The **Edit Schedule Switch** page appears.

3 Enter the settings for the new schedule.

4 Click the Save Changes button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Add Holiday to Schedule

STEPS:

1 From the **Destinations➔Schedules➔Edit Schedule Switch** page, scroll down to the **Holidays** section of the page.

2 Enter the dates and times for the new holiday schedule. Click the **Add** button to add the new holiday schedule.

3 Click on the newly added holiday, select a destination from the Destination dropdown and click **Set**

4 Click the Save Changes button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Remove Holiday from Schedule

STEPS:

1 From the **Destinations➔Schedules➔Edit Schedule Switch** page, scroll down to the **Holidays** section of the page.

2 Select the holiday (appearing in the list) that you want to remove. Click the **Remove** button to delete the holiday schedule. The item in the Holiday list is removed.

3 Click the Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Edit Schedule

STEPS:

1 From the **Destinations➔Schedules** page, locate the schedule name that you want to edit.

2 Click on icon to the right of the **Name** of the schedule you want to update. The **Edit Schedule Switch** window appears.

3 Edit the necessary parameters to configure the schedule.

4 Click the Save Changes button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Delete Schedule

STEPS:

1 From the **Destinations➔Schedule** page, locate to the **Schedule Name** from the listing that you want to delete.

2 Click on icon to the right of the **Name** of the schedule you want to delete. The schedule is removed from the listing page.

3 Click the Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Branch Offices

Branch Offices provide a powerful tool for interconnecting PBX Systems. Through Branch Offices you can create a network of systems where dialing phones on remote systems is as easy as dialing local extensions. You can route calls inbound to a destination on any branched PBX, and outbound through the trunks of other systems networked in this way. Branch office extensions can even participate in ring groups.

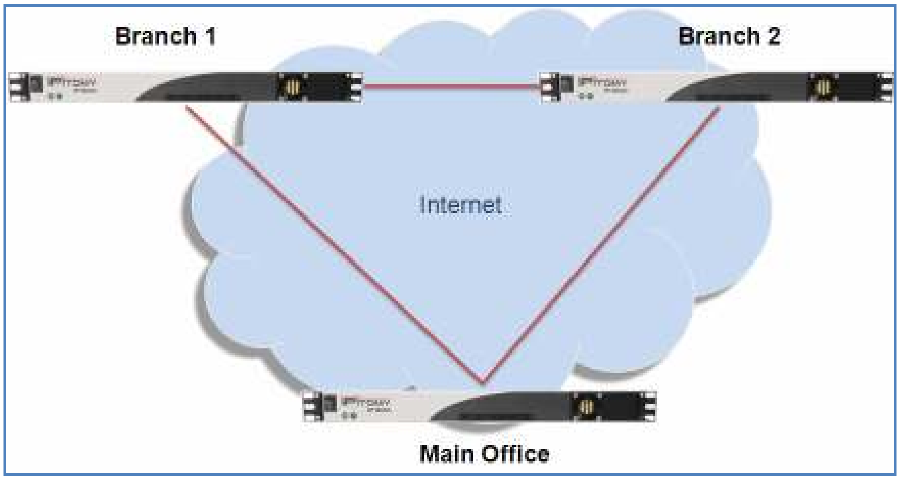


Figure 68 – Sample Branch Office Networking

!	**IMPORTANT: Branch Offices communicate using IAX2 protocol over Port 4569. Refer to the Port Forwarding Configuration Table earlier in this manual for more information.**

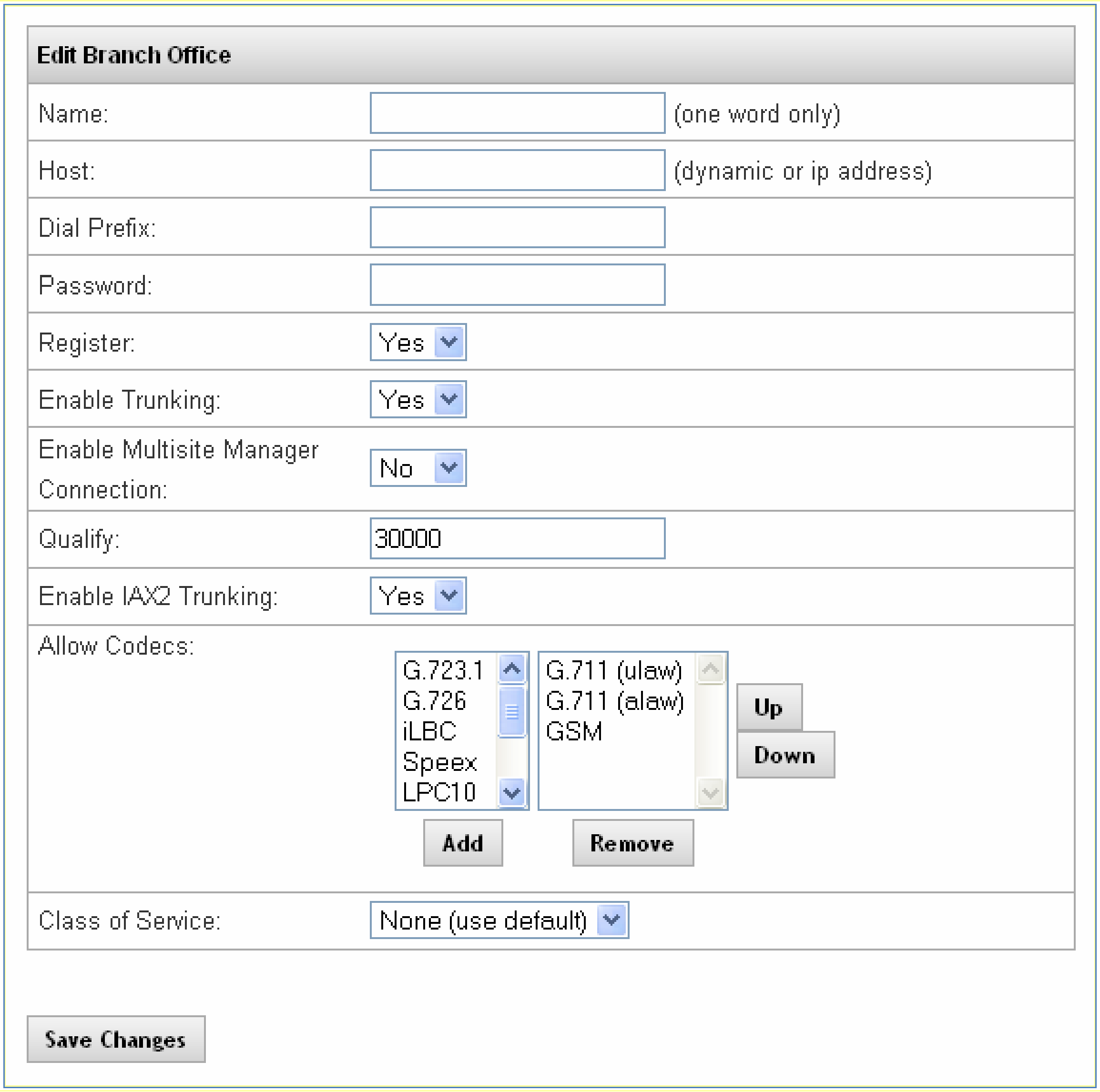


Figure 69 – Edit Branch Office Page

Sections/Fields	Description
Name	The name that is associated with the branch office. Both sites involved in the Branch Office connection must use the same Name. **NOTE: This field will not allow spaces in the name. It must be all one word. Symbols can be used (i.e. underscore "_" dashes "-", etc.) can be used in place of spaces.**
Host	Enter the static IP of the system your PBX is connecting to. If they do not have a static IP address, enter the word **dynamic.**
Dial Prefix	The dial prefix must be an asterisk, plus two numbers (ie. *22). This prefix must be dialed to reach any destination on another PBX that has not been entered as a Branch Extension.
Password	This is the password that will allow access to the PBX. Both sites involved in the Branch Office connection must use the same Password.
Register	**YES**= to require that the branch office register with the PBX system. Only used if the Host is set to dynamic. **NO** = does not require that the branch office register with the PBX system.
Enable Trunking	**YES** = Branched PBXs are able to dial calls out your PBXs trunks. **NO** = Branched PBXs are not able to dial calls out your PBXs trunks.
Enable Multisite Manager Connection:	Defines if Multi-Site Call Manager functionality is enabled. This feature requires a license from IPitomy to be functional.
Qualify	The time in milliseconds that the PBX should check that the branch office is still online. Setting this to zero may increase Branch Office stability.
Enable IAX2 Trunking	Indicates whether to use IAX trunking. If enabled (**YES**), this setting can help eliminate packet overhead by cutting the cost of continuous communication.
Allow Codecs:	Defines what codecs this branch office can use for calls. Priority for codecs are defined from top to bottom.
Class of Service:	Allows for individual Branch Offices to use different COS. Setting to None will follow the default Branch Office COS defined under PBX Setup➔General.

Table 49 – Branch Office Settings and Descriptions

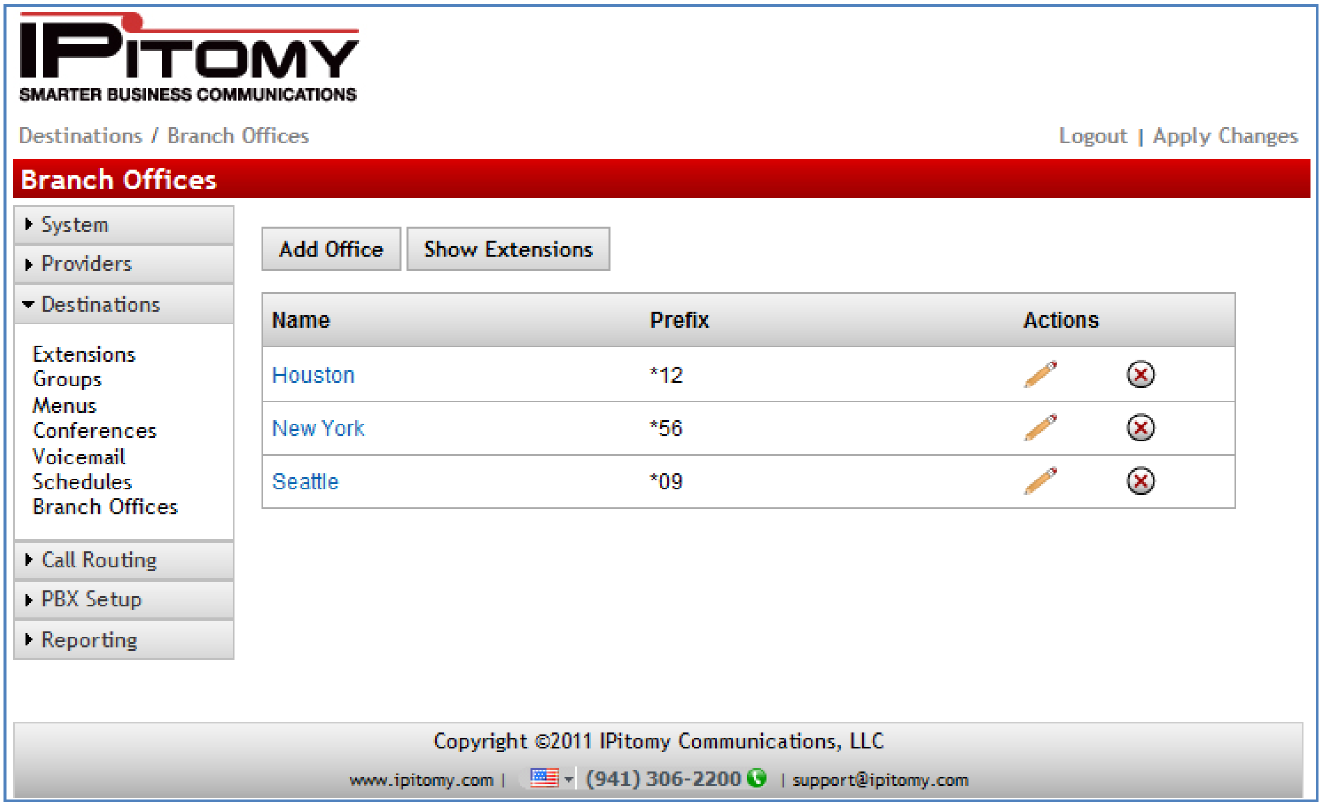


Figure 70 – Branch Offices Page

Configuring Office 1

STEPS:

1 Click on **Destinations→Branch Offices**. The **Branch Offices** page is displayed.

2 Click on the **ADD office** button. The **Edit Branch Office** page appears.

3 Give a unique **NAME** for the connection. The name should only contain alpha-numeric characters and no spaces. It should be ***ONE WORD ONLY***.

4 Enter the **External IP** of the PBX in the **Host** field.

5 Enter a unique **DIALING PREFIX** of an asterisk followed by two numbers.

6 Give a unique **PASSWORD** for the connection.

7 Select **NO** for **REGISTER** *(Note: that registration is not required if host is known).*

8 Configure **Enable Trunking** as needed.

9 Configure **Enable Multisite Manager Connection** as needed.

10 Set **QUALIFY** at **0**.

11 Configure **Enable IAX2 Trunking** as needed.

12 Click the Save Changes button to save the changes.

13 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Configuring Office 2

STEPS:

1 Click on **Destinations➔Branch Offices**. The **Branch Offices** page is displayed.

2 Click on the **ADD Office** button. The **Edit Branch Office** page appears.

3 Enter a name in the **NAME** field that matches that given to the Office 1 PBX. The name should only contain alpha-numeric characters and no spaces. It should be **ONE WORD ONLY**.

4 Enter an **external IP address** or **domain** corresponding to the IP of the main office PBX in the **HOST** field.

5 Enter a **UNIQUE** dialing prefix for the extensions connected to the Office 2 PBX.

6 The **PASSWORD** for Branch Office 2 needs to be the **same** as the one assigned in the Branch Office 1 PBX.

7 Select **NO** for **REGISTER**.

8 Configure **Enable Trunking** as needed.

9 Configure **Enable Multisite Manager Connection** as needed.

10 Set **QUALIFY** at **0**.

11 Configure **Enable IAX2 Trunking** as needed.

12 Click the Save Changes button to save the changes.

13 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

To setup additional Branch Offices in the PBX, use the steps outlined above. Keep in mind to use the same logic (pattern) and make sure that the dialing prefix is unique to each office.

Edit Branch Office

STEPS:

1 From the **Destinations➔Branch Office** page, locate the schedule name that you want to edit.

2 Click on icon to the right of the **Name** of the Branch Office you want to update. The **Edit Branch Office** page appears.

3 Edit the necessary parameters to configure the branch office.

4 Click the Save Changes button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Delete Branch Office

STEPS:

1 From the **Destinations➔Branch Offices** page, locate to the **Name** from the Branch Office listing that you want to remove.

2 Click on icon to the right of the **Name** of the **Branch Office** you want to delete. The Branch Office is removed from the listing page.

3 Click the Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database

Branch Extensions

Branch Extensions are created through the add extension field at the bottom of the branch office edit page. Branch extensions can be dialed directly without a prefix, provided that these extensions are properly configured on the PBX for which the branch extensions are defined. Branch extensions will appear in call routing drop down lists throughout the system after they are created. If needed, you can add Ring Group and Menu numbers as Branch Extensions.

Configuring Office 2 with Branch Extensions

Branch Extensions [ ] Add

Name	Extension	Action
Branch Extension	222	⊗
Branch Extension	245	⊗
Branch Extension	288	⊗
Branch Extension	212	⊗
Branch Extension	100	⊗
Branch Extension	101	⊗

Figure 71 – Branch Office Extension Section

STEPS:

1 Click on **Destinations➔Branch Offices.** The Branch Offices page appears.

2 Click on the **Branch Office** connection (**Name**) assigned to Office 1.

3 Assuming Office 1 has extension numbers 100 thru 110, enter the numbers 100 to 110 in the field above the Add button then click **ADD**. You will enter (add) each extension one at a time. **Note: Using the format X-Y will add all extensions in the specified range.**

4 If the extension number is valid (not already in use), the new extension will appear in the list of Branch Extensions.

5 Click the **Save Changes** button to save the changes.

6 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

7 Verify the new extension by calling the number from phones located (registered) at Branch Office 2.

To place a call from one Branch Office to another simply dial the prefix that was assigned to that locations PBX + the extension of the user trying to be reached at the other office. This same easy concept works for transferring calls from one Branch Office to another. Or, if you have the extension configured as a Branch Extension on your PBX, you can simply dial the extension number and it will be automatically routed to the branch.

View Branch Office Extensions

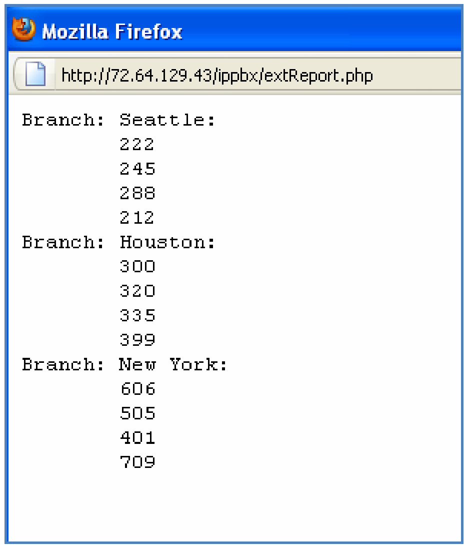


Figure 72 – Show Extensions Page

STEPS:

1 From the **Destinations➔Branch Office** page, click on the **Show Extensions** button located at the top of the Branch Office list page.

2 A popup window appears displaying all the extension information for each Branch Office that is currently in the system.

3 Click on the “**X**” to close the popup window.

Call Routing

Call routing sends callers to specific inbound destinations within the system, and routing outbound callers over specific outbound routes like local, long distance, international and emergency.

Incoming Call Routing

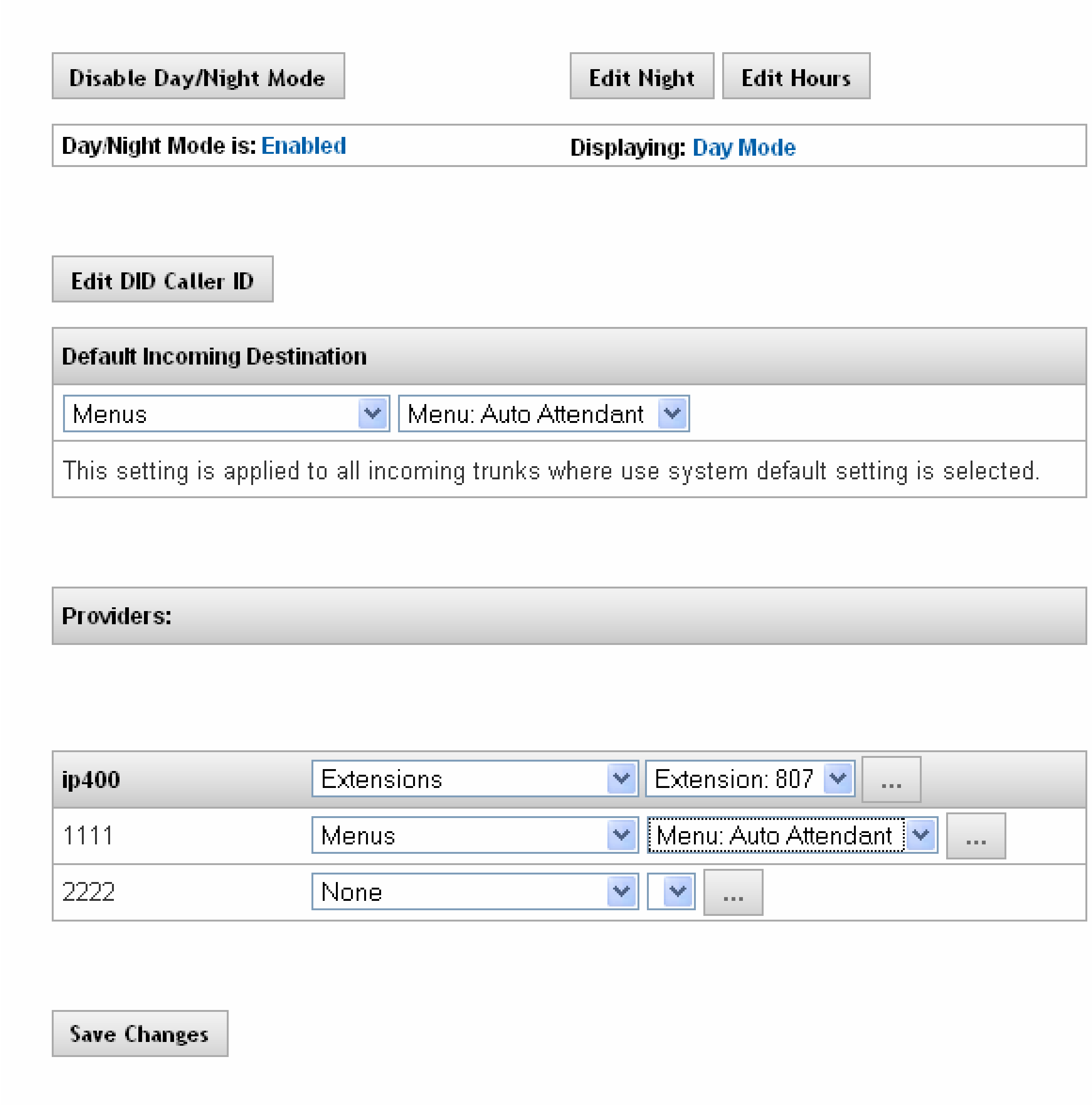


Figure 73 – Incoming Call Routing Page

This table describes the parameters that can be set for incoming calls. Calls can be routed to destinations and you can also enable (set) the time parameter when the calls will get sent to their defined route.

Sections/Fields	Description
Disable Day/Night Mode Button	**The PBX system defaults to Day/Night Mode Disabled**. To toggle between **Enabled/Disabled** settings for the **Day/Night Mode** setting. When enabled the Edit Night and Hours button will appear which will allow you to configure the night destinations and hour parameters.
Edit Night/Day Button	Toggles between **Day/Night Mode** settings. When selected, it will change the **Displaying** field to either "**Day Mode**" or "**Night Mode**", allowing you to configure the destination for calls when in and out of hours.
Edit Hours	Allows you to edit the hours for the incoming calls to routing. **IMPORTANT: If you enable day night mode service you must populate the system hours schedule otherwise all calls will be routed to the night destinations.**
Edit DID Caller ID	Allows you to edit CID information for incoming calls based on DID. If you wish to retain the original CID information as well as display the override info, be sure to check the box for **Prepend**.
Providers	Each provider will display in a gray box. DIDs for each provider will display beneath the gray box in white. If a DID destination is set to None, it will follow the settings for the provider itself. If a provider destination is set to None, calls will route to the Default Incoming Destination.
...	This button allows for configuration of destination specific settings. Groups: Allows you to define a priority, increasing or decreasing the calls importance in the queue. The higher the number, the higher the priority. Extension: Allows you to define a group of numbers that will be dialed once the call has been answered. Typically used when connecting an extension to a fax server that routes faxes to people based on the digits read at the time of answering the call.

Table 50 – Incoming Call Routing Settings and Descriptions

Set Default Incoming Destination

STEPS:

1 From the **Call Routing→Incoming Routing** page, go to the **Default Incoming Destination** section.

2 Select the desired Destination category from the first drop-down list, then select the specific destination from the second drop-down list.

3 Click the Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Set Provider Trunk Incoming Destination

1 From the **Call Routing➔Incoming Routing** page, go to the section that corresponds to either your SIP Provider or Channel Group name.

2 Select the desired Destination category from the first drop-down list, then select the specific destination from the second drop-down list.

3 Repeat step 2 for all other providers. Any provider that can follow the Default Incoming Destination can be left as None.

4 Once you have made the necessary changes, click the Save Changes button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Enable/Disable Day/Night Mode

The following section outlines steps to enable or disable the Day/Night Mode for Incoming Calls.

STEPS:

1 From the **Call Routing➔Incoming Routing** page, click on the Enable/**Disable Day/Night Mode** button located at the top of the page.

2 The Day/Night Mode changes either Enabled or Disabled (depending on its previous setting).

3 Once you have set the parameter to the desired mode, click Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Switch Day/Night Mode

This feature allows you to change whether you are editing the Day destinations or the Night destinations for incoming calls. The following section outlines the steps to switch or toggle to/from night or day.

STEPS:

1 From the **Call Routing➔Incoming Routing** page, click on the **Edit Night or Edit Day** button located at the top of the page.

2 The Displays field changes either **Day Mode** or **Night Mode** (depending on its previous setting).

3 Modify the destinations for whichever mode you are working with, then click Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Edit Hours

STEPS:

1 From the **Call Routing→Incoming Routing** page, click on the **Edit Hours** button located at the top of the page, which will open the schedule page for Day/Night mode.

2 Set the hour parameters for the incoming route to meet the business requirements. See the previous section pertaining to Schedules for more details.

3 Click Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Edit Incoming Caller DID

If you want a particular DID to display the override CID information followed by the original inbound CID information, be sure to check the box for **Prepend**, otherwise the original CID information will be lost.

DID Number	Caller ID Name		Caller ID Number	
9413062201		☐ Prepend		☐ Prepend
9413062203		☐ Prepend		☐ Prepend
9413062204		☐ Prepend		☐ Prepend
9413062205		☐ Prepend		☐ Prepend
9413062206		☐ Prepend		☐ Prepend

Figure 74 – Incoming Routing Edit Incoming DID CID Page

STEPS:

1 From the **Call Routing→Incoming Routing** page, click on the **Edit DID Caller ID** button located at the top of the page.

2 A list of existing DID numbers is displayed. Enter **Caller ID Name** and corresponding **Caller ID Number** for the desired DID Number.

3 Enable Prepend for each by checking the box, if desired.

4 Once you have made the necessary changes to the DID numbers, click the Save Changes button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Outgoing Call Routing

By default, the PBX has five dialing routes built and included in the System Default Class of Service. These routes are 7 Digit, 10 Digit, 11 Digit, International, and Emergency, and should handle the normal day to day calling needed on a PBX. Once you have configured trunks on the PBX, you must add them to whichever dialing routes you wish to use those trunks. IPitomy advises you add ALL available trunks to the Emergency route, and to check with the end user before adding trunks to the International dialing route. Some users may require specific dialing routes to be built to meet their needs.

!	**IMPORTANT: Please keep in mind that all new routes need to be added to a Class of Service before the route can be used.**

Sections/Fields	Description
Add Route	Click this button to create a new **Outbound Dialing Route**.
Add Custom Route	Advanced feature only available through licensing. Contact IPitomy for more information.
Class of Service	A Class of Service (COS) pertains to a group of outbound dialing routes. You can define a specific COS for an extension or branch office, limiting which routes are usable by those entities. Select a specific COS from the dropdown to view the routes included for that COS. Click on the blue words Class of Service to create, modify, or delete a Class of Service.
7 Digit Local Dialing	By default this route looks for a pattern of any number 7 digits long.
10 Digit Dialing	By default this route looks for a pattern of any number 10 digits long.
1+ Dialing (11 Digit)	By default this looks for a pattern of numbers that start with a 1 and is 11 digits long.
International	By default this looks for a pattern of numbers that start with 011 that is between 4 and 15 digits long.
Emergency	By default, this looks for the specific pattern of 911.

Table 51 - Outbound Routing Page Descriptions

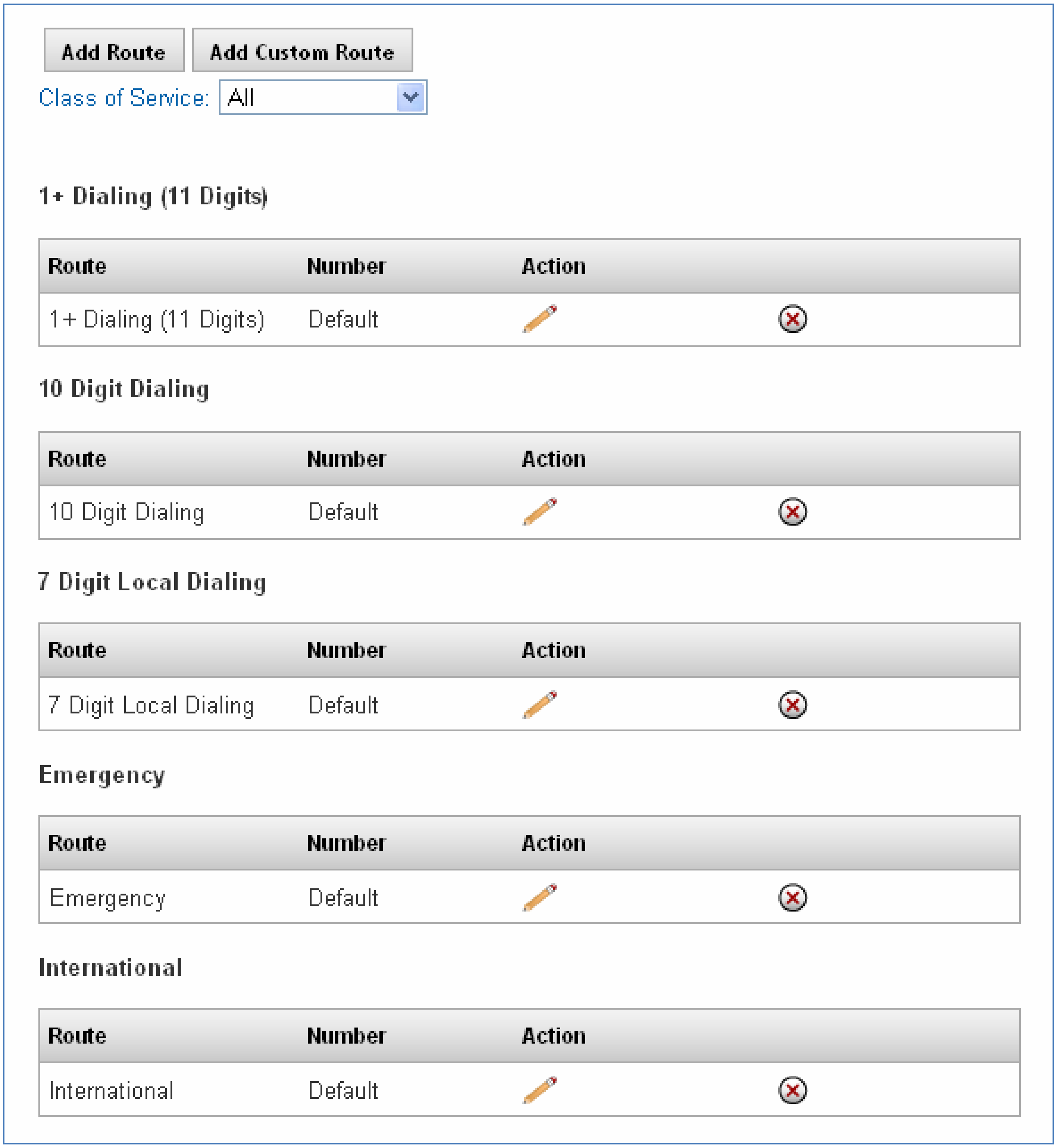


Figure 75 – Outgoing Routing Page

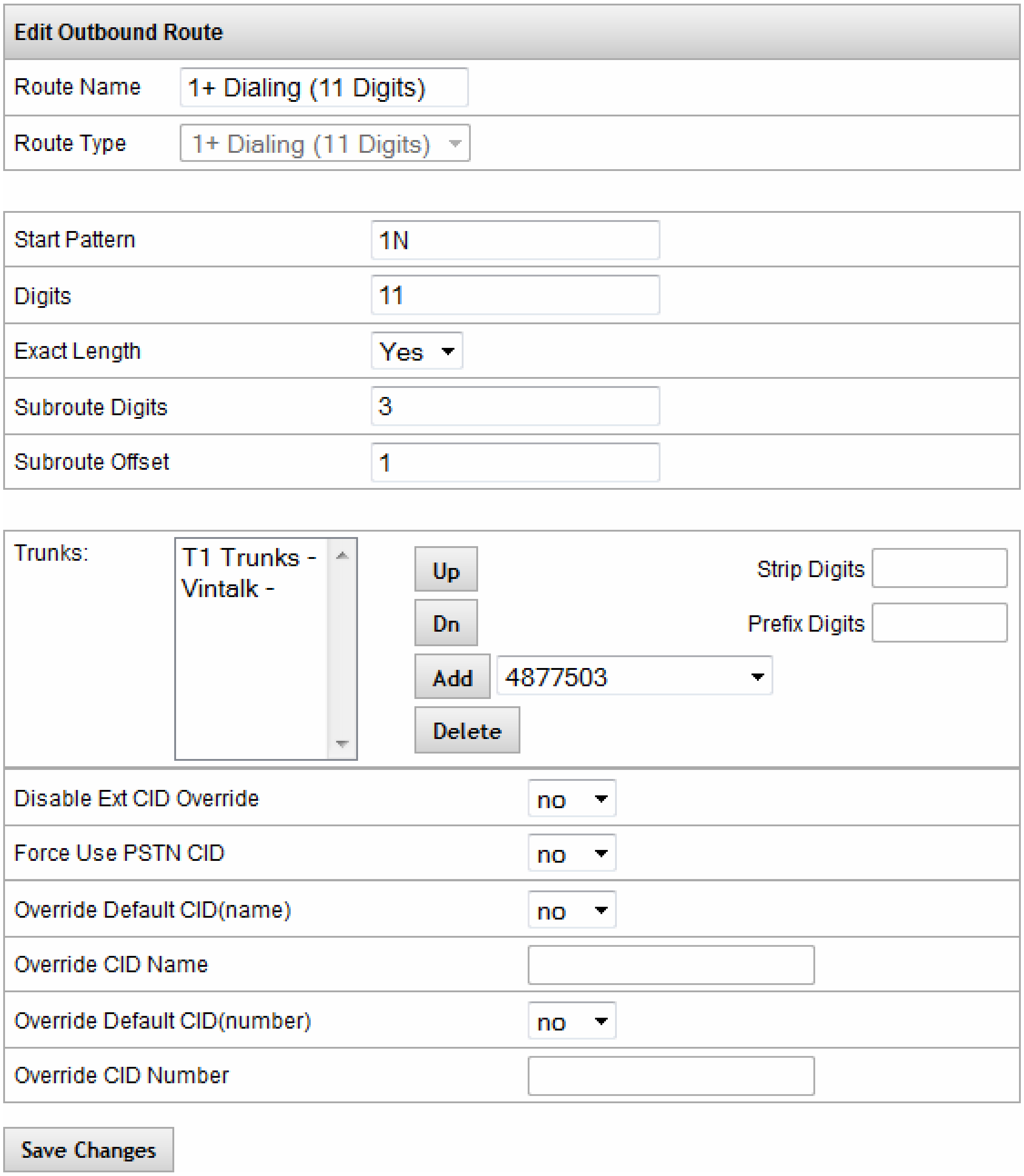


Figure 76 – Add New Outgoing Route Page

IMPORTANT: Please keep in mind that all new routes need to be added to a Class of Service before the route can be used. Subroutes are automatically added under the parent route they use.

Sections/Fields	Description
Route Name	This is the name associated with the outgoing route.
Route Type	Set to New Route for a brand new dialing route, or select an existing route to build a subroute.
Start Pattern	Defines what type of digit pattern will match the route. Acceptable entries are: Numbers X = Any number N = Any number, excluding 1 and 0
Digits	Length of the pattern that will match the route.
Exact Length	Defines if the number dialed has to exactly match the pattern length defined under Digits. **Yes –** Does need to match exactly. **No –** Does not need to match exactly.
Subroute Digits	How many digits of the number dialed to read for comparison to any subroutes that may be built for this route.
Subroute Offset	How many digits of the number dialed to skip before starting to read **Subroute Digits**
Number	When building a subroute off of an existing **Outbound Dialing** Route, this is the number that will match the subroute digits of the parent outbound route. For example, the 11 digit dialing route reads the area code for subroute digits by default. Here, you would enter the area code that needs to be subrouted.
Trunks	In order to dial outbound from a dialing route, you need to define what trunks it will use. **Add** – Use this button to add the trunk selected from the dropdown to the route. **Delete** – This button allows you to delete the trunk selected from the route. **Up** – Highlight a trunk and click this button to move it up the list, and in priority. **Dn** – Highlight a trunk and click this button to move it down the list, and in priority. **Strip Digits** – This indicates whether the PBX will strip any digits before sending the call out a particular Trunk. This is set per trunk. **Prefix Digits** – Defines the digits to be prepended to the dialed number before sending the call out a particular Trunk. This is set per trunk.

CID Override for Outbound Route Parameters	In addition to the Extension and Trunk, Caller ID (CID) override can be set on a per route basis. Each outbound route can be configured to override the CID name and/or CID number that is sent when dialing out each trunk in the route. Each outbound route also has control over whether extensions with Ext CID Override configured are able to send their CID information. **Disable Ext CID Override -** Set to "YES" if you want to disable the CID override for the outgoing route, including Extension CID settings. **Force Use PSTN CID –** New in version: 3.4.1, it is now possible to deliver the CID received on PSTN inbound calls out on calls being transferred outbound. This feature is subject to the ability of the provider associated to the outbound call. This provider must be able to allow substitution of CID. Set to "YES" if you want to send the received CID with a call transferred outbound.. **Override Default CID (name) -** Set to "YES" if you want to enable custom Name CID override for the outgoing route. **Override CID Name –** If the Override Default CID (name) is set to "YES", enter the new CID override name for the outgoing route. **Override Default CID (number) -** Set to "YES" if you want to enable custom Number Default CID override for the outgoing route. **Override CID Number –** If the Override Default name is set to "YES", enter the new CID override number for the outgoing route.

Table 52 – Outbound Route Settings and Descriptions

Add Outgoing Route

STEPS:

1. From the **Call Routing➔Outgoing Routing** page, click on the **Add Route** button located at the top of the page. The **Edit Outbound Route** page appears.

2. Enter the parameters to configure the outbound call route.

3. Once you have entered all the necessary information, click the Save Changes button to save the changes.

4. Add the new route to any Class of Service it will be used with.

5. Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Edit Outgoing Route

STEPS:

1. From the **Call Routing➔Outgoing Routing** page, locate the Outbound Route name that you want to edit.

2. Click on icon to the right of the **Route** you want to update. The **Edit Outbound Route** page appears.

3. Edit the necessary parameters to configure the outbound route.

4 Click the **Save Changes** button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Delete Outgoing Route

STEPS:

1 From the **Call Routing➔Outgoing Routing** page, locate to the **Route Name** from the Route listing that you want to delete.

2 Click on icon to the right of the **Route Name** you want to delete. The Route is removed from the listing page.

3 Click the **Save Changes** button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Class of Service

A Class of Services is used to define which outbound dialing routes a given feature can use. By clicking the blue words **Class of Service** on the Outgoing routing page, you will be able to delete, create, clone, and populate your Class of Service with the desired outbound routes.

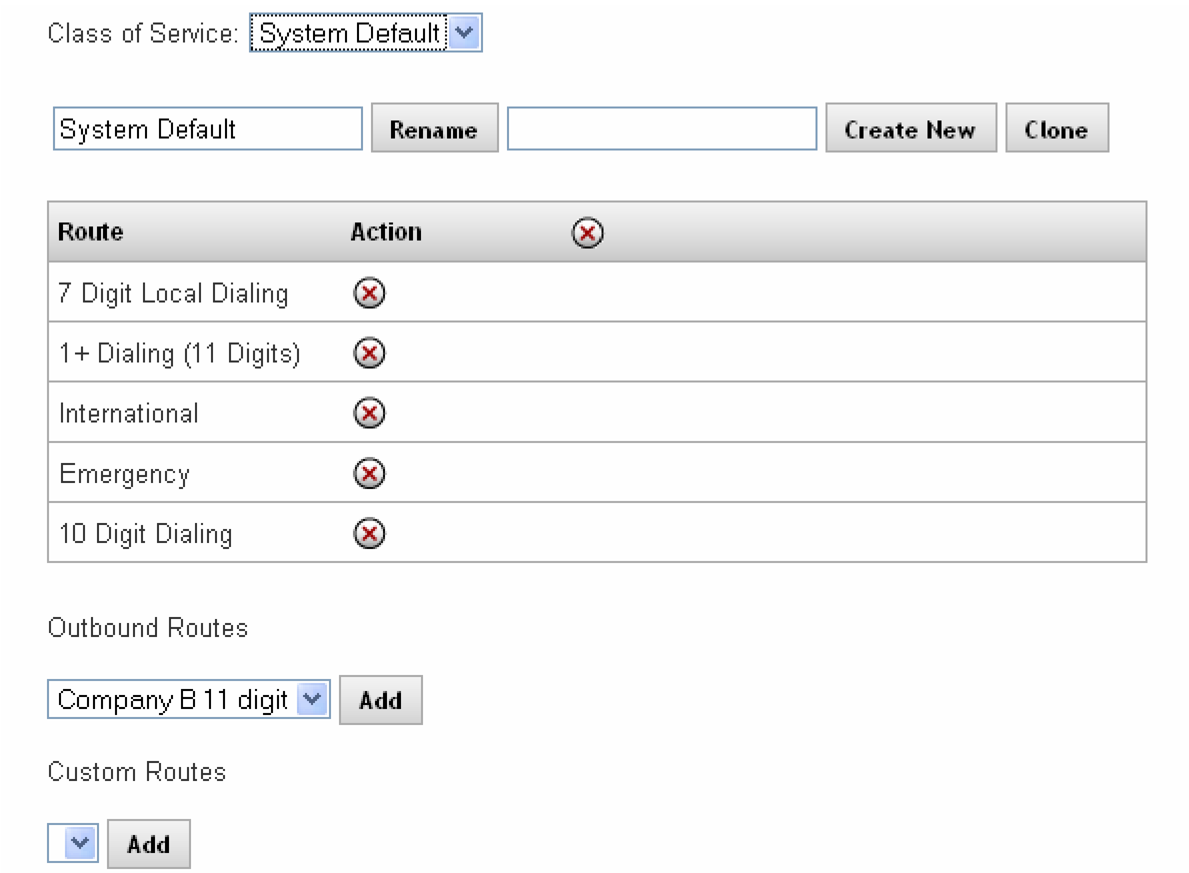


Figure 77 - Class of Service

Sections/Fields	Description
Class of Service	Use the drop down box to view Class of service. You can then take actions pertaining to that COS.
Rename	Enter a new name in the far left text field, then click the **Rename** button to change the name of the COS currently being viewed.
Create New	Allows you to create a new, blank, COS. Name the COS in the text field to the left of the Create New button, then click the button to create.
Clone	Select from the drop down the COS you wish to clone. Enter a name in the field to the left of the Create New button. When you click the Clone button, a new COS will be created with the new name which will be populated with the routes of the original COS.
Add	Clicking the Add button will add the route displayed in the dropdown to the COS that is currently being viewed.

	Clicking this button in the gray bar will delete the COS. Clicking this button next to a route will delete that route from the viewed COS.

Outgoing Call Routing

This section demonstrates, by example, some of the common configurations used with Outbound Routing.

Configure Block Calls Subroutes

In this example, your customers want to block their users from dialing 1-900 numbers. To achieve this, you will need to create a subroute and configure it using the specifications below.

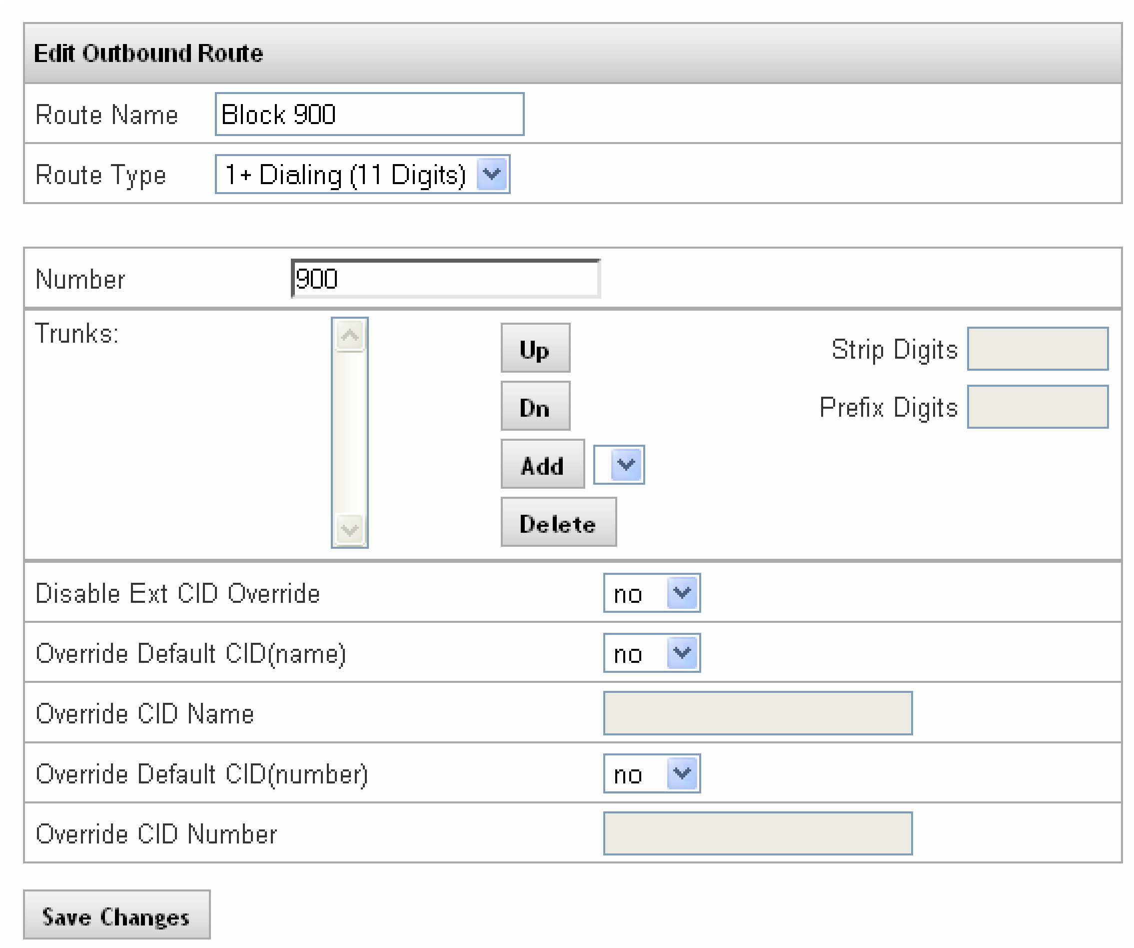


Figure 78 – Blocking Dialing 1-900

Sections/Fields	Description
Route Name	Enter a meaningful name for the outbound route. In this case you may want to call it “**Block 900**”.
Route Type	Select 1+ Dialing, as this will be a subroute
Number	900, since the default 1+ Dialing route skips the 1 that is dialed, and reads the next 3 digits for subroutes

Trunks	None should be added, as without a trunk to dial out, any 1-900 number will be dead in the system.

Table 53 – Block 1900 Outbound Call Configuration

Configure Least Cost Routing Subroutes

In this example, the end user has multiple providers in different regions, and wants dialed calls to use the most cost effective trunk when dialing out. The default 1+ (11 Digit) dialing route has Subroute Offset = 1 (ignores the first digit dialed, this case being the 1 for a long distance number) and Subroute Digits = 3 (after ignoring the first digit from subroute offset, this reads the next 3 digits (area code) in the dialed string and compares them to any subroutes created). The subroute that will be built specifically routes calls dialed to the 941 area code through a different, more cost efficient trunk.

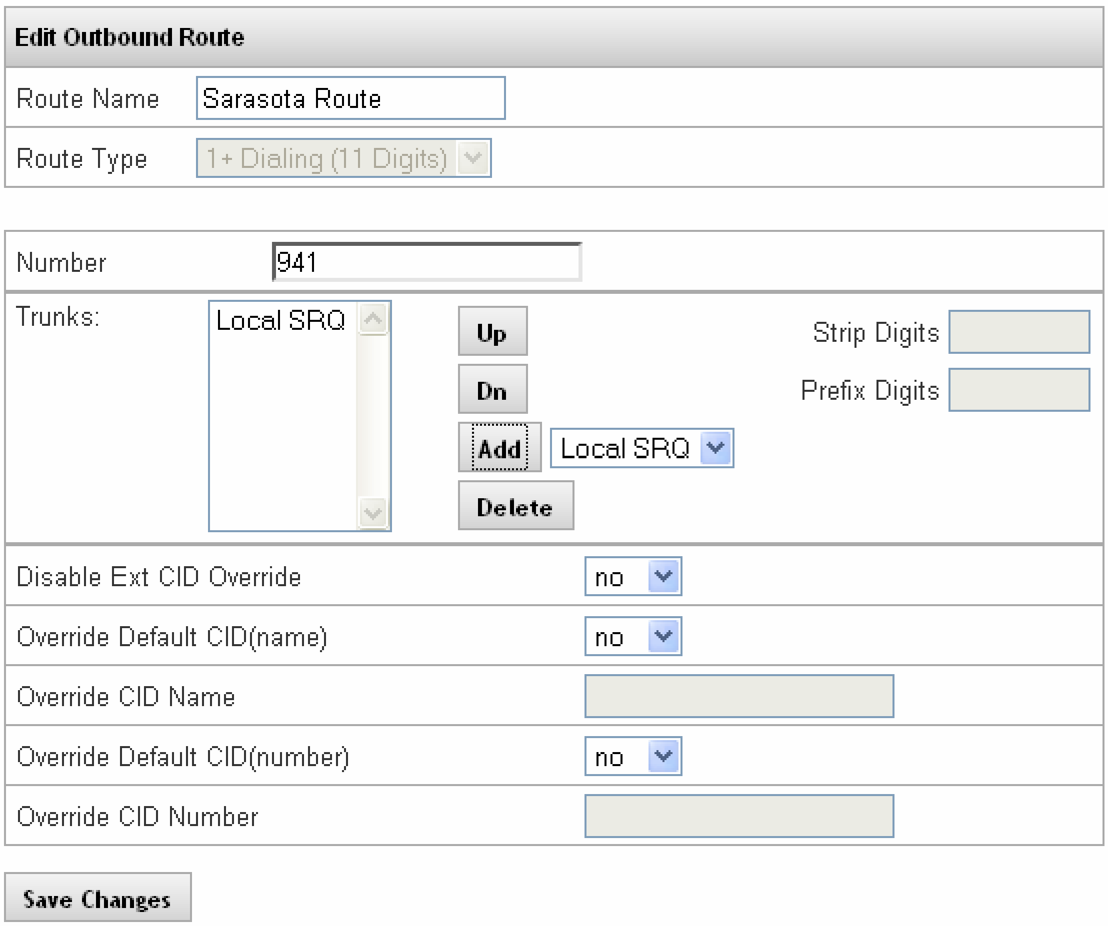


Figure 79 – Least Cost Routing Example

Sections/Fields	Description
Route Name	Enter a meaningful name for the outbound route. In this case you may want to call it "**Sarasota Subroute**".
Route Type	Select the 1+ Dialing as the route type, as this will be a subroute of that dialing route.

Number	Enter 941 as this is the area code for the
Trunks	Add the trunks that will call this particular subroute in the most cost effective manner.

Table 54 – Least Cost Outbound Call Subroute Configuration

IMPORTANT: The normal 11 Digit route is using analog 1 and analog 2 for trunks, while the 941 sub-route uses a trunk that can dial locally to Sarasota without accruing long distance charges.

Configure Information (411/1411) Subroute

If you would like callers to have the ability to use Information Service provided by your carrier then you must configure the route. If they must dial 411 to get information then you will need to configure a route for 411. To do this configure a route as below,

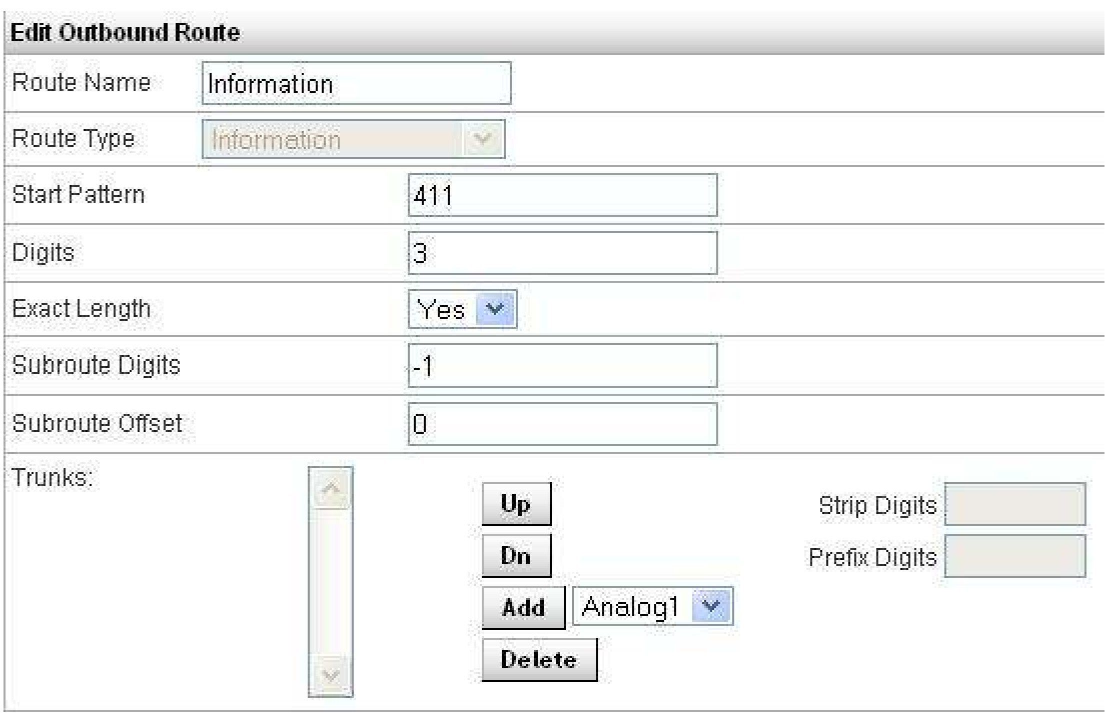


Figure 80 – Information (411) Subroute Setting

Sections/Fields	Description
Route Name	Enter a meaningful name for the outbound route. In this case you may want to call it "**Information**".
Route Type	Select New Route, as this will not be a sub-route of any other route.

Start Pattern	Enter the start pattern of as the code your provider uses for information, in this case “**411**”
Digits	Enter the length of the code your provider uses for information, in this case “**3**”
Exact Length	Set the exact length to “**YES**”
Subroute Digits	Set the Subroute digits to “**-1**” (minus 1).
Subroute Offset	Set the Subroute offset to “**0**”
Trunks	Add any trunks that are appropriate for dialing Information with the code created.

Table 55 – Information (411) Subroute Configuration

IMPORTANT: Please keep in mind that all new routes need to be added to a Class of Service before the route can be used. Subroutes are automatically added under the parent route they use.

PBX Setup

PBX setup is used by a System Administrator to manage system-specific and system-wide settings. Typically these pages will be visited once during installation, and then won’t be changed in the future.

General System Setup

While not visited often, the General settings page pertains to some core PBX components, like administrator Username and Password, Time Settings, System Operator, Mailbox Exit Destination, Directory Type, **Autoprovisioning**, and Security Settings.

Admin Settings Section

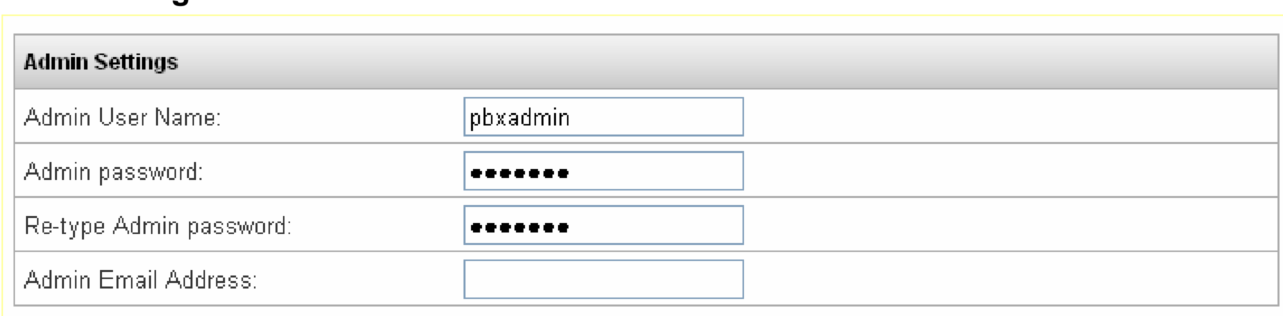


Figure 81 – PBX Admin Settings Section

Sections/Fields	Description
Admin User Name	The administrator user name for the PBX system, default is **pbxadmin**
Admin Password	The administrator password for the PBX system, default is **ipitomy**. Once installed, we recommend you change this to a strong password.

Re-type Admin Password	This requires that you re-enter the password to confirm that they match.
Admin Email Address	This is the email address where administrative alerts will be sent.

Table 56 – PBX Admin Settings Parameters and Descriptions

Edit Admin Settings

STEPS:

1 From the **PBX Setup➔ General System Setup** page, locate the **Admin Settings** section.

2 Enter the administrative information required.

3 Click the 

button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

General Settings Section

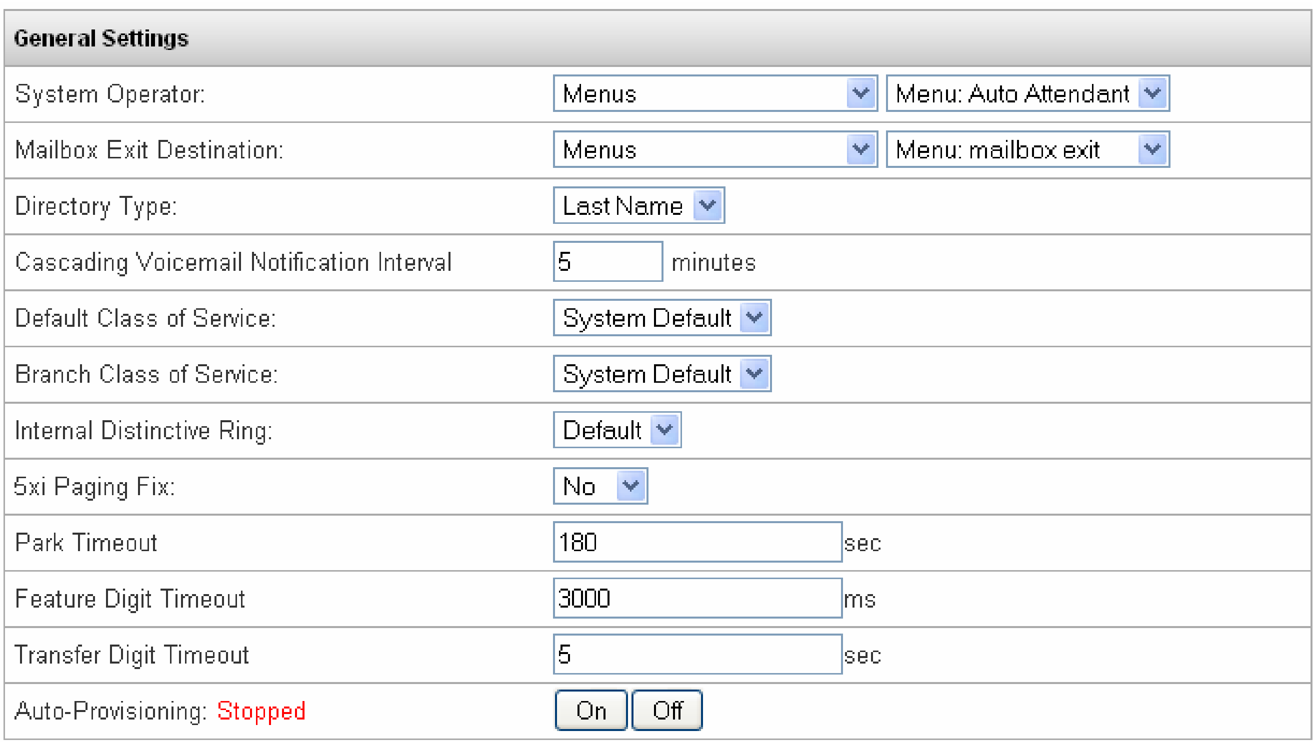


Figure 82 – PBX General Settings Section

Sections/Fields	Description

System Operator	This is the default destination that callers will go to when pressing **'0'** while in a voicemail box, if Allow Operator is enabled on the voicemail box. Additionally, if an extension dials zero, this will be where they are directed.
Mailbox Exit Destination	This is the default destination for a caller if they press **'#'**after leaving a voicemail message.
Directory Type	This determines if the name directory search will use the first or last name.
Cascading Voicemail Notification Interval	This is the frequency with which the PBX will check and send out cascading message notifications.
Default Class of Service	Defines what COS each newly created extension will use unless it is changed.
Branch Class of Service	This is the default COS for a Branch Office using Branch Office Trunking.
Internal Distinctive Ring	This is the type of ring that will be used for internal calls.
5xi Paging Fix	Enabled (set to YES) this parameter only if IPitomy's technical support personnel advises.
Park Timeout	This is how long the PBX will wait before timing out to an extension that has a parked call, which if unanswered will failover to the System Operator. Default value is **180** seconds.
Feature Digit Timeout	This is the timeout parameter for the feature code digit press attempts (in milliseconds). Default value is **3000**.
Transfer Digit Timeout	This is the timeout parameter for inter-digit timing on feature code based transfers. Default value is **5**.
Extension FAX Detect Timer	This is how long, in seconds, the PBX will detect to see if a call ringing to an extension is a fax. The setting is enabled under extension settings. Default value is **4.5**.
Auto-Provisioning	While auto-provisioning is Running, users may use the phone-based auto-provisioning mechanism in IPitomy phones to assign or create extensions. Click on to start, and Off to stop the service. This does not require saving or applying changes.

Table 57 – General PBX Admin Settings and Descriptions

Edit General PBX Settings

STEPS:

1 From the **PBX Setup→ General System Setup** page, locate the **General Settings** section.

2 Enter the information required for the **PBX Security** settings

3 Click the Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Security Settings Section

Log Watch/Ban Service

This feature allows the PBX to monitor the SIP registration traffic and ban any IP address that makes 5 failed registration attempts. The IP address will be banned for either 5 days or until the service is restarted. Configuration settings allow you to ignore (not monitor) IP addresses that are communicating from the LocalNet, as well as allowing you to manually add addresses you want the system to ignore.

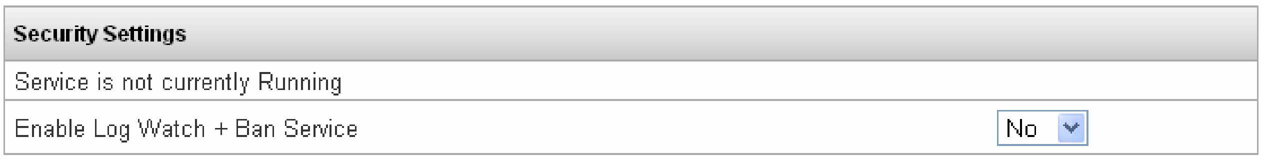


Figure 83 – PBX Security Log Watch and Ban Service Status

Enable Log Watch/Ban Service

STEPS:

1. From the **PBX Setup→ General System Setup** page, locate the **Security Settings** section.

2. **Enable** the **Log Watch Ban Service** by setting the parameter to "**Yes**" on the drop-down list.

3. Click the 

 button to save the changes.

4. Scroll back to the Log Watch Ban Service panel and click the **Reload** button.

Set Log Watch/Ban Service Parameters

Once you have enabled the Log Watch/Ban Services, you will be able to configure the parameters for the security settings.

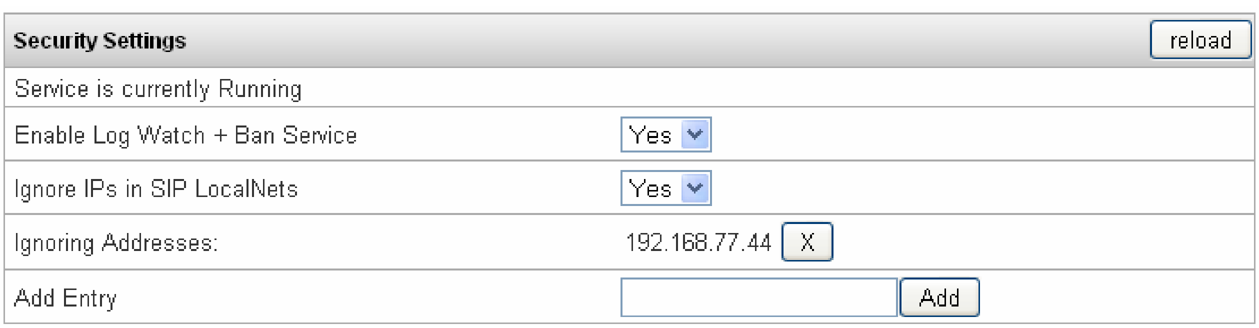


Figure 84 – Security Log Watch and Ban Security Settings

The following table describes the parameters and descriptions (recommended settings) for the Log Watch and Ban feature.

Sections/Fields	Description
Service is currently Running	This indicates the status of the service, either **Running** or **Not Running**.
Reload Button	Restarts the Ban service, clearing any existing banned IP addresses.
Enable Log Watch & Ban Service	Set to Yes and the PBX will monitor failed SIP registration attempts. 5 failed attempts and the IP is blocked for: • 5days • Until system reboot • Until Reload button is clicked
Ignore IPs in SIP LocalNets	Set to **Yes** and the PBX will ignore failed SIP registration attempts from IPs from the LocalNets configured in the system. When set to **No**, IP addresses from the LocalNets are ignored for failed attempts.
Ignoring Addresses	Lists the addresses that were manually entered to be ignored in regards to failed SIP registration attempts
Add Entry	Enter the IP addresses you wish to be ignored by the PBX in regards to failed SIP registration attempts.

Table 58 – PBX Security Log Watch/Ban Settings and Descriptions

Configure Log Watch/Ban Service

STEPS:

1 Navigate to PBX Setup→General.

2 Scroll down to the **Security Settings** panel, set the parameters based on your general business requirements.

3 Click the Save Changes button to save the changes.

4 Navigate to the **PBX Setup→General** page. Locate the **Admin Settings** section (located at the top of the page).

5 Enter an IP address for **Admin Email Address** field. Depending on whether you have Unified Messaging configured, the PBX will send an email to this address any time that it bans an IP address. *See the Unified Messaging section of this guide for details on how to configure this parameter.*

6 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

> **IMPORTANT:** **Please contact IPitomy's Technical Support Group if you need further assistance by email at support@ipitomy.com or phone 941-306-2200 option 2. You can also search the FAQ page on our website at www.ipitomy.com.**

Disable Log Set Security Setting

STEPS:

1 From the **PBX Setup➔ General System Setup** page, locate the **Security Settings** panel.

2 Disable the security setting by selecting "No" from the drop-down list for the **Log Watch Ban Service.**

3 Click the Save Changes button to save the changes.

4 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Time Settings Section

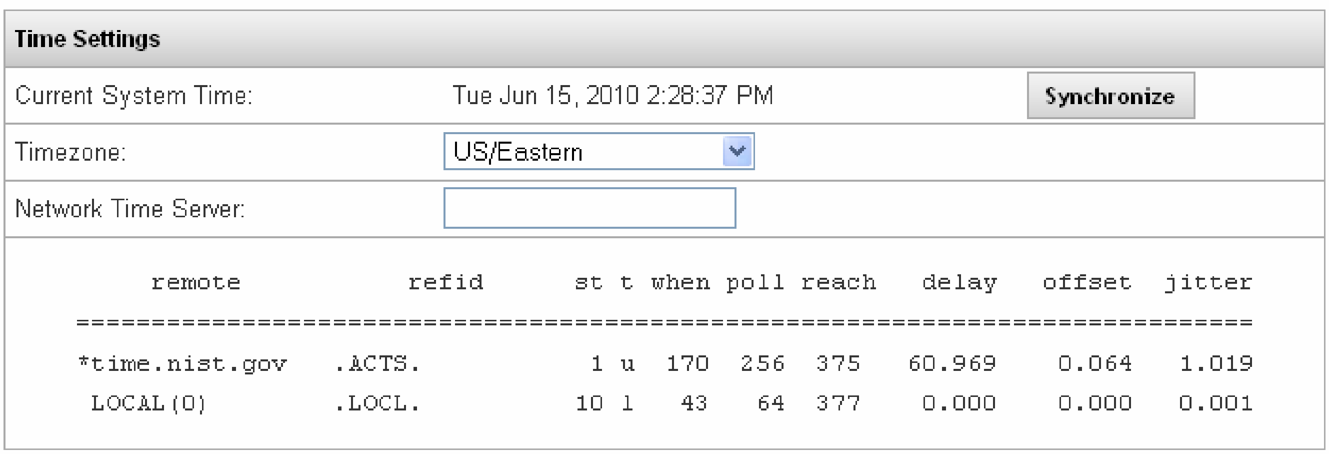


Figure 85 – PBX Time Settings Section

Sections/Fields	Description
Current System Time	This is the current time of the PBX system.
Synchronize	Clicking this button will cause the PBX to sync to the time server. This will result in a restart of services, interrupting calls. If the time is off by more than an hour, we advise you to reboot the PBX after synchronization.
Timezone	This is the time zone for the PBX system.

Network Time Server	This is the location of the PBX time server. If nothing is entered in the field, the PBX is using the default value of time.nist.gov.
Time Server Info	Below the Network Time Server field is a display showing information about the system's connection to its time server.

Table 59 – PBX Time Settings and Descriptions

Set Time Settings

STEPS:

1 From the **PBX Setup➔ General System Setup** page, locate the **Time Settings** panel.

2 Set the Timezone to match your location.

3 If needed, set the Network Time Server to the desired address. Typically the default value will work.

4 Click the Save Changes button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Outbound Transfers Numbers Section

This setting allows outside numbers, such as cell phones or land lines, which receive calls through the PBX to press ## to transfer the call. Once connected, the call can be transferred to someone else at the office or another number, depending on configuration. Simply ask them to hold, then press ##. After a moment the PBX will say “Transfer”, after which you can dial the number where the call should be transferred to. Any numbers listed here will be able to perform this type of transfer.

Use the Transfer Permissions setting to define where calls may be transferred to. Internal Only restricts the transfers to extensions only. If you set it to an extension, then you can transfer anywhere that extension could, following the Class of Service for outbound dialing.

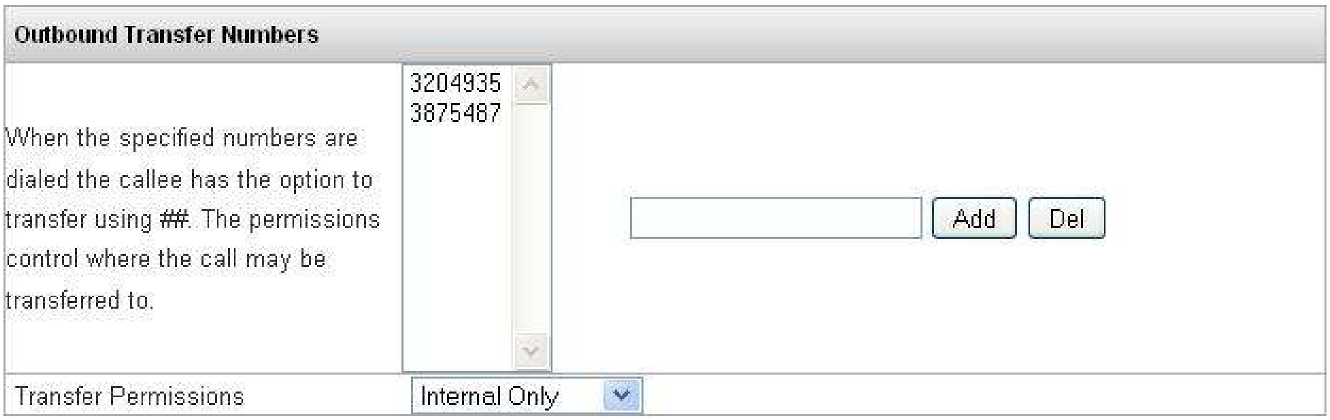


Figure 86 – Outbound Transfer Numbers Section

Add Outbound Transfers

STEPS:

1 From the **PBX Setup➔ General System Setup** page, locate the **Outbound Transfer Numbers** section.

2 To in the box to the left of the **Add** button, enter the phone number you wish to be able to use this transfer feature. Click **Add** to move the number to the list. Repeat this step for all the numbers you want to add.

3 Select the desired permission from the Transfer Permissions drop-down list.

4 Click the 

button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

Delete Outbound Transfers

STEPS:

1 From the **PBX Setup➔General System Setup** page, locate the **Outbound Transfer Numbers** section.

2 From the list of transferable numbers, click on the number(s) that you want to delete. You can use the **SHIFT** or **CTRL** button to select multiple numbers on the list.

3 Click the **DEL** button. The selected numbers are removed from the list .

4 Click the Save Changes button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

InGenius Connector

InGenius is a 3rd party software licensable on the IPitomy PBX. The settings under PBX Setup➔General are where you would configure the user name and password InGenius. Contact IPitomy for more information.

Figure 87 – InGenius Connector Settings Section

Database Administration

Create Backup Section

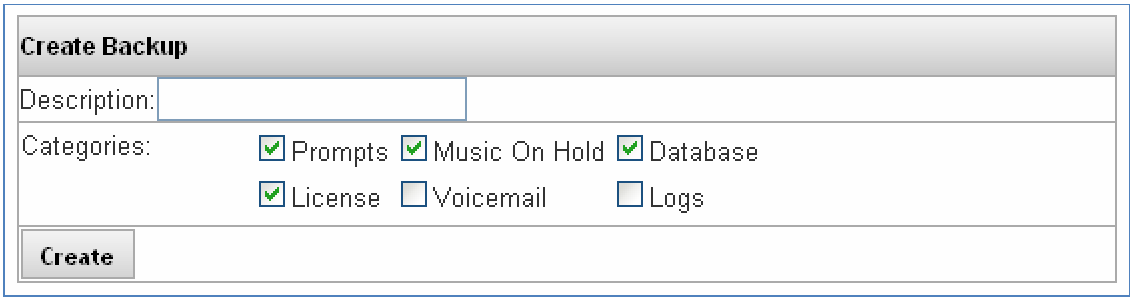


Figure 88 – Create Backup Section

Sections/Fields	Description
Description	Use this field to enter keywords to identify what the backup you are making pertains to.
Categories	This is the types of files that you want to create backups for. Select the desired type by clicking in the box to the left of the item. This will place a checkmark next to the item. Available categories are: • Prompts • Music On Hold • Database • License • Voicemail • Logs
Create	Clicking the Create button will generate a backup file on the PBX containing the information selected under Categories.

Table 60 – Create Backup Settings and Descriptions

Creating a Backup

STEPS:

1 From the **PBX Setup➔Database Administration** page, locate the **Create Backup** section.

2 From the **Create Backup** section, enter the keywords in the **Description** field for the backup file you want to create.

3 Select the categories you want included in the backup.

4 Click on the **CREATE** button to create the backup file.

5 Scroll down to the **Date/Time** section of the **Database** page. The backup file that was just created will appear at the bottom of the list with the corresponding date and time the backup was made. Use your mouse to hover over the date and time field and the keywords entered will be displayed.

Automatic Backups Section

With this feature, you can have the PBX automatically create backups at a set interval, as well as send those backups to an external FTP. The system will only store a total of 4 automated backups at any given time, overwriting the oldest when a new on is created.

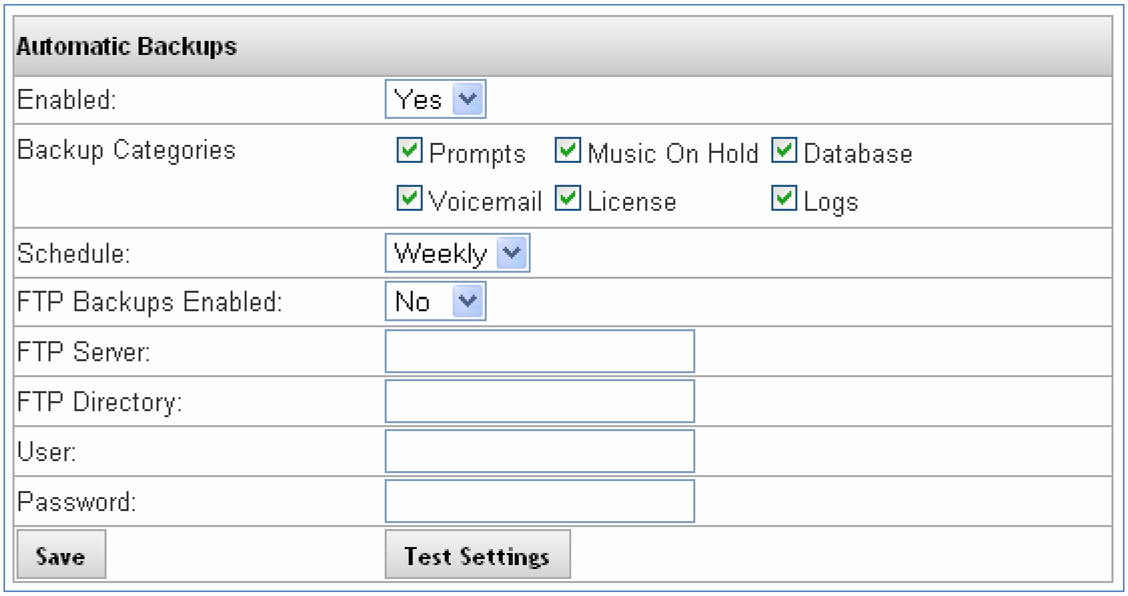


Figure 89 – PBX Database Automatic Backups Section

Sections/Fields	Description
Enabled	If **Enabled** (set to **YES**), a backup will be performed automatically based on the interval set under Schedule.
Backup Categories	This is the types of files that you want to create backups for. Select the desired type by clicking in the box to the left of the item. This will place a checkmark next to the item. Available categories are: • Prompts • Music On Hold • Database • License • Voicemail • Logs
Schedule	This is the frequency for which you want the automated backup to be performed. Options are Daily, Weekly and Monthly.
FTP Backups Enabled	If **Enabled** (set to **YES**), the PBX will send automatic backups to the configured FTP server
FTP Server	This is the FTP server address that the FTP backup will be stored.

FTP Directory	This is the FTP directory on the server where a backup will be stored.
User	Enter the Username required to write files to your FTP
Password	Enter the password associated with the Username used to access your FTP
SAVE Button	This button will save the changes made to the automated backup settings.
Test Settings Button	This button will initiate a test to validate the FTP server, directory, username, and password. A message will display the success or failure, and if successful, a test file will be placed on the FTP

Table 61 – PBX Database Automatic Backup Settings and Descriptions

Set Automatic Backup

STEPS:

1 From the **PBX Setup➔Database Administration** page, locate the **Automatic Backup** section.

2 Enabled the Automated Backup process by setting the **Enabled** field to "**YES**".

3 Select the **Category** that you want to have backed up.

4 Set the **Schedule** to the desired interval.

5 Enter **FTP parameters** and **Username** and P**assword** if you want the backup to be sent to an external FTP.

6 Click the **Save** button to save the settings.

7 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

8 Click the **Test Settings** button to validate the automated backup settings. You should receive a backup "**Successful**" message. If the setup test failed, you receive an "**Error**" with a message indicating what parameters failed.

9 Make the necessary adjustments to the backup parameter, Save, Apply Changes, and try the test again.

Upload Backup File Section

This feature allows you to upload backup files to the PBX system.

Figure 90 – PBX Database Upload Backup File Section

Upload Backup Files

STEPS:

1 From the **PBX Setup→Database Administration** page, locate the **Upload Backup File** section.

2 Click on the **Browse** button to search for the file you want to upload.

3 Double click on the file that you want to upload. The file directory will appear in the box next to the **Browse** button.

4 Click on the **Upload File** button to initiate the upload process. You should receive a message stating that the upload was "**Successful**". If the upload process failed you will receive an "**Error**" message indicating what failed during the process.

Backup Storage Section

The bottom of the **Database Backup** page lists the backup files that are stored on the PBX. The list will provide the following information for each backup file created:

Sections/Fields	Description
Date/Time	The Date and time the backup was created. If you mouse over the date/time for a particular entry, a tooltip with further information like categories and description will display. An asterisk denotes the backup was created by the automated backup feature.
Version	Version of the database when the back up was created.
Size	The size of the backup file.
Action	Delete Backup Restore Backup Download Backup

Table 62 - Saved Backups

Date/Time	Version	Size	Action
May 23, 2010 4:22 AM*	1.0.39	13.19 MB	
May 30, 2010 4:22 AM*	1.0.39	13.29 MB	
Jun 06, 2010 4:22 AM*	1.0.39	13.33 MB	
Jun 13, 2010 4:22 AM*	1.0.39	13.42 MB	
May 21, 2010 2:40 PM	1.0.39	12.23 MB	
May 21, 2010 4:43 PM	1.0.39	13.2 MB	
May 27, 2010 1:12 PM	1.0.39	13.27 MB	
Jun 08, 2010 2:22 PM	1.0.39	13.35 MB	
Jun 10, 2010 10:25 AM	1.0.39	13.38 MB	

* Denotes an automatic backup.

Figure 91 – Backup Date/Time Information

Delete Backup File

STEPS:

1. From the **PBX Setup➔Database Administration** page, scroll to the bottom of the page to find the backup files stored on the PBX.

2. Find the backup file that you want to delete.

3. Select the icon to the right of the backup file name. The file is removed from the list.

4. Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database. This will permanently remove the backup file from the server.

Restore Backup File

STEPS:

1. From the **PBX Setup➔Database** page, scroll to the bottom of the page to find the backup files stored on the PBX.

2. Find the backup file that you want to restore.

3. Select the icon to the right of the backup file name you want to restore.

4. A box will appear prompting you on which values you wish to restore. Select per your needs, and click Restore.

5. Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Download Backup File

STEPS:

1 From the **PBX Setup→Database Administration** page, scroll to the bottom of the page to find the backup files stored on the PBX.

2 Find the backup file you want to download.

3 Select the icon to the right of the backup file name. The system will ask you whether want to open or save the file. Choose Save.

4 Depending on your browser settings, you may have to define where the file should be downloaded to.

Voicemail Setup

This page is used to configure global voicemail box settings, interact with the voicemail archive, as well as configuring the PBX to send out email notifications.

General Settings Section

Used to configure global mailbox settings that are not found elsewhere in the PBX.

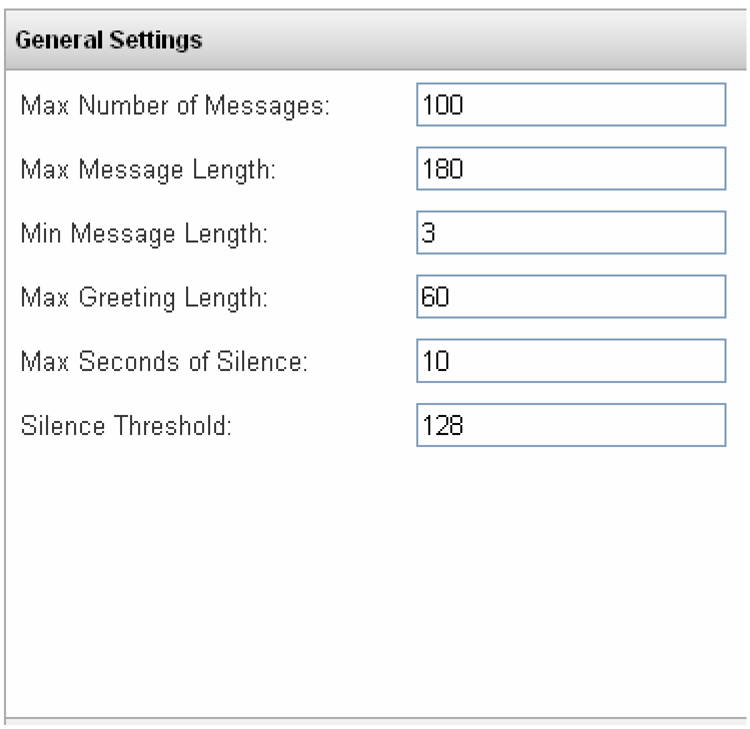


Figure 92 – PBX Voicemail General Settings Section

Sections/Fields	Description
Max Number of Messages	Maximum number of messages allowed for a voicemail box. Default value is **100.**
Max Message Length	Maximum length of a message in seconds. Default value is **180.**
Min Message Length	Minimum length of a message in seconds. Default value is **3.**
Max Greeting Length	Maximum length of a greeting in seconds. Default value is **60.**
Max Seconds of Silence	Maximum seconds of silence before the message is considered complete. Set this to zero for an infinite time period. Default value is **10.**
Silence Threshold	When using the maximum Seconds of Silence setting, it is sometimes necessary to adjust the silence detection threshold to eliminate false triggering on background noise. The higher the number, the more background noise is needed to break the silence. Default value is **128.**

Table 63 – PBX General Settings and Descriptions

Set General Voicemail Settings

STEPS:

1. From the **PBX Setup➔Voicemail Setup** page. Scroll to find the **General Settings** section.
2. Set the General Voicemail parameters base on your business. In most scenarios, the default settings are fine.
3. Click the **Save** button to save the settings.
4. Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Voicemail Menu Section

Configure settings that an individual voicemail box can follow globally.

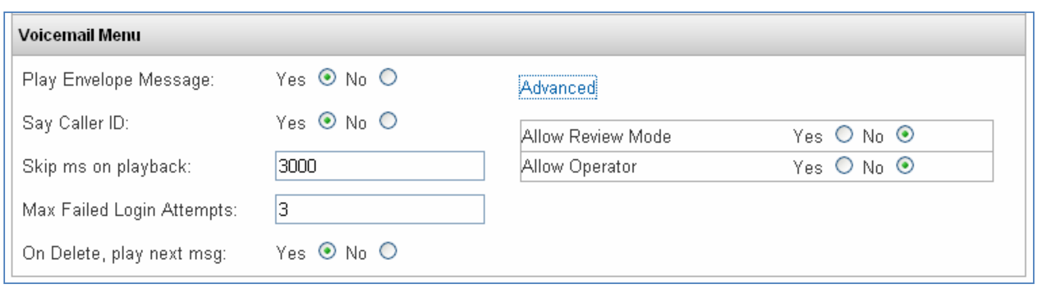


Figure 93 – PBX Voicemail Menu Section

Sections/Fields	Description
Play Envelope Message	Play the envelope message (date/time) before playing the voicemail message. Default value is **Yes.**
Say Caller ID	Play the Caller ID information prior to the message, if available. Default value is **Yes.**
Skip ms on playback	This setting defines an interval in milliseconds to use when skipping forward or reverse while a voicemail message is being played. Default value is **3000.**
Max Failed Login Attempts	The number of retries a user has to enter voicemail passwords before the PBX will disconnect the user. Default value is **3.**
On Delete, play next msg.	Defines if the PBX will automatically play the next message after deleting a voicemail message. Default value is **Yes.**
Allow Review Mode	Global setting that defines if callers are able to listen to a message they left, and then decide if they want to re-record it or leave it as it is. Default value is **No.**
Allow Operator	Global setting in regards to if a voicemail box allows callers to press "**0**" to reach the System Operator

Table 64 – PBX Voicemail Menu Setting and Descriptions

Set Voicemail Menu Options

STEPS:

1 From the **PBX Setup→Voicemail Setup** page, locate the **Voicemail Menu** section.

2 Set the **Voicemail E-mail** parameters base on your business requirements. Typically the default setting will work.

3 Click the 

button to save the settings.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

E-mail Settings Section

This section is used to define the email address and server the PBX will use to send out Unified Messaging and notification emails. Unified Messaging allows a user to receive emails whenever they receive new voicemail messages. If the PC where you check your email has the capacity to play .wav files, the you will be notified when an email is received, and will be able to listen to the new message directly from the email. Notification emails are sent out for features like Log Watch/Ban.

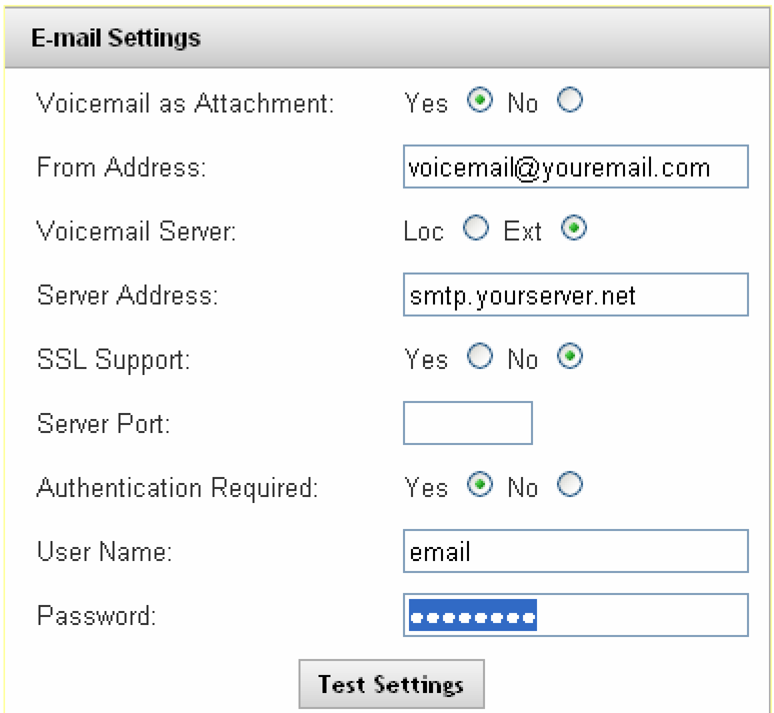


Figure 94 – PBX Voicemail Settings Section

Sections/Fields	Description
Voicemail as Attachment	Defines if new voicemail messages will be attached and set in an email.
From Address	The email address to be used in the From address of the voicemail message. . Note: Most providers require that this is set to the address of the account used for sending the messages.
Voicemail Server	Allows voicemail services to be provided by an external server. It is not recommend to use the “Loc(local)” setting. This setting will be removed in a future release.
Server Address	This is the mail server address. Enter either a fully qualified host name or an IP address.
SSL Support	Enable if the email server uses SSL encryption.
Server Port	If the email server uses a port other than 25, define that port here.

Authentication Required	Defines if the email server requires a username and password to gain access and send messages.
User Name	The user name associated with the use of the external email server.
Password	The external email server password.

Table 65 – PBX Voicemail Settings and Descriptions

Set E-mail Settings

STEPS:

1 From the **PBX Setup→Voicemail Setup** page, scroll to find the **Email Settings** section.

2 Set the **Voicemail E-mail** parameters base on your business requirements or what is recommended by IPitomy.

3 Click the **Save** button to save the settings.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Test Settings Button

STEPS:

1 From the **PBX Setup→Voicemail Setup** page, locate the **E-mail Settings** section.

2 Located at the button of the **E-mail Settings** section is the **Test Settings** button. Click on this button to generate a test of the email settings entered.

3 The “**Email to send a Test message to:**” window appears. Enter the email address that you want the test message sent to then click the **OK** button.

4 If the email setting and address is valid, you will receive a “**SENDEMAIL Successful!**” message at the button of the Test Settings button. The email recipient that was entered will also receive an email with a subject titled “**Voicemail Server Test**”.

5 If the email setting test failed, you will receive an “**Error!**” message at the button of the **Test Settings** button. Check the message for hints as to what caused the failure (ie. SSL enabled but not needed).

6 Reconfigured the email parameters per the Set Email Settings steps above, then test again.

Refer to the Extension Settings Section for information on configuring email addresses for Unified Messaging. Refer to the PBX Admin➔General Section for information on configuring email addresses for Log Watch/Ban.

Voicemail Archive Section

Use this section to view which mailboxes currently have voicemail messages, and to download, upload, or erase the voicemail messages archived on the PBX.

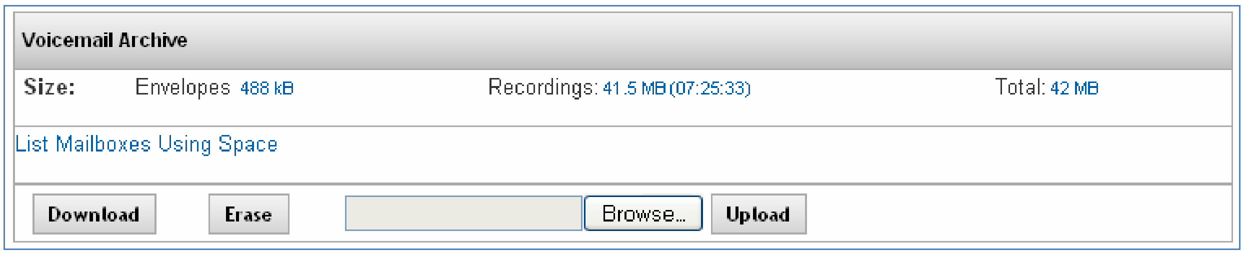


Figure 95 – PBX Voicemail Archive Section

Sections/Fields	Description
Size	Displays the amount of space being used on the PBX for voicemail messages (Envelopes), greetings (Recordings), and the total.
List Mailboxes Using Space	Clicking on this box will display the list of mailboxes and the number of voice messages stored for each.
Download Button	This allows you to download a backup of all user voicemail messages, message envelopes and greeting files to your computer.
Erase Button	Deletes all user voicemail data on the PBX, including personal greetings.
Upload Button	Used to upload a previously downloaded voicemail archive. **IMPORTANT: Uploading a voicemail archive will overwrite existing voicemail files on the PBX.**

Table 66 – PBX Voicemail Archive Settings and Descriptions

Download Voicemail Archive Settings

STEPS:

1 From the **PBX Setup➔Voicemail Setup** page, locate the **Voicemail Archive** section.

2 Click on the **Download** button. Select **Save File** if prompted by your browser.

3 Define the location you wish to save the file if prompted by your browser.

Erase Voicemail Archive Settings

STEPS:

1 From the **PBX Setup→Voicemail Setup** page, locate the **Voicemail Archive** section.

2 Click on the **Erase** button. The warning message "*This will delete all Voicemail recordings and user greeting. If you are sure you want to do this, click OK.*"

3 Click the **OK** button to continue. The system performs the erase and returns you to the **Voicemail Setup** page.

Upload Voicemail Archive Settings

STEPS:

1 From the **PBX Setup→Voicemail Setup** page, locate the **Voicemail Archive** section.

2 Click on the **Browse** button to search for the file location of the file you want to upload.

3 Click on the file that you want to upload. The file directory will appear in the box next to the **Browse** button.

4 Click on the **Upload** button to initiate the upload process. You should receive a message stating that the upload was "**Successful**". If the upload process failed you will receive an "**Error**" message indicating what failed during the process.

5 Click the Save Changes button.

6 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

View Voicemail Listing (Usage Space)

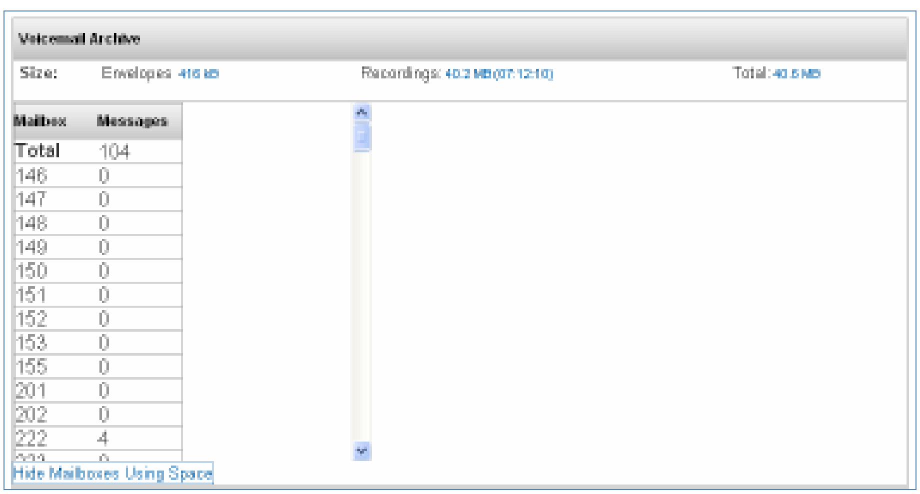


Figure 96 – PBX Voicemail Listing Page

STEPS:

1 From the **PBX Setup➔Voicemail Setup** page, locate the **Voicemail Archive** section.

2 Click the **List Mailboxes Using Space** link. A listing of the Mailboxes (extensions) and number of messages for each mailbox appears.

3 Click on the **Hide Mailboxes Using Space** link to close the listing.

SIP Setup

SIP Networking Settings Section

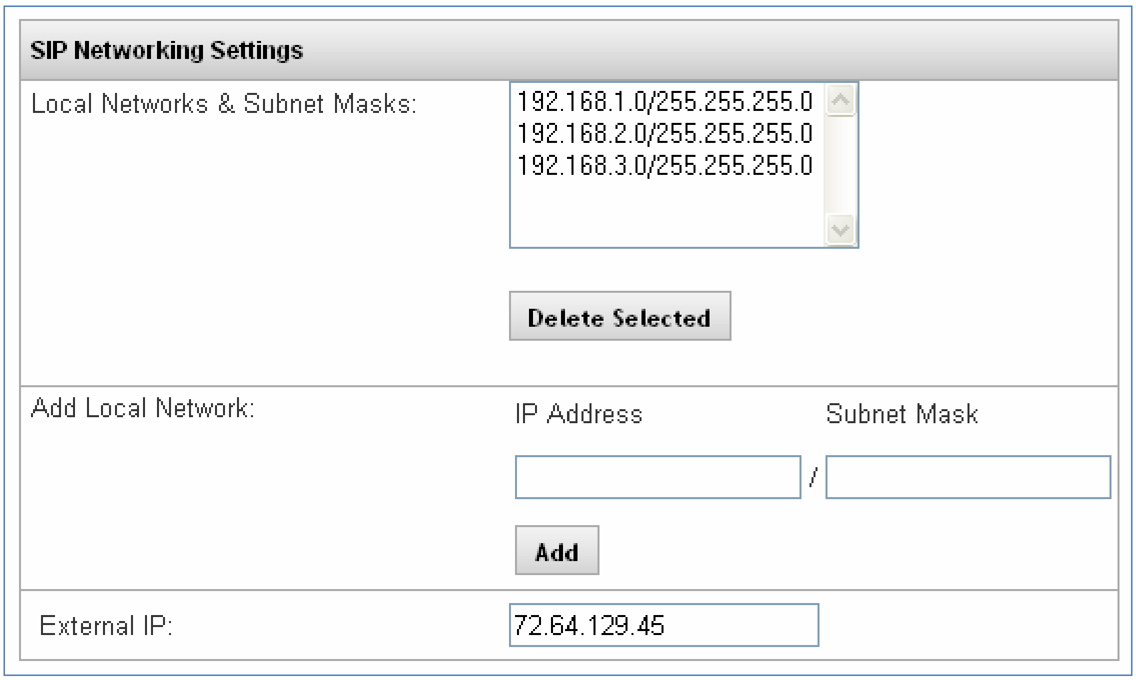


Figure 97 – SIP Networking Settings Page

Sections/Fields	Description
Local Network & Subnet Masks	This is the IP address to the local network and associated subnet masks.
Delete Selected	Allows you to delete the item selected from list of networks.
Add Local Network	Allows you to add local network information. This information will appear in the list of networks.
External IP	The public IP address associated with the SIP network.
Add Button	This button will create/add the network information to the list of local network section.

Table 67 – SIP Networking Settings and Descriptions

Add SIP Networking Settings

STEPS:

1 From the **PBX Setup→SIP** page, locate the **SIP Networking Settings** section.

2 Enter the IP Address and Subnet Mask for the network the PBX is being installed on.

3 Click the Add button.

4 Click the Save Changes button.

5 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Delete SIP Networking Settings

STEPS:

1 From the **PBX Setup→SIP** page, locate the **SIP Networking Settings** section.

2 Highlight the listing you wish to delete. You can use Shift/Ctrl click functionality to select multiple listings.

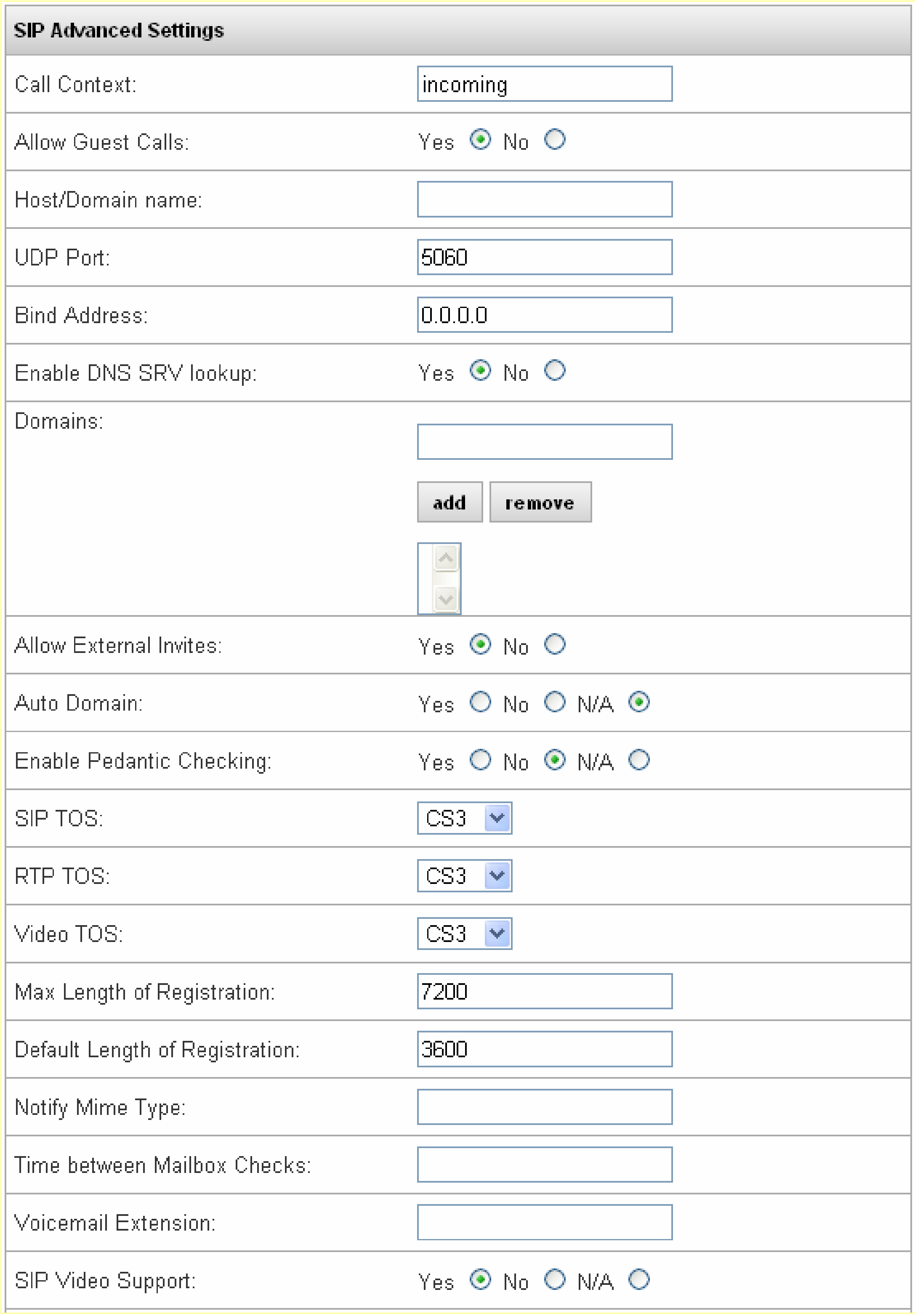
3 Click the Delete Selected button.

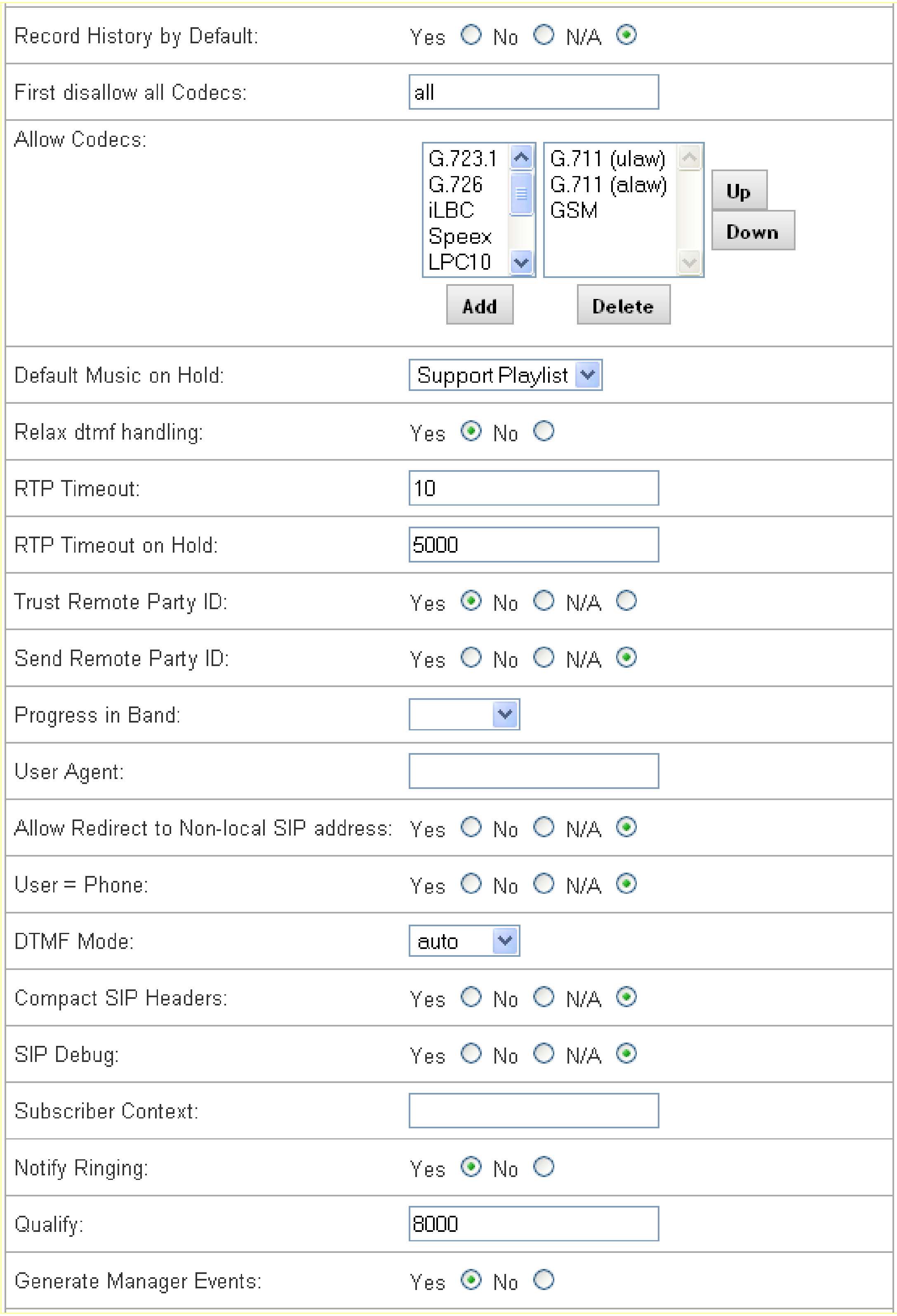
4 Click the Save Changes button.

5 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database

SIP Advanced Settings Section

Advanced SIP settings define in more detail the management of network traffic. These settings are automatically provisioned when the system registers with the router. In most business implementations it is not necessary to make changes to these defaulted settings


SIP Advanced Settings
Call Context:
incoming
Allow Guest Calls:
Yes No
Host/Domain name:
UDP Port:
5060
Bind Address:
0.0.0.0
Enable DNS SRV lookup:
Yes No
Domains:
add
remove
Allow External Invites:
Yes No
Auto Domain:
Yes No N/A
Enable Pedantic Checking:
Yes No N/A
SIP TOS:
CS3
RTP TOS:
CS3
Video TOS:
CS3
Max Length of Registration:
7200
Default Length of Registration:
3600
Notify Mime Type:
Time between Mailbox Checks:
Voicemail Extension:
SIP Video Support:
Yes No N/A


Record History by Default: Yes No N/A
First disallow all Codecs: all
Allow Codecs:
G.723.1
G.726
iLBC
Speex
LPC10
G.711 (ulaw)
G.711 (alaw)
GSM
Up
Down
Add
Delete
Default Music on Hold: Support Playlist
Relax dtmf handling: Yes No
RTP Timeout: 10
RTP Timeout on Hold: 5000
Trust Remote Party ID: Yes No N/A
Send Remote Party ID: Yes No N/A
Progress in Band:
User Agent:
Allow Redirect to Non-local SIP address: Yes No N/A
User = Phone: Yes No N/A
DTMF Mode: auto
Compact SIP Headers: Yes No N/A
SIP Debug: Yes No N/A
Subscriber Context:
Notify Ringing: Yes No
Qualify: 8000
Generate Manager Events: Yes No

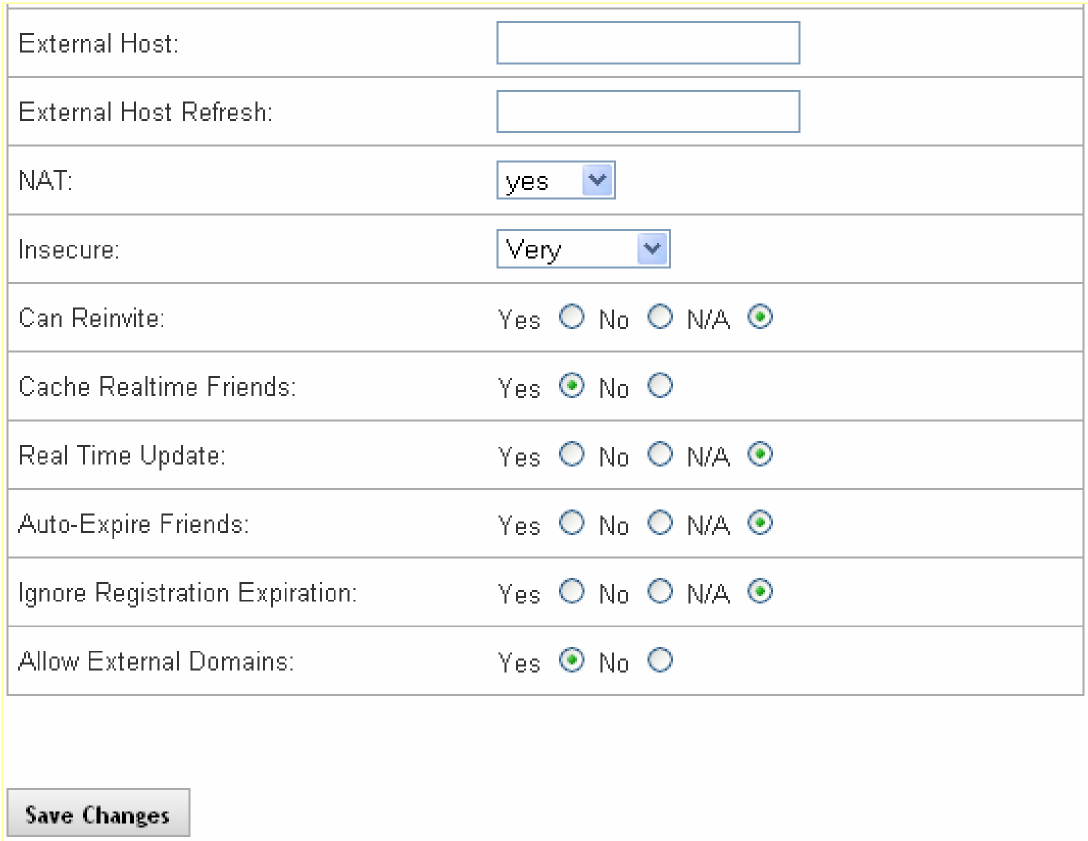


Figure 98 – SIP Advanced Settings Page

IMPORTANT: The default settings for the SIP configuration should not require any changes. If it is necessary for you to do so to meet your customer's business requirements, we recommend that you contact IPitomy's Technical Support for assistance.

Sections/Fields	Description/Default Parameters
Call Context	Default: **INCOMING**
Allow Guest Calls	Default: **YES**
Host/Domain Name	Default: **BLANK**
UDP Port	Default: **5060**
Bind Address	Default: **0.0.0.0**
Enable DNS SRV Lookup	Default: **NO**
Domains	Default: **BLANK**

Allow External Invites	Default: **YES**
Auto Domain	Default: **N/A**
Enable Pedantic Checking	Default: **N/A**
SIP TOS	Default is **CS3**. To configure QOS on your LAN, set your managed switches to prioritize packets flagged with CS3
RTP TOS	Default is **CS3**. To configure QOS on your LAN, set your managed switches to prioritize packets flagged with CS3
Video TOS	Default is **CS3**. To configure QOS on your LAN, set your managed switches to prioritize packets flagged with CS3
Max Length of Registration	Default: **7200**
Default Length of Registration	Default: **3600**
Notify Mime Type	Default: **BLANK**
Time Between Mailbox Checks	Default: **BLANK**
Voicemail Extension	Default: **BLANK**
SIP Video Support	Default: **YES**
Record History of Default	Default: **N/A**
First disallow all Codecs	Default: **ALL**
Allow Codecs	Default: **G.711 ulaw, G.711 alaw, GSM**
Default Music on Hold	This will display whatever playlist is set to default on the PBX Setup=>Music On Hold page
Relax DTMF Handling	Default: **YES**
RTP Timeout	Default is **BLANK**. Set to a value, in seconds, if you wish the PBX to end a call when no RTP traffic is detected for that long. Typically used in regards to lines that are not disconnecting correctly.
RTP Timeout on Hold	Default: **BLANK**
Trust Remote Party ID	Default: **N/A**

Send Remote Party ID	Default: **N/A**
Progress in Band	Default: **BLANK**
User Agent	Default: **BLANK**
Allow Redirect to Non-local SIP Address	Default: **N/A**
User = Phone	Default: **N/A**
DTMF Mode	Default: **AUTO**
Compact SIP Headers	Default: **N/A**
SIP Debug	Default: **N/A**
Subscriber Context	Default: **BLANK**
Notify Ringing	Default: **YES**
Qualify	Default: 8000
Generate Manager Events	Default: **YES**
External Host	Default is **BLANK**. When the site is using a dynamic public IP, you can go to dyndns.com, set up a domain with them, and enter that domain name here.
External Host Refresh	Default is **BLANK**. Set this under the interval used at dyndns.com for checking the current IP.
NAT	Default: **YES**
Insecure	Default: **VERY**
Can Reinvite	Default: **N/A**
Cache Realtime Friends	Default: **YES**
Real Time Update	Default: **N/A**
Auto-Expire Friends	Default: **N/A**
Ignore Registration Expiration	Default: **N/A**
Allow External Domains	Default: **YES**

Table 68 – SIP Advanced Settings and Descriptions

Edit Advanced SIP Networking Settings

STEPS:

1 From the **PBX Setup→SIP** page, click on the **Advanced** link.

2 The **Advanced SIP Networking Settings** page is displayed.

3 Set the **SIP Network** parameters base on your business requirements or what is recommended by IPitomy.

The default settings should not require any changes. If it is necessary for you to do so to meet your customer's business requirements, we recommend that you contact IPitomy's Technical Support for assistance..

4 Click the Save Changes button.

5 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Prompts

Prompts are voice recordings used by a Menu to define the options that can be selected in the menu, or to simply convey information.

Upload Voice Prompt Section

This feature allows you to upload a voice prompt file and store it in the PBX server for use. Once loaded, you can select which menus you wish to use the Prompt file.

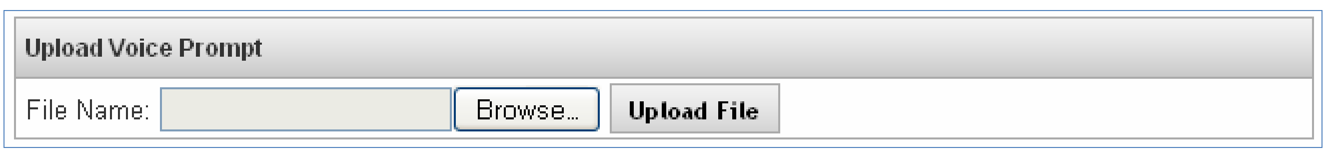


Figure 99 – Upload Voice Prompt Section

Upload Voice Prompt

STEPS:

1 From the **PBX Setup→Edit Prompts** page, locate the **Upload Voice Prompt** section.

2 Click on the **Browse** button in the **Upload Voice Prompt** section of the page.

3 Specify the location of the voice prompt file you want to upload (the accept file format is .gsm, the file name should not contain any spaces or special characters)

4 Click the **Upload File** button.

5 Once the file is uploaded it will be listed in the **Prompt Files on Server** section.

6 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Record New Voice Prompt Section

This feature allows you to record a new voice prompt and store on the PBX server. Once created you can define which menus you wish to use the Prompt file.

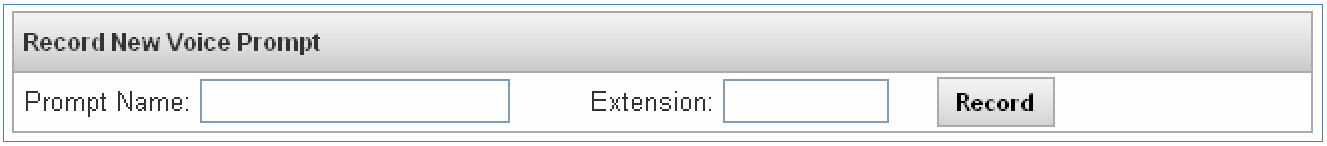


Figure 100 – Record New Voice Prompt Section

Record New Voice Prompt

STEPS:

1. From the **PBX Setup➔Edit Prompts** page, locate the **Record New Voice Prompt** section.

2. Enter a description in the **Prompt Name** and the **Extension** number of the person who will be recording the prompt.

3. Click on the **Record** button. The system will display a message stating that it is trying to call the extension. When the connection is successful, the phone at that extension will ring.

4. Answer the call and record the message. Be sure to follow the in-call directions through till the system says ‘Goodbye”.

5. When you have completed the recording, press the **Continue** button. The system will return you to the **Edit Prompts** page.

6. Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Prompt Files on Server Section

All record prompt files are stored on the PBX server and will appear in the **Prompt Files on Server** listing of the **Edit Prompts** page.

Prompt Files on Server			
File Name	**Size**	**Delete**	**Download**
1-sec-silence.gsm	1650		
1.gsm	1650		
aable.gsm	7722		
AnimalHealth_day_1-2.gsm	40491		
Christmas.gsm	16896		
enter-ext-of-person.gsm	4620		

Figure 101 – Prompt Files on the Server Section

Delete Prompt Files on Server

STEPS:

1 From the **PBX Setup→Edit Prompts** page, locate the **Prompt Files on Server** list.

2 Select the icon to the right of the prompt file, listed under the **File Name** column, that you want to delete. The file is removed from the list.

3 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Download Prompt Files on Server

STEPS:

1 From the **PBX Setup→Edit Prompts** page, locate the **Prompt Files on Server** list.

2 Select the icon to the right of the prompt file, listed under the **File Name** column, you want to download.

3 Specify the location where you want to download voice prompt file. Once the download is complete, the system will return you to the **Edit Prompts** page.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Music on Hold

The PBX allows you to create multiple play lists that can be populated with music files. Once a playlist is configured, you can define for this to be the System Default Hold music, or specify to use this playlist under individual Extensions, Groups, or Menus.

Systems Default Music on Hold Section

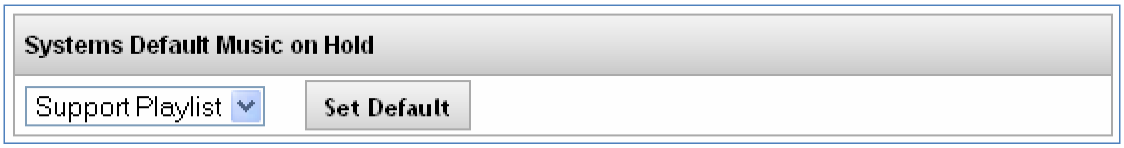


Figure 102 – System Default Music on Hold Section

Set Default Music on Hold Section

STEPS:

1 From the **PBX Setup→Music on Hold** page, locate the **Systems Default Music on Hold** section.

2 Select the desired **Music on Hold** Playlist from the drop-down list.

3 Click on the **Set Default** button.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Create a Playlist Section

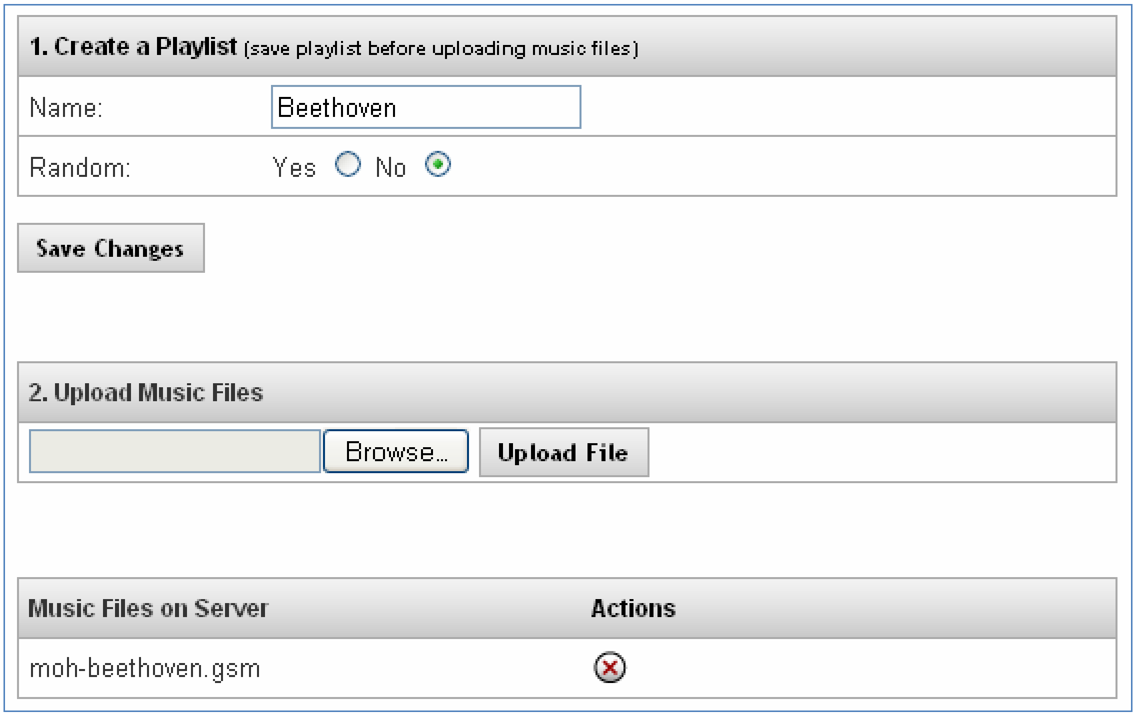


Figure 103 – Create a Playlist Section

Add New Music on Hold

STEPS:

1 From the **PBX Setup➔Music on Hold** page, locate the **Create a Playlist** section.

2 Enter a single word description of the playlist in the **Name** field.

3 Set the **Random** preference to either **YES** or **NO**.

4 Click the 

button to save the changes.

5 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

6 The new Playlist name is added to the playlist on the **Music on Hold** page.

Upload Music Files

Name	Play Mode	Random	Action
Beethoven	files	no	
Rock	files	no	
jazz	files	no	
Christmas	files	yes	
Ringing	files	no	
Support Playlist	files	no	

Figure 104 – Music on Hold Playlist

The following steps describe how to upload Music on Hold file to the PBX system.

STEPS:

1 From the **PBX Setup→Music on Hold** page, locate the **Name** of the **Music on Hold** playlist you wish to upload the music file to.

2 Click on the **Name** link from the list of Music on Hold files or the icon to the right of the name.

3 The **Edit Music on Hold** page appears. Click on the **Browse** button in the **Upload Music on Hold** section of the page.

4 Specify the location of the music file you want to upload (accept formats are .mp3, .wav or .gsm) to the PBX system then click the **Upload File** button.

5 Click the Save Changes button to save the changes.

6 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Delete Music Files on Server

STEPS:

1 From the **PBX Setup→Music on Hold** page, locate the **Name** of the **Music on Hold** file that you want to delete.

2 Select the icon to the right of the music file **Name** that you want to remove. The file name is removed from the list of music files.

3 Click the Save Changes button to save the changes.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Feature Codes

This section lists a number of basic codes users can dial from their phone to access certain features on the PBX. Additionally, at the bottom of the page you can find a section that allows for configuration of System Speed Dials.

Speed Dialing Section

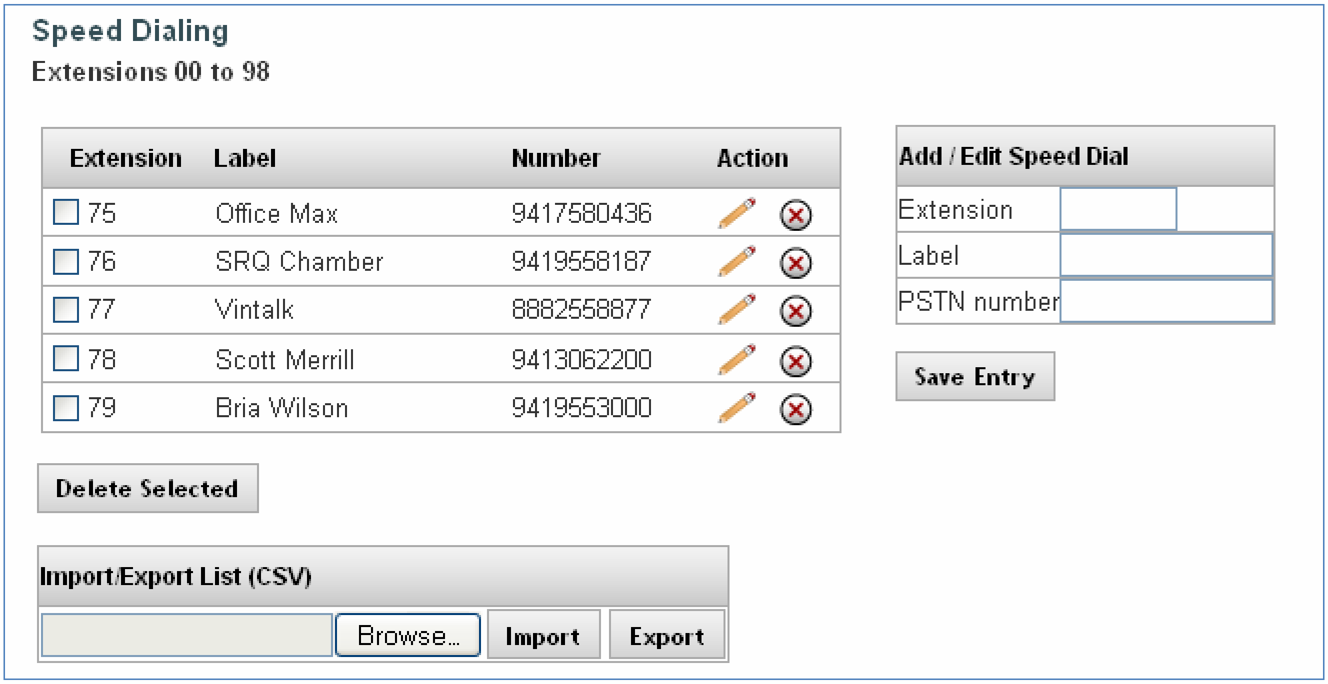


Figure 105 – Features Code Speed Dialing Section

Sections/Fields	Description
Extension	Enter a 2 digit code between 00 and 98. This is the code the user will enter to access the speeddial.
Label	Enter a recognizable value that relates to the number being dialed
PSTN number	This is the phone number that will be dialed when the code is dialed. Only enter digits that could be dialed from a phone (numbers, *, #).

Table 69 – Features Code Settings and Descriptions

Add Speed Dialing

STEPS:

1 From the **PBX Setup→Feature Codes** page, locate the **Speed Dialing** section.

2 Enter the **Extension** number, **Label** name and **PSTN** number for the speed dial in the **Add/Edit Speed Dial** section.

3 Click on the **Save Entry** button to save the information. The system will create the speed dialing code and it will appear in the listing.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Edit Speed Dialing

STEPS:

1 From the **PBX Setup→Feature Codes** page, locate the **Speed Dialing** section.

2 Click on the icon to the right of the speed dial code you want to edit. The information for the speed dial code will appear in the **Add/Edit Speed Dial** section.

3 Edit the necessary information then click on the **Save Entry** button to save the information.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Delete Speed Dialing

STEPS:

1 From the **PBX Setup→Feature Codes** page, locate the **Speed Dialing** section.

2 Select the icon to the right of the speed dial code you want to remove. To remove multiple speed codes at one time, click on the box to the left of the codes then click the **Delete Selected** button.

3 The file name is removed from the list of speed dial codes.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Import/Export Speed Dialing List (CSV)

STEPS:

1 From the **PBX Setup→Feature Codes** page, locate the **Import/Export List** section.

2 Click on the **Browse** button in that section of the page.

3 Specify the location of the file you want to import from or export to and then click the appropriate function button (import or export).

4 The system will perform the function you selected and will tell you whether it was successful in the attempting to process the request.

5 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Services

System Information Section

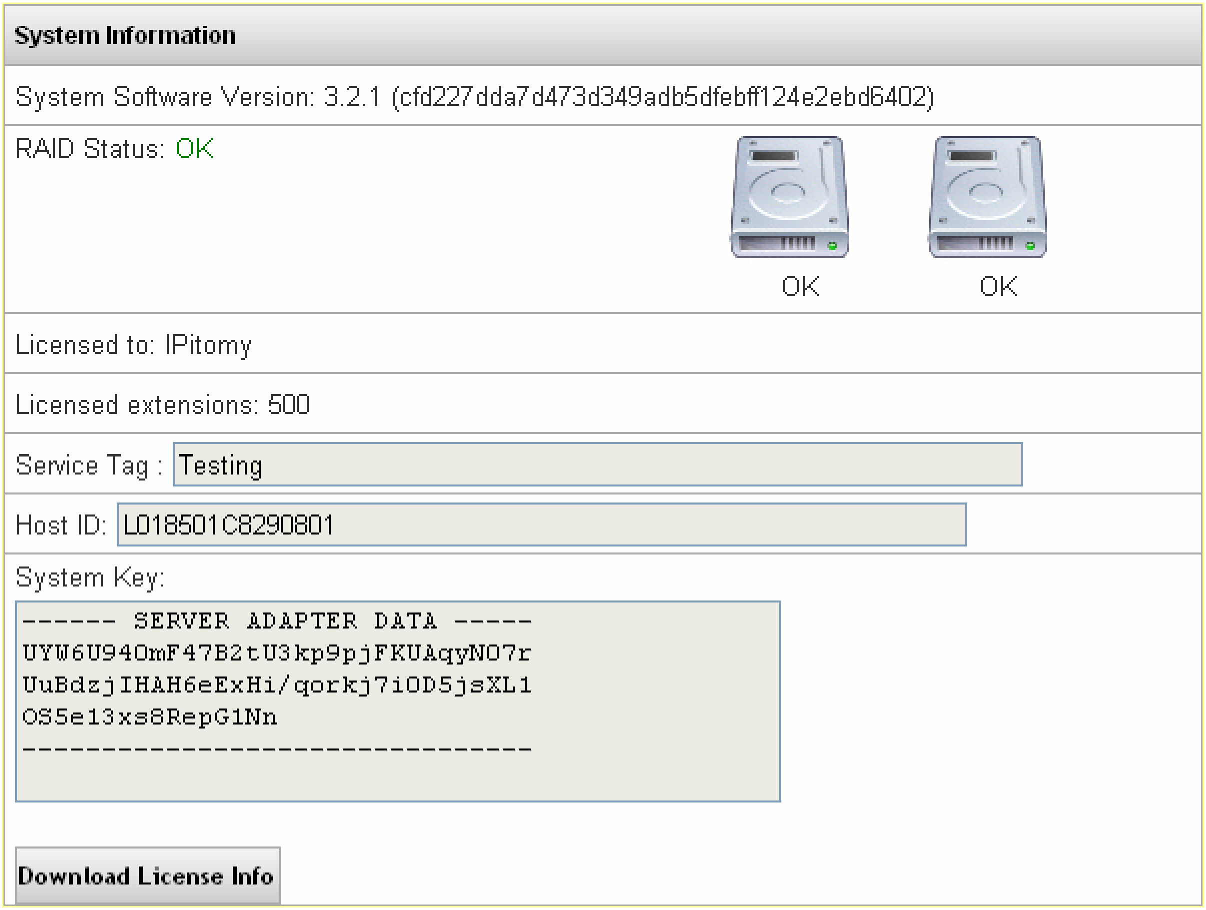


Figure 106 – Services System Information Page

Sections/Fields	Description
System Software Version	This is the software version that is running on the PBX system.
RAID Status	Defines the status the system Raid (if it has one). RAID is Redundant Array of Independent Disks. RAID is available in Software and Hardware versions. 1100 and 1200 systems can only deploy SW RAID. 2000 and 5000 systems are typically deployed using Hardware RAID.
Licensed to	Displays the name of the company the PBX is licensed to.
Licensed extensions	Displays how many extensions the system is licensed for. If you try to create more extensions they the PBX has licenses, a warning will display to alert you. There are licenses for both IPitomy branded telephones and non-IPitomy branded telephones. Telephones are assigned the correct license automatically.
Service Tag	Unique serial number for the PBX, used when licensing the system.

Host ID	Unique ID used for licensing the system.
System Key	Unique key used for licensing the system.
Download License Info Button	Clicking this button will save a text file on your PC. This file contains the Unique ID, Serial, and Key of the PBX. Be sure to email this file along with your purchase order when buying additional licenses,

Table 70 – Services System Settings and Descriptions

View System Information

STEPS:

1 From the **PBX Setup→Services** page, locate the **System Information** section.

2 Use the table of settings and their description for an explanation of how the system is configured.

License Information Section

This feature allows you view what licensed features are available on the system. You can upload or download licenses for the PBX applications. You can also assign Call Manager licenses to users. The following section describes how to perform these functions.

Figure 107 – Services Download License Information Section

Sections/Fields	Description
Features	Defines which additional features are licensed on the system. Available features are: ACD InGenius Multisite Call Scheduling
Desktop Call Manager Licenses	Displays how many of each Call Manager member type are licensed on the PBX. Currently **Operator** and **User** are the only functional license. Clicking the **Assign User Licenses** button will open a new window for assigning call manager licenses.
License File	In this panel you can download the current license file, or upload a new license.

Table 71 - License Info Descriptions

Upload License File

STEPS:

1 From the **PBX Setup➔Services** page, locate the **License Info** section.

2 Click on the **Browse** button in that section of the page.

3 Specify the location of the file you want to upload, then click **Upload File** button.

4 The system will log you out of the PBX.

5 Log back into the PBX.

6 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Download License File

STEPS:

1 From the **PBX Setup➔Services** page, locate the **License Info** section.

2 **Click the Download File** button**.**

3 Your browser may prompt you as to where you wish to save the file.

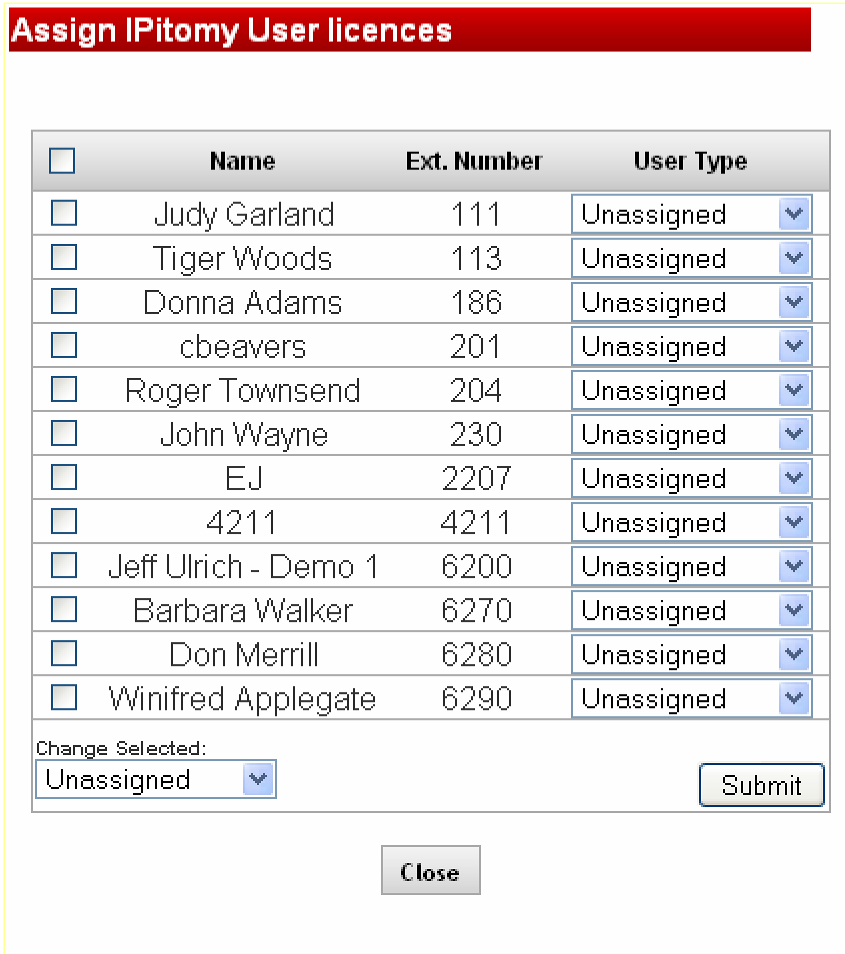


Figure 108 – Services Assign Licenses Page

Assign Single User License

STEPS:

1 From the **PBX Setup➔Services** page, locate the **License Info** section.

2 Click the **Assign User Licenses** button. The "**Assign IPitomy User Licenses**" window appears. Find the name of the extension that you want to assign the license to. Choose the desired license type from the drop-down list (User/Operator).

3 Click the **Submit** button to save the changes. Click the **Close** button to close the active window.

4 Click the 

button to save the changes.

5 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

*If this is the first time a license has been assigned, be sure to return to the servies page and click the **Restart Call Manager** button.*

Assign Multiple User Licenses

STEPS:

1 From the **PBX Setup➔Services** page, locate the **Download License info** section.

2 Click the **Assign User Licenses** button. The "**Assign IPitomy User Licenses**" window appears. Click on the box (place a checkmark) to the left of each of the extensions that you want to assign the licenses to.

3 Scroll down to the button of the page and select the license type (User/Operator) to assign to these extensions.

4 Click the **Submit** button to save the changes. Click the **Close** button to close the active window.

5 Click the 

button to save the changes.

6 Click the **Apply Changes** link located at the top right hand corner of the page, to commit the changes to the database.

System Functions Section

This section allows the administrator to restart some or all services in the PBX, as well as clear the Diagnostic log.

Figure 109 – Services System Functions Section

Sections/Fields	Description
Restart PBX Daemon	This will stop and restart all asterisk services on the PBX. **IMPORTANT: Performing this function will drop all active calls.**
Reboot the PBX	This reboot the PBX. **IMPORTANT: Performing this function will drop all active calls.**
Restart All Services	This will restart all services, including asterisk, drivers, and web services of the PBX. **IMPORTANT: Performing this function will drop all active calls.**
Restart CallManager Daemon	This will reinitialize Call Manager Sevices. It will drop any active instances of Call Manager.
Clear Diagnostic Log	This will clear the Diagnostics logs from the PBX system.

Table 72 – Services System Functions and Descriptions

Restart PBX

STEPS:

1 From the **PBX Setup➔Services** page, locate the **System Functions** section.

2 Click on the **Restart PBX** button. All asterisk services on the PBX and return you to the **Services** page when it is done.

Reboot PBX

STEPS:

1 From the **PBX Setup➔Services** page, locate the **System Functions** section.

2 Click on the **Reboot PBX** button. The will perform a software reboot, and require you to log back in when it is done.

Restart Services

STEPS:

1 From the **PBX Setup➔Services** page, locate the **System Functions** section.

2 Click on the **Restart Services** button. The system will restart all the services that are running and return you to the **Services** page when it is done.

Restart Call Manager

STEPS:

1 From the **PBX Setup➔Services** page, locate the **System Functions** section.

2 Click on the **Restart Call Manager** button. The system will restart process for the CallManager application and return you to the **Services** page when it is done.

Clear Diagnostics

STEPS:

1 From the **PBX Setup➔Services** page, locate the **System Functions** section.

2 Click on the **Clear Diagnostics** button. The system will clear all the diagnostic messages on the PBX server and return you to the **Services** page when it is done.

Load File System

This area is used to load software files when performing an upgrade.

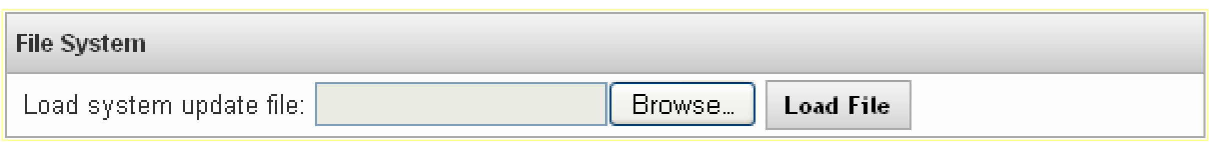


Figure 110 - Services Load File System Section

STEPS:

1 From the **PBX Setup➔Services** page, locate the **File System** section.

2 Click on the **Browse** button in that section of the page.

3 Specify the location of the file you want to load. The file must have the .tgz file extension.

4 The PBX will load the file, and the PBX will reboot.

5 Upon logging in, a prompt declaring the database version does not match will appear, Click **OK** and the database will be converted.

6 Click on the **Apply Changes** link at the top of the page.

TFTP Files

This panel allows the administrator to view, load, download, and delete files on the TFTP in the PBX. Typically, the files you would be loading will be phone firmware files.

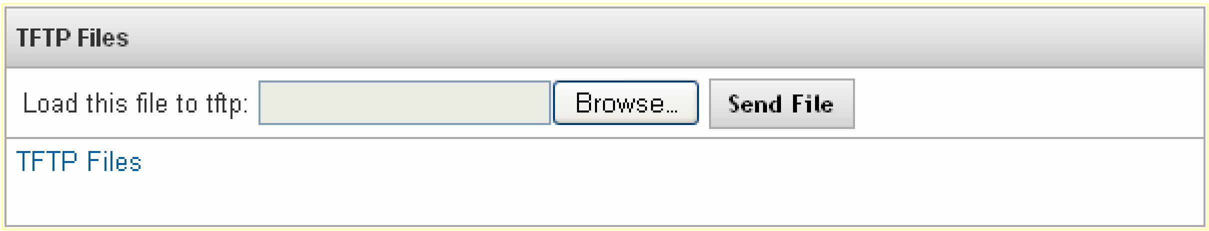


Figure 111 – Services TFTP Files Section

Load TFTP File

STEPS:

1 From the **PBX Setup→Services** page, locate the **TFTP Files** section.

2 Click on the **Browse** button in that section of the page.

3 Specify the location of the file you want to load onto the PBX.

Download TFTP File

STEPS:

1 From the **PBX Setup→Services** page, locate the **TFTP Files** section.

2 Click on the blue **TFTP Files** link. A listing of the files that have been loaded on the TFTP server will be displayed.

3 Click on the blue link for the file you want to download. Your browser may require you to specify the location where you want to download the TFTP file.

Delete TFTP File

STEPS:

1 From the **PBX Setup→Services** page, locate the **TFTP Files** section.

2 Click on the blue **TFTP Files** link. A listing of the files that have been loaded on the TFTP server will be displayed.

3 Find the name of the file you want to delete from the list.

4 Click on the **Delete** button. The file is removed from the list and the system will return you to the **Services** page.

Log File Settings

This feature allows you indicate whether you want the logs to be written to a file or displayed on the PBX webpage.

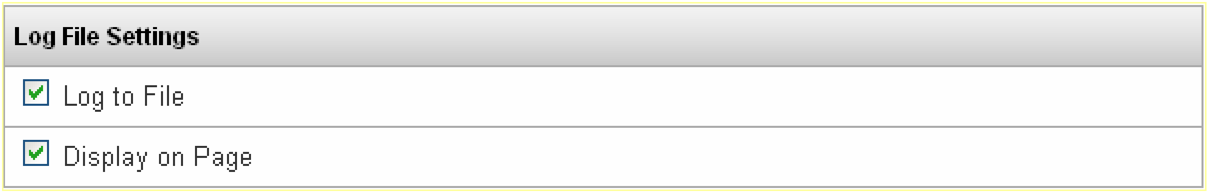


Figure 112 – Services Log File Settings Section

Sections/Fields	Description
Log to File	If selected – the system will write the log messages to the **ipitomy.log** file, accessible by IPitomy Support. Default is **enabled.**
Display on Page	If selected – the system will display the log messages on the System Diagnostics page, accessable by anyone with Administrator access to the PBX. Default is **enabled.**

Table 73 – Services Log File Settings and Descriptions

Select Log File Settings

STEPS:

1 From the **PBX Setup➔Services** page, locate the **Log File Settings** section.

2 Select (placing a checkmark in the box) the method that you want the log files to be handled. You can elect to do both, if so desired.

3 Click the 

button to save the changes.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Logging Level Section

This section of the Services page allows you to specify the type of messages that will be captured and available on the **System Diagnostics** page.

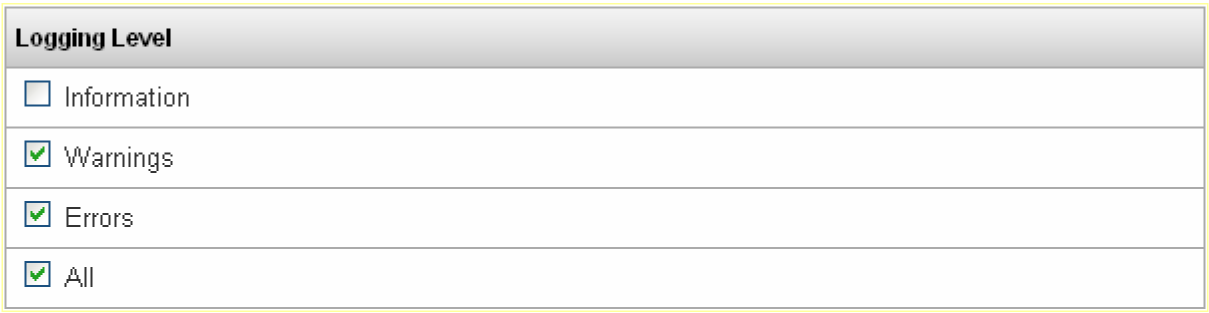


Figure 113 – Services Logging Level Section

Sections/Fields	Description
Information	If selected – the system will log Information level messages.
Warnings	If selected – the system will log Warning level messages.
Errors	If selected – the system will log Error level messages.
All	If selected – the system will log all the message types above and some other types such as Notice.

Table 74 – Services Logging Levels and Descriptions

Select Logging Level

STEPS:

1. From the **PBX Setup➔Services** page, locate the **Logging Level** section.
2. Select (placing a checkmark in the box) the method that you want the log files to be handled. You can elect to select all the types, if so desired.
3. Click the Save Changes button to save the changes.
4. Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Call Event Log Section

This allows you to enable web service and indicate the call event logs retention time.

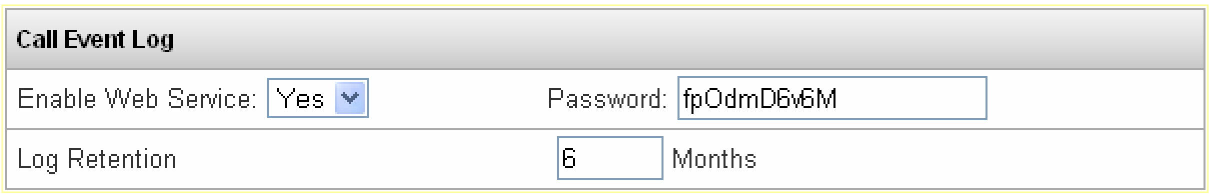


Figure 114 – Services Call Event Log Section

Sections/Fields	Description
Enable Web Service	If Enabled (set to **YES**), the system will allow call events to be polled via the web service.
Password	This is the password that will be used to access the call event log files.
Log Retention	This indicates the number of months that the log files will be retained on the server. Default is **6** months.

Table 75 – Services Call Event Logging and Descriptions

Set Call Event Logging

STEPS:

1 From the **PBX Setup→Services** page, locate the **Call Event Log** section.

2 From the **Enable Web Service** field, select **YES** to enable the system to start polling the event logs. Otherwise, select NO if you do not want the system to perform this function.

3 In the **Password** field, enter the password that will be used to access the polling service.

4 Enter the number of months you want to retain the call event logs in the **Log Retention** field.

5 Click the 

button to save the changes.

6 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Automated Phone Firmware Updates

This allows you to set the parameters for the system to perform automated firmware updates. Currently, when enabled, the PBX will check at midnight to see if there are more current phone firmware files available, and will download them if they exist.

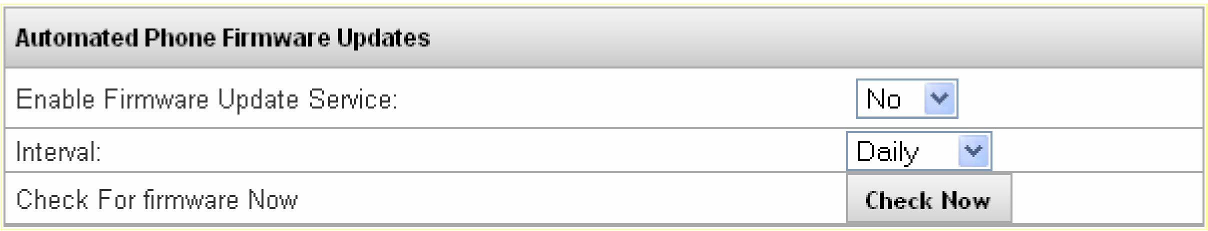


Figure 115 – Services Automated Phone Firmware Updates

The following table provides the parameters and their descriptions for the Automated Phone Firmware Updates feature.

Sections/Fields	Description
Enable Firmware Update Service	If enabled (set to **YES**), then the system will perform an automated check for firmware updates.
Interval	This allows you to indicate how often the system will check for new firmware. Available internals are Daily, Weekly or Monthly.
Check for Firmware Now	This allows you to check for firmware on demand.

Table 76 – Services Automated Firmware Parameters and Descriptions

Set Automated Firmware Update Parameters

STEPS:

1 From the **PBX Setup➔Services** page, locate the **Automated Phone Firmware Updates** section.

2 Set the parameters defining if the system should automatically check for firmware, and the interval is should check.

3 Click the 

button to save the changes.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Check for New Firmware Now

STEPS:

1 From the **PBX Setup➔Services** page, locate the **Automated Phone Firmware Updates** section.

2 Click on the **Check Now** button. The system will display the results once the process is complete. The following is a

```
Comparing checksum for IPitomy 550 file
IPitomy 550 Checksums match.  Nothing to do
Comparing checksum for IPitomy 120 file
IPitomy 120 Checksums match.  Nothing to do
```

sample of what the message that will be displayed. This message demonstrates that the firmware version in the PBX and at IPitomy match.

Scheduled Calling

This is a licensed feature that allows for periodic automated calls to be made by the PBX. This can be used for announcements, bells, alarms etc.

Figure 116 - Scheduled Calling

Enabling Scheduled Calling

STEPS: From the PBX Setup➔Services page, locate the Scheduled Calling section.

2 Change the dropdown to Enabled.

3 Click the Save Changes button to save the changes.

4 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database

Select	Enabled	Name	Start	End	Interval	Execute	Target	Destination	
☐	0	AName	Jul 19, 2010		Every 1 Weeks on M,W,Sa	9:30 am	**201	4444	
☐	0	Bell	Aug 6, 2010	Aug 31, 2010	Everyday	2:30 pm		1900	
☐	1	no_touch	Aug 17, 2010	Aug 24, 2011	Everyday	3:23 pm	4888	1900	

Edit Delete Add

Back To Services

Figure 117 - List of Scheduled Calls

Sections/Fields	Description
Select	Use this column to select which multiple entries to delete or edit.
Enabled	Displays if the entry is active. 1: Enabled 0: Disabled
Name	Displays the name of the Scheduled Call.
Start	Displays the date which a particular entry is set to start.
End	Display the date which a particular entry is set to end
Interval	Displays information on when the Scheduled Call will take place.
Execute	Shows the time the Scheduled Call will happen.
Target	Displays the **Number to Call** as defined in the Scheduled Call.
Destination	Displays the **Number to Connect** as defined in the Scheduled Call.
	Click to edit individual entries.
	Click to test an individual entry. A call between the Target and Destination will be established.
	Click to delete individual entries.
Edit	Click to edit multiple entries if checked in the Select column.
Delete	Click to delete multiple entries if checked in the Select column.
Add	Click to create a new Scheduled Call.

Table 77 - Scheduled Call List Descriptions

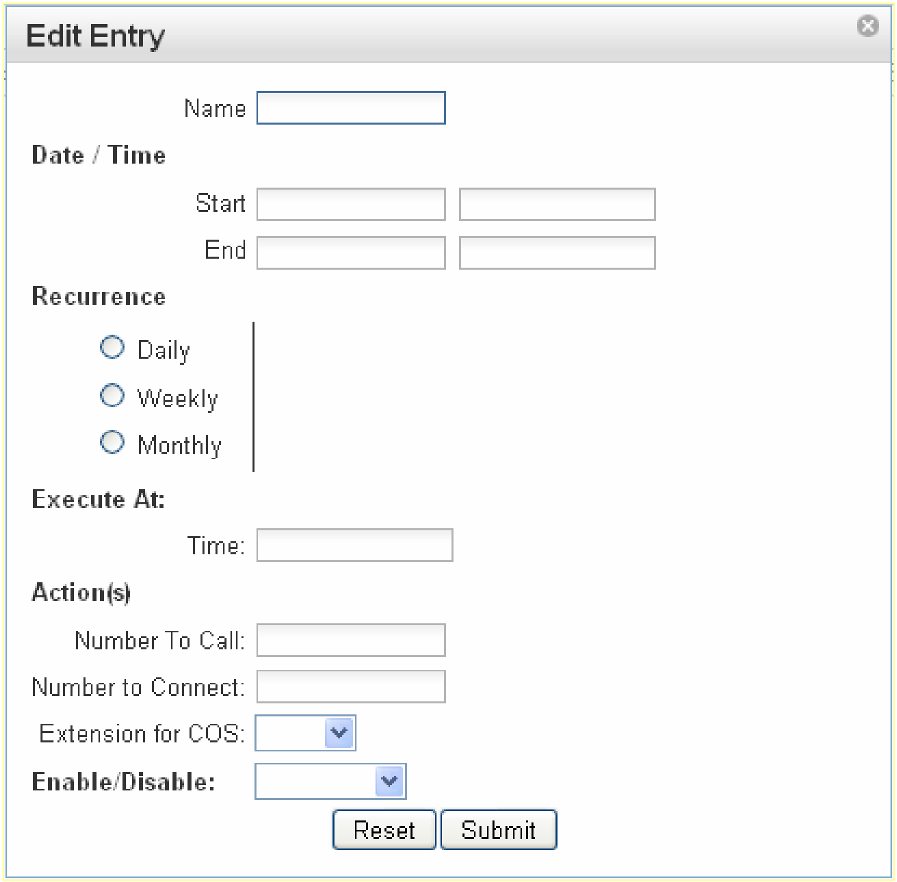


Figure 118 - Edit Scheduled Call

Sections/Fields	Description
Name	Enter a name to identify this Scheduled Call.
Date/Time	Define what date and time the Scheduled Call should start and end. You do not need to set an end time.
Recurrence	Defines the interval the Scheduled Call will take place. Calls can be set to Daily, Weekly, or Monthly. See the Recurrence table below for more details.
Execute At	Defines the time the call will take place for each Recurrence.
Number to Call	Defines the entity that will be called.
Number to Connect	Defines what destination the called party will be connected to.
Extension for COS	Set this to the extension number that is using the Class of Service you wish this Scheduled Call to use when making outbound calls.
Enable/Disable	Choose Enable for this Scheduled Call to be active, and Disable for it to be inactive.

Table 78 - Edit Scheduled Call Descriptions

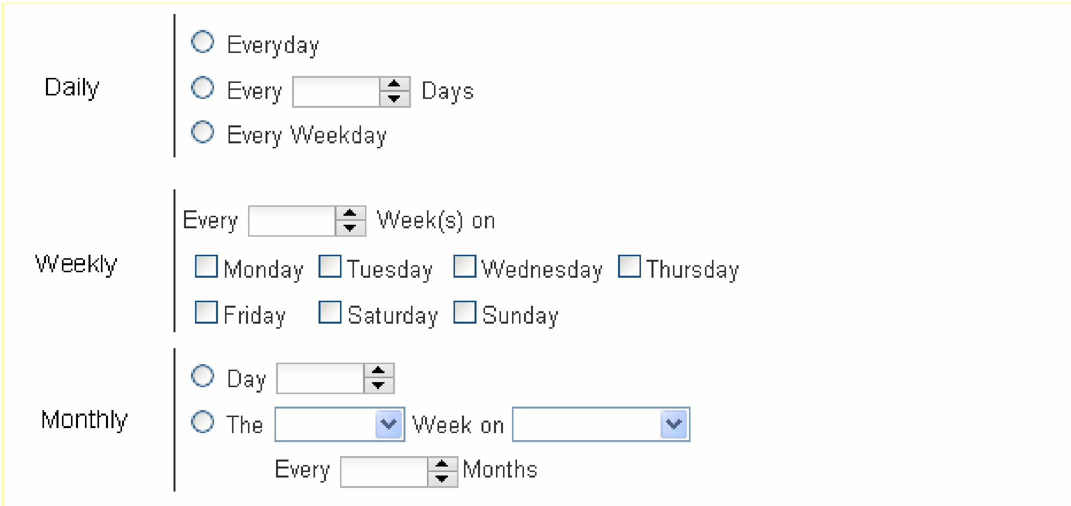


Figure 119 - Recurrence

Sections/Fields	Description
Daily	**Everyday:** Will call at the defined Time every day. **Every X Days:** Will call at the defined Time every (1-1550) days. **Every Weekday:** Will call at the defined Time every M-F.
Weekly	**Every X Weeks on Y:** Will call at the defined Time every (1-1550) weeks on the day defined.
Monthly	**Day:** Will call at the defined Time every (1st-31st) day of the month. **The X Week on Y:** Will call at the defined Time on the 1st, 2nd, 3rd, or 4th week of the month on the day defined. **Every X Months:** Will call at the defined Time every (1-31) months.

Table 79 - Recurrence Descriptions

Adding a New Scheduled Call

STEPS: From the **PBX Setup➔Services** page, locate the Scheduled Calling section.

2 If enabled, click the button for Manage Calls, which will take you to the **List of Scheduled Calls** page.

3 Click the **Add** button.

4 Configure your new Scheduled Call. If errors are made, you can click **Rese**t to start over.

5 Click **Submit** to save.

6 Click the blue link Back to Services to return to the **PBX Setup➔Services** page.

7 Click the Save Changes button to save the changes.

8 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database

Edit a Scheduled Call

STEPS: Navigate to the List of Scheduled Calls page.

2 Click the icon to edit an individual entry, or place a check next to multiple entries and click the **Edit** button.

3 Make the necessary changes.

4 Click **Submit** to save.

5 Click the blue link Back to Services to return to the **PBX Setup➔Services** page.

6 Click the Save Changes button to save the changes.

7 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database

Edit a Scheduled Call

STEPS: Navigate to the List of Scheduled Calls page

2 Click the icon to delete an individual entry, or place a check next to multiple entries and click the **Delete** button.

3 Click the blue link Back to Services to return to the **PBX Setup➔Services** page.

4 Click the Save Changes button to save the changes.

5 Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database

Test a Scheduled Call

STEPS: Navigate to the **List of Scheduled Calls** page.

2 Click the button to test the call connection immediately.

3 If the call does not work to your end users desires, reconfigure the Scheduled Call and test again.

REPORTING

Reports

The Call Data Records (CDR) contain information about calls collected from the IPitomy IP PBX for a specified period of time. The report details the number of calls, call duration, call origination, call destination, and status.

Call Data Records (CDR) can be sorted by specific time periods. CDR may also be exported in the same date range format. The records have also been enhanced for easier reading, and comprehension of data. To sort simply select the date range you wish to be displayed, and then press the view button. The display will be arranged and sorted according to your date selections.

CDR-Report Queue Graphs

CDR Report

From 9/20/2010 To 9/21/2010 View Export

Date/Time	Direction	Source	Destination	Trunk	Duration	Status
Sep 21 03:45 pm	Incoming	5551234567	Bill Wood	Zap/2-1	0:00:18	ConnectEvt
Sep 21 03:39 pm	Incoming	5551234567	Elaine Blodgett	Zap/2-1	0:03:59	AGENT COMPLETED
Sep 21 03:39 pm	Incoming	5551234567	Bill Wood	Zap/3-1	0:05:56	CALLER COMPLETED
Sep 21 03:39 pm	Incoming	5551234567	John Wolfe	Zap/1-1	0:00:03	-
Sep 21 03:33 pm	Outgoing	John Wolfe	5551234567	Vintalk	0:03:45	ANSWER
Sep 21 03:36 pm	Internal	807	Paul Falanga	-	0:00:18	NOANSWER
Sep 21 03:33 pm	Internal	Tonie Office	EJ Donovan	-	0:02:04	ANSWER
Sep 21 03:29 pm	Incoming	5551234567	Northeast Region Sales	Zap/1-1	0:00:58	TIMEOUT EXIT
Sep 21 03:20 pm	Internal	Nick Branica	Tonie Office	-	0:07:38	ANSWER
Sep 21 03:23 pm	Internal	5551234567	TestNick	-	0:00:18	HANGUP
Sep 21 03:22 pm	Incoming	5551234567	John Wolfe	Zap/2-1	0:05:49	CALLER COMPLETED
Sep 21 03:22 pm	Incoming	5551234567	Elaine Blodgett	Zap/3-1	0:14:03	AGENT COMPLETED
Sep 21 03:20 pm	Incoming	5551234567	Main Menu	Zap/2-1	0:00:08	HANGUP
Sep 21 03:05 pm	Outgoing	IP650 Two	5551234567	Vintalk	0:14:16	ANSWER
Sep 21 03:09 pm	Incoming	5551234567	Aastra Test Mobile	Zap/2-1	0:09:00	ANSWER
Sep 21 03:17 pm	Incoming	5551234567	Support	Zap/5-1	0:01:02	ABANDONED
Sep 21 03:11 pm	Incoming	5551234567	John Wolfe	Zap/4-1	0:07:54	CALLER COMPLETED
Sep 21 03:08 pm	Outgoing	Elaine Blodgett	5551234567	Vintalk	0:02:20	ANSWER
Sep 21 03:08 pm	Internal	Tonie Office	Aastra Test Mobile	-	0:00:38	ANSWER
Sep 21 03:04 pm	Outgoing	BlackBox Test	5551234567	Vintalk	0:00:01	CANCEL

First < **1** 2 3 4 5 6 7 8 9 10 11 > Last

Figure 120 – CDR Reports Page

Sections/Fields	Description
Date/Time	The date and time of the call.
Direction	Defines where the call originated: Internal: The call took place entirely within the PBX Outgoing: An internal entity made a call out a trunk. Incoming: An external party made a call inbound over a trunk.
Source	The originating party for the call.
Destination	The end location for the call.
Trunk	Defines which trunk was used for the call.
Duration	The length of the call in hours:minutes:seconds.
Status	The status describes the final state of a call through the PBX: Adandoned: Call entered a Queue, but the caller hung up before the call was answered or met the failover requirement. ChanUnavail: No channel was able to be created for the call due to a possible error state for the intended destination. Agent Completed: Connected Queue call was terminated at the extension. Hangup: Call to a Menu, but the caller hung up before taking any actions. Caller Completed: Connected Queue call was terminated at the calling parties end. Cancel: Call from an Extension, but the caller hangs up before the call is connected. Answer: A connected call. No Answer: A call that met the ring time of an extension without being answered. Timeout Exit: Call to a Queue that meets the Timeout. VM Not Left: Call ended with the caller being sent to voicemail where they did not leave a message. VM Left: Call ended with the caller being sent to voicemail where they left a message. Transferred: Connected call successfully transferred. Busy: Call out a trunk returned busy status, or extension was busy. - : Defines the call is still active, and the status will update when completed. Click on the call record for details. Exited Empty Queue: Call to Queue that follows Exit Empty rule. Exited With Key: Call to Queue that follows the Exit Menu key press.

Table 80 - CDR Report Descriptions

View CDR Report – Smart Personal Console

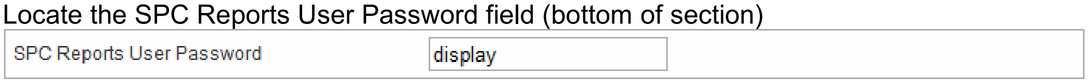
IPitomy PBX Software version 3.4.1 and above have the added ability to view CDR Reports and Queue data from Smart Personal Console (SPC) using a special user name and password.

STEPS to access CDR and Queue Graphs via Smart Console Console:

1 Within PBX Administration, navigate to the **PBX Setup➔General** page

2 Locate to the **General Settings** section (second section from the top).

3 Locate the SPC Reports User Password field (bottom of section)

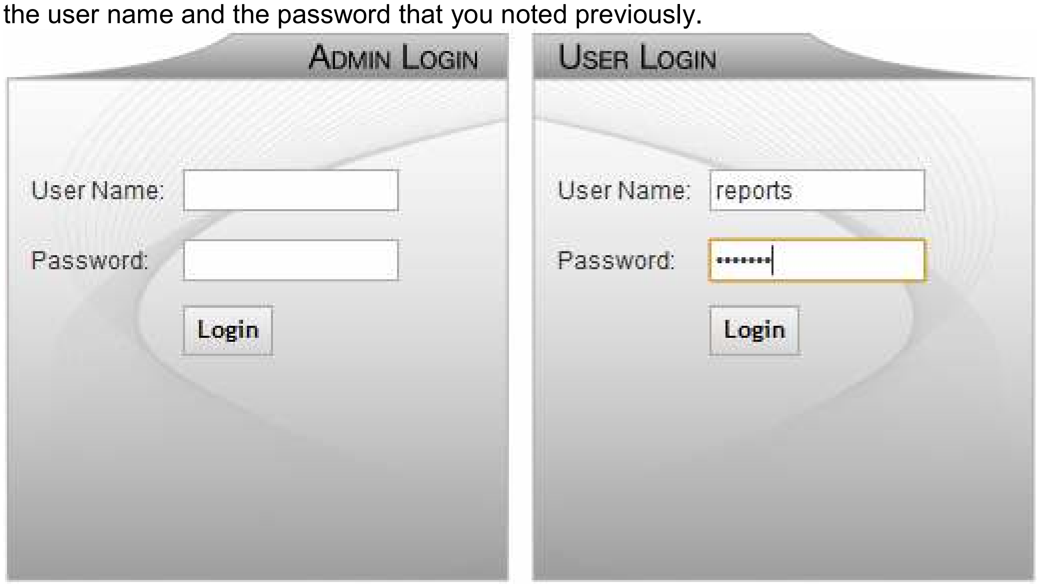


4 It may be helpful to change this to a user-friendly password. If you change the password be sure to Save Changes and Apply Changes.

5 Make a note of this password.

6 On the PC that will be used to access CDR Reports and Queue Graphs via SPC, open a web browser and navigate to the IPitomy PBX login screen

7 Using the right side – Smart Personal Console (User Login) input the word “reports” as the user name and the password that you noted previously.



8 Click on Login.

9 The CDR-Report page is opened, use the instructions below to extract the data you wish. Or,

10 Click on Queue Graphs to open that interface and use the instructions below to extract that data.

View CDR Report – PBX Administration

STEPS:

1 From the **Reporting→Reports** page, click on the **CDR Report** button. The **CDR Report** page appears.

2 Select the **From** and **To** date range by click on the box next to the From and To fields. Use the calendar control page that appears to select the desired dates.

3 Click the **View** button. A report with call information from the date range parameter that was entered is displayed.

4 Click on an individual call record and a new window will appear displaying more detailed information for that call.

Export CDR Report

The Export function will provide you with a tabulated report to download to a PC.

STEPS:

1 From the **Reporting→Reports** page, click on the **CDR Report** button. The **CDR Report** page appears.

2 Enter the date range and retrieve the report that you want to export (see steps for Viewing Report).

3 Click the **Export** button then specify the location where you want to download the report.

Queue Graphs

Queue Graphs display additional call information for ACD Queues. A PBX must be licensed for ACD for this button to appear.

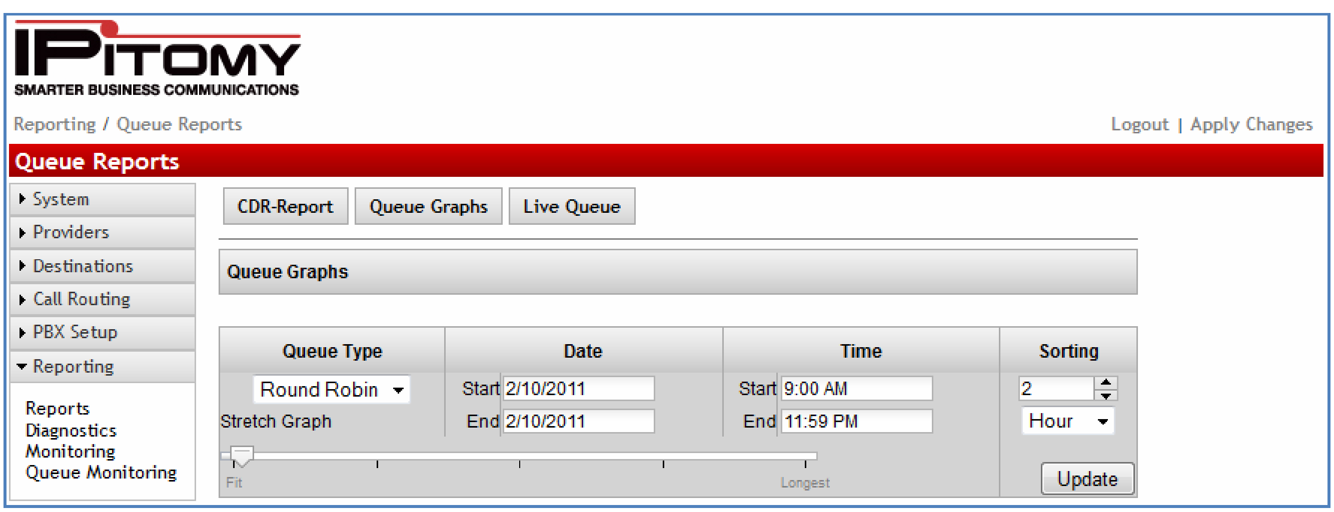


Figure 121 – Queue Graphs Page

Sections/Fields	Description
Queue Type	This is the listing of the queue you want to view. The drop-down list is populated with the various groups that have been configured.
Date	This allows you to specify a range of dates for which call records will display.
Time	This allows you to specify a start and end time within the date range for the reporting to display.
Setting	This is the increment with which to sort by. The default increment is 2 hour.
Stretch Graph	Move the slider to adjust the size of the graph to meet your viewing requirements.

Table 81 – Queue Graphs Search Parameters

Run Queue Graph Report

STEPS:

1 From the **Reporting➔Reports** page, click on the **Queue Graphs** button. The **Queue Graphs** page appears.

2 Select the **Queue Type** from the drop-down list.

3 Select the **Start** and **End** date using the calendar control that appears when you click on the date fields. Select the desired **Start** and **End** time for the reporting period you want to search.

4 Change the **Sorting** parameters to the desired setting.

5 Click the **Update** button. The system will perform the search based on the criteria that is specified and display the results.

Overview Tab

The Overview Tab offers an overall summary of the calls for the period in question for that queue. The overview will display textual information for answered and unanswered calls, as well as a graph summarizing the information.

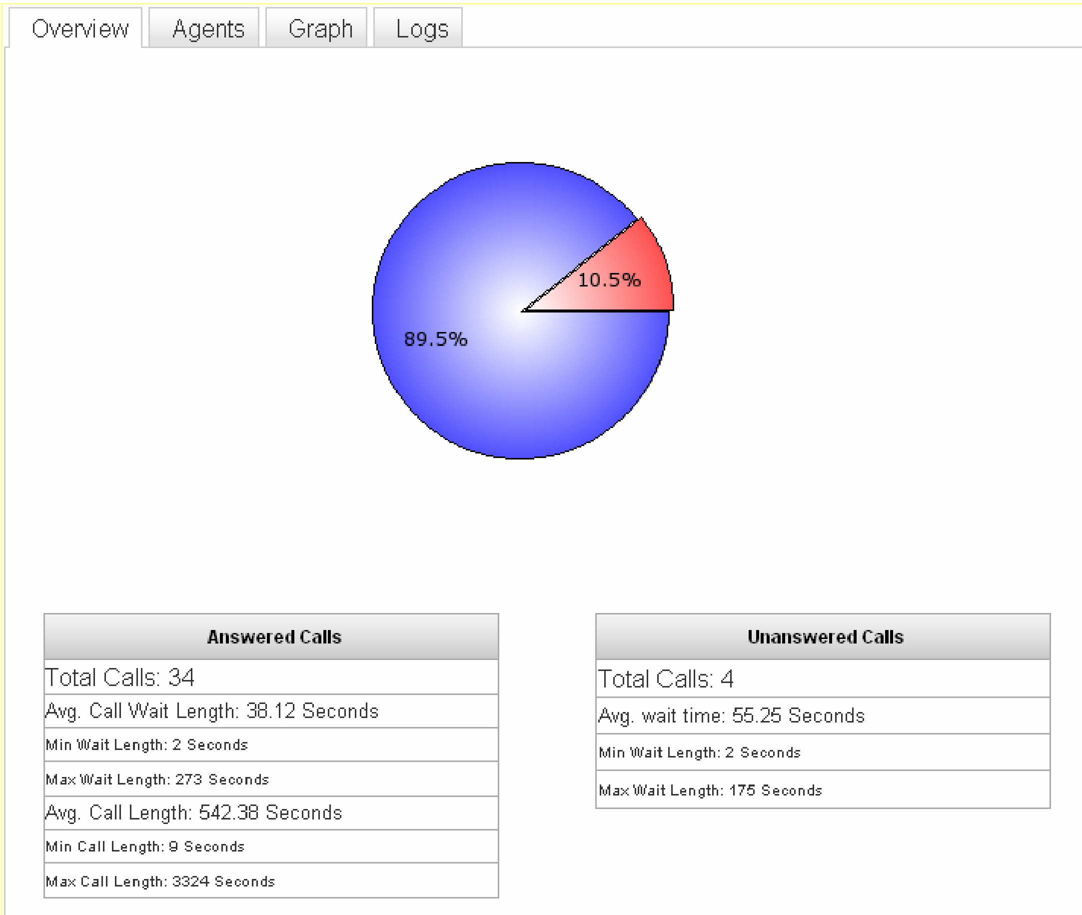
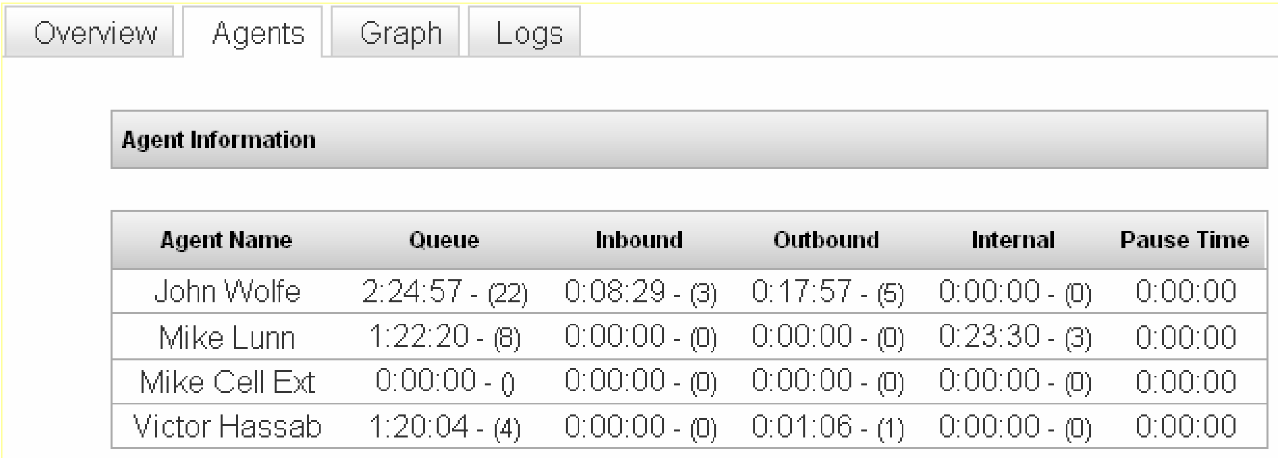


Figure 122 – Reports Queue Graph Overview Tab

Sections/Fields	Description
Total Calls	Total number of calls answered, or unanswered.
Avg. Call Wait Length	An average of the wait time for calls answered, or unanswered.
Min. Wait Length	Shortest wait time for calls answered, or unanswered.
Max Wait Length	Longest wait time for calls answered, or unanswered.
Avg. Call Length	Average duration of an answered call.
Min. Call Length	Shortest duration of an answered call.

Max Call Length	Longest duration of an answered call.

Agents Tab



Agent Name	Queue	Inbound	Outbound	Internal	Pause Time
John Wolfe	2:24:57 - (22)	0:08:29 - (3)	0:17:57 - (5)	0:00:00 - (0)	0:00:00
Mike Lunn	1:22:20 - (8)	0:00:00 - (0)	0:00:00 - (0)	0:23:30 - (3)	0:00:00
Mike Cell Ext	0:00:00 - ()	0:00:00 - (0)	0:00:00 - (0)	0:00:00 - (0)	0:00:00
Victor Hassab	1:20:04 - (4)	0:00:00 - (0)	0:01:06 - (1)	0:00:00 - (0)	0:00:00

Figure 123 – Reports Queue Graphs Agents Tab

The Agents tab displays details for queue members.

Sections/Fields	Description
Queue	Time spent on queue calls. Number of calls displayed in parenthesis.
Inbound	Time spent on inbound, non-queue calls. Number of calls displayed in parenthesis.
Outbound	Time spent on outbound, non-queue calls. Number of calls displayed in parenthesis.
Internal	Time spent on internal, non-queue calls. Number of calls displayed in parenthesis.
Pause Time	Time spent paused out of the queue. Number of calls displayed in parenthesis.

Graph Tab

The Graph Tab provides a graphical view of queue calls only. Inbound and outbound calls are tabulated but not graphed. The graph tab divided into time segments defined by the Sorting setting set previously. The information presented in this tab will allow you to determine busiest times and when call coverage is most needed.

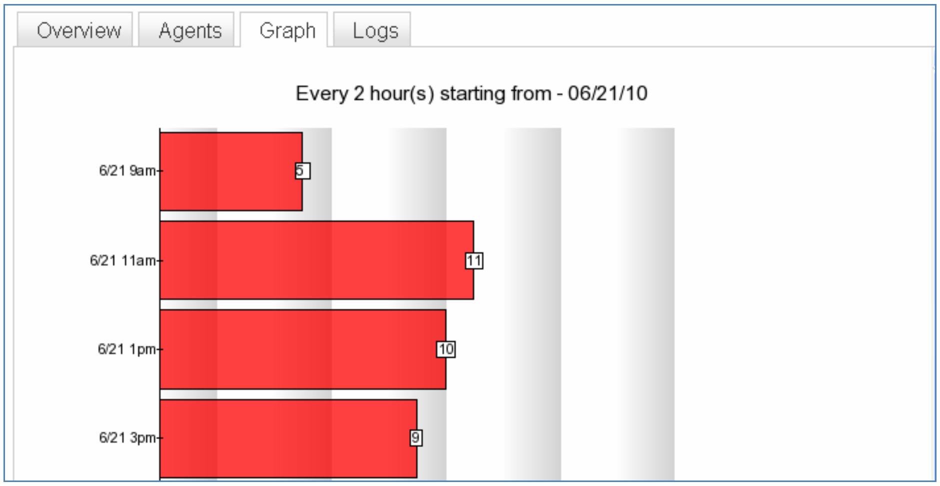


Figure 124 – Reports Queue Graphs Tab

Logs Tab

The Logs Tab will display the complete call log for the queue from your search setting. It will display the time of the call, the direction of the call either inbound or outbound, the caller ID that was delivered with the call, the location that the call was delivered to, the wait time for the caller, the total duration of the call, and finally the outcome of the call or the final status of the call. See the table for CDR Report Descriptions for detailed information about the fields.

Overview | Agents | Graph | Logs

ACD Details

Time	Direction	From	To	Wait Length	Duration	Outcome
Jun 21 09:47 am	Incoming	unknown	John Wolfe	0:00:04	0:01:28	AGENT COMPLETED
Jun 21 10:14 am	Incoming	6135919000	John Wolfe	0:00:20	0:13:52	AGENT COMPLETED
Jun 21 10:45 am	Internal	2115	John Wolfe	0:00:12	0:04:55	AGENT COMPLETED
Jun 21 11:01 am	Outgoing	John Wolfe	9557101	0:00:00	0:03:17	ANSWER
Jun 21 11:09 am	Incoming	8454940148	John Wolfe	0:00:16	0:09:48	AGENT COMPLETED
Jun 21 11:10 am	Incoming	9095933972	Victor Hassab	0:00:05	0:08:13	AGENT COMPLETED
Jun 21 11:27 am	Incoming	4409421900	Victor Hassab	0:00:07	0:04:41	AGENT COMPLETED
Jun 21 11:33 am	Incoming	5613338008	Mike Roaming*	0:00:12	0:10:27	CALLER COMPLETED
Jun 21 11:45 am	Incoming	7187057451	Victor Hassab	0:00:08	0:30:02	AGENT COMPLETED
Jun 21 11:48 am	Outgoing	John Wolfe	3718980	0:00:00	0:00:51	ANSWER

Figure 125 – Reports Queue Graphs Logs Tab

Diagnostics

This section, when configured under PBX Setup➔Services, displays a message log of actions in the PBX. Typically this page will be referenced by Tech Support to pinpoint where troubleshooting for a specific problem should begin.

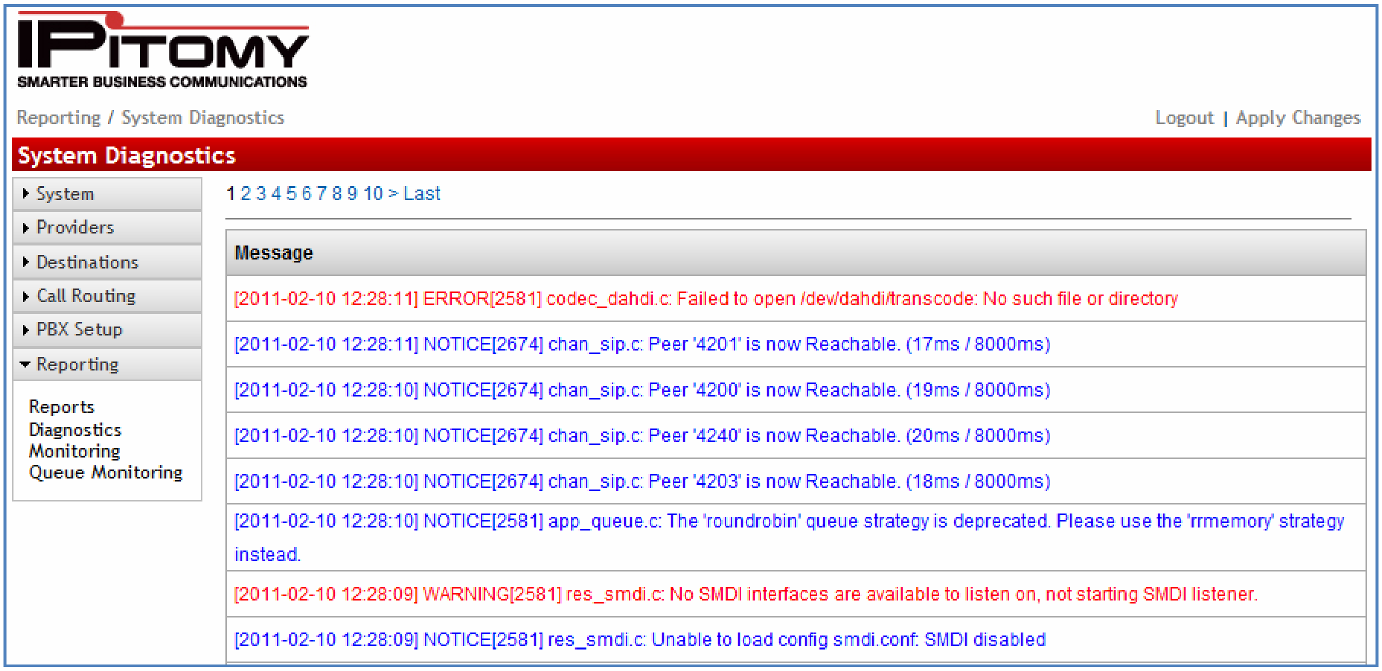


Figure 126 – System Diagnostics Page

View System Diagnostics

STEPS:

1 Click the **Reporting➔Diagnostics** link on the Admin Page.

2 The **System Diagnostics** page appears displaying the messages that have been processed and their status.

3 The type of message that is displayed on this page is set in the **Logging Level** section of the **Systems** page.

Monitoring

This page gives a snapshot of the status of your SIP network. Extensions, SIP Providers, Branch Offices, and active calls will display. Entries in Green are connected, entries in Red are unable to be reached, and entires in Grey are not present. At the bottom of the page, active calls will display on a channel by channel basis. The page will refresh every 10 seconds.

SIP:

Account	Address	Port	Status
111	(Unspecified)	0	UNKNOWN
113	(Unspecified)	0	UNKNOWN
186	(Unspecified)	0	UNKNOWN
201	(Unspecified)	0	UNKNOWN
204	(Unspecified)	0	UNKNOWN
230	(Unspecified)	0	UNKNOWN
2207	97.96.48.119	5060	OK (119 ms)
4211	(Unspecified)	0	UNKNOWN
6200	174.22.163.81	5060	OK (136 ms)
6270	(Unspecified)	0	UNKNOWN
6280	(Unspecified)	0	UNKNOWN
6290	(Unspecified)	0	UNKNOWN
SIP Trunk/siplines	216.146.210.100	5060	UNREACHABLE

Branches:

Name/Username	Host		Mask	Port		Status
New_York/New_Yo	(Unspecified)	(D)	255.255.255.255	0	(T)	UNKNOWN
Houston/Houston	105.26.45.11	(S)	255.255.255.255	4569	(T)	UNREACHABLE
Seattle/Seattle	11.123.15.2	(S)	255.255.255.255	4569		UNREACHABLE

3 iax2 peers [0 online, 3 offline, 0 unmonitored]

Channels:

Channel	Location	State	Application(Data)

0 active channels

0 active calls

Figure 127 – Monitoring Page

APPENDICES

Appendix 1: Key Types and Codes

This table describes the function of each key type, and how to manually configure them. It is not advised to manually configure unless the phone is in a situation where you cannot easily use the PBX to set the keys (ie: remote phones). Unless specified, the Line field when manually configuring should be set to All.

Key Type	Description
BLF	Can be set to an extension or park orbit. Pressing the key will call the extension or retrieve a call from the park orbit. When an extension is ringing the light will flash, and when an extension is connected to a call, the light will be solid. Park BLF keys will lamp when a call is parked in that particular orbit. To manually configure, log into the phone and set the key to: Type: BLF Label: Meaningful Descriptor Value: The extension or park orbit number to monitor Line: 1
Speed Dial	Dials the digits entered as for the Value and sends them to the PBX. To manually configure, log into the phone and set the key to: Type: Speed dial Label: Meaningful Descriptor Value: Any number or feature code that the PBX can recognize
Park	Takes the connected call and parks it in an available orbit from 701-720, starting at the lowest available slot. When parking, the PBX will tell you audibly which orbit the call was sent to. To manually configure, log into the phone and set the key to: Type: Park Label: Park Value: ##700#
DND	Places the phone itself in a state where it will not receive calls. This does not communicate your state to the PBX (see Pause key). To configure manually, log into the phone and set the key to: Type: Do Not Disturb
Blind Transfer	This key starts the transfer process and allows you to enter a transfer destination. Once entered, press Dial or wait for the digit timeout and the call will transfer. This key is most useful if a phone has no open lines to perform a transfer using the fixed key on the phone, as it does not open a new line to transfer. To configure manually, log into the phone and set the key to: Type: Prefix Label: Blind Transfer Value: ##

Call Pickup	Allows the user to answer calls ringing direct to extensions that are in the Call Group that matches the phones Pickup Group. To manually configure, log into the phone and set the key to: Type: Pickup Label: Meaningful Descriptor Value: 99
Voicemail	Allows the user to check their voicemail messages. User will be prompted for their password(PIN). To manually configure, log into the phone and set the key to: Type: Speed dial Label: Voicemail Value: 923
Voicemail Gateway	Allows the user to check any voicemail box for messages. User will be prompted for mailbox number and password(PIN). To manually configure, log into the phone and set the key to: Type: Speed dial Label: VM Gateway Value: 924
Record	Toggles on and off the record feature on the current active call. The file will be saved in the voicemail box for the extension, under the Work folder. To manually configure, log into the phone and set the key to: Type: Speed dial Label: Record Value: *#
Fwd On	With a single press, a user can enable their unconditional forwarding. If no destination is set, you will need to define one before forwarding can be enabled. To manually configure, log into the phone and set the key to: Type: Speed dial Label: Fwd On Value: *91
Fwd Off	With a single press, a user can disable their unconditional forwarding. To manually configure, log into the phone and set the key to: Type: Speed dial Label: Fwd Off Value: *90
Set Fwd	With a singe press, a used can define their forwarding destination. To manually configure, log into the phone and set the key to: Type: Speed dial Label: Set Fwd Value: *92

Fwd Gateway	This takes the user to a gateway that allows them to control forwarding for extensions other than their own. User will be prompted for an extension number and password (PIN) before they can make changes. Once logged in, you can enable, disable, and set the forward destination. To manually configure, log into the phone and set the key to: Type: Speed dial Label: Fwd Gateway Value: *9
Intercom	This key starts the intercom process and allows you to enter an extension or group number. Once entered, press Dial or wait for the digit timeout, and the page will happen. To manually configure, log into the phone and set the key to: Type: Intercom Label: Intercom
Pause	Similar to the DND key, except pressing this key communicates to the PBX, and will pause you out of call queues. The key is a BLF and will lamp when the phone is paused. To manually configure, log into the phone and set the key to: Type: BLF Label: Pause Value: ABC<extnum> Line: 1
Day/Night BLF	When pressed, a call is established to the PBX telling the user the state of Day/Night mode override, and gives instructions on how to change the settings. If Day/Night mode is forced into night mode, the light will blink; and if forced into day mode, the light will be solid. If Day/Night mode is following the schedule the BLF will not lamp. To manually configure, log into the phone and set the key to: Type: BLF Label: Day/Night Value: ff Line: 1
Page BLF	Allows BLF monitoring of an extension, as well as one press paging to that extension. TO manually configure, log into the phone and set the key to: Type: BLF Label: Meaningful Descriptor Value: **<extnum> Line: 1

Appendix 2: IP Telephones

IPitomy 550

The IPitomy **IP550** SIP based telephone is designed for busy business environments. The **IP550** is the workhorse phone for any business with a wide range of telecommunications requirements. Equally, at home on the desk of a corporate executive or on a conference table, the **IP550** fits just about any business situation.

The **IP550** is an advanced open standards business telephone. With features that are based on the technology of the future, the **IP550** also includes most of the familiar functions available on traditional business telephone systems. The combination of high technology and practical functionality results in a standards-based telephone that is simple to use and won't require a long learning curve. The 20 soft keys, 6 programmable keys and 16 fixed feature keys, provide all of the flexibility required for any business.

Because the **IP550** is designed as a SIP based telephone, its robust capabilities perform well in or out of the office. Remote users can enjoy the same great features as though they were right in the office. The large graphical display conveniently displays caller ID information. Missed calls, dialed calls and received calls are stored and can be viewed, dialed or stored in the on board phone book. The full duplex speakerphone provides world class speaker phone quality. The attractive appearance and sturdy stand provide a professional look and feel. The **IP550** enhances the performance of the IPitomy IP PBX systems and is far more integrated that other open standards SIP based endpoints.

IPitomy 55i

The **55i** from IPitomy offers powerful features and flexibility in a standards based carrier-grade advanced level expandable IP telephone. With a sleek and elegant design, 144 x 75 pixel backlit LCD display and 6 dynamic context-sensitive softkeys, the **55i** is fully interoperable with leading IP Telephony platforms, offering advanced XML capability to access custom applications and support for up to 9 calls simultaneously. Part of the IPitomy family of IP telephones, the **55i** is ideally suited for moderate to heavy telephone users who require more one touch feature keys and XML based programs.

IPitomy 53i

The **53i** from IPitomy offers powerful features and flexibility in a standards based carrier-grade basic level IP telephone. With a sleek and elegant design and 3 line LCD displays, the **53i** is fully interoperable with leading IP Telephony platforms, offering advanced XML capability to access custom applications and support for up to 9 calls simultaneously. Part of the IPitomy family of IP telephones, the **53i** is ideally suited for light to regular telephone requirements.

IPitomy 53i IP Phone

The **53i IP phone** offers powerful features and flexibility in a standards based, carrier-grade basic level IP telephone. The 53i has a sleek and elegant design and 3 line LCD display, the 53i is fully interoperable with leading IP Telephony platforms, offering advanced XML capability to access custom applications and support for up to 9 calls simultaneously.

IPitomy 57i

The 57i CT from IPitomy offers powerful features and flexibility in a standards based, carrier-grade advanced level, expandable, IP telephone that includes an integrated WDCT cordless mobility handset for coverage up to 300,000 sq ft.* With a sleek and elegant design, large 144 x 128 pixel graphical backlit LCD display and 6 dynamic context-sensitive softkeys, the 57i CT base unit is fully interoperable with leading IP Telephony platforms, offering advanced XML capability to access custom applications and support for up to 9 calls simultaneously. Part of the IPitomy family of IP telephones, the 57i CT is ideally suited for executives, heavy telephone users who require more one touch feature keys and a large screen to take full advantage of XML based programs and mobile warehouse, support and retail staff.

6739i

The new 6739i, a new member of 67xi SIP portfolio, brings leading edge communications technology to the desktop in a stylish design. This phone features a giant touch screen display, Bluetooth, 1 GB Ethernet and enhanced audio components which make G.711 sound almost as good as Hi-Q our G.722 codec. This product is available in December 2009.

IP120

The IP120 is a value priced SIP based IP Telephone/Speakerphone. It has two Ethernet ports, supports Power over Ethernet and has a two line display. The IP120 supports two calls and has 5 programmable buttons. The IP120 is designed for applications where a low cost instrument is required such as kitchen, breakroom, warehouse, exam room or anywhere that does not require a large screen or more lines. The IP120 uses IPitomy's advanced Auto Provisioning features for easy deployment.

IP51i

The 55i from IPitomy offers powerful features and flexibility in a standards based, carrier-grade advanced level expandable IP telephone. With a sleek and elegant design, 144 x 75 pixel backlit LCD display and 6 dynamic context-sensitive softkeys, the 55i is fully interoperable with leading IP Telephony platforms, offering advanced XML capability to access custom applications and support for up to 9 calls simultaneously. Part of the IPitomy family of IP telephones, the 55i is ideally suited for moderate to heavy telephone users who require more one touch feature keys and XML based programs.

IPitomy 536M

The 536M expansion module is designed to increase the power and flexibility of the 5i Series SIP telephones. Up to three modules can be used with the 55i, 57i or 57i CT telephone to create a powerful, feature rich console option. The 536M shares power and signaling with the phone, eliminating the need for additional wiring. Designed for receptionists, administrative assistants call center agents, power users, and executives who need to monitor and manage a large volume of calls on a regular basis, the 5i Series Expansion Modules provide an intelligent choice for all Enterprise IP environments.

IPitomy 560M

The 560M expansion module is designed to increase the power and flexibility of the 5i Series SIP telephones. Up to three modules can be used with the 57i or 57i CT telephone to create a powerful, feature rich console option. The 560M shares power and signaling with the phone, eliminating the need for additional wiring. Designed for receptionists, administrative assistants call center agents, power users, and executives who need to monitor and manage a large volume of calls on a regular basis, the 5i Series Expansion Modules provide an intelligent choice for all Enterprise IP environments.

Detailed information regarding these phones including technical specifications can be found at www.aastra.com.

APPENDIX 3: SOFTPHONES

CounterPath™ eyeBeam® 1.5 and X-Lite® 3.0

What is a Softphone?

CounterPath's eyeBeam® 1.5 and X-Lite® 3.0 are Web-based telephones that operate from a PC. These next-generation Voice Over IP (VoIP) telephony client's are designed to enhance a user's communication experience by keeping them connected to callers anyplace and anytime through the convenience of an intuitive and user-friendly desktop.

Based on open standards, CounterPath™ Softphones use a telephone-centric interface that allows users to manage voice, video, instant messaging (IM) and presence applications on their desktop. This comprehensive suite of carrier-grade solutions, give users the flexibility to meet the fast-paced and changing demands of any business.

X-Lite® 3.0 Free Softphone

- **Intuitive user interface** makes it easy for both novice and power users to make and receive calls, initiate video conferencing, and communicate using Instant Messaging.
- **Comprehensive Personal Address Book**, including detailed calls lists and history.
- **Microsoft Outlook® integration** allowing users to import their address book into their eyeBeam® contact list.
- Zero-Touch Configuration of audio or video devices.
- Instant messaging (IM) and presence management.
- Multi-party and ad-hoc voice and video conferencing (IP and PSTN).
- Voice and video call recording.
- **Pop-up management** of incoming calls.

eyeBeam® 1.5 (Pricing available at www.counterpath.com)

- **Intuitive user interface** that makes it easy for both novice and power users to make and receive calls, initiate video conferencing, and communicate using Instant Messaging.
- **Comprehensive Personal Address Book**, including detailed calls lists and history.
- **Microsoft® Outlook® integration** allowing users to import an address book into the eyeBeam® contact list and dial directly from the application.
- Zero-Touch Configuration of audio or video devices.
- Instant messaging (IM) and presence management.
- Multi-party and ad-hoc voice and video conferencing (IP and PSTN).
- Voice and video call recording.
- **Pop-up management** of incoming calls.
- Security offering signaling and media encryption via TLS and SRTP streams.
- **Performance management** of the SIP end-point (Softphone).
- High compression CODEC support.

Softphone Installation

STEPS:

1 Using the Internet Explorer, go to the **CounterPath** website at the www.counterpath.com.

2 **Download the CounterPath™ Softphone** to be used with the system. The installation utility will install a phone icon in the toolbar of the operating system. It looks like a green light.

3 **Left click the Softphone** Icon in the operating system toolbar. The Softphone will appear.

4 **Right click on the Softphone** and select **SIP Account Settings** from the drop-down menu.

5 Click **Properties**. The properties window for the Softphone will appear.

6 Enter a **Display Name**. This is the name of the person or department associated with the phone.

7 Create a **User Name**. This is the extension the phone will be off of the IPitomy IP PBX. Be sure to use a number that is not being used by an existing extension.

8 Enter a **Password**. This password will need to be the same as the one used in the IPitomy IP PBX Extension Setup (Add New) page.

9 Enter the **Extension Number** in the **Authorization User Name** field.

10 Enter the **Domain (IP Address)** of the system to which the Softphone is to be connected.

11 Click **Apply** then select OK. Softphone Account Settings page will close.

12 Log into the IPitomy IP PBX (if not already logged in). Click **Destinations** and **Extensions** in the navigation bar of the system's administration menu. The Extensions page will appear.

13 Click **Add New**. The Edit Extensions page will appear. Note that each new extension added automatically has a voice mailbox created.

14 Insert the **Name** or department associated with the extension being created.

15 Create an **Extension Number** for this person or department.

16 Populate the **Email address** for the person or extension. This will allow the system to forward email messages to the address of the person at the extension.

17 **Select a status** from the drop-down menu. An extension can be:

- **Active** – Currently in use.
- **Disabled** – Not currently in use.

18 **Create a voicemail PIN** for the extension. PIN numbers must be between 3 and 4 characters long. The default setting is for the PIN to be the extension number. Be sure to instruct users to change the PIN to avoid unauthorized use.

19 **Enter a Ring Time**. This is the time in seconds that a call will ring before it is considered unanswered. Ring time must be between **1** and **360** seconds in length.

20 **Define a Call Limit**. This is the number of concurrent calls allowable at an extension. The Call Limit selected must be between **0** and **9**. This limit will depend on the phone being installed.

21 **Create a Call Group number**. This number assigns this extension to a group with a similar purpose (e.g., Sales or Customer Service). Multiple call groups can be assigned to each extension by putting a comma between the group numbers. The call groups also define which Pickup Groups can answer calls to this extension.

22 **Create a Pickup Group**. This number must match the Call Group number(s). It defines the Call Group Numbers this extension can pickup remotely by pressing ***8**.

23 Click **Apply Schedule**. When an extension is created, a schedule destination is created. This schedule is not activated until the Apply Schedule box is selected. When it is selected, you can setup a schedule for this extension by selecting Schedule under the Destinations Menu and clicking on the schedule for that extension. Extension schedules will appear with the name of the extension (e.g., Extension 123 would appear as “ext_123”). See the Schedules section of this guide for more information.

Forward Settings

The forwarding settings are made to be very user friendly. The settings may be modified from the Smart Personal Console, changed from your telephone extension or changed remotely from any telephone (including cell phones) using the touch-tone key pad of any telephone.

Forward settings routes calls to a different destination. These settings can be:

- **Unconditional** – Always route calls to a specific destination.
- **Busy** – Route calls to a specific destination when the extension is in use or do not disturb is selected.
- **No Answer** – Route calls to a specific destination when a call is not answered.
- **Unavailable** – Route calls to a specific destination when a phone is turned off, is not registered with the system or has reached its call limit (as set In the IP PBX).

Provisioning Forward Settings

STEPS:

1 **Pick the setting to be provisioned** – Unconditional, Busy, No Answer or Unavailable.

2 Select **Enabled** or **Disabled**. Disabled turns the forward setting off. Enabled turns the forward setting on.

3 If the Forward setting is Enabled, you can choose to select a destination from the drop-down list. The IPitomy IP PBX allows calls to be forwarded to a PSTN. Forward calls to a PSTN number by entering it into the field provided. Calls can be forwarded to any destination (or telephone number) in the drop-down list or any telephone number.

Changing a Forwarding Number from an Extension

Only unconditional forwarding can be changed from a touch-tone keypad.

- Dial ***90** to disable forwarding.
- Dial ***91** to enable forwarding.
- Dial ***92** to set the forwarding number.

Changing a Forwarding Number from a PC

STEPS:

1. Browse to the **Smart Personal Console** page.
2. **Login**.
3. Select a **Destination** for the chosen forward type.
4. Enter the **telephone number**.

Changing a Forwarding Number While Away from an Extension

Only unconditional forwarding can be changed from a touch-tone keypad.

When it is necessary to modify the forwarding setting while away from the office, the IPitomy IP PBX has a forwarding application built into the system. It is necessary to have an automated attendant menu accessible from outside the system. The forwarding gateway is selectable as an option from the Smart Personal Console. When away from the office, it is possible to call into the Automated Attendant, enter the digit setup to be the forwarding gateway. Here users can turn forwarding on or off and enter a different number to forward calls to.

STEPS:

1. Call into the **Automated Attendant** menu.
2. Select the touch-tone digit that has been set for modifying forwarding settings.
3. The system will prompt for an **Extension Number** and **Password**.
4. The system will indicate if extension forwarding is **Enabled** or **Disabled**.
5. Pressing “**1**” toggles between **Enabled** and **Disabled**.
6. Pressing “**2**” allows the forwarding destination to be modified.

Advanced Settings

Network Settings

When installing a Softphone change the SIP Password in Network Settings to match the password created in the Softphone Account Settings. The rest of these settings represent service provider permissions and identification information. **These other system (extension) defaults should not be changed.**

APPENDIX 4: IP ADDRESSES

Determining Local Network Address (on a Microsoft® Windows® Machine)

To determine the local network address on a Microsoft® Windows® machine:

STEPS:

1. Go to **Start** and click on **Run**. Type "**cmd**" and press **Enter**.
2. At the Command Prompt type "**ipconfig /all**" and press **Enter**. Windows will display the current connection details **(Diagram 63).** These connection details contain important information used in the setup of the IPitomy IP PBX.
3. Write the following information in the IPitomy IP PBX Installation Checklist:
   - **IP Address**
   - **Subnet Mask**
   - **Default Gateway**

Determining an Available IP Address

Adding the IPitomy IP PBX to an existing Local Area Network (LAN) requires knowing what IP address can be given to the system. Every new piece of equipment added to the network must have its own IP address. Pinging an IP address determines if an IP address is available to be used. A ping is a call to the IP address to see if it exists. If a ping gets a response then the IP address is in use.

To ping an IP address on a Windows®-based machine already configured on the local network:

STEPS:

1. Go to **Start** and click on **Run**. Type "**cmd**" and press **Enter**.
2. At the **Command Prompt** type "**ping *[192.168.1.249]**" and press **Enter**. This IP Address is the IPitomy IP PBX default static IP Address. The IP Addresses you ping on your local network will be defined by your router. For example, even though the default IP Address for the IPitomy IP PBX is 192.168.1.249, the IP Addresses that you ping to determine if one is available may differ. This will be determined by your gateway (router) address.
3. If you receive a "**Destination net unreachable**" message like in the number *[192.168.1.249] is an **IP Address not in use** and can be assigned to the IPitomy IP PBX.

   NOTE: *Sample configurations are for a Windows XP® operating system and Linksys® router and are for demonstration purposes only. This IP Address is determined by the default gateway (router) you discovered in Determining the Local Network Address.*

4. If you get a return or "**Reply from *192.168.1.249**", the **IP Address is in use**. Continue pinging IP addresses on "*[192.168.1.XXX]" that are greater than "*[192.168.1.249]" until you find a "Destination net unreachable." Make a record of this IP address in the IPitomy IP PBX Installation Checklist to use later in the installation.

APPENDIX 5: DHCP SETTINGS

Internet Protocol (IP) addresses are used to facilitate communication between equipment within a network. Each piece of equipment on a network has a unique IP address. These addresses are maintained in a list on a server (a computer that manages information traffic throughout a network). The list has both static (non-changing) and dynamic (varying) IP addresses. Those that are static do not change and can be used by Web servers that require a permanent connection to the Internet. Those that are dynamic are assigned as needed to allow communication between equipment or servers on the network. Dynamic Host Configuration Protocol (DHCP) lets Network Administrators manage an organization's network centrally and automate the assignment of IP addresses using the internet protocol TCP/IP. DHCP supports the management of IP addresses and internet traffic by:

Automatically assigning an IP address to a computer if it moves to another location on the network. If DHCP didn't exist the new IP address would have to be manually configured.

Allowing a Network Administrator to supervise and distribute IP addresses from a central point— automatically sending a new IP address when a computer is plugged into a different place in the network.

Leasing Time for an IP Address

DHCP uses the concept of a "lease" or an amount of time that a given IP address will be valid for a computer. The lease time can vary depending on how long a user is likely to require the Internet connection at a particular location. DHCP is especially useful in education and other environments where network users change frequently. In these settings, by using very short leases, DHCP can dynamically reconfigure networks in which there are more computers than there are available IP addresses.

Determining/Viewing DHCP Settings on a Linksys® Router

STEPS:

1. **Log into the router** using the router's IP address (e.g., 192.168.1.1).
2. Enter a **Username** and **Password** at the Login page.
3. Click the **Advanced Tab** and **DHCP**.

Tips for Understanding the IP Address List

Listed below are some tips that explain the DHCP Settings:

- The starting IP address determines the first IP that will be auto assigned by the router (e.g., 192.168.1.100).
- The number of DHCP users determines the amount of auto assigned IP addresses that can be generated (e.g., 50).
- In a scenario with 50 users, "192.168.1.49" would be the last auto assigned IP address in the pool.
- To set an IP address outside the DHCP pool, statically assign the system to a number beyond the pool (e.g., 192.168.1.249).

APPENDIX 6: ROUTER CONFIGURATION

The sample setup instructions below use a Linksys® router and are meant for demonstration purposes only.

STEPS:

1 **Log into the router** using the router's IP address (e.g., 192.168.1.1).

2 Open a Web browser. Enter the IP address of the router (e.g., 192.168.1.1) and press Enter.

3 Login to the router using the Username and Password. For help refer to router documentation.

4 Navigate to Port Forwarding. For help refer to router documentation. Open ports 5060 to 5060 on the IP address for the system being installed (e.g., 192.168.1.249) for both UDP and TCP (Name = SIP).

5 Open ports 10000 to 20000 on the IP address for the system being installed for both UDP and TCP (Name = RTP).

6 Open ports 4569 to 4569 on the IP address for the system being installed for both UDP and TCP (Name = IAX2).

7 Click Apply Changes to commit the changes to the database.

APPENDIX 7: NETWORK CONSOLE

The system comes with an application that allows the IP Address and other network attributes to be changed to meet the requirements of the specific network. Using the Network Console allows you to view and change settings without requiring an external web browser.

Accessing the Network Console

STEPS:

1 Verify that the IP1500 is powered and that a keyboard and monitor are connected directly to it.

2 If the monitor screen is blank go to step 4. However, if you see the Network Console, go to step 5.

3 If you just powered up the IP1500, you will see various system housekeeping messages scrolling on the screen. Do not press any keys at this time. Soon the messages will stop moving and be replaced with a "login as" prompt.

4 Press **CTRL** + **ALT** + **F7**, simultaneously. If nothing happens, go to step 1.

5 View or make changes as needed.

Selecting Options

Options are single letters or digits within brackets. You select an option by typing the option letter/digit and then pressing the Enter key. For example, on the Main menu, **[1]** indicates you can select the **Network Type** by typing **1** and pressing **Enter**. A blank option **[ ]** indicates that you cannot edit the value in the current context.

Menus

- Each menu screen contains 4 sections:
- Menu Location.
- “Using” (Live) System Settings.
- Database Settings.
- Commands.

Menu Location

The location “bar” tells you where in the configuration system you are. For example, the Main Menu is “IPitomy Networking – TCP/IP Settings”. Another menu is “IPitomy Networking – TCP/IP Settings – Subnet Mask”

Using the “Live” System Settings

This section always contains the current network system information.

Database Settings

Most values in this section are IP Addresses. The values can be edited. To edit a value you first select the corresponding option. Options here are always numbered. Edits do not modify actual network settings until you “Save & Update”.

Command Options

To issue a command select the desired option. Which options are available vary depending on where you are within the menu system. Some important options are:

Refresh View [R]

- When to use:
- Undo edits you do not wish to save.
- See the latest live network information.
- See if the database has been modified by someone else.

Default Settings [D]

Changes the values listed in the menu to factory defaults. Remember, you must save your changes to actually update your network interface.

Save & Update [S]

Save settings and update network connections.

Disable [Z]

IP address values can be disabled with this option.

Network Type

DHCP

DHCP may be used to simplify networking administrative tasks such as initial system configuration. With Network Type set to DHCP, several IP Address options become unavailable. For more information about DHCP and IP1500 networking requirements, refer to the appropriate section of this manual.

Acquiring a DHCP Lease

STEPS:

1 Change **Network Type** to "DHCP".

2 Save & Update.

Renewing DHCP Lease

STEPS:

1 Verify **Network Type** is set to "DHCP".

2 Save & Update.

Other Menus

- [W] Web Server
- [H] Host Access
- [F] Fix Database

Web Server

This menu allows you to restart the Web Server, and modify which IPs and/or domains can access the web server. To apply changes you must restart the web server. This will not interrupt phone service. A reboot of the PBX will also apply changes made here. The "Allow" format may be: domain name, full IP address, partial IP address, network/netmask pair, network/nnn CIDR specification.

Changes to the Web Server access list are database independent so custom changes do not migrate from one box to another via a database backup file.

Host Access

Limit access to special services on the PBX.

An "allow from" entry is a list of one or more host names, host addresses, patterns or wildcards that will be matched against the client host name or address. List elements should be separated by blanks and/or commas.

Changes to the Host access list are installed immediately. They are database independent so custom changes do not migrate from one box to another via a database backup file.

*Changes made to Web Server and Host Access settings are installed immediately. You must restart the Web Server ([U] option, or the PBX) before Web Server access changes are used by the Web Server. The Web Server and Host Access console menus provide all of the functionality of the **System > Access Control** web page.

Fix Database

From here you can factory default your database. This will permanently change your system. Be sure this is really what you want to do before proceeding.

Tips for Using the IPitomy IP PBX Network Console

Always read the text near the cursor before pressing keys. Otherwise you could miss an important warning message.

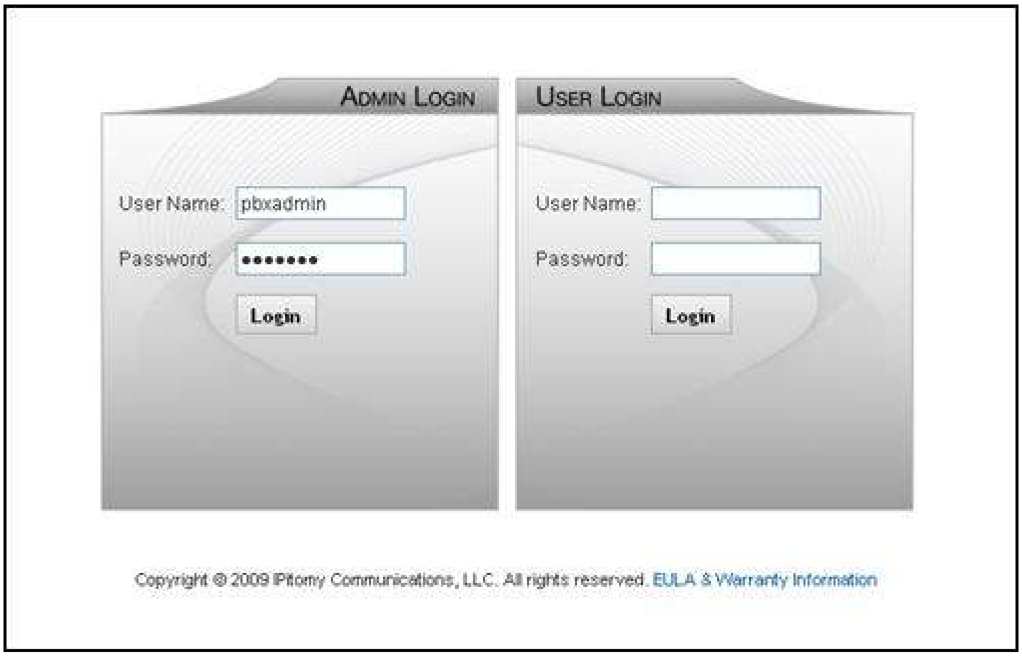
If the IP1500 console has been left unattended with the Network Console Menu present on the monitor screen, and you believe someone else may have modified network settings remotely since the last time the console was used, be sure to select Refresh Database View before making changes or using Save & Update. This ensures that you do not accidentally revert settings to an undesired prior state and that you are not relying on out of date information.

While unlikely, it is possible for the actual database settings to be out of sync with the “Using” (live) network settings. If you want to switch the actual settings to the database values, simply select Save & Update. If you want to re-configure the network, edit the settings first. Then Save & Update.

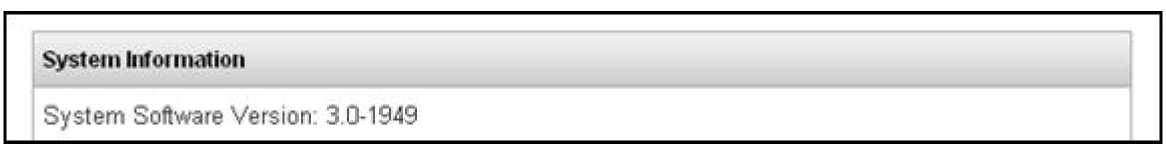
APPENDIX 8: SOFTWARE UPGRADE

Checking for Upgrade Availability

1. Log into the PBX



2. Navigate to PBX Setup=>Services and note the firmware version. If the PBX firmware is earlier than 3.x-xxxx then your system requires a Hard Drive upgrade before you will be able to load any 3.x firmware onto the system. Please contact your IPitomy Sales Representative if an HD upgrade is needed; otherwise continue with the steps laid out in this guide.



System Backup

1. Go to PBX Setup=>Database
2. Click Backup to create a backup of the current database
3. Find the most recent backup and click the Down action arrow to download this database to your computer

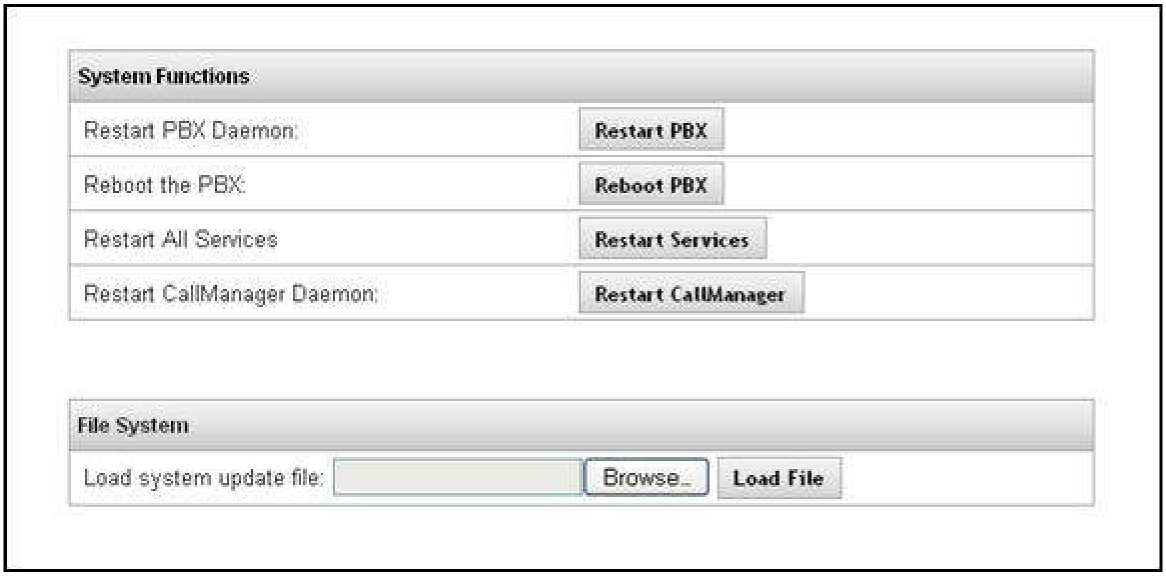
Downloading the Firmware File

1. In a web browser, navigate to http://www.ipitomy.com/pbx_files/ip1500/

2. Download the Current Release

Loading the Firmware File

1. Log into the PBX and navigate to PBX Setup=>Services



2. Scroll down to the File System panel

3. Click Browse and find the current firmware file *ipitomy_pbx_<build#>-uf.tgz*

4. Click Load File to begin the upgrade process

5. Depending on your starting firmware version, one of two things will occur:
    1. A dialog box will pop up announcing that the PBX will reboot, click OK and wait 5 minutes. After the 5 minutes has passed, log out of the PBX, then log back into the PBX (you should see a message when logging in that the Database does not match and will be converted)
    2. A progress bar will appear while the PBX upgrades and reboots, after which you will be returned to the login screen.
6. Log into the PBX and a message will display stating the Database version does not match. Click OK and the DB will update.
7. Click Apply Changes

Restoring Backups (Does not need to be done if you want to use the same DB as was previously running on the system)

1. Navigate to PBX Setup=>Database

2. Find the backup you wish to load and click the curved arrow to restore it as the active database.
3. Once the Database loads, Click Apply Changes.

3. Navigate to PBXSetup->Services and Click "Restart All Services"

For systems with Digium© Analog Cards/T1/PRI Cards or Xorcom© USB Devices

1. If your system has any of these products installed, navigate to Providers=>Hardware Providers

2. Scroll down to the Hardware Functions panel

3. Click Restart USB Devices & PBX Services

APPENDIX 9: TROUBLESHOOTING (FAQ)

(Q) Why does my Aastra phone freeze with 100% done written on the screen?

(A) The phone is attempting to access an unreachable network. You need to manually change the TFTP server IP on the phone to match the PBX IP. Also make sure that PBX Network Settings IP matches your actual PBX IP. If it does not, change it and SAVE the change. Make sure that the PBX Sip IP has been properly set:

STEPS:

1 Change the setting in **PBX Setup / SIP**.

2 Click on “**Apply Changes**” button to commit the changes to the database

3 **Restart** the phone.

(Q) Why did my phone restart with an old configuration after using Auto Discovery to create a new extension, assign and restart the phone?

(A) When you create a new extension you are taken to the Create Extensions page. After you click on the "Create" button you must then click on "Apply Changes" before restarting the affected phones.

(Q) Why does my Aastra phone hang (or freeze) with "40% Done" written on the screen?

(A) This is a known issue with the Aastra 480i model that sometimes occurs after Factory Defaulting the phone. You need to turn the phone off and turn it back on. Remove the power connection, wait a few seconds and then re-connect the power.

(Q) Why won’t my time zone on my voicemail change after my system settings have been changed?

(A) You have to go to PBX>Services and click on the Restart PBX button in order for the changes to take effect.

(Q) Why can’t I delete a menu?

(A) Be sure the menu is not being used in incoming routing.

(Q) What is the default IP Address for the IP PBX?

(A) The IP Address for the PBX is **192.168.1.249**

For the most recent troubleshooting FAQ’s please visit www.ipitomy.com

GLOSSARY

The following terminologies relate to either standard Internet Protocol (IP) and/or IPitomy's suite of products.

A

Analogue Telephone Adapter (ATA) – Connects a telephone to a high-speed modem and facilitates VoIP or fax calls over the internet.

***Automatic Call Distribution (ACD)* -** The IPitomy's ACD product is an IP-based component that allows management of calls and features Virtual Call Centers, Skills-based Routing, Multiple Queue Assignments and Overflow/Load Balancing.

B

Backbone – Global network connections that route voice and data traffic from one major metropolitan area to another.

Bandwidth – The transmission capacity of a given device or network.

Broadband – An internet connection that is always-on and fast.

Browser – A software application that allows users to view and navigate to information on the Web. Microsoft® Explorer® and Mozilla Firefox® are two common browsers.

Busy Lamp Field (BLF) –This is an LED on a phone that designates the status or another phone or feature in the PBX.

C

Caller ID – Displays the name and telephone number of a person calling.

Call Detail Record (CDR) – Information about calls collected from the IPitomy IP PBX for a specified period of time. This report is downloadable. The report details the numbers of calls, call duration, call origination and call destination.

Class of Service (CoS) – Class of Service defines which outbound routes a particular extension or feature has access to for outbound dialing.

Cloud Computing - refers to applications and services offered over the Internet. These services are offered from data centers all over the world, which collectively are referred to as the "cloud." This metaphor represents the intangible, yet universal nature of the Internet.

CODEC (Compression-Decompression) – Defines the algorithm used to compress and decompress audio, video, etc. The IPitomy PBX has many codecs to choose from.

D

Digital Subscriber Line (DSL) – This service provides digital phone service over an analog line.

Direct Inward Dial (DID) – When a trunk has multiple lines, you can use a DID to route calls in the PBX to specific destination.

Domain Name Server (DNS) – These are servers on the public internet which resolve the domain name into an IP address. Without a valid DNS server, many of the services used by the PBX (Time, Email, etc) will not be accessible if they use a domain name.

***Do Not Disturb* –** Prevents notification of incoming calls.

DTMF (Dual-tone Multi-frequency) – This is the touch-tone or audio signal a phone sends to a phone system to get it to perform some action.

Dynamic IP– An IP address that changes and is not static.

***Dynamic Host Configuration Protocol (DHCP)* -** This is a computer networking protocol used by hosts (DHCP clients) to retrieve IP address assignments and other configuration information.

E

Encryption – The process of scrambling data to prevent the accurate interpretation of this data by anyone except those for whom it is intended.

Ethernet - a family of frame-based computer networking technologies for local area networks (LANs).

F

***Firewall* -** A firewall is a part of a computer system or network that is designed to block unauthorized access while permitting authorized communications. It is a device or set of devices which is configured to permit or deny computer based application upon a set of rules and other criteria.

Forward – the action of automatically configuring a call that was intended for one party to go to another destination.

G

Gateway – A device that interconnects networks with different, incompatible communications protocols.

H

I

Institute of Electrical and Electronics Engineers (IEEE) – An independent institute that develops networking standards.

Infrastructure – Currently installed computing and networking equipment.

IP Telephony – Phone service (voice calls) carried over a network using Session Initiation Protocol.

Interactive Voice Response (IVR) - IVR is a telephony technology that can read a combination of touch tone and voice input. It gives users the ability to access a database of information via phone. A typical IVR system has several menus of prerecorded options that the caller can choose from. While many choices are as basic as choosing a number, some options may require the caller to speak detailed information such as his name or account number. This input is read by the IVR system and is used to access the appropriate information in the database.

Internet Protocol (IP) – A protocol used to send data over a network. These are currently set to IPv4 meaning 4 octets ranging from 0-255. The public internet takes up a majority of these addresses, but there are certain ranges reserved for private LANs.

Reserved Private IP Address Ranges			
	Start	**End**	**Number of Addresses**
24-bit Block (/8 prefix, 1 × A)	10.0.0.0	10.255.255.255	16,777,216
20-bit Block (/12 prefix, 16 × B)	172.16.0.0	172.31.255.255	1,045,576
16-bit Block (/16 prefix, 256 × C)	192.168.0.0	192.168.255.255	65,536

Internet Service Provider (ISP) – A company that provides access to the Internet.

J

Jitter - This is the deviation in or displacement of some aspect of the pulses in a high-frequency digital signal. As the name suggests, jitter can be thought of as shaky pulses. The deviation can be in terms of amplitude, phase timing, or the width of the signal pulse.

K

L

Local Area Network (LAN) – A group of computers and other devices that share a common communications line. These devices most often share a server and are located within a small geographic area.

Loop Back - The term loopback is generally used to describe methods or procedures of routing electronic signals, digital data streams, or other flows of items, from their originating facility quickly back to the same source entity without intentional processing or modification. This is not something you want to do on a LAN.

M

MAC Address - A **Media Access Control address** (**MAC address**) is a unique identifier assigned to network interfaces for communications on the physical network. http://en.wikipedia.org/wiki/MAC_address

Message Waiting Light – A light on a phone indicating that a voicemail message is waiting.

Music on Hold – Music or announcements callers listen to while on hold.

N

Network – A group of computers or devices that share a common communication line and are typically used for the transmission of data and voice traffic.

Network Address Translation (NAT) – Network address translation converts your LAN IP to the Public IP, allowing traffic to go out to the internet and route back to the correct device on your LAN. http://en.wikipedia.org/wiki/Network_address_translation

O

***Original Equipment Manufacturer (OEM)* -** This refers to a company that produces hardware to be marketed under another company's brand name.

P

Packet – A unit of data transmitted over a network. This is a small amount of computer data sent over a network. Any time you receive data from the Internet, it comes to your computer in the form of many little packets. Each packet contains the address of its origin and destination, and information that connects it to the related packets being sent.

Park – Parks a call in a reserved extension (park slot) and allows the call to be retrieved from another extension.

POE (Power over Ethernet) – A method of providing power to a network device over the network cabling, eliminating the need for a power supply.

Port – Ports can describe two different devices.

An Internet port. This is a number that indicates what kind of protocol a server on the Internet is using. For example, Web servers typically are listed on port 80. Web browsers use this port by default when accessing Web pages

A hardware port. This refers to any one of the ports that are on the back of a computer where devices can be hooked up (like a keyboard, mouse, printer, etc). Some common ports found on today's computers are USB, Firewire, and Ethernet.

PRI (Primary Rate Interface) – ISDN service provides 23 64-Kbps B (Bearer) channels and one 64-Kbps D (Data) channel (23 B and D). The D Channel is used for control in signaling information.

Private Branch Exchange (PBX) – An in-house telephone system that connects extensions and the Public Switched Telephone Network.

Public Switched Telephone Network (PSTN) – This is the global circuit-switched telephone network. It is similar to the Internet. However, on the Internet packets of data are sent and received using Internet protocol over a network.

Q

***Quality of Service (QoS)* -** The ability to provide different priority to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow.

R

Redundant Array of Independent Disks (RAID) – This is a method of storing data on multiple hard disks. When disks are arranged in a RAID configuration, the computer sees them all as one large disk. However, they operate much more efficiently than a single hard drive. Since the data is spread out over multiple disks, the reading and writing operations can take place on multiple disks at once. This can speed up hard drive access time significantly. Multiple hard drives may not improve hard disk performance as much as multiple processors may enhance the CPU performance, but it is based on a similar logic.

Real-Time Protocol (RTP) - a standardized packet format for delivering audio and video over IP networks.

Router – A networking device that connects multiple networks together, such as a local network and the Internet.

S

Server – Any computer in a network that provides users access to files, printing, communications, etc.

Server Time – this is the time set on the server where the phone or device is located. The server time for

IPitomy's IP PBX system is synched to time.nist.gov server time.

Session Initiation Protocol (SIP) – A signaling protocol that establishes data sessions. For example when making a call from one extension to another on a VoIP phone system SIP sets up the call and creates the connection between the two extensions.

Smart Personal Console (SPC) – This user-friendly Web page gives a person the ability to set basic phone features (mailbox settings, phone key settings, call forwarding, etc) from anywhere. Additionally, the user can check their voicemail and call logs from the SPC.

Stateful Packet Inspection (SPI) – A stateful inspection is a firewall architecture that works at the network layer. Unlike static packet filtering, which examines a packet based on the information in its header, a stateful inspection tracks each connection traversing all interfaces of the firewall and makes sure they are valid.

Static IP – an IP address that does not change.

Subnet Mask – A subnet mask is a number that defines a range of IP Addresses that can be used in a network. Subnet masks are used to designate subnetworks, or subnets, which are typically local networks LANs that are connected to the Internet. Systems within the same subnet can communicate directly with each other, while systems on different subnets must communicate through a router. Therefore, subnetworks can be used to partition multiple networks and limit the traffic between them.

Switch – A network device that connects network segments. These come in POE and non-POE varieties.

T

T1 – A dedicated digital voice circuit that has 24 channels. This point-to-point circuit delivers 1.544 Mbps of bandwidth.

TFTP (Trivial File Transfer Protocol) – a simple protocol to transfer files. IPitomy uses this method in regards to phone firmware and configuration files. http://en.wikipedia.org/wiki/Trivial_File_Transfer_Protocol

Time to Live (TTL) – This refers to an aspect of the Internet Protocol. TTL is used when a "ping," or a request for a response, is sent to another computer, such as a server. The TTL represents the number of hops, or servers in different locations, the request can travel to before returning a failed attempt message.

Transfer – Sends a call to another extension.

Trunk – A communications channel between two points.

Transmission Control Protocol (TCP) – Packets sent via this method are verified end to end.

U

Upload - While downloading is receiving a file from another computer, uploading is the exact opposite. It is sending a file from your computer to another system. It is possible to upload and download at the same time, but it may cause slower transfer speeds, especially if you have a low bandwidth connection. Because most files are located on Internet servers, people generally do a lot more downloading than uploading.

Uniform Resource Locator (URL) - A URL is the address of a specific Web site or file on the Internet. It cannot have spaces or certain other characters and uses forward slashes to denote different directories. Some examples of URLs are http://www.cnet.com/, http://web.mit.edu/, and ftp://info.apple.com/.

Uninterruptible Power Supply (UPS) – A devise that maintains continual electrical power.

User Datagram Protocol (UDP) - Packets sent via this method are fire and forget.

V

Virtual Private Network (VPN) – a computer network that uses a public telecommunication infrastructure, such as the Internet, to provide remote offices or individual users with secure access to their organization's network.

Voice Over Internet Protocol (VoIP) – The routing of voice traffic over the internet. VoIP is basically a telephone connection over the Internet. The data is sent digitally, using the Internet Protocol (IP) instead of analog telephone lines.

W

Wide Area Network (WAN) – A computer network that crosses geographic boundaries like cities, states or countries.

Wireless Local Area Network (WLAN) – A link between two or more computers in a network without wires. Wireless LANs use radio waves to communicate between computers in a limited area.

Notes: