IPitomy IP PBX User Guide

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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 like IPitomy Exchange, 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 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, adds 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.

How This Guide Works

Web-based System Setup

This is a Reference Guide designed to help you install and use the IPitomy IP PBX. Each section of the guide provides easy-to-follow instructions regarding installation of the system. Within each section of the Reference Guide you will find:

- **Step-by-Step Instructions** – Use these easy-to-follow steps as part of any system implementation.

- **Advanced Settings** – These options are settings for handling some of the more sophisticated capabilities of the IPitomy IP PBX.

- **Installation Notes** – These business scenarios and tips describe applications where or when a specific feature might be used.

- **Quick Reference** – These are tips about completing fields throughout the administration of the IPitomy IP PBX. Just move your mouse over the ⓘ and a brief description of the field pops up.

The Installation Worksheet

Use the **IPitomy Setup Worksheet** to make collecting information used in the implementation of the system simple. This **Worksheet** can be downloaded from IPitomy.com in the Dealer Section of the site.

Product Overview

IPitomy IP PBX Components

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

Powerful All-In-One Communications Platform

The IPitomy IP PBX **(Diagram 1)** 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 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)

(Diagram 1)

Entering System Information

The system is configured by entering information into the appropriate fields on the menu screens. External data sources such as CSV files or network scans can optionally be used to enter extension data. Some fields are populated with data that is entered directly; other fields are completed by selecting from data presented in a drop-down menu. Drop-down menus are populated by completing information in other sections of the system. To simplify system setup it is recommended that information be entered in the following order:

- Extensions
- Groups
- Prompts
- Menus
- Providers
- Call Routing

Entering information in this sequence will reduce the time it takes to get the system running because:

- Extensions will be populated in the drop-down menus for creating Groups.
- Extensions, Groups, and Prompts will be populated for creating automated attendant (Menu) routing.
- Destinations will be populated for use in setting up Providers and Hardware Trunks.
- Destinations, Providers, and Hardware Trunks will be available to select for Incoming and Outgoing Call Routing.

General Settings		Forwarding Settings	
Name:	chris chandler	Unconditional:	Disabled
Number:	105		PSTN Number
Email:	chris@phonesanddata.com		
Status:	active	Busy:	Disabled
Class of Service:	System Default		PSTN Number
PIN:	105		
Ring Time:	32	No Answer:	Disabled
Call Group:	1		PSTN Number
Pickup Group:	1		
Apply Schedule:	☐	Unavailable:	Disabled
CID Override:	Enabled ○ Disabled ●		PSTN Number
CID Name:			
CID Number:			

(Diagram 2)

System Administration

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 get 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.

- **Reports** – These reports display system usage, monitor activity, and provide diagnostic information.

System Overview

The system is designed to be quick to setup and install. Using the Installation Worksheet to organize system information and plan the application in advance will reduce the time it takes to install the system. Most businesses will have some common communication needs. The system is organized based on these common needs.

Icon System

- Edit main settings.

- Edit an extension's phone settings.

- Delete an item from the system.

- Download a file from the system.

- Restore a file to the system.

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 when 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 by using the Group feature.

Automated Attendant (Menu)

To create an automated attendant use the system's Menu function. The Menu function routes calls to a destination in the system like a group, extension or another menu. 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.

Advanced Routing Functions

When building an automated attendant (menu) all routable destinations in the system will appear in the drop-down menu. 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 must 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 (menu). 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, 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.

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. This can be an individual extension, group, conference or menu.

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.

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 (menu) option to be programmed.

Voicemail Gateway

From the automated attendant (menu), 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.

Supported IP Phone Sets

The IPitomy IP PBX works with a variety of business-grade IP phone sets. See **Appendix 1: IP Telephones** for a complete list.

### *It is highly recommended to Use IPitomy IP550 SIP Telephones*

Before Getting Started

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 Web site.

Connecting the System

Hardware Setup

The IPitomy IP PBX comes assembled and ready to install. The system requires connection to the PSTN for analog or T1 lines. It requires telephones to be connected to the local area network (LAN). Broadband access must also be established for VoIP connectivity (allowing remote extensions and remote management).
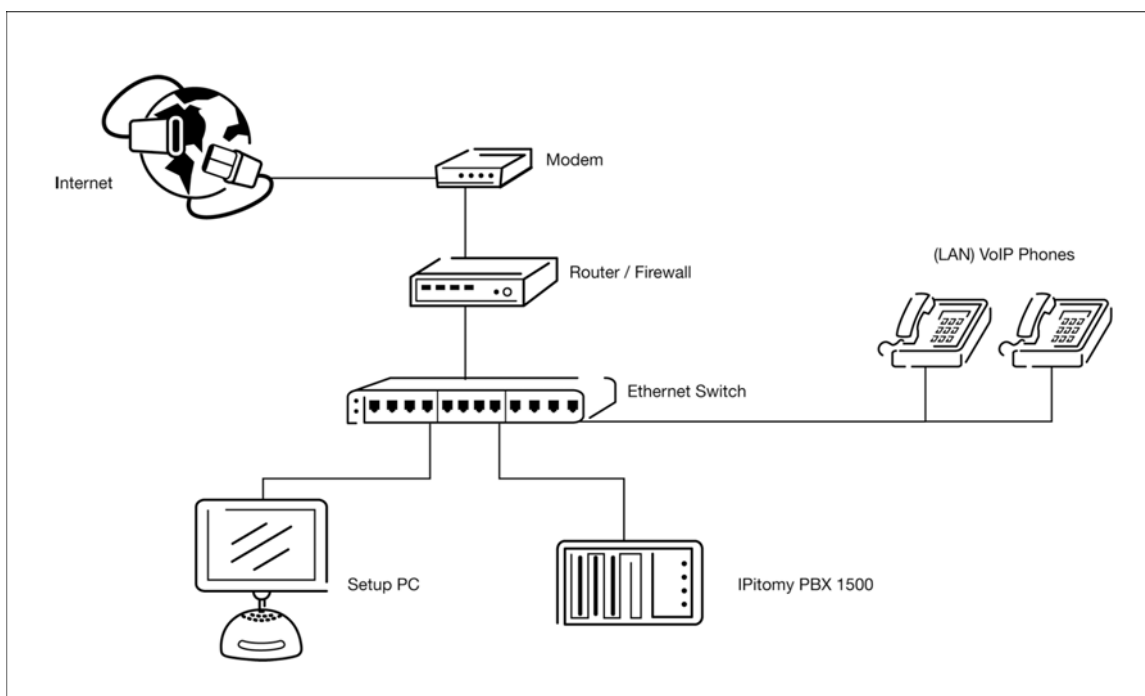
(Diagram 3)

Connecting the Phone Lines

The IPitomy IP PBX is equipped to support analog, digital, gateway or SIP connections. Analog lines or a T1 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.

- **The Digium TDM04B Analog Line Card (4 PSTN line connections) (Diagram 4)** – The 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. These connections are single pair; one line per jack. It may be necessary to echo tune the system. Please call IPitomy Support (941-306-2200) if you notice an echo on the line.

- **The Digium TDM808B Analog Line Card (8 PSTN line connections)** – The 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. These connections are single pair; one line per jack. It may be necessary to echo tune the system. Please call IPitomy Support (941-306-2200) if you notice an echo on the line.

- **The 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.

Internal Digital T1 Cards

The IPitomy IP PBX uses two devises for digital connectivity.

- **Digium TE122P T1 Card (a single T1/E1/J1 connection) (Diagram 5)** – 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 4)

(**Diagram 5)**

Connecting Using an External Gateway (Diagram 6)

PSTN lines are 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.

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.
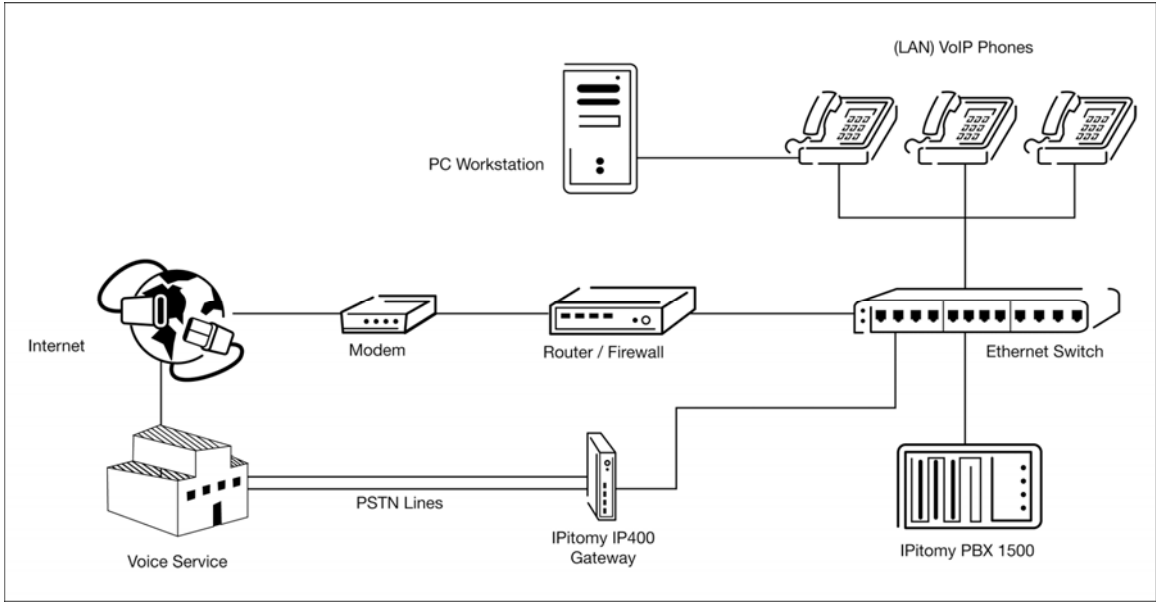
(Diagram 6)

Connecting Telephones

(**Diagram 7)**
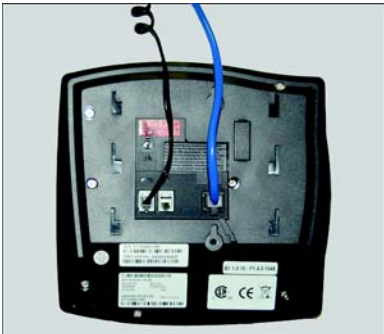
(Diagram 8)
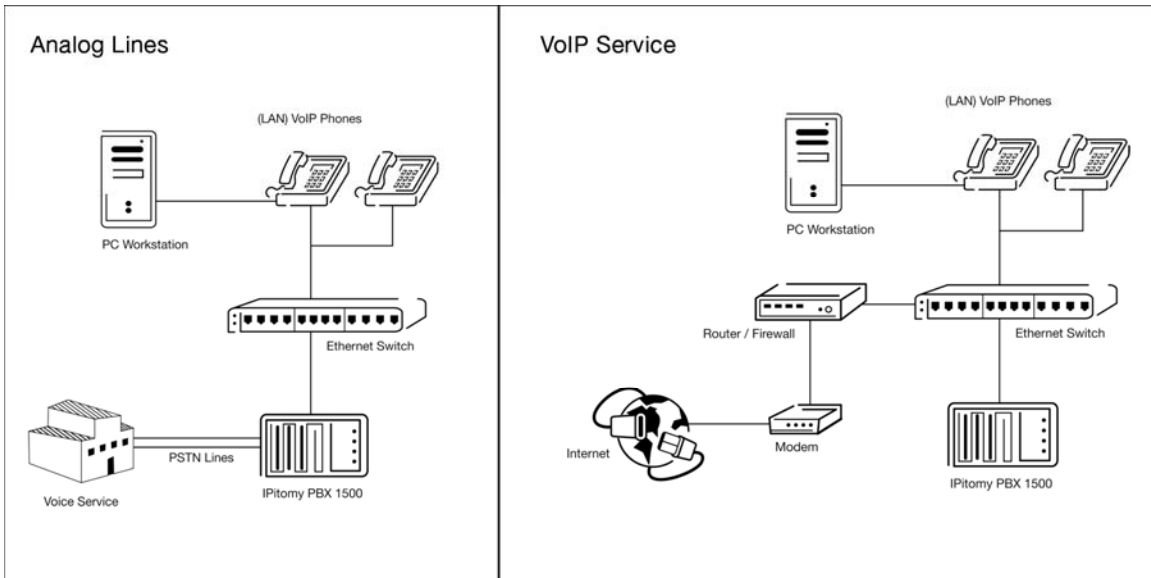
(Diagram 9)

Connecting to a LAN
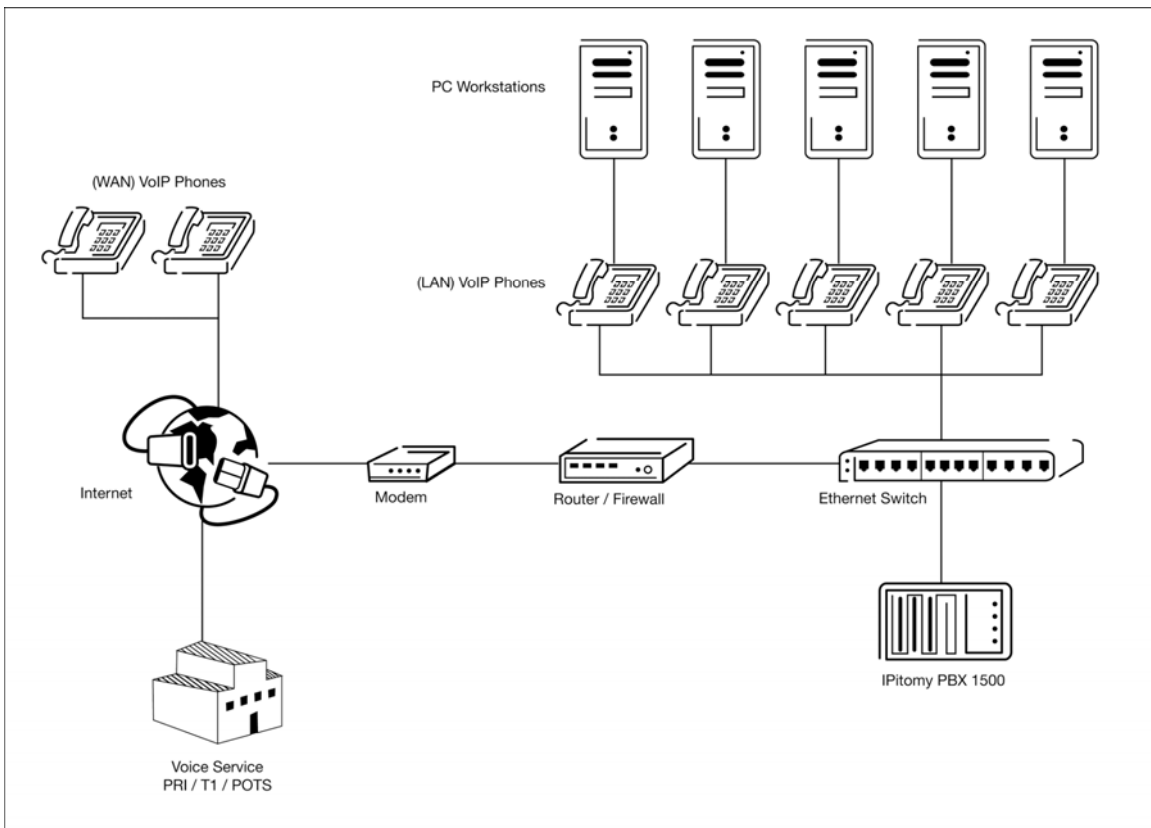
(Diagram 10)

System Requirements

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 (option 66) 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.

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. 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**. 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).

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. Plug a keyboard and monitor into the IP PBX. Once at the PBX login screen just press **CTRL + ALT + F7** and follow the onscreen instructions. For further details on using the Net Console please refer to appendix 5.
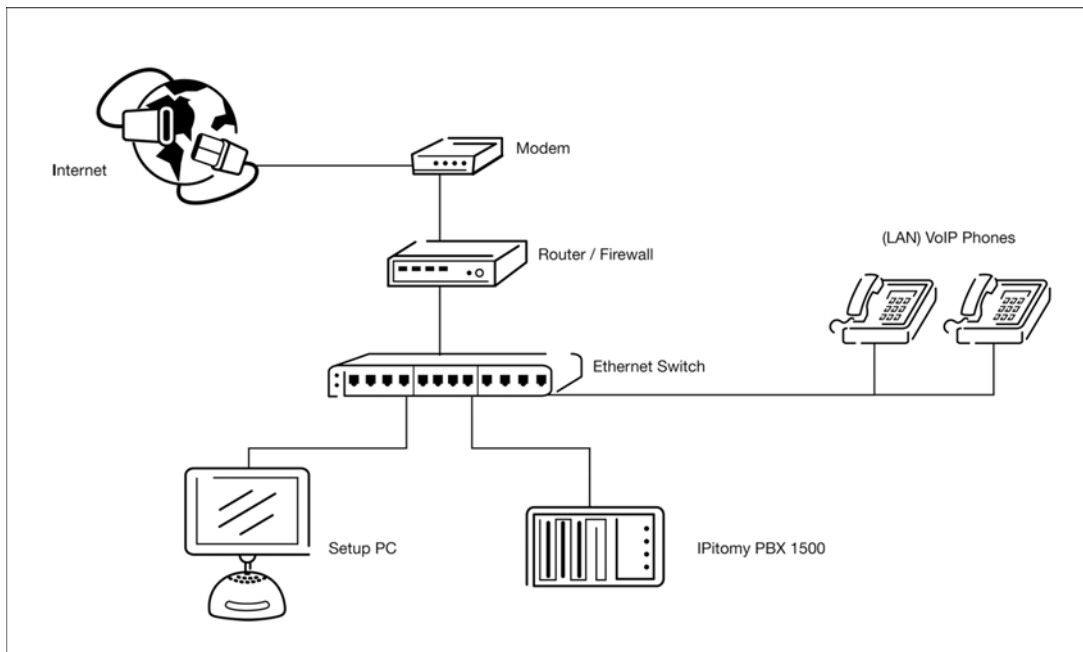
(Diagram 11)

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

Record this information on the **IPitomy Setup Worksheet**.

System Administration
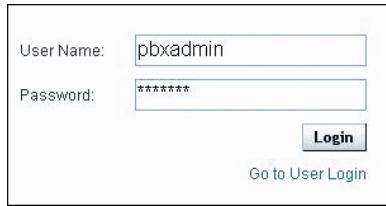
About the Administration Menu

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. 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.

- **PBX Setup** – System settings allow global configuration settings for system applications like PBX timers, voice messaging settings and music-on-hold.

- **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.

- **Reports** – The system displays usage reports, diagnostic information and monitors system activity.

Each Online Administration page also contains:

- **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.

- **Save Changes** – Located in the bottom left corner of the screen, this button saves changes to the page currently being administered. **This button must be pressed before leaving a page or changes will be lost**.

- **Apply Changes** – **To apply changes to the system you must click the Apply Changes button.** Located in the top right corner of the screen, this button globally applies changes to the system.

- **Edit** – To make **changes to an existing administration page** click **Edit**.

- **Add New** – The Add New button **creates another destination, provider, route or schedule**. For example, to add a new extension, click the Add New button on the extension administration page.

- **Advanced** – In several sections of the online administration there is an **Advanced** button where the **most sophisticated capabilities of the IPitomy IP PBX** can be configured. The Advanced button is located on the lower left side of each page.

- **About Us** – Located at the bottom left corner of each page, this link provides additional information about **IPitomy**.

- **Contact Us** – 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.

Log In

(Diagram 12)

User Name: pbxadmin
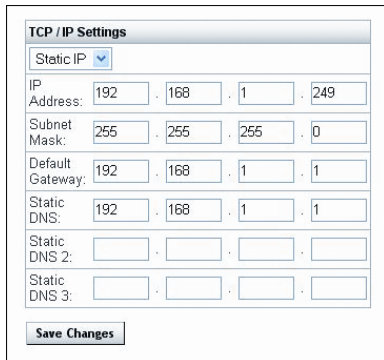
Password: ipitomy

System

(Diagram 13)

The IPitomy System Menu is for setting up network attributes. For example the IP address of the system and router information.

(Diagram 14)

Networking

The Networking Setup Menu defines 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.

Default values for the IP Address, Subnet Mask, Default Gateway and Static DNS will appear in the Networking Setup Menu when this administration page is opened.

Setting Network Attributes

- **Router Configuration** – Though it is not seen on this page, be sure to **configure an existing router for the fixed IP address of the IPitomy IP PBX (192.168.1.249)**. Select another address if the default address is not outside the range of IP addresses currently assigned through DHCP on the router.

- **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)**. Provided by the router, this mask tells the network what communication traffic to listen to. For example, the setting 255.255.255.0 tells the system to listen to communication traffic sent to "192.168.1" for any variation of the fourth number (designate by the zero at the end).

- **Default Gateway** – **The default gateway provided is 192.168.1.1**. Though this default is common to many routers if this is not the gateway for the router used, change it to the correct value for the target network.

- **Static DNS** – **Enter the DNS IP address of the router 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.

- **Save Changes** – **Click the Save Changes** button to save changes made to the hardware setup.

Access Control

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.

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 provisioned. The system is equipped to handle two types of provider settings Hardware Trunks and SIP Providers.

 (Diagram 15)

 (Diagram 16)

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.

(Diagram 17)　　　　　　　　　　　**(Diagram 18)**

(Diagram 19)

Connecting Phone Lines

Before hardware trunks can be provisioned, they must be connected to the system.

Connecting Using Internal Analog Line Cards

- **The Digium TDM04B Analog Line Card (4 PSTN line connections) (Diagram 17)** – Connect the phone lines to the RJ11 jacks at the rear of the IPitomy IP PBX and start making calls. These connections are single pair; one line per jack. It may be necessary to Echo tune the system. Please call IPitomy Support (941-306-2200) if you notice an echo on the line.

- **The Digium TDM808B Analog Line Card (8 PSTN line connections)** – Connect the phone lines to the RJ11 jacks at the rear of the IPitomy IP PBX and start making calls. These connections are single pair; one line per jack. It may be necessary to Echo tune the system. Please call IPitomy Support (941-306-2200) if you notice an echo on the line.

- **The Digium 2404B 16 and 2406B 24 Analog Line Cards (16 or 24 PSTN line connections) (Diagram 18)** – These lines require an Amphenol connector for two wire connection to a 66 block cross connect, breakout box with RJ11 connectors or a patch panel with RJ11 telephone jacks.

Connecting Using Internal Digital T1 Cards

- **Digium TE120P T1 Card (a single T1/E1/J1 connection) (Diagram 19)** – This card supports industry standard telephony and data protocols. The card has a default configuration for the system, but this configuration may be adjusted based on preference. On the rear of the system, plug the RJ45 connector into the T1 TDM source supplied by the T1 provider. The T1 cable is usually red.

- **Digium TE205P T1 Card (a dual T1/E1/J1 connection) (Diagram 15)** – This card supports industry standard telephony and data protocols. The card has a default configuration for the system, but this configuration may be adjusted based on preference. On the rear of the system, plug up to 2 RJ45 connectors into the T1 TDM sources supplied by the T1 provider.

Provisioning a New Hardware Trunk Group

Channel Group Parameters Card 1

Group Name:	Signalling Type:	Generate Ringing on outbound calls? ☐	Answer Incoming?
Nuvox	fxs_ks ▾		Yes ▾

Start Channel:	Area Code:	Allow Caller to transfer an outbound call? ☐	Answer After:
1	941		0 sec

End Channel:	Dial Prefix:	Allow Call Recording? ☑	Ext CID Override? ☑
23	w		

Use Caller ID:	Echo Cancelation:	RX Gain:	Restrict CID Override? ☐
No ▾	Yes (128) ▾	0.0 db	

Inbound Caller ID:	Echo Cancelation Bridged:	TX Gain:	Busy Detect:
asreceived	No ▾	0.0 db	Yes ▾ : 8

Outbound Caller ID Name:	Echo Training:	Route Calls to:	
IPitomy	800	Menus ▾	Menu: Auto Attendant ▾
		...	

Outbound Caller ID Number:
941-306-2200

Phone Numbers

This section contains phone numbers, (sometimes called DIDs) associated with this provider.

[ ] add

```
941-306-2201
941-306-2202
941-306-2204
```

remove

Destination: None ▾ Set

(Diagram 20)

Provisioning a hardware trunk tells the system what phone numbers are associated with the trunk. It also establishes rules for the system to follow when processing incoming and outgoing calls through this physical network connection. To provision a hardware trunk:

1. Click on **Providers** and **Hardware Trunks** in the navigation bar on the left of the administration menu. The Hardware Trunk page will appear. This is where hardware trunks will appear when they have been provisioned.

2. The system can contain 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** for each card in order to tell the system which cards you have.

3. Once you have set the card you can click the Add Lines button to the right of the corresponding card to define line/channel/trunk groups associated with that card. Click **Add Lines**. The Edit Hardware Provider

page will appear.

4. Enter the **Name** you would like associated with this group of channels.

5. Define the **start** and **end channel** fields relative to the board. If you would like a 1 channel group set the start and end channel to the same value. Port 1 on analog card 1 would be set as start channel 1 and end channel 1. You will then be able to address these channels individually in incoming and outgoing routing.

6. Select a **Signaling Type**. For analog lines this should generally be fxs_ks. For T-1 this will be normally be fxs_ls, and for T-1 PRI this should be set to pri_cpe.

7. Select **Use CallerID** yes or no. This tells the system to attempt to detect caller ID on incoming calls for the corresponding channels.

8. **Define the Inbound Caller ID** for inbound calls on this trunk. If set to 'asreceived' the calls will receive 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. If you need to configure the system this way, contact IPitomy Tech Support at (941)306-2200 for the proper syntax.

9. **Define the Outbound Caller ID Name** for outbound calls on this trunk. If this is left blank, the name displayed on outbound calls will match what is registered with the provider. Otherwise the name typed here will override the outbound caller ID name, pending the provider supports it.

10. **Define the Outgoing Caller ID Number –** If the trunks support setting outgoing caller id (primarily only on T-1 channels), this field will set the caller ID number for outgoing calls on the associated channels.

11. **Set the Area Code -** this field is obsolete and will be removed in a future release.

12. The **Route Calls To** field is the default incoming destination for this line. The default destination will typically be a Menu, Extension, Schedule, Voicemail, or a Ring Group. The default destination is the destination all calls from this provider will be routed to unless they are routed specifically to a different destination as with a specific DID number. (Note that DID numbers are only available for routing on a trunk that provides DNIS such as a group of T-1 PRI channels). It will be necessary to come back to this screen once all of the destinations have been configured to complete any routing if the destinations have not been entered, or you can set this through **Call Routing->Incoming**. Once all of the destinations are created, they appear in a drop-down menu and are selectable. Note that by selecting None here you are specifying that undefined (non-DID) numbers will be routed to the system default incoming destination.

13. **Check Allow Call Recording** if users of these trunks will be allowed to record calls. In a standard business application this feature is typically not needed.

14. **Set the Dial Prefix** if your trunk lines require a specific dialing pattern to reach the PSTN. Typically this is a 9w which means dial 9 then wait 1 second. 'w' can be used repeatedly to cause a 1 second pause.

15. **Set the TX Gain %** - to boost the transmit gain on the corresponding channels. Typically these settings don't need to be changed from default, unless there are issues with the volume of calls.

16. **Set the RX Gain %** - to boost the receive gain on the corresponding channels. Typically these settings don't need to be changed from default, unless there are issues with the volume of calls.

17. **Set Echo Cancellation** to yes if you want echo cancellation on the line. This should be on normally, and can be adjusted if echo issues occur.

18. **Set Echo Cancellation Bridged** to yes if you want echo cancellation on calls that are bridged from one port to another on the board.

19. **Set the Echo Training** field to adjust the time spent training the echo cancellation algorithm before a call is picked up.

20. **Set Answer Incoming** to yes if you want the PBX to answer incoming calls on this line.

21. **Set Answer After** to the number of seconds to wait before answering the line once it starts ringing.

22. **Enable Ext CID Override** to allow the users to set an alternate caller ID under their extension that will override the outgoing caller ID.

23. **Enable Restrict CID Override** to limit an extensions ability to override CID, unless it is specifically one of the DID numbers associated with that trunk.

24. **Set Busy Detect** 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 hangs up.

25. **Add the Phone Numbers associated with the provisioned trunk**. (Note that DID numbers are only available for routing on a trunk that provides DNIS such as a group of T-1 PRI channels). Type the phone number in the field provided and click Add. The phone number will appear in the field to the right. To remove a number from the list click the Phone Number and the Remove button.

26. **Define a Destination for the number** by clicking it and selecting where it is to be routed to from the drop-down menu. Destinations can be:

    - **Extensions** where specific people or departments can be reached.

    - **Menus** through which callers can personally select from a variety of destinations in the system.

    - **Locations** in the business like Conference Rooms.

    - **Ring Groups** through which people respond to similar types of calls.

    - **Schedules** that route callers to different destinations based on the time and date.

    - **Voicemail Boxes** where callers can leave a message or get information.

Installation Note 1

How Destinations Populate

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. Hardware can be provisioned without the destinations and set once they have been added to the system.

Assigning a Destination

The destinations are places to route calls to in the system:

Menu destination is used to setup an Auto Attendant or other set of voice instructions for callers, such as driving directions. This is where callers can select specific persons or departments within an organization after receiving instructions from a voice prompt.

Group destination is where calls are routed to a group of extensions, such as, sales, customer service, accounting or possibly everyone. Groups are groups of extensions.

Extensions are destinations for Individual telephones. Calls can be routed directly to extensions.

Schedules are destinations that supersede the destination that calls are routed to and execute a routing plan based upon the schedule times.

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27. Click the **Save Changes** button to save changes made to the hardware provider setup.
28. Click **Apply Changes** when ready to implement these changes to the system.

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 just like a trunk (T1 or plain old telephone line).

SIP Provider settings vary widely. To simplify the provisioning we recommend that you ask the provider for a sample Asterisk configuration.

SIP Provider

Field	Value
Name:	IP1Network Demo Sip T
User Type:	friend
DTMF Mode:	auto
Host:	204.10.106.225
Register:	◯ Yes ◉ No ◯ Custom
Authorization:	◯ Yes ◉ No ◯ Custom
From Domain:	◉ Default ◯ Custom
Outbound Proxy:	◉ Disabled ◯ Enabled
Username:	2396773181
Secret:	sipdgpyr
Inbound Caller ID:	
Outbound Caller ID Name:	IP1-IPITOMY demo trnk
Outbound Caller ID Number:	2396773181
Call Limit:	5
Qualify:	30000
Default Destination:	None
Dial Prefix:	
Area Code:	
Generate Ringing on outbound calls:	☐
Allow Outbound Caller to transfer:	☐
Allow Call Recording:	☐
Ext CID Override:	☐
Restrict CID Override:	☐
Can Reinvite:	◯ Yes ◉ No ◯ N/A
Insecure:	Very

Allow Codecs:

Codec	
ITU G.711 ulaw 64Kbps, (US)	☑
ITU G.711 alaw 64Kbps	☑
ITU G.723.1 (5.3/6.3 Kbps, 20ms frame size	☐
ITU G.726 - 16/24/32/40 Kbps	☐
GSM - 13 Kbps (full rate), 20ms frame size	☐
iLBC - 15Kbps,20ms frame size, 13.3 Kbps, 30ms frame size	☐
Speex - 2.15 to 44.2 Kbps	☐
LPC10 - 2.5 Kbps	☐

Phone Numbers

This section contains phone numbers, (sometimes called DIDs) associated with this provider.

[input] [Add]	7545818408 2396773181 [Remove]

Destination: None [Set]

(Diagram 21)

Add a New SIP Provider

1. Click on **Providers** and **SIP Providers**. The SIP Providers landing page will appear.

2. Click on **Add Provider**. The Edit SIP Providers page appears.

3. **Assign a Name** to the SIP Provider.

4. **Select the User Type** associated with the SIP Provider.

5. **Select DTMF Mode** associated with the SIP Provider.

6. **For the Host** enter the domain or IP address associated with the SIP Provider.

7. **Select the Register** option needed for the SIP Provider.
   Yes – automatically registers with provided settings.
   No – Doesn't register.
   Custom – Allows for any special registration rules needed for the SIP provider.

8. **Select Authentication Mode** for the SIP Provider.
   Yes – automatically generates based upon provided settings.
   No – Doesn't use authentication for the SIP Provider.
   Custom – Allows for any special authentication rules needed for the SIP provider.

9. **From Domain** is hardly used, and if needed the SIP Domain information will be provided to you from the SIP provider.

10. **Set Outbound Proxy** if the provider requires you to use a different outbound IP for SIP.

11. **Enter User Name** associated with the SIP Provider.

12. **Enter the Secret** "password" for the SIP Provider.

13. **Inbound Caller ID** 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. If you need to configure the system this way, contact IPitomy Tech Support at (941)306-2200 for the proper syntax.

14. **Enter Outbound Caller ID Name** to override the caller ID name that is displayed on outgoing calls through this trunk.

15. **Enter Outbound Caller ID Number** to override the caller ID number that is displayed on outgoing calls through this trunk, pending it is permitted by the provider.

16. **Set the Call Limit** "number of lines" for the SIP Provider.

17. **Set the Qualify** to the number of milliseconds 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.

18. **Set the Default Destination** where inbound calls will be routed to, unless they are individually routed, as with a DID.

19. **Set the Dial Prefix** if it is required by the SIP Provider.

20. **Area Code** is an obsolete field and will be removed in a future release.

21. **Enable Generate Ringing on Outbound Calls** in the rare case when the provider does not generate the ringing for you.

22. **Enable Allow Outbound Caller to Transfer** to allow a user to transfer a call that they originated.

23. **Check Call Recording** if users of this provider will be allowed to record calls. In a standard business application this feature is typically not needed.

24. **Enable Ext CID Override** to allow the users to set an alternate caller ID under their extension that will override the outgoing caller ID.

25. **Enable Restrict CID Override** to limit an extensions ability to override CID, unless it is specifically one of the DID numbers associated with that trunk.

26. **Can Reinvite**
    Yes – select this if the phone type allows the re-invite feature.
    No – select this if the phone type doesn't allow the re-invite mechanism for reconnecting the call mid stream.

27. **Select Insecure** option from provided drop down list. This specifies how to handle connections with peers.
    Port - ignore the port number where authentication came from.

    Invite - do not require 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 port.
    Very - To allow registered hosts to call without re-authenticating.

28. **Allow Codecs** – encodes a stream or signal for transmission. Select which codecs will be enabled for this provider.

29. **Add the Phone Numbers** associated with the provisioned trunk. Type the phone number in the field provided and click Add. The phone number will appear in the field to the right. To remove a number from the list click the Phone Number and the Remove button.

30. **Define a Destination** for the number by clicking it and selecting where it is to be routed to from the drop-down menu. Destinations can be:

    - Locations in the business like Conference Rooms.

    - Extensions where specific people or departments can be reached.

    - Ring Groups through which groups of people respond to similar types of calls.

    - Schedules that when applied route callers to different destinations or people in the organization during specified times and dates.

    - Voicemail Boxes where callers can leave a message.

    - Menus through which callers can personally select from a variety of destinations in the system.

Installation Note 2

How Destinations Populate

The destinations dropdown list is populated as destinations are added to the system. During system implementation these destinations can be populated first and then the hardware provisioned or the hardware can be provisioned without the destinations and set once they have been added to the system.

Assigning a Destination

Most often businesses assign phone numbers to a:

- Main Menu destination where callers can select specific people or departments within an organization.

- Ring All Ring Group destinations where everyone in the business has access to answering all calls.

31. Click the **Save Changes** button.
32. Click **Apply Changes** when ready to implement these changes to the system.

Edit an Existing SIP Provider

1. Click on **Providers** and **SIP Provider**. The SIP Providers landing page will appear.

2. Select a provider from the list in the SIP Provider Window on the landing page.

3. Select a **Provider** by clicking on the **Provider Name** or **Edit**.

4. Make changes to the provider.

5. Click on **Save Changes**.

6. Click on **Apply Changes** when ready to implement these changes to the system.

Delete a SIP Provider

1. Click on **Providers** and **SIP Provider**. The SIP Provider page will appear.

2. Discern which provider from the list in the SIP Provider Window on the landing page is to be deleted.

3. Click on **Delete** button next to the selected provider. The SIP provider will be removed from the SIP Provider Window on the landing page.

Destinations

Destinations are the various places that a call can be routed to within the system. Destinations in the IPitomy IP PBX include:

- **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 a destination within the system. A voice recording can also be used to play information like driving directions.

- **Meet-me Conference Rooms** are where callers can call into a conference call. Callers can be routed in any way to a Meet-me Conference; direct dialed, 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 to 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.

These system destinations (where and to whom calls will be routed) should be planned in advance. In most cases, there will be a menu to setup for an automated attendant. There will also be business hours to setup. The **IPitomy Setup Worksheet** helps organize provisioning information prior to installation.

Extensions

Extensions define where specific people or departments can be reached in an organization. They should be setup first in the system. Once extensions are setup then the rest of the application can be provisioned.
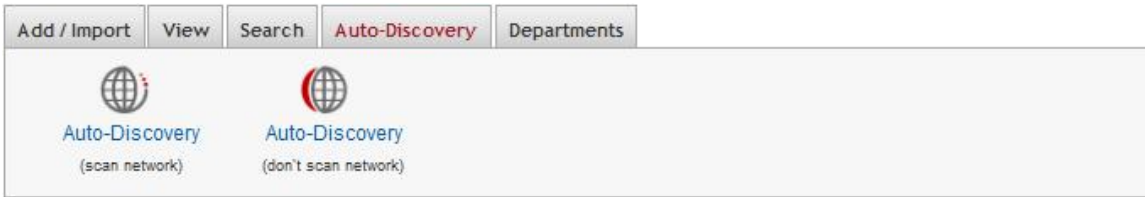
(Diagram 22)

- **Auto Discovery** - This network discovery tool identifies phones and other devices connected to the network and allows you to remotely configure networked phones.

- **Add/Import** – This tab is where you would go to create new extensions or edit existing extensions.

  o **Add/Delete Extensions** – Allows the user to add one or a number of extensions.
    - **Mass Edit** – The user is able to select multiple extensions and edit certain fields for all of those extensions all at once. Only fields that have been modified will be saved to the extensions selected.

  o **Import/Export CSV file** – Easily upload a CSV file with information to auto create extensions, or export previously created extension information into an easy to store CSV file for backup or template purposes.

- **View** – This tab allows the user to sort the display of extensions by Phone Model

- **Search** – 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

- **Phone Settings** - This tool allows you to modify a phone's keys. It includes support for all Aastra and IPitomy model phones.

- **General Settings** – Identifies the owner of the extension, email address, extension and password information.

- **Forward Settings** – Sets the forwarding behavior. Calls can be forwarded to local destinations, PSTN and cell numbers.

- **Network Settings** – These are extension authentication settings for SIP registration with the IPitomy IP PBX. This is also where selecting the model of telephone for Auto Configuration and changing button mapping is done.

- **CODEC Permissions** – The quality of voice transmission and bandwidth utilization are dictated by CODECs. Different CODECs compress voice packets differently. G.711 is the most common and highest quality, but consumes the most bandwidth. G.729 requires license fees to use in conjunction with the IPitomy IP PBX services like voicemail, conferencing and music on hold.

- **Voicemail Settings** – Manages voicemail messaging and routing.

- **Calling Permissions** – Defines types of calls that can be originated, received and some advanced options like park and record calls.

- **Follow-Me -** Allows the user to establish a failover routine to find them across an assortment of locations (your extension, your cell phone, etc) that can be assigned priorities.

Auto Discovery

This network discovery tool identifies phones and other devices connected to the network and allows you to remotely configure networked phones. You are given the choice to scan the network, or if a scan has been done previously you could select No Scan to save time.

(Diagram 23)

(Diagram 23a)

Toolbar: Edit Selected

Button	Function	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	No

	to selected phones.	
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

Toolbar: Command Selected

Button	Function	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 specified under Advanced Settings. Then the phone is commanded to restart.	Yes
Assign, Configure & Restart	Combines the functions of the Assign button with the Configure & Restart button.	Yes

Toolbar: View Settings

Button	Function	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	No

| | using the settings in Advanced Scan Settings. The scan results displayed depend on the active filters (controlled via Advanced Filter Settings). | |
| Filter Only | Displays results of the last network scan after applying the filters set in Advanced Filter Settings. | No |

Tip: You must check the checkbox beside each item you wish to modify before issuing any commands.

List of Devices & Extensions

The list of discovered network devices uses two sources of information:

- o Network
- o PBX Database

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. The help/info question mark to the right can be hovered over to get more specific info about the device.

⑦ Hover your cursor over the info icon to obtain details about the networked device or phone. This can be an important source of information about which phones are available for live configuration.

If you click refresh-view more than once in a short time period, the network is not re-scanned. The minimum time period allowed between scans is indicated under advanced scan settings (Min Refresh Interval).

Tip: The network is only scanned for new information when you click [Refresh]. Both Refresh & [Filter Only] apply whatever filters you have chosen to the displayed list.

Advanced Filter Settings

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.

- **Sort Order**

Determine what order items are listed in. Hover your cursor over the help icon next to the Sort Order fields for options.

- **Hide**
  You can hide different status types from view.

- **Filter Patterns**
  You can enter a partial MAC or IP here. The limitation is that the pattern must match from the first character onward. i.e. "000D" would match "000DE" but not "A000DE".

  More advanced pattern filtering:
  Full Regular Expression Matching is supported when you enclose your search pattern in double quotes.

Pattern	Equivalent to
1.2.3	^1\.2\.3
"1\.2\.3.*"	1\.2\.3.*

- **Filter by Type:**
  You can select multiple device types to filter for. 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.

  Tip: Only Aastra phones are shown by default because they are set in the Filter by Type.

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.

The 'Reset' button will return all Scan Settings to original default values.

Advanced Scan Setting	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 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.

Troubleshooting Network Scanning Problems

Scan settings have been tuned to work in a network environment with two switches and a router between the PBX and the phone. 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.

To tune scanning for your network:

1. Click on Advanced Scan Settings

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.

Tip: Phones do not show up as 'online' in a network scan if they are in the process of restarting, turned off, or operating on a different network.

Example Workflow # 1 – Create, Assign & Configure phones.

This assumes that the phones are in a Factory Defaulted state.

1. Go to the **Extensions page**. Click the "**Create**" button.

2. Enter the new extension information or upload a **CSV file** for the extensions. You do not need to provide MAC addresses. Then Click the "**Create**" button. If successful, Click "**Return to Extensions**"

3. Click the "**Auto-Discovery**" button.

4. Next to each phone you wish to configure, click the checkbox and use the drop down **"Assigned"** list to select the extension you wish to assign to the phone.

5. Click the "**Assign**, **Configure** & **Restart**" button under the Commands tab.

Example Workflow # 2 – Create, Factory Default, Assign & Configure phones.

1. Go to the **Extensions** page. Click the "**Auto-Discovery**" Button. This will scan the network for phones and other devices.

2. Click the **checkboxes** next to phones you wish to configure. Then Click the "**Create**" button. This takes you to the Extension Creation page.

3. Enter the new extension information or upload a CSV file for the extensions. Then Click the **"Create**" button. If successful, click the "**Return to Extensions**" button.

4. Click the "**Auto-Discovery**" button.

5. All of the phones you are attempting to setup are highlighted. Click the checkboxes next to these phones (verifying that you are only checking the ones from step 2.).

6. Click the "**Factory Default**" under the Commands tab. 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.

7. Click the **checkboxes** next to the phones from step 2. Click the "**Assign**, **Configure**, and **Restart**" button under the Commands tab.

CSV Upload

Each line is a record consisting of 1-5 comma separated fields. The order of the fields must be:

Name, Email, Number, Phone Type, MAC

- o   Number: the extension number.

- o   Phone Type: This should be spelled out in English. Example Aastra 480i

You may omit fields starting from the right hand side of a record but the order of the remaining fields must match the file format above.

Comments may be placed in the CSV file by using '#' marks. Comments must either be at the start of a line or have only spaces in front of the mark. Comments must not be placed on the same line as a record entry.
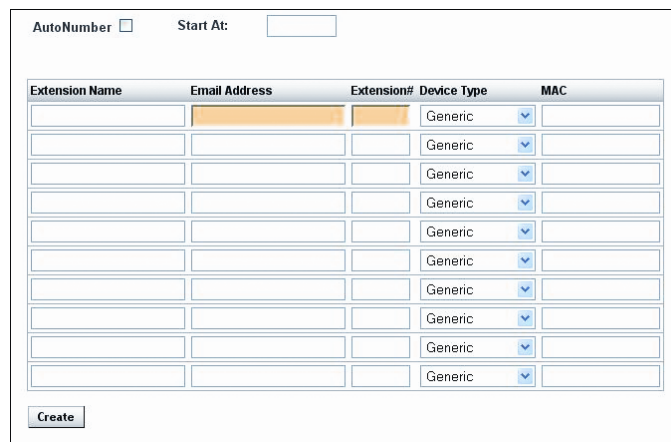
Using the CSV upload and including the phone type "IP550" will facilitate easy setup with the IP550 auto configuration process.

Add Range

(Diagram 24)

1. Go to **Destinations** > **Extensions**.

2. Enter the number of extensions to be created in the provided text box.

3. Click the **Add** button.

4. Give the system a starting extension #.

(Diagram 25)

5. Check the **Auto Number** check box.

6.   If Auto-Discovery is going to be used skip to **Step7**. If Auto Discovery is not going to be used than fill in all the pertinent information for each

extension.

7. Click the **Create** button.

8. The screen that follows displays the results of each extension. If satisfied with the results click "**Return to Extensions**".

Phone Settings
This tool allows you to modify a phone's keys. It includes support for all Aastra and IPitomy model phones. It includes support for Aastra phones running firmware version 1.4.2.1081

Buttons:

- **[Save Changes]** – Save changes but does not restart the phone.

- **[Save & Restart Phone]** - Save changes and restart the phone. The phone is only restarted if there have been changes.

- **[Exit]** - Leave the Phone Settings page without modifying any phone settings.

Table of supported phones, and key counts.

Phone Model	Programmable Keys	Top & Bottom Key Sections?	Expansion Module Support
IP550	26	Yes	No
9112i	2	No	-
9133i	7	No	-
480i & 480iCT	20	No	-
53i	4	No	-
55i	26	Yes	536EM & 560EM (Add up to 180 keys)
57i and 57i CT	30	Yes	536EM & 560EM (Add up to 180 keys)

Phones with top and bottom key areas will display shortcut links for Top & Bottom.

(Diagram 26)

The total number of keys displayed on the phone at any moment depends not only on the display area available, but also on the state of the phone lines (see Key State for more information). Expansion modules display a fixed number of keys and are not limited by state.

Tip: Although the 9112i has many physical keys, most of them were never supported by Aastra so they remain unusable.

A list of available keys fills most of the Phone Settings Page. Each row of the list contains: Key Type, Label, Value, and States (if applicable).

Key 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. Here is a complete list of programmable key types:

Key Type	Additional Information
BLF	Busy Lamp Field
Speed Dial	
Park	
DND	Do Not Disturb
Blind Transfer	
Call Pickup	
Voicemail	
Voicemail Gateway	
Record	
Fwd On	Forwarding On
Fwd Off	Forwarding Off
Set Fwd	Set Forwarding Number
Fwd Gateway	
Park Menu	
Intercom	
Directory	
Callers	

Services

Xfer

Aastra Park

Pause

Tip: Keys will only work if the System Administrator has configured the system for the particular function.

Key Label

The label is the display name to use for the key. It is visible on the phone's LCD screen (except for models 9112i and 9113i).

Key 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.

Keys that require users to enter values:

- o BLF
- o Speed Dial

Key State

Generally, a key is only active when the current extension line status matches the enabled states. Some special keys are always active, regardless of state. When a key is active you can see the key's label printed on the phone's display screen. When a key is inactive it is not displayed and cannot be used. Default states for each key type will be automatically filled in for you.

Key State	Description
Idle	No current call activity.
Connected	A call is connected.
Incoming	A call is ringing the extension.
Outgoing	A call is being placed from the extension.

To program a key's state:

Check the desired boxes (any of: Idle, Connected, Incoming, Outgoing). If no boxes are available for checking it means that the key will always be active.

Advanced Settings

Custom Sip Settings

Network	*LAN
	*WAN
	*Custom
Server	The IP address or name of the server to use for the Network type selected. A user must use the Custom Network type to set a user defined Server.

Time Settings

Network	*PBX
	*Custom
Time Server	The IP address or name of the server to obtain time information from. A user must use the Custom Network type to set a user defined Time Server.
Time Zone	*US-Eastern
	*US-Central
	*US-Mountain
	*US-Pacific
	*US-Alaska
	*US-Aleutian
	*US-Hawaii

Volume Controls

Headset Transmit Gain	This is the audio volume level (dB) transmitted by the PBX to the phone.
Headset Side Tone Gain	This is the audio volume level (dB) of your

	voice signal sent back into your ear so you can hear yourself talk and know that the line is not dead.
Handset Transmit Gain	This is the audio volume level (dB) transmitted by the PBX to the phone.
Handset Side Tone Gain	This is the audio volume level (dB) of your voice signal sent back into your ear so you can hear yourself talk and know that the line is not dead.
Hands free transmit Gain	This is the audio volume level (dB) transmitted by the PBX to the phone's speaker.
Microphone Mute	Disable the phone's microphone(s).

Aastra Expansion Modules

Module Type	*None
	*536 EM
	*560 EM

Working with Expansion Modules

 (Diagram 27)

When a phone has a Module Type set you will see the Module Type name and quick links to module keys at the top of the page.

The most advanced module is the 560 EM. Its LCD screen displays button names in two columns. Each column can be given a name via Phone Settings.

Module Descriptions and Options:
- o 560 Expansion Module -
  This module has 60 keys. It has a LCD screen split into two columns. There are 20 physical keys (10 per column) plus buttons to switch screens. There are 3 screens, bringing the total number of programmable keys to 60. You can Daisy Chain up to 3 modules to a single phone.
- o 536 Expansion Module -
  This module has 36 physical keys. Each key has an indicator lamp. The module requires printed labels to be inserted. You can Daisy Chain up to

3 modules to a single phone.

To Change Module Type

Warning: When changing module types, you will lose any existing module key settings.

1. Click on "Advanced Settings"

2. Select the Module Type connected to the phone.

3. Configure the module keys as desired.

4. Click on Save & Restart to immediately update the phone.

To Add a Module of the Same Type

Do not change the module type. Support for up to three modules of the same type is automatic. Simply go to the desired module via the quick links at the top of the Phone Settings page. Modules are numbered from left to right so Module 1 is the first module, and Module 3 is the last one.

Add a New Extension

1. Click **Destinations** and **Extensions** in the navigation bar of the system's administration menu. The Extensions page will appear.

2. Enter the number of extensions you wish to create and Click **Add.** The Create Extensions page will appear.

(Diagram 28)

3. Insert the **Name** to be associated with the extensions being created.

4. Enter an **Extension Number** for this person or department.

5. 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 when properly configured

6. Click **Create,** and then click Return to Extensions if the display shows the creation was a success. If the extension number selected is already in use by the system then the result will be conflict and the extension will not be created.

7. Find the extension that was created and click the Pencil icon to edit.

8. **Select a status** from the drop-down menu. An extension can be:

    -   -**Active** – Currently in use.

    -   -**Disabled** – Not currently in use.

9. **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.

10. **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.

11. **Define a Call Limit**. This is the number of concurrent calls allowable at an extension. The Call Limit selected must be between 1 and 99.

12. **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.

13. **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 99.

14. 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.

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 IPitomy IP PBX).

Changing Forward Settings

To enable or disable forward settings:

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.

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 public switched telephone number (PSTN) by entering it into the field provided. Calls can be forwarded to any destination from 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.

o Dial ***90** to disable forwarding.

o Dial ***91** to enable forwarding.

o Dial ***92** to set the forwarding number.

Changing a Forwarding Number from a PC

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 menu. 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.

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

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.

Advanced Settings

Network Settings

SIP Password	password [Generate]
Location	LAN (Local)
NAT	☑
Host	dynamic
Phone Type	Aastra 57i Settings
Phone MAC	
Qualify	30000
DTMF Mode	rfc2833
User Type	Friend
Call Limit	0
Can Reinvite	○ Yes ○ No ⊙
Insecure	Very

Voicemail Settings

Mailbox	500
Attach to Email	Yes ⊙ No ○ N/A ○
Delete After Emailing	Yes ○ No ○ N/A ⊙
Turn Old After Emailing	Yes ○ No ⊙
Say Caller ID	Yes ⊙ No ○ N/A ○
Allow Review	Yes ○ No ○ N/A ⊙
Allow Operator	Yes ⊙ No ○ N/A ○
Play Envelope Message	Yes ⊙ No ○ N/A ○
Auto Delete Voicemail in	90

Allow Codecs

ITU G.711 ulaw 64Kbps, (US)	☑
ITU G.711 alaw 64Kbps	☑
ITU G.723.1 (5.3/6.3 Kbps, 30ms frame size	☐
ITU G.726 - 16/24/32/40 Kbps	☑
GSM - 13 Kbps (full rate), 20ms frame size	☐
iLBC - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size	☐
Speex - 2.15 to 44.2 Kbps	☐
LPC10 - 2.5 Kbps	☐
H.263 - Video	☐

Calling Permissions

Local Calls	☑
Long Distance Calls (US)	☑
International Calls	☐
Incoming Outside Calls	☑
Internal Calls	☑
Allow Incoming Intercom Paging	☑
Allow Outgoing Intercom Paging	☑
Allow User to Forward Calls	☑
Allow User to Record Calls	☑
Allow User to Monitor Calls	☑
Allow Call Park	☑
Is Operator	☐

[Save Changes]

Allow Codecs		Calling Permissions	
ITU G.711 ulaw 64Kbps, (US)	☑	Allow User to control Forwarding	☑
ITU G.711 alaw 64Kbps	☑	Allow User to control Follow-Me	☐
ITU G.723.1 (5.3/6.3 Kbps, 30ms frame size	☐	Allow User to change Phone Key Settings	☐
ITU G.726 - 16/24/32/40 Kbps	☐	Internal Calls	☑
GSM - 13 Kbps (full rate), 20ms frame size	☑	Allow Incoming Intercom Paging	☑
		Allow Outgoing Intercom Paging	☑
iLBC - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size	☐	Allow User to Forward Calls	☑
Speex - 2.15 to 44.2 Kbps	☐	Allow User to Record Calls	☐
LPC10 - 2.5 Kbps	☐	Allow User to Monitor Calls	☐
H.263 - Video	☐	Allow Call Park	☑
		Is Operator	☐

Follow-Me	
Manage	**Numbers & Settings**

Save Changes

(Diagram 29)

(Diagram 30)

SIP Password

The password used to when the extension attempts to communicate with the PBX.

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

This setting should be checked unless specifically told to change it by an IPitomy representative.

HOST

This should be set to dynamic unless otherwise instructed.

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 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, 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 in tact 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 of rfc2833, auto, info, and inband.

User Type

This is set 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.

Can Reinvite

This setting should not be changed unless instructed to do so by an IPitomy representative.

Insecure

This setting should not be changed unless instructed to do so by an IPitomy representative.

Music OnHold

This setting allows the user to select a different Music On Hold playlist for their extension then the system default playlist.

CODEC Permissions (Allow CODECs)

These transmission speeds are delivered by the service provider and automatically register in the extension through the system. **These extension defaults should not be changed.**

Voicemail Settings

These settings manage voicemail messaging and routing. When a voicemail box is created with an extension, it is not possible to change the voicemail box in the extension screen.

- o **Attach to Email** – Send a voicemail message to an email address by attaching it to an email message as an audio file (.Wav).

- o **Delete After Emailing** – Delete the voicemail after it has been emailed to the email address provided for the extension in General Settings.

- o **Turn Old After Emailing** – Turns all voicemail in the voicemail box old so the message waiting light will not be turned on.

- o **Say Caller ID** – State Caller ID prior to playback of the message.

- o **Allow Review** – Allow callers to review a message after it has been played.

- o **Allow Operator** – Allow pressing "0" during the voicemail greeting to reach the system-wide operator.

- o **Play Envelope Message** – Play caller ID and time of call prior to audio version of a message delivered through email.

- o **Auto Delete Voicemail In** – Define the number of days in which voicemail messages are to be automatically deleted from a mailbox.

> **Installation Note 3**
> Not applicable (NA) accepts the system-wide default set in the System Setup section of the administration menu. If the global settings are acceptable, leave the NA setting.

Calling Permissions

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.

1. Check the permission box if an extension is to have the calling capability. The following permissions can be set for an extension:

   - **Allow User to Control Forwarding –** When checked, this allows the user to modify their forwarding settings when they log into the Smart Personal Console user interface.

   - **Allow User to Control Follow-Me –** When 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 –** When checked, this allows the user to modify their Phone Key settings when they log into the Smart Personal Console user interface.

   - **Internal Calls** – Permits calls made from internal extensions.

- **Allow Incoming Intercom Paging** – Permits a page to be heard through this extension.

- **Allow Outgoing Intercom Paging** – Permits a page to be made through this extension.

- **Allow User to Forward Calls** – Permits an extension to forward a call or voicemail message to another extension on the system.

- **Allow User to Record Calls** – Permits the extension to record phone conversations.

- **Allow User to Monitor Calls** – Permits a user to listen to another extension's (person's) phone calls.

- **Allow Call Park** – Permits extension to park a call.

- **Is Operator** – Designates the extension as an operator.

2. Click on Save Changes.

3. Click on Apply Changes when ready to implement changes to the system.

Edit an Existing Extension

1. Click on **Destinations** and **Extensions**. The Extensions page will appear.

2. **Select an extension** from the Extension Window on the page.

3. Select an **Extension** by clicking on the **Extension Name** or 🖉 **Edit**.

4. **Make changes to the extension.**

5. Click **Save Changes**.

6. Click **Apply Changes** when ready to implement changes to the system.

Delete an Extension

1. Click on **Destinations** and **Extensions**. The Extensions page will appear.

2. **Select an extension**.

3. Click ⊗ **Delete**. The extension will be removed from the Extensions Window on the page.

Auto-Provisioning an IP550

This Chapter will guide the technician through Auto-Provisioning of IP550 Phone sets. The purpose is to familiarize the Technician with the settings in the Phone sets as well as the appropriate areas of PBX programming, and how to utilize advanced features of this functionality as well as maintenance of the functions and the Phone sets. It should be noted that the standard method of Auto-Discovery of phones that IPitomy users are familiar with on Aastra Phones is also available on IP550. Auto-Provisioning is a feature enhancement though, for the IP550 and is not available on Aastra Phone sets.

The IP550 is supplied with 20 programmable Soft-Keys, as well as 6 pre-programmed

feature buttons. These buttons are pre-labeled and are as follows:

Park- This key will park the call in an available park assignment.

Voicemail- This button will take the caller to their voicemail login.

Pause- Is a BLF Key that can be assigned to a letter and the Extension number to indicate when an Agent in the ACD is paused. Pause is the equivalent of DND except it refers to Agents in a queue.

Pickup 701 – Picks up the call that is parked in Park 701

Pickup 702 – Picks up the call that is parked in Park 702

Pickup 703 – Picks up the call that is parked in Park 703

The functionality provided by the park pick up keys allows users to more easily pick up calls and is designed to provide legacy phone users with a more familiar experience of using the phone. It is designed to provide features that are similar to line appearances that users of legacy systems have come to expect. We have chosen this button programming to assist installers and technicians with the initial setup of phones with commonly required buttons which will abbreviate the install time considerably.

In order to Auto Provision you must follow the steps outlined briefly below.

- Go to PBX setup->General and turn on Auto-Provisioning
- Create an extension and under Advanced set the phone type to IPitomy 550
- On the phone press Menu followed by 7
- Enter the password (1234)
- Press 8
- Select YES for Auto PBX Search
- Enter the extension number of the extension that was just created
- The phone will be registered to the selected extension once it finishes rebooting
- Be sure to go back to PBX Setup -> General and Turn Off Auto-Provisioning when complete

To provide more detail, you must first browse to the PBX and then log in as an Administrator. The default Administrator credential is

Username = pbxadmin

Password = Ipitomy

Once logged in browse to PBX Setup and then go to the General Link and find the Auto Provisioning field and turn Auto- Provisioning to On. Once on it will indicate the status of "Running".

(Diagram 31)

Create an extension or extensions to be used by IP550 phones by browsing to Destinations and then Extensions and create either a single extension or a group of extensions to be utilized. You can also use the Import .CSV feature. This import feature is particularly relevant with Auto-Provisioning as you can in almost a single step import the Extensions list and create extensions. Then using the rest of the Auto-Provision setup the installer can rapidly deploy the phones to the desktop which is usually one of the most time consuming tasks in the installation process. This is a brief explanation of the import .CSV feature.

CSV Upload

When creating the CSV file each line is a record consisting of 1-5 comma separated fields. The order of the fields

Must be as follows:

Name, Email, Number, Phone Type, MAC Address

Number: the extension number.

Phone Type: This should be spelled out in English. Example IPitomy 550

You may omit fields starting from the right hand side of a record but the order of the remaining fields must match the file format above.

Comments may be placed in the CSV file by using '#' marks. Comments must either be at the start of a line or have only spaces in front of the mark. Comments must not be placed on the same line as a record entry. You may also utilize the IPitomy setup worksheet which is available at the IPitomy web site which will provide you with a quick way of gathering this information as well as generating the .CSV file.

For further instruction on the .CSV feature please refer to the Product Installation Manual for the PBX product that you are installing. Once you have created your extensions then edit the extensions, and browse to Advanced and set the phone type to IP550 as indicated in **Diagram 32**.

(Diagram 32)

The remainder of the configuration is done via the Phone interface. You must access the Menu button of the phone to be Auto-Provisioned. Then select option 7. You can do this without using the scroll keys by just pressing Menu then 7. It will prompt you for the Admin password which by default is 1234. You may then, using either the scroll key to select option 8, or just simply pressing 8, select the Auto-Provisioning Scan. When you are prompted, select Yes to scan for the Extension. At this point you must enter either the extension number you created or one from the range created to provision the phone. At this point the phone will re-boot and when it finishes its re-boot it will be provisioned with the Extension credential from that extension.

This feature can also be used to do Hot-Desk from any extension. If the Auto Provisioning is left running on the PBX then users can be trained to follow this procedure to Hot-Desk their phones. However care must be taken if the feature is left running for users who do not wish to Hot-Desk. If your users do not wish to use the Hot-Desk functionality, then it is recommended after you complete the Auto-Provisioning to return to PBX-Setup General and turn the feature off until needed once more. It should be noted that if no extension exists in the PBX database this process from the phone will create the basic Extension information.

Standard Button Programming is available as well on IP550 Phone through the PBX administration interface. Just as with Aastra Phones you can program the buttons and Soft-

Keys from within the PBX interface. To access the button mapping features you do this as you would Aastra phones, just browse to Destinations- Extensions, and then find the extension or group of extensions that you wish to edit, and then access the Phone edit interface

It should also be noted that you can always perform the standard Auto-Discovery that has been available for Aastra Phone sets. This process can now be performed on IP550 also. It is the same as discussed in the Auto-Discovery section of the Manual.

Top

Key	Type	Label	Value	Idle	Connected	Incoming	Outgoing
1.	None			☐	☐	☐	☐
2.	None			☐	☐	☐	☐
3.	None			☐	☐	☐	☐
4.	None			☐	☐	☐	☐
5.	None			☐	☐	☐	☐
6.	None			☐	☐	☐	☐
7.	None			☐	☐	☐	☐
8.	None			☐	☐	☐	☐
9.	None			☐	☐	☐	☐
10.	None			☐	☐	☐	☐
11.	None			☐	☐	☐	☐
12.	None			☐	☐	☐	☐
13.	None			☐	☐	☐	☐
14.	None			☐	☐	☐	☐
15.	None			☐	☐	☐	☐
16.	None			☐	☐	☐	☐
17.	None			☐	☐	☐	☐
18.	None			☐	☐	☐	☐
19.	None			☐	☐	☐	☐
20.	None			☐	☐	☐	☐

Bottom

Key	Type	Label	Value	Idle	Connected	Incoming	Outgoing
21.	Park	Park Call		☐	☐	☐	☐
22.	Voicemail	Voicemail		☐	☐	☐	☐
23.	Pause	Pause		☑	☑	☑	☑
24.	BLF	Pickup 701	701:Park_701	☐	☑	☐	☐
25.	BLF	Pickup 702	702:Park_702	☐	☑	☐	☐
26.	BLF	Pickup 703	703:Park_703	☐	☑	☐	☐

Advanced Settings

Save Changes Save & Restart Phone Exit

(Diagram 33)

Make you appropriate button changes from this screen and then select Save and Restart the Phone. Once the phone restarts the new buttons should be accessible. This process is unchanged from how it was performed with Aastra phones and should be recognizable to users of Aastra phones. Additionally these buttons mapping controls can be used via the Smart Personal Console for users with permissions to use the SPC. To access the Button mapping feature via the Smart Personal Console simply log in to your SPC, then browse to My Account and then access the My Phone Settings Link. This will bring you to the same Button Mapping interface as shown in **Diagram 33**.

Groups

The Group function allows incoming calls to be distributed to a group of extensions rather than just one extension. Within the Group function, different distributions strategies may be selected based on the call coverage required by the application. Groups can be intercom paging groups too. By dialing the intercom button or code followed by the group number, 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 group for Sales, Customer Service or Engineering departments. 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 call DID, the Auto Attendant, the choice selected by the incoming caller, or the time of day. Note that all calls to ring group members from the ring group number or routed to the ring group will bypass forwarding settings set on the PBX. The exception to this is that some phones have forwarding settings which are independent of the PBX (not set through the PBX Web GUI). These settings will still be applied.

Ring Group Examples

Ring groups define a set of extensions (people) that answer calls. Ring groups can be created for departments (e.g., Sales or Engineering) or business 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 option is selected from an Automated Attendant Menu, the call is routed to the group. Calls will be distributed to the members in the group 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

▪ Round Robin

▪ Round Robin (with memory)

▪ Least Recent

▪ Fewest Recent

▪ Fewest Calls

▪ Random

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.

Example Ring Group 1 – Departmental Grouping

Group 1 Sales	Ext.	Group 2 Customer Service	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** for this menu and group arrangement would read as follows:

Thank you for calling ATI Connect 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 the **ring group strategy** might be to have calls answered **Round Robin** or distributed to one Sales person after the other.

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 ATI Connect 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 in this group calls may be set to ring Least Recent. Using this strategy the person having answered the smallest number of recent calls would get the next incoming call.

Add a New Group

(Diagram 34)

1. Click **Destinations** and **Groups**. The Groups page will appear.
2. Click **Add Group**. The Edit Ring Group page will appear.
3. Enter a **Name** for the group.
4. Enter a **Group Number**. This number must be three or four digits in length.
5. Check **Allow Paging with ** + the Group Number** if this group will need to be paged. Note that only Aastra® phones support this off-hook paging

functionality at this point.

6. Define a **Ring Strategy** for the group by selecting it from the drop-down menu. Calls can ring:

- **Ring All** – Ring all phones in the group.

- **Round Robin** – Distributes calls to extensions consecutively one after the other. 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)** – Distributes calls to extensions consecutively one after the other. 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** – Distributes a call to an extension in the group with the longest time between calls.

- **Fewest Recent** – Distributes a call to an extension in the group that has taken the fewest recent calls.

- **Fewest Calls** – Distributes a call to an extension in the group that has taken the fewest total calls

- **Random** – Distributes calls randomly to the group.

7. **Choose a Fail over Destination** for the group's calls. 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. Selecting "None" will give the caller a fast busy.

8. **Define the time in which calls will timeout**, end the ring strategy and be sent to the fail over destination. Timeout can be between 1 and 32 seconds.

9. Add **Members** to the groups, and define priorities if so desired. On a scale of 0-9, 0 is the highest priority.

10. Click **Save Changes**.

11. Click **Apply Changes** when ready to implement changes to the system.

Advanced Settings

Call Ring Settings

Caller Ring Settings – Defines what a caller will here while they are waiting for someone to pick up a call from the call group. Callers can either hear:

- o **Ringing** – The phone continues to ring while the caller is waiting.

- o **Music on Hold** – The caller hears music while waiting for a group member to pick up the call.

Queue Dial String

Queue Dial String is an advanced group setting reserved for IPitomy administrative support. **This field should not be populated**.

Edit an Existing Group

1. Click on **Destinations** and **Groups**. The Groups page will appear.

2. Select a Group by clicking on the **Group Name** or ✎ **Edit**.

3. **Make changes** to the Group.

4. Click **Save Changes**.

5. Click **Apply Changes** when ready to implement changes to the system.

Delete a Group

1. Click on **Destinations** and **Groups**. The Groups page will appear.

2. Click ⊗ **Delete**. The Group will be removed.

ACD

ACD Settings - Agents	
Agent Ring Time	10
Agent Retry Timer	1
Weight	0
Wrap-up Time	0
Autofill	yes
Autopause	no
Maximum # of people in queue	0
Periodic Announce Frequency	0
Announce Holdtime	no
Join Empty Queue	yes
Leave Empty Queue	no
Report Hold Time	no
Ring in use	yes
Member Delay before connect	0
Timeout Restart	yes
Service Level	0
Announce Filename	None
Exit Menu	none

Advanced

(Diagram 35)

ACD Settings & Agents

Systems licensed to use ACD will have access to additional functionality pertaining to Ring Groups (**Diagram 35**). The 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.

Agent Ring Time – This defines, in seconds, how long the queue will ring a particular agent.

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.

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.

Autopause – This will place an agent or phone in a pause state in relation to the queue when they ignore or answer a call. To become an active member of the queue again unpause using the 1* feature code to for a phone, and 1*+agent# for an agent.

Maximum # of People in Queue – Defines how many callers can join the queue before it is considered full.

Periodic Announce Frequency – This defines how much time passes between playing the recording that was set for this queue.

Announce Holdtime – When set to YES, this will announce a callers status in the queue with regards to their expected hold time.

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: Callers can only join the queue with non-busy members
- STRICT: Callers can join the queue as long as there are active members, be they available or busy

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
- 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

Report Hold Time – Set to Yes if you would like the caller to hear an announcement declaring the estimated hold time before their call will be handled.

Ring in Use – Sets whether currently busy members will ring when a call enters the queue.

Member Delay before Connect – Delay, in seconds, before a member is connected to the queue.

Timeout Restart – When 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

Allow Recording – Determines if calls from this queue can be recorded.

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 – Set the destination to send a caller when the queue they are trying to join is Full

Exit Status: JOINEMPTY – Set the destination to send a caller when the queue they are trying to join is Empty

Exit Status: LEAVEEMPTY – Set the destination to send a caller when the queue they are trying to leave is Empty

Exit Status: JOINUNAVAIL – Set the destination to send a caller when the queue they are trying to join is Unavailable

Exit Status: LEAVEUNAVAIL – Set the destination to send a caller when the queue they are trying to leave is Unavailable.

Announce Filename – Define what audio file to use for the queue announcement.

**Exit Menu –** Set the destination to send a caller when they exit the queue.

Destinations / Ring Groups

Ring Groups

- System
- Providers
- Destinations
- Call Routing
- PBX Setup
- Reporting

	Name	ID	PIN
☐	Bill Smith	1234	1111
☐	Mary Jones	5678	2222
☐	John Johnson	4321	3333
☐	Tim Stevens	3219	4444

Save Changes | Add | Delete Selected

(Diagram 36)

To create an agent, select "Agent" link next to "ACD Settings" under any group. Type in name of agent, PIN, and ID#, and select "Add" (**Diagram 36**). Agents appear under the drop down menu and can be added to any group (**Diagram 37**). Agents can also have a priority assigned specific to the individual group.
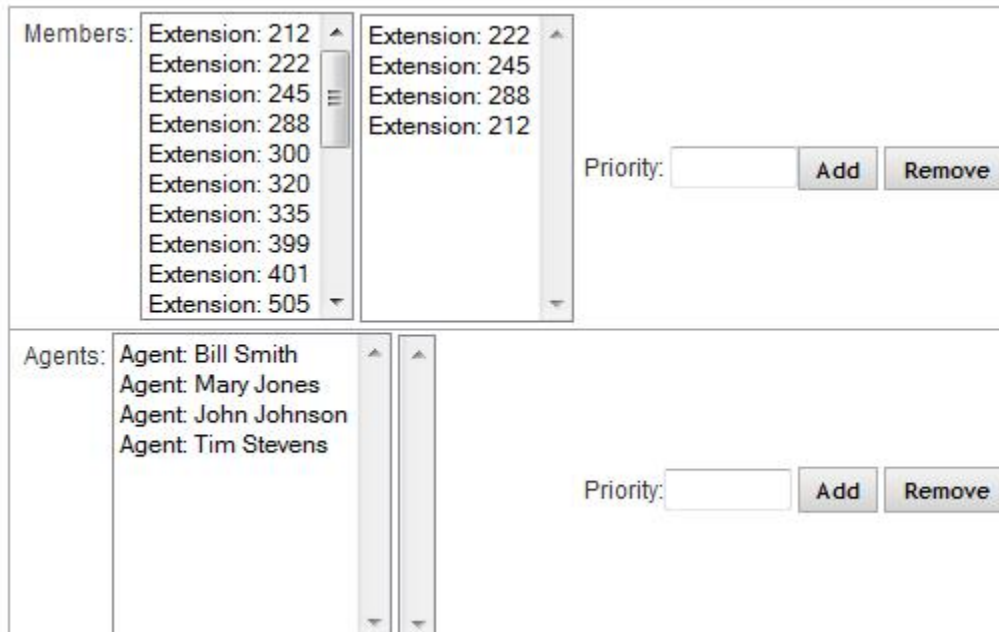
(Diagram 37)

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. A 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.

Menu Examples

A business can create a variety of different menus to direct calling traffic through a business. Some common menu examples are defined for a business with the departments and people listed below.

(Diagram 38)

Menu Example 1 – Main Menu (Auto Attendant)

This menu example is used to automatically route callers to the destination (person or group) they would like to speak to. The automated attendant helps minimize the number of calls to the receptionist and frees up time for other tasks. A Main Menu for this company might be listed as follows:

- Sales (Destination - Group 1).

- Customer Service (Destination - Group 2).

- Office Manager (Destination - Ext. 113).

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

Thank you for calling ATI Connect 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.

- Customer Service, press 2.

- Accounting, press 3.

Calls sent to the Sales department in this case might use a Round Robin ring strategy, answering calls consecutively one after the other. The Sales team may have times during the day that there are extremely high call volumes and the team might need to alleviate the problem of loading one Representative with all of the calls. In this case the company might set a Fewest Recent ring strategy, which sends a call to the Sales Representative that has received the smallest number of recent calls. If the Customer Service team is missing a team member their calls might be set to Ring All, so that the first available Representative can take a call from a customer.

Menu Example 2 – Regional Sales

This menu example allows the business to direct callers to regional sales groups once the Sales destination has been selected from the main menu. Then, through Menu 2, routes callers to Representatives by the region of the country they are calling from:

Menu 1

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

Menu 2

- New York (Destination - Ext. 123, Cathy Caldwell).
- Florida (Destination - Ext. 124, David Dawson).
- California (Destination - Ext. 125, Susan Smith).
- Arizona (Destination - Ext. 126, Robert Reed).

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

Thank you for calling ATI Connect 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.
- Customer Service, press 2.
- Accounting, press 3.

When a caller pressed "1" they would hear a **menu prompt for Menu 2**:

For sales in:

- New York, press 1.
- Florida, press 2.
- California, press 3.
- Arizona, press 4.

In Menu 1 Customer Service might use a Ring All strategy, which would allow the first available Customer Service Representative to pick up a call. In Menu 2 calls would be directed to specific Representatives. The Company can set the fail over Destination to Group 1 (Sales) and set the ring strategy for this group to Ring All. This would direct calls to any available Sales Representative.

Menu Example 3 – After Hours Menu

This menu directs traffic when the office is closed. After hours calls would be routed

through a Schedule to the After Hours Menu which might be listed as follows for this company:

- 1 – Sales (Destination - Sales General Mailbox).
- 2 – Customer Service (Destination - Customer Service General Mailbox).
- 3 – Office Manager (Ext. 113).

A **menu prompt** for this menu might read as follows:

Thank you for calling ATI Connect a leading manufacturer of cable assemblies and wiring harnesses. We are closed right now, but will respond to your call on the next business day. If you know the extension of the party you are calling you may dial it at any time. To leave a message for:

- Sales, press 1.
- Customer Service, press 2.
- All other callers press 3.

Add a New Menu

1. Click on **Destinations** and **Menus**. The Menus page will appear.
2. Click **Add Menu**. The Edit Menu page will appear.
3. Enter a **Name** for the menu.
4. Enter a **Menu Number**. This number must be three or four digits long.
5. Select a **Greeting (Prompt)** to be used for the Menu from the drop-down box. If a greeting does not exist. Record one and apply it to the Menu later in the system installation.
6. Select a **Fail over Destination** for the call. This is where the call will be directed to if the caller does not select a menu option as directed by the greeting (prompt).

Select a destination for the dialed digits from the drop-down menu. The system permits twelve menu selections "0" through "9", * and #. Destinations can be Extensions, Groups, Menus, Conferences, Voice Mail Boxes, Schedules, Branch Offices, Services, or the Follow Me feature.

Advanced Menu Settings

Advanced menu settings create rules by which the menu will operate. They also permit other dialing options if callers do not want to use the menu to reach a destination within the business.

Response Time Out

Enter the number of seconds the system will wait for a menu selection from the caller after the greeting (prompt).

Digit Time Out

Enter the number of seconds the system will wait for a digit to be pressed by a caller after the greeting (prompt).

Number of Times to Play the Greeting Before Time Out

Enter the number of times the greeting is to play before timing out. The minimum number of times it will play is "1" and the maximum number of times it will play is "10."

Fail Over Caller Hears

Setting currently disabled – reserved for future use.

Local Exchange Dialing

This allows callers to dial the extension of the person they are trying to reach, if they know it. Select "Yes" from the drop-down menu to activate this feature and No if this is not a menu option.

Edit an Existing Menu

1. Click on **Destinations** and **Menus**. The Menus page will appear.

2. Select a Menu by clicking on the **Menu Name** or ✎ **Edit**.

3. Make changes to the Menu.

4. Click **Save Changes**.

5. Click **Apply Changes** when ready to implement changes to the system.

Delete a Menu

1. Click on **Destinations** and **Menus**. The Menus page will appear.

2. Click ⊗ **Delete**. The Menu will be removed.
   The menu will be removed from the Menu Window on the page.

Meet-me Conferences

The IPitomy Meet-me Conference Bridge allows groups of callers (large and small) to dial into a conference call. To gain access to the conference, participants dial into an extension and are admitted to the conference room by entering a PIN number. A conference call can also be established without a PIN. To reach the two pre-programmed conference rooms from an extension dial:

- **Conference 1** – Dial, 901.

- **Conference 2** – Dial, 902.

The Administrator of the conference call is the person who establishes the call. This person uses an Administrative Personal Identification Number (PIN). The administrator can also control how callers enter the conference. If the "Announce on enter" checkbox is checked, callers will be prompted to record their name as they enter the conference room. The new participants name will be announced to other participants in the conference. If the "Enter Muted" checkbox is checked, conference callers will enter the conference with their phones muted. Users may press the (*1) to disable muting or enable muting on their telephones during the conference call.

PINs are not permanent, which offers a business the security of knowing that the conference extensions can be used multiple times without fear of interruption. By providing a new Administrative and General PIN number each time the room is to be used for a different purpose the business can prevent conference interruptions from parties not participating in a call.

(Diagram 39)

Setting Up a Conference

1. Click on **Destinations** and **Conferences**. The Conferences page will appear.

2. Select a conference room.

3. Enter an **Administrative PIN (Admin PIN)**. PIN numbers must be three or four characters long.

4. Enter a General PIN.

5. Check "**Announce on enter**" if caller's should announce their names to other participants in the conference.

6. Check "**Enter Muted**" if callers should enter the conference with their phones muted.

7. Click on **Save Changes**.

8. Click on **Apply Changes** when ready to implement the changes to the system.

Meet-me Conferences Features

Conference Calling offers several features that are available to participants in the

conference. All features are invoked by participants pressing the star (*) key on their telephone plus a feature code number.

*1 – Mutes and un-mutes your telephone.

*2 – Locks or un-locks the conference to additional participants. (Only available to the conference administrator).

*3 – Ejects the last participant to enter the conference call. (Only available to the conference administrator).

*4 – Decrease the Conference volume (voice volume from other participants).

*6 – Increase the Conference volume (voice volume from other participants).

*7 – Decrease User Volume (voice volume from your phone).

*9 – Increase User Volume (voice volume from your phone).

*8 – Exit Menu (terminate volume adjusting input)

Voicemail

There are two places in the system that will allow the management of a voicemail box. Voicemail boxes associated with an extension are automatically created when an extension is created and can be administered from the Advanced Settings section of the Extension. Voicemail boxes that do not have an extension associated with them are administered as a destination independently. Voicemail boxes not associated with an extension can be used as a general mailbox for groups or for employees that do not have extensions, but require a voicemail box.

 (Diagram 40)

Creating a Voicemail Box without an Associated Extension

1. Click **Destinations** and **Voicemail**. The Voicemail page will appear with any extensions that have already been created.

2. Click **Add Mailbox.** The Edit Voicemail page will appear.

3. Enter the **Mailbox Number**. Be sure to verify that the number you select is not being used for a conference (extensions 901 and 902), a group or an existing extension within the organization.

4. Enter the **Name** of the Mailbox.

5. Enter a **Password** for the Mailbox.

6. Enter an **Email Address** for the Mailbox user.

7. Check **Yes**, **No** or Not Applicable (**N/A**) to the following:

- **Attach to E-mail** – Send a voicemail message to an Email address by attaching it to an email message as an audio file (.Wav).

- **Delete After Email** – Delete the voicemail after it has been emailed to the address provided for the extension in General Settings.

- **Turn Old After Emailing** – Turn the message old after emailing.

- **Say Caller ID** – State Caller ID prior to playback of the message.

- **Allow Review** – Permits callers to review a voicemail message after it has been recorded.

- **Allow Operator** – Allows people calling this mailbox to press '0' to go to the system operator destination.

- **Play Envelope Message** – Plays caller ID and time of call prior to audio version of a message delivered through Email.

Installation Note 4
Not applicable (N/A) denotes that this mailbox should use the system wide default settings for this setting.

8. Define the number of days in which voicemail messages are to be automatically deleted from the mailbox. The minimum number of days is 1 and the maximum number of days is 365.

9. Click Save Changes.

10. Click Apply Changes when ready to implement changes to system.

Edit an Existing Voicemail Box

1. Click on **Destinations** and **Voicemail**. The Voicemail page will appear.

2. Select a Voicemail Box by clicking on the Voicemail Box **Name** or **Edit**.

3. **Make changes to the Voicemail Box**.

4. Click **Save Changes**.

5. Click **Apply Changes** when ready to implement changes to the system.

Delete a Voicemail Box

1. Click on **Destinations** and **Voicemail**. The Menus page will appear.

2. Click **Delete**. The Voicemail box will be removed.

Schedules

A schedule is a destination. A schedule defines the destination to which calls are to be routed for a specified time period. A schedule can be used to define when a business,

department, group or extension is open, closed or out to lunch. The IPitomy IP PBX automatically creates a schedule when an extension is provisioned in the system. These schedules are very flexible. They are designed to accommodate even the most complex business hour scenarios.

Multiple schedules can be created that account for different business, customer, departmental or system user needs. Schedules do not need to be applied to an entire company, they can be applied to individual destinations or routes based on the needs of a business. This works well in businesses where departments have different business hours, but are served by the same system.

(Diagram 41)

Create A Schedule

1. Click on **Destinations** and **Schedules**. The Schedules page will appear.

2. Click **Add Schedule.** The Edit Schedules page will appear.

3. Enter a **Name** for the Schedule.

4. Define **Hours of Operation** for the schedule by selecting times from the **Start** and **Stop** drop-down boxes for each day of the week.

5. Select a **destination for calls that are In Hours**. Destinations can be Extensions, Groups, Menus, Conferences, Voice Mail Boxes, Schedules, Branch Offices, Services, or the Follow Me feature.

6. Check **Apply Forward Settings**. This will apply the Forward Settings established within each destination. Note there are only forwarding settings on extensions currently, so this will not effect destinations that do not have forwarding settings.

7. Select a destination for calls that are "Outside of Hours."

8. Check Apply Forward Settings.

9. Define **Lunch Hours** for the schedule by selecting times from the **Start** and **Stop** drop-down boxes for each day of the week. Lunch hours must be within the hours of operation for the respective day.

10. Select a destination for calls during Lunch Hours.

11. Check Apply Forward Settings.

12. Click the "**Save Changes**" button to save the schedule data.

13. Creating Holiday Schedules:

   ▪ Enter a **Name** the Holiday.

   ▪ Clicking the "**add"** button to create the holiday, the Holiday name appears in the Holiday list.

   ▪ **Highlight (Select)** the new holiday in the Holiday list.

   ▪ Enter a **Start** and **End Date** for the Holiday in the Month, Day fields.

   ▪ Enter a **Start** and **End Time** for the span.

   ▪ Enter a **destination** for calls during the Holiday.

   ▪ Click the "**Set**" button to save the holiday data.

   ▪ **Repeat this process** for each Holiday observed.

Note that Holidays override the operating hours for the time frame specified. This time is all inclusive, meaning it includes ALL times between the start and the stop times.

14. Click **Save Changes**.

15. Click **Apply Changes** when ready to implement changes to the system.

Editing an Existing Schedule

1. Click on **Destinations** and **Schedules**. The Schedules page will appear.

2. Select a Schedule by clicking on the Schedule **Name** or ✏ **Edit**.

3. Make changes to the Schedule.

4. Click **Save Changes**.

5. Click **Apply Changes** when ready to implement changes to the system.

Deleting a Schedule

1. Click on **Destinations** and **Schedules**. The Schedules page will appear.

2. Click ✖ **Delete**. The Schedule will be removed.

Deleting a Holiday

1. Click on **Destinations** and **Schedules**. The Schedules page will appear.

2. Click the Schedule **Name** containing the Holiday to be deleted.

3. In the Holidays section, highlight the Holiday name you want to delete.

4. Click the "**remove**" button.

5. Click the "**Save Changes**" button.

Branch Offices

Branch Offices is a powerful tool that is used to link up multiple PBX's located in external offices. By utilizing Branch Offices you are able to create direct extension dialing from one office location to another with a simple dialing prefix.

(Diagram 42)

Configuring Router

Please see Appendix on Router Configuration

Configuring Office 1

1. Click on **Destinations** and **Branch Offices**. The Branch Offices page will appear.

2. Click on **Add Office**.

3. Give a unique **NAME** for the connection

4. Type dynamic for **HOST**.

5. Create a unique **DIALING PREFIX** for the extensions connected to the Office 1 PBX.

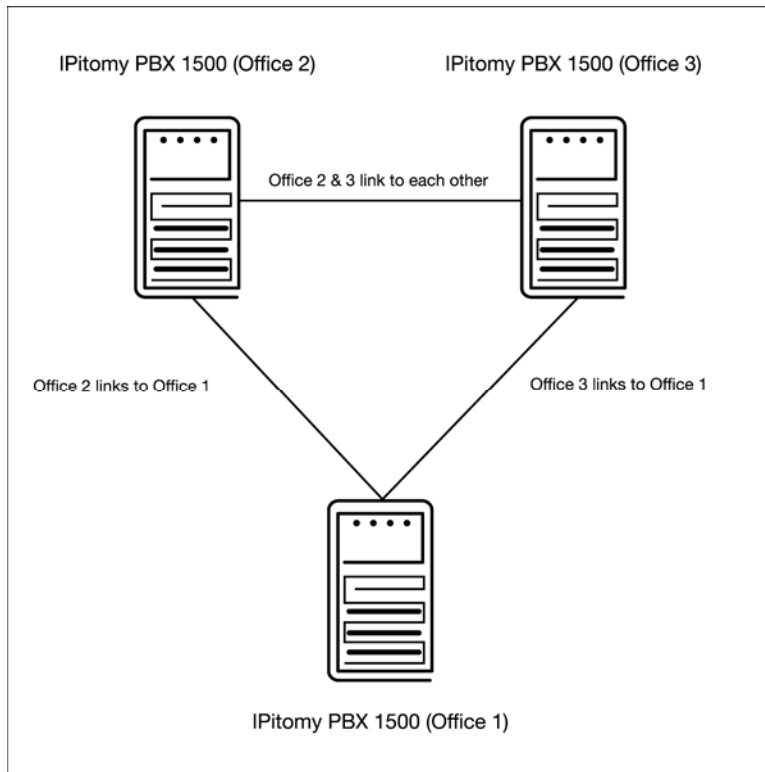| Edit Branch Office		
Name:	Sarasota	(one word only)
Host:	68.53.255.72	(dynamic or ip address)
Dial Prefix:	*12	
Password:	password	
Register:	No	
Enable Trunking:	Yes	
Qualify:	30000	

(Diagram 42)

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

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

8. Select Yes to **Enable Trunking** to allow Office 2 to use Office 1's trunks for outbound routing.

9. Leave **QUALIFY** at 30000.

10. Click **Save Changes**

11. Click **Apply Changes**

Configuring Office 2

1. Click on **Destinations** and **Branch Offices**. The Branch Offices page will appear.

2. Click on **Add Office**.

3. **NAME** should match that given to the Office 1 PBX. The name should only contain alpha-numeric characters and no spaces.

4. **HOST** needs to be the external IP address or domain corresponding to the IP of the main office PBX.

5. Create a unique dialing prefix for the extensions connected to the Office 2 PBX.

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

7. Select Yes for **REGISTER**.

8. Select Yes to **Enable Trunking** to allow Office 1 to use Office 2's trunks for outbound routing.

9. Leave **QUALIFY** at 30000.

10. Click **Save Changes**

11. Click **Apply Changes**

To setup up multiple PBX's just follow the above pattern linking each PBX to each other and use a unique dialing prefix for each one.

(Diagram 43)

To place a call from one Branch Office to another just 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.

Branch Extensions

Branch Extensions can also be added to a branch office. These 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.

Configuring Office 2 with branch extensions

1. Click on **Destinations** and **Branch Offices**. The Branch Offices page will appear.

2. Click on the branch office connection for Office 1

3. Assuming Office 1 has extensions 100-110. Enter 100-110 in the field above the add button. You can also enter individual extension numbers. These numbers are checked against existing system extensions to insure conflicts are not created.

4. Click the **Add** button

5. **Apply Changes**

6. Verify dialing the new extensions from phones registered to Office 2.

Call Routing

Routing tells the system what destination to send calls to within the business. It also tells the system what provider services to use when making outbound calls.
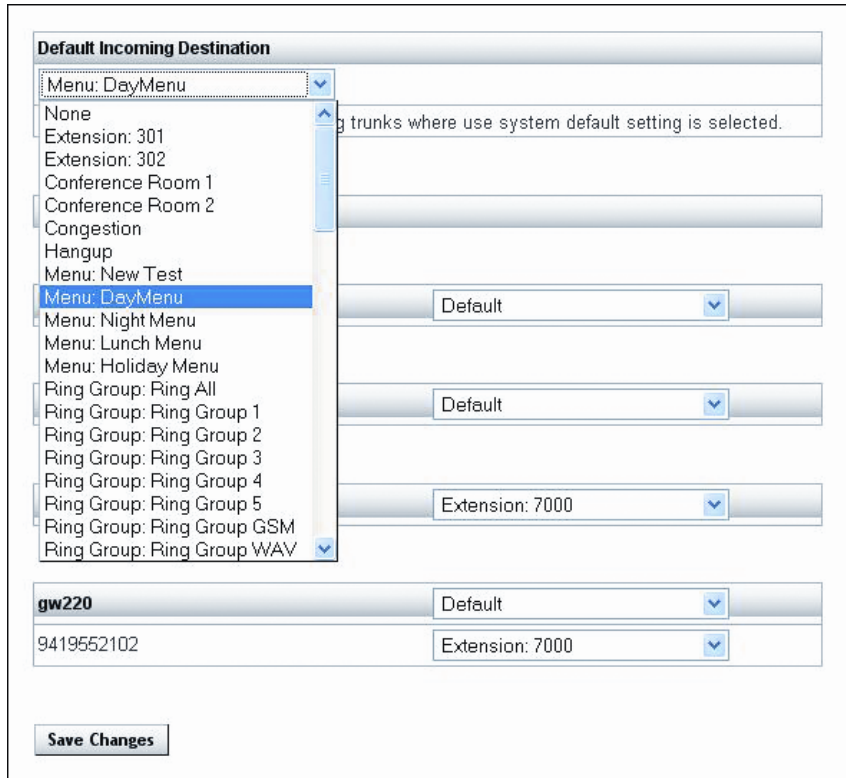
Incoming Routing

The Incoming Routing administration specifies system routing, the service provider and DID level for incoming calls. The different levels may overlap at times so it is important to understand their order of precedence:

- Phone Number
- Provider (how that phone number arrives at the system)
- System Default

The system default destination is the weakest precedent. The provider levels and the specific numbers being routed follow in precedent. When the stronger precedent is set to default it will follow the weaker precedents. If the route is set to something other than default it will override the weaker precedents.

For example, if a user selects default for a phone numbers incoming destination and the provider of that phone number is a SIP provider, when that number is called from the PSTN it will route to the default destination of the provider. If the provider destination is set to default, the system will route calls to the system's default incoming destination. If the provider had a destination other than the system default, the call would route to that destination rather than the system default. If the particular phone number is set to a destination other than default, the system will route a call to that destination rather than the provider or system defaults.

The IPitomy IP PBX allows for flexible routing arrangements. Calls can initially be routed to a Menu, Schedule, Extension or Group. Establishing Incoming Routing sets the default destination for all calls.

(Diagram 44)

Setting the Default Incoming Destination

To set the Default Incoming Destination:

1. Click on **Call Routing** and **Incoming**. The Incoming Call Routing page will appear.

2. Select a **Default Incoming Destination** from the drop-down box.

3. Click the **Save Changes** button.

Defining the Service Provider Destination

A service provider destination can also be defined. The service provider destination acts as the default incoming destination for a particular provider. To set the service provider destination:

1. Select an **Incoming Destination** from the drop-down box.

2. Click on **Save Changes** button.

3. Click on **Apply Changes** when ready to implement these changes to the system.

Outgoing Routing

Outgoing routing tells the system what service providers certain types of calls should use. The IPitomy IP PBX comes with common outgoing call routes already provisioned in the system:

- 7 digit dialing

- 10 digit dialing

- 11 digit (1+ dialing)

- International
- Emergency Calls
- Custom Dialing

These routes should not be deleted from the system.

Adding A New Call Subroute

A sub route is a specific pattern within a route type that will be routed differently from other routes. For instance, if you add 858 as a route to 10 digit or 11 digit dialing, you can define the trunks that specifically route to that area code. If you do not add any trunks to the newly created sub route, that dialing pattern is effectively blocked and users of the phone system will hear "all outgoing lines are unavailable" when attempting to dial the 858 area code.

1. Click **Call Routing** and **Outgoing**. The Outgoing Call Routing page will appear.
2. Click the **Add Route** button. The Edit New Outgoing Route page will appear.
3. Enter a Route Name.
4. **Select the Route** Type the call is to take. This identifies the call as:
   a) Local (7-digit)
   b) 10 Digit
   c) Long Distance (1+10 digit) dialing
   d) International
5. Assign a routing Number.

Editing An Existing Route

To change a call route to a different service provider or to change the order in which calls are routed over providers:

1. Click on **Call Routing** and **Outgoing Routing**. The Outgoing Routing page will appear.
2. Select a **Route** in the Route Window to be changed.
3. Click **Edit**. The Edit Route page will appear.
4. Change the **Route Name**, Type and Number as needed.
5. Change the Trunk associated with the Route.

Changing the Order Service Providers are Selected

From the service providers available:

1. Click on the **Name** of a provider.
2. Click the **Up** or **Down** button to position the provider. Calls will be routed in the order the service providers appear in the list.
3. **Repeat this process** until the providers are in the order calls should be routed over the available service providers.

Adding a Service Provider

1. **Select a service provider** from the drop-down box to the left of the

Trunks list.

2. Click **Add**. The service provider will appear in the list.

3. Click on the **Name** of the provider.

4. Click the **Up** or **Down** button to position the service provider in the list in the order calls are to be routed over this resource.

Deleting a Service Provider

1. Click on a **Service Provider** in the list.

2. Click **Delete**.

3. Click on **Save Changes**.

4. Click on **Apply Changes** when ready to implement these changes to the system.

Installation Note 5

Default Settings For Existing Routes

The Outgoing Call Routes already in the IPitomy IP PBX have the following default

settings Route Type and route Number. These default settings cannot be changed.

The Name of the route, service provider associated with the route and the order in

which service providers are selected for the route can be changed.

PBX Setup

PBX setup is used by a System Administrator to manage system-specific and system-wide capabilities of the IPitomy1500.

General

General setup manages the Administrative Settings for the system: User Name, Password and Time Zone.

Edit General Settings

1. Click on **PBX Setup** and **General**. The General System Setup page will appear.

2. Enter an **Administrative User Name**.

3. Enter an **Administrative Password**.

4. Re-enter the **Password**.

5. Select a **Time Zone** from the drop-down box provided.

6. Select a **System Operator** Destination. This is the destination that people will go to when pressing '0' while in a voicemail box, if Allow Operator is enabled on a voicemail box.

7. Select **Mailbox Exit** destination – this is the destination calls will be routed to when leaving a voicemail box.

8. Select **Directory Type** – this is the type of directory to use for the Directory destination.

9. Click on **Save Changes**.

10. Click on **Apply Changes** when ready to implement changes to the system.

Database

The IPitomy IP PBX is like a computer, in that; information can be stored in the system or backed up in case it goes down. For additional protection, the system also allows a copy of the system setup to be stored in an external source like a computer or CD. The Database section of online administration manages the process of downloading and storing copies of the system setup to the system itself or an external source.

Backing Up a Copy of the System's Setup

The system can be backed up to an internal database or to an external source like a computer or CD.

1. Click on **PBX Setup** and **Database**. The Database Setup page will appear.

2. To back up a copy of the **System's Setup to the internal database**, click the **Backup** button. The backup version of the file will appear in the Databases Window with the Date, Time and Version of the backup.

3. To back up a copy of the **System's Setup to an external source** like a computer or CD, click on ⊕ **Download** button. **Give the file a name and store it in a place that is easily accessible**.

Restoring a Copy of System's Setup

The system can be restored from a copy of the System's Setup in the internal database or from a copy on an external source.

1. Click on **PBX Setup** and **Database**. The Database Setup page will appear.

2. To restore a copy of the **System's Setup using the internal database**, **select the Backup version** to be restored from the Database Window and click the ⊕ **Restore** button. This will restore the version of the System's Setup selected.

3. To restore a copy of the **System's Setup using an external source**, **Browse for the file** in the external source (click Open from the operating system for the file to appear in the Browse Window) and click **Send File**. The Backup file will appear in the Database Window. Click **Restore**.

4. Click on **Apply Changes** when ready to implement these changes to the system.

Deleting a Backup Version from the System

1. Click on **PBX Setup** and **Database**. The Database Setup page will appear.

2. To delete a copy of the **System's Setup from the internal database**,

select the Backup version to be deleted from the Database Window and click the ⊗ **Delete** button. This will delete the version of the System's Setup selected.

3. To delete a copy of the **System's Setup from an external source**, use the **delete function of the operating system** or throw away the CD that contains the Backup file.

Voicemail

System-wide voicemail defaults define the rules by which voicemail boxes operate. In the IPitomy IP PBX General, Menu and Email settings can be set for voicemail boxes system-wide.

General

General settings relate to the capacity (storage time and message length) of a voicemail box. To define the General Settings:

1. Click on **PBX Setup** and **Voicemail**. The Voicemail Settings page will appear. Within this page are the General Settings.

2. Enter the following General Settings:

   - **Maximum Number of Messages** – This is the greatest number of voicemail messages a box can hold. The system limit is 300.

   - **Maximum Message Length** – This is the longest duration of time allowable for each message. The system limit is 3 minutes.

   - **Minimum Message Length** – This is the shortest duration of time allowable for each message. The system minimum is 2 seconds.

   - **Maximum Greeting Length** – This is the longest duration of time allowable for a voicemail greeting. The system maximum is 30 seconds.

   - **Maximum Seconds of Silence** – This is the longest duration of time a caller can be silent before the system considers the call complete. Setting this to zero will end the voicemail recording when the caller terminates the call or connection. The system maximum is 20 seconds.

   - **Silence Threshold** – This is the amount of time that the system will wait for a response from a caller leaving a message.

3. Click **Save Changes**.

4. Click on **Apply Changes** when ready to implement these changes to the system.

Voicemail Settings

1. Select and enter Voicemail Menu settings:

   - **Play the Envelope Message** – This is the Time and Date of the call.

- **Say Caller ID** – This plays the caller's phone number when available.

- **Skip MS on Playback** – This is the interval (in Milliseconds) that voicemail will skip forward or backward on playback. This number must be between 1 and 4.

- **Maximum Failed Login Attempts** – This is the maximum number of times a user may try to login to a voicemail box before the system disconnects the call. This number must be between 3 and 5.

- **On Delete, Play Next Message** – This tells the system to play the next message in queue when a message has been deleted.

2. Click **Save Changes**.

3. Click on **Apply Changes** when ready to implement these changes to the system.

Email Settings

1. Select **Yes** or **No** to allowing voicemail messages to be **attached as an audio file (.Wav)** to the email address associated with the voicemail box.

2. Set the from address for the email address you would like the attached voicemail to be sent from.

3. Voicemail Server can either be Loc (local) or Ext (External).

   Local – internal PBX mail server

   External – an external mail server such as (smtp.ipitomy.com)

4. Authentication Required can either be Yes or No

   Yes – authentication is required for this mail server

   No – authentication is not required for this mail server

5. User Name – user name required for authentication

6. Password – password for authentication that is associated with the above user name.

7. Click on **Save Changes**.

8. Click on **Apply Changes** when ready to implement the changes to the system.

   * For external mail servers please check with your ISP or hosting provider for their required settings.

   * There is no guarantee that an external mail server will work, some ISP's or hosts may have mail filters and could block a message.

Advanced Settings

- **Allow Review** – Allows users to review a message after it has been played.

- **Allow Operator** – Allows people calling this mailbox to press '0' to go to the system operator destination.

Voicemail Archive

Download

    Creates a backup of all user Voicemail messages, message envelopes and greeting files. The backup file is downloaded to your computer.

Erase

    Deletes all user Voicemail data on the PBX, including personal greetings.

Upload

    Restores a previously made backup.

    Tip: Voicemail recordings may contain sensitive information. You should only store Voicemail archives in a secure location.

Session Initiation Protocol (SIP) Settings

These settings tell the system the IP address of calls coming into the system and the local network. In addition, they define to which communication traffic the system is to listen. To provision SIP settings enter the:

- **External IP** – This is the IP address of the ISP and can be obtained from the network router. The external IP address can also be obtained online at whatismyip.com. Type the address in a Web browser click Enter and the external IP address will appear on a page.

- **Local Network** – This is the default IP address of the router with the last digit replaced by a zero. In most cases the default router address will be 192.168.1.1. By replacing the last digit (1) with a zero this indicates that the local network (traffic addressed to 192.168.1) includes any variation of the fourth number in the IP address. This means that the local network includes traffic addressed to: 192.168.1.1, 192.168.1.2, 192.168.1.3 etc.

- **Subnet Mask** – Leave the default setting for the Subnet Mask as is (255.255.255.0). Provided by the router, this mask tells the network what communication traffic to listen to. For example, the setting 255.255.255.0 tells the system to listen to communication traffic sent to "192.168.1" for any variation of the fourth number (designate by the zero at the end).

> **Installation Note 6**
>
> **For the Local Network and Subnet Mask to work correctly the fourth digit in both numbers must be a "0."** The zero in the Subnet Mask indicates that the network is to listen to traffic addressed to any IP address within the Local Network. The Zero in the Local Network indicates that the Local Network can include an IP address with any variation of the fourth number.

Advanced Settings

    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 settings.

Prompts

A prompt or greeting can welcome a caller to the business, direct them to a destination, provide instructions or deliver information. The IPitomy IP PBX makes managing prompts easy.

Upload Voice Prompt

1. Click **PBX Setup** and **Prompts**. The Edit Prompts page will appear.

2. Select Browse and locate the prompt file to be uploaded.

3. **Open the file using the operating system**. The file will appear in the File Name Window.

4. Enter a **Name** for the File in the Description field.

5. Click on **Upload File**. The file will appear in the Prompt Files on Server Window. It will display the File Name, Description and Size.

6. Click on **Save Changes**.

7. Click on **Apply Changes** when ready to implement these changes to the system.

Record New Voice Prompt

1. Click **PBX Setup** and **Prompts**. The Edit Prompts page will appear.

2. Define a **Prompt Name** and **Description**.

3. **Assign an Extension** to the prompt.

4. Click **Record**. The system will dial the extension assigned to the prompt. A message screen will appear on the computer indicating that the system is trying to reach the extension.

5. **Answer the call from the system and record the prompt**.

6. Click the **Continue** button on the message screen. The new prompt will appear in the Prompt Files on Server Window. It will display the File Name, Description and Size.

7. Click on **Save Changes**.

8. Click on **Apply Changes** when ready to implement these changes to the system.

(Diagram 45)

Download a Prompt

1. Click **PBX Setup** and Prompts. The Edit Prompts page will appear.

2. **Select a prompt** from the Prompt Files on Server Window.

3. Click ⊕ **Download**. The prompt will download.

Delete a Prompt

1. Click **PBX Setup** and **Prompts**. The Edit Prompts page will appear.

2. **Select a prompt** from the Prompt Files on Server Window.

3. Click ⊗ **Delete**. The prompt will be deleted from the list.

4. Click **Save Changes**.

5. Click **Apply Changes** when ready to implement these changes to the system.

Music on Hold

In a busy business it is sometimes necessary to place callers on hold. Playing music while a caller waits can make this time more pleasant. Music files must be in .MP3 format.
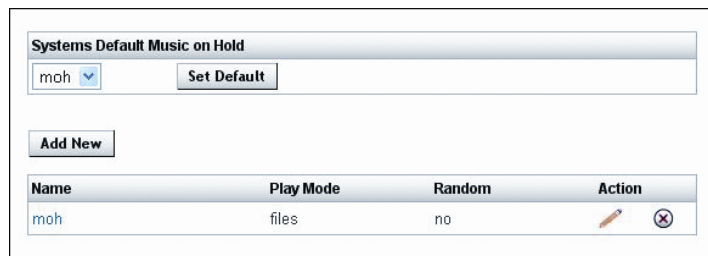
(Diagram 46)

Add New Music Files

1. Click **PBX Setup** and **Music on Hold**. The Music on Hold page will appear.

2. Click **Add New**. The Edit Music on Hold Page will appear.

3. **Name the music play list.**

4. Set **Random** to **Yes** or **No**. If the music file is to play randomly, select **Yes**. If it is to play sequentially with files that already exist, then select **No**.

5. Click **Save Changes.**

6. Select **Browse and locate the music file to be uploaded**.

7. **Open the file** using the operating system. The file will appear in the Load This Music File Window.

8. Click on **Upload File**. The file will appear under the play list of which it was uploaded too.

 (Diagram 47)

Edit Existing Music Files

1. Click **PBX Setup** and **Music on Hold**. The Music on Hold page will appear.

2. **Select a music file** from the Music on Hold Main page.

3. Select a play list by clicking on the play list **Name** or ✏ **Edit**.

4. Change the Name, Random Setting, Load a New File, or Delete the existing files.

5. Click **Save Changes**.

6. Click **Apply Changes**.

Delete a Play list

1. Click **PBX Setup** and **Music on Hold**. The Music on Hold page will appear.

2. **Select a Music on Hold** play list.

3. Click ⊗ **Delete**. The play list will be deleted from the list.

* Note if the individual music files aren't deleted from inside the play list, the music files will be generated inside the next play list created.

Setting Music on Hold

1. Click **PBX Setup** and **Music on Hold**. The Music on Hold page will appear.

2. Select your default Music on Hold play list from the dropdown menu.

3. Click **Set System Default**.

4. Click **Apply Changes**.

Feature Codes

Found in the PBX Setup section of online administration, the IPitomy IP PBX provides a set of system feature codes. These codes allow system users to manually manage calls from an extension. To access these feature codes. Click **PBX Setup** and **Feature Codes**. The **Feature Code Chart** will appear. Note that the feature descriptions in online administration can be found by sliding a mouse over the "?" next to the feature code.

Feature	Code
Personal Voicemail:	923
Voicemail Main:	924
Transfer to Voicemail:	Transfer + *+(ext #)
Directory:	926
Blind Transfer:	# #
Attended Transfer:	# *
One Touch Record:	* #
Pickup Extension:	99
Directed Pickup Extension:	99 + ext
Ring Group Page:	** (ext #)
Pause Phone Queues:	0*
Unpause Phone Queues:	1*
Pause Agent Queues:	0* + Agent #
Unpause Agent Queues:	1* + Agent #
Agent Callback Login:	2*
Agent Login:	3*

(Diagram 48)

Services

The Services online administration page is a utility that allows the uploading of system software, configurations and databases. The **Services page is designed to be used with the support of an IPitomy Customer Service Representative. The default settings in this page are not to be changed during a system implementation.**

Reports

The IPitomy IP PBX offers System Administrators a set of reporting tools to help manage calling traffic. Reporting tools also includes a diagnostic report used by IPitomy to evaluate the system's performance.

CDR Reports

A CDR Report can be used by the System Administrator to track calling traffic. The CDR Report for the IPitomy IP PBX tracks:

- **Date/Time** – The date and time of the call.

- **From** – The telephone number where the call is from.

- **Destination** – The extension, voicemail box or group to which the call is going.

- **Trunk** – The phone number the call came in on.

- **Duration** – The length of the call in hours, minutes and seconds.

- **Status** – A description of what happened to the call once it reached its destination (e.g., Not Answered, Disconnected or Successful).

To access the CDR Reports click Reporting and Reports. The CDR Report will appear:

(Diagram 49)

Diagnostics

The System Diagnostics page in the system's online administration is a utility used by IPitomy Customer Service Representatives for system diagnostics. **The System Diagnostics page is meant only to be used with the support of an IPitomy Customer Service Representative.**

Monitoring

The Monitoring page is a great place to quickly check the status of extensions on the system. This page lists the Extensions, their Status, and which providers call traffic is utilizing. To access the Monitoring Report click Reporting and Monitoring. The Monitoring Report will appear:

```
SIP:

Name/username            Host              Dyn Nat ACL Port    Status
4008/4008                (Unspecified)     D   N        0       UNKNOWN
4007/4007                (Unspecified)     D   N        0       UNKNOWN
4000/4000                210.66.55.59      D   N        5060    OK (400 ms)
2222/2222                192.168.2.193     D   N    A   5060    OK (19 ms)
2225/2225                192.168.2.100     D   N    A   5060    OK (175 ms)
VinTalk                  209.203.104.37        N        5060    OK (189 ms)
teliax/nbranica          70.42.223.22          N        5060    Unmonitored
4877503/19414877503      69.30.45.229          N        5060    OK (459 ms)
tollfree_support/18663054 69.30.45.229         N        5060    OK (337 ms)
9418943980/Ipitomy       208.51.24.111         N        5060    UNREACHABLE
ipex/19414877531         69.30.45.229          N        5060    Unmonitored
IPExhangeINT/19414877502 69.30.45.229          N        5060    OK (132 ms)
95 sip peers [Monitored: 31 online, 62 offline Unmonitored: 2 online, 0 offline]

Branches:

Name/Username     Host               Mask              Port        Status
Boston/Boston     24.62.32.63    (S) 255.255.255.255  4569 (T)    Unmonitored
DemoRoom/DemoRo   192.168.2.40   (S) 255.255.255.255  4569 (T)    Unmonitored
IPitomyIP1/IPit   74.2.44.178    (S) 255.255.255.255  4569 (T)    Unmonitored
V1Conf/V1Conf     192.168.2.136  (S) 255.255.255.255  4569 (T)    Unmonitored

Channels:

Channel             Location            State   Application(Data)
SIP/2222-0a07ad20   (None)              Up      Bridged Call(IAX2/Siesta-4155)
IAX2/Siesta-4155    s@macro-sipexten:8  Up      Dial(SIP/2222|32|twM(setfeat^c
2 active channels
1 active call
```

(Diagram 50)

Recordings

Recordings found:

1. July 27, 2007 at 1:04 pm by **s**, while connected to **507** - Listen - Download - Delete
2. July 27, 2007 at 1:00 pm by **s**, while connected to **507** - Listen - Download - Delete

**Record a conversation by pressing *#
on your phone. Then check back here.**

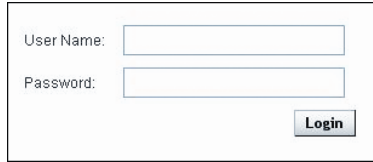Check again

(Diagram 51)

The Recordings section allows you to:

> o   Listen to recordings.
>
> o   Download recordings.
>
> o   Delete recordings.

This page will display all recorded calls on the PBX. Only extensions with allow call recording selected are able to use this feature.

Smart Personal Console

The IPitomy IP PBX offers users a set of tools to manage Call Traffic, Phone Settings and Voicemail. Collectively, these tools are known as your Smart Personal Console.

Login Screen
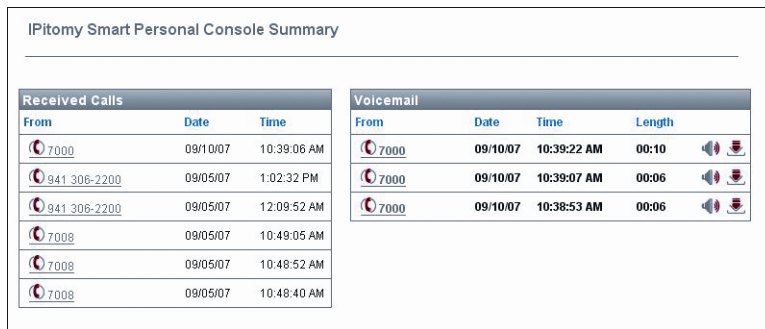
User Name:	
Password:	
	Login

(Diagram 52)

Before you can use the Smart Personal Console you must log into the system. Direct your web browser to the IP PBX network address followed by '/ippbx/spc'.

Example: 192.168.1.249/ippbx/spc

- If you are unsure what the address is, ask your IP PBX System Administrator.

- Enter the 'User Name' and Password given to you by your System Administrator.

- Upon successful logon you will be presented with the Smart Personal Console summary page, called Home.

Home

IPitomy Smart Personal Console Summary

Received Calls				Voicemail					
From	Date	Time		From	Date	Time	Length		
7000	09/10/07	10:39:06 AM		7000	09/10/07	10:39:22 AM	00:10		
941 306-2200	09/05/07	1:02:32 PM		7000	09/10/07	10:39:07 AM	00:06		
941 306-2200	09/05/07	12:09:52 AM		7000	09/10/07	10:38:53 AM	00:06		
7008	09/05/07	10:49:05 AM							
7008	09/05/07	10:48:52 AM							
7008	09/05/07	10:48:40 AM							

(Diagram 53)

This is your Smart Personal Console starting point. It displays a short list of recent calls received and new voicemail messages.

On this page you may:

- Click on a phone icon to initiate a call between your extension and the indicated caller.

- Listen to a recent Voicemail Message. Click on a speaker icon to listen.

- Save a Voicemail Message to your Computer. Click on the download icon to save a message.

 My Account

(Diagram 54)

Voicemail Settings

- o **Attach to Email** – Send a voicemail message to an email address by attaching it to an email message as an audio file (.Wav).

- o **Delete After Emailing** – Delete the voicemail after it has been emailed to the email address provided for the extension in General Settings.

- o **Turn Old After Emailing** – Turns all voicemail in the voicemail box old so the message waiting light will not be turned on.

- o **Say Caller ID** – State Caller ID prior to playback of the message.

- o **Allow Review** – Allow callers to review a message after it has been played.

- o **Allow Operator** – Allow pressing "0" during the voicemail greeting to reach the system-wide operator.

- o **Play Envelope Message** – Play caller ID and time of call prior to audio version of a message delivered through email.

- o **Delete Email After** – Define the number of days in which voicemail messages are to be automatically deleted from a mailbox.

Forwarding

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:

- o **Unconditional** – Always route calls to a specific destination.

- o **Busy** – Route calls to a specific destination when the extension is in use or do not disturb is selected.

- o **No Answer** – Route calls to a specific destination when a call is not answered.

- o **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

- o **Pick the setting to be provisioned** – Unconditional, Busy, No Answer or Unavailable.

- o Select **Enabled** or **Disabled**. Disabled turns the forward setting off. Enabled turns the forward setting on.

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.

- o Dial ***90** to disable forwarding.

- o Dial ***91** to enable forwarding.

- o Dial ***92** to set the forwarding 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 Menu. 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.

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.

Phone Settings

- Review or change phone button functions (Aastra models only).
- Voicemail Settings.
- Forwarding. This feature will only be available if it has been enabled by your system administrator.

Personal Schedule (Can be allowed or denied from Advanced Extension page)

(Diagram 55)

Follow Me (Can be allowed or denied from Advanced Extension page)

(Diagram 56)

Follow Me

(Diagram 57)

Follow me is designed to allow users to create a list of numbers that if they are not available at office extension, the caller will be placed on hold and the list will be tried in a priority weighted list in order to complete the call. This allows a new element of flexibility and freedom to users of IPitomy phone systems.

To access the Follow Me feature users must log in to their Smart Personal Console (SPC). This is accessed from the main log in page of the IP PBX. See **Diagram 54.** Administrators may access this from the advanced features for the Extension.

(Diagram 58)

You would choose the User Login and enter your credentials. The default for a user credential is the Extension Number as User Name, and also as the Password as well. Bear in mind users can change this from default and they must keep track of the credential if they change it. Once logged in access the My Account section as seen in **Diagram 59.**

(Diagram 59)

From this screen select the Follow Me Manage Numbers and Settings button this will give you the Management Screen for Follow Me Settings. See **Diagram 60.**

(Diagram 60)

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 additionally you can weight the priority from 1-20 (20 being lowest 1 being highest) as 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.

Call Log

IPitomy Call Log

Call History

	Status	From	To	Date	Time	Length
☐	answered	7000	mbox_exit	09/10/07	10:39:21 AM	00:00:13
☐	answered	7000	7000	09/10/07	10:39:06 AM	00:00:11
☐	answered	7000	mbox_exit	09/10/07	10:38:51 AM	00:00:11
☐	answered	7000	306-2200	09/05/07	3:40:27 PM	00:00:10
☐	answered	7000	941 306-2200	09/05/07	3:40:19 PM	00:00:02
☐	answered	7000	7008	09/05/07	3:38:05 PM	00:00:00
☐	answered	7000	306-2200	09/05/07	3:37:31 PM	00:00:09
☐	answered	7000	306-2200	09/05/07	3:12:04 PM	00:00:07
☑	answered	7000	306-2200	09/05/07	3:11:34 PM	00:00:07
☐	answered	7000	306-2200	09/05/07	3:11:16 PM	00:00:08
☐	answered	7000	306-2200	09/05/07	3:10:51 PM	00:00:07
☐	answered	7000	306-2200	09/05/07	3:10:31 PM	00:00:10
☐	answered	7000	306-2200	09/05/07	1:07:23 PM	00:00:03
☐	answered	7000	7008	09/05/07	1:07:06 PM	00:00:00
☐	answered	941 306-2200	7000	09/05/07	1:02:32 PM	00:00:09
☐	answered	7000	7008	09/05/07	12:10:48 AM	00:00:00
☐	answered	941 306-2200	7000	09/05/07	12:09:52 AM	00:00:07
☐	answered	7000	7008	09/05/07	12:00:06 AM	00:00:00
☐	no answer	7008	7000	09/05/07	10:49:05 AM	00:00:00
☐	no answer	7008	7000	09/05/07	10:48:52 AM	00:00:00

(Diagram 61)

Displays a history of calls made to and from your extension.

- Calls are listed from most recent to oldest.

- You can initiate calls here the same as on the Smart Personal Console Home Page. Click on the phone icon to initiate a call.

- When there are too many records to display on the page at once, your call log is split into multiple pages. To the right and above your call log you will see Next and Previous links.

    Example: 200 Messages | Previous 180-200 Next

    In the example, you are viewing messages 180 through 200 out of a total of 200 messages. Remember that calls are listed from most recent to oldest, so the first

call on the page is number 200, followed by 199, 198, etc.

- Clicking on 'Previous' or 'Next' will take you to the indicated page.

 Voicemail

(Diagram 62)

Your voicemail box can be entirely maintained via the Smart Personal Console. If you need to setup your Voicemail Greetings, go to the My Greetings section.

The voicemail box is divided into folders: New, Family, Friends, Work, and Old. New messages appear in the New Folder. You may move messages from one folder to another, delete messages, listen to messages and download messages.

Messages are listed in order of most recent to oldest. When using a phone to call the Voicemail system, messages are listed from oldest to most recent.

Icon Legend

- Click on a phone icon to initiate a call between your extension and the indicated caller.

- Listen to a recent Voicemail Message. Click on a speaker icon to listen.

- A sound file that has already been listened to.

- Save a Voicemail Message to your Computer. Click on the download icon to save a message.

- Upload a sound file to use as a greeting.

- ▪ 🗑 Empty a folders contents

Navigating Folders

📄 Click on the folder you wish to view. A list of items in the folder is then displayed.

* Download Folder button will not be displayed if the current folder is empty.

Navigating Pages

- ▪ Calls are listed from most recent to oldest.

- ▪ You can initiate calls here the same as on the Smart Personal Console Home Page. Click on the phone icon to initiate a call.

- ▪ When there are too many records to display on the page at once, your call log is split into multiple pages. To the right and above your call log you will see Next and Previous links.

> Example:
>
> 200 Messages | Previous 180-200 Next
>
> In the example, you are viewing messages 180 through 200 out of a total of 200 messages. Remember that calls are listed from most recent to oldest, so the first call on the page is number 200, followed by 199, 198, etc.

- ▪ Clicking on 'Previous' or 'Next' will take you to the indicated page.

Usually "most recent" is in relation to the date but if you move messages between folders "most recent" means the message most recently moved into the current folder. Regardless of the order of the messages you can sort them.

Moving Messages

1. Select messages by checking the checkbox to the left of each message.
2. Use the Folder drop down list to choose a location to move the messages to.
3. Click **Move Selected To**.

Deleting Messages

Deleting all of the messages in a folder

🗑 Click the trashcan icon beside the folder you wish to empty

Deleting small numbers of messages

1. Select messages by checking the checkbox to the left of each message.
2. Click **Delete Selected**.

   Warning: Deleted messages cannot be recovered.

Sorting
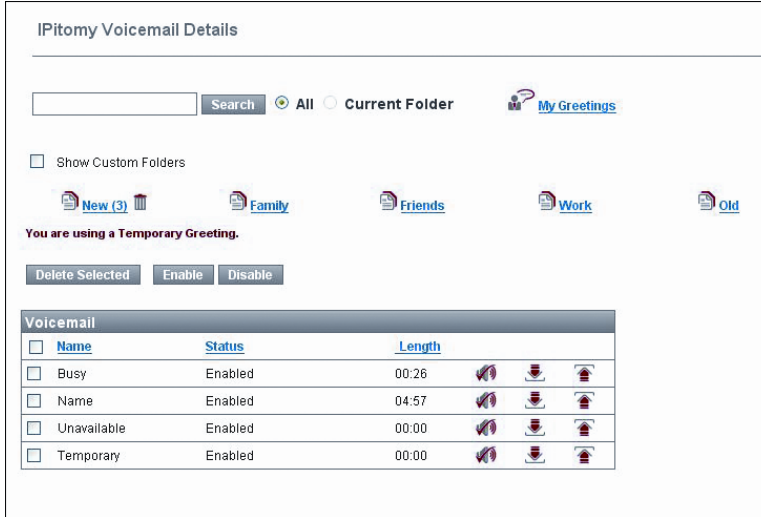
You can sort by Name, Number, Time and message Length by clicking on the appropriate column heading.

Searching

You can search for names and phone numbers. The search can be of all folders or the

current folder. Simply type the text you wish to search into the search box, and then click 'Search'. Select 'all' or 'current folder' to change the scope of the search.

My Greetings

(Diagram 63)

Click on the My Greetings link. This folder contains greetings sound files specific to this voice mail box (if you or your administrator have installed any). A sound file has a status of Enabled, Disabled or System Default.

How to Setup My Greetings

1. Consult Which Greetings to Use (section A.) and decide which greeting(s) you want. Make a list of greetings you want. For each greeting:

2. If the status is Enabled, click on the play icon. If the message sounds exactly how you want, you do not need to make any more changes to this item. Otherwise go to step 5.

3. If the status is Disabled, click on the play icon. If the message sounds exactly how you want, follow the Enable part of How to Modify Personal Greetings (section C.) Otherwise go to step 5.

4. If the status is System Default go to step 5.

5. Follow How To Upload a Personal Greeting (section B.)

Tip: If a greeting has both an Enabled and Disabled version, the Enabled version will be used. If you disable the Enabled version it will replace the Disabled version (when applicable).

How to Disable all Greetings

1. Click the checkboxes next to enabled items.

2. Click the "Disable" button.

A. Which Greetings to Use?

▪ **Temporary** – This overrides all other Voicemail greetings. If you just want a single

greeting to be used regardless of your extension's Do Not Disturb setting, use this.

- **Busy** – This is only played when you have set the Do Not Disturb indicator on your extension.

- **Unavailable** – This is only played when you don't answer your phone and you have not set the Do Not Disturb indicator on your extension.

- **Name** – This is used by Voicemail when you have no other greetings enabled. It is also used by Directory Service to speak your name.

Tip: You can install more than one greeting. You can also have multiple greetings disabled and only enable the greeting(s) you wish to be used by the system now.

B. How To Upload a Personal Greeting

1. From the Voicemail Page, Click on My Greetings.

2. You will need a sound file on your computer in one of these formats: CDR, CVS, HCOM, GSM, MP3, and WAV. If you don't have a file, use your favorite sound recording program to create a file.

3. Click on the upload icon next to the Greeting you wish to create or change. A popup window will appear.

4. Click the Browse button. A file browsing window will appear. Search for the sound file. When you find it, click on it. Then (if needed) click the "Open" button. You are returned to the first popup window.

5. Click the Upload Button. The message "Upload OK" appears if the sound file was successfully uploaded. If a problem occurs other messages will appear. Follow the instructions to correct the problem.

Tip: when you install a greeting its status is ENABLED. It will overwrite any existing enabled sound file with the same name. To change the status see 'How to Modify personal Greetings'.

Tip: When you upload a sound file it is converted to 8Khz sample rate, single channel (mono). This is standard phone quality and takes up less space on the voicemail server. If you download this file to your computer it will be of a lower quality than your original file (unless the original was also sampled at 8Khz).

C. How To Modify Personal Greetings

Delete – Permanently remove a greeting.

1. Click on the check box beside each greeting you wish to delete.

2. Click on the Delete Selected.

Disable – Prevent a greeting from being used but keep the file around.

1. Click on the check box beside each greeting you wish to disable.

2. Click on the Disable button.

Enable – Start using a previously disabled greeting.

1. Click on the check box beside each greeting you wish to enable.

2. Click on the Enable button.

Advanced Tip: If you would like to change a greeting but not delete the greeting you are changing, first Disable the greeting. Then Install (upload) the new greeting. Both the

original greeting and the new greeting will be available via your Smart Personal Console. The old greeting is marked disabled. The new greeting is marked enabled. Be careful not to disable the new greeting or the old greeting will be lost. To remove the new greeting, delete it OR enable the old greeting.

Appendices

Appendix 1: IP Telephones

IPitomy 480i

The Model 408i is an advanced, fully featured, IP screen Telephone that provides a flexible IP solution designed with interoperability and ease of use in mind.

IPitomy 9133i

The Model 9133i is an advanced, fully featured multi-line IP Telephone that takes full advantage of VoIP technology by offering a flexible, interoperable solution at an affordable price.

IPitomy 9112i

The Model 9112i is a value priced, basic single line IP Telephone with speakerphone offering a flexible, interoperable IP solution that meets VoIP budgets for most applications.

480i CT Cordless IP Phone

IPitomy 480i CT

IPitomy Communications is pleased to announce the newest addition to our Enterprise IP telephone portfolio. Consisting of an advanced, fully featured IP screen Telephone base station and separate cordless handset, the Model 408i CT provides high performance VoIP cordless mobility within an office or warehouse.

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 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. Part of the IPitomy family of IP telephones, the 53i is ideally suited for light to regular telephone requirements.

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 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. Since the IP550 is designed as a SIP based telephone, it is capable of operation equally as 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 57i

The 57i from IPitomy offers powerful features and flexibility in a standards based carrier-grade advanced level expandable IP telephone. With a sleek and elegant design, large 144 x 128 pixel graphical backlit LCD display and 6 dynamic context-sensitive softkeys, the 57i 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 is ideally suited for executives and heavy telephone users who require more one touch feature keys and a large screen to take full advantage of XML based programs.

IPitomy 57i CT

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.

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.

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**.

(Diagram 64)

Softphone Installation

1. **Download the CounterPath™ Softphone** to be used with the system from www.counterpath.com. The installation utility will install a phone icon in the toolbar of the operating system. It looks like a green light.

2. **Left click the Softphone** Icon in the operating system toolbar. The Softphone will appear.

3. **Right click on the Softphone** and select **SIP Account Settings** from the drop-down menu.

4. Click **Properties**. The properties window for the Softphone will appear.

5. Enter a **Display Name**. This is the name of the person or department associated with the phone.

6. 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.

7. 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.

8. Enter the **Extension Number** in the **Authorization User Name** field.

9. Enter the **Domain (IP Address)** of the system to which the Softphone is to be connected.

10. Click **Apply**.

11. Select **OK**. The 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

- **Pick the setting to be provisioned** – Unconditional, Busy, No Answer or Unavailable.

- o Select **Enabled** or **Disabled**. Disabled turns the forward setting off. Enabled turns the forward setting on.

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.

- o Dial ***90** to disable forwarding.
- o Dial ***91** to enable forwarding.
- o Dial ***92** to set the forwarding number.

Changing a Forwarding Number from a PC
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.

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.**
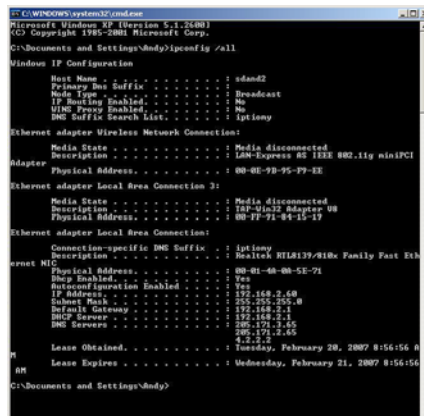
Appendix 2: IP Addresses

Determining Local Network Address (on a Microsoft® Windows® Machine)

To determine the local network address on a Microsoft® Windows® machine:

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.

(Diagram 65)

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:

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 **(Diagram 64),** the number *[192.168.1.249] is an **IP Address not in use** and can be assigned to the IPitomy IP PBX.

(Diagram 66)

*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**" like in **(Diagram 65)** 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.

(Diagram 67)

Implementation Note 7

Exact IP addresses will be determined by your specific network.

Appendix 3: 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

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 4: Router Configuration

The sample setup instructions below use a Linksys® router and are meant for demonstration purposes only.

1. Open a Web browser. Enter the IP address of the router (e.g., 192.168.1.1) and press Enter.

2. Login to the router using the Username and Password. For help refer to router documentation.

3. 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).

4. Open ports 10000 to 20000 on the IP address for the system being installed for both UDP and TCP (Name = RTP).

5. Open ports 4569 to 4569 on the IP address for the system being installed for both UDP and TCP (Name = IAX2).

Port Range					
Application	Start	End	Protocol	IP Address	Enabled
SIP	5060 to	5082	Both ▼	192.168.2.249	☑
RTP	10000 to	20000	Both ▼	192.168.2.249	☑
IAX2	4569 to	4569	Both ▼	192.168.2.249	☑
	0 to	0	Both ▼	192.168.2.0	☐
	0 to	0	Both ▼	192.168.2.0	☐
	0 to	0	Both ▼	192.168.2.0	☐
	0 to	0	Both ▼	192.168.2.0	☐
	0 to	0	Both ▼	192.168.2.0	☐
	0 to	0	Both ▼	192.168.2.0	☐
	0 to	0	Both ▼	192.168.2.0	☐
			Save Settings	Cancel Changes	

(Diagram 68)

6. Click Apply Changes.

Appendix 5: 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

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.

```
IPitomy Networking - TCP/IP Settings
USING:      IP Address :      192.168.2.14
            Mask       :      255.255.255.0
            MAC Address:      00:19:D1:3F:82:E7

DATABASE:
            NAME              VALUE
            ------------      -------------

    [1] Network Type          Static IP
    [2] IP Address            192.168.2.14
    [3] Subnet Mask           255.255.255.0
    [4] Default Gateway       192.168.2.1
    [5] Static DNS            192.168.2.1
    [6] Static DNS 2
    [7] Static DNS 3
                              Other Menus
    [R] Refresh View          --------------
                              [F] Fix Database
    [D] Default Settings      [W] Web Server
    [S] Save & Update         [H] Host Access

        SELECT OPTION: ▮
```

Main Menu (Diagram 69)

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" (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".

```
IPitomy Networking - TCP/IP Settings - IP Address

USING:      IP Address :      192.168.2.14
            Mask       :      255.255.255.0
            MAC Address:      00:19:D1:3F:82:E7

DATABASE:
            NAME              VALUE
            ------------      -------------

    [ ] IP Address            192.168.2.14

    PRESS "ENTER" TO CANCEL
        OR ENTER NEW VALUE: ▮
```
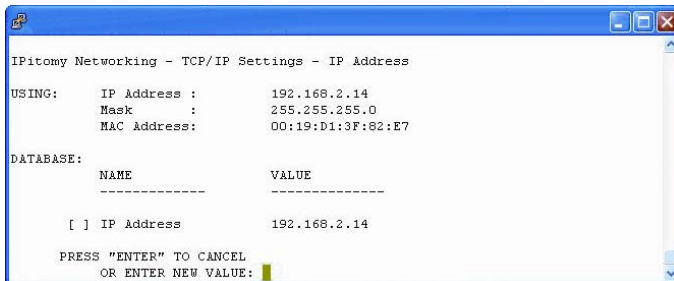
Editing IP Address (Diagram 70)

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

1. Change **Network Type** to "DHCP".

2. Save & Update.

Renewing DHCP Lease

1. Verify **Network Type** is set to "DHCP".

2. Save & Update.

```
IPitomy Networking - TCP/IP Settings - Network Type

USING:     IP Address :      192.168.2.14
           Mask       :      255.255.255.0
           MAC Address:      00:19:D1:3F:82:E7

DATABASE:
           NAME              VALUE
           -------------     --------------

      [ ] Network Type       Static IP

OPTIONS:

           Static IP

      [2] DHCP


      PRESS "ENTER" TO CANCEL
          OR SELECT OPTION:
```
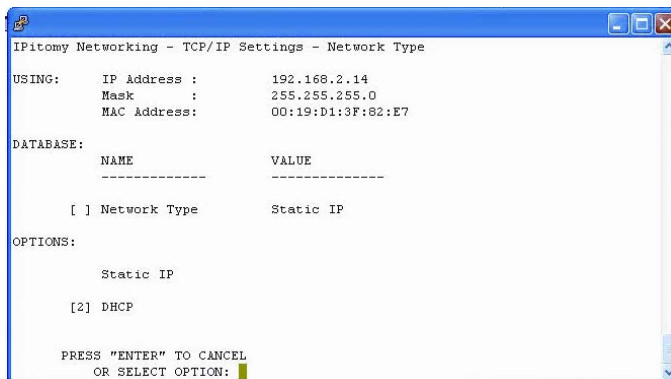
Network Type Sub Menu (Diagram 71)

Other Menus

- [W] Web Server

- [H] Host Access

- [F] Fix Database

Web Server

```
IPitomy - Networking - Web Server Settings
Changes here are saved immediately.

      [U]  Restert Web Server

           Allow Access From
           -----------------

      [1]  192.168.
      [2]  72-164-254-211.dia.static.qwest.net
      [3]  72.164.254.211
      [ ]  IPitomy.com

      [R]  Refresh View
      [D]  Load Factory Defaults
      [#]  Delete Item
      [+]  Add Item

      [M]  Main Menu

    SELECT NUMBER TO DELETE
          OR ENTER OPTION:
```
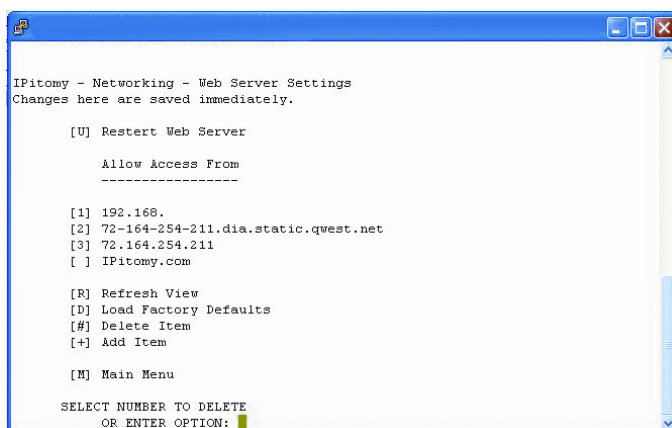
Web Server Sub Menu (Diagram 72)

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

```
          SELECT OPTION: h




IPitomy - Networking - Host Access
Changes here are saved immediately.


            Service       Allow Access From
          ----------      --------------------

      [1]     sshd      192.168.
      [2]     sshd      72-164-254-211.dia.static.qwest.net
      [3]     sshd      IPitomy.com

      [R]  Refresh View
      [D]  Load Factory Defaults
      [#]  Delete Item
      [+]  Add Item

      [M]  Main Menu

          SELECT OPTION:
```
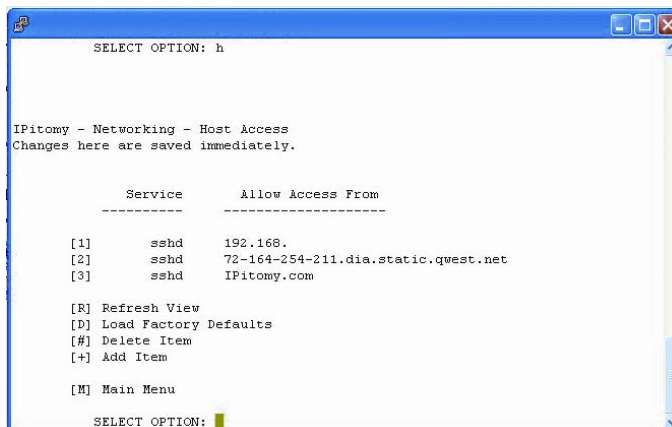
Host Access Sub Menu (Diagram 73)

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 6: Troubleshooting

Why does my Aastra phone freeze with 100% done written on the screen?

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:

1. Change the setting in PBX Setup / SIP.

2. Click on "Apply Changes".

3. Restart the phone.

Why did my phone restart with an old configuration after using Auto Discovery to create a new extension, assign and restart the phone?

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.

Why does my Aastra phone hang (or freeze) with "40% Done" written on the screen?

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.

Why won't my time zone on my voicemail change after my system settings have been changed?

You have to go to PBX>Services and click on the Restart PBX button in order for the changes to take effect.

Why can't I delete a menu?

Be sure the menu is not being used in incoming routing.

What is the default IP Address for the IP PBX?

192.168.1.249

For all most recent troubleshooting FAQ's please visit www.ipitomy.com

Glossary

Analogue Telephone Adapter (ATA) – Connects a telephone to a high-speed modem and facilitates VoIP or fax calls over the internet.

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 Indicator (BLI) – An LED on a telephone showing which line is in use.

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 number of calls, call duration, call origination and call destination.

Digital Subscriber Line (DSL) – This service provides digital phone service over an analog line.

Direct Inward Dial (DID) – A telephone number assigned exclusively to an extension or person. This number allows a caller to reach a person directly without using a menu.

CODEC (Compression-decompression) – This voice compression-decompression algorithm defines the rate of speech compression, quality of decompressed speech and processing power requirements. In VoIP, ITU-T G.723.1 and G.729 (AB) are the most often used CODECS.

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.

Encryption – The process of scrambling data to prevent the accurate interpretation of this data by anyone except those for whom it is intended.

Forward – Automatically forwards an incoming call to another telephone number.

Gateway – A device that interconnects networks with different, incompatible communications protocols.

IEEE – The Institute of Electrical and Electronics Engineers – An independent institute that develops networking.

Infrastructure – Currently installed computing and networking equipment.

IP Telephony – Phone service (voice calls) carried over a network using Session Initiation Protocol.

Internet Protocol (IP) – A protocol used to send data over a network.

Internet Service Provider (ISP) – A company that provides access to the Internet.

LAN (Local Area Network) – 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.

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.

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.

Packet – A unit of data transmitted over a network.

Park – Parks a call in a reserved extension (park slot) and allows the call to be retrieved from another extension.

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.

Router – A networking device that connects multiple networks together, such as a local network and the Internet.

Server – Any computer in a network that provides users access to files, printing, communications, etc.

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 Operator Console (SOC) – This is a Web-based and intuitive attendant station. It graphically depicts call traffic and with the click of a mouse allows a user to manage this traffic by transferring calls, placing callers on a park slot and/or putting a caller into an existing conference.

Smart Personal Console (SPC) – This user-friendly Web page gives a person the ability to set basic phone features (e.g., mailbox settings and call forwarding) from anywhere.

Switch – Software used to bridge a public switched telephone network and voice over Internet. The switch performs call control functions such as protocol conversion, authorization and other administrative operations.

Uninterruptible Power Supply (UPS) – A devise that maintains continual electrical power.

T1 – A dedicated digital voice circuit that has 24 channels. This point-to-point circuit delivers 1.544 Mbps of bandwidth.

Transfer – Sends a call to another extension.

Trunk – A communications channel between two points.

Virtual Private Network (VPN) – A private communication network that companies use to transmit information securely by encrypting traffic sent from one network to another.

Voice Over Internet Protocol (VoIP) – The routing of voice traffic over the internet.

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.

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