IPitomy IP1000 User Guide

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Introduction

About the IP1000

The IP1000 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 50 users and the system will work with analog lines for traditional Public Switched Telephone Network (PSTN) connectivity. In addition to traditional telephone lines, the IP1000 can use VoIP service providers replacing traditional PSTN lines with a broadband connection.

Benefits of VoIP Technology

The IP1000 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 voices 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.
- **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.
- **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 IP1000 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 IP1000. 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 IP1000.
- **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 IP1000. Just move your mouse over the and a brief description of the field will pop up.

The Installation Worksheet

Use the **IP1000 Installation Worksheet** to make collecting information used in the implementation of the system simple.

Product Overview

IP1000 Components

Understanding the IP1000's architecture and how it works will make installing the system simple.

Powerful All-In-One Communications Platform

The IP1000 IP PBX is an all-in-one business communications system. This powerful system includes a complete suite of business communication applications in one appliance:

- Fully-featured Business Phone System
- Automated Attendant and 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
- Remote Extensions
- Browser-based Administration

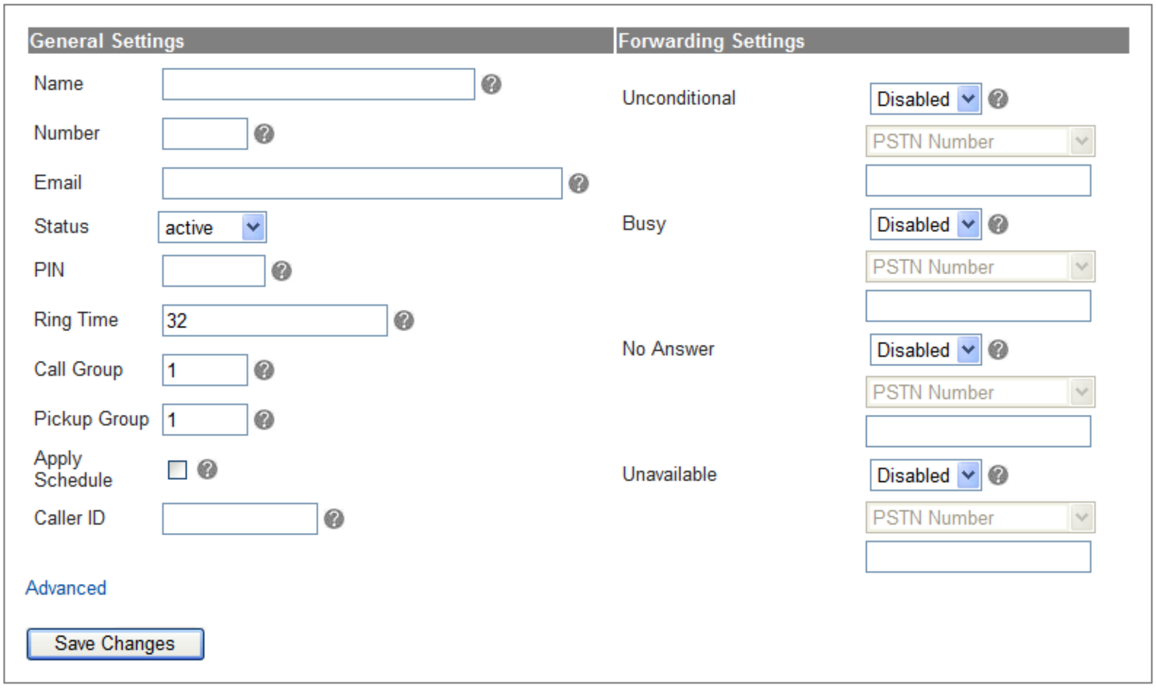
Entering System Information

The system is configured by entering information into the appropriate fields on the menu screens. Some fields are populated with data that is entered 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 recommend that information be entered in the following order:

- Extensions
- Groups
- Menus
- Providers

Entering information in this sequence will reduce the time it takes to up the system:

- Extensions will be populated in the drop-down menus for creating groups.
- Groups and extensions will be populated for creating automated attendant (menu) routing.
- Destinations will be populated for use in setting up providers.



System Administration

IP1000'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 IP1000 is simple and intuitive. The system is designed with six primary areas of functionality.

- **Networking** – Networking 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.

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 IP1000, 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 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 function.

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.

Direct Inward Dialing (DID) Numbers

A Direct Inward Dialed (DID) number is a telephone number assigned by a service provider. DIDs allow direct routing of a call to a destination within the system. This can be an individual extension, group, conference or menu.

Conferencing

A Meet-me Conference is an extension on the system used for conference calls. Participants of a conference 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 IP1000 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 setup in the Extensions setup 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.

Supported IP Phone Sets

The IP1000 works with a variety of business-grade IP phone sets. See **Appendix 1: IP Telephones** for a complete list.

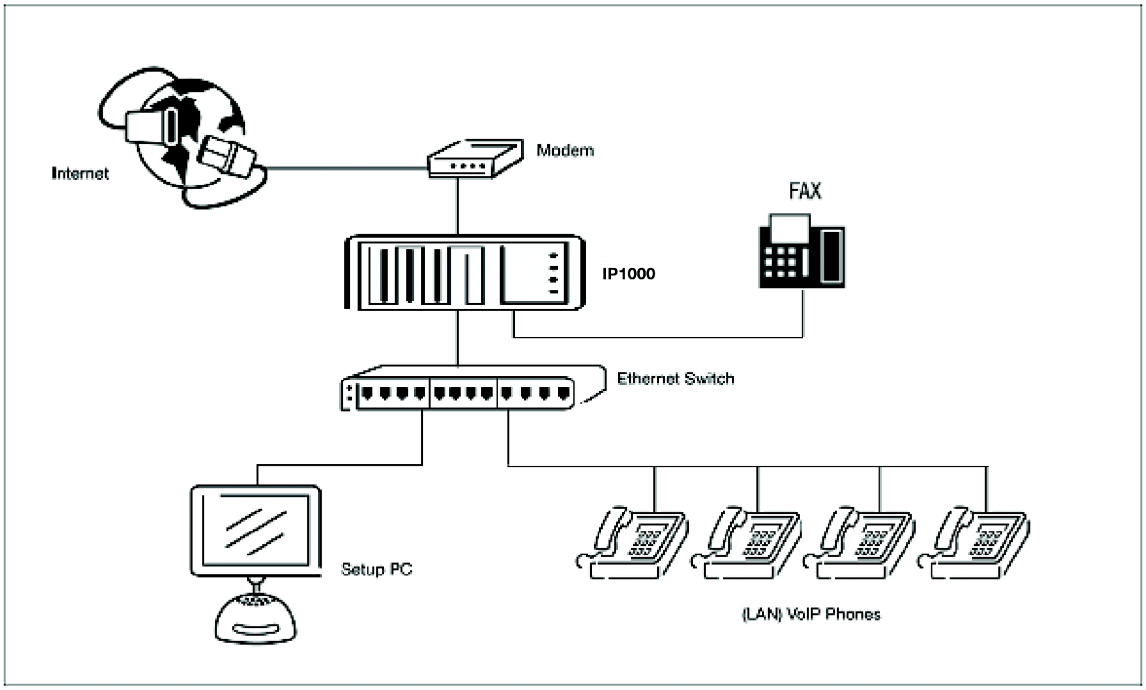
Before Getting Started

Planning before getting started will make the setup and installation of the IP1000 simple. An installation Worksheet is created to assist in recording business and system information used in planning system setup and installation. This Checklist is in Appendix 3 of this Guide.

Connecting the System

Hardware Setup

The IP1000 comes assembled and ready to install. The system requires connection to the PSTN for analog lines. It requires telephones to be connected to the local area network (LAN) via the IP1000. Broadband access must also be established for VoIP connectivity (allowing remote extensions and remote management).



Connecting the Phone Lines and FAX Machines

The IP1000 is equipped to support analog, gateway or SIP connections. Analog lines 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.

Embedded Analog Phone Ports

The IP1000 has analog phone ports and analog line ports embedded on board:

- **Two Analog Phone Ports** – The IP1000 has two analog phone ports embedded on board for connectivity to FAX machine, analog phones or cordless phones with FXS interfaces.
- **Two Analog Line Ports** – The IP1000 has two analog line ports embedded on board for PSTN connectivity.

Expandable Analog Line/Phone Card

The IP1000 equipped an expansion slot:

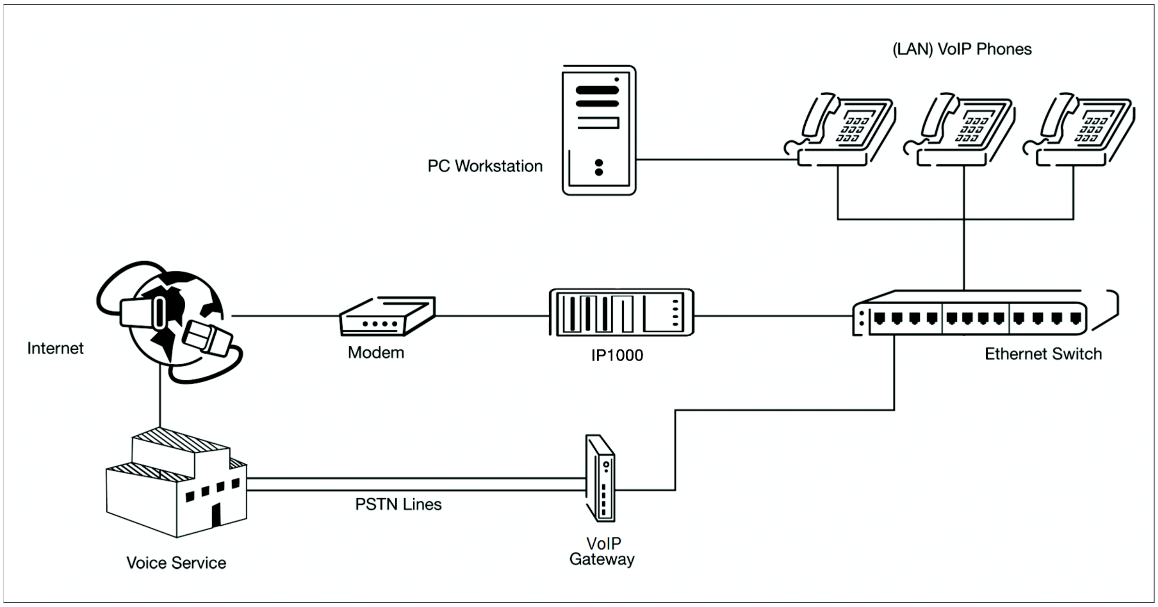
▪ **Analog Line Card** – This card supplies four analog lines to expand the PSTN connectivity. The card supporting these connections is already installed and completely configured. Simply connect the phone lines to the RJ11 jacks at the rear of the IP1000 and start making calls. These connections are single pair; one line per jack.

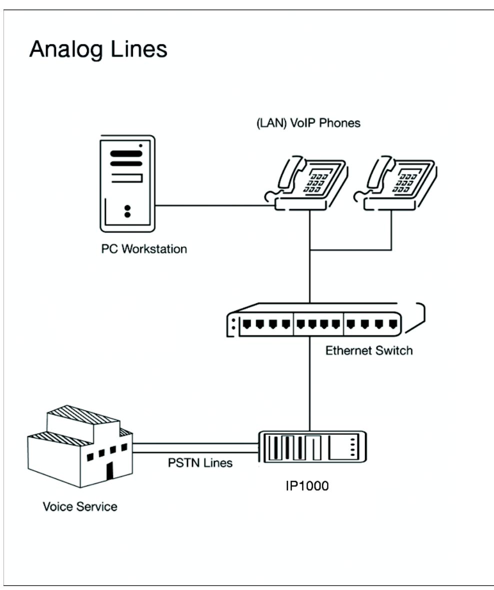
Connecting Using an External Gateway

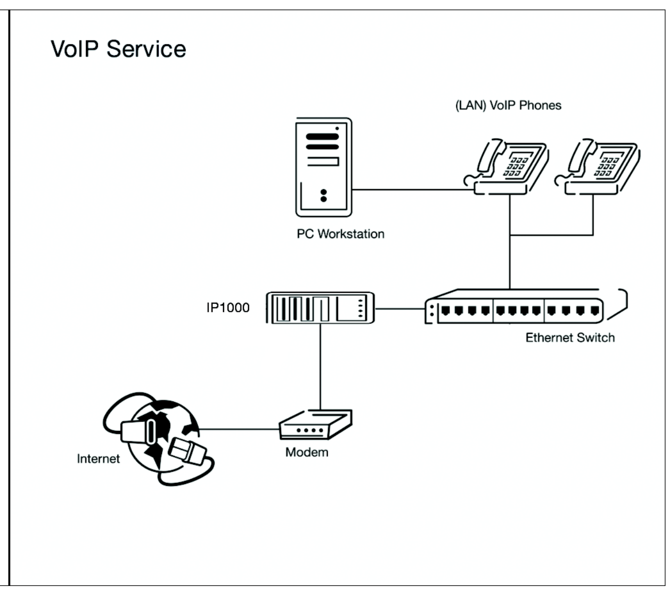
PSTN lines are connected to a Gateway device. The gateway device is connected to the LAN. The Gateway is then registered as a SIP provider in the system.

Connecting Using SIP Providers

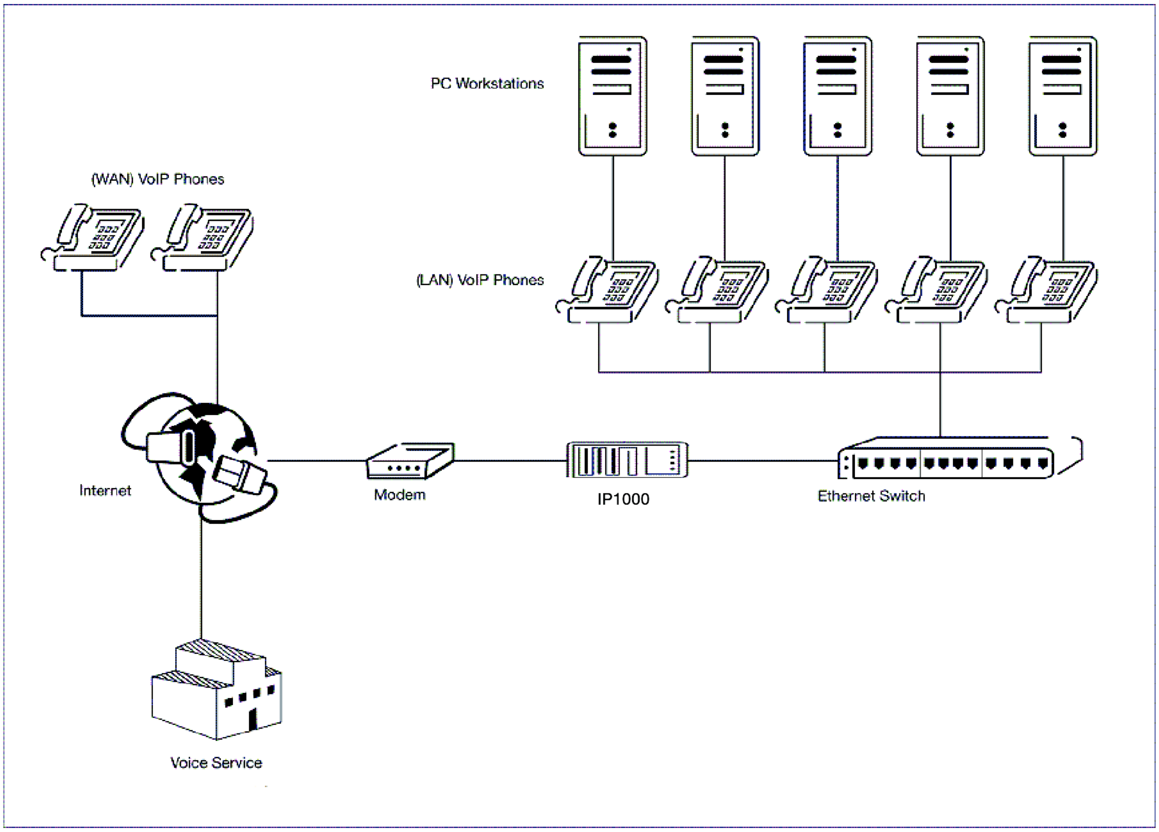
Once connected to the LAN, the LAN's broadband connection provides a pathway for SIP VoIP Providers. Use the SIP Provider pages to setup a connection.







Connecting to a LAN



System Requirements

Network Requirements

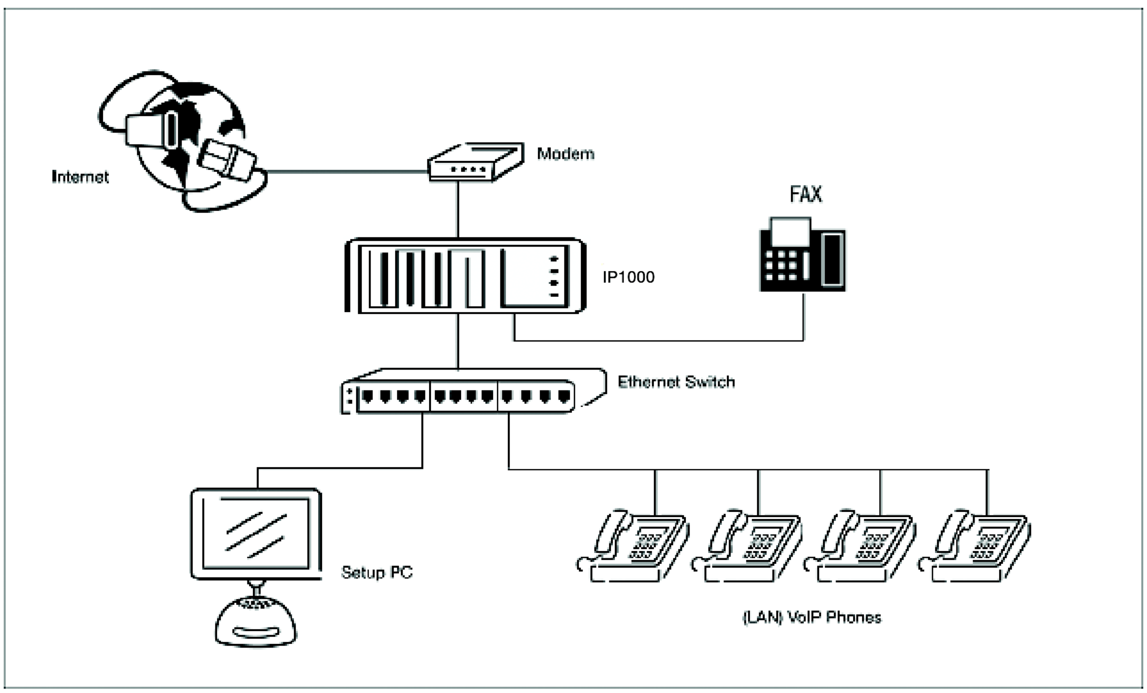
Making preparations for the network in advance will assure there are no surprises. If you are going to have remote extensions, you will need access to the router to setup a network address translation (NAT) and port forwarding.

A LAN with a broadband connection is required for operation of the system. It is must be on fast Ethernet (100baseT or better). The system must also use Ethernet data switches. The router inside IP1000 can use DHCP or not, depending on preference. If IP1000 is connected to Internet through broadband modem, use PPPoE for connection to Internet and use DHCP server for local network is recommended.

IP Addresses

It is important to know the LAN configuration and IP addresses of the specific network the system is becoming a part of to make installation of the IP1000 simple. The IP1000 is required to have a fixed (static) IP address. To get the information about public IP check the network administrator.

By default, the IP address used by IP1000's router for local network is 192.168.1.1. The devices including PC, IP phones and other network devices will get IP address from IP1000's DHCP server and those IP addresses will be 192.168.1.xxx. To connect to IP1000 by PC in local network for changing system settings or monitoring system's status, login to IP1000 with IP address 192.168.1.1. The public IP address used for IP1000 to connect to Internet can also be viewed from IP1000's web management pages.



Service Providers

In order to provision the IP1000 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 Service (POTS). 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 or SIP)
- Phone Numbers Associated with the Service
- Password and Login Information for SIP Service

Record this information on the **IP1000 Installation Worksheet**.

System Administration

About the Administration Menu

IP1000'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.

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

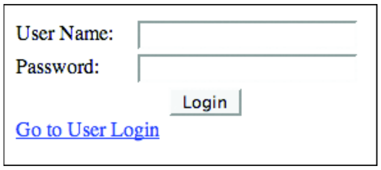
Each Online Administration page also contains:

- **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 is a useful feature for making changes to the system prior to applying them. **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. Once changes have been saved, clicking the Apply Changes button will apply the changed administration pages 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 IP1000** can be configured. The Advanced button is located on the lower left side of each page.

Log In

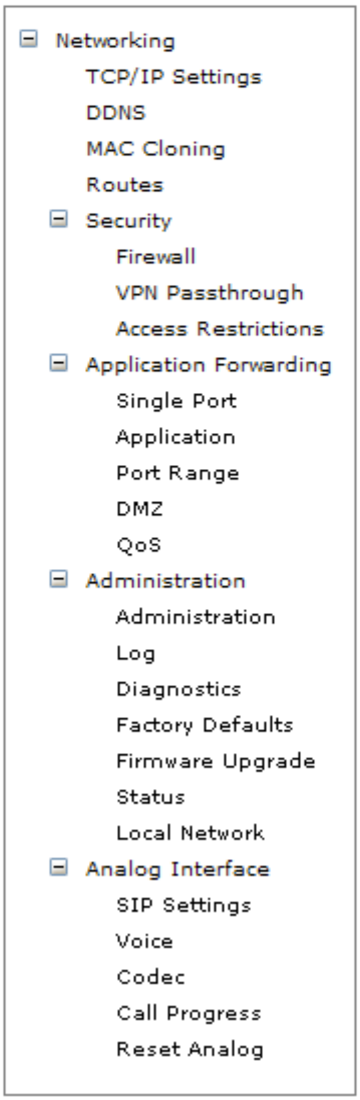


User Name: pbxadmin

Password: ipitomy

Networking

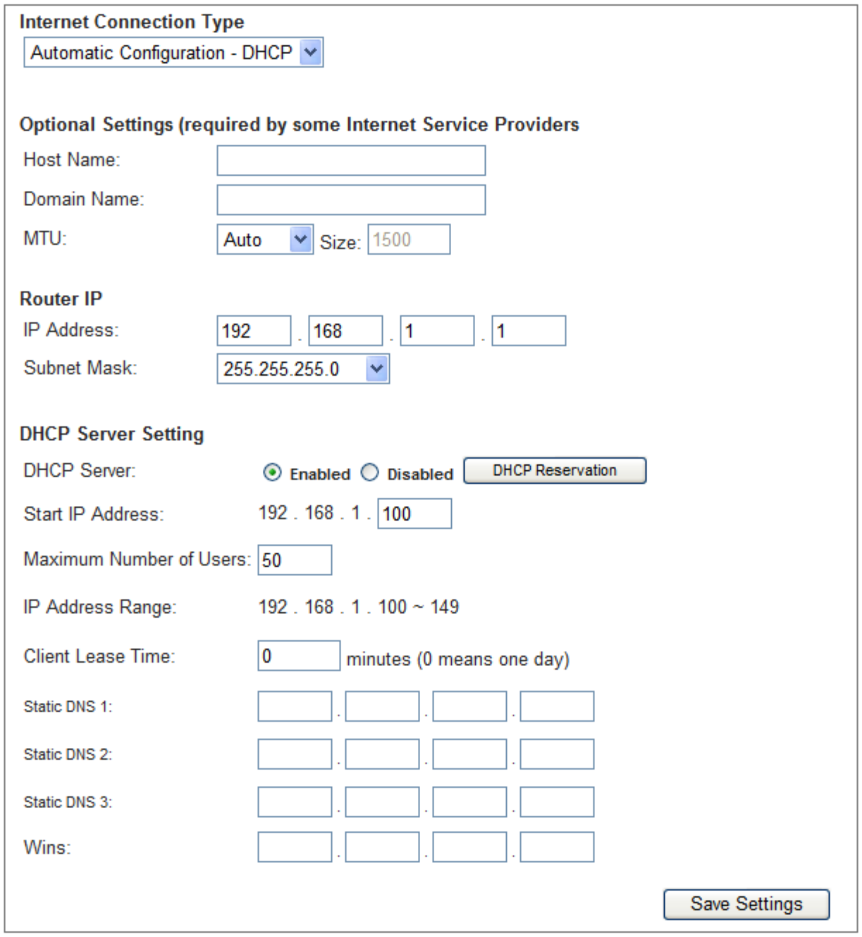
The System Menu is for setting up network attributes. For example the IP address of the system and router information.



TCP/IP Settings

The Networking Setup Menu defines the Internet Setup for the system's hardware. Either to get IP address through DHCP or PPPoE, or to be assigned by user, the system must operate using an IP address. The TCP/IP Settings section configures the IP PBX for your Internet connection type. This information can be obtained from the service provider.

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 Internet Connection Type

The IP1000 supports six connection types: Automatic Configuration – DHCP, Static IP, PPPoE, PPTP, L2TP, and Telstra Cable. Each setup screen and available features will differ depending on what kind of connection type you select.

- **Automatic Configuration (DHCP)** – By default, the IP1000's Internet Connection Type is set to Automatic Configuration (DHCP), and it should be used only if your ISP supports DHCP or you are connecting through a dynamic IP address.
- **Static IP** – If a permanent IP address is provided then select Static IP. And settings for following network attributes are required. Contact service provide or the network administrator for any Information missed.
  - **Internet IP Address** – Normally the static IP address is a public IP address provided by service provider; it is used for connecting to Internet. If it is a local IP address assigned by the network administrator; the router function in IP1000 may need to be disabled.
  - **Subnet Mask** – The subnet mask Information should be provided along with IP address.

- **Default Gateway** – The IP address for the default network gateway, it is the information service provider should provide.
- **DNS 1-3** – Service provider will provide at least IP address for one DNS (Domain Name System) server. At most three DNS servers can be set.

▪ **PPPoE** –Some DSL service providers use PPPoE (Point-to-Point Protocol over Ethernet) to establish Internet connections for end-users. If DSL connection to Internet is using, check service provider for the connection provisioning type. Enable it if PPPoE is used.

- **User Name and Password** – Enter the user name and password provided by service provider. User name and password will be used for authentication while establishing PPPoE connection.
- **Max Idle Time and Connect on Demand** – If the connection stay inactive for over a specific period time (Max Idle Time) the PPPoE connection may be cut off. Assign 0 to Max Idle Time field will always keep the connection no matter it is active or not. If Internet connection has been terminated due to inactivity, automatic re-establishment for Internet connection will be invoked by any attempt of access to Internet if the Connect on Demand field is checked.
- **Keep Alive and Redial Period** – If Keep Alive is enabled, system will periodically check the Internet connection. If the connection is down, then the system will automatically re-establish the connection. To use this option, click the radio button next to Keep Alive. The Redial Period is the time period to trigger system to check the Internet connection; default Redial Period is **30** seconds.

▪ **PPTP** – Point-to-Point Tunneling Protocol (PPTP) is a service that deployed in Europe and Israel only.

- **Internet IP Address** – Normally the static IP address is a public IP address provided by service provider; it is used for connecting to Internet. This IP address must be assigned from IP provider.
- **Subnet Mask** – The subnet mask Information should be provided along with IP address.
- **Default Gateway** – The IP address for the default network gateway, it is the information service provider should provide.
- **User Name and Password** – Enter the user name and password provided by service provider. User name and password will be used for authentication while establishing PPTP connection.
- **Max Idle Time** – If the connection stay inactive for over a specific period time (Max Idle Time) the Internet connection may be cut off. Assign 0 to Max Idle Time field will always keep the connection no matter it is active or not.
- **Connect on Demand** – If Internet connection has been terminated due to inactivity, automatic re-establishment for Internet connection will be invoked by any attempt of access to Internet if the Connect on Demand field is checked.
- **Keep Alive and Redial Period** – If Keep Alive is enabled, system will periodically check the Internet connection. If the connection is down, then the system will automatically re-establish the connection. To use this option, click the radio button next to Keep Alive. The Redial Period is the time period to trigger system to check the Internet connection; default Redial Period is **30** seconds.

▪ **L2TP** –Layer 2 Tunneling Protocol (L2TP) is a service that tunnels Point-to-Point Protocol (PPP) across the Internet. It is used mostly in European countries. Check with service provider for necessary setup information.

  - **Internet IP Address** – Normally the static IP address is a public IP address provided by service provider; it is used for connecting to Internet. This IP address must be assigned from IP provider.
  - **User Name and Password** – Enter the user name and password provided by service provider. User name and password will be used for authentication while establishing PPTP connection.
  - **Max Idle Time and Connect on Demand** – If the connection stay inactive for over a specific period time (Max Idle Time) the PPPoE connection may be cut off. Assign 0 to Max Idle Time field will always keep the connection no matter it is active or not. If Internet connection has been terminated due to inactivity, automatic re-establishment for Internet connection will be invoked by any attempt of access to Internet if the Connect on Demand field is checked.
  - **Keep Alive and Redial Period** – If Keep Alive is enabled, system will periodically check the Internet connection. If the connection is down, then the system will automatically re-establish the connection. To use this option, click the radio button next to Keep Alive. The Redial Period is the time period to trigger system to check the Internet connection; default Redial Period is **30** seconds.

- **Telstra Cable** –Telstra Cable is a service used in Australia only. Check with service provider for necessary setup information.
  - **Server IP Address** – Normally the static IP address is a public IP address provided by service provider; it is used for connecting to Internet.
  - **User Name and Password** – Enter the user name and password provided by service provider. User name and password will be used for authentication while establishing Telstra cable connection.

Optional Settings

Some server providers may require the following settings. Check with the service provider before making any changes.

- **Host Name and Domain Name** – Some service providers require these names as identification. You may need to check with service provider to see if it is required. In most cases, leaving these fields blank will work.
- **MTU** – The MTU (Maximum Transmission Unit) setting specifies the largest packet size permitted for network transmission. To manually set a value, select **Manual** and enter the value desired in the **Size** field. MTU value should be in the range from 1200 to 1500. Normally the value 1492 is used. The default is **Auto**, which allows the system to select the best MTU for your Internet connection.

Router IP

The local IP address and Subnet Mask are shown here. In most cases, keeping the defaults is recommended.

- **IP Address** – The default value is **192.168.1.1**.
- **Subnet Mask** – The default Subnet Mask is 255.255.255.0.

DHCP Server Setting

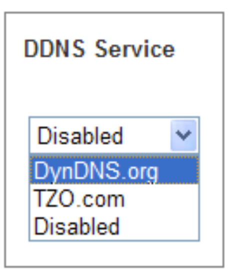
The IP1000 can be used as a Dynamic Host Configuration Protocol (DHCP) server, hence no router device is required. DHCP server automatically assigns an IP address to each computer or network equipment in a local network. It is highly recommended to utilize IP1000's DHCP server function.

- **DHCP Server** – By Factory default DHCP is enabled.
- **Start IP Address** – Enter an initial IP address for the DHCP server to start with when assigning IP addresses. Because the default local IP address for the IP1000 is 192.168.1.1, the Start IP Address must be one between 192.168.1.2 and 192.168.1.254. The default Start IP Address is **192.168.1.100**
- **Minimum Number of Users** – The maximum number of IP addresses that allow the DHCP server to assign to. This number cannot exceed 253. The default number is **50**.
- **Client Lease Time** – The Client Lease Time is the amount of time a network device will be allowed to use the dynamically assigned IP address by IP1000. After the Client Lease Time expires the assigned IP address will be released and be assigned with a new dynamic IP address. The default value is **0** minutes, which means one day.
- **Static DNS 1-3** – The Domain Name System (DNS) is how the Internet translates domain or website names into Internet addresses or URLs. Service providers should provide at least one DNS Server IP Address to each user.
- **WINS** – The Windows Internet Naming Service (WINS) converts NetBIOS names to IP addresses. If a WINS server is used, enter the servers IP address here, otherwise leave this field blank.

DDNS

The IP1000 offers a Dynamic Domain Name System (DDNS) feature. DDNS assigns a fixed host name and a domain name to a dynamic Internet IP address. It is useful when users are hosting their own website, FTP server, or other server behind the IP1000.

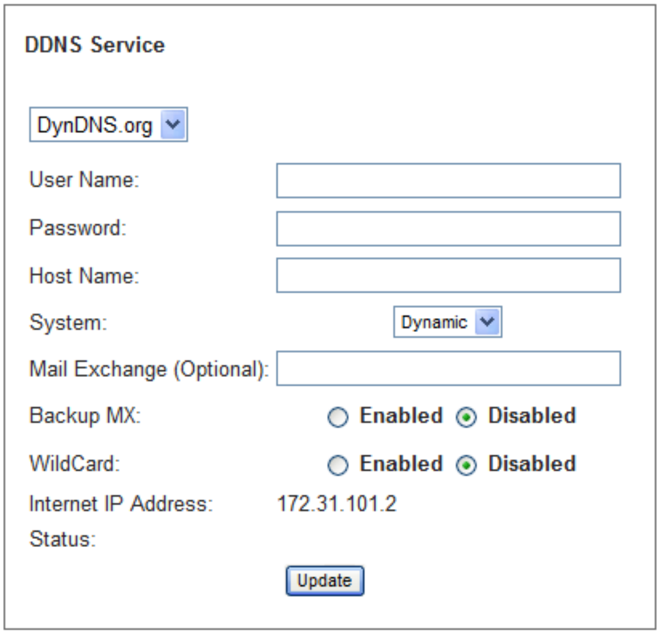
Before using this service, it is required to sign up DDNS service at one of two DDNS service providers such as DynDNS.org or TZO.com. Default setting is **Disabled**.



DDNS Service

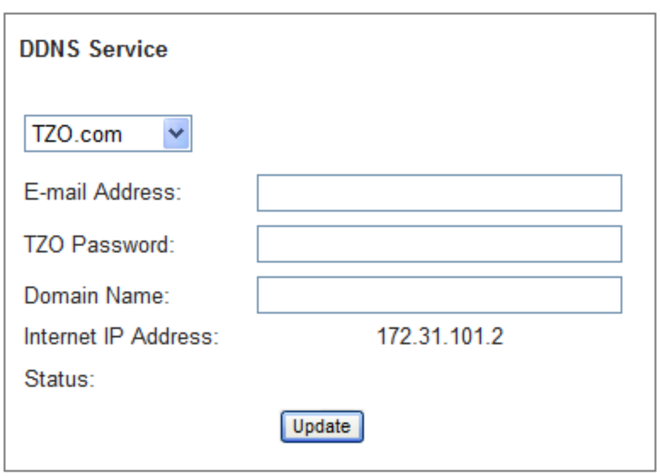
Select the DDNS service provider from the drop-down menu. There are two options in the menu, **DynDNS.org** and **TZO.com**. The features available on the DDNS screen will vary depending on which DDNS service provider is used.

- **DynDNS.org**



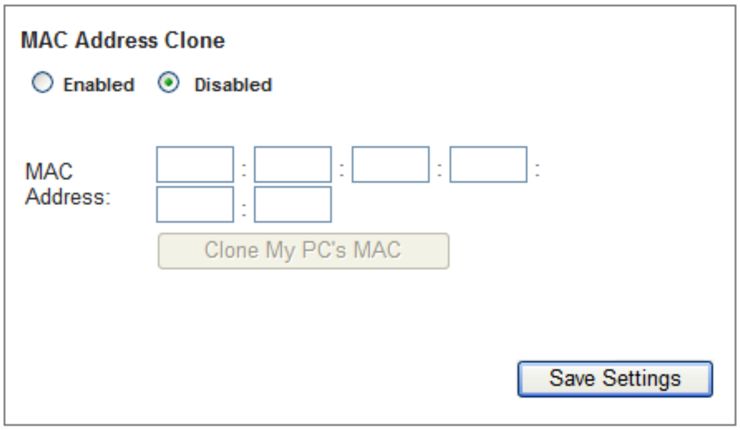
  - **User Name, Password and Host Name** – Enter the settings of the account you set up with DynDNS.org.
  - **System** – Select the DynDNS service you use: **Dynamic, Static,** or **Custom**.
  - **Mail Exchange** – Enter the settings of the account set up with DynDNS.org.
  - **Backup MX** – This feature allows the mail exchange server to be a backup. By default, this feature is **Enabled.** To disable this feature, select **Disabled.**
  - **WildCard** – This setting is for enabling or disabling wildcards. For example, if your DDNS address is myplace.dyndns.org and you enable wildcards, then x.myplace.dyndns.org will work as well (x is the wildcard). By default wildcards is **Enabled.** To disable wildcards, select **Disabled.**
  - **Internet IP Address** – The IP1000's Internet IP address is displayed here.
  - **Status** – The status of the DDNS service connection is displayed here.
  - **Update** – To manually trigger an update, click this button.

- **TZO.com**



- **E-mail Address, TZO Password,** and **Domain Name** – Enter the settings of the account set up with TZO.
- **Internet IP Address** – The IP1000's Internet IP address is displayed here.
- **Status** – The status of the DDNS service connection is displayed here.
- **Update** – To manually trigger an update, click this button.

MAC Clone



A MAC address is a 12-digit code assigned to a unique piece of hardware for identification, like a social security number. Some ISPs will require you to register a MAC address in order to access the Internet. If you do not wish to re-register the MAC address with your ISP, you may assign the MAC address you have currently registered with your ISP to the IP PBX with the MAC Address Clone feature.

MAC Address Clone

To use MAC address cloning, select **Enabled.** Otherwise, keep the default, **Disabled.**

MAC Address

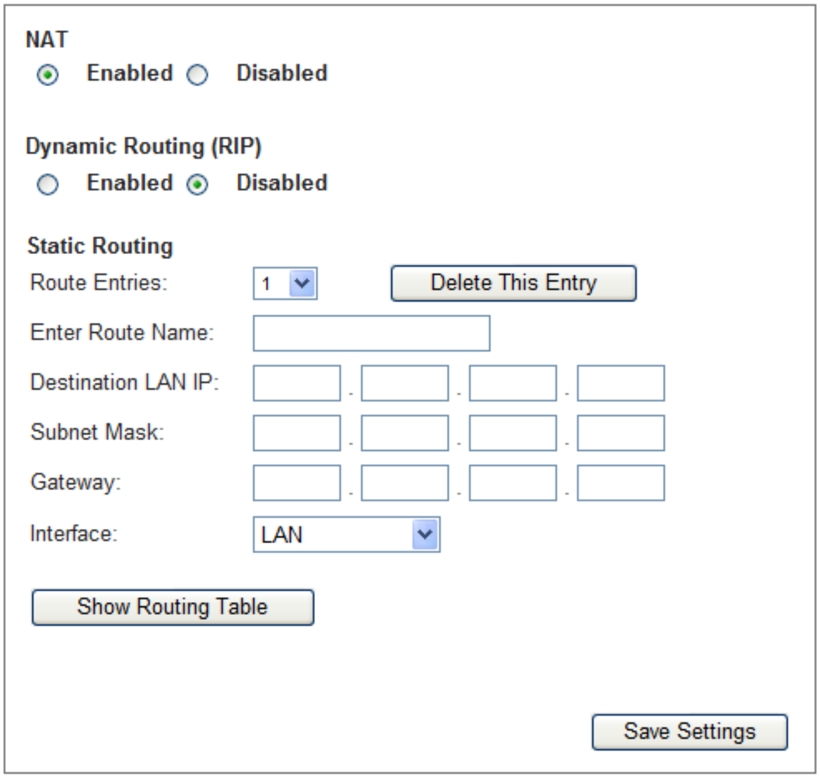
Enter the MAC Address registered with service provider.

Clone My PC's MAC

Click this button to clone the MAC address of the PC be currently using to configure the IP1000. The IP1000 will automatically detect PC's MAC address. It is recommended that the PC registered to the service provider is used to open the MAC Address Clone screen.

Routes

The Routes screen allows user to configure the dynamic and static routing settings.



NAT

If IP1000 is hosting your network's connection to the Internet, select **Enabled.** If another Router exists in front of IP1000, select **Disabled.** When the NAT setting is disabled, dynamic routing will be enabled.

Dynamic Routing (RIP)

This feature enables the IP1000 to automatically adjust to physical changes in the network's layout and exchange routing tables with the other router(s). The IP PBX determines the network packets' route based on the fewest number of hops between the source and the destination locations. To use dynamic routing, select **Enabled.** Otherwise, select **Disabled.** When the NAT setting is disabled, dynamic routing will be enabled.

Static Routing

A static route is a pre-determined pathway that network information must travel to reach a specific host or network. Use this feature to set up a static route, alter the following settings:

- **Route Entries** – Select the number of the static route from the drop-down menu.
- **Enter Route Name** – Enter a name for the static route, using a maximum of 25 alphanumeric characters.
- **Destination LAN IP** – The Destination LAN IP Address is the address of the remote network or host to which you want to assign a static route. Enter the IP address of the host for which you wish to create a static route.
- **Subnet Mask** – The Subnet Mask determines which portion of a Destination IP address is the network portion, and which portion is the host portion.
- **Gateway** – This is the IP address of the gateway device that allows for contact between the IP PBX and the remote network or host.

- **Interface** – Select **LAN** or **WAN (Internet)** depending on the location of the final destination.

Delete This Entry

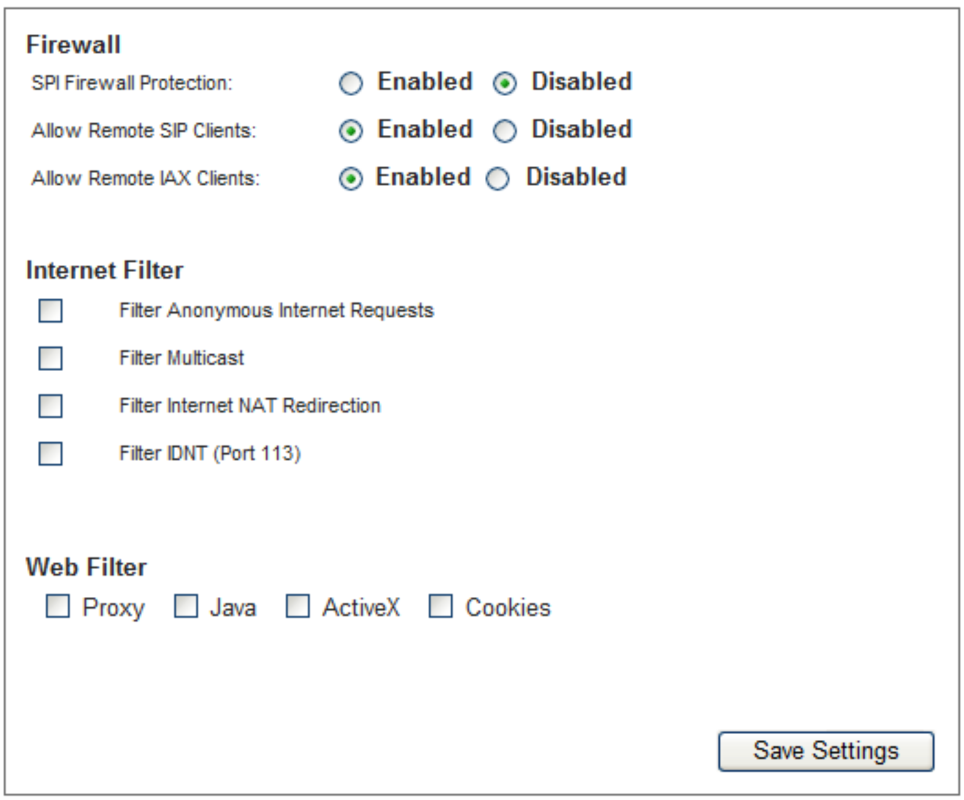
To delete a route, select its number from the drop-down menu, and click this button.

Show Routing Table

Click the **Show Routing Table** button to open a screen displaying how data is routed through your local network. For each route, the Destination LAN IP address, Subnet Mask, Gateway, and Interface are displayed. Click the **Refresh** button to update the information. Click the **Close** button to exit this screen.

Security

Firewall



The Firewall screen offers a firewall and filters that block specific Internet data types.

Firewall

- **SPI Firewall Protection** – A firewall enhances network security and use Stateful Packet Inspection (SPI) or more detailed review of data packets entering your network. Select **Enabled** to use a firewall, or **Disabled** to disable it.
- **Allow Remote SIP Clients** – Enabling **Allow Remote SIP Clients** setting will allow the SIP packets to pass through the firewall. This allows administrators to setup the connection between this IP PBX and the external SIP phones or SIP trunks from the Internet. To enable the communication with remote SIP devices, select **Enabled.** Otherwise, select **Disabled.**

- **Allow Remote IAX Clients** – Enabling **Allow Remote IAX Clients** setting will allow the IAX packets to pass through the firewall. This will allow the IP1000 to setup a peer-to-peer connection with another IP1000. To enable the IAX peer-to-peer communication with remote IP PBX, select **Enabled.** Otherwise, select **Disabled.**

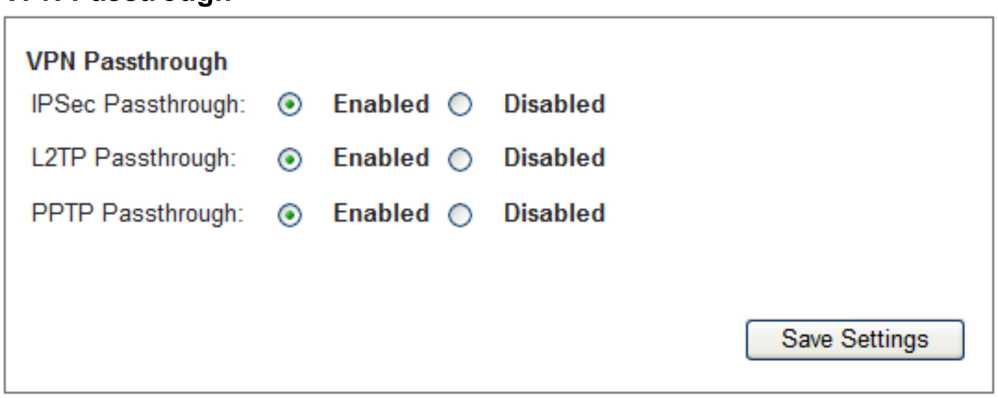
Internet Filter

- **Filter Anonymous Internet Requests** – When enabled, this feature protects the network behind IP1000 from being "pinged" or detected by other Internet users. It also hides the used network ports. This filter is enabled by default. Click the check box to enable or disable.
- **Filter Multicast** – Multicasting allows for multiple transmissions to specific recipients at the same time. If multicasting is permitted, then IP1000 will allow IP multicast packets to be forwarded to the appropriate computers. Click the check box to enable or disable.
- **Filter Internet NAT Redirection** – This feature uses port forwarding to block access to local servers from local network computers. Click the check box to enable or disable.
- **Filter IDENT (Port 113)** – This feature protects port 113 from being scanned by devices outside of your local network. Click the check box to enable or disable.

Web Filter

- **Proxy** – Use of WAN proxy servers may compromise the Gateway's security. Denying Filter Proxy will disable access to any WAN proxy servers. To enable proxy filtering, click the checkbox.
- **Java** – Java is a programming language for websites. If Java is filtered, it may fail to access to Internet sites created by using Java. To enable Java filtering, click the checkbox.
- **ActiveX** – ActiveX is a programming language for websites. If ActiveX is filtered, it may failed to access to Internet sites created by using ActiveX. To enable ActiveX filtering, click the checkbox.
- **Cookies** – A cookie is data stored on your computer and used by Internet sites when you interact with them. To enable cookie filtering, click the checkbox.

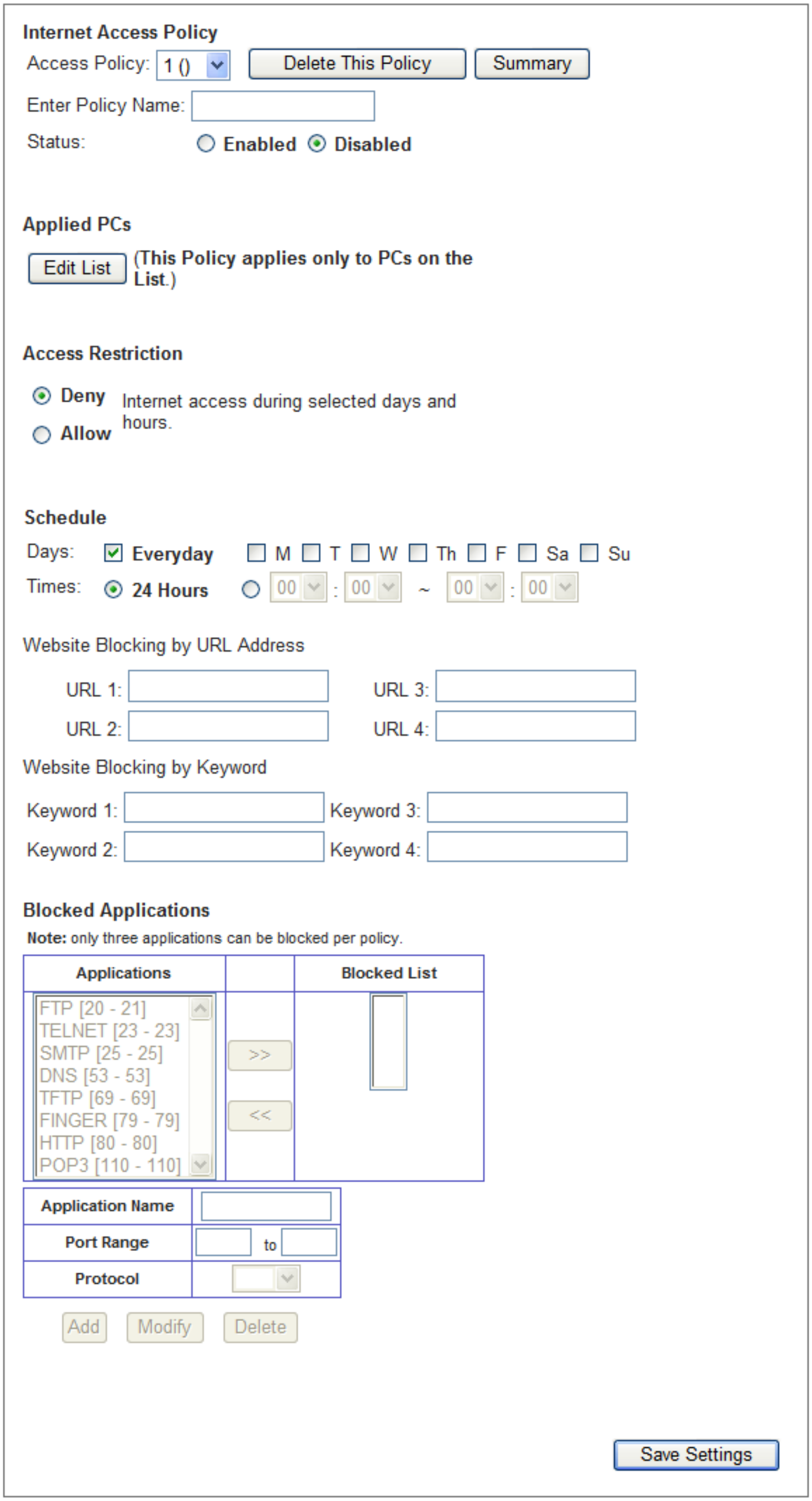
VPN Passtrough



The VPN Passthrough allows VPN tunneling using IPSec, L2TP or PPTP protocols to pass through the IP1000.

- **IPSec Passthrough –** IPSec (Internet Protocol Security) is a suite of protocols used to implement secure exchange of packets at the IP layer.
- **L2TP Passthrough –** Layer 2 Tunneling Protocol is the method used to enable Point-to-Point Protocol (PPP) to be tunneled through an IP network.
- **PPTP Passthrough –** PPTP (Point-to-Point Tunneling Protocol) Passthrough allows the Point-to-Point Protocol (PPP) to be tunneled through an IP network.

Access Restriction



Internet Access Policy

The Internet Access Policy screen allows you to block or allow specific kinds of Internet applications and traffic such as Internet access, designated services, websites, and inbound traffic during specific days and times.

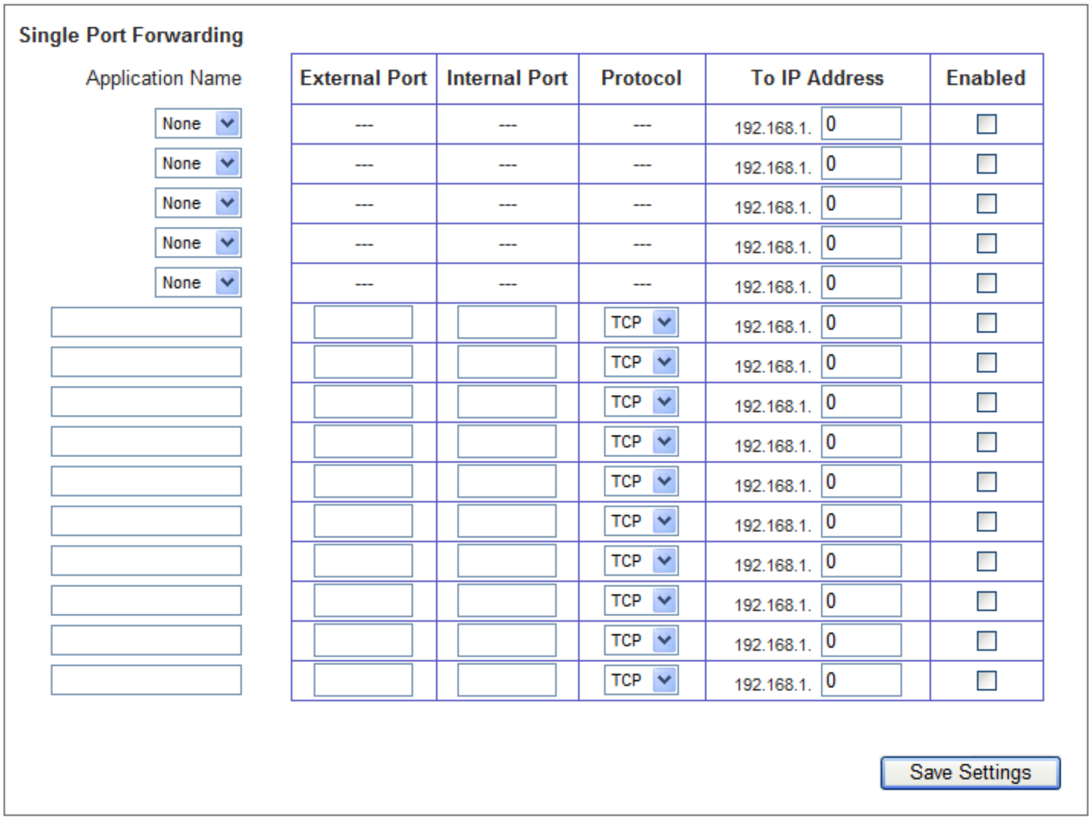
- **Access Policy –** Access can be managed by a policy by using the settings on this screen to establish an access policy. Selecting a policy from the drop-down menu will display that policy's settings. To delete a policy, select that policy's number and click the **Delete This Policy** button. To view all the policies, click the **Summary** button.
- **Policy Table –** On the Summary screen, the policies are listed with the following information: No., Policy Name, Access, Days, Time, and status (Enabled). To enable a policy, click the **Enabled** checkbox. To delete a policy, click its **Delete** button. Click the **Save Settings** button to save your changes, or click the **Cancel Changes** button to cancel your changes. To return to the Internet Access Policy screen, click the **Close** button.
- **Status –** Policies are disabled by default. To enable a policy, select the policy number from the drop-down menu. And click the radio button beside Enabled.

To create a policy:

- Select a number from the Access Policy drop-down menu.
- Enter a Policy Name.
- Enable this policy by checking the **Enabled**.
- Click the **Edit List** button to select the PCs to be affected by the policy. The List of PCs screen will appear. You can select a PC by MAC address or IP address. You can also enter a range of IP addresses if you want this policy to affect a group of PCs.
- Select **Deny** or **Allow** to block or allow Internet access for the PCs you listed on the screen.
- Decide the days and times you want this policy to be enforced. Select the individual days during which the policy will be in effect, or select **Everyday**.
- Enter a range of hours and minutes during which the policy will be in effect, or select **24 Hours**.
- To block websites with specific URL addresses, enter URL address in a separate field next to Website Blocking.
- To block websites using specific keywords, enter each keyword in a separate field next to Website Blocking.
- To filter access to various services over the Internet, choose the access to be blocked such as FTP or Telnet. Up to three kinds of access methods can be blocked per each policy.
- From the Applications list, select the applications to be blocked. Then click the >> (move right) button to move it to the Blocked List. To remove an application from the Blocked List, select it and click the << (move left) button.
- To add an application to block or to edit a service's settings, enter the application's name and its range in the Port Range fields, select its protocol from the Protocol drop-down menu. Then click the **Add** button.
- To modify a service, select it from the Application list. Then click the **Delete** button.
- Click the **Save Settings** button to save the policy's settings.

Application Forwarding

Single Port



On this screen, forwarding applications per port basis to specified network servers is customized. Once configured, the requests received from Internet for the configured application and through the specified port will be forwarded to the appropriate servers (computers). Before using forwarding, static IP addresses should be assigned to the designated servers (use the DHCP Reservation feature on the Networking/TCP/IP Settings screen).

Single Port Forwarding

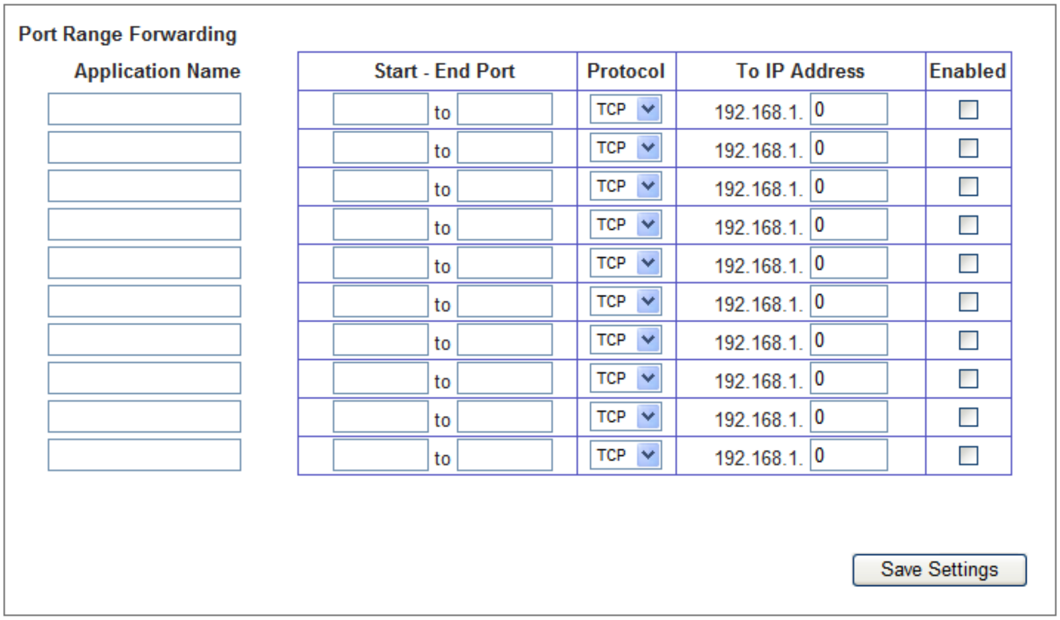
Common applications are available for the first five entries. Select the appropriate application, then enter the IP address of the server that should receive these requests. Click the **Enabled** checkbox to activate this entry

For additional applications, complete the following fields:

- **Application Name –** Enter the name of the application.
- **External Port –** Enter the external port number used by the server or Internet application. Check the Internet application documentation for more information.
- **Internal Port –** Enter the internal port number used by the server or Internet application. Check the Internet application documentation for more information.
- **Protocol –** Select the protocol **TCP** or **UDP,** or select **Both**.
- **To IP Address –** Enter the IP address of the server that should receive the requests.

- **Enabled –** Click the **Enabled** checkbox to enable the applications you have defined. This is disabled (unchecked) by default.

Application



Application screen sets up public services on local network, such as web servers, ftp servers, e-mail servers or other specialized Internet applications. (Specialized Internet applications are any applications that use Internet access to perform functions such as videoconferencing or online gaming. Some Internet applications may not require any forwarding.)

When the types of requests for configured applications are received via Internet, IP1000 will forward those requests to the appropriate servers (computers). Before using forwarding, assigning static IP addresses to the designated servers (use the DHCP Reservation feature on the Networking/TCP/IP Settings screen) is recommended.

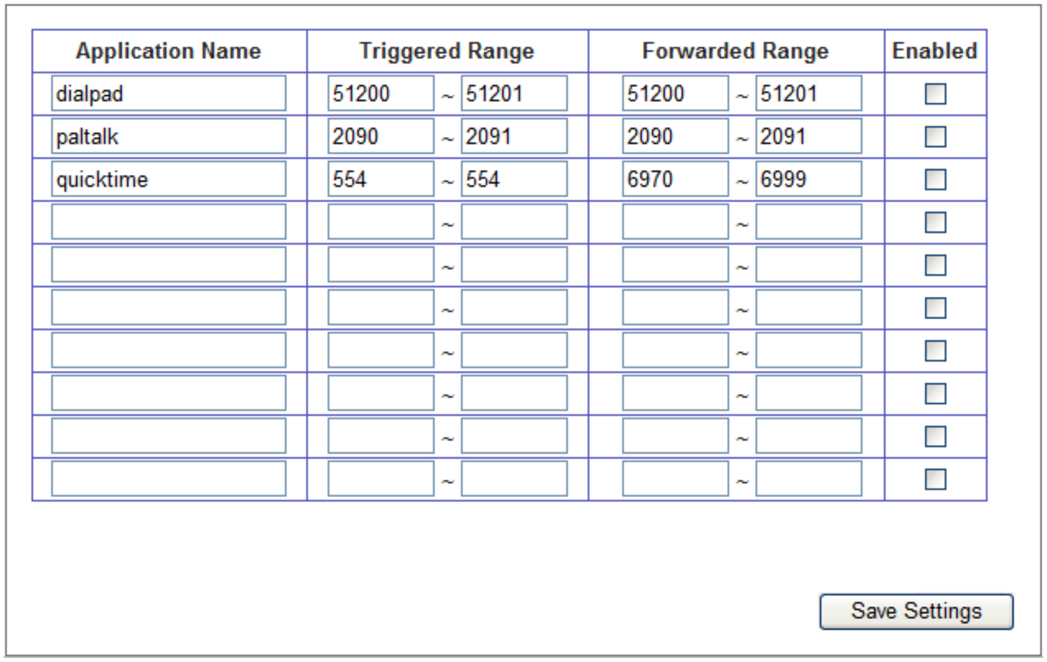
If you need to forward all ports to one PC, using DMZ is recommended.

Port Range Forwarding

To add an application, complete the following fields:

- **Application Name –** Enter the name of the application
- **Start – End Port –** Enter the number or range of port(s) used by the server or Internet application. Check with the Internet application documentation for more information.
- **Protocol –** Select the protocol **TCP** or **UDP**, or select **Both**.
- **To IP Address –** Enter the IP address of the server that allows Internet users to access.
- **Enabled –** Click the **Enabled** checkbox to enable the applications you have defined. This is disabled (unchecked) by default.

Port Range

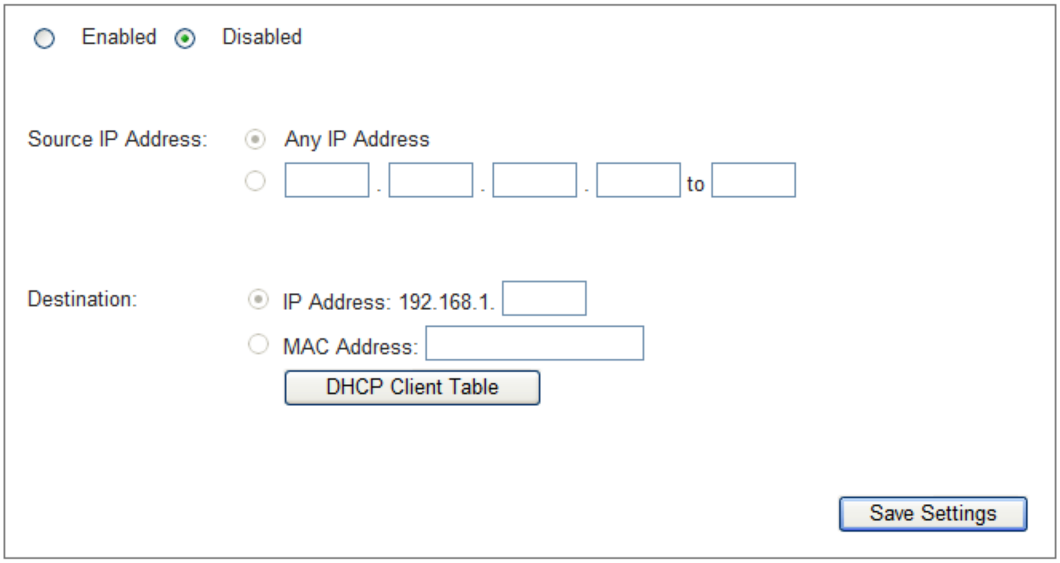


This screen instructs the IP1000 to watch outward data for specific port numbers. The IP address of the computer that matched is remembered by IP1000, so that when the requested data returns through the IP PBX, the data is sent to the proper computer by way of IP address and port mapping rules.

To add an application, complete the following fields:

- **Application Name –** Enter the name of the application
- **Triggered Range –** Enter the range of port numbers. Check with the Internet application documentation for the port number(s) needed.
- **Forwarded Range –** Enter the range of the forwarded port numbers. Check with the Internet application documentation for the port number(s) needed.
- **Enabled –** Click the **Enabled** checkbox to enable the applications you have defined. This is disabled (unchecked) by default.

DMZ



The DMZ screen allows one local user to be exposed to the Internet for use of a special purpose service such as Internet gaming and videoconferencing. DMZ hosting forwards all the ports at the same time to one PC. The Port Range Forwarding is more secure because only the configured ports are opened. If DMZ hosting opens all the ports of one computer, it exposes the computer to the Internet.

Any PC whose port is being forwarded must have its DHCP client function disabled and should have a new static IP address assigned to it because its IP address may change when using the DHCP function.

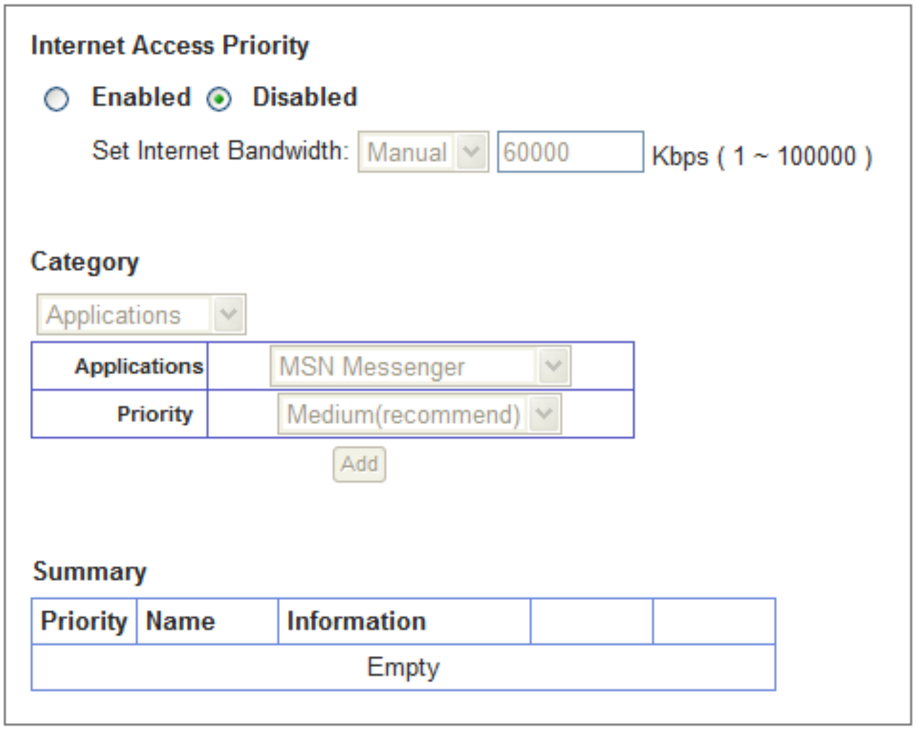
To use this feature, select **Enabled**. To disable DMZ hosting, select **Disabled**.

- **Source IP Address –** If any IP address can be the source, select **Any IP Address**. Want to specify an IP address or a range of IP addresses as the designated source, click the second radio button and enter the IP address.
- **Destination –** Want to specify the DMZ host by IP address, select **IP Address** and enter the IP address. Or want to specify the DMZ host by MAC address, select **MAC Address** and enter the MAC address. To retrieve this information, click the **DHCP Client Table** button.

The DHCP Client Table lists computers and other devices that have been assigned IP addresses by the IP1000. The Client table is sorted by Client Name, Interface, IP Address, MAC Address and Expired Time (how much time is left for the current IP address). To select a DHCP client, click the **Select** button. To retrieve the most up-to-date information, click the **Refresh** button. To exit this screen and return to the DMZ screen, click the **Close** button.

QoS

Quality of Service (QoS) ensures better service to high-priority types of network traffic, which may involve demanding and real-time applications such as videoconferencing.



Internet Access Priority

Select **Enabled** to use the QoS policy and system will allow user to setup the QoS policy.

Administrators can choose to manually set the Internet bandwidth or let system to determine it automatically.

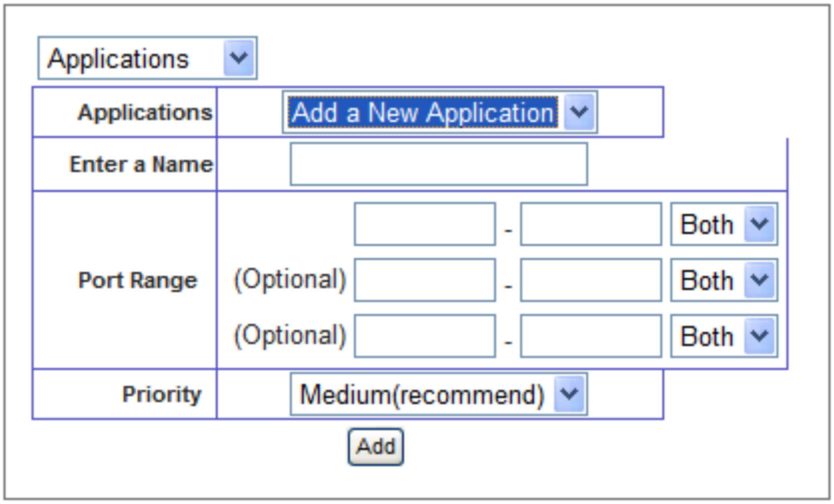
Category

There are four categories available. Select one of the following: Applications, Online Games, MAC Address, Ethernet Port, or Voice Device. In this section, you can select the bandwidth priority for a variety of applications and devices. There are four levels priority: High, Medium, Normal and Low. When setting priority do not set all applications to **High**, because this will defeat the purpose of allocating the available bandwidth. Select **Low** for those require normal bandwidth. A few attempts to establish the appropriate bandwidth priority may be required. It depends on the application.

Applications

- **Applications –** Select the appropriate application, If you select **Add a New Application –** follow the **Add a New Application** Instructions.
- **Priority –** Select the appropriate priority: **High**, **Medium**, **Normal**, or **Low**.

Add a New Application:



- Enter a Name – Enter any name to indicate the name of the entry.
- **Port Range –** Enter the port range that the application will use. For example, if administrators want to allocate bandwidth for FTP, enter 21-21; if need services for an application that uses from 1000 to 1250, then enter 1000-1250. There are totally up to three ranges to define for this bandwidth allocation. Port numbers can range from 1 to 65535. Check your application's documentation for details on the service ports used.
- **Protocol –** Select the protocol **TCP** or **UDP**, or select **Both**.
- **Priority –** Select the appropriate priority: **High**, **Medium**, **Normal**, or **Low**.

Online Games

- **Select a Game –** Select the appropriate game.
- **Priority –** Select the appropriate priority: **High**, **Medium**, **Normal**, or **Low**.
- Click the **Add** button to save your changes. New entry will appear in the Summary List

MAC Address

- **Enter a Name –** Enter a name for your device.
- **Mac Address –** Enter the MAC address of your device
- **Priority –** Select the appropriate priority: **High**, **Medium**, **Normal**, or **Low**.
- Click the **Add** button to save your changes. New entry will appear in the Summary List

Ethernet Port

- **Ethernet –** Select the appropriate Ethernet port.
- **Priority –** Select the appropriate priority: **High**, **Medium**, **Normal**, or **Low**.
- Click the **Add** button to save your changes. New entry will appear in the Summary List

Voice Device.

- **Enter a Name –** Enter a name for voice device.
- **Mac Address –** Enter the MAC address of your voice device.

- **Priority –** Select the appropriate priority: **High**, **Medium**, **Normal**, or **Low**.
- Click the **Add** button to save your changes. New entry will appear in the Summary List

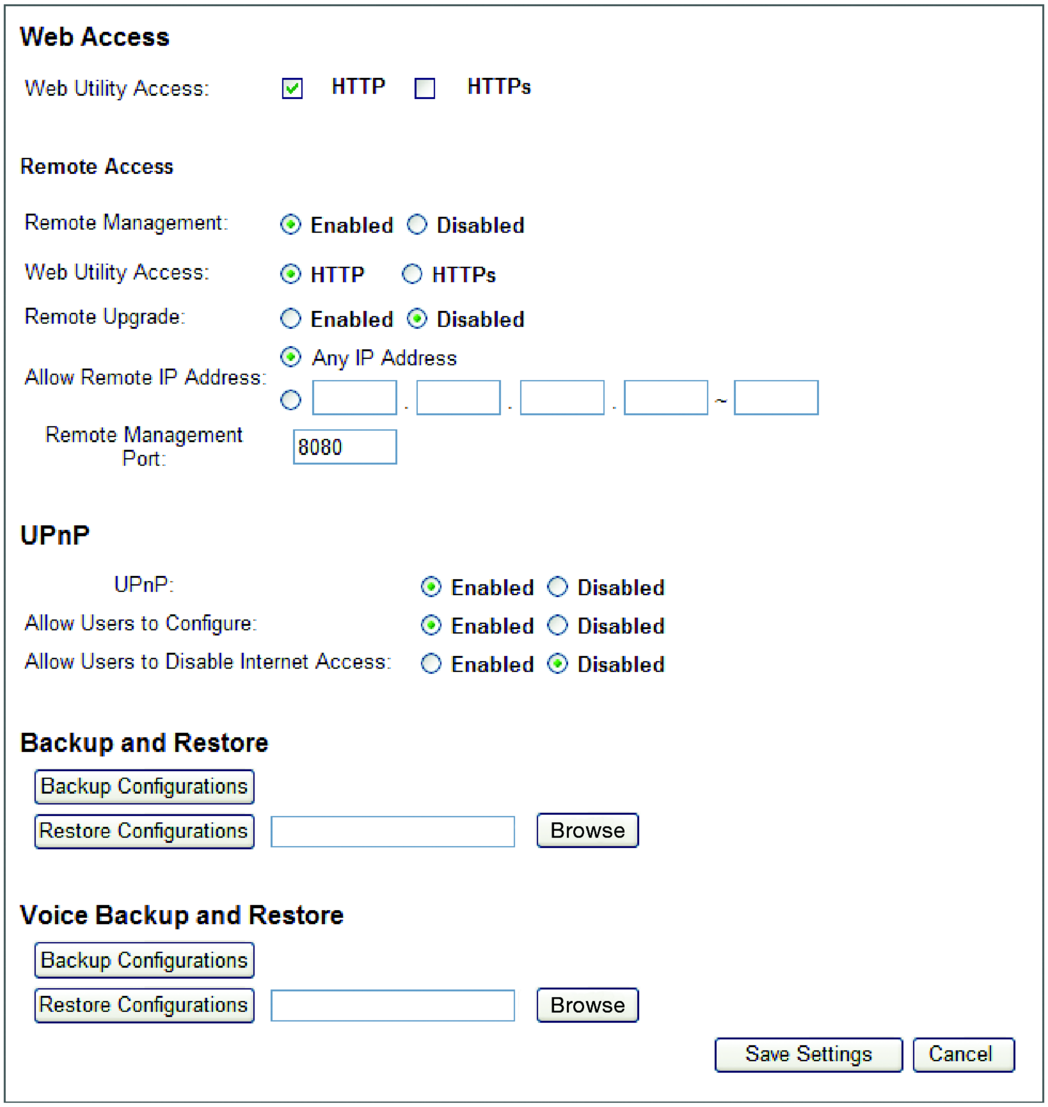
Summary

This lists the QoS entries you have created for your applications and devices

- **Priority –** This displays the bandwidth priority of High, Medium, Normal, or Low.
- **Name –** This displays the application, device, or port name.
- **Information –** This displays the port range or MAC address entered for your entry. If a pre-configured application or game was selected, there will be no valid entry shown in this section.
- **Remove –** Click this button to remove an entry.

Administration

Administration



The Administration screen allows user to change the IP PBX's access settings and configure the UPnP (Universal Plug and Play) features as well as to backup and restore the IP PBX's configuration data.

Web Access

- **Web Utility Access –** HTTP (HyperText Transport Protocol) is the communications protocol used to connect to servers on the World Wide Web. HTTPS uses SSL (Secured Socket Layer) to encrypt transmitted data for higher security. IP1000 supports two types of protocols, **HTTP** or **HTTPS,** for web access.

Remote Access

Settings for this field can only be configured from LAN network.

- **Remote Management – If** remote access to the IP1000 from outside the local network is permitted, choose **Enabled**. Otherwise, keep the default setting, **Disabled**.

- **Web Utility Access –** HTTP (Hyper Text Transport Protocol) is the communications protocol used to connect to servers on the World Wide Web. HTTPS uses SSL (Secured Socket Layer) to encrypt transmitted data for higher security. IP1000 supports two types of protocols, **HTTP** or **HTTPS,** for web access.
- **Remote Upgrade –** If remote upgrade from outside the local network is allowed, select **Enabled**. (You must have the **Remote Management** feature enabled as well.) Otherwise, keep the default setting, **Disabled**.
- **Allow Remote IP Address –** If allow remote IP address from outside the local network is allowed, select **Any IP Address**. If administrators want to specify an external IP address or a range of IP addresses, then select the second option and complete the fields provided.
- **Remote Management Port –** Enter the port number that will be open to outside access.

UPnP

Universal Plug and Play (UPnP) allows Windows system to automatically configure the IP PBX for various Internet applications, such as gaming and videoconferencing.

- **UPnP –** To use UPnP, keep the default setting, **Enabled**. Otherwise, select **Disabled**.
- **Allow Users to Configure –** Select **Enabled** if users are allowed to configure manually while using the UPnP feature. Otherwise, keep the default setting, **Disabled**.
- **Allow Users to Disable Internet Access –** Select **Enabled** if users are allowed to configure to prohibit all Internet connections. Otherwise, keep the default setting, **Disabled.**

Backup and Restore

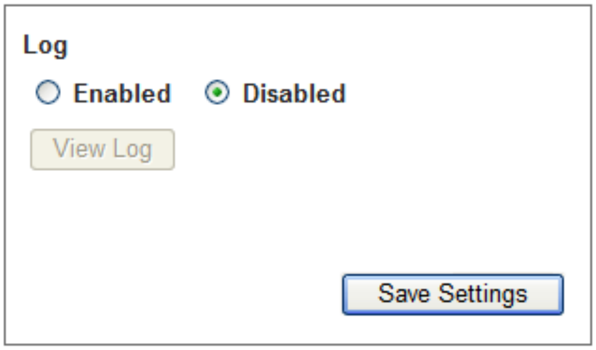
- **Backup Configurations –** To back up the IP1000 network configuration settings, click this button and follow the on-screen instructions.
- **Restore Configurations –** To restore the IP1000 network configuration settings, click this button and follow the on-screen instructions. (You must have previously backed up the IP1000 network configuration settings.)

Voice Backup and Restore

- **Backup Configurations –** To back up the IP1000 PBX configuration settings, click this button and follow the on-screen instructions.
- **Restore Configurations –** To restore the IP1000 PBX configuration settings for on-board analog devices, click this button and follow the on-screen instructions. (You must have previously backed up the IP1000 PBX configuration settings.)

Log

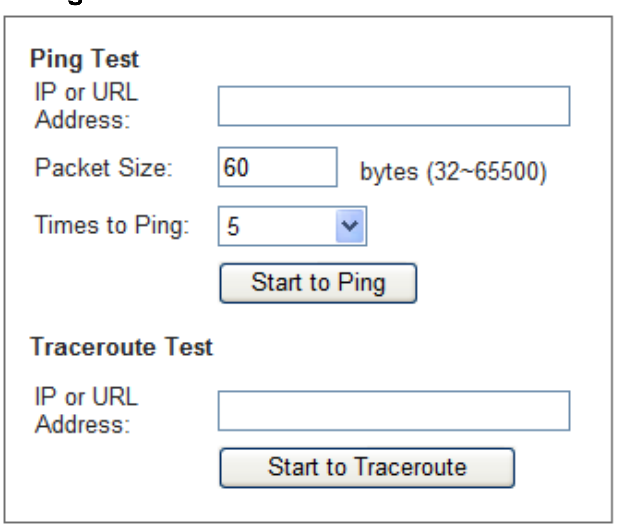
The Log screen provides you with a log of all incoming and outgoing URLs or IP addresses for your Internet connection.



- **Log –** To access activity logs, select the **Enabled** radio button. While logging is enabled, users can choose to view temporary logs. Click the **Disabled** button to disable this function
- **View Log –** To view the logs, click **View Log**, a new screen will appear with logged information shown on it. Four types of logging are supported, **Incoming Log**, **Outgoing Log**, **Security Log** or **DHCP Client Log**, choose one from the Type drop-down menu.
    - **Incoming Log** –The Incoming Log displays a temporary log of the source IP addresses and destination port numbers for the incoming Internet traffic.
    - **Outgoing Log** –The Outgoing Log displays a temporary log of the local IP addresses, destination URLs/IP addresses, and service/port numbers for the outgoing Internet traffic.
    - **Security Log** –The Security log displays the login information for the Web-based Utility.
    - **DHCP Client Log** – The DHCP Client Log displays the LAN DHCP server status information.

Click the **Refresh** button to update the log. Click the **Clear Log** button to clear all the information that is displayed. Click the **Close** button to close the log window.

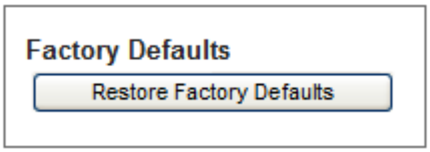
Diagnostics



The diagnostic tests (Ping and Traceroute) allow you to check the connections of your network devices including the connection to the Internet.

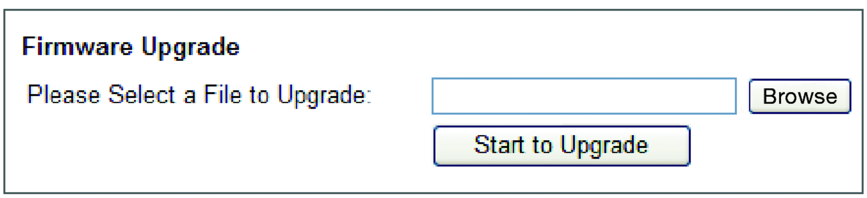
- **Ping Test** – The Ping test will check the status of a connection. Enter the IP address or URL of the PC whose connection you wish to test, the packet size (default is **60** bytes), and how many times you wish to test it. Then, click the **Start to Ping** button. The Ping screen will then display the test results. Click the **Close** button to return to the Diagnostics screen.
- **Traceroute Test** – To test the performance of a connection, enter the IP address or URL of the PC whose connection and click the **Start to Traceroute** button. The Traceroute screen will then display the test results. Click the **Close** button to return to the Diagnostics screen.

Factory Defaults



- **Factory Defaults** – The Factory Defaults screen allows administrators to restore the IP PBX's configuration to its factory default settings.
- **Restore Factory Defaults** – To clear all of the IP PBX's settings and reset them to its factory defaults, click the **Restore Factory Defaults** button.

Firmware Upgrade

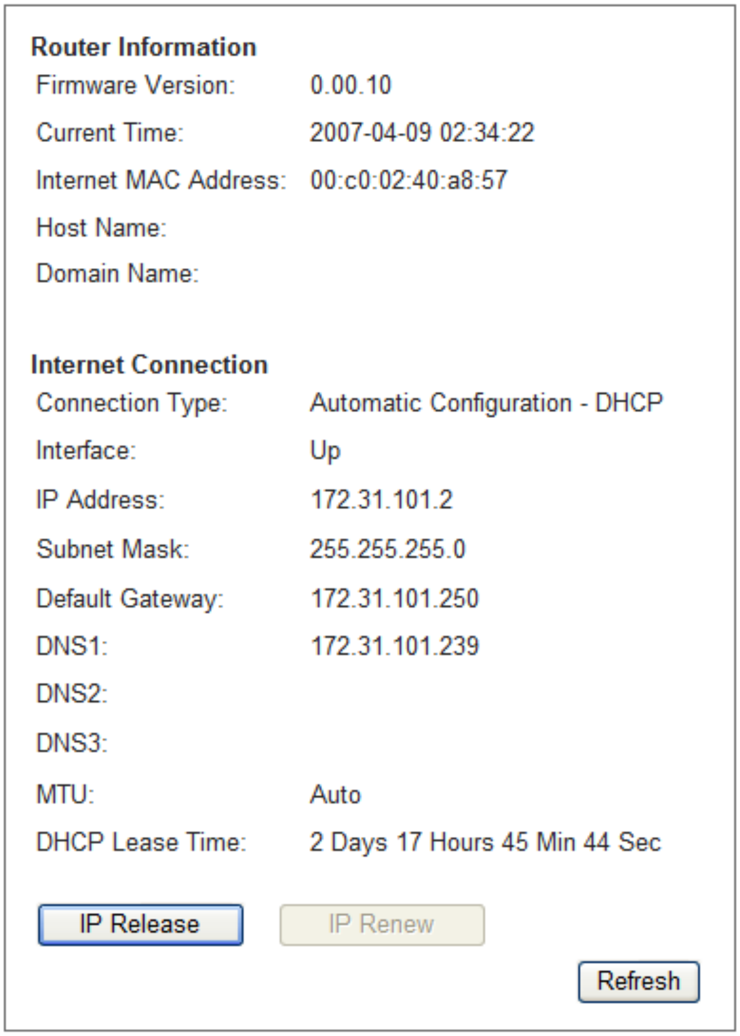


The Firmware Upgrade screen allows you to upgrade the IP1000's firmware.

- **Please Select a File to Upgrade** – Enter the name of the new firmware file, or click the **Browse** button to find this file.
- **Start to Upgrade** – After the appropriate file is selected, click this button and follow the on-screen instructions to perform firmware upgrading

Status

The Status screen displays information about the Routing function in the IP1000 and its current settings. The on-screen information will vary depending on the Internet Connection Type selected on the **TCP/IP Settings** screen.



Router Information

- **Firmware Version** – This shows version number of the IP1000's firmware
- **Current Time** – This shows the time set on the IP1000
- **Internet MAC Address** – This is the IP1000's MAC address.
- **Host Name** – The Host Name entered when set TCP/IP Settings screen.
- **Domain Name** – The Domain Name entered when set TCP/IP Settings screen

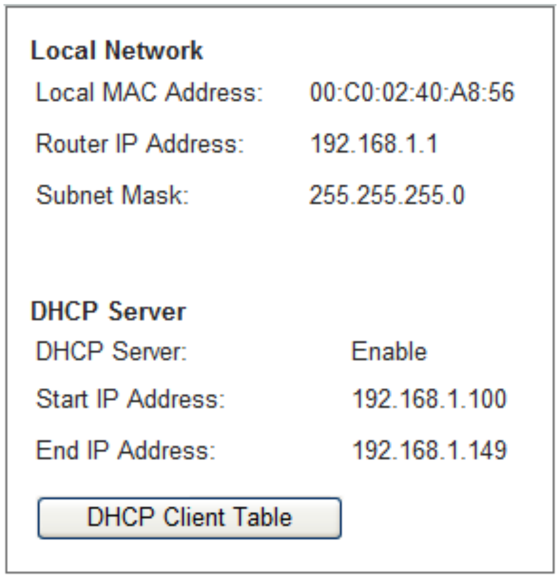
Internet Connection

- **Connection Type** – This indicates the type of Internet connection you are using. For dial-up style connections such as PPPoE or PPTP, there is a **Connect** button to re-establish the Internet connection if there is no connection.
- **Interface** – This indicates the Internet connection of the IP PBX, up or down.
- **IP Address** – Show IP1000's Internet IP address.
- **Subnet Mask** and **Default Gateway** – The IP PBX's Subnet Mask and Default Gateway address are displayed here for DHCP and static IP connections.
- **DNS1-3** – Show the DNS (Domain Name System) IP addresses currently used by

the IP1000.

- **MTU** – Show the MTU (Maximum Transmission Unit) setting for the IP1000.
- **IP Release** – It is available for a DHCP connection, click this button to release the current IP address got from DHCP server.
- **IP Renew** – It is available for a DHCP connection, click this button to release the current IP address and get a new IP address from DHCP server.

Local Network



The local Network screen displays the information about the local network.

Local Network

- **Local MAC Address** – The MAC Address of the IP1000 for local interface.
- **Router IP Address** – This shows the IP address used by IP1000 for appearing on local network.
- **Subnet Mask** – The IP1000's Subnet Mask is shown here.

DHCP Server

- **DHCP Server** – Display the status of the IP1000 embedded DHCP server.
- **Start IP Address** – The starting IP address of the range of IP addresses are used by DHCP server for being assigned to devices on local network.
- **End IP Address** –The ending IP address of the range of IP addresses are used by DHCP server for being assigned to devices on local network.

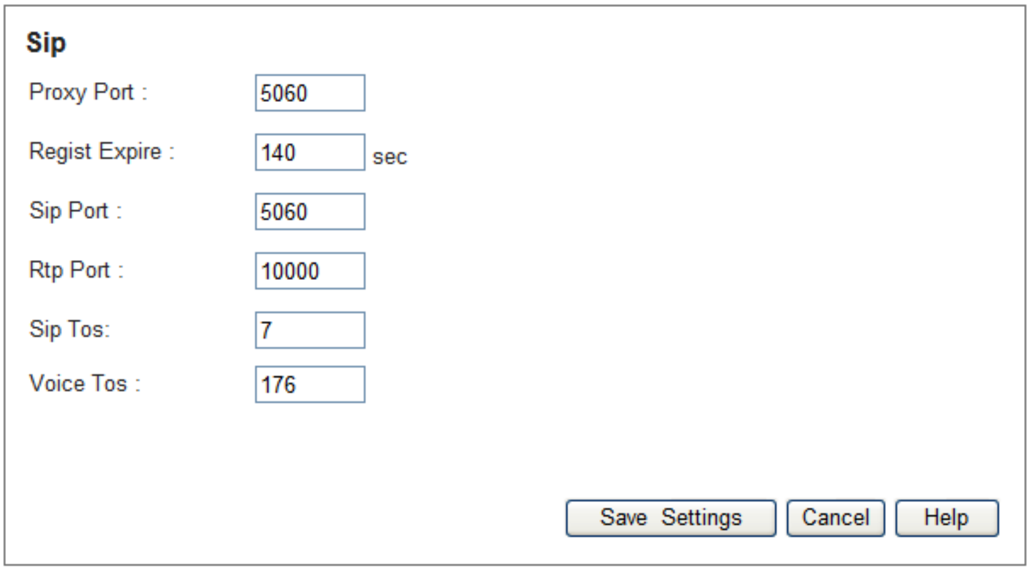
DHCP Client Table

Click the **DHCP Clients Table** button to view the DHCP Client Table. It lists computers and other devices that have been assigned IP addresses by the IP1000 embedded DHCP server. Sorting by Client Name, Interface, IP Address, MAC Address or Expired Time (how much time is left for the current IP address) is supported. To remove a DHCP client, select it and click the **Delete** button. To retrieve the most up-to-date information, click the **Refresh** button. To exit this screen and return to the Local Network screen, click the **Close** button.

Analog Interface

The Analog Interface screen is used to configure the on-board analog ports. In IP1000, it includes 2 FXS ports and 6 FXO ports. Each analog port uses SIP protocol to register to the SIP proxy server embedded in IP1000

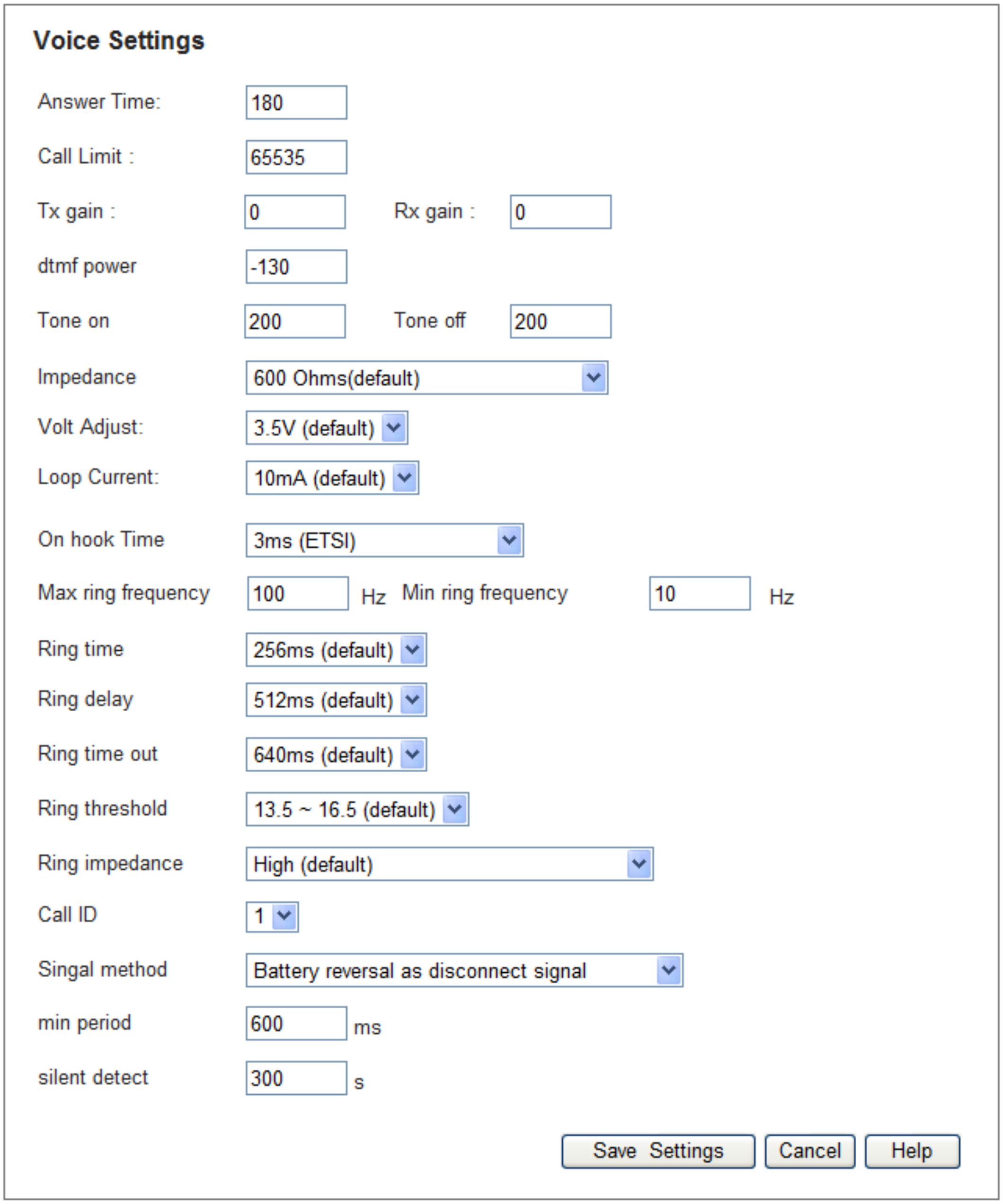
SIP Settings



This screen lets you configure the SIP server and the related parameters that the analog ports will register to. The SIP server address is the same gateway address of IP PBX, so you don't need to specify the SIP server address.

- **Proxy Port** – The port used for initiating connections to the SIP server, the default port number is 5060
- **Regist Expire** – It is a timer for monitoring the registration to embedded SIP proxy server in IP1000. When connection is idle for over the Register Expire time, the connection will be terminated automatically. The default value for this item is 120 seconds.
- **SIP Port** – The UDP port number that the analog ports use for incoming call setup request. The default value is 5060
- **RTP Port** – The base UDP port that the analog ports uses for transmitting RTP and RTCP packets. The analog ports use a block of port numbers for sending/receiving RTP and RTCP packets from this port number. The default value is 10000.
- **SIP TOS** – TOS field in IP header used in outgoing SIP packets. The default value is 7.
- **Voice TOS** – TOS field in IP header used in outgoing RTP/RTCP packets. The default value is 176.

Voice



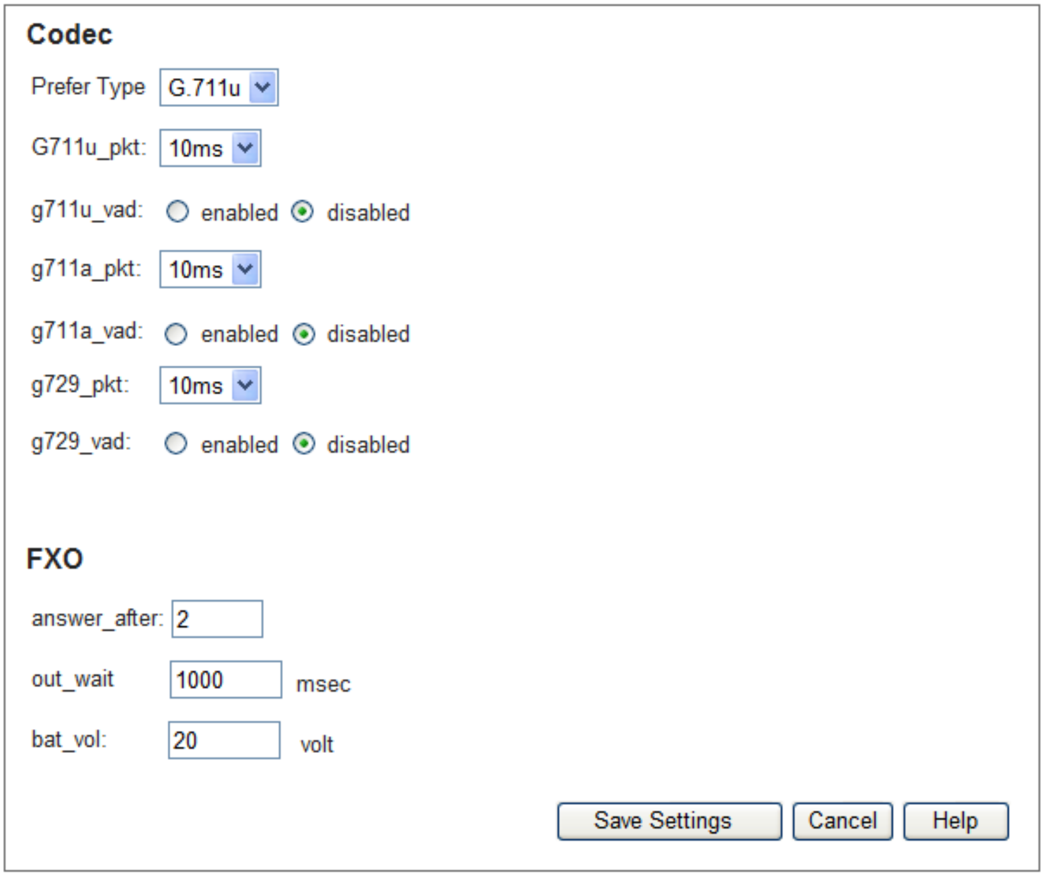
The Voice Settings screen is for selecting and configuring the FXO line settings.

- **Answer Time** – Specify the time in seconds that the analog ports wait for the called party to answer the call. If the called party does not answer the call within this time period, the call is terminated automatically. The default value is 180 seconds.
- **Call Limit** – Specify the maximum number of seconds for a call conversation. When the duration of a call exceeds this value, the call is terminated automatically. The default value is 65535 seconds.
- **Tx Gain** – The FXO ports may increase or attenuate the power level before transmitting to the telephony port, changing gain level manually may be required. This field allows user to set the transmitter gain level in dB
- **Rx Gain** –The FXO ports may increase or attenuate the power level of the telephony

port, changing gain level manually may be required. This field allows user to set the receiver gain level in dB

- **DTMF Power** – Enter the desired value for the DTMF power that FXO ports dial toward PSTN. Each level for changing is 0.1dBm.The default value is -130*0.1 dB. This setting will only affect the DTMF tones sent by SIP INFO.
- **Tone On** – Specify the Tone-On time in millisecond for an out dialing DTMF digit. The default value is 200 milliseconds. This setting will only affect the DTMF tones sent by SIP INFO.
- **Tone Off** – Specify the Tone-Off time in millisecond for an out dialing DTMF digit. The default value is 200 milliseconds. This setting will only affect the DTMF tones sent by SIP INFO.
- **Impedance** – Select the impedance of the lines connecting to PSTN ports.
- **Volt Adjust** – Select the TIP/RING voltage adjust value, low-voltage countries should use a lower voltage.
- **Loop Current** – Select the minimum operational loop current at which the FXO ports will operate.
- **On Hook Time** – Select the amount of time to wait for the FXO port to go on-hook
- **Min Ring Frequency** – Enter the minimum ring frequency for the FXO port to detect. The default minimum ring frequency is 10Hz
- **Max Ring Frequency** – Enter the maximum ring frequency for the FXO port to detect. The default maximum ring frequency is 100Hz
- **Ring Time** – Select the amount of ringing time in millisecond that the FXO port detects to be a valid ring time.
- **Ring Delay** – Select the amount of time in millisecond as the duration starts when a ring signal is validated and till a valid ring signal is confirmed.
- **Ring Time Out** – Select the amount of time in millisecond for determining that ring signal is stopped.
- **Ring Threshold** – Select the minimum voltage level the incoming ringing signal must be presented with for the FXO port to detect it.
- **Ring Impedance** – Select the desired value to satisfy the maximum ringer impedance specification.
- **Call ID** – We don’t use this setting anymore, but still keep it in the configuration file for compatibility issues.
- **Signal Method** – Select the line disconnection signal method. Two options to be selected::
    - 1. Battery reversal as disconnect signal.
    - 2. Loop period shut-down as disconnect signal.
- **Min Period** – Enter the minimum period of time for the above hardware to receive a disconnect signal. The default value is 600 milliseconds.
- **Silent Detect** – The amount of time to wait for the FXO port to disconnect after not receiving RTP packets on the port. The default value is 300 seconds.

Codec



This screen is for selecting and configuring the voice codec, voice parameters.

Codec

- **Prefer Type** – Select this preferred voice codec that analog ports (FXO and FXS ports) used to negotiate with SIP sever for determining the voice codec. Available codecs are G.711u, G.711a, and G.729.
- **G711u_pkt** – Select this packetization time for G.711u. The packetization time is the duration that the analog port samples voice signal and compresses it into a packet before sending to the remote SIP device.
- **G711u_vad** – Select this button to enable or disable Voice Activity Detection for G.711u. This should be disabled for Asterisk application requirements.
- **G711a_pkt** – Select this packetization time for G.711a
- **G711a_vad** – Select this button to enable or disable Voice Activity Detection for G.711a. This should be disabled for Asterisk application requirements.
- **G729_pkt** – Select this packetization time for G.729.
- **G729_vad** – Select this button to enable or disable Voice Activity Detection for G.729. This should be disabled for Asterisk application requirements.

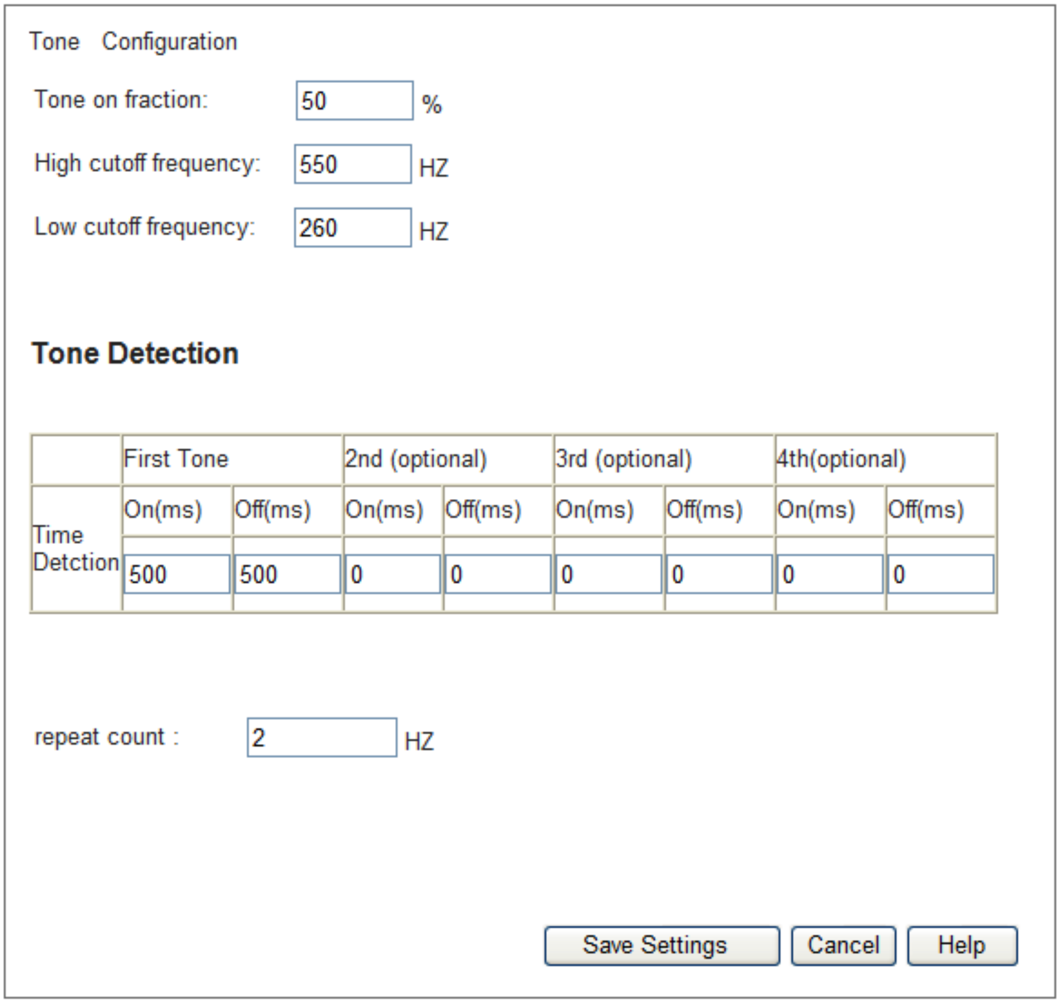
FXO

This portion is for configuring the FXO dial in/out parameters

- **Answer_after** – Input the number of rings that the FXO port will keep waiting before answering the incoming calls. The default value is 2 rings.

- **Out_wait** – Enter the time in milliseconds that the FXO port keeps waiting after seizing a telephony port and before dialing out DTMF signals. The default value is 1000 millisecond.
- **Bat_vol** – Before seizing a FXO port for dialing out, the FXO port detects voltage level on the port to ensure that the port is connected and available. If the voltage level is below this threshold level, the port is declared unavailable

Call Progress



This screen is for configuring the CP (Call Progress) tone detection for FXO port. For normal application, it is not recommended to modify the default settings.

- **Tone Detection** – This specifies the ON/OFF of CP tone (busy tone for general case) cadence periods. For instance, **On** of Tone 1 is 500 and **Off** of Tone 1 is 500 means the cadence is ON for 500ms then OFF for 500ms. If more than one ON/OFF cadence detections are required, for example, ON 300ms/OFF 400ms, then ON 500ms/OFF 600ms, then program **On** of Tone 1 to be 300, **Off** of Tone 1 to be 400, **On** of Tone 2 to be 500, **Off** of Tone 2 to be 600.
- **Repeat Count** – Input the minimum detection cycles for the above CP tone cadence.

- **Tone On Fraction** – Input the CP tone sensitivity level. This parameter controls the SNR and frequency offset. The default value is 50%.
- **High Cutoff Frequency** – Input the high cut-off frequency that the CP tone detection will take. The default value is 550 Hz
- **Low Cutoff Frequency** – Input the low cut-off frequency that the CP tone detection will take. The default value is 260 Hz.

Reset Analog

- The **Reset Analog** button allows administrators to reset the analog ports including all FXS and FXO ports. For some reason, the analog ports may lose their connection with the SIP server, in this case, the user can reset the analog ports to recover the connection.

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.

CO 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.

Connecting Phone Lines

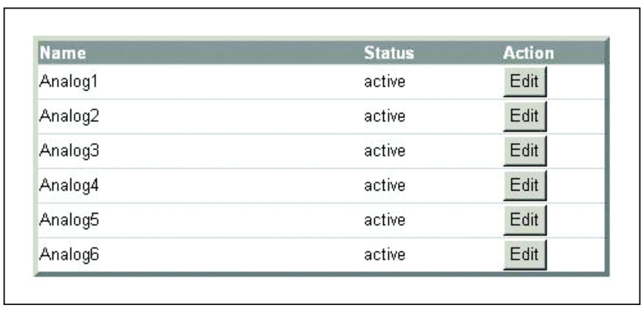
Before hardware trunks can be provisioned, they must be connected to the system.

Connecting Using Internal Analog Line Cards

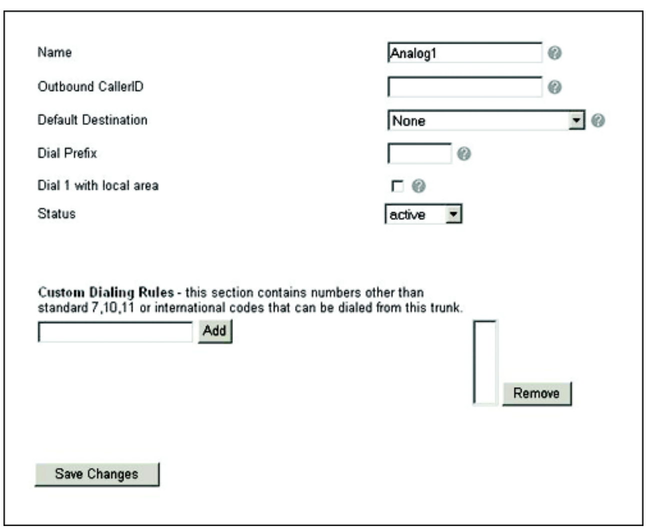
- **The VB400 Analog Line Card (4 PSTN line connections)** – Connect the phone lines to the RJ11 jacks at the rear of the IP1000 and start making calls. These connections are single pair; one line per jack.

Provisioning a New Hardware Trunk

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:



Name	Status	Action
Analog1	active	Edit
Analog2	active	Edit
Analog3	active	Edit
Analog4	active	Edit
Analog5	active	Edit
Analog6	active	Edit

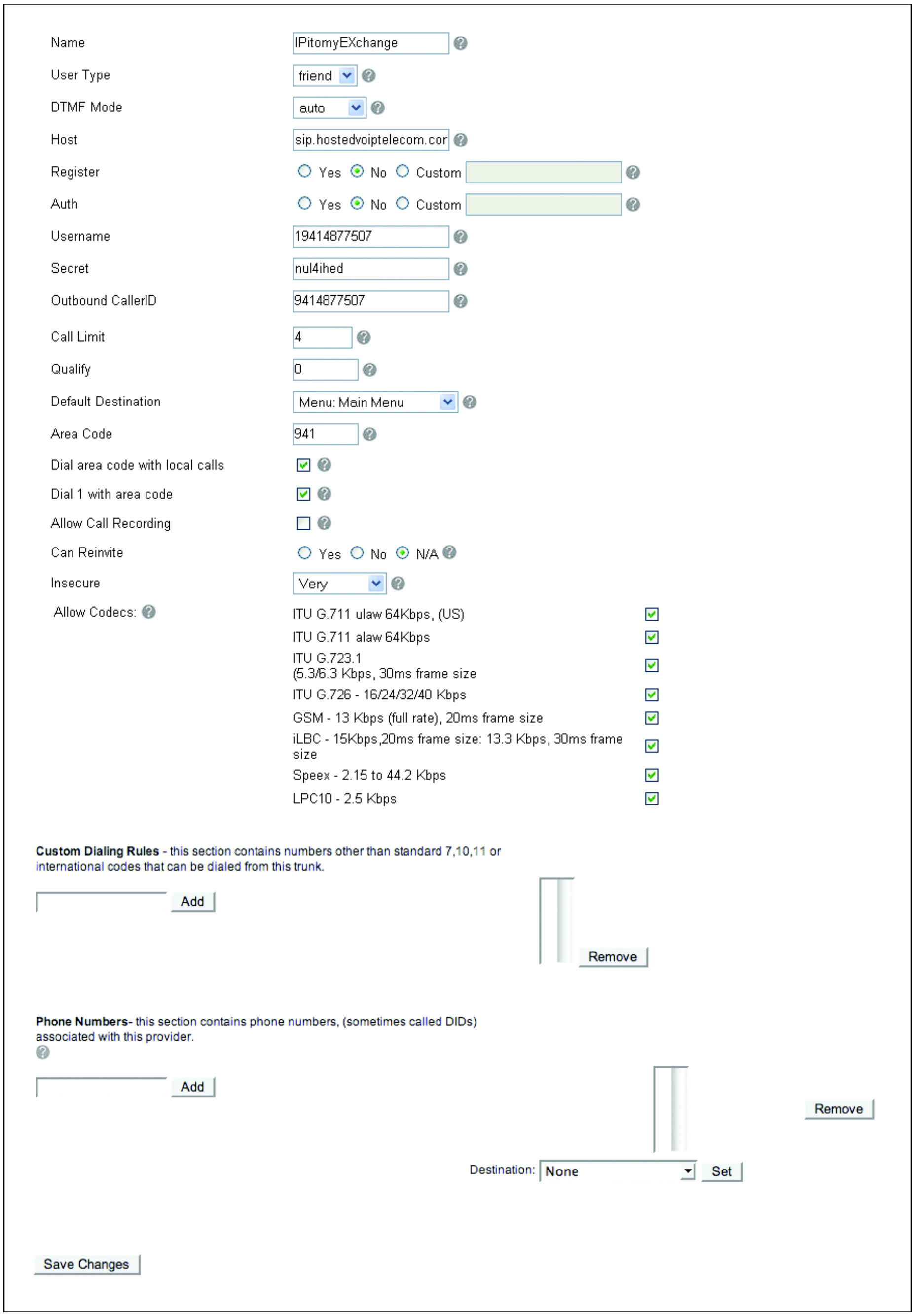


SIP Providers

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 **SIP Providers** 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. Click **Add New**. The Edit Hardware Provider page will appear:



3. Enter the **Name** of the service provider.

4. Select the **User Type**, options are friend, user and peer.

5. Select DTMF mode, options are rfc2833, auto, info and inband.

6. Enter URL or IP address for SIP provider **Host**.

7. Click **Yes** for **Register** option to enable registration to SIP provider if required.

8. Click **Yes** for **Auth** option to enable authentication with SIP provider if

required.

9. Enter **Username** and **Secret** (password) for authentication with SIP provider.
10. **Define the Outbound Caller ID** for calls routed through this trunk. This must be a ten-digit telephone number. This is the number those receiving the call will associate with the business.
11. **Create a Zap Group Number**. A Zap Group associates the telephone numbers with the service provider being provisioned. It tells the system on which trunk calls to and from these numbers should be routed.
12. Enter a number for **Call Limit** as the limit of simultaneous calls.
13. Enter a number for Qualify as the time in milliseconds for checking the availability of SIP provider periodically.
14. **Select a Default Destination** using the drop-down menu for calls to be routed to if they are not to be routed to a specific destination. In most cases the default destination will be an automated attendant, menu or a 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. 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. Once all of the destinations are created, they appear in a drop-down menu and are selectable.
15. Enter the **Area Code** of the trunk being provisioned.
16. Check **Dial Area Code With Local Calls** if the system should always dial an area code with outbound local calls. In some cases it may be best NOT to check this box.
17. Check **Dial 1 With Area Code** if the system should always dial “1” with the area code when making outbound calls. In some cases it may be best NOT to check this box. See Installation Note 1.

Installation Note 1

In some business locations it can be convenient to have the system add an area code and a “1” when placing a local or long distance call. However, in locations where businesses are close to several telecommunication transport areas it can be confusing. In these cases it is best to not check the boxes next to Dial Area Code With Local Calls and Dial 1 With Area Code. Example: Bradenton, Sarasota and Englewood Florida are all within 50 miles of each other. A business located in Bradenton Florida makes local calls by dialing a standard seven digit number (223-4567)*. Calls made from Bradenton to the neighboring city of Sarasota require the use of the area code or a ten digit number (941-223-4567)*.

Calls made from Bradenton to Englewood require the use of a “1” in front of the ten digit number (1-941-223-4567)*.

*These phone numbers are not real and are for demonstration purposes only.

18. Check **Allow Call Recording** if users on the system will be allowed to record calls. In a standard business application this feature is typically not needed.
19. Select the option to allow **Can Reinvite** or **Don't Care**.
20. Select a **Insecure** level, options have Port, Invite, Port+Invite, Yes and Very.
21. Check options for **Allow Codecs** to enable the Codecs to be used for this SIP provider.
22. 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.
23. 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 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 or get information.

Installation Note 2

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 person 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.

24. Click the **Save Changes** button to save changes made to the hardware provider setup.

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

Edit an Existing SIP Provider

26. Click on **Providers** and **SIP Providers**. The SIP Providers page will appear.

27. Select an entry from the list on the page.

28. Click **Edit**. The Edit SIP Providers page will appear.

29. Make changes.

30. Click **Save Changes**.

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

Delete an Existing SIP Provider

32. Click on **Providers** and **SIP Providers**. The SIP Providers page will appear.

33. Select an entry from the list on the page.

34. Click **Delete**. The SIP Provider will be removed from the list on the page.

Destinations

Destinations are the various places that a call can be routed to within the system. Destinations in the IP1000 include:

- **Extensions** are individual extensions with a telephone. When an extension is created, a voicemail box is also created.
- **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 a caller 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.
- **Schedules** route callers to different destinations or people 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.

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 **IP1000 Installation Worksheet** helps organize provisioning information prior to installation.

Extensions

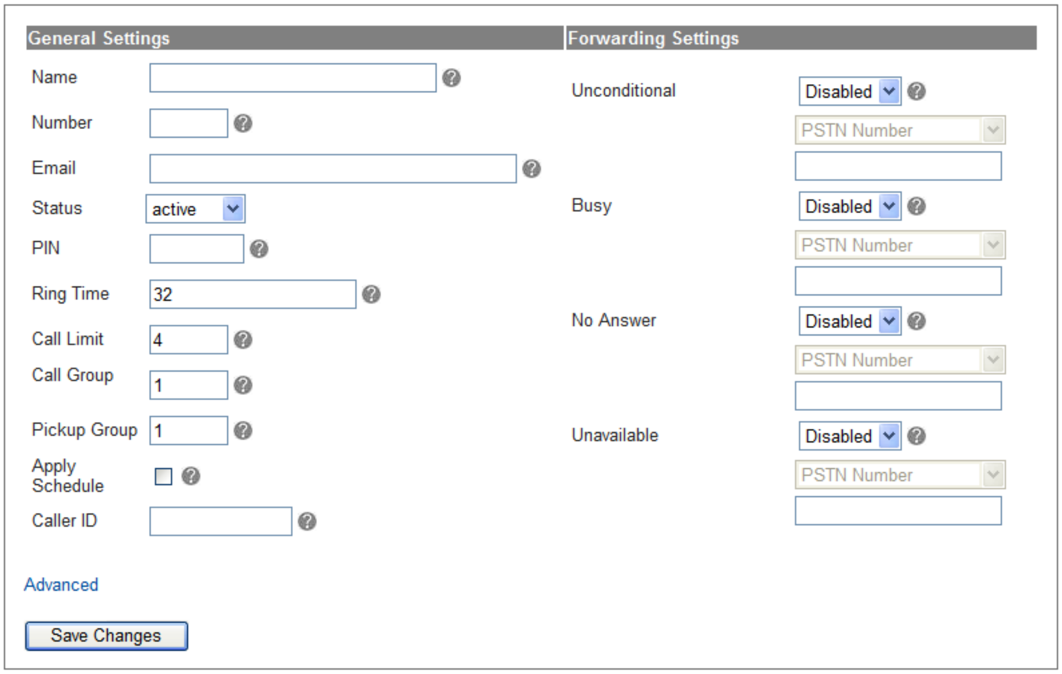
Extensions define where specific people or departments can be reached in an organization. They should be setup first in the system. Once extensions are setup then the rest of the application can be provisioned. Extensions have six groups of settings that need to be established:

- **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 IP1000. 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 IP1000 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.

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. Click **Add New**. The Edit Extensions page will appear. Note that each new extension added automatically has a voice mailbox created.



3. Insert the **Name** or department associated with the extension being created.

4. Create 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.

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

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

7. **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.

8. **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.

9. **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.

10. **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.

11. **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.

12. 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.

13. Enter **Caller ID**. The numbers programmed will be carried to called destination as the caller ID.

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 IP1000).

Provisioning Forward Settings

To provision forward settings:

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

2. Select **Enabled** or **Disabled**. Disabled turns the forward setting off. Enabled

turns the forward setting on.

3. If the Forward setting is **Enabled**, you can choose to select a destination from the drop-down list. The IP1000 allows calls to be forwarded to a public switched telephone number (PSTN). Forwarding calls to a PSTN number 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.

- Dial ***90** to disable forwarding.
- Dial ***91** to enable forwarding.
- 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.

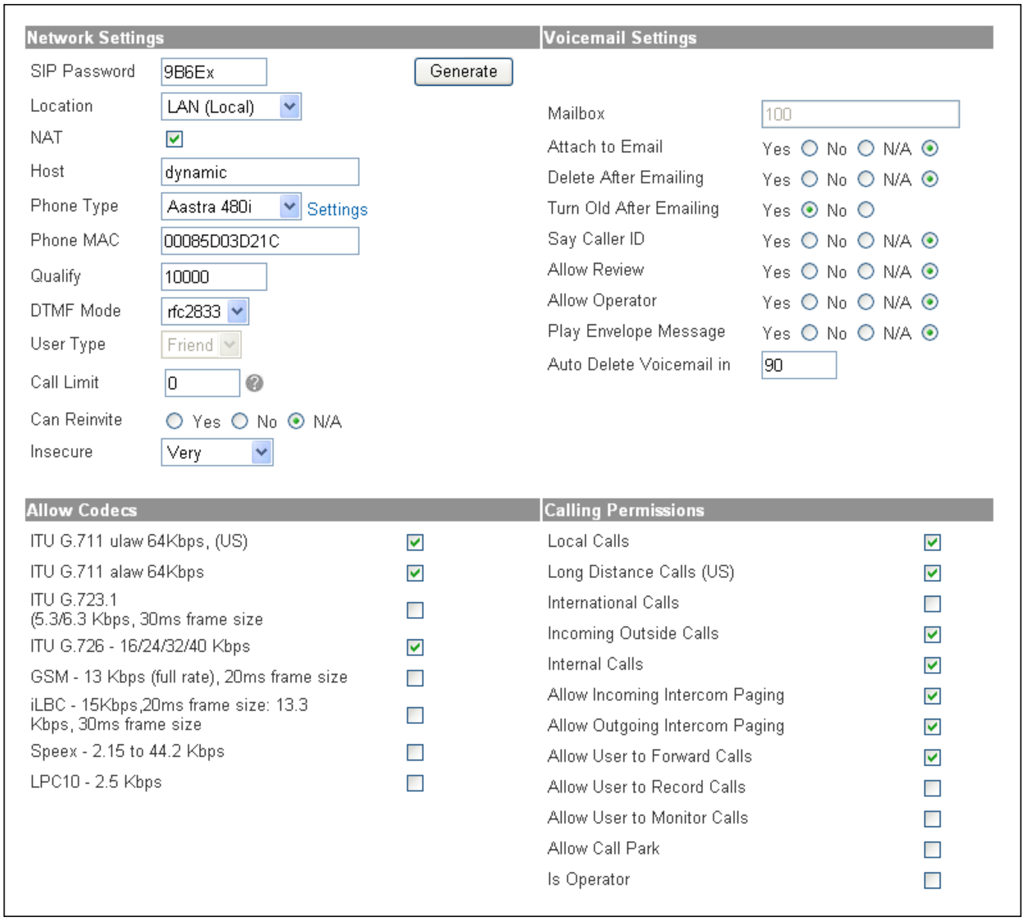
When it is necessary to modify the forwarding setting while away from the office, the IP1000 has a forwarding application built into the system. It is necessary to have an automated attendant menu accessible from outside the system. The forwarding gateway is selectable as an option from the menu. When 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.

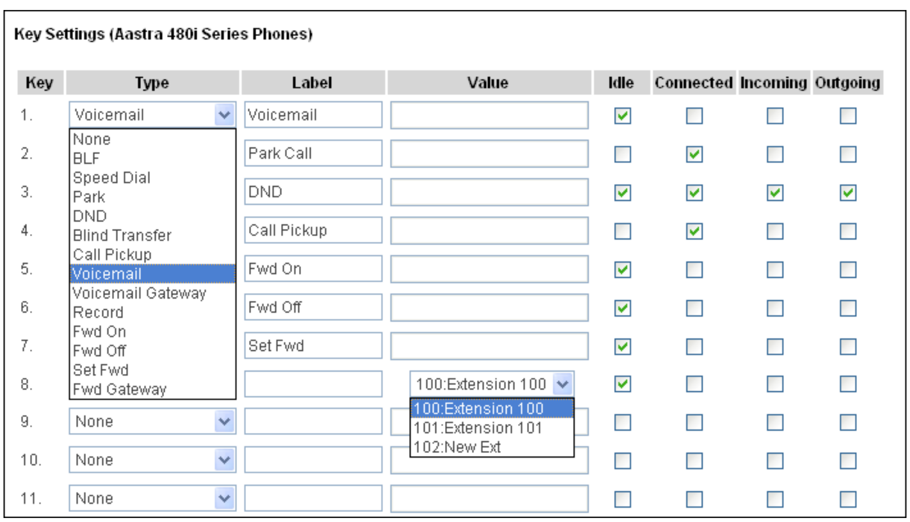
1. **Call into the Automated Attendant** (menu).
2. **Select the touch-tone digit** that has been set for modifying forwarding settings.
3. The system will prompt for an **Extension Number** and **Password**.
4. The system will indicate if extension forwarding is **Enabled** or **Disabled**.
5. Pressing “**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 consulting support.





Phone Type

It is necessary to enter the MAC address of each telephone. The telephones have a barcode with the MAC ID printed on them. The phone type is a drop down list for selecting which IP phone hardware is being used on the extension.

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 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 IP1000 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, the settings on the phone take priority and will be kept in tact during the upgrade.

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.

- **Attach to Email** – Send a voicemail message to an email address by attaching it to an email message as an audio file (.Wav).
- **Delete After Emailing** – Delete the voicemail after it has been emailed to the email address provided for the extension in General Settings.
- **Say Caller ID** – State Caller ID prior to playback of the message.
- **Allow Review** – Allow callers to review a message after it has been played.
- **Allow Operator** – Allow pressing ”0” during the voicemail greeting to reach the system-wide operator.
- **Play Envelope Message** – Play caller ID and time of call prior to audio version of a message delivered through email.
- **Delete Email After** – 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. This feature is useful if there are certain permissions that an entire group needs to have, but extensions within that group that need to be restricted. For example, a phone in the lobby of a small business may be permitted to make local calls and dial extensions within the business, but be prohibited from making long distance or international calls.

1. Check the permission box if an extension is to have the calling capability.

   The following permissions can be set for an extension:

   - **Local Calls** – Permits calls that do not require a prefix or “1” prior to dialing.

  - **Long Distance Calls** – Permits both calls that require a prefix and calls requiring “1” and the prefix before the seven digit number.
  - **International Calls** – Permits calls that begin with “010.”
  - **Incoming Outside Calls** – Permits calls made to the business from people outside the business.
  - **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** – To assign here an extension as the 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. Click **Edit**. The Edit Extensions page will appear.
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.

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, or the choice selected by the incoming caller or the time of day.

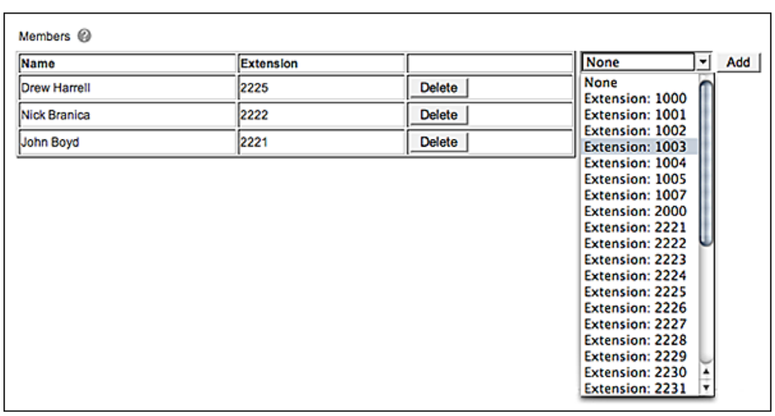
Ring Group Examples

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
- Round Robin with Memory

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:

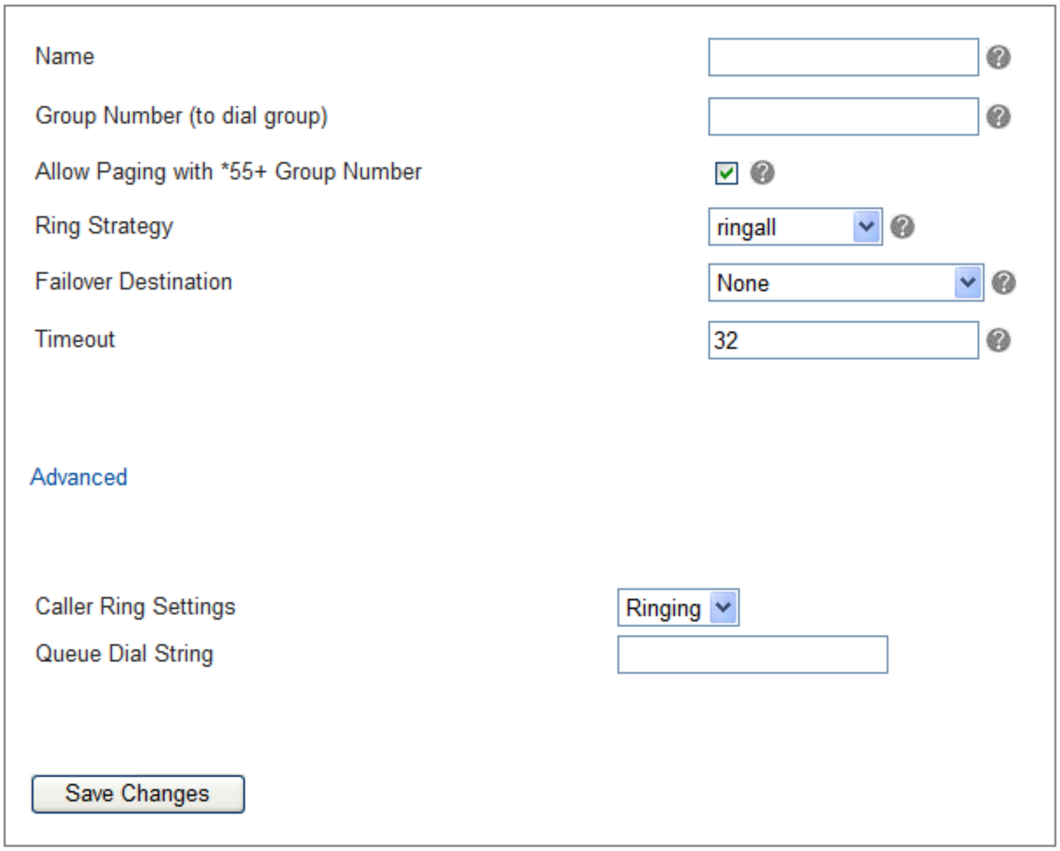
Thank you for calling ATI Connect a leading manufacturer of cable assemblies and wiring harnesses. If you know the extension of the party you would like to reach you may dial it at any time. For:

- 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



1. Click **Destinations** and **Groups**. The Groups page will appear.
2. Click **Add New**. The Edit Ring Group page will appear.
3. Enter a **Name** for the group.
4. Define a **Group Number**. This number must be three or four digits in length.
5. Check **Allow Paging with *55 + the Group Number** if this group will need to be paged.
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 takes 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 30 minutes.
9. Click **Save Changes**.
10. 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:

- **Ringing** – The phone continues to ring while the caller is waiting.
- **Music on Hold** – The caller hears music while waiting for a group member to pick up the call.

Queue Dial String

Queue Dial String is an advanced group setting reserved for 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** from the Groups Window on the page.
3. Click **Edit**. The Edit Groups page will appear.

4. **Make changes** to the Group.

5. Click **Save Changes**.

6. 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. **Select a Group**.

3. Click **Delete**. The group will be removed from the Groups Window on the page.

Menus

Menus direct calling traffic to destinations within a business. A menu is always associated with a menu prompt. The 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.

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:

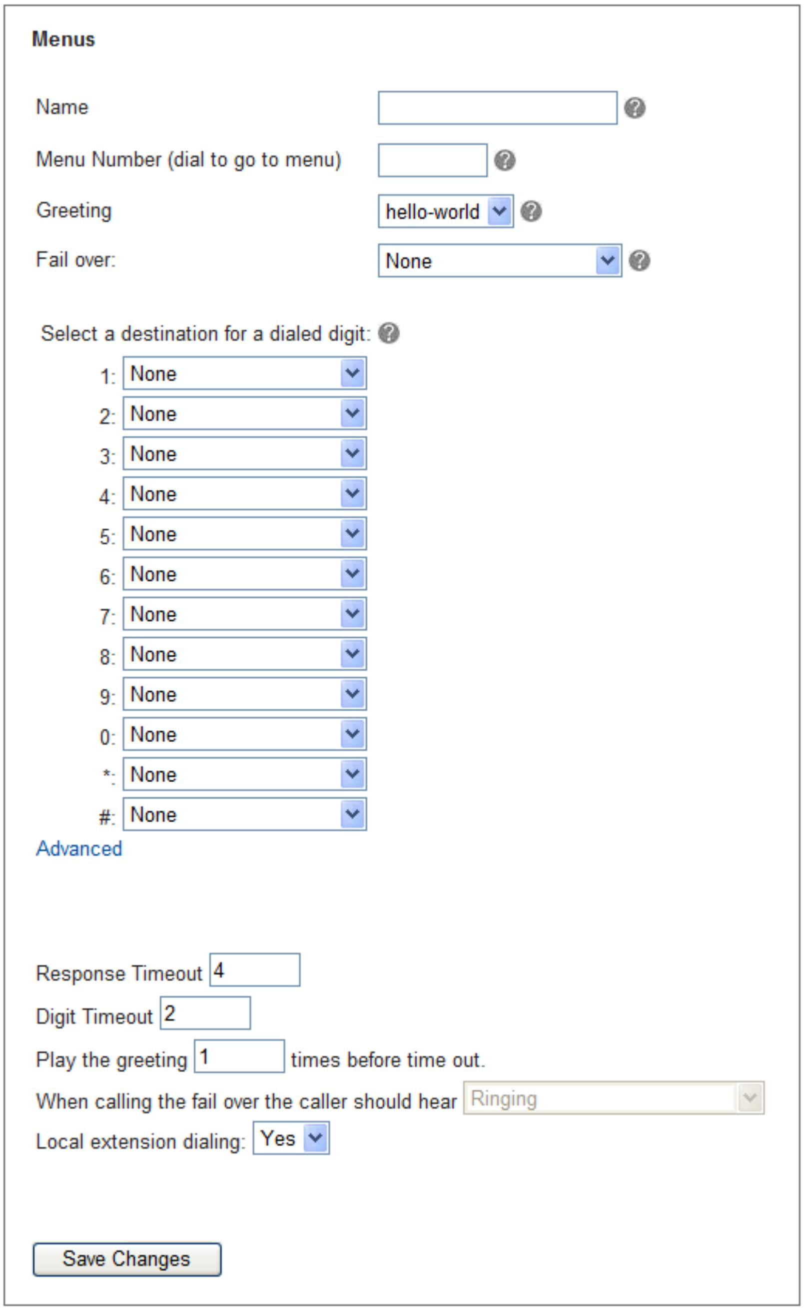
- 1 – Sales (Destination - Sales General Mailbox).
- 2 – Customer Service (Destination - Customer Service General Mailbox).
- 3 – Office Manager (Ext. 113).

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 New**. The Edit Menu page will appear.
3. Enter a **Name** for the menu.
4. Create 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. Define 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 Groups, Extensions, Locations in the Business (Conference Rooms), People, Voicemail Boxes, or Branch Offices (other office locations).

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

Select from the drop-down menu if the caller will hear Ringing or Music on Hold when a call is sent to the fail over destination.

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 an additional menu option.

Edit an Existing Menu

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

2. Select a **Menu** from the Menus Window on the page.

3. Click **Edit**. The Edit Menus page will appear.

4. **Make changes to the Menu**.

5. Click **Save Changes**.

6. 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. **Select a Menu**.

3. Click **Delete**. The group will be removed from the Menu Window on the page.

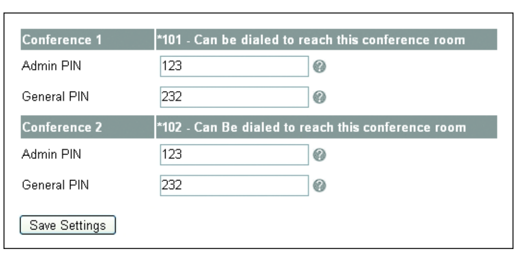
Meet-me Conferences

The 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. The number of people that can be in a conference at the same time is defined by the number of trunks coming into the system. 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). All other call participants access the call using a General PIN.

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.



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. Click on **Save Changes**.

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

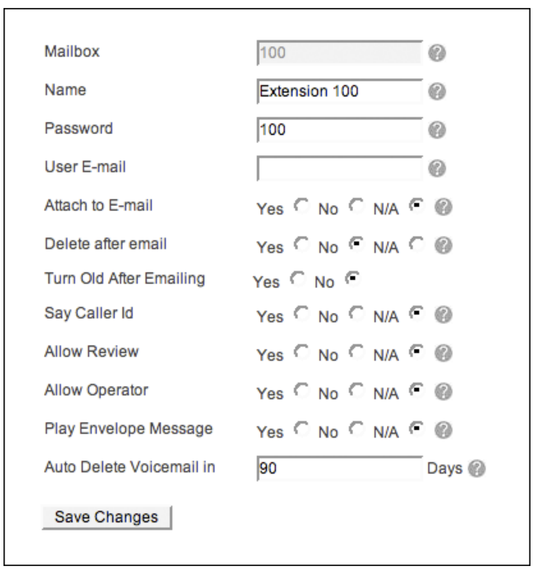
Voicemail

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.

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 New**. 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. Create a **Password** for the Mailbox.

6. Assign an **Email Address** to the Mailbox.

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

    - **Attach to Email** – Send a voicemail message to an Email address by attaching it to an email message as an audio file (.Wav).
    - **Delete After Emailing** – Delete the voicemail after it has been emailed to the address provided for the extension in General Settings.
    - **Say Caller ID** – State Caller ID prior to playback of the message.
    - **Allow Review** – Permits a user to review a voicemail message after it has been played.
    - **Allow Operator** – Defines an extension (person) as a system operator.
    - **Play Envelope Message** – Plays caller ID and time of call prior to audio version of a message delivered through Email.

    Implementation Note 4
    Not applicable (N/A) would most often apply if a Mailbox does not have an Email address associated with it. For example a Mailbox created for a Customer Service group to take calls from customers after hours.

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** from the Mailbox Window on the page.
3. Click **Edit**. The Edit Voicemail page will appear.
4. **Make changes to the Voicemail Box**.
5. Click **Save Changes**.
6. Click **Apply Changes** when ready to implement changes to the system.

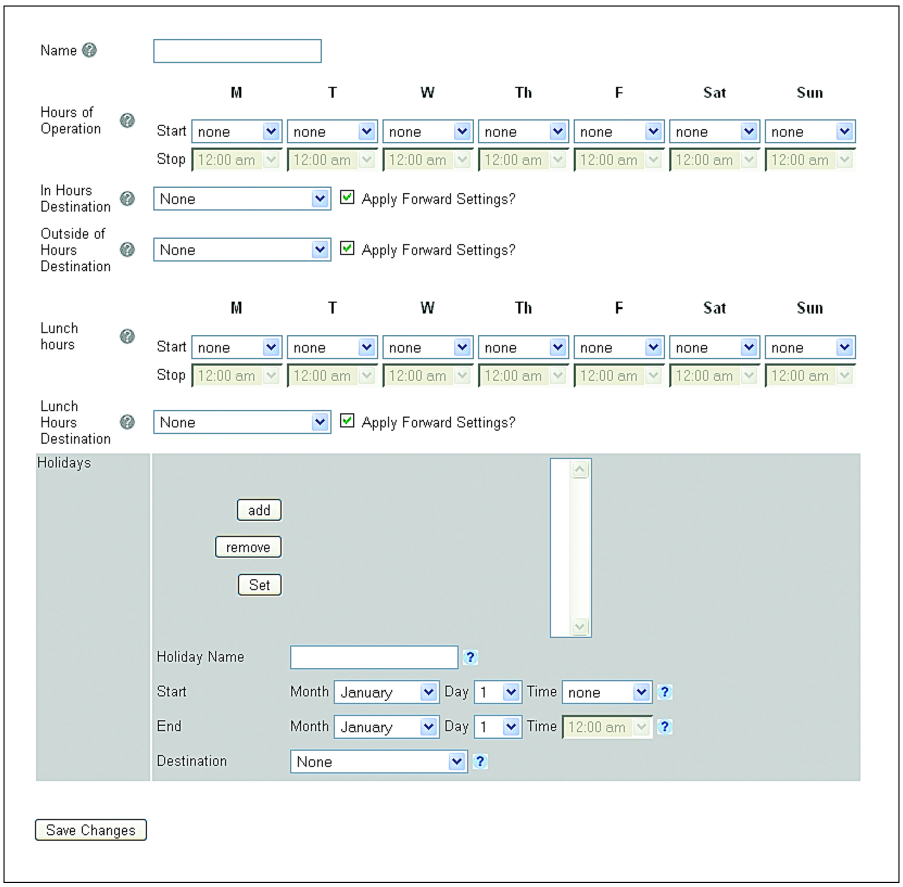
Delete a Voicemail Box

1. Click on **Destinations** and **Voicemail**. The Menus page will appear.
2. Select a **Voicemail** Box.
3. Click **Delete**. The group will be removed from the Menu Window on the page.

Schedules

A schedule is a destination. A schedule defines for a specified time period the destination to which calls are to be routed. A schedule can be used to define when a business, department, group or extension is open, closed or out to lunch. The IP1000 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.



Create A Schedule

1. Click on **Destinations** and **Schedules**. The Schedules page will appear.
2. Click **Add New**. 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. Define a **destination for calls that are In Hours**. Destinations can be

Groups, Extensions, Locations in the Business (Conference Rooms), People, Voicemail Boxes, or Branch Offices (other office locations).

6. Check **Apply Forward Settings**. This will apply the Forward Settings established with each destination.
7. Define a **destination for calls that are** "**Outside of Hours**."
8. Create **Lunch Hours** for the schedule by selecting times from the **Start** and **Stop** drop-down boxes for each day of the week.
9. Define a **destination for calls during Lunch Hours**.
10. Check **Apply Forward Settings**.
11. Create **Holiday Schedules** by:
    - Naming the Holiday.
    - Defining a **Start** and **End Date** for the Holiday in the Month, Day and Time fields.
    - **Define a destination for calls** during the Holiday.
    - Click **Add**, the Holiday appears in the list.
    - **Repeat this process** for each Holiday observed.
    - Click the **Holiday** and **Remove** to take a Holiday off the list.
12. Click **Save Changes**.
13. Click **Apply Changes** when ready to implement changes to the system.

Editing an Existing Schedule

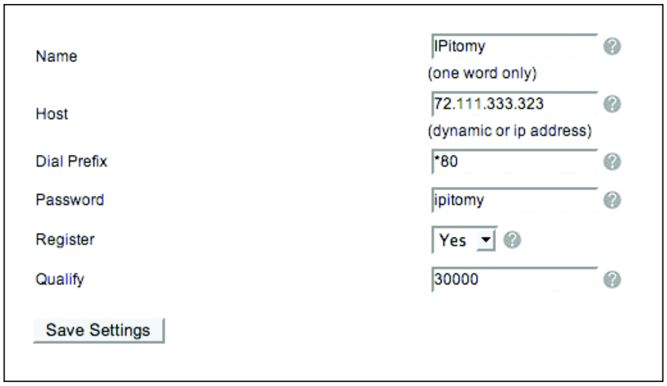
1. Click on **Destinations** and **Schedules**. The Schedules page will appear.
2. **Select a Schedule** from the Schedules Window on the page.
3. Click **Edit**. The Edit Schedule page will appear.
4. **Make changes to the Schedule**.
5. Click **Save Changes**.
6. Click **Apply Changes** when ready to implement changes to the system.

Deleting a Schedule

1. Click on **Destinations** and **Schedules**. The Schedules page will appear.
2. **Select a Voicemail Box**.
3. Click **Delete**. The schedule will be removed from the Menu Window on the page.

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.



Configuring Office 1

1. Click on **Destinations** and **Branch Offices**. The Branch Offices page will appear.
2. Click on **Add New**.
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.
6. Give a unique password for the connection.
7. Select No for register.
8. Leave qualify at 30000.
9. **Save Changes**
10. **Apply Changes**

Configuring Office 2

1. Click on **Destinations** and **Branch Offices**. The Branch Offices page will appear.
2. Click on **Add New**.
3. Name should match that given to the Office 1 PBX.
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.

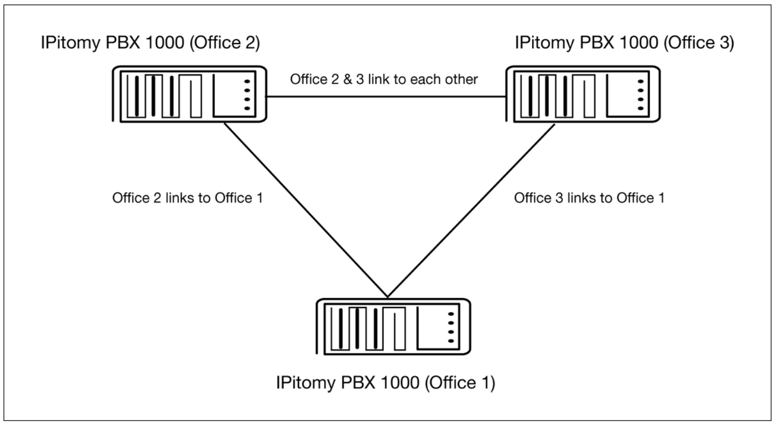
7. Select Yes for register.

8. Leave qualify at 30000.

9. **Save Changes**

10. **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.



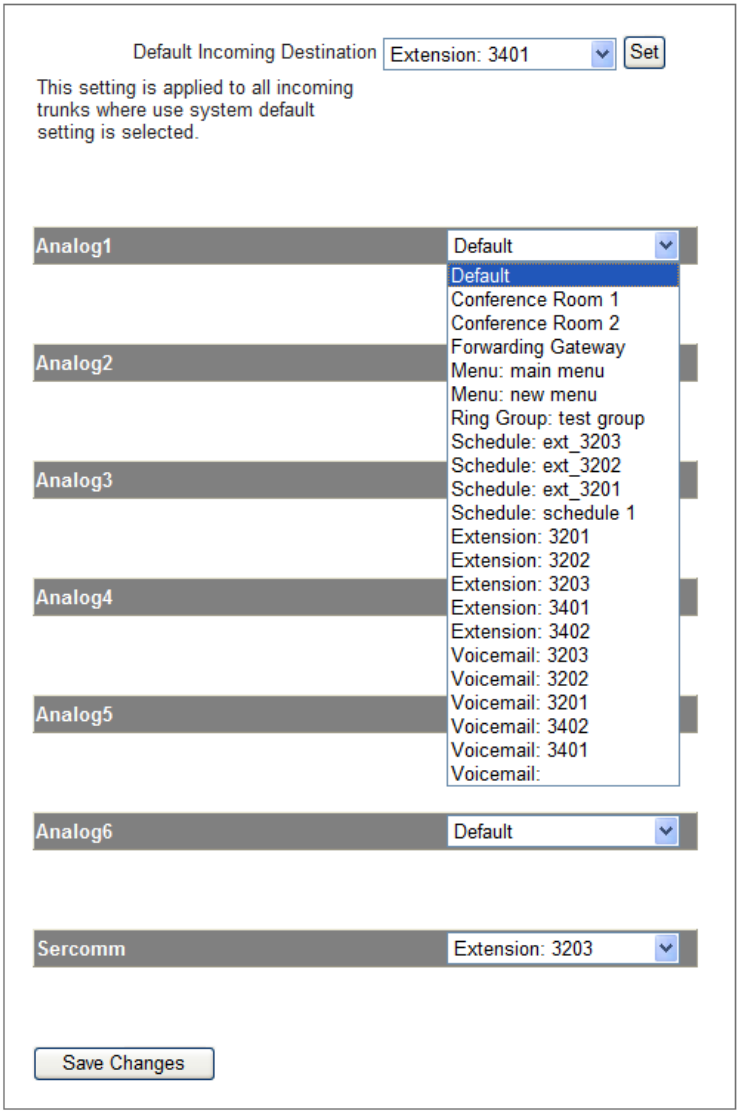
To place a call from one Branch Office to another just dial the prefix that was assigned to that locations PBX + the extension of the user trying to be reached at the other office. This same easy concept works for transferring calls from one Branch Office to another.

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 routes for all incoming calls to the system.



The Default Incoming Destination acts as a catch all for all available providers. If any specific providers or numbers are not specified this is where they will be routed too. The routing menu works in a hierarchy, it is important to understand this hierarchy:

- Phone Number (The DID number or phone number of an analog line)
- Provider (how that phone number arrives at the system e.g. analog, T1 or SIP)
- System Default (Catch all for all numbers ringing into the system)

The Hierarchy: a specific phone number or line coming in to a hardware provider card is more specific than the card itself. So a specific route other than default for the line or number that comes through a provider will override the provider default which in turn overrides the system default.

This hierarchy is determined by the routes chosen for each specific phone number, provider and system default (in that order) applied in the menu. The default destination is a specific location in the system such as a ring group, automated attendant menu or an extension. This can be set system wide or for each individual provider (analog lines, T1 or Sip provider). If an individual number associated with a provider is not specifically routed to a location, than the providers default destination will be used for that number. If a provider does not have a specific route set than the default for all unspecified numbers associated with that provider will go to the overall system default destination.

The IP1000 allows for flexible routing arrangements. Calls can initially be routed to a Menu, Schedule, Extension, Group, Conference, or Voicemail Box. Establishing Incoming Routing sets the default destination for all calls.

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 **Set** button.
4. Click the **Save Changes** button.

Defining the Service Provider Destination

A service provider destination must also be defined. The service provider destination must match the Default Incoming Destination for the system. 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 IP1000 comes with common outgoing call routes already provisioned in the system:

- Local
- Long Distance
- International
- Emergency Calls

These routes cannot 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.

4. Click **Call Routing** and **Outgoing**. The Outgoing Call Routing page will appear.

5. Click the **Add New** button. The Edit New Outgoing Route page will appear.

6. Enter a **Route Name**.

7. **Select the Route** Type the call is to take. This identifies the call as:

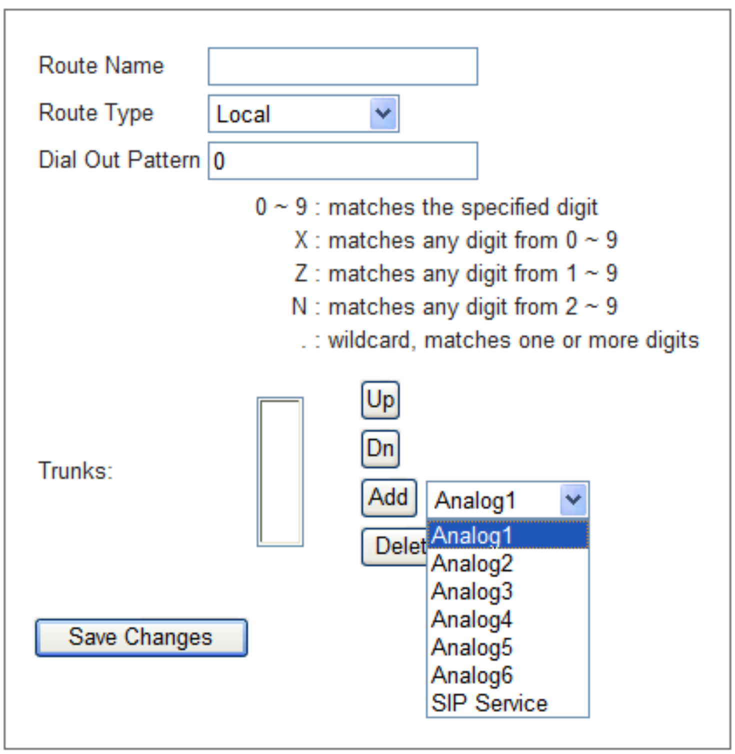
   a. Local (7-digit)
   b. 10 Digit
   c. Long Distance (1+10 digit) dialing
   d. International

8. Assign a routing Number.

Adding A New Call Route

1. Click **Call Routing** and **Outgoing**. The Outgoing Call Routing page will appear.

2. Click the **Add New** 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:

   o Local

- Long Distance
- 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.

Implementation Note 5 Default Settings For Existing Routes The Outgoing Call Routes already in the IP1000 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

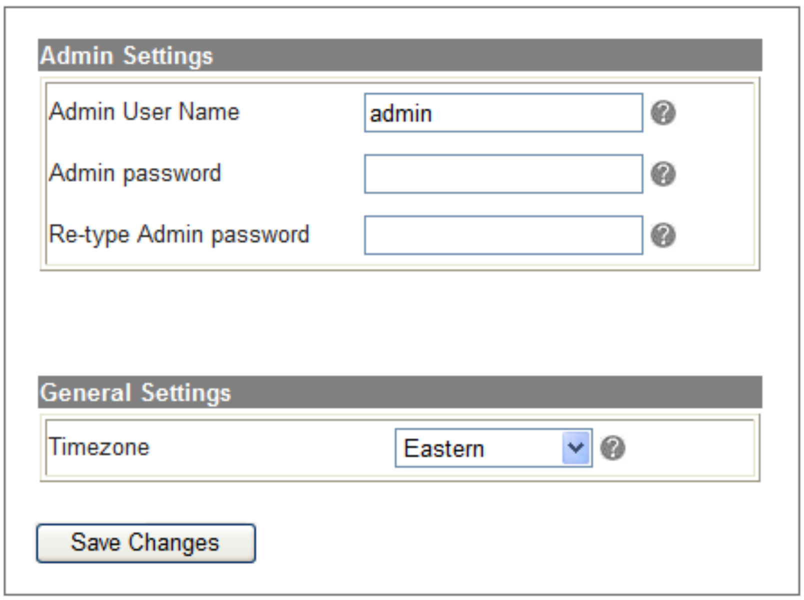
PBX setup is used by a System Administrator to manage system-specific and system-wide capabilities of the IP1000.

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

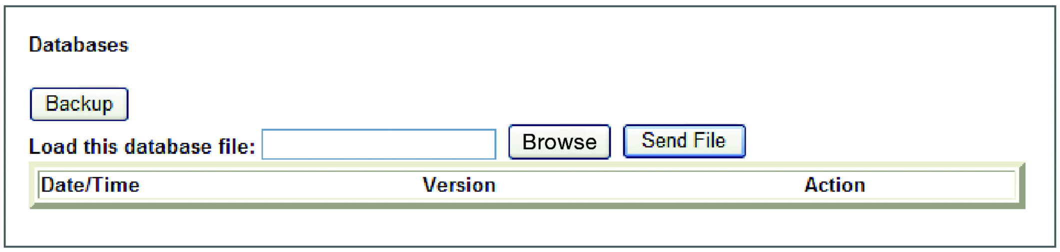
Database

The IP1000 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. Give the file a name and store it in a place that is easily accessible.

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

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.

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

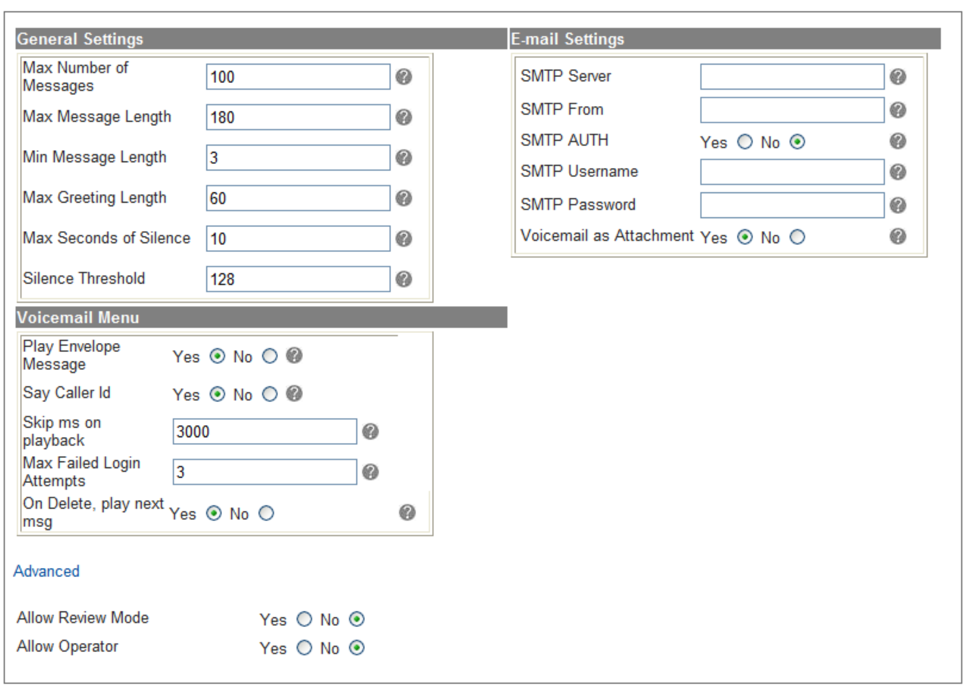
Voicemail

System-wide voicemail defaults define the rules by which voicemail boxes operate. In the IP1000 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 10 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 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 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.

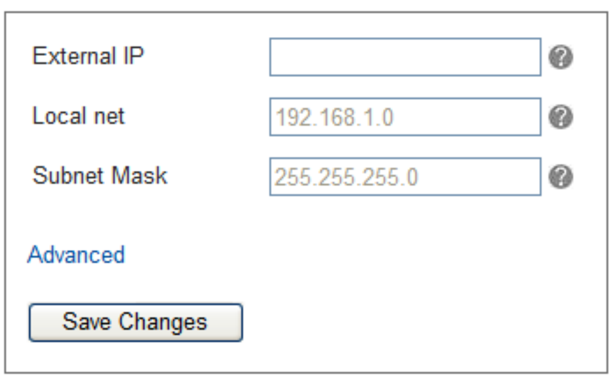
   * IPitomy Communications does not guarantee that any external mail server will work, some ISP's or hosts may have heavy mail filters and could block an incoming message.

Advanced Settings

- **Allow Review** – Allows users to review a message after it has been played.
- **Allow Operator** – Defines a specific extension (person) to act as an operator.

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:



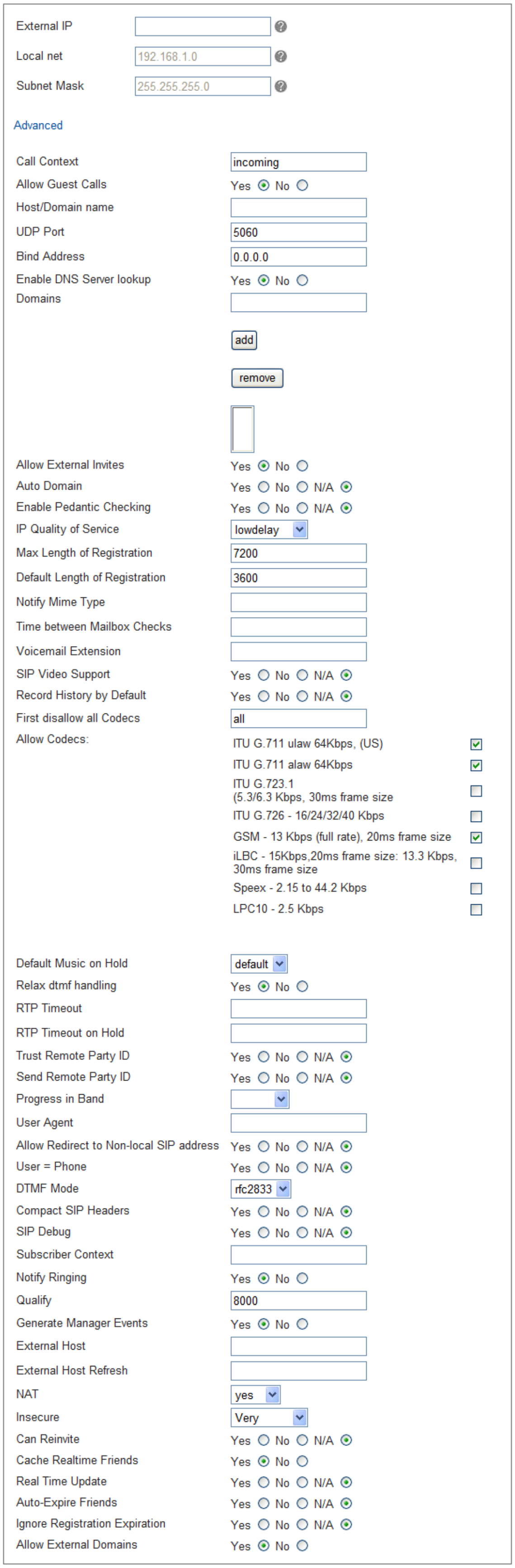
- **External IP** – This is the IP address of the ISP and can be obtained from the network router. The external IP address can also be obtained online at whatismyip.com. Type the address in a Web browser click Enter and the external IP address will appear on a page.
- **Local Network** – This is the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

> **Implementation 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.

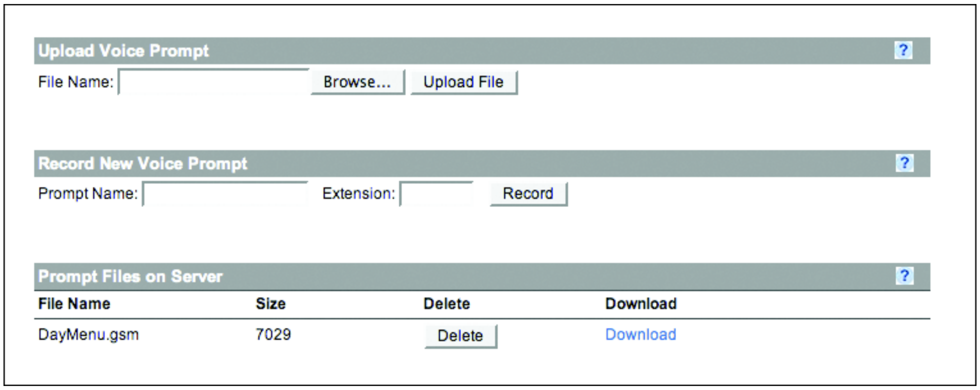

External IP
Local net
192.168.1.0
Subnet Mask
255.255.255.0
Advanced
Call Context
incoming
Allow Guest Calls
Yes No
Host/Domain name
UDP Port
5060
Bind Address
0.0.0.0
Enable DNS Server lookup
Yes No
Domains
add
remove
Allow External Invites
Yes No
Auto Domain
Yes No N/A
Enable Pedantic Checking
Yes No N/A
IP Quality of Service
lowdelay
Max Length of Registration
7200
Default Length of Registration
3600
Notify Mime Type
Time between Mailbox Checks
Voicemail Extension
SIP Video Support
Yes No N/A
Record History by Default
Yes No N/A
First disallow all Codecs
all
Allow Codecs:
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
Default Music on Hold
default
Relax dtmf handling
Yes No
RTP Timeout
RTP Timeout on Hold
Trust Remote Party ID
Yes No N/A
Send Remote Party ID
Yes No N/A
Progress in Band
User Agent
Allow Redirect to Non-local SIP address
Yes No N/A
User = Phone
Yes No N/A
DTMF Mode
rfc2833
Compact SIP Headers
Yes No N/A
SIP Debug
Yes No N/A
Subscriber Context
Notify Ringing
Yes No
Qualify
8000
Generate Manager Events
Yes No
External Host
External Host Refresh
NAT
yes
Insecure
Very
Can Reinvite
Yes No N/A
Cache Realtime Friends
Yes No
Real Time Update
Yes No N/A
Auto-Expire Friends
Yes No N/A
Ignore Registration Expiration
Yes No N/A
Allow External Domains
Yes No

Prompts

A prompt or greeting can welcome a caller to the business, direct them to a destination, provide instructions or deliver information. The IP1000 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 **Send 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.

Play 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 **Play**. The prompt will play.

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.

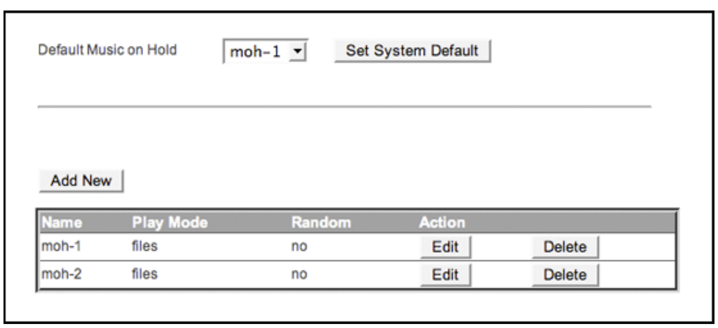
3. Click **Delete**. The prompt will be deleted from the list.

4. Click **Save Changes**.

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

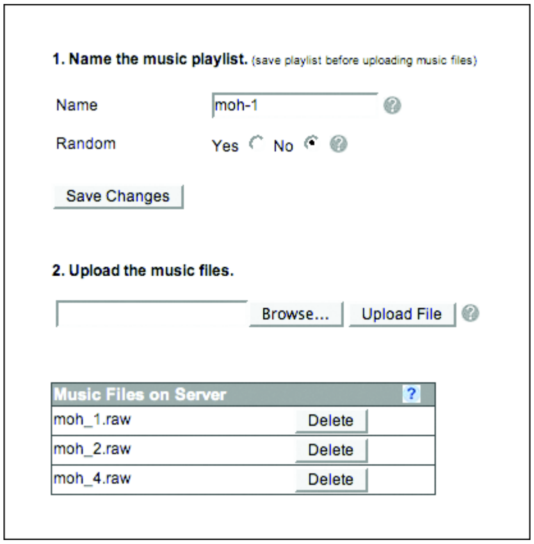
Music on Hold

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.



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 file** that will be uploaded.

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

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

7. Click on **Send File**. The file will appear in the Music Files on Server Window. It will also appear on the Music on Hold page where it will display the file Name, Play Mode and Random designation.

8. Click **Save Changes**.

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

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. Click **Edit**. The Edit Music on Hold page will appear.

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

5. Click **Save Changes**.

6. Click **Apply Changes**. The revised music fill will appear on the Music on Hold page where it will display the new file details.

Delete a Music File

1. Click **PBX Setup** and **Music on Hold**. The Music on Hold page will appear.
2. **Select a Music on Hold file** from the files on the Music on Hold page.
3. Click **Delete**. The Music on Hold file will be deleted from the list.
4. Click on **Apply Changes** when ready to implement this change to the system.

Setting Music on Hold

5. Click **PBX Setup** and **Music on Hold**. The Music on Hold page will appear.
6. Select your default Music on Hold play list from the dropdown menu.
7. Click **Set System Default.**
8. Click **Apply Changes.**

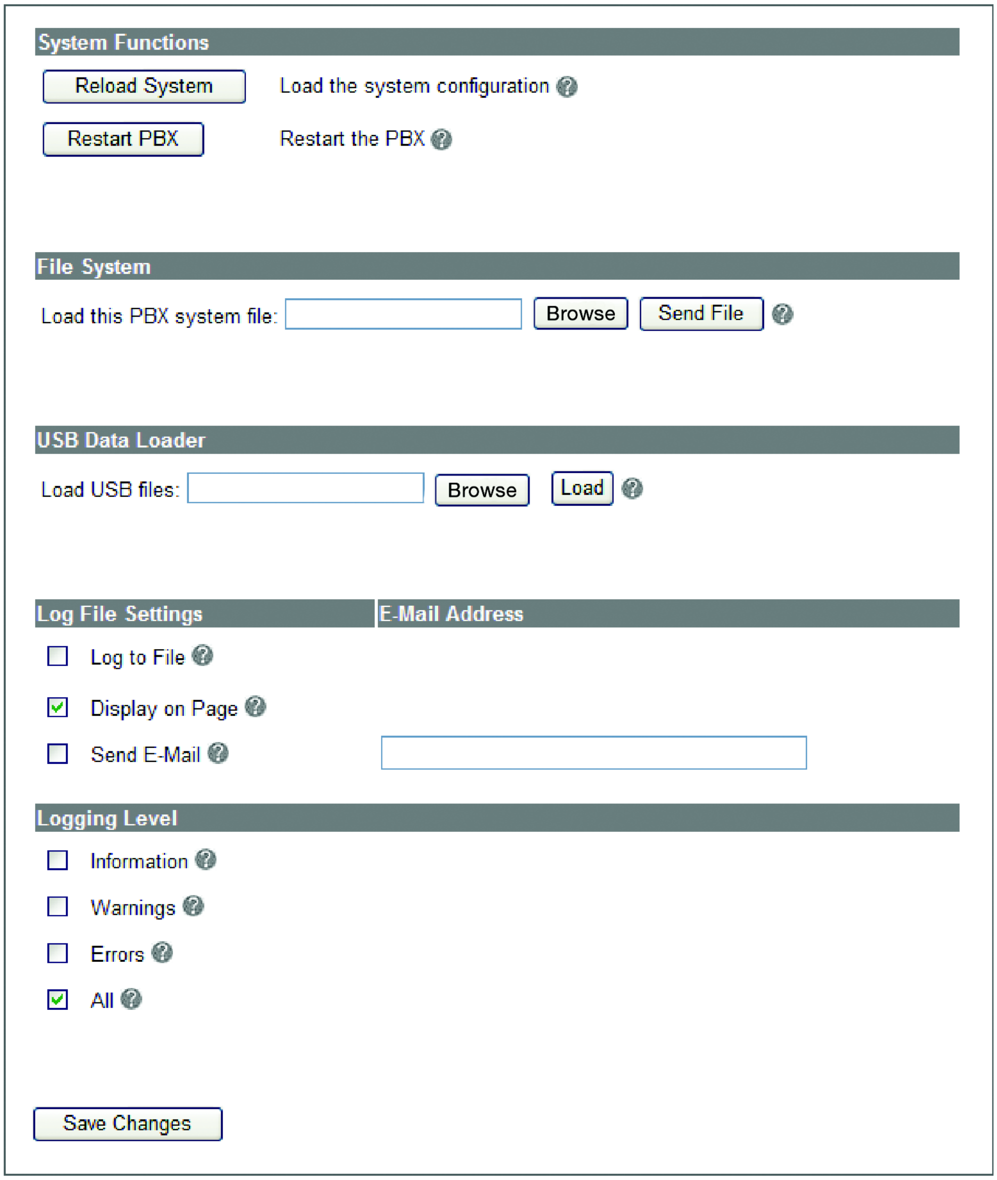
Feature Codes

Found in the PBX Setup section of online administration, the IP1000 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	Description
Personal Voicemail	* 123	Provides access to personal voicemail.
Voicemail Main	* 124	Provides access to main voicemail box.
Transfer to Voicemail	*+(ext #)	Transfers a caller to a voicemail box.
Directory	* 126	Access to the Company directory.
Blind Transfer	# #	Transfers a caller without announcing their call.
Attended Transfer	# *	Allows person transferring a call to stay on the call until it is received by the intended party.
One Touch Record	* #	Turns Call Recording on.
Pickup Extension	* 8	Allows a person to pick up a call at a different extension than their own.
Ring Group Page	** (ext #)	Pages a ring group.

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 a Customer Service Representative. The default settings in this page are not to be changed during a system implementation.



Reports

The IP1000 offers System Administrators a set of reporting tools to help manage calling traffic. Reporting tools also includes a diagnostic report used 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 IP1000 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:

1 2 3 4 5 6 7 8 9 10 11 > Last

Date/Time	From	Destination	Trunk	Duration	Status
2007-03-07 15:57:33	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:43:56	2222	Voice Mail	SIP/2222	00:00:34	answered
2007-03-07 15:39:56	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:36:43	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:34:23	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:33:54	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:33:20	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:33:08	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:32:58	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:32:03	9413062200	9552101	SIP/2221	00:00:12	answered
2007-03-07 15:31:54	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:31:43	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:31:37	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:31:09	9413062200	9552101	SIP/2221	00:00:13	answered
2007-03-07 15:30:27	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:30:09	9413062200	9552101	SIP/2221	00:00:05	answered
2007-03-07 15:29:04	4155368923	2222	Zap/3	00:01:06	answered
2007-03-07 15:29:39	9413062200	9552101	SIP/2221	00:00:23	answered
2007-03-07 15:29:36	9413062200	9552101	SIP/2221	00:00:00	no answer
2007-03-07 15:29:33	9413062200	9552101	SIP/2221	00:00:00	no answer

Diagnostics

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

```
Name/username             Host            Dyn Nat ACL Port     Status
102/102                   (Unspecified)    D   N   A  0        UNKNOWN
101/101                   (Unspecified)    D   N   A  0        UNKNOWN
100/100                   (Unspecified)    D   N   A  0        UNKNOWN
3 sip peers [0 online , 3 offline]

Channel             Location             State   Application(Data)
0 active channels
0 active calls
```

Recordings

The Recordings section allows you to:

1. Listen to recordings.

2. Download recordings.

3. 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.

Appendix 1: IP Telephones

SP320

SP320 100 is a full-feature business-grade IP phone offering a:

- **Complete business feature set**.
- **Simplified installation** using unique discovery protocol.

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 40)

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. This icon 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 IP1000. 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 IP1000 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 IP1000 (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.
- 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 IP1000 allows calls to be forwarded to a PSTN. Forward calls to a PSTN number by entering it into the field provided. Calls can be forwarded to any destination (or telephone number) in the drop-down list or any telephone number.

Changing a Forwarding Number from an Extension

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

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

Changing a Forwarding Number from a PC

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 IP1000 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.**

Select **Save Changes**.

Click **Apply Changes** when ready to implement the extension to the system.

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 IP1000 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