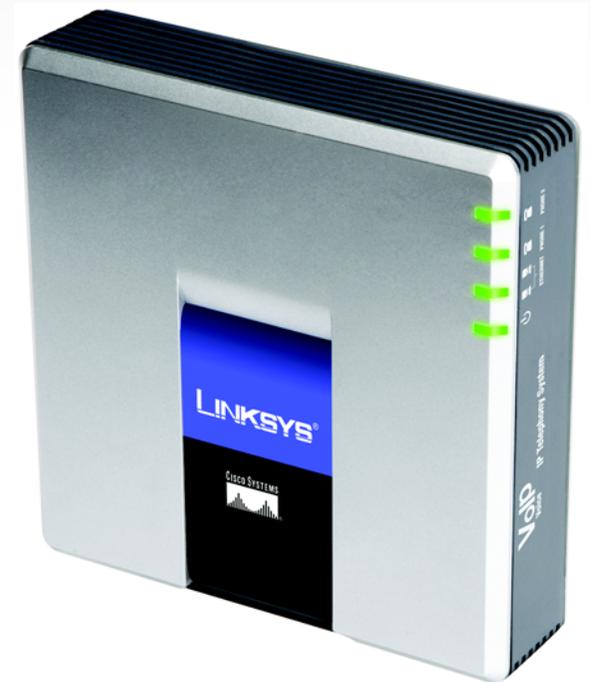


# LINKSYS®

A Division of Cisco Systems, Inc.



## IP Telephony System

## User Guide

**VoIP**  
Voice

Model No. **SPA9000**

CISCO SYSTEMS



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**WARNING:** This product contains chemicals, including lead, known to the State of California to cause cancer, and birth defects or other reproductive harm. ***Wash hands after handling.***

## How to Use this Guide

The guide to the IP Telephony System has been designed to make understanding networking with the IP Telephony System easier than ever. Look for the following items when reading this User Guide:



This checkmark means there is a note of interest and is something you should pay special attention to while using the IP Telephony System.



This exclamation point means there is a caution or warning and is something that could damage your property or the IP Telephony System.



This question mark provides you with a reminder about something you might need to do while using the IP Telephony System.

In addition to these symbols, there are definitions for technical terms that are presented like this:

***word: definition.***

Also, each figure (diagram, screenshot, or other image) is provided with a figure number and description, like this:

**Figure 0-1: Sample Figure Description**

Figure numbers and descriptions can also be found in the “List of Figures” section.

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# Chapter 1: Introduction

## Welcome

Thank you for choosing the Linksys IP Telephony System. The System combines the rich feature set of legacy PBX (Private Branch eXchange) telephone systems with the convenience and cost advantages of Internet telephony. It supports common key system features such as an auto-attendant, music-on-hold, call forwarding, three-way call conferencing, and more.

The System is so easy to configure that a fully working system can be set up in minutes. New Linksys SPA-family Internet telephones are automatically detected and registered when they are connected to the System. While the System will work with any SIP-compatible Internet telephone, it is the ideal host for Linksys business telephones, including model number: SPA941. The System supports the advanced features of these phones, such as shared line appearances, hunt groups, call transfer, call park, and group paging. Plus, with its two FXS ports, the System can support traditional analog devices such as telephones, fax machines, answering machines, media adapters.

How does the System do all of this? By connecting your analog phones or fax machines to the System and connecting the System and Internet phones to your router, then the System can direct voice communications for your network.

But what does all of this mean?

Networks are useful tools for sharing Internet access and computer resources. Multiple computers can share Internet access, so you don't need more than one high-speed Internet connection. With Internet phone service, your Internet access can now be shared by your Internet phones as well. You will be able to make phone calls using your Internet phone service account, even while another colleague is web browsing. Plus, you can access one printer from different computers and access data located on another computer's hard drive (with the right permissions).

PCs on a wired network create a LAN, or Local Area Network. They are connected with Ethernet cables, which is why the network is called "wired". The System takes your wired network and lets you integrate Internet phones and Internet phone service.

When you first install the System, Linksys strongly recommends that you use the Setup Wizard, which you can download from [www.linksys.com](http://www.linksys.com). If you do not wish to run the Setup Wizard, then use the instructions in the Quick Installation or this User Guide to help you. These instructions should be all you need to get the most out of the IP Telephony System.



**NOTE:** Some of these features are set up from the Internet phones.

**network:** a series of computers or devices connected for the purpose of data sharing, storage, and/or transmission between users.

**lan (local area network):** the computers and networking products that make up the network in your home or office.

**ethernet:** an IEEE standard network protocol that specifies how data is placed on and retrieved from a common transmission medium.

## What's in this Guide?

This user guide covers the steps for setting up a network with the System. Most users will only need to use “Chapter 4: Getting Started.” When you're finished, then you are ready to make calls within your system as well as calls to the outside world.

You also have other chapter available for reference:

- **Chapter 1: Introduction**  
This chapter describes the System and this User Guide.
- **Chapter 2: Applications**  
This chapter discusses the most common scenarios for the System.
- **Chapter 3: Getting to Know the IP Telephony System**  
This chapter describes the physical features of the System.
- **Chapter 5: Using the Interactive Voice Response Menu**  
This chapter explains how to configure the System's network settings when you access its Interactive Voice Response Menu.
- **Chapter 6: Using the Web-based Utility**  
This chapter explains how to configure the settings of the System through the Web-based Utility.
- **Appendix A: Troubleshooting**  
This appendix describes some possible problems and solutions, as well as frequently asked questions, regarding installation and use of the System.
- **Appendix B: Configuring the Nighttime Auto-Attendant**  
This appendix explains how to set up the auto-attendant for nighttime (non-business) hours.
- **Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users**  
This appendix explains how to define the dial plan and auto-attendant instructions. (These instructions are for advanced users only.)
- **Appendix D: New Music for the Music-on-Hold Feature**  
This appendix explains how to replace the System's default music file with your own music file.
- **Appendix E: Finding the MAC Address and IP Address for Your Ethernet Adapter**  
This appendix instructs you on how to find the MAC address or Ethernet address of your PC's Ethernet network adapter.

## **IP Telephony System**

- **Appendix F: Windows Help**  
This appendix describes how you can use Windows Help for instructions about networking, such as installing the TCP/IP protocol.
- **Appendix G: Glossary**  
This appendix gives a brief glossary of terms frequently used in networking.
- **Appendix H: Specifications**  
This appendix provides the technical specifications for the System.
- **Appendix I: Warranty Information**  
This appendix supplies the warranty information for the System.
- **Appendix J: Regulatory Information**  
This appendix supplies the regulatory information regarding the System.
- **Appendix K: Contact Information**  
This appendix provides contact information for a variety of Linksys resources, including Technical Support.

# Chapter 2: Applications for the IP Telephony System

## How Does the IP Telephony System Fit into My Business or Home?

High-speed Internet access is a valuable resource. When you have more than one computer, chances are you want to share that Internet access with all of your computers. That's when you create a network, a collection of devices connected to each other. A device called a router connects computers and other devices, so they can share a high-speed Internet connection and other resources, including data and printers.

One of the biggest benefits of the Internet is data communications, either e-mail or web browsing, whether you send a file to a client or download the latest software upgrade. With the System, you also get voice communications.

## What Does the IP Telephony System Do?

The System connects multiple Internet phones to an Internet phone service. The System manages and routes all calls. Incoming calls go to the auto-attendant, an automated greeting system, or correct internal extension (each phone has its own extension number). Outgoing calls go to the correct external phone number (you can have more than one external phone number).

You can have not only more than one external phone number, but also up to four Internet Telephony Service Providers (ITSPs) for maximum flexibility.



**NOTE:** The basic configuration of the System lets you connect up to four Internet phones and use up to four ITSPs. To expand the basic configuration, contact your primary ITSP for more information.

## A Typical Scenario

Typically, you connect the Internet port of the System to a local network port of your router. Then connect a switch to another local network port of your router. Use this switch to connect Internet phones, computers, and other devices. Then connect an administration computer to the Ethernet port of the System.

If you have analog telephones or fax machines, you can connect them to the Phone ports, so you can also use those phones to make Internet phone or fax calls. (More details are available in "Chapter 4: Getting Started.")

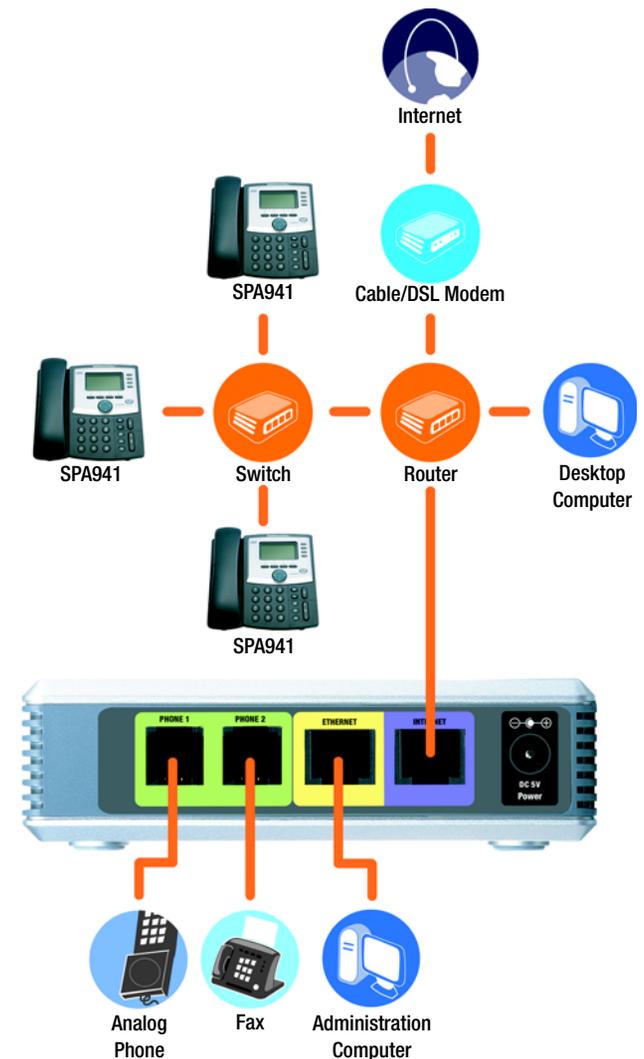


Figure 2-1: A Scenario for the IP Telephony System

### What Kind of Router Should I Use?

For your network, get the highest-performance router possible. For best results, use a QoS (Quality of Service) router, so it can assign top priority to voice traffic.

### What Kind of Switch Should I Use?

Again, performance is key. For best results, use a switch that offers QoS (Quality of Service) and full wire-speed switching. QoS enables the switch to give top priority to voice traffic, while full wire-speed switching lets it forward packets as fast as your network can deliver them. The next best choice is a switch featuring QoS (Quality of Service).

### What if I Keep My Traditional Phone Service?

Traditional phone service, also known as Plain Old Telephone Service (POTS), runs on a network called the Public Switched Telephone Network (PSTN). If you decide to keep traditional phone service, then connect the Analog Telephone Adapter (model number: SPA3000) to the switch. (For more information, refer to the SPA3000 documentation.)

## Which Call Management Features Does the IP Telephony System Offer?

Beyond basic call routing, the System offers several powerful and sophisticated features:

- **Auto-Attendant.** An automated system guides each caller to the appropriate contact.
- **Music-on-Hold.** You can combine the auto-attendant feature with the music- or information-on-hold feature, so the caller has a richer experience with your call system.
- **Call Hunt.** You can designate which Internet phones receive outside calls. You can even have calls ring multiple phones, either simultaneously or one at a time.
- **Paging.** When you want to page all of the Internet phones, you can use the System.
- **Dial Plans.** When you have more than one dial plan, you can route outgoing calls to take advantage of the best rates available for the different types of calls.

After setup of the System, you will have dynamic and feature-rich Internet voice communications for your business or home.



**NOTE:** If your ITSP configured the System for you, then these features may already be set up. Check with your ITSP for more information.

(To set up these features yourself, refer to “Chapter 6: Using the Web-based Utility.”)

# Chapter 3: Getting to Know the IP Telephony System

## The Back Panel

The System's ports are located on its back panel.

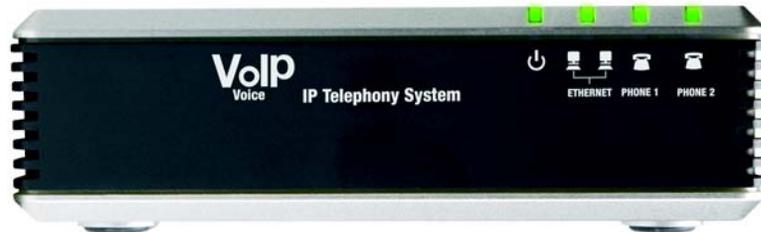


**Figure 3-1: Back Panel**

- PHONE 1/2** The **PHONE 1/2** ports allow you to connect analog telephones (or fax machines) to the System using RJ-11 telephone cables (not included).
- ETHERNET** The **ETHERNET** port connects to an administration computer, so you can access the System's Web-based Utility for configuration.
- INTERNET** This **INTERNET** port connects to either a router or broadband modem.
- Power** The **Power** port is where you will connect the power adapter.

## The Front Panel

The System's LEDs are located on its front panel.



**Figure 3-2: Front Panel**

- Power** Green. The **power** LED is solidly lit when the System is powered on and connected to the Internet. It flashes when there is no Internet connection.
- ETHERNET** Green. The **ETHERNET** LED is solidly lit when there is an Internet connection. It flashes when there is network activity.
- PHONE 1/2** Green. The **PHONE 1/2** LED is solidly lit when the phone is on-hook and registered. (The connection is registered if your Internet phone service account is active.) The LED is not lit when the phone is on-hook and not registered. It flashes when the phone is off-hook.

# Chapter 4: Getting Started

## Overview

For first-time installation of the System, Linksys strongly recommends using the Setup Wizard, which you can download from [www.linksys.com](http://www.linksys.com). For advanced users, you may follow the instructions in this chapter, and then use the Web-based Utility for additional configuration (refer to “Chapter 6: Using the Web-based Utility”). To use the Interactive Voice Response Menu, proceed to “Chapter 5: Using the Interactive Voice Response Menu.”

## Before You Begin

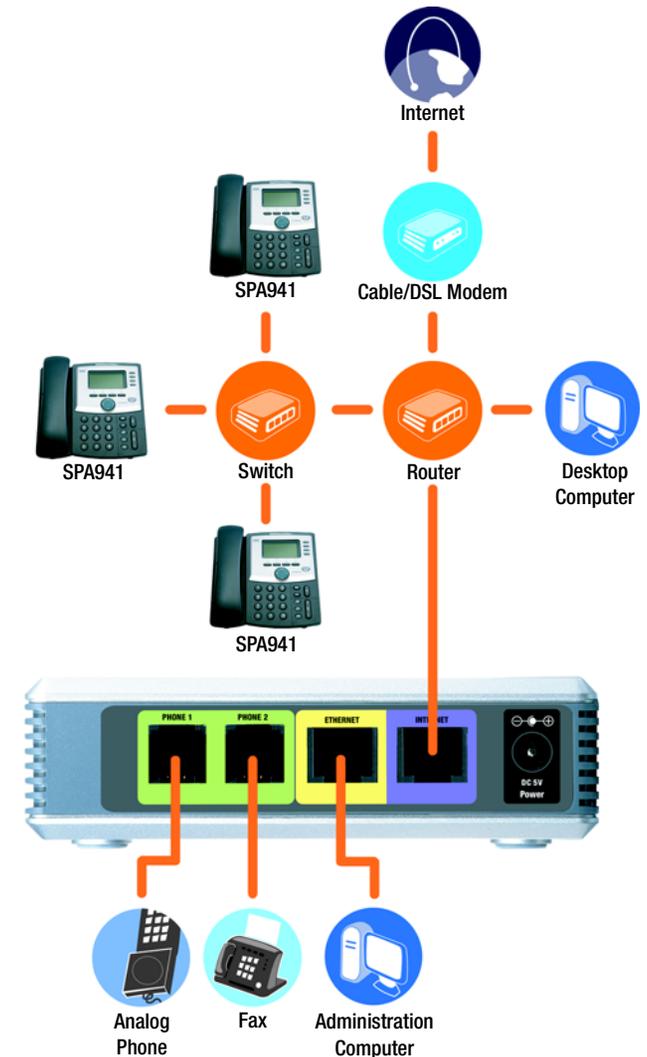
Make sure you have the following:

- IP Telephony System (model number: SPA9000)
- One or more Internet phones (for example, Linksys SPA-family IP Phones, model number: SPA941)
- A router and cable/DSL modem (or gateway)
- One or more Ethernet network switches (so you can connect Internet phones or computers)



**NOTE:** For best results, use a switch that offers QoS (Quality of Service) and full wire-speed switching. QoS enables the switch to give top priority to voice traffic, while full wire-speed switching lets it forward packets as fast as your network can deliver them. The next best choice is a QoS (Quality of Service) switch.

- At least one active Internet phone service account and its settings if you want to make external calls
- An active Internet connection if you want to make external calls
- At least one computer for configuration of the System and Internet phones
- Two or more Ethernet network cables
- Analog telephones or fax machines (optional)



**Figure 4-1: A Typical Scenario for the IP Telephony System**

## Instructions for Installing the IP Telephony System

### Internal Calls

To install the System for internal calls, you will do the following:

- connect and configure the System
- connect the Internet phones

### Connect and Configure the System

1. (optional) Plug an analog telephone into the Phone 1 port of the System.
2. (optional) If you have a second analog telephone or fax machine, plug it into the Phone 2 port of the System.

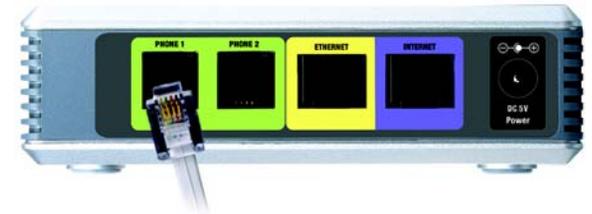


**IMPORTANT:** Do not connect the Phone port to a telephone wall jack. Make sure you only connect a telephone or fax machine to the Phone port. Otherwise, the System or the telephone wiring in your home or office may be damaged.

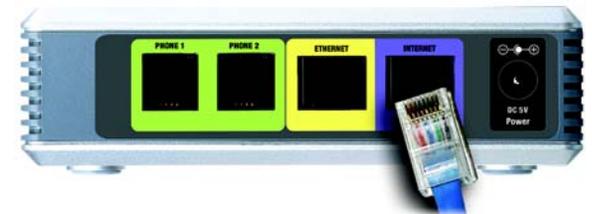
3. Connect an Ethernet network cable to the Internet port of the System. Then connect the other end of the cable to one of the Ethernet ports on your router.
4. Connect a different Ethernet network cable to the Ethernet port of the System. Then connect the other end to the computer you will use to manage the System (this will be called the administration computer).
5. Connect the included power adapter to the Power port of the System, and then plug the power adapter into an electrical outlet.
6. Launch the web browser on the administration computer.
7. Enter **192.168.0.1/admin/voice/advanced** in the *Address* field (**192.168.0.1** is the default local IP address of the System). Then press the **Enter** key.

*ip (internet protocol): a protocol used to send data over a network.*

*ip address: the address used to identify a computer or device on a network.*



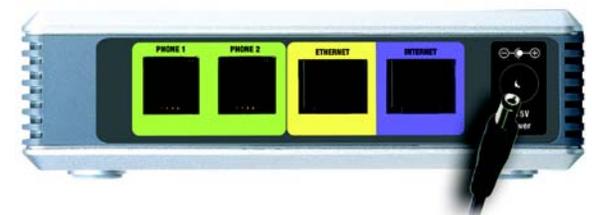
**Figure 4-2: Connect to the Phone 1 Port**



**Figure 4-3: Connect to the Internet Port**



**Figure 4-4: Connect to the Ethernet Port**



**Figure 4-5: Connect to Power**

## IP Telephony System

- The *Voice - Info* screen will appear. Click the **SIP** tab.
- In the PBX Parameters section, select **WAN** from the *Proxy Network Interface* drop-down menu.
- Click the **Submit All Changes** button.
- The *Voice - Info* screen will appear. Click the **Router** tab.
- Click the **WAN Setup** tab.
- From the *Connection Type* drop-down menu, select **Static IP**.
- In the Static IP Settings section, complete the *Static IP*, *NetMask*, and *Gateway* fields.

**Static IP.** Enter a static IP address appropriate for your network. Write this down; you will use it later.



**NOTE:** Make sure your router will not assign the System's static IP address to any other network device. For example, you can assign a static IP address outside of your router's DHCP IP address range; however, it must be within the router's subnet range.

For more information about IP addressing, refer to the router's documentation.

**NetMask.** Enter the subnet mask of your network router.

**Gateway.** Enter the local IP address of your network router or gateway.

- In the Optional Settings section, complete the *Primary DNS* field.

**Primary DNS.** Enter the DNS IP address of your network router.

- In the Remote Management section, select **yes** from the *Enable WAN Web Server* drop-down menu.
- Click the **Submit All Changes** button.

Figure 4-6: Voice - SIP Screen - PBX Parameters

Figure 4-7: Router - WAN Setup Screen

## IP Telephony System

18. The *Router - Status* screen will appear. Verify that the following settings match your entries:

- WAN Connection Type - Static IP
- Current IP
- Current Netmask
- Current Gateway
- Primary DNS

**Proceed to the next section, “Connect the Internet Phones.”**

### Connect the Internet Phones

1. Connect an Ethernet network cable to one of the Ethernet ports on your router. Then connect the other end of the cable to an Ethernet port on a network switch.
2. Connect the switch’s power adapter to its power port, and then plug the power adapter into an electrical outlet.
3. Connect an Ethernet network cable to an Internet phone. Then connect the other end to one of the Ethernet ports on the switch.

(If the Internet phone has been used before, reset it to its factory default settings first. Refer to its documentation for more information.)

4. Connect the Internet phone’s power adapter to its power port, and then plug the power adapter into an electrical outlet.
5. The Internet phone will reboot two to three times (each reboot may take up to one minute). The System will automatically assign an extension number to the Internet phone. When the Internet phone displays its extension number, then it is ready for use.



**NOTE:** The System automatically registers Linksys SPA-family Internet phones (including model number SPA941). If you connect a different SIP-compatible phone, then registration will be manual. Refer to the documentation for your phone.



**NOTE:** The default SIP port of the System is **6060**.

6. Repeat steps 3-5 until you have installed all of your Internet phones.

**Congratulations! Now you can make calls from one Internet phone to another by dialing an extension number.**

**Continue to the next section to configure the System for external calls.**

## External Calls

For external calls, make sure you have an active Internet connection. Then configure the settings for your Internet phone service account on the System.

1. Launch the web browser on the administration computer.
2. Enter **<IP address of the System>/admin/voice/advanced** in the *Address* field (use the static IP address you previously assigned to the System). Then press the **Enter** key.
3. The *Voice - Info* screen will appear. Click the **Line 1** tab.
4. On the *Line 1* screen, enter the settings for your Internet phone service account.

### Subscriber Information

**User ID.** Enter the user ID (also called the account number) supplied by your ITSP. Do not use any hyphens, spaces, or other punctuation.

**Password.** Enter the case-sensitive password supplied by your ITSP.

### Proxy and Registration

**Proxy.** Enter the proxy address supplied by your ITSP.

If your ITSP supplied additional settings, enter those as well. Refer to the instructions your ITSP gave you.

5. Click the **Submit All Changes** button to save your new settings.
6. The System will reboot itself. Then the Internet phones will reboot themselves.
7. The *Voice - Info* screen will appear. In the Line 1 Status section, make sure that the Registration Status says, "Registered."

You are now ready to make your first external call. Use any phone connected to the System, and dial **9** first when you make an external call with the default US dial plan.

You can use analog telephones to make external calls; however, you cannot receive calls on any analog telephones unless you configure the appropriate settings. Refer to the Voice - FXS 1 section of "Chapter 6: Using the Web-based Utility" for instructions.

**Congratulations! Now you can make external calls using the System.**



**NOTE:** If your Internet Telephony Service Provider (ITSP) supplied the System, then it may be pre-configured for you, and you do not need to change any settings. Refer to the instructions supplied by your ITSP for more information.

**Figure 4-8: Voice - Line 1 Screen**



**NOTE:** If you cannot make calls with the default US dial plan, visit [www.linksys.com/kb](http://www.linksys.com/kb) for additional dial plans, or refer to "Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users" to write your own script.

## Receiving and Handling External Phone Calls

To receive external phone calls, you need to know the Direct Inward Dialing (DID) number assigned to you by your ITSP. Usually this is the same as your user ID, but it can be a different number. Check with your ITSP to find out what your DID number is.

Then decide which Internet phones will ring when an outside caller calls your DID number. The default is **aa**, which stands for auto-attendant, an automated system that picks up external calls and plays pre-recorded voice messages. If you want only the auto-attendant to receive a call, keep the default setting. When the auto-attendant receives a call, it will prompt the caller to dial the appropriate extension.

If you want specific Internet phones to ring when your DID number is called, then refer to “Chapter 6: Using the Web-based Utility” for instructions about the Contact List setting.



**NOTE:** If you decide to keep traditional phone service, which is also known as Plain Old Telephone Service (POTS), then you will use the Linksys Analog Telephone Adapter (model number: SPA3000). For details, refer to the Analog Telephone Adapter’s documentation.

## Configuring the Auto-Attendant

By default, the daytime auto-attendant is enabled, so the first message it plays (“If you know your party’s extension, you may enter it now”) is suitable for business hours. If you want a caller to hear a different greeting during nighttime (non-business) hours, then refer to “Appendix B: Configuring the Nighttime Auto-Attendant.”

**To use the Web-based Utility for additional configuration, refer to “Chapter 6: Using the Web-based Utility.” To use the Interactive Voice Response Menu, proceed to “Chapter 5: Using the Voice Interactive Response Menu.”**

# Chapter 5: Using the Interactive Voice Response Menu

## Overview

You may need to manually configure the System by entering the settings provided by your Internet Telephony Service Provider (ITSP). This chapter explains how to use the Interactive Voice Response Menu to configure the System's network settings and record auto-attendant messages. You will use the telephone's keypad to enter your commands and select choices, and the System will use voice responses.

For more advanced configuration, refer to "Chapter 6: Using the Web-based Utility."



**NOTE:** If your ITSP sent you the System, then it may be pre-configured for you, and you do not need to change any settings. Refer to the instructions supplied by your ITSP for more information.

## Accessing the Interactive Voice Response Menu

1. Use a telephone connected to the Phone 1 or Phone 2 port of the System. (You can only access the Interactive Voice Response Menu through an analog telephone, not any of the Internet phones.)
2. Press \*\*\*\* (in other words, press the star key four times).
3. Wait until you hear "Linksys configuration menu. Please enter the option followed by the # (pound) key or hang up to exit."
4. Refer to the following table that lists actions, commands, menu choices, and descriptions. After you select an option, press the # (pound) key. To exit the menu, hang up the telephone.

## Using the Interactive Voice Response Menu

While entering a value, such as an IP address, you may exit without entering any changes. Press the \* (star) key twice within half a second. Otherwise, the \* will be treated as a decimal point or dot.

After entering a value, such as an IP address, press the # (pound) key to indicate you have finished your selection. To save the new setting, press **1**. To review the new setting, press **2**. To re-enter the new setting, press **3**. To cancel your entry and return to the main menu, press \* (star).

## IP Telephony System

For example, to enter the IP address *191.168.1.105* by keypad, press these keys: **191\*168\*1\*105**. Press the # (pound) key to indicate that you have finished entering the IP address. Then press **1** to save the IP address or press the \* (star) key to cancel your entry and return to the main menu.

If the menu is inactive for more than one minute, the System will time out. You will need to re-enter the menu by pressing \*\*\*\*.

The settings you have saved will take effect after you have hung up the telephone. The System may reboot at this time.

### Interactive Voice Response Menu

Action	Command (press these keys on the telephone)	Choices	Description
Enter Interactive Voice Response Menu	****		Use this command to enter the Interactive Voice Response Menu. Do not press any other keys until you hear, "Linksys configuration menu. Please enter the option followed by the # (pound) key or hang up to exit."
Check Internet Connection Type	100		Hear the Internet connection type of the System.
Check Internet IP Address	110		Hear the IP address assigned to the System's Internet (external) interface.
Check Network Mask (or Subnet Mask)	120		Hear the network or subnet mask assigned to the System.
Check Gateway IP Address	130		Hear the IP address of the gateway (usually the network router).
Check MAC Address	140		Hear the MAC address of the System in hexadecimal string format.
Check Firmware Version	150		Hear the version number of the firmware currently running on the System.

**ip (internet protocol):** a protocol used to send data over a network.

**ip address:** the address used to identify a computer or device on a network.

**subnet mask:** an address code that determines the size of the network.

**gateway:** a device that forwards Internet traffic from your local area network.

**mac address:** the unique address that a manufacturer assigns to each networking device.

**firmware:** the programming code that runs a networking device.

## Interactive Voice Response Menu

Action	Command (press these keys on the telephone)	Choices	Description
Check Primary DNS Server IP Address	<b>160</b>		Hear the IP address of the primary DNS (Domain Name Service) server.
Check Internet Web Server Port	<b>170</b>		Hear the port number of the Internet Web server used for the Web-based Utility.
Check Local IP Address	<b>210</b>		Hear the local IP address of the System.
Set Internet Connection Type	<b>101</b>	Press <b>0</b> to use DHCP. Press <b>1</b> to use a static IP address. Press <b>2</b> to use PPPoE.	Select the type of Internet connection you are using. Refer to the documentation supplied by your Internet service provider.
Set Static IP Address	<b>111</b>	Enter the IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	First, set the Internet Connection Type to static IP address; otherwise, you will hear, "Invalid Option," if you try to set the static IP address.
Set Network (or Subnet) Mask	<b>121</b>	Enter the network or subnet mask using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	First, set the Internet Connection Type to static IP address; otherwise, you will hear, "Invalid Option," if you try to set the network or subnet mask.
Set Gateway IP Address	<b>131</b>	Enter the IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	First, set the Internet Connection Type to static IP address; otherwise, you will hear, "Invalid Option," if you try to set the gateway IP address.

**dhcp** (dynamic host configuration protocol): a protocol that lets one device on a local network, known as a DHCP server, assign temporary IP addresses to the other network devices, typically computers.

**static ip address:** a fixed address assigned to a computer or device that is connected to a network.

**pppoe:** a type of broadband connection that provides authentication (username and password) in addition to data transport.

## Interactive Voice Response Menu

Action	Command (press these keys on the telephone)	Choices	Description
Set Primary DNS Server IP Address	<b>161</b>	Enter the IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	First, set the Internet Connection Type to static IP address; otherwise, you will hear, "Invalid Option," if you try to set the IP address of the primary DNS server.
Set the Mode	<b>201</b>	Press <b>0</b> to select the router/NAT mode. Press <b>1</b> to select the bridge/switch mode.	Use the router/NAT mode when the Internet phones are on the Local Area Network (LAN) side.  Use the bridge/switch mode when the Internet phones are on the Wide Area Network (WAN) side.
Configure Auto-Attendant Messages	<b>72255</b>		Refer to the "Configuring the Auto-Attendant Messages" section at the end of this chapter.
Enable/Disable WAN Access to the Web-based Utility	<b>7932</b>	Press <b>1</b> to enable. Press <b>0</b> to disable.	Use this setting to enable or disable WAN access to the Web-based Utility. (This Utility lets you configure the System.)
Manual Reboot	<b>732668</b>		After you hear, "Option successful," hang up the phone. The System will automatically reboot.
Factory Reset	<b>73738</b>	Press <b>1</b> to confirm. Press * (star) to cancel.	If necessary, enter the password. The System will request confirmation; enter <b>1</b> to confirm. You will hear, "Option successful." Hang up the phone. The System will reboot, and all settings will be reset to their factory default settings.



**NOTE:** This feature may be protected by a password available only from your ITSP.

If you need to enter a password, refer to the following section, "Entering a Password."

### Interactive Voice Response Menu

Action	Command (press these keys on the telephone)	Choices	Description
Change Auto-Attendant	<b>79228</b>	Press <b>0</b> to use the auto-attendant based on day and time. Press <b>1</b> to use the Daytime Auto-Attendant. Press <b>2</b> to use the Nighttime Auto-Attendant. Press <b>3</b> to use the Weekend/Holiday Auto-Attendant.	Use this setting to select the auto-attendant you want to use. You can have the auto-attendant change depending on the day and time, or you can use one auto-attendant for all days and hours. (Make sure the auto-attendant you select has been enabled through the Web-based Utility; otherwise, the auto-attendant feature will not work.)  For more information, refer to "Chapter 6: Using the Web-based Utility."
User Factory Reset	<b>877778</b>	Press <b>1</b> to confirm. Press * (star) to cancel.	The System will request confirmation; enter <b>1</b> to confirm. You will hear, "Option successful." Hang up the phone. The System will reboot and all user-configurable settings will be reset to their factory default settings.

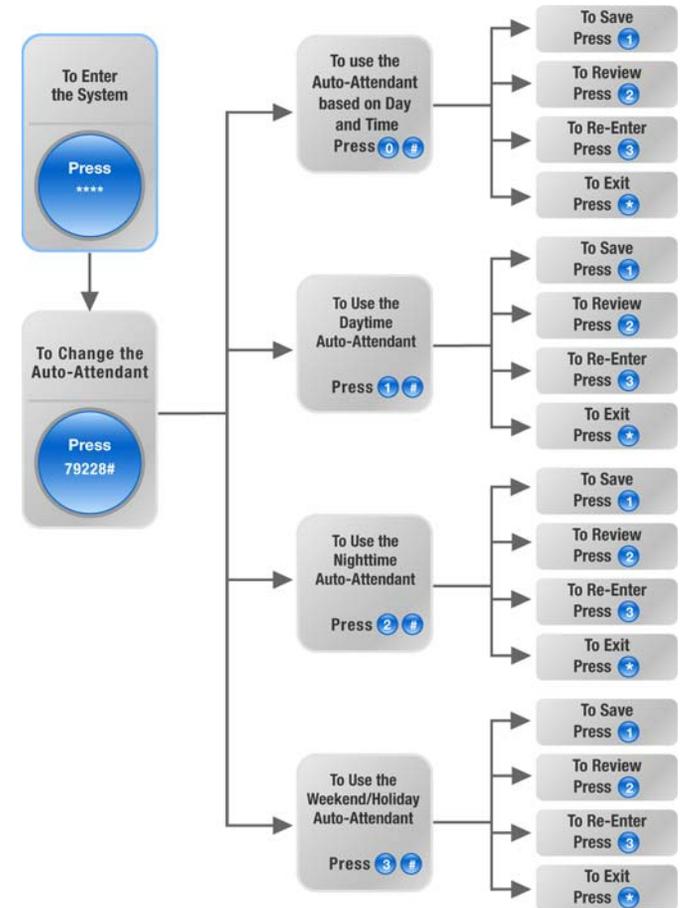


Figure 5-1: Auto-Attendant Options

## Entering a Password

You may be prompted to enter a password when you want to reset the System to its factory default settings. To enter the password, use the phone's keypad, and follow the appropriate instructions.

- To enter A, B, C, a, b, or c — press **2**.
- To enter D, E, F, d, e, or f — press **3**.
- To enter G, H, I, g, h, or i — press **4**.
- To enter J, K, L, j, k, or l — press **5**.
- To enter M, N, O, m, n, or o — press **6**.
- To enter P, Q, R, S, o, q, r, or s — press **7**.
- To enter T, U, V, t, u, or v — press **8**.
- To enter W, X, Y, Z, w, x, y, or z — press **9**.
- To enter all other characters, press **0**.



**NOTE:** These bulleted instructions only apply when you are entering a password. At all other times, pressing a number only selects a number, not a letter or punctuation mark.

For example, to enter the password *phone@321* by keypad, press these keys: **746630321**. Then press the # (pound) key to indicate that you have finished entering the password. To cancel your entry and return to the main menu, press \* (star).

## Configuring the Settings for Your Internet Phone Service

If you want to change the settings for your Internet phone service, refer to the instructions provided by your ITSP and “Chapter 6: Using the Web-based Utility.”

## Configuring the Auto-Attendant Messages

The System provides a feature called the auto-attendant, which automatically answers incoming calls with greetings or directory messages. It can handle up to 10 incoming calls and uses the default user ID **aa**.

### Auto-Attendant Messages

You can save up to 10 customized greetings. The first four have default messages, which can be changed through the Interactive Voice Response Menu.

Prompt ID	Default Audio Message
1	"If you know your party's extension, you may enter it now."
2	"Your call has been forwarded."
3	"Not a valid extension, please try again."
4	"Goodbye."

The recorded messages will be encoded with G711U and saved in flash memory. These messages will be erased whenever you reset the System to its factory default settings. The maximum length of any message is one minute. You can record up to 94.5 seconds of audio, excluding the default messages. When there is not enough memory left, the Interactive Voice Response Menu will automatically end the recording.

You can access the auto-attendant prompt settings through the Interactive Voice Response Menu.

- Using one of the analog telephones connected to the System, press **\*\*\*\*** (in other words, press the star key four times).
- Wait until you hear "Linksys configuration menu. Please enter the option followed by the # (pound) key or hang up to exit."
- Press **72255#** to access the auto-attendant message settings.
- You will hear, "Please enter the message number followed by the # (pound) key." Enter the number of the message you wish to record, review, or delete.
- The Interactive Voice Response Menu will say, "Enter **1** to record. Enter **2** to review. Enter **3** to delete. Enter **\*** to exit." Follow the instructions for your selection.

### 1 to Record

- a. If you entered 1, you will hear, “You may record your message after the tone. When finished, press #.”
- b. After you record the message, you will hear, “To save, enter 1. To review, enter 2. To re-record, enter 3. To exit, enter \*.”
- c. Follow the instructions for the entry you have selected.

If you entered 1, the new message will be saved. You will be returned to the menu described in step 5.

If you entered 2, you will hear the message played. You will be returned to the menu described in step b.

If you entered 3, you will be returned to the menu in step a.

If you entered \*, you will be returned to the menu in step 5.

### 2 to Review

If you entered 2, you will hear the message played. You will be returned to the menu described in step 5.

### 3 to Delete

- a. If you entered 3, you will hear, “Enter 1 to confirm; enter \* to exit.”
- b. If you entered 1, the message will be erased. You will be returned to the menu described in step 5.

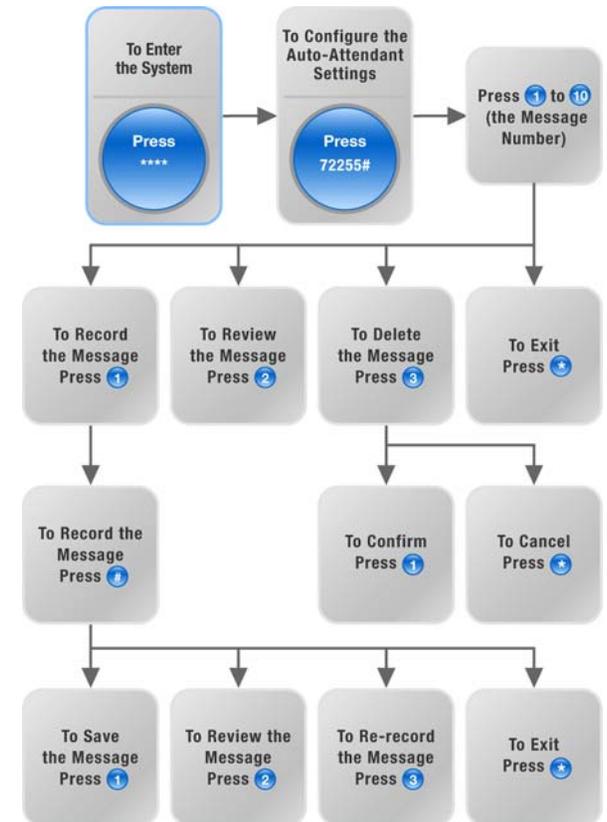
If you entered \*, you will be returned to the previous menu described in step 5.

### \* to Exit

If you entered \*, you will be returned to the previous menu in step 4.

Through the Web-based Utility, you can configure the auto-attendant to answer calls in a specific number of seconds. By default, the auto-attendant answer delay is set to **12** seconds for the daytime hours, while it is set to **0** seconds for nighttime hours and weekends.

For status information about the auto-attendant messages or to configure additional settings, such as the auto-attendant answer delay, refer to “Chapter 6: Using the Web-based Utility.”



**Figure 5-2: Auto-Attendant Message Options**



**NOTE:** If there is not enough memory left to record a new message, then you will hear, “Option failed” and be returned to step 4.



**NOTE:** If the message you want to save is longer than 15 seconds, then you will hear, “One moment, please.” This indicates that it will take several seconds to save the message. After the message has been saved, you can continue to use the Interactive Voice Response Menu.

# Chapter 6: Using the Web-based Utility

## Overview

When you first install the System, Linksys strongly recommends that you use the Setup Wizard, which you can download from [www.linksys.com](http://www.linksys.com). If you do not wish to run the Setup Wizard, you can use the Web-based Utility to configure the System.

The System may have been pre-configured by your Internet Telephony Service Provider (ITSP), so you may not have to make any changes. If you do wish to make changes, follow the instructions in this chapter.

The Web-based Utility offers two levels of access: user and admin (administrator). Your level of access depends on your service provider's policies. Also, access to some settings may be protected or blocked, so that service settings cannot be accidentally changed. For more information, contact your ITSP.

This chapter will describe each web page of the Web-based Utility and each page's key functions. The Internet connection settings are configured on the *Router - WAN Setup* screen, while some of the most popular features: auto-attendant, music-on-hold, and call hunt are configured on the *Voice - SIP* screen. The Utility can be accessed via your web browser through use of a computer on your network.

There are two main tabs: Router and Voice. Additional tabs will be available after you click one of the main tabs.

## Router

- **Status.** This screen displays routing information about the System.
- **WAN Setup.** Use this screen to configure the Internet connection, MAC clone, remote management, QoS, VLAN, and optional settings.
- **LAN Setup.** Use this screen to configure the local network, dynamic DHCP, and static DHCP lease settings.
- **Application.** On this screen, configure port forwarding, DMZ, and reserved ports range settings.

## Voice

- **Info.** This screen displays voice-related information about the System.
- **System.** Use this screen to configure system settings. In most cases, you should not change these settings unless instructed to do by your ITSP.



**NOTE:** If you are not sure how to configure the settings, then keep the default settings.

## IP Telephony System

- **SIP.** Configure service, music-on-hold, group paging, call hunt, and auto-attendant settings on this screen. In most cases, do not change service settings unless instructed to do so by your ITSP.
- **Provisioning.** Use this screen to configure service provisioning settings. In most cases, you should not change these settings unless instructed to do by your ITSP.
- **Regional.** Use this screen to configure call settings. In most cases, you should not change these settings unless instructed to do by your ITSP.
- **FXS 1/2.** Use the appropriate screen to configure settings for each FXS (Phone) port on the System.
- **Line 1/2/3/4.** Use the appropriate screen to configure settings for each external Internet phone line.

## How to Access the Web-based Utility

To access the Web-based Utility of the System, launch Internet Explorer or Netscape Navigator on the administration computer connected to the System's Ethernet port. If the System uses its default address, then enter **192.168.0.1** in the *Address* field. If you have assigned a static IP address to the System, then enter **<IP address of the System>** in the *Address* field. Press the **Enter** key.

Enter your user name and password. The default user name for administrative access is **admin**, and the default user name for user access is **user**. (These user names cannot be changed.) Then enter the password supplied by your ITSP. (By default, there is no password, so if you were not given a password, then leave this field blank.)

To view the status information for the phones and their calls, click **PBX Status**. To switch to a different login, click **User Login** or **Admin Login**. Enter the appropriate login information. Two views of the Web-based Utility are available. Click **basic** to view basic settings, or click **advanced** to view advanced settings.

When you have finished making changes on a screen, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes. When changes are saved, the System may reboot.

## The PBX Status Screen

This screen shows status information for the phones and their calls.

### Registration

This section shows the registration information for the phones.

**Registration.** To remove a phone's registration, click its checkbox. Then click the **Delete** button.



**NOTE:** If your ITSP supplied the System, then it may be pre-configured for you, and you do not need to change any settings. Refer to the instructions supplied by your ITSP for more information.

## IP Telephony System

**Station.** Shown here is the station name assigned to the phone. (This setting is configured through the phone.)

**User ID.** Shown here is the extension number assigned to the phone.

**IP Address.** Shown here is the local IP address of the phone.

**Reg Expires.** This indicates the number of seconds left before the phone needs to re-register with the System.

## Parking Lot

This section shows the calls that have been parked. Call park is a convenient feature that lets a call be put on hold and picked up from any extension number.

**Parking Lot.** To remove a call from the Parking Lot, click its checkbox. Then click the **Delete** button.

**Caller ID.** Shown here is the phone number of the caller.

**Parked By.** Shown here is the extension number that parked the call.

**Parked At.** Shown here is the call park number that you should use to pick up this call.

**Duration.** Shown here is the length of time that the call has been parked.

## Line 1 Calls

This section shows the current incoming and outgoing calls.

**Line 1 Calls.** To remove a call, click its checkbox. Then click the **Delete** button.

**External.** Shown here is the external phone number of the caller.

**Station.** Shown here is the extension number of the call; it displays the word “callpark” when the call has been parked for pickup from any extension number.

**Direction.** Shown here is the direction of the call, Inbound or Outbound.

**State.** Shown here is the status of the call, Connected or Proceeding.

**Duration.** Shown here is the length of time the call has been active.

delete					
Registration	Station	User ID	IP Address	Reg Expires(s)	
<input type="checkbox"/>	Receptionist	200	192.168.0.4	1860	
<input type="checkbox"/>	Oscar	200	192.168.0.2	1859	
<input type="checkbox"/>	BigBird	200	192.168.0.3	1859	
<input type="checkbox"/>	BigBird	300	192.168.0.3	1859	
<input type="checkbox"/>	Oscar	300	192.168.0.2	1859	
<input type="checkbox"/>	Receptionist	501	192.168.0.4	1860	
<input type="checkbox"/>	Oscar	502	192.168.0.2	41	
<input type="checkbox"/>	BigBird	503	192.168.0.3	41	
<input type="checkbox"/>		505	192.168.0.6	2158	
Parking Lot	Caller ID	Parked By	Parked At	Duration	
<input type="checkbox"/>	19495551319	501	1	00:00:14	
Line 1 Calls	External	Station	Direction	State	Duration
<input type="checkbox"/>	19495551319	callpark	Inbound	Connected	00:00:28

Figure 6-1: PBX Screen - Parking Lot

delete					
Registration	Station	User ID	IP Address	Reg Expires(s)	
<input type="checkbox"/>	Receptionist	200	192.168.0.4	1883	
<input type="checkbox"/>	Oscar	200	192.168.0.2	1882	
<input type="checkbox"/>	BigBird	200	192.168.0.3	1882	
<input type="checkbox"/>	BigBird	300	192.168.0.3	1882	
<input type="checkbox"/>	Oscar	300	192.168.0.2	1882	
<input type="checkbox"/>	Receptionist	501	192.168.0.4	1883	
<input type="checkbox"/>	Oscar	502	192.168.0.2	64	
<input type="checkbox"/>	BigBird	503	192.168.0.3	64	
<input type="checkbox"/>		505	192.168.0.6	2181	
Line 1 Calls	External	Station	Direction	State	Duration
<input type="checkbox"/>	19495551319	501	Inbound	Connected	00:00:05

Figure 6-2: PBX Screen - Inbound Call

delete					
Registration	Station	User ID	IP Address	Reg Expires(s)	
<input type="checkbox"/>	Receptionist	200	192.168.0.4	1662	
<input type="checkbox"/>	Oscar	200	192.168.0.2	1661	
<input type="checkbox"/>	BigBird	200	192.168.0.3	1661	
<input type="checkbox"/>	BigBird	300	192.168.0.3	1661	
<input type="checkbox"/>	Oscar	300	192.168.0.2	1661	
<input type="checkbox"/>	Receptionist	501	192.168.0.4	1662	
<input type="checkbox"/>	Oscar	502	192.168.0.2	39	
<input type="checkbox"/>	BigBird	503	192.168.0.3	39	
<input type="checkbox"/>		505	192.168.0.6	1960	
Line 1 Calls	External	Station	Direction	State	Duration
<input type="checkbox"/>	19495551319	8001	Outbound	Proceeding	

Figure 6-3: PBX Screen - Outbound Call

## The Router Tab

### The Router - Status Screen

This screen displays product and system information.

#### Product Information

**Product Name.** Shown here is the model number of the System.

**Serial Number.** Shown here is the serial number of the System.

**Software Version.** Shown here is the version number of the System software.

**Hardware Version.** Shown here is the version number of the System hardware.

**MAC Address.** Shown here is the MAC address of the System.

**Client Certificate.** Shown here is the status of the client certificate. It authenticates the System for use in the ITSP's network.

**Licenses.** This indicates how many additional licenses you have acquired for the System.

#### System Status

**Current Time.** Displayed here is the current date and time of the System.

**Elapsed Time.** Displayed here is the amount of time elapsed since the last reboot of the System.

**WAN Connection Type.** Displayed here is the Internet connection type of the System.

**Current IP.** Displayed here is the Internet IP address of the System.

**Host Name.** Displayed here is the host name of the System.

**Domain.** Displayed here is the domain name of the System.

**Current Netmask.** Displayed here is the netmask or subnet mask of the System.

**Current Gateway.** Displayed here is the IP address of the gateway.

**Primary DNS.** Displayed here is the IP address of the primary DNS server.



Figure 6-4: Router - Status Screen

**mac address:** the unique address that a manufacturer assigns to each networking device.

**ip (internet protocol):** a protocol used to send data over a network.

**ip address:** the address used to identify a computer or device on a network.

**subnet mask:** an address code that determines the size of the network.

**gateway:** a device that forwards Internet traffic from your local area network.

## IP Telephony System

**Secondary DNS.** Displayed here is the IP address of the secondary DNS server.

**LAN IP Address.** Displayed here is the local IP address of the System.

**Broadcast Pkts Sent.** Displayed here is the number of broadcast packets sent.

**Broadcast Bytes Sent.** Displayed here is the number of broadcast bytes sent.

**Broadcast Pkts Recv.** Displayed here is the number of broadcast packets received and processed.

**Broadcast Bytes Recv.** Displayed here is the number of broadcast bytes received and processed.

**Broadcast Pkts Dropped.** Displayed here is the number of broadcast packets received but not processed.

**Broadcast Bytes Dropped.** Displayed here is the number of broadcast bytes received but not processed.

***packet:*** a unit of data sent over a network.

## The Router - WAN Setup Screen

This screen lets you configure the Internet connection, MAC clone, remote management, QoS, VLAN, and optional settings. Information about your Internet connection type should be provided by your Internet Service Provider (ISP). If you do not have this information, contact your service provider.

### Internet Connection Settings

**Connection Type.** Select the connection type you use: **DHCP**, **Static IP**, or **PPPOE**.

If you already have a router for your network, select **Static IP** and assign an address that is appropriate for your network. (Refer to the router's documentation for more information about IP addressing.)

### Static IP Settings

If you selected Static IP, complete the Static IP Settings section.

**Static IP.** Enter the static or fixed IP address of the System (this should be provided by your ISP).

**NetMask.** Enter the net or subnet mask of the System (this should be provided by your ISP).

**Gateway.** Enter the IP address of the gateway (this should be provided by your ISP).

### PPPOE Settings

If you selected PPPOE, complete the PPPOE Settings section.

**PPPoE Login Name.** Enter the name provided by your ISP.

**PPPOE Login Password.** Enter the password provided by your ISP.

**PPPOE Service Name** (optional). Enter the service name provided by your ISP.

### Optional Settings

**HostName.** Enter the host name, if provided by your ISP.

**Domain.** Enter the domain name, if provided by your ISP.

**Primary DNS.** Enter the IP address of the primary DNS server.

**Secondary DNS** (optional). Enter the IP address of the secondary DNS server.

The screenshot shows the 'Router' tab with the 'WAN Setup' sub-tab selected. The interface is organized into several sections:

- Internet Connection Settings:** Connection Type is set to 'DHCP'.
- Static IP Settings:** Static IP is '192.168.10.1' and NetMask is '255.255.255.0'.
- PPPOE Settings:** Fields for Login Name, Login Password, and Service Name are present.
- Optional Settings:** Fields for HostName, Domain, Primary DNS, Secondary DNS, DNS Server Order (set to 'Manual'), DNS Query Mode (set to 'Parallel'), Primary NTP Server, and Secondary NTP Server.
- MAC Clone Settings:** Enable MAC Clone Service is set to 'no'.
- Remote Management:** Enable WAN Web Server is set to 'no'.
- QoS Settings:** QoS QDisc is set to 'NONE' and Maximum Uplink Speed is '128 (Kbps)'.
- VLAN Settings:** Enable VLAN is set to 'no' and VLAN ID is '1'.

At the bottom, there are buttons for 'Undo All Changes' and 'Submit All Changes', along with a 'Fix Status' link and user login options.

Figure 6-5: Router - WAN Setup Screen

**dhcp** (dynamic host configuration protocol): a protocol that lets one device on a local network, known as a DHCP server, assign temporary IP addresses to the other network devices, typically computers.

**static ip address:** a fixed address assigned to a computer or device that is connected to a network.

**pppoe:** a type of broadband connection that provides authentication (username and password) in addition to data transport

## IP Telephony System

**DNS Server Order.** Select the order in which the DNS servers should be used: **Manual**; **Manual, DHCP**; or **DHCP, Manual**. The default is **Manual**.

**DNS Query Mode.** Select how the DNS servers should be queried: **Parallel** or **Sequential**. The default is **Parallel**.

**Primary NTP Server.** Enter the IP address of the primary NTP server, which the System uses to keep the date and time current.

**Secondary NTP Server** (optional). Enter the IP address of the secondary NTP server.

## MAC Clone Settings

**Enable MAC Clone Service.** Select whether you want to clone a MAC address onto the System, **yes** or **no**. The default is **no**.

**Cloned MAC Address.** Enter the MAC address you want to clone.

## Remote Management

**Enable WAN Web Server.** This feature lets you enable or disable access to the Web-based Utility from the WAN side. Select **yes** or **no** from the drop-down menu. The default is **no**.

**WAN Web Server Port.** Enter the port number used to access the Utility from the WAN side. The default is **80**.

## QoS Settings

**QoS QDisc.** QoS prioritizes voice communications when different types of traffic are competing for bandwidth. Select the method you want to use: **NONE**, **CBQ**, or **TBF**. The default is **NONE**.

**Maximum Uplink Speed.** Enter the maximum upload speed of your Internet connection. The default is **128Kbps**.

## VLAN (Virtual Local Area Network) Settings

**Enable VLAN.** VLAN (802.1Q) settings let you use the System in a virtual LAN environment. Select **yes** or **no** from the drop-down menu. The default is **no**.

**VLAN ID.** Enter the ID number used by the System. The default is **1**.

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.

## The Router - LAN Setup Screen

This screen lets you configure the local network, dynamic DHCP, and static DHCP lease settings.

**Networking Service.** Select the service you want to use, **NAT** or **Bridge**. The default is **NAT**.

### LAN Network Settings

**LAN IP Address.** Enter the local IP address of the System. The default is **192.168.0.1**.

**LAN Subnet Mask.** Select the local subnet mask: **255.255.255.0**, **255.255.255.128**, **255.255.255.192**, **255.255.255.224**, **255.255.255.240**, **255.255.255.248**, or **255.255.255.252**. The default is **255.255.255.0**.

**Enable DHCP Server.** To use the System as a router assigning IP addresses, select **yes**. Otherwise, select **no**. The default is **yes**.

**DHCP Lease Time.** Enter the lease time used by the System to distribute IP addresses. The default is **24 Hours**.

**DHCP Client Starting IP Address.** When the System issues IP addresses, it starts with the first value of its DHCP client IP address range. Enter that value here. The default is **192.168.0.2**.

**Number of Client IP Addresses.** Enter the number of IP addresses that can be distributed. The default is **50**.

### Static DHCP Lease Settings

**Enable.** You can have the System assign the same IP address to a specific device. To disable this feature, select **no**. To use this feature, select **yes**. The default is **no**.

**Host MAC Address.** Enter the MAC address of the device whose IP address you want to specify.

**Host IP Address.** Enter the IP address you want to assign to the device, 192.168.0.x (x being a different number for each device you specify).

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.

The screenshot shows the 'LAN Setup' screen of a router. At the top, there are tabs for 'Router', 'Voice', 'Status', 'Wan Setup', 'LAN Setup', and 'Application'. The 'LAN Setup' tab is active. Below the tabs, there are links for 'Full Status', 'User Login', 'Basic', and 'Advanced'. The main configuration area includes:
 

- Networking Service:** A dropdown menu set to 'NAT'.
- LAN Network Settings:**
  - LAN IP Address:** 192.168.0.1
  - LAN Subnet Mask:** 255.255.255.0
- Enable DHCP Server:** A dropdown menu set to 'yes'.
- DHCP Lease Time:** 24 Hours
- DHCP Client Starting IP Address:** 192.168.0.2
- Number of Client IP Addresses:** 50

 Below these are the **Static DHCP Lease Settings**, which is a table with columns for 'Enable', 'Host Mac Address', and 'Host IP Address'. The table contains 10 rows, each with a 'no' in the 'Enable' column, '0000000000' in the 'Host Mac Address' column, and '192.168.0.0' in the 'Host IP Address' column. At the bottom of the form are two buttons: 'Undo All Changes' and 'Submit All Changes'. At the very bottom, there are links for 'Full Status', 'User Login', 'Basic', and 'Advanced'.

Figure 6-6: Router - LAN Setup Screen

## The Router - Application Screen

This screen lets you configure port forwarding, DMZ, and reserved ports range settings.

### Port Forwarding Settings

**Enable.** Select **yes** or **no** for each port forwarding entry, which defines a port range to be forwarded to a server. The default is **no**.

**Service Name.** Enter the name of the service or application.

**Starting Port.** Enter the starting port number of the forwarded port range.

**Ending Port.** Enter the ending port number of the forwarded port range.

**Protocol.** Select the protocol used, **TCP**, **UDP**, or **Both**. The default is **TCP**.

**Server IP Address.** Enter the IP address of the server, 192.168.0.x (x being a different number for each server you specify).

### DMZ Settings

**Enable DMZ.** DMZ hosting forwards all ports at the same time to one computer. This allows one local user to be exposed to the Internet for use of special-purpose services such as videoconferencing. Select **yes** or **no** from the drop-down menu. The default is **no**.

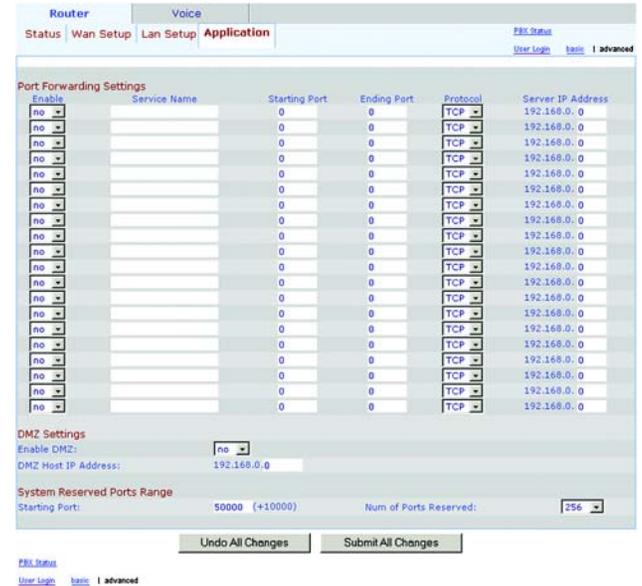
**DMZ Host IP Address.** Enter the IP address of the DMZ host, 192.168.0.x (x being the number for the computer you want to specify). Use the Static DHCP Lease Settings section on the *LAN Setup* screen, so the DMZ Host keeps this IP address; otherwise, its IP address may change.

### System Reserved Ports Range

**Starting Port.** This port range defines the random TCP/UDP ports used by the application running on the System. They cannot be used by port forwarding or DMZ. Enter the starting port number of the reserved ports range. The default is **50000**.

**Num of Ports Reserved.** Select the number of ports you want to reserve: **256**, **512**, or **1024**. The default is **256**.

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.



**Figure 6-7: Router - Application Screen**

**tcp:** a network protocol for transmitting data that requires acknowledgement from the recipient of data sent.

**udp:** a network protocol for transmitting data that does not require acknowledgement from the recipient of the data that is sent.

## The Voice Tab

### The Voice - Info Screen

This screen shows voice-related settings for the System.

#### Product Information

**Product Name.** Shown here is the model number of the System.

**Serial Number.** Shown here is the serial number of the System.

**Software Version.** Shown here is the version number of the System software.

**Hardware Version.** Shown here is the version number of the System hardware.

**MAC Address.** Shown here is the MAC address of the System.

**Client Certificate.** Shown here is the status of the client certificate, which indicates that the System has been authorized by your ITSP.

**Licenses.** This indicates how many additional licenses you have acquired for the System.

#### System Status

**Current Time.** Displayed here is the current date and time of the System.

**Elapsed Time.** Displayed here is the amount of time elapsed since the last reboot of the System.

#### FXS 1/2 Status

The FXS 1 and FXS 2 ports are the Phone ports of the System. (You can connect analog phones or fax machines to both ports.) They have the same status information available.

**Hook State.** Displayed here is the status of the phone's readiness. **On** indicates that the phone is ready for use, while **Off** indicates that the phone is in use.

**Message Waiting.** This indicates whether you have new voicemail waiting.

**Call Back Active.** This indicates whether a call back request is in progress.

**Last Called Number.** Displayed here is the last number called.

Router		Voice									
Info	System	SIP	Provisioning	Regional	FXS 1	FXS 2	Line 1	Line 2	Line 3	Line 4	FXS Status
User Login <a href="#">Home</a> <a href="#">Advanced</a>											
<b>Product Information</b>											
Product Name:	SPA-9000			Serial Number:	88016GA00020						
Software Version:	3.2.3(0)			Hardware Version:	1.0.0(5439)						
MAC Address:	00E08EC76EC			Client Certificate:	Installed						
Licenses:	None										

**Figure 6-8: Voice - Info Screen - Product Information**

System Status	
Current Time:	12/31/2002 00:26:17
Elapsed Time:	02:31:42

**Figure 6-9: Voice - Info Screen - System Status**

## IP Telephony System

**Last Caller Number.** Displayed here is the number of the last caller.

Calls 1 and 2 have the same status information available.

**Call 1/2 State.** Displayed here is the status of the call.

**Call 1/2 Tone.** Displayed here is the type of tone used by the call.

**Call 1/2 Encoder.** Displayed here is the codec used for encoding.

**Call 1/2 Decoder.** Displayed here is the codec used for decoding.

**Call 1/2 FAX.** Displayed here is the status of the fax pass-through mode.

**Call 1/2 Type.** Displayed here is the direction of the call.

**Call 1/2 Remote Hold.** This indicates whether the far end has placed the call on hold.

**Call 1/2 Callback.** This indicates whether the call was triggered by a call back request.

**Call 1/2 Peer Name.** Displayed here is the name of the internal phone.

**Call 1/2 Peer Phone.** Displayed here is the phone number of the internal phone.

**Call 1/2 Duration.** Displayed here is the duration of the call.

**Call 1/2 Packets Sent.** Displayed here is the number of packets sent.

**Call 1/2 Packets Recv.** Displayed here is the number of packets received.

**Call 1/2 Bytes Sent.** Displayed here is the number of bytes sent.

**Call 1/2 Bytes Recv.** Displayed here is the number of bytes received.

**Call 1/2 Decode Latency.** Displayed here is the number of milliseconds for decoder latency.

**Call 1/2 Jitter.** Displayed here is the number of milliseconds for receiver jitter.

**Call 1/2 Round Trip Delay.** Displayed here is the number of milliseconds for delay.

**Call 1/2 Packets Lost.** Displayed here is the number of packets lost.

**Call 1/2 Packet Error.** Displayed here is the number of invalid packets received.

FXS 1 Status			
Hook State:	Off	Message Waiting:	No
Call Back Active:	No	Last Called Number:	919495551319
Last Caller Number:		Call 1 State:	Idle
Call 1 State:	Idle	Call 2 State:	Proceeding
Call 1 Tone:	None	Call 2 Tone:	Ring Back
Call 1 Encoder:		Call 2 Encoder:	G711u
Call 1 Decoder:		Call 2 Decoder:	G711u
Call 1 FAX:		Call 2 FAX:	No
Call 1 Type:		Call 2 Type:	Outbound
Call 1 Remote Hold:		Call 2 Remote Hold:	No
Call 1 Callback:		Call 2 Callback:	No
Call 1 Peer Name:		Call 2 Peer Name:	
Call 1 Peer Phone:		Call 2 Peer Phone:	919495551319
Call 1 Duration:		Call 2 Duration:	
Call 1 Packets Sent:		Call 2 Packets Sent:	0
Call 1 Packets Recv:		Call 2 Packets Recv:	404
Call 1 Bytes Sent:		Call 2 Bytes Sent:	0
Call 1 Bytes Recv:		Call 2 Bytes Recv:	116160
Call 1 Decode Latency:		Call 2 Decode Latency:	110 ms
Call 1 Jitter:		Call 2 Jitter:	0 ms
Call 1 Round Trip Delay:		Call 2 Round Trip Delay:	0 ms
Call 1 Packets Lost:		Call 2 Packets Lost:	0
Call 1 Packet Error:		Call 2 Packet Error:	0
FXS 2 Status			
Hook State:	On	Message Waiting:	No
Call Back Active:	No	Last Called Number:	
Last Caller Number:		Call 2 State:	Idle
Call 1 State:	Idle	Call 2 Tone:	None
Call 1 Tone:	None	Call 2 Encoder:	
Call 1 Encoder:		Call 2 Decoder:	
Call 1 Decoder:		Call 2 FAX:	
Call 1 FAX:		Call 2 Type:	
Call 1 Type:		Call 2 Remote Hold:	
Call 1 Remote Hold:		Call 2 Callback:	
Call 1 Callback:		Call 2 Peer Name:	
Call 1 Peer Name:		Call 2 Peer Phone:	
Call 1 Peer Phone:		Call 2 Duration:	
Call 1 Duration:		Call 2 Packets Sent:	
Call 1 Packets Sent:		Call 2 Packets Recv:	
Call 1 Packets Recv:		Call 2 Bytes Sent:	
Call 1 Bytes Sent:		Call 2 Bytes Recv:	
Call 1 Bytes Recv:		Call 2 Decode Latency:	
Call 1 Decode Latency:		Call 2 Jitter:	
Call 1 Jitter:		Call 2 Round Trip Delay:	
Call 1 Round Trip Delay:		Call 2 Packets Lost:	
Call 1 Packets Lost:		Call 2 Packet Error:	
Call 1 Packet Error:			

Figure 6-10: Voice - Info Screen - FXS Status

## Line 1/2/3/4 Status

Lines 1, 2, 3, and 4 have the same status information available.

**Registration State.** Shown here is the status of the line's registration with the ITSP.

**Last Registration At.** Shown here are the last date and time the line was registered.

**Next Registration In.** Shown here is the number of seconds until the next registration.

**Message Waiting.** This indicates whether you have new voicemail waiting.

**Mapped SIP Port.** Shown here is the port number of the mapped SIP port.

## Auto Attendant Prompt Status

**Prompt 1-4.** The first four greetings or messages are defaults. If you change a default, then the screen will show the new prompt's duration in milliseconds.

**Prompt 5-10.** For each prompt, the screen shows its duration in milliseconds.

**Space Remaining.** Shown here is the number of milliseconds available.

**Current AA.** Shown here is the auto-attendant in use.

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.

Line 1 Status			
Registration State:	Registered	Last Registration At:	12/31/2002 00:26:09
Next Registration In:	10 s	Message Waiting:	No
Mapped SIP Port:			
Line 2 Status			
Registration State:	Failed	Last Registration At:	0/0/0 00:00:00
Next Registration In:	22 s	Message Waiting:	No
Mapped SIP Port:			
Line 3 Status			
Registration State:	Not Registered	Last Registration At:	
Next Registration In:		Message Waiting:	No
Mapped SIP Port:			
Line 4 Status			
Registration State:	Not Registered	Last Registration At:	
Next Registration In:		Message Waiting:	No
Mapped SIP Port:			

Figure 6-11: Voice - Info Screen - Line Status

Auto Attendant Prompt Status			
Prompt 1:	Default	Prompt 2:	Default
Prompt 3:	Default	Prompt 4:	Default
Prompt 5:	5250 ms	Prompt 6:	5250 ms
Prompt 7:	15750 ms	Prompt 8:	5250 ms
Prompt 9:	15750 ms	Prompt 10:	0 ms
Space Remaining:	47250 ms	Current AA:	Daytime

[Fix Status](#)  
[User Login](#) [Home](#) | [Advanced](#)

Figure 6-12: Voice - Info Screen - Auto Attendant Prompt Status

## The Voice - System Screen

This screen lets you configure system settings.



**IMPORTANT:** In most cases, you should not change these settings unless instructed to do by your ITSP.

### System Configuration

**Restricted Access Domains.** Enter the domain names permitted to access the System.

**Enable Web Admin Access.** This setting lets you enable or disable local access to the Web-based Utility. Select **yes** or **no** from the drop-down menu. The default is **yes**.

**Admin Passwd.** Enter the password for the administrator. (By default, there is no password.)

**User Password.** Enter the password for the user. (By default, there is no password.)

### Miscellaneous Settings

**Syslog Server.** Enter the IP address of the syslog server, which logs system information and critical events of the System.

**Debug Server.** Enter the IP address of the debug server, which logs debug information of the System.

**Debug Level.** This determines the level of debug information that will be generated. Select **0**, **1**, **2**, or **3** from the drop-down menu. The higher the debug level, the more debug information will be generated. The default is **0**, which indicates that no debug information will be generated.

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.

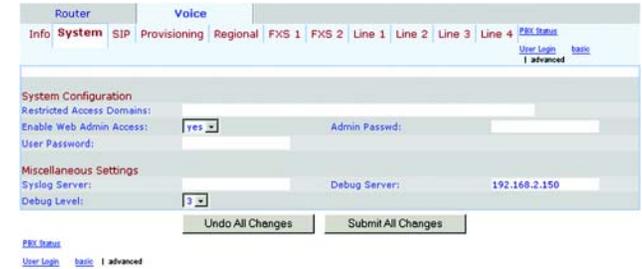


Figure 6-13: Voice - System Screen

## The Voice - SIP Screen

This screen lets you configure service, music-on-hold, group paging, call hunt, and auto-attendant settings.



**IMPORTANT:** In most cases, you should not change the service settings unless instructed to do by your ITSP.

### SIP Parameters

**Max Forward.** This is the SIP Max Forward value, which can range from 1 to 255. The default is **70**.

**Max Redirection.** This is the number of times an invite can be redirected to avoid an infinite loop. The default is **5**.

**Max Auth.** This is the maximum number of times (from 0 to 255) a request may be challenged. The default is **2**.

**SIP User Agent Name.** This is the User-Agent header used in outbound requests. The default is **\$VERSION**.

**SIP Server Name.** This is the Server header used in responses to inbound responses. The default is **\$VERSION**.

**SIP Reg User Agent Name.** This is the User-Agent name to be used in a REGISTER request. If this is not specified, then the SIP User Agent Name will also be used for the REGISTER request.

**SIP Accept Language.** This is the Accept-Language header used by the System. There is no default (this indicates the System does not include this header).

**DTMF Relay MIME Type.** This is the MIME Type used in a SIP INFO message to signal a DTMF event. The default is **application/dtmf-relay**.

**Hook Flash MIME Type.** This is the MIME Type used in a SIP INFO message to signal a hook flash event. The default is **application/hook-flash**.

**Remove Last Reg.** This feature lets you remove the last registration before registering a new one if the value is different. Select **yes** or **no** from the drop-down menu. The default is **no**.

**Use Compact Header.** This feature lets you use compact SIP headers in outbound SIP messages. Select **yes** or **no** from the drop-down menu. The default is **no**.

**Escape Display Name.** This feature lets you keep the Display Name private. Select **yes** if you want the System to enclose the string (configured in the Display Name) in a pair of double quotes for outbound SIP messages. Any

SIP Parameters	
Max Forward:	70
Max Auth:	2
SIP Server Name:	\$VERSION
SIP Reg User Agent Name:	\$VERSION
DTMF Relay MIME Type:	application/dtmf-relay
Hook Flash MIME Type:	application/hook-flash
Remove Last Reg.:	no
Use Compact Header:	no
Escape Display Name:	no

**Figure 6-14: Voice - SIP Screen - SIP Parameters**

## IP Telephony System

occurrences of “ or \ in the string will be escaped with \” and \\ inside the pair of double quotes. Otherwise, select **no**. The default is **no**.

### SIP Timer Values (sec)

**SIP T1.** This is the RFC 3261 T1 value (RTT estimate), which can range from 0 to 64 seconds. The default is **.5**.

**SIP T2.** This is the RFC 3261 T2 value (maximum retransmit interval for non-INVITE requests and INVITE responses), which can range from 0 to 64 seconds. The default is **4**.

**SIP T4.** This is the RFC 3261 T4 value (maximum duration a message will remain in the network), which can range from 0 to 64 seconds. The default is **5**.

**SIP Timer B.** This is the INVITE time-out value, which can range from 0 to 64 seconds. The default is **32**.

**SIP Timer F.** This is the non-INVITE time-out value, which can range from 0 to 64 seconds. The default is **32**.

**SIP Timer H.** This is the INVITE final response, time-out value, which can range from 0 to 64 seconds. The default is **32**.

**SIP Timer D.** This is the ACK hang-around time, which can range from 0 to 64 seconds. The default is **32**.

**SIP Timer J.** This is the non-INVITE response, hang-around time, which can range from 0 to 64 seconds. The default is **32**.

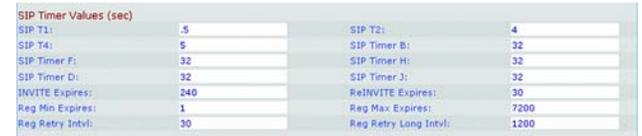
**INVITE Expires.** This is the INVITE request Expires header value. If you enter 0, then the Expires header is not included in the request. The default is **240**.

**ReINVITE Expires.** This is the ReINVITE request Expires header value. If you enter 0, then the Expires header is not included in the request. The default is **30**.

**Reg Min Expires.** This is the minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If the proxy returns a value less than this setting, then the minimum value is used. The default is **1**.

**Reg Max Expires.** This is the maximum registration expiration time allowed from the proxy in the Min-Expires header. If the value is larger than this setting, then the maximum value is used. The default is **7200**.

**Reg Retry Intvl.** This is the interval to wait before the System retries registration after failing during the last registration. The default is **30**.



SIP Timer Values (sec)			
SIP T1:	5	SIP T2:	4
SIP T4:	5	SIP Timer B:	32
SIP Timer F:	32	SIP Timer H:	32
SIP Timer D:	32	SIP Timer J:	32
INVITE Expires:	240	ReINVITE Expires:	30
Reg Min Expires:	1	Reg Max Expires:	7200
Reg Retry Intvl:	30	Reg Retry Long Intvl:	1200

**Figure 6-15: Voice - SIP Screen - SIP Timer Values**

**Reg Retry Long Intvl.** When registration fails with a SIP response code that does not match, the System will wait for the specified length of time before retrying. If this interval is 0, then the System will stop trying. This value should be much larger than the Reg Retry Intvl value. The default is **1200**.

### Response Status Code Handling

**SIT1-4 RSC.** Enter the SIP response status code for the appropriate SIT Tone (SIT stands for Special Information Tone). For example, if you set the SIT1 RSC to 404, then when the user makes a call and a failure code of 404 is returned, the SIT1 tone is played.

**Try Backup RSC.** This is the SIP response code that retries a backup server for the current request.

**Retry Reg RSC.** This is the interval to wait before the System retries registration after failing during the last registration.

### RTP Parameters

**RTP Port Min.** This is the minimum port number for RTP transmission and reception. The default is **16384**.

**RTP Port Max.** This is the maximum port number for RTP transmission and reception. The default is **16482**.

**RTP Packet Size.** This is the packet size in seconds, which can range from 0.01 to 0.16. Valid values must be a multiple of 0.01 seconds. The default is **0.030**.

**Max RTP ICMP Err.** This indicates that the RTP data stream has failed due to ICMP errors. The default is **0**.

**RTCP Tx Interval.** This is the interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. The default is **0**.

**No UDP Checksum.** Select **yes** if you want the System to calculate UDP header checksum for SIP messages. Otherwise, select **no**. The default is **no**.

**Stats in BYE.** This sets whether the System will include the P-RTP-Stat header or response to a BYE message. The header contains RTP statistics of the current call. Select **yes** or **no** from the drop-down menu. The default is **no**.

### SDP Payload Types

**NSE Dynamic Payload.** This is the NSE dynamic payload type. The default is **100**.

**AVT Dynamic Payload.** This is the AVT dynamic payload type. The default is **101**.

Response Status Code Handling			
SIT1 RSC:	<input type="text"/>	SIT2 RSC:	<input type="text"/>
SIT3 RSC:	<input type="text"/>	SIT4 RSC:	<input type="text"/>
Try Backup RSC:	<input type="text"/>	Retry Reg RSC:	<input type="text"/>

**Figure 6-16: Voice - SIP Screen - Response Status Code Handling**

RTP Parameters			
RTP Port Min:	<input type="text" value="16384"/>	RTP Port Max:	<input type="text" value="16482"/>
RTP Packet Size:	<input type="text" value="0.030"/>	Max RTP ICMP Err:	<input type="text" value="0"/>
RTCP Tx Interval:	<input type="text" value="0"/>	No UDP Checksum:	<input type="text" value="no"/>
Stats in BYE:	<input type="text" value="no"/>		

**Figure 6-17: Voice - SIP Screen - RTP Parameters**

SDP Payload Types			
NSE Dynamic Payload:	<input type="text" value="100"/>	AVT Dynamic Payload:	<input type="text" value="101"/>
INFOREQ Dynamic Payload:	<input type="text"/>	G726r16 Dynamic Payload:	<input type="text" value="98"/>
G726r24 Dynamic Payload:	<input type="text" value="97"/>	G726r40 Dynamic Payload:	<input type="text" value="96"/>
G729b Dynamic Payload:	<input type="text" value="99"/>	NSE Codec Name:	<input type="text" value="NSE"/>
AVT Codec Name:	<input type="text" value="telephone-event"/>	G711u Codec Name:	<input type="text" value="PCMU"/>
G711a Codec Name:	<input type="text" value="PCMA"/>	G726r16 Codec Name:	<input type="text" value="G726-16"/>
G726r24 Codec Name:	<input type="text" value="G726-24"/>	G726r32 Codec Name:	<input type="text" value="G726-32"/>
G726r40 Codec Name:	<input type="text" value="G726-40"/>	G729a Codec Name:	<input type="text" value="G729a"/>
G729b Codec Name:	<input type="text" value="G729ab"/>	G723 Codec Name:	<input type="text" value="G723"/>

**Figure 6-18: Voice - SIP Screen - SDP Payload Types**

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**INFOREQ Dynamic Payload.** This is the INFOREQ dynamic payload type. There is no default.

**G726r16 Dynamic Payload.** This is the G726-16 dynamic payload type. The default is **98**.

**G726r24 Dynamic Payload.** This is the G726-24 dynamic payload type. The default is **97**.

**G726r40 Dynamic Payload.** This is the G726-40 dynamic payload type. The default is **96**.

**G729b Dynamic Payload.** This is the G729b dynamic payload type. The default is **99**.

**NSE Codec Name.** This is the NSE codec name used in SDP. The default is **NSE**.

**AVT Codec Name.** This is the AVT codec name used in SDP. The default is **telephone-event**.

**G711u Codec Name.** This is the G711u codec name used in SDP. The default is **PCMU**.

**G711a Codec Name.** This is the G711a codec name used in SDP. The default is **PCMA**.

**G726r16 Codec Name.** This is the G726-16 codec name used in SDP. The default is **G726-16**.

**G726r24 Codec Name.** This is the G726-24 codec name used in SDP. The default is **G726-24**.

**G726r32 Codec Name.** This is the G726-32 codec name used in SDP. The is **G726-32**.

**G726r40 Codec Name.** This is the G726-40 codec name used in SDP. The default is **G726-40**.

**G729a Codec Name.** This is the G729a codec name used in SDP. The default is **G729a**.

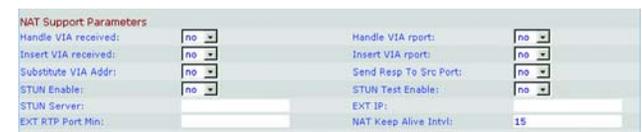
**G729b Codec Name.** This is the G729b codec name used in SDP. The default is **G729ab**.

**G723 Codec Name.** This is the G723 codec name used in SDP. The default is **G723**.

### NAT Support Parameters

**Handle VIA received.** If you select yes, the System will process the received parameter in the VIA header (this is inserted by the server in a response to any one of its requests). If you select no, the parameter will be ignored. Select **yes** or **no** from the drop-down menu. The default is **no**.

**Handle VIA rport.** If you select yes, the System will process the rport parameter in the VIA header (this is inserted by the server in a response to any one of its requests). If you select no, the parameter will be ignored. Select **yes** or **no** from the drop-down menu. The default is **no**.



NAT Support Parameters	
Handle VIA received:	<input type="text" value="no"/>
Insert VIA received:	<input type="text" value="no"/>
Substitute VIA Addr:	<input type="text" value="no"/>
STUN Enable:	<input type="text" value="no"/>
STUN Server:	<input type="text"/>
EXT RTP Port Min:	<input type="text"/>
Handle VIA rport:	<input type="text" value="no"/>
Insert VIA rport:	<input type="text" value="no"/>
Send Resp To Src Port:	<input type="text" value="no"/>
STUN Test Enable:	<input type="text" value="no"/>
EXT IP:	<input type="text"/>
NAT Keep Alive Intvl:	<input type="text" value="15"/>

**Figure 6-19: Voice - SIP Screen - NAT Support Parameters**

## IP Telephony System

**Insert VIA received.** This lets you insert the received parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Select **yes** or **no** from the drop-down menu. The default is **no**.

**Insert VIA rport.** This feature lets you insert the rport parameter into the VIA header of SIP responses if the received-from port and VIA sent-by port numbers differ. Select **yes** or **no** from the drop-down menu. The default is **no**.

**Substitute VIA Addr.** This feature lets you use NAT-mapped IP:port values in the VIA header. Select **yes** or **no** from the drop-down menu. The default is **no**.

**Send Resp To Src Port.** This feature lets you send responses to the request source port instead of the VIA sent-by port. Select **yes** or **no** from the drop-down menu. The default is **no**.

**STUN Enable.** This feature lets you use STUN to discover NAT mapping. Select **yes** or **no** from the drop-down menu. The default is **no**.

**STUN Test Enable.** If the STUN Enable feature is enabled and a valid STUN server is available, then the System can perform a NAT type discovery operation when it powers on. It will contact the configured STUN server, and the result of the discovery will be reported in a Warning header in all subsequent REGISTER requests. If the System detects symmetric NAT or a symmetric firewall, NAT mapping will be disabled.

The STUN Test Enable feature lets you use the STUN test. Select **yes** or **no** from the drop-down menu. The default is **no**.

**STUN Server.** Enter the IP address of the STUN server to contact for NAT mapping discovery.

**EXT IP.** Enter the external IP address to substitute for the actual IP address of the System in all outgoing SIP messages. If 0.0.0.0 is specified, then no IP address substitution will be performed.

**EXT RTP Port Min.** This is the external port mapping number of the RTP Port Min. number. If this value is not zero, then the RTP port number in all outgoing SIP messages will be substituted for the corresponding port value in the external RTP port range.

**NAT Keep Alive Intvl.** This is the interval between NAT-mapping, keep alive messages. The default is **15**.

## PBX Parameters

**Proxy Network Interface.** This tells the System how the clients (usually phones) are connected. Select **LAN** or **WAN**. The default is **WAN**.

**Proxy Listen Port.** This is the port used by the System when it listens for client messages at the selected interface. The default is **6060**.

PBX Parameters			
Proxy Network Interface:	LAN	Proxy Listen Port:	6060
Multicast Address:	224.168.168.168:6061	Group Page Address:	224.168.168.168:3456
Max. Expires:	3600	Force Media Proxy:	no
Proxy Debug Option:	full		
Call Routing Rule:	<-1,2,3,4>9xx.		
Internal Music URL:	http://192.168.0.4/aria711u.dat		
Internal Music Script:	2(58/133152),(133152/518000)		
Internal MOH Refresh Intvl:	0	Call Park MOH Server:	
Call Park DLG Refresh Intvl:	0	Default Group Line:	1,2,3,4
Group 1 User ID:		Group 1 Line:	
Group 2 User ID:		Group 2 Line:	
Group 3 User ID:		Group 3 Line:	
Group 4 User ID:		Group 4 Line:	
Hunt Groups:			
SIP DIDN Field:	TO UserID	SIP DIDN Param Name:	didn

Figure 6-20: Voice - SIP Screen - PBX Parameters

**Multicast Address.** This is the IP address (and port number) used by the System to send control messages to all clients at the same time. It must be a multicast address and must contain a port number. The default is **224.168.168.168:6061**.

**Group Page Address.** This is the IP address (and port number) used by the System to tell clients to send and receive group page RTP packets. It must be a multicast address and must contain a port number. The default is **244.168.168.168:34567**.

**Max Expires.** This sets the maximum allowed Registration Expires value (in seconds) for clients. The default is **3600**.

**Force Media Proxy.** This feature forces external clients to use the System's media proxy when exchanging RTP traffic with external peers. Select **yes** or **no** from the drop-down menu. The default is **no**.

**Proxy Debug Option.** SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Select **none** for no logging. Select **1-line** to log the start-line only for all messages. Select **1-line excl. OPT** to log the start-line only for all messages except OPTIONS requests/responses. Select **1-line excl. NTFY** to log the start-line only for all messages except NOTIFY requests/responses. Select **1-line excl. REG** to log the start-line only for all messages except REGISTER requests/responses. Select **1-line excl. OPTINTFYIREG** to log the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. Select **full** to log all SIP messages in full text. Select **full excl. OPT** to log all SIP messages in full text except OPTIONS requests/responses. Select **full excl. NTFY** to log all SIP messages in full text except NOTIFY requests/responses. Select **full excl. REG** to log all SIP messages in full text except REGISTER requests/responses. Select **full excl. OPTINTFYIREG** to log all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. The default is **full**.

**Call Routing Rule.** This is a special dial plan that determines which line can be used for an external, outbound call request from a phone based solely on the target public number. When you create this rule, follow this format:

(rule|rule|rule|...|rule)

The most specific rules should be placed first.

Each rule should be in this format: <:Lx>*pattern*

L indicates Line (phone line).

The variable x is **1**, **2**, **3**, or **4** depending on which line you want to specify.

The word *pattern* indicates any digit pattern (see the Dial Plan setting for more information).

## IP Telephony System

The default is (<L1,2,3,4>9xx.); this indicates that any of the four lines can be used for any target number starting with 9. For example, with this dial plan, the caller dials 9 before entering the external phone number.

**Internal Music URL.** Enter the Uniform Resource Locator (URL), also known as a web address, to download a music file for the music-on-hold and call park features. This is its format: [tftp://]server\_IP\_address[:port]/path. TFTP is the only protocol supported for music download. The default port is **69**. Saving a new URL will reboot the System. After its reboot, the System will download the file and store the samples in flash memory.

The music samples are encoded in G711u format at 8000 samples/second. This file should not contain any extra header information, and its maximum length is 65.536 seconds (524,288 bytes). For more information, refer to “Appendix D: New Music for the Music-on-Hold Feature.”

**Internal Music Script.** This script tells the System how to play the downloaded music file. This is its format:

```
[section[(section[,...]]]
```

Each section should be in this format: [n (start/end[/pause])] [pause2]

The variable n is the number of times you want a section to repeat before moving to the next section.

The start/end is the starting and 1+ending sample for the section. Note that the samples are numbered from 0 to total-length - 1. You may enter -1 or a very large number if the end of the file should be the ending sample. The default start value is **0**, and the default end value is the end of the file.

The variable pause is the number of samples to pause after the ending sample has been played. The default is **0**.

The variable pause2 is the additional number of samples to pause after the entire n repetitions of the section have been played. The default is **0**.

A maximum of 16 sections can be specified. Samples should be encoded in G711u format at 8000 samples/second. After all sections are played, they are replayed, starting with the first section.

For example, the default Internal Music Script setting is **2(0/230954),2(230954/444720),(0/230954)40000**. The first section is 2(0/230954); samples 0 through 230954 will be played twice. The second section is 2(230954/444720); samples 230954 through 444720 will be played twice. The third section is (0/230954); samples 0 through 230954 will be played once.

The last section is 40000. The ending pause will last for 40,000 samples. Each sample lasts 1/8000 of a second, so 40,000 samples equals 5 seconds. When this pause is over, the sections are replayed.

**Internal MOH Refresh Intvl.** The System can refresh an internal music session periodically. The default is **0**, which disables the refresh function.

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**Call Park MOH Server.** Enter the name or IP address of the music-on-hold server that should be used to handle a parked call. If you do not have a music-on-hold server for the call park feature, then keep the default, **imusic**, and the parked caller will hear the internal music file. Otherwise, if this setting is not specified, the parked caller will hear silence.

**Call Park DLG Refresh Intvl.** The System can refresh a call park session periodically. The default is **0**, which disables the refresh function.

**Default Group Line.** This is the default group of lines, **1,2,3,4**.

**Group 1-4 User ID.** A group designates specific phones that should be paged as a group, use the same phone lines, and receive the same type of calls. For example, sales calls should go to the sales group. You can designate up to four groups. For each group, enter a comma-separated list of User IDs, each representing a different client. For example, if the sales group is Group 1, then enter the sales extensions: **501,502,503** in the *Group 1 User ID* field. A client can belong to more than one group. If a client does not belong to any group, then the client belongs to the default group assigned to the Default Group Line. Each User ID pattern can use \* and ? wildcards as well as %xx escaped characters (refer to “Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users” for more details). The default is a blank field, which means that all clients belong to the default group.

**Group 1-4 Line.** For each group, enter a comma-separated list of phone lines the clients can use (this list determines the order in which the lines will be used). The System will make external calls for clients using the phone lines listed here. For example, for a group whose setting is 1,3, then System will use Line 1. If that is not successful, then it will use Line 3.

**Hunt Groups.** This defines one or more hunt groups that can be called directly by any client like a regular extension. The syntax is the same as the syntax for the Contact List. Note that a member of one group can also be the extension of another group (i.e., one level of recursion is allowed).

**SIP DIDN Field.** This determines which field is used to indicate the Direct Inward Dialing (DID) number for an incoming INVITE to a line interface. Select **TO UserID** to use the User-ID field of the TO header, or select **TO Param** to use a parameter in the TO header with the name specified in the SIP DIDN PARAM Name. The default is **TO UserID**.

**SIP DIDN Param Name.** This indicates the DID number in an incoming INVITE message. The default is **didn**.

### Auto Attendant Parameters

**AA Dial Plan 1.** This is used to define the first dial rule in the auto-attendant. The default is **(10xlxxx.)**. Refer to “Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users” for more details.

**AA Dial Plan 2.** This is used to define the second dial rule in the auto-attendant. The default is **(<:10>xlxxx.)**.

Auto Attendant Parameters			
AA Dial Plan 1:	<:0:501>[1]<:2:200>[<:3:300> 5xx 8001]		
AA Dial Plan 2:	<:0:501>[<:2:200>[<:3:300> 5xx 8001]		
AA script 1:	<aa><form id="dt" type="menu"><audio src="prompt5" bargein="T"/><a		
AA script 2:	<aa><form id="nt" type="menu"><audio src="prompt6" bargein="F"/><a		
AA script 3:			
DayTime AA:	<input type="checkbox"/>	Day Time:	
DayTime AA Script:	<input type="checkbox"/>	DayTime Answer Delay:	12
NightTime AA:	<input type="checkbox"/>	NightTime AA Script:	<input type="checkbox"/>
NightTime Answer Delay:	0	Weekend/Holiday AA:	<input type="checkbox"/>
Weekends/Holidays:		Weekend/Holiday AA Script:	<input type="checkbox"/>
Weekend/Holiday Answer Delay:	0		

Figure 6-21: Voice - SIP Screen - Auto Attendant Parameters

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**AA script 1-3.** These are used to define the three auto-attendant scripts. Refer to “Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users” for more details.

**DayTime AA.** To enable the daytime auto-attendant, select **yes**. Otherwise, select **no**. The default is **yes**.

**DayTime.** Enter the daytime hours for the daytime auto-attendant in 24-hour format. Enter the start and end times in this format:

start=hh:mm:ss;end=hh:mm:ss

(hh for hours, mm for minutes, and ss for seconds)

For example, start=9:0:0;end=17:0:0 means the start time is 9 AM and the end time is 5 PM. The other hours (5 PM to 9 AM) are considered nighttime hours.

If you do not enter start and end times, then the whole day (24 hours) is considered as daytime, so the nighttime auto-attendant will not be used, even if it is enabled.

**DayTime AA Script.** Select the daytime auto-attendant script that you want to use, **1**, **2**, or **3**. The default is **1**.

**DayTime Answer Delay.** Select the number of seconds you want the daytime auto-attendant to wait before answering. The default is **12** seconds.

**NightTime AA.** To enable the nighttime auto-attendant, select **yes**. Otherwise, select **no**. The default is **no**.

**NightTime AA Script.** Select the nighttime auto-attendant script that you want to use, **1**, **2**, or **3**. The default is **1**.

**NightTime Answer Delay.** Select the number of seconds you want the nighttime auto-attendant to wait before answering. The default is **0** seconds.

**Weekend/Holiday AA.** To enable this auto-attendant, select **yes**. Otherwise, select **no**. The default is **no**.

**Weekends/Holidays.** When the weekend/holiday auto-attendant is enabled, you can use this setting to specify the weekends and holidays. Up to four weekend days can be defined. Use this format:

[wk=n1[,ni];][hd=mm/dd/yyyy|mm/dd/yyyy-mm/dd/yyyy[,mm/dd/yyyy|mm/dd/yyyy-mm/dd/yyyy];]

(wk for weekend, which can be 1 for Monday to 7 for Sunday)

(hd for holiday, which does not have to include the year)

For example, wk=6,7;hd=1/1,2/21/2006,5/30/2006,12/19/2006-12/30/2006 means that Saturdays and Sundays are the weekends. Holidays are January 1-2, 2006; May 30, 2006; and December 19-30, 2006.

## IP Telephony System

**Weekend/Holiday AA Script.** Select the auto-attendant script that you want to use for weekends and holidays, **1, 2, or 3**. The default is **1**.

**Weekend/Holiday Answer Delay.** Select the number of seconds you want the weekend/holiday auto-attendant to wait before answering. The default is **0** seconds.

### PBX Phone Parameters

**Next Auto User ID.** This is the User ID assigned to the next new client that requests to register with the System.

**Phone Ext Password.** This is a REGISTRATION password that applies to Ext 1 of all clients. If there is no password, then all clients will be allowed to register without being challenged by the System. The default is blank (no password).

**Phone Upgrade Rule.** This is the upgrade rule for all clients. The default is blank (no rule).

**Phone Dial Plan.** Enter the dial plan for all clients. The default is **(9,[3469]11S019,<:1408>[2-9]XXXXXXI9,<:1>[2-9]xxxxxxxxxS019,1[2-9]xxxxxxxxxS019,011xx.I9,xx.I[1-8]xxx)**. This dial plan tells the phone to do the following:

- play the outside dial tone if the first digit is 9
- dial 9311, 9411, 9611, and 9911 immediately
- dial 9 + [2-9] + 6 digits after a short timeout and insert the 1 + 408 area code
- dial 9 + [2-9] + 9 digits immediately and insert the 1 (domestic long distance)
- dial 91 + [2-9] + 9 digits immediately (domestic long distance)
- dial 9011 + 1 or more digits after timeout or the # (pound) key (international)
- dial 9 + 1 or more digits after timeout or the # (pound) key (catch-all)
- dial [1-8] + 3 more digits immediately (internal calls)

Refer to “Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users” for more details.

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.

PBX Phone Parameters  
Next Auto User ID: 504 Phone Ext Password:  
Phone Upgrade Rule:  
Phone Dial Plan: (9,[3469]11S019,<:1408>[2-9]XXXXXXI9,<:1>[2-9]xxxxxxxxxS019,1[2-9]xxxxxxxxxS019,011xx.I9,xx.I[1-8]xxx)  
Undo All Changes Submit All Changes  
PBX Status  
User Login Basic Advanced

**Figure 6-22: Voice - SIP Screen - PBX Phone Parameters**

## The Voice - Provisioning Screen

Use this screen to configure service provisioning settings.



**IMPORTANT:** In most cases, you should not change these settings unless instructed to do by your ITSP.

### Configuration Profile

**Provision Enable.** The configuration profile must be requested by the System and cannot be pushed from a provisioning server, although a service provider can effectively push a profile by remotely triggering the request operation via SIP NOTIFY. To enable the provisioning feature, select **yes**. Otherwise, select **no**. The default is **yes**.

**Resync On Reset.** This feature lets you force the System to resync with the provisioning server when it powers on or reboots. Select **yes** or **no** from the drop-down menu. The default is **yes**.

**Resync Random Delay.** The System uses this feature to uniformly scatter resync requests from multiple devices over a period of time. Enter the period of time in seconds. The default is **2**.

**Resync Periodic.** The System uses this feature to resync on a periodic basis. Enter the interval in seconds. The default is **3600**.

**Resync Error Retry Delay.** If a resync attempt fails, the System will retry after a period of time. Enter the period of time in seconds. The default is **3600**.

**Forced Resync Delay.** This feature tells the System how long to wait before undergoing a forced resync. Enter the period of time in seconds. The default is **14400**.

**Resync From SIP.** This feature permits a service provider to trigger a profile resync via a SIP NOTIFY message. To enable this feature, select **yes**. Otherwise, select **no**. The default is **yes**.

**Resync After Upgrade Attempt.** If you want the System to resync after an upgrade attempt, select **yes**. Otherwise, select **no**. The default is **yes**.

**Resync Trigger 1/2.** Enter the first and second triggers you want to use.

**Resync Fails On FNF.** If you want the resync to fail when the FNF (File Not Found) error occurs, select **yes**. Otherwise, select **no**. The default is **yes**.

**Profile Rule.** This script identifies the provisioning server to contact when the System is performing a profile resync. Enter the appropriate script. The default is **/spa\$PSN.cfg**.

Router		Voice	
Info	System	SIP	Provisioning
Regional	FXS 1	FXS 2	Line 1
Line 2	Line 3	Line 4	FXS Status
User Login: basic 1 advanced			
<b>Configuration Profile</b>			
Provision Enable:	<input type="text" value="yes"/>	Resync On Reset:	<input type="text" value="yes"/>
Resync Random Delay:	<input type="text" value="2"/>	Resync Periodic:	<input type="text" value="3600"/>
Resync Error Retry Delay:	<input type="text" value="3600"/>	Forced Resync Delay:	<input type="text" value="14400"/>
Resync From SIP:	<input type="text" value="yes"/>	Resync After Upgrade Attempt:	<input type="text" value="yes"/>
Resync Trigger 1:	<input type="text"/>		
Resync Trigger 2:	<input type="text"/>		
Resync Fails On FNF:	<input type="text" value="yes"/>		
Profile Rule:	<input type="text" value="/spa\$PSN.cfg"/>		
Profile Rule B:	<input type="text"/>		
Profile Rule C:	<input type="text"/>		
Profile Rule D:	<input type="text"/>		
Log Resync Request Msg:	<input type="text" value="\$PN \$MAC -- Requesting resync \$SCHEME://\$SERVIP:\$PORT\$PATH"/>		
Log Resync Success Msg:	<input type="text" value="\$PN \$MAC -- Successful resync \$SCHEME://\$SERVIP:\$PORT\$PATH"/>		
Log Resync Failure Msg:	<input type="text" value="\$PN \$MAC -- Resync failed: \$ERR"/>		
Report Rule:	<input type="text"/>		

**Figure 6-23: Voice - Provisioning Screen - Configuration Profile**

**Profile Rule B, C, and D.** Enter profile rules B, C, and D.

**Log Resync Request Msg.** This script defines the message sent to the configured syslog server whenever the System attempts to resync with the provisioning server. Enter the appropriate script. The default is **\$PN \$MAC -- Requesting resync \$SCHEME://\$SERVIP:\$PORT\$PATH.**

**Log Resync Success Msg.** This script defines the message sent to the configured syslog server whenever the System successfully completes a resync with the provisioning server. Enter the appropriate script. The default is **\$PN \$MAC -- Successful resync \$SCHEME://\$SERVIP:\$PORT\$PATH.**

**Log Resync Failure Msg.** This script defines the message sent to the configured syslog server whenever the System fails to complete a resync with the provisioning server. Enter the appropriate script. The default is **\$PN \$MAC -- Resync failed: \$ERR.**

**Report Rule.** Enter the report rule.

## Firmware Upgrade

**Upgrade Enable.** The firmware profile must be requested by the System and cannot be pushed from an upgrade server, although a service provider can effectively push a new firmware load by remotely triggering the request operation via the configuration file. To enable the upgrade feature, select **yes**. Otherwise, select **no**. The default is **yes**.

**Upgrade Error Retry Delay.** If an upgrade attempt fails, the System will retry after a period of time. Enter the period of time in seconds. The default is **3600**.

**Downgrade Rev Limit.** Enter the downgrade firmware revision limit.

**Upgrade Rule.** Enter the upgrade rule.

**Log Upgrade Request Msg.** This script defines the message sent to the configured syslog server whenever the System attempts an upgrade from the upgrade server. Enter the appropriate script. The default is **\$PN \$MAC -- Requesting upgrade \$SCHEME://\$SERVIP:\$PORT\$PATH.**

**Log Upgrade Success Msg.** This script defines the message sent to the configured syslog server whenever the System successfully completes an upgrade from the upgrade server. Enter the appropriate script. The default is **\$PN \$MAC -- successful upgrade \$SCHEME://\$SERVIP:\$PORT\$PATH -- \$ERR.**

**Log Upgrade Failure Msg.** This script defines the message sent to the configured syslog server whenever the System fails to complete an upgrade from the upgrade server. Enter the appropriate script. The default is **\$PN \$MAC -- Upgrade failed \$ERR.**



**Figure 6-24: Voice - Provisioning Screen - Firmware Upgrade**

*firmware: the programming code that runs a networking device.*

*upgrade: to replace existing software or firmware with a newer version.*

## IP Telephony System

**License Keys.** There are additional license keys you can acquire to upgrade the System. You can upgrade from support of 4 phones to support of 16 phones, and/or you can upgrade from a two-line appearance per phone to a four-line appearance per phone. Enter the license keys in this field. For more information about licensing, contact your ITSP.

### General Purpose Parameters

**GPP A-P.** These can be used by both the provisioning and upgrade logic to hold any string value. Then any of the values can be incorporated in other scripted parameters. Enter the appropriate string value in each field.

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.

General Purpose Parameters

GPP A:	
GPP B:	
GPP C:	
GPP D:	
GPP E:	
GPP F:	
GPP G:	
GPP H:	
GPP I:	
GPP J:	
GPP K:	
GPP L:	
GPP M:	
GPP N:	
GPP O:	
GPP P:	

Undo All Changes Submit All Changes

[PBL Status](#)  
[User Login](#) [Basic](#) | [Advanced](#)

**Figure 6-25: Voice - Provisioning Screen - General Purpose Parameters**

## The Voice - Regional Screen

Use this screen to configure call settings.



**IMPORTANT:** In most cases, you should not change these settings unless instructed to do by your ITSP.

### Call Progress Tones

**Dial Tone.** This is played to prompt the user to enter a phone number. The default is **350@-19,440@-19;10(\*0/1+2)**.

**Second Dial Tone.** This is an alternative to the Dial Tone when the user dials a three-way call. The default is **420@-19,520@-19;10(\*0/1+2)**.

**Outside Dial Tone.** This is an alternative to the Dial Tone. It prompts the user to enter an external phone number, as opposed to an internal extension. It is triggered by a , character (a comma) encountered in the dial plan. The default is **420@-19;10(\*0/1)**.

**Prompt Tone.** This is played to prompt the user to enter a call forwarding phone number. The default is **520@-19,620@-19;10(\*0/1+2)**.

**Busy Tone.** This is played when a 486 RSC is received for an outbound call. The default is **480@-19,620@-19;10(.5/5/1+2)**.

**Reorder Tone.** This is played when an outbound call has failed or after the far end hangs up during an established call. The default is **480@-19,620@-19;10(.25/.25/1+2)**.

**Off Hook Warning Tone.** This is played when the caller has not properly placed the handset on the cradle. The default is **480@10,620@0;10(.125/.125/1+2)**.

**Ring Back Tone.** This is played during an outbound call when the far end is ringing. The default is **440@-19,480@-19;\*(2/4/1+2)**.

**Confirm Tone.** This brief tone notifies the user that the last input value has been accepted. The default is **600@-16; 1(.25/.25/1)**.

**SIT1 Tone.** This is an alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the *SIP* screen. The default is **985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)**.

Router	Voice
Info	System SIP Provisioning <b>Regional</b> FXS 1 FXS 2 Line 1 Line 2 Line 3 Line 4 <a href="#">FXS Status</a>
	<a href="#">User Login</a> <a href="#">Home</a>
	<a href="#">1 advanced</a>
<b>Call Progress Tones</b>	
Dial Tone:	350@-19,440@-19;10(*0/1+2)
Second Dial Tone:	420@-19,520@-19;10(*0/1+2)
Outside Dial Tone:	420@-16;10(*0/1)
Prompt Tone:	520@-19,620@-19;10(*0/1+2)
Busy Tone:	480@-19,620@-19;10(.5/5/1+2)
Reorder Tone:	480@-19,620@-19;10(.25/.25/1+2)
Off Hook Warning Tone:	480@-10,620@0;10(.125/.125/1+2)
Ring Back Tone:	440@-19,480@-19;*(2/4/1+2)
Confirm Tone:	600@-16;1(.25/.25/1)
SIT1 Tone:	985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT2 Tone:	914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0)
SIT3 Tone:	914@-16,1371@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT4 Tone:	985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0)
MWI Dial Tone:	350@-19,440@-19;2(.1/1/1+2);10(*0/1+2)
Cfwd Dial Tone:	350@-19,440@-19;2(.2/2/1+2);10(*0/1+2)
Holding Tone:	600@-19;*(.1/1/1,.1/1/1,.1/9.5/1)
Conference Tone:	350@-19;20(.1/1/1,.1/9.7/1)
Secure Call Indication Tone:	397@-19,507@-19;15(0/2/0,.2/1/1,.1/2.1/2)
Feature Invocation Tone:	350@-16;*(.1/1/1)

**Figure 6-26: Voice - Regional Screen - Call Progress Tones**

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**SIT2 Tone.** This is an alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the *SIP* screen. The default is **914@-16,1371@-16,1777@-16;20(.274/0/1,274/0/2,380/0/3,0/4/0)**.

**SIT3 Tone.** This is an alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the *SIP* screen. The default is **914@-16,1371@-16,1777@-16;20(.380/0/1,380/0/2,380/0/3,0/4/0)**.

**SIT4 Tone.** This is an alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the *SIP* screen. The default is **985@-16,1371@-16,1777@-16;20(.380/0/1,274/0/2,380/0/3,0/4/0)**.

**MWI Dial Tone.** This tone is played instead of the Dial Tone when there are unheard messages in the caller's mailbox. The default is **350@-19,440@-19;2(.1/.1/1+2);10(\*0/1+2)**.

**Cfwd Dial Tone.** This tone is played when all calls are forwarded. The default is **350@-19,440@-19;2(.2/.2/1+2);10(\*0/1+2)**.

**Holding Tone.** This lets the local caller know that the far end has placed the call on hold. The default is **600@-19\*(.1/.1/1,1/.1/1,1/9.5/1)**.

**Conference Tone.** This is played to all parties when a three-way conference call is in progress. The default is **350@-19;20(.1/.1/1,1/9.7/1)**.

**Secure Call Indication Tone.** This is played when a call has been successfully switched to secure mode. It should be played only for a short while, less than 30 seconds, and at a reduced level, less than -19 dBm, so it will not interfere with the conversation. The default is **397@-19,507@-19;15(0/2/0,2/.1/1,1/2.1/2)**.

**Feature Invocation Tone.** This is played when a feature is implemented. The default is **350@-16;\*(.1/.1/1)**.

### Distinctive Ring Patterns

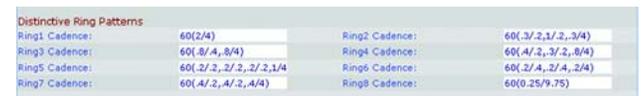
**Ring1 Cadence.** This is the cadence script for distinctive ring 1. The default is **60(2/4)**.

**Ring2 Cadence.** This is the cadence script for distinctive ring 2. The default is **60(.3/.2,1/.2,3/4)**.

**Ring3 Cadence.** This is the cadence script for distinctive ring 3. The default is **60(.8/.4,8/4)**.

**Ring4 Cadence.** This is the cadence script for distinctive ring 4. The default is **60(.4/.2,3/.2,8/4)**.

**Ring5 Cadence.** This is the cadence script for distinctive ring 5. The default is **60(.2/.2,2/.2,2/.2,1/4)**.



Distinctive Ring Patterns			
Ring1 Cadence:	60(2/4)	Ring2 Cadence:	60(.3/.2,1/.2,3/4)
Ring3 Cadence:	60(.8/.4,8/4)	Ring4 Cadence:	60(.4/.2,3/.2,8/4)
Ring5 Cadence:	60(.2/.2,2/.2,2/.2,1/4)	Ring6 Cadence:	60(2/4,2/4,2/4)
Ring7 Cadence:	60(.4/.2,4/.2,4/4)	Ring8 Cadence:	60(0.25/9.75)

**Figure 6-27: Voice - Regional Screen - Distinctive Ring Patterns**

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**Ring6 Cadence.** This is the cadence script for distinctive ring 6. The default is **60(.2/.4,.2/.4,.2/4).**

**Ring7 Cadence.** This is the cadence script for distinctive ring 7. The default is **60(.4/.2,.4/.2,.4/4).**

**Ring8 Cadence.** This is the cadence script for distinctive ring 8. The default is **60(0.25/9.75).**

### Distinctive Call Waiting Tone Patterns

**CWT1 Cadence.** This is the cadence script for distinctive CWT (Call Waiting Tone) 1. The default is **30(.3/9.7).**

**CWT2 Cadence.** This is the cadence script for distinctive CWT2. The default is **30(.1/.1, .1/9.7).**

**CWT3 Cadence.** This is the cadence script for distinctive CWT3. The default is **30(.1/.1, .3/.1, .1/9.3).**

**CWT4 Cadence.** This is the cadence script for distinctive CWT4. The default is **30(.1/.1,.1/.1,.1/9.5).**

**CWT5 Cadence.** This is the cadence script for distinctive CWT5. The default is **30(.3/.1,.1/.1,.3/9.1).**

**CWT6 Cadence.** This is the cadence script for distinctive CWT6. The default is **30(.1/.1,.3/.2,.3/9.1).**

**CWT7 Cadence.** This is the cadence script for distinctive CWT7. The default is **30(.3/.1,.3/.1,.1/9.1).**

**CWT8 Cadence.** This is the cadence script for distinctive CWT8. The default is **2.3(.3/2).**

### Distinctive Ring/CWT Pattern Names

**Ring1 Name.** In an INVITE's Alert-Info header, this is the name that picks distinctive ring/CWT 1 for the inbound call. The default is **Bellcore-r1.**

**Ring2 Name.** In an INVITE's Alert-Info header, this is the name that picks distinctive ring/CWT 2 for the inbound call. The default is **Bellcore-r2.**

**Ring3 Name.** In an INVITE's Alert-Info header, this is the name that picks distinctive ring/CWT 3 for the inbound call. The default is **Bellcore-r3.**

**Ring4 Name.** In an INVITE's Alert-Info header, this is the name that picks distinctive ring/CWT 4 for the inbound call. The default is **Bellcore-r4.**

**Ring5 Name.** In an INVITE's Alert-Info header, this is the name that picks distinctive ring/CWT 5 for the inbound call. The default is **Bellcore-r5.**

Distinctive Call Waiting Tone Patterns	
CWT1 Cadence:	30(.3/9.7)
CWT2 Cadence:	30(.1/.1, .1/9.7)
CWT3 Cadence:	30(.1/.1, .3/.1, .1/9.3)
CWT4 Cadence:	30(.1/.1,.1/.1,.1/9.5)
CWT5 Cadence:	30(.3/.1,.1/.1,.3/9.1)
CWT6 Cadence:	30(.1/.1,.3/.2,.3/9.1)
CWT7 Cadence:	30(.3/.1,.3/.1,.1/9.1)
CWT8 Cadence:	2.3(.3/2)

**Figure 6-28: Voice - Regional Screen - Distinctive Call Waiting Tone Patterns**

Distinctive Ring/CWT Pattern Names	
Ring1 Name:	Bellcore-r1
Ring2 Name:	Bellcore-r2
Ring3 Name:	Bellcore-r3
Ring4 Name:	Bellcore-r4
Ring5 Name:	Bellcore-r5
Ring6 Name:	Bellcore-r6
Ring7 Name:	Bellcore-r7
Ring8 Name:	Bellcore-r8

**Figure 6-29: Voice - Regional Screen - Distinctive Ring/CWT Pattern Names**

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**Ring6 Name.** In an INVITE's Alert-Info header, this is the name that picks distinctive ring/CWT 6 for the inbound call. The default is **Bellcore-r6**.

**Ring7 Name.** In an INVITE's Alert-Info header, this is the name that picks distinctive ring/CWT 7 for the inbound call. The default is **Bellcore-r7**.

**Ring8 Name.** In an INVITE's Alert-Info header, this is the name that picks distinctive ring/CWT 8 for the inbound call. The default is **Bellcore-r8**.

### Ring and Call Waiting Tone Spec

**Ring Waveform.** Select the waveform of the ringing signal, **Sinusoid** or **Trapezoid**. The default is **Sinusoid**.

**Ring Frequency.** Enter the frequency of the ringing signal, which can range from 10 to 100 Hz. The default is **25**.

**Ring Voltage.** Enter the ringing voltage value, which can range from 60 to 90 volts. The default is **70**.

**CWT Frequency.** Enter the frequency script of the call waiting tone. All distinctive CWTs are based on this tone. The default is **440@-10**.

### Control Timer Values (sec)

**Hook Flash Timer Min.** This is the minimum on-hook time before off-hook to qualify as a hook-flash event. If the on-hook time is less than the minimum, then it is ignored. The range is 0.1 to 0.4 seconds. The default is **.1**.

**Hook Flash Timer Max.** This is the maximum on-hook time before off-hook to qualify as a hook-flash event. If the on-hook time is more than the maximum, then it is ignored. The range is 0.4 to 1.6 seconds. The default is **.9**.

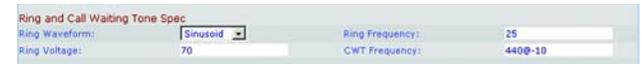
**Callee On Hook Delay.** The phone must be on-hook for this length of time before the IP Telephony ends the current inbound call. (This does not apply to outbound calls.) The range is 0 to 255 seconds. The default is **0**.

**Reorder Delay.** This is the delay after the far end hangs up before the Reorder Tone is played. To play the tone immediately, enter **0**. To have the tone never play, enter **inf**. The range is 0 to 255 seconds. The default is **5**.

**Call Back Expires.** This is the expiration time for a call back activation. The range is 0 to 65,535 seconds. The default is **1800**.

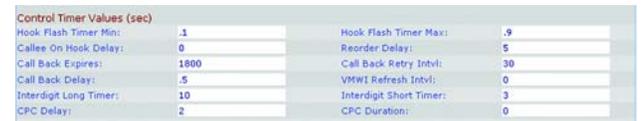
**Call Back Retry Intvl.** This is the interval for a call back retry. The range is 0 to 255 seconds. The default is **30**.

**Call Back Delay.** This is the delay between the System receiving the first SIP 18x response and the System declaring that the far end is ringing. If a busy response is received during this time, then the System considers the call a failure and continues to retry. The default is **.5**.



Ring and Call Waiting Tone Spec	
Ring Waveform:	Sinusoid
Ring Frequency:	25
Ring Voltage:	70
CWT Frequency:	440@-10

**Figure 6-30: Voice - Regional Screen - Ring and Call Waiting Tone Spec**



Control Timer Values (sec)	
Hook Flash Timer Min:	.1
Hook Flash Timer Max:	.9
Callee On Hook Delay:	0
Reorder Delay:	5
Call Back Expires:	1800
Call Back Retry Intvl:	30
Call Back Delay:	.5
VMWI Refresh Intvl:	0
Interdigit Long Timer:	10
Interdigit Short Timer:	3
CPC Delay:	2
CPC Duration:	0

**Figure 6-31: Voice - Regional Screen - Control Timer Values**

**VMWI Refresh Intvl.** This is the interval between the VMWI refresh events to the CPE. The default is **0**.

**Interdigit Long Timer.** This is the long timeout between entering digits when a caller is dialing. The range is 0 to 64 seconds. The default is **10**.

**Interdigit Short Timer.** This is the short timeout between entering digits when a caller is dialing. The range is 0 to 64 seconds. The default is **3**.

**CPC Delay.** CPC stands for Calling Party Control. The CPC Delay is the delay after the caller hangs up when the System will begin removing the tip-and-ring voltage from the attached equipment of the called party. The range is 0 to 255 seconds, and the resolution is 1 second. The default is **2**.

**CPC Duration.** This is the length of time the tip-to-ring voltage is removed after the caller hangs up. After that, the tip-to-ring voltage is restored, and the dial tone will apply if the attached equipment is still off-hook. The CPC is disabled if this value is set to 0. The range is 0 to 1.000 seconds, and the resolution is 0.001 seconds. The default is **0**.

### Vertical Service Activation Codes

**Call Return Code.** This code calls the last caller. The default is **\*69**.

**Call Redial Code.** This code redials the last number called. The default is **\*07**.

**Blind Transfer Code.** This code begins a blind transfer of the current call to the extension specified after the activation code. The default is **\*98**.

**Call Back Act Code.** This code starts a callback when the last outbound call is not busy. The default is **\*66**.

**Call Back Deact Code.** This code cancels a callback. The default is **\*86**.

**Call Back Busy Act Code.** This code starts a callback when the last outbound call is busy. The default is **\*05**.

**Cfwd All Act Code.** This code forwards all calls to the extension specified after the activation code. The default is **\*72**.

**Cfwd All Deact Code.** This code cancels call forwarding of all calls. The default is **\*73**.

**Cfwd Busy Act Code.** This code forwards busy calls to the extension specified after the activation code. The default is **\*90**.

**Cfwd Busy Deact Code.** This code cancels call forwarding of busy calls. The default is **\*91**.

Vertical Service Activation Codes			
Call Return Code:	*69	Call Redial Code:	*07
Blind Transfer Code:	*98	Call Back Act Code:	*66
Call Back Deact Code:	*86	Call Back Busy Act Code:	*05
Cfwd All Act Code:	*72	Cfwd All Deact Code:	*73
Cfwd Busy Act Code:	*90	Cfwd Busy Deact Code:	*91
Cfwd No Ans Act Code:	*92	Cfwd No Ans Deact Code:	*93
Cfwd Last Act Code:	*63	Cfwd Last Deact Code:	*63
Block Last Act Code:	*60	Block Last Deact Code:	*80
Accept Last Act Code:	*64	Accept Last Deact Code:	*84
CW Act Code:	*56	CW Deact Code:	*57
CW Per Call Act Code:	*71	CW Per Call Deact Code:	*70
Block CID Act Code:	*67	Block CID Deact Code:	*68
Block CID Per Call Act Code:	*81	Block CID Per Call Deact Code:	*82
Block ANC Act Code:	*77	Block ANC Deact Code:	*87
DND Act Code:	*78	DND Deact Code:	*79
CID Act Code:	*65	CID Deact Code:	*85
CW/CID Act Code:	*25	CW/CID Deact Code:	*45
Dist Ring Act Code:	*26	Dist Ring Deact Code:	*46
Speed Dial Act Code:	*74	Secure All Call Act Code:	*16
Secure No Call Act Code:	*17	Secure One Call Act Code:	*18
Secure One Call Deact Code:	*19	Conference Act Code:	
Attn-Xfer Act Code:		Modern Line Toggle Code:	*99
FAX Line Toggle Code:	*99		
Referral Services Codes:			
Feature Dial Services Codes:			

**Figure 6-32: Voice - Regional Screen - Vertical Service Activation Codes**

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**Cfwd No Ans Act Code.** This code forwards no-answer calls to the extension specified after the activation code. The default is **\*92**.

**Cfwd No Ans Deact Code.** This code cancels call forwarding of no-answer calls. The default is **\*93**.

**Cfwd Last Act Code.** This code forwards the last inbound or outbound calls to the extension specified after the activation code. The default is **\*63**.

**Cfwd Last Deact Code.** This code cancels call forwarding of the last inbound or outbound calls. The default is **\*83**.

**Block Last Act Code.** This code blocks the last inbound call. The default is **\*60**.

**Block Last Deact Code.** This code cancels blocking of the last inbound call. The default is **\*80**.

**Accept Last Act Code.** This code accepts the last outbound call. It lets the call ring through when the do not disturb or call forwarding of all calls features are enabled. The default is **\*64**.

**Accept Last Deact Code.** This code cancels the code to accept the last outbound call. The default is **\*84**.

**CW Act Code.** This code enables call waiting on all calls. The default is **\*56**.

**CW Deact Code.** This code disables call waiting on all calls. The default is **\*57**.

**CW Per Call Act Code.** This code enables call waiting for the next call. The default is **\*71**.

**CW Per Call Deact Code.** This code disables call waiting for the next call. The default is **\*70**.

**Block CID Act Code.** This code blocks caller ID on all outbound calls. The default is **\*67**.

**Block CID Deact Code.** This code removes caller ID blocking on all outbound calls. The default is **\*68**.

**Block CID Per Call Act Code.** This code blocks caller ID on the next outbound call. The default is **\*81**.

**Block CID Per Call Deact Code.** This code removes caller ID blocking on the next inbound call. The default is **\*82**.

**Block ANC Act Code.** This code blocks all anonymous calls. The default is **\*77**.

**Block ANC Deact Code.** This code removes blocking of all anonymous calls. The default is **\*87**.

**DND Act Code.** This code enables the do not disturb feature. The default is **\*78**.

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**DND Deact Code.** This code disables the do not disturb feature. The default is \*79.

**CIC Act Code.** This code enables caller ID generation. The default is \*65.

**CIC Deact Code.** This code disables caller ID generation. The default is \*85.

**CWCID Act Code.** This code enables call waiting, caller ID generation. The default is \*25.

**CWCID Deact Code.** This code disables call waiting, caller ID generation. The default is \*45.

**Dist Ring Act Code.** This code enables the distinctive ringing feature. The default is \*26.

**Dist Ring Deact Code.** This code disables the distinctive ringing feature. The default is \*46.

**Speed Dial Act Code.** This code assigns a speed dial number. The default is \*74.

**Secure All Call Act Code.** This code makes all outbound calls secure. The default is \*16.

**Secure No Call Act Code.** This code makes all outbound calls not secure. The default is \*17.

**Secure One Call Act Code.** This code makes the next outbound call secure. (It is redundant if all outbound calls are secure by default.) The default is \*18.

**Secure One Call Deact Code.** This code makes the next outbound call not secure. (It is redundant if all outbound calls are not secure by default.) The default is \*19.

**Conference Act Code.** If this code is specified, then the user must enter it before dialing the third party for a conference call. Enter the code for a conference call.

**Attn-Xfer Act Code.** If the code is specified, then the user must enter it before dialing the third party for a call transfer. Enter the code for a call transfer.

**Modem Line Toggle Code.** This code toggles the line to a modem. The default is \*99.

**FAX Line Toggle Code.** This code toggles the line to a fax machine. The default is #99.

**Referral Services Codes.** These codes tell the System what to do when the user places the current call on hold and is listening to the second dial tone. One or more \* codes can be entered here. For example, the blind transfer code is \*98. After the user dials \*98, the System will wait for the user to enter a phone number. After the number has been entered, the System will complete the blind transfer for the call on hold.

**Feature Dial Services Codes.** These codes tell the System what to do when the user is listening to the first or second dial tone. One or more \* codes can be entered here. For example, the code for call forwarding of all calls

## IP Telephony System

is \*72. After the user dials \*72, the System will wait for the user to enter a phone number. After the number has been entered, the System will forward all calls for that phone number.

### Vertical Service Announcement Codes

**Service Annc Base Number.** Enter the base number for service announcements.

**Service Annc Extension Codes.** Enter the extension codes for service announcements.

### Outbound Call Codec Selection Codes

**Prefer G711u Code.** This is the dialing code that will make this codec the preferred codec for the associated call. The default is **\*017110**.

**Force G711u Code.** This is the dialing code that will make this codec the only codec that can be used for the associated call. The default is **\*027110**.

**Prefer G711a Code.** This is the dialing code that will make this codec the preferred codec for the associated call. The default is **\*017111**.

**Force G711a Code.** This is the dialing code that will make this codec the only codec that can be used for the associated call. The default is **\*027111**.

**Prefer G723 Code.** This is the dialing code that will make this codec the preferred codec for the associated call. The default is **\*01723**.

**Force G723 Code.** This is the dialing code that will make this codec the only codec that can be used for the associated call. The default is **\*02723**.

**Prefer G726r16 Code.** This is the dialing code that will make this codec the preferred codec for the associated call. The default is **\*0172616**.

**Force G726r16 Code.** This is the dialing code that will make this codec the only codec that can be used for the associated call. The default is **\*0272616**.

**Prefer G726r24 Code.** This is the dialing code that will make this codec the preferred codec for the associated call. The default is **\*0172624**.

**Force G726r24 Code.** This is the dialing code that will make this codec the only codec that can be used for the associated call. The default is **\*0272624**.

Vertical Service Announcement Codes  
Service Annc Base Number:   
Service Annc Extension Codes:

**Figure 6-33: Voice - Regional Screen - Vertical Service Announcement Codes**

Outbound Call Codec Selection Codes			
Prefer G711u Code:	*017110	Force G711u Code:	*027110
Prefer G711a Code:	*017111	Force G711a Code:	*027111
Prefer G723 Code:	*01723	Force G723 Code:	*02723
Prefer G726r16 Code:	*0172616	Force G726r16 Code:	*0272616
Prefer G726r24 Code:	*0172624	Force G726r24 Code:	*0272624
Prefer G726r32 Code:	*0172632	Force G726r32 Code:	*0272632
Prefer G726r40 Code:	*0172640	Force G726r40 Code:	*0272640
Prefer G729a Code:	*01729	Force G729a Code:	*02729

**Figure 6-34: Voice - Regional Screen - Outbound Call Codec Selection Codes**

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**Prefer G726r32 Code.** This is the dialing code that will make this codec the preferred codec for the associated call. The default is **\*0172632**.

**Force G726r32 Code.** This is the dialing code that will make this codec the only codec that can be used for the associated call. The default is **\*0272632**.

**Prefer G726r40 Code.** This is the dialing code that will make this codec the preferred codec for the associated call. The default is **\*0172640**.

**Force G726r40 Code.** This is the dialing code that will make this codec the only codec that can be used for the associated call. The default is **\*0272640**.

**Prefer G729a Code.** This is the dialing code that will make this codec the preferred codec for the associated call. The default is **\*01729**.

**Force G729a Code.** This is the dialing code that will make this codec the only codec that can be used for the associated call. The default is **\*02729**.

## Miscellaneous

**Set Local Date (mm/dd).** Set the local date (mm stands for months and dd stands for days). The year is optional and uses two or four digits.

**Set Local Time (hh/mm).** Set the local time (hh stands for hours and mm stands for minutes). Seconds are optional.

**Time Zone.** For caller ID generation, select the number of hours to add to GMT to generate the local time. The default is **GMT-08:00**.

**FXS Port Impedance.** This sets the electrical impedance of the FXS port. Select one of these choices: **600, 900, 600+2.16uF, 900+2.16uF, 270+750i150nF, 220+850i120nF, 220+820i115nF, or 370+620i310nF**. The default is **600**.

**Daylight Saving Time Rule.** Enter the rule for calculating daylight saving time; it should include the start, end, and save values. This rule is comprised of three fields. Each field is separated by ; (a semicolon) as shown below. Optional values inside [ ] (the brackets) are assumed to be 0 if they are not specified. Midnight is represented by 0:0:0 of the given date.

This is the format of the rule: Start = <start-time>; end=<end-time>; save = <save-time>

The <start-time> and <end-time> values specify the start and end dates and times of daylight saving time. Each value is in this format: <month> /<day> / <weekday>[/HH:[mm[:ss]]]

Miscellaneous

Set Local Date (mm/dd):  Set Local Time (HH/mm):

Time Zone: GMT-08:00 FXS Port Impedance: 600

Daylight Saving Time Rule: start=4/1/7;end=10/1/7;save=1

FXS Port Input Gain: -3 FXS Port Output Gain: -3

DTMF Playback Level: -16 DTMF Playback Length: .1

Detect ABCD: yes Playback ABCD: yes

Caller ID Method: Bellcore(N-Amer,China) Caller ID FSK Standard: bell 202

Feature Invocation Method: Default

Undo All Changes Submit All Changes

PBX Status

User Login Home Advanced

Figure 6-35: Voice - Regional Screen - Miscellaneous

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The <save-time> value is the number of hours, minutes, and/or seconds to add to the current time during daylight saving time. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition. The <save-time> value is in this format: `[/[+|-]HH:[mm[:ss]]]`

The <month> value equals any value in the range 1-12 (January-December).

The <day> value equals [+|-] any value in the range 1-31.

If <day> is 1, it means the <weekday> on or before the end of the month (in other words the last occurrence of <weekday> in that month).

The <weekday> value equals any value in the range 1-7 (Monday-Sunday). It can also equal 0.

If the <weekday> value is 0, this means that the date to start or end daylight saving is exactly the date given. In that case, the <day> value must not be negative.

If the <weekday> value is not 0 and the <day> value is positive, then daylight saving starts or ends on the <weekday> value on or after the date given.

If the <weekday> value is not 0 and the <day> value is negative, then daylight saving starts or ends on the <weekday> value on or before the date given.

The abbreviation HH stands for hours (0-23).

The abbreviation mm stands for minutes (0-59).

The abbreviation ss stands for seconds (0-59).

The default Daylight Saving Time Rule is **start=4/1/7;end=10/-1/7;save=1**.

**FXS Port Input Gain.** Enter the input gain in dB, up to three decimal places. The range is 6.0 to -infinity. The default is **-3**.

**FXS Port Output Gain.** Enter the output gain in dB, up to three decimal places. The range is 6.0 to -infinity. The default is **-3**.

**DTMF Playback Level.** Enter the local DTMF playback level in dBm, up to one decimal place. The default is **-16**.

**DTMF Playback Length.** Enter the local DTMF playback duration in milliseconds. The default is **.1**.

**Detect ABCD.** To enable local detection of DTMF ABCD, select **yes**. Otherwise, select **no**. The default is **yes**.

## IP Telephony System

**Playback ABCD.** To enable local playback of OOB DTMF ABCD, select **yes**. Otherwise, select **no**. The default is **yes**.

**Caller ID Method.** You have a choice of methods to use for caller ID. Select **Bellcore(N.Amer, China)** for CID, CIDCW, and VMWI. FSK is sent after the first ring, and there is no polarity reversal or DTAS. Select **DTMF(Finland,Sweden)** for CID only. DTMF is sent after polarity reversal (with no DTAS) and before the first ring. Select **DTMF(Denmark)** for CID only. DTMF is sent after polarity reversal (with no DTAS) and before the first ring. Select **ETSI DTMF** for CID only. DTMF is sent after DTAS (with no polarity reversal) and before the first ring. Select **ETSI DTMF With PR** for CID only. DTMF is sent after polarity reversal and DTAS and before the first ring. Select **ETSI DTMF After Ring** for CID only. DTMF is sent after the first ring (with no polarity reversal or DTAS). Select **ETSI FSK** for CID, CIDCW, and VMWI. FSK is sent after DTAS (with no polarity reversal) and before the first ring. It will wait for ACK from CPE after DTAS for CIDCW. Select **ETSI FSK With PR(UK)** for CID, CIDCW, and VMWI. FSK is sent after polarity reversal and DTAS and before the first ring. It will wait for ACK from CPE after DTAS for CIDCW. Polarity reversal is applied only if the equipment is on-hook. The default is **Bellcore(N.Amer, China)**.

**Caller ID FSK Standard.** The System supports bell 202 and v.23 standards for caller ID generation. Select the FSK standard you want to use, **bell 202** or **v.23**. The default is **bell 202**.

**Feature Invocation Method.** Select the method you want to use, **Default** or **Sweden default**. The default is **Default**.

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.

## The Voice - FXS 1/2 Screen

Use the appropriate screen to configure settings for each FXS port, which is called the Phone port on the System.



**IMPORTANT:** In most cases, you should not change the service settings unless instructed to do by your ITSP.

**Line Enable.** To enable this line for service, select **yes**. Otherwise, select **no**. The default is **yes**.

### Network Settings

**SIP ToS/DiffServ Value.** Enter the TOS/DiffServ field value in UDP IP packets carrying a SIP message. The default is **0x68**.

**SIP CoS Value.** Enter the CoS value for SIP messages. The default is **3**.

**RTP ToS/DiffServ Value.** Enter the ToS/DiffServ field value in UDP IP packets carrying RTP data. The default is **0xb8**.

**RTP CoS Value.** Enter the CoS value for RTP data. The default is **6**.

**Network Jitter Level.** This setting determines how jitter buffer size is adjusted by the System. Jitter buffer size is adjusted dynamically. The minimum jitter buffer size is 30 milliseconds or (10 milliseconds + current RTP frame size), whichever is larger, for all jitter level settings. However, the starting jitter buffer size value is larger for higher jitter levels. This setting controls the rate at which the jitter buffer size is adjusted to reach the minimum. Select the appropriate setting: **low**, **medium**, **high**, **very high**, or **extremely high**. The default is **high**.

**Jitter Buffer Adjustment.** This controls how the jitter buffer should be adjusted. Select the appropriate setting: **up and down**, **up only**, **down only**, or **disable**. The default is **up and down**.

### SIP Settings

**SIP Port.** Enter the port number of the SIP message listening and transmission port. The default is **5080**.

**SIP Remote-Party-ID.** To use the Remote-Party-ID header instead of the From header, select **yes**. Otherwise, select **no**. The default is **yes**.

**SIP Debug Option.** SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Select **none** for no logging. Select **1-line** to log the start-line only for all messages. Select **1-line excl. OPT** to log the start-line only for all messages except OPTIONS requests/responses. Select **1-line**

Router		Voice	
Info	System	SIP	Provisioning
Regional	FXS 1	FXS 2	Line 1
Line 2	Line 3	Line 4	FXS Status
User Login   Home   Advanced			
Line Enable:	[ yes ]		
<b>Network Settings</b>			
SIP ToS/DiffServ Value:	0x68	SIP CoS Value:	3 [0-7]
RTP ToS/DiffServ Value:	0xb8	RTP CoS Value:	6 [0-7]
Network Jitter Level:	[ high ]	Jitter Buffer Adjustment:	[ up and down ]

Figure 6-36: Voice - FXS 1 Screen - Network Settings

SIP Settings			
SIP Port:	5080	SIP Remote-Party-ID:	[ yes ]
SIP Debug Option:	[ none ]	RTP Log Intvl:	0
Restrict Source IP:	[ no ]	Referer Bye Delay:	4
Refer Target Bye Delay:	0	Referee Bye Delay:	0
Refer-To Target Contact:	[ no ]	Sticky 183:	[ no ]

Figure 6-37: Voice - FXS 1 Screen - SIP Settings

## IP Telephony System

**excl. NTFY** to log the start-line only for all messages except NOTIFY requests/responses. Select **1-line excl. REG** to log the start-line only for all messages except REGISTER requests/responses. Select **1-line excl. OPTINTFYIREG** to log the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. Select **full** to log all SIP messages in full text. Select **full excl. OPT** to log all SIP messages in full text except OPTIONS requests/responses. Select **full excl. NTFY** to log all SIP messages in full text except NOTIFY requests/responses. Select **full excl. REG** to log all SIP messages in full text except REGISTER requests/responses. Select **full excl. OPTINTFYIREG** to log all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. The default is **none**.

**RTP Log Intvl.** Periodically, the System will log RTP statistics via syslog, depending on debug level. Enter the period of time in seconds. The default is **0**.

**Restrict Source IP.** If Lines 1 and 2 use the same SIP Port value and the Restrict Source IP feature is enabled, then the proxy IP address for Lines 1 and 2 is treated as an acceptable IP address for both lines. To enable the Restrict Source IP feature, select **yes**. Otherwise, select **no**. The default is **no**.

**Referor Bye Delay.** This controls when the System sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referor Bye Delay, enter the appropriate period of time in seconds. The default is **4**.

**Refer Target Bye Delay.** For the Refer Target Bye Delay, enter the appropriate period of time in seconds. The default is **0**.

**Referee Bye Delay.** For the Referee Bye Delay, enter the appropriate period of time in seconds. The default is **0**.

**Refer-To Target Contact.** To contact the refer-to target, select **yes**. Otherwise, select **no**. The default is **no**.

**Sticky 183.** If this feature is enabled, then the IP Telephony will ignore further 180 SIP responses after receiving the first 183 SIP response for an outbound INVITE. To enable this feature, select **yes**. Otherwise, select **no**. The default is **no**.

### Subscriber Information

**Display Name.** Enter the display name for caller ID.

**User ID.** Enter the extension number for this line.



Subscriber Information	
Display Name:	<input type="text"/>
User ID:	<input type="text" value="8001"/>

**Figure 6-38: Voice - FXS 1 Screen - Subscriber Information**

## Dial Plan

**Dial Plan.** Enter the dial plan script for this line. Refer to “Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users” for more details.

## Streaming Audio Server (SAS)

**SAS Enable.** To enable the use of the line as a streaming audio source, select **yes**. Otherwise, select **no**. If enabled, the line cannot be used for outgoing calls. Instead, it auto-answers incoming calls and streams audio RTP packets to the caller. The default is **no**.

**SAS DLG Refresh Intvl.** If this is not zero, it is the interval at which the streaming audio server sends out session refresh (SIP re-INVITE) messages to determine if the connection to the caller is still active. If the caller does not respond to the refresh message, then the System will end this call with a SIP BYE message. The range is 0 to 255 seconds (0 means that the session refresh is disabled).The default is **30**.

**SAS Inbound RTP Sink.** This setting works around devices that do not play inbound RTP if the streaming audio server line declares itself as a send-only device and tells the client not to stream out audio. Enter a Fully Qualified Domain Name (FQDN) or IP address of an RTP sink; this will be used by the System’s streaming audio server line in the SDP of its 200 response to an inbound INVITE message from a client.

## Call Feature Settings

**Blind Attn-Xfer Enable.** This settings lets the System perform an attended transfer operation by ending the current call leg and performing a blind transfer of the other call leg. If this feature is disabled, the System performs an attended transfer operation by referring the other call leg to the current call leg while maintaining both call legs. To use this feature, select **yes**. Otherwise, select **no**. The default is **no**.

**MOH Server.** Enter the user ID or URL of the auto-answering streaming audio server. When only a user ID is specified, the current or outbound proxy will be contacted. Music-on-hold is disabled if the MOH Server is not specified.

**Xfer When Hangup Conf.** This setting makes the System perform a transfer when a conference call has ended. Select **yes** or **no** from the drop-down menu. The default is **yes**.

**Conference Bridge URL.** This feature supports external conference bridging for n-way conference calls ( $n > 2$ ), instead of mixing audio locally. To use this feature, set this parameter to that of the server's name, e.g., *conf@myserver.com:12345* or *conf* (which uses the Proxy value as the domain).

**Conference Bridge Ports.** Select the maximum number of conference call participants. The range is 3 to 10. The default is **3**.

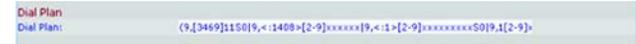


Figure 6-39: Voice - FXS 1 Screen - Dial Plan



Figure 6-40: Voice - FXS 1 Screen - Streaming Audio Server

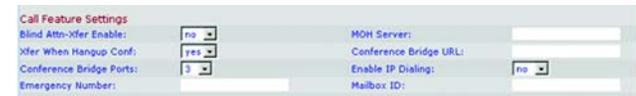


Figure 6-41: Voice - FXS 1 Screen - Call Feature Settings

## IP Telephony System

**Enable IP Dialing.** To use IP dialing, select **yes**. Otherwise, select **no**. The default is **no**.

**Emergency Number.** This is a comma-separated list of emergency number patterns. If the outbound call matches one of the patterns, then the System will disable hook flash event handling. Hook flash event handling will be restored to normal when the phone is on-hook again. If you leave this field blank, then the System will have no emergency number.

**Mailbox ID.** Enter the ID number of the mailbox for this line.

### Audio Configuration

**Preferred Codec.** Select a preferred codec for all calls. (The actual codec used in a call still depends on the outcome of the codec negotiation protocol.) Select one of the following: **G711u**, **G711a**, **G726-16**, **G726-24**, **G726-32**, **G726-40**, **G729a**, or **G723**. The default is **G711u**.

**Silence Supp Enable.** To enable silence suppression so that silent audio frames are not transmitted, select **yes**. Otherwise, select **no**. The default is **no**.

**Use Pref Codec Only.** To only use the preferred codec for all calls, select **yes**. (The call will fail if the far end does not support this codec.) Otherwise, select **no**. The default is **no**.

**Silence Threshold.** Select the appropriate setting for the threshold: **high**, **medium**, or **low**. The default is **medium**.

**G729a Enable.** To enable the use of the G729a codec at 8kbps, select **yes**. Otherwise, select **no**. The default is **yes**.

**Echo Canc Enable.** To enable the use of the echo canceller, select **yes**. Otherwise, select **no**. The default is **yes**.

**G723 Enable.** To enable the use of the G723a codec at 6.3kbps, select **yes**. Otherwise, select **no**. The default is **yes**.

**Echo Canc Adapt Enable.** To enable the echo canceller to adapt, select **yes**. Otherwise, select **no**. The default is **yes**.

**G726-16 Enable.** To enable the use of the G726 codec at 16kbps, select **yes**. Otherwise, select **no**. The default is **yes**.

**Echo Supp Enable.** To enable the use of the echo suppressor, select **yes**. Otherwise, select **no**. The default is **yes**.

Audio Configuration	
Preferred Codec:	G711u
Use Pref Codec Only:	no
G729a Enable:	yes
G723 Enable:	yes
G726-16 Enable:	yes
G726-24 Enable:	yes
G726-32 Enable:	yes
G726-40 Enable:	yes
DTMF Process INFO:	yes
DTMF Process AVT:	yes
DTMF Tx Method:	Auto
Hook Flash Tx Method:	None
Release Unused Codec:	yes
Silence Supp Enable:	no
Silence Threshold:	medium
Echo Canc Enable:	yes
Echo Canc Adapt Enable:	yes
Echo Supp Enable:	yes
FAX CED Detect Enable:	yes
FAX CNG Detect Enable:	yes
FAX Passthru Codec:	G711u
FAX Codec Symmetric:	yes
FAX Passthru Method:	NSE
FAX Process NSE:	yes
FAX Disable ECAI:	no
FAX Enable T38:	yes

Figure 6-42: Voice - FXS 1 Screen - Audio Configuration

## IP Telephony System

**G726-24 Enable.** To enable the use of the G726 codec at 24kbps, select **yes**. Otherwise, select **no**. The default is **yes**.

**FAX CED Detect Enable.** To enable detection of the fax Caller-Entered Digits (CED) tone, select **yes**. Otherwise, select **no**. The default is **yes**.

**G726-32 Enable.** To enable the use of the G726 codec at 32kbps, select **yes**. Otherwise, select **no**. The default is **yes**.

**FAX CNG Detect Enable.** To enable detection of the fax Calling Tone (CNG), select **yes**. Otherwise, select **no**. The default is **yes**.

**G726-40 Enable.** To enable the use of the G726 codec at 40kbps, select **yes**. Otherwise, select **no**. The default is **yes**.

**FAX Passthru Codec.** Select the codec for fax passthrough, **G711u** or **G711a**. The default is **G711u**.

**DTMF Process INFO.** To use the DTMF process info feature, select **yes**. Otherwise, select **no**. The default is **yes**.

**FAX Codec Symmetric.** To force the System to use a symmetric codec during fax passthrough, select **yes**. Otherwise, select **no**. The default is **yes**.

**DTMF Process AVT.** To use the DTMF process AVT feature, select **yes**. Otherwise, select **no**. The default is **yes**.

**FAX Passthru Method.** Select the fax passthrough method: **None**, **NSE**, or **ReINVITE**. The default is **NSE**.

**DTMF Tx Method.** Select the method to transmit DTMF signals to the far end: **InBand**, **AVT**, **INFO**, **Auto**, **InBand+INFO**, or **AVT+INFO**. InBand sends DTMF using the audio path. AVT sends DTMF as AVT events. INFO uses the SIP INFO method. Auto uses InBand or AVT based on the outcome of codec negotiation. The default is **Auto**.

**FAX Process NSE.** To use the fax process NSE feature, select **yes**. Otherwise, select **no**. The default is **yes**.

**Hook Flash Tx Method.** Select the method for signaling hook flash events: **None**, **AVT**, or **INFO**. None does not signal hook flash events. AVT uses RFC2833 AVT (event = 16). INFO uses SIP INFO with the single line signal=hf in the message body. The MIME type for this message body is taken from the Hook Flash MIME Type setting. The default is **None**.

**FAX Disable ECAN.** If enabled, this feature will automatically disable the echo canceller when a fax tone is detected. To use this feature, select **yes**. Otherwise, select **no**. The default is **no**.

## IP Telephony System

**Release Unused Codec.** This feature allows the release of codecs not used after codec negotiation on the first call, so that other codecs can be used for the second line. To use this feature, select **yes**. Otherwise, select **no**. The default is **yes**.

**FAX Enable T38.** To enable the use of the ITU-T T.38 standard for faxing, select **yes**. Otherwise, select **no**. The default is **yes**.

### FXS Port Polarity Configuration

**Idle Polarity.** Select the polarity before a call is connected, **Forward** or **Reverse**. The default is **Forward**.

**Caller Conn Polarity.** Select the polarity after an outbound call is connected, **Forward** or **Reverse**. The default is **Forward**.

**Callee Conn Polarity.** Select the polarity after an inbound call is connected, **Forward** or **Reverse**. The default is **Forward**.

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.



**Figure 6-43: Voice - FXS 1 Screen - FXS Port Polarity Configuration**

## The Voice - Line 1/2/3/4 Screen

Use the appropriate screen to configure settings for each external Internet phone line.



**IMPORTANT:** In most cases, you should not change the service settings unless instructed to do by your ITSP.

**Line Enable.** To enable this line for service, select **yes**. Otherwise, select **no**. The default is **yes**.

### Network Settings

**SIP ToS/DiffServ Value.** Enter the TOS/DiffServ field value in UDP IP packets carrying SIP messages. The default is **0x68**.

**SIP CoS Value.** Enter the CoS value for SIP messages. The default is **3**.

### SIP Settings

**SIP Port.** Enter the port number of the SIP message listening and transmission port. The default is **5060**.

**SIP 100REL Enable.** To enable the support of 100REL SIP extension for reliable transmission of provisional responses (18x) and use of PRACK requests, select **yes**. Otherwise, select **no**. The default is **no**.

**Auth Resync-Reboot.** If this feature is enabled, the System will authenticate the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select **yes**. Otherwise, select **no**. The default is **yes**.

**SIP Proxy-Require.** The SIP proxy can support a specific extension or behavior when it sees this header from the user agent. If this field is configured and the proxy does not support it, then it will respond with the message, “unsupported.” Enter the appropriate header in the field provided.

**SIP Remote-Party-ID.** To use the Remote-Party-ID header instead of the From header, select **yes**. Otherwise, select **no**. The default is **yes**.

**SIP Debug Option.** SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Select **none** for no logging. Select **1-line** to log the start-line only for all messages. Select **1-line excl. OPT** to log the start-line only for all messages except OPTIONS requests/responses. Select **1-line excl. NTFY** to log the start-line only for all messages except NOTIFY requests/responses. Select **1-line excl. REG** to log the start-line only for all messages except REGISTER requests/responses. Select **1-line excl. OPTINTFYIREG** to log the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER

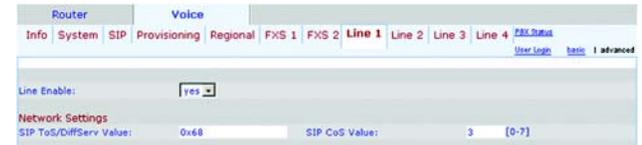


Figure 6-44: Voice - Line 1 Screen - Network Settings

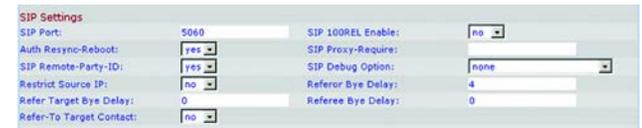


Figure 6-45: Voice - Line 1 Screen - SIP Settings

## IP Telephony System

requests/responses. Select **full** to log all SIP messages in full text. Select **full excl. OPT** to log all SIP messages in full text except OPTIONS requests/responses. Select **full excl. NTFY** to log all SIP messages in full text except NOTIFY requests/responses. Select **full excl. REG** to log all SIP messages in full text except REGISTER requests/responses. Select **full excl. OPTINTFYIREG** to log all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. The default is **none**.

**Restrict Source IP.** If Lines 1 and 2 use the same SIP Port value and the Restrict Source IP feature is enabled, then the proxy IP address for Lines 1 and 2 is treated as an acceptable IP address for both lines. To enable the Restrict Source IP feature, select **yes**. Otherwise, select **no**. The default is **no**.

**Referor Bye Delay.** This controls when the System sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referor Bye Delay, enter the appropriate period of time in seconds. The default is **4**.

**Refer Target Bye Delay.** For the Refer Target Bye Delay, enter the appropriate period of time in seconds. The default is **0**.

**Referee Bye Delay.** For the Referee Bye Delay, enter the appropriate period of time in seconds. The default is **0**.

**Refer-To Target Contact.** To contact the refer-to target, select **yes**. Otherwise, select **no**. The default is **no**.

### Subscriber Information

**Display Name.** Enter the display name for caller ID.

**User ID.** Enter the extension number for this line.

**Password.** Enter the password for this line.

**Use Auth ID.** To use the authentication ID and password for SIP authentication, select **yes**. Otherwise, select **no** to use the user ID and password. The default is **no**.

**Auth ID.** Enter the authentication ID here.

**Call Capacity.** Select the maximum number of calls allowed on this line. (The System does not distinguish between incoming and outgoing calls when it determines its call capacity.)

**Contact List.** This is a list of clients that System should alert when there is an incoming call on this line. Each rule is also known as a hunt group. The default method to ring a group is to ring all the members simultaneously, unless a hunt rule is specified. The default contact list is **aa** (auto-attendant).



The screenshot shows a web-based form titled "Subscriber Information". It contains several input fields and dropdown menus:

Display Name:	14088501231	User ID:	14088501231
Password:		Use Auth ID:	no
Auth ID:		Call Capacity:	
Contact List:	aa		
Cfwd No Ans Delay:	20		

**Figure 6-46: Voice - Line 1 Screen - Subscriber Information**

## IP Telephony System

When you create this rule, follow this format:

```
rule[rule[rule[...]]]
```

The most specific rules should be placed first.

Each rule should be in this format: [*did:*]*extf*,*extf*,*extf*[...]][,name=*gname*][,hunt=*hrule*][,cfwd=*targetf*]

The term *did* indicates an embedded Direct Inward Dialing (DID) number. If it is not specified, then the rule applies to any DID number.

The term *ext* indicates the client extension number pattern. It accepts \* and ? wildcards as well as %xx escaped characters.

The term *name* is a name for the call group.

If a *hrule* is specified, then the listed clients are contacted sequentially (also known as hunting); otherwise, they will ring simultaneously. When you create this *hrule*, follow this format:

```
hunt=<algo>;<interval>;<max>
```

The term <*algo*> determines the order to ring the clients. It can be one of the following:

- **restart** or **re=** so it always starts from the beginning of the list
- **next** or **ne=** so it starts from the next on the list to the last client that rings
- **random** or **ra=** so the order is random for each call

The term <*interval*> is the time, in seconds, to ring each client.

The term <*max*> is the total time, in seconds, to hunt before the call is rejected or forwarded to voicemail. If <*max*> is less than <*interval*>, it is interpreted as the number of cycles to go through the hunt group before call hunting stops. If <*max*> is 0, call hunting will continue indefinitely until the caller hangs up or someone answers the call.

If necessary, the call is forwarded to a user ID, called the *target*, in the *hrule*. If the *target* is a voicemail mailbox, then the *target* starts with **vm**. For example, the *target* **vm3456** will have calls forwarded to voicemail with the mailbox ID 3456.

For example, the Contact List is **501,502,hunt=ne,4,1;cfwd=aa**. This means that 501 will ring first for four seconds. If 501 does not pick up or is already on another call, then 502 will ring for four seconds. This cycle will repeat once before the call hunting stops. Then the call goes to the auto-attendant.

## IP Telephony System

**Cfwd No Ans Delay.** Enter the delay, in seconds, before the call forwarding of no-answer calls feature is triggered. The default is **20**.

### Dial Plan

**Dial Plan.** Enter the dial plan script for this line. Refer to “Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users” for more details. The default is (<9:>xx).

### NAT Settings

**NAT Mapping Enable.** To use externally mapped IP addresses and SIP/RTP ports in SIP messages, select **yes**. Otherwise, select **no**. The default is **no**.

**NAT Keep Alive Enable.** To send the configured NAT keep alive message periodically, select **yes**. Otherwise, select **no**. The default is **no**.

**NAT Keep Alive Msg.** Enter the keep alive message that should be sent periodically to maintain the current NAT mapping. If the value is \$NOTIFY, then a NOTIFY message is sent. If the value is \$REGISTER, then a REGISTER message without contact is sent. The default is **\$NOTIFY**.

**NAT Keep Alive Dest.** Enter the destination that should receive NAT keep alive messages. If the value is \$PROXY, then the messages will be sent to the current or outbound proxy. The default is **\$PROXY**.

**EXT SIP Port.** Enter the port number of the external port to substitute for the actual SIP port of the System in all outgoing SIP messages.

### Proxy and Registration

**Proxy.** Enter the SIP proxy server for all outbound requests.

**Use Outbound Proxy.** To use the outbound proxy, select **yes**. Otherwise, select **no**. The default is **no**.

**Outbound Proxy.** Enter the SIP outbound proxy server, where all outbound requests are sent for the first hop.

**Use OB Proxy In Dialog.** To force SIP requests to be sent to the outbound proxy within a dialog, select **yes**. Otherwise, select **no**. The default is **yes**.

**Register.** To require periodic registration with the proxy server, select **yes**. Otherwise, select **no**. The default is **yes**.

**Make Call Without Reg.** To allow outbound calls to be made without successful registration by the System, select **yes**. Otherwise, select **no**. The default is **no**.

Dial Plan  
Dial Plan: (9,[3469]1150[9,<1:1408>[2-9]xxxxxx]9,<1>[2-9]xxxxxxxxx50]9,1[2-9])

Figure 6-47: Voice - Line 1 Screen - Dial Plan

NAT Settings  
NAT Mapping Enable: no  
NAT Keep Alive Enable: no  
NAT Keep Alive Msg: \$NOTIFY  
NAT Keep Alive Dest: \$PROXY  
EXT SIP Port:

Figure 6-48: Voice - Line 1 Screen - NAT Settings

Proxy and Registration  
Proxy:  
Outbound Proxy:  
Register: yes  
Register Expires: 3600  
Use DNS SRV: no  
Proxy Fallback Intvl: 3600  
Mailbox Subscribe URL:  
Mailbox Manage URL:  
Use Outbound Proxy: no  
Use OB Proxy In Dialog: yes  
Make Call Without Reg: no  
Ans Call Without Reg: no  
DNS SRV Auto Prefix: no  
Proxy Redundancy Method: Normal  
Mailbox Deposit URL:  
Mailbox Status:  
Undo All Changes Submit All Changes  
PST Status  
User Login Basic Advanced

Figure 6-49: Voice - Line 1 Screen - Proxy and Registration

## IP Telephony System

**Register Expires.** This is the expiration value, in seconds, of a REGISTER request. The System will periodically renew registration shortly before the current registration has expired. The default is **3600**.

**Ans Call Without Reg.** To allow inbound calls to be answered without successful registration by the System, select **yes**. Otherwise, select **no**. The default is **no**.

**Use DNS SRV.** To use the DNS SRV lookup for proxy and outbound proxy, select **yes**. Otherwise, select **no**. The default is **no**.

**DNS SRV Auto Prefix.** To have the System automatically prepend the proxy or outbound proxy name with `_sip._udp` when it is performing a DNS SRV lookup on that name, select **yes**. Otherwise, select **no**. The default is **no**.

**Proxy Fallback Intvl.** This sets the delay, in seconds, after which the System will retry from the highest priority proxy (or outbound proxy) servers after it has failed over to a lower priority server. It will work only if the primary and backup proxy server list is provided to the System via the DNS SRV record lookup on the server name. The default is **3600**.

**Proxy Redundancy Method.** The System will create an internal list of proxies returned in DNS SRV records. You have a choice of two modes. Select **Normal** mode if you want this list to contain proxies organized by weight and priority. Select **Based on SRV Port** mode if you want the System to use Normal mode first and then inspect the port number based on the first proxy's port on the list. The default is **Normal**.

**Mailbox Subscribe URL.** Enter the URL that should receive the SUBSCRIBE messages, so the System will receive voicemail status notification for all mailboxes on this line.

**Mailbox Deposit URL.** Enter the URL that the System will contact when clients and external callers need to deposit voicemail in any of the mailboxes on this line.

**Mailbox Manage URL.** Enter the URL that the IP Telephony will contact when it needs to check voicemail for any of the mailboxes on this line.

**Mailbox Status.** Displayed here is the status for all mailboxes on this line. The status is automatically updated when the System receives voicemail status notification from the ITSP. The information is shown in this format:

`[mailboxID:number of new messages/number of old messages[,mailboxID:number of new messages/number of old messages[,mailboxID:number of new messages/number of old messages[,...]]]]`

When you have finished making changes, click the **Submit All Changes** button to save the changes, or click the **Undo All Changes** button to undo your changes.

# Appendix A: Troubleshooting

This appendix provides solutions to problems that may occur during the installation and operation of the IP Telephony System. Read the description below to solve your problems. If you can't find an answer here, check the Linksys website at [www.linksys.com](http://www.linksys.com).

## Common Problems and Solutions

**1. The System did not automatically assign an extension number to the Linksys SPA-family Internet phone, and the phone's Ext LED is yellow instead of green.**

Follow these steps:

- A. Open the web browser on the administration computer.
- B. Enter **<http://192.168.0.1/admin/router/status>**.
- C. If the phone is on the WAN side, write down the Current IP of the System. (This is the Internet IP address.)

If the phone is on the LAN side, write down the LAN IP Address of the System. (This is the local IP address.)

- D. Access the phone's Web-based Utility.
- E. Make sure the configured proxy server on the phone matches the System's IP address. (Refer to the phone's documentation for details.)

**2. The Internet phone can make internal calls to other Internet phones and analog phones; however, it cannot make external calls.**

Check to see if the Internet phone's line is registered. Follow these steps:

- A. Open the web browser on the administration computer.
- B. Enter **<http://192.168.0.1/admin/voice/advanced>**.
- C. On the *Voice - Info* screen, check to see if the Line 1 Status indicates that the Registration State says, "Registered."
- D. If it is not registered, then verify that the User ID, Proxy, and Password supplied by your Internet Telephony Service Provider (ITSP) are valid (these settings are configured on the *Line 1* screen).

**3. I made a call from an outside line, and I did not hear a ring tone after I entered the extension number.**

First, try again and make sure you entered the extension number correctly. If you still do not hear a ring tone, then follow these steps:

- A. Open the web browser on the administration computer.
- B. Enter **<http://192.168.0.1/admin/voice/status>**.
- C. On the *PBX Status* screen, make sure the Internet phone for that extension number is registered.

**4. I made a call from an outside line, and the auto-attendant says, “Not a valid extension, please try again.” However, I can make outgoing calls from the Internet phone with that extension number.**

Follow these steps:

- A. Open the web browser on the administration computer.
- B. Enter **http://192.168.0.1/admin/voice/advanced**.
- C. Click the **SIP** tab.
- D. On the *Voice - SIP* screen, add the extension number to the Auto-Attendant dial plan.

**5. When an outside line calls the System, it rings one time and then goes to the auto-attendant.**

By default, if no one answers the call after four seconds, then the call will go to the auto-attendant. To change this setting, follow these steps:

- A. Open the web browser on the administration computer.
- B. Enter **http://192.168.0.1/admin/voice/advanced**.
- C. Click the **SIP** tab.
- D. On the *Voice - SIP* screen, change the appropriate Answer Delay setting (DayTime, NightTime, or Weekends/Holidays).

**6. How can I change greetings for the auto-attendant?**

Use the Interactive Voice Response Menu to record or change greetings; refer to “Chapter 5: Using the Interactive Voice Response Menu” for instructions.

**7. I want to use another computer on the network (not the administration computer) to access the Web-based Utility. I entered http://192.168.0.1, but this address did not work.**

Any computer connected to your router should use the Internet (WAN) IP address of the System. (The administration computer is directly connected to the System’s Ethernet port, so it can use http://192.168.0.1, which is the System’s local IP address.) Use the Interactive Voice Response Menu to find out the System’s Internet IP address. Follow these steps:

- A. Use a telephone connected to the Phone 1 port of the System.
- B. Press \*\*\*\* (in other words, press the star key four times).
- C. Wait until you hear “Linksys configuration menu. Please enter the option followed by the # (pound) key or hang up to exit.”
- D. Press **110#**.
- E. You will hear the IP address assigned to the System’s Internet (external) interface. Write it down.
- F. Press **7932#**.
- G. Press **1** to enable WAN access to the Web-based Utility.
- H. Open the web browser on a networked computer.
- I. Enter **http://(Internet IP address of the System)**.

**8. I'm trying to access the System's Web-based Utility, but I do not see the login screen. Instead, I see a screen saying, "404 Forbidden."**

If you are using Windows Explorer, perform the following steps until you see the Web-based Utility's login screen (Netscape Navigator will require similar steps):

- A. Click **File**. Make sure *Work Offline* is NOT checked.
- B. Press **CTRL + F5**. This is a hard refresh, which will force Windows Explorer to load new webpages, not cached ones.
- C. Click **Tools**. Click **Internet Options**. Click the **Security** tab. Click the **Default level** button. Make sure the security level is Medium or lower. Then click the **OK** button.

**9. I need to set a static IP address on a PC.**

The System, by default, assigns an IP address range of 192.168.0.100 to 192.168.0.150 using the DHCP server on the System. To set a static IP address, you can only use the ranges 192.168.0.2 to 192.168.0.99 and 192.168.0.151 to 192.168.0.254. Each PC or network device that uses TCP/IP must have a unique address to identify itself in a network. If the IP address is not unique to a network, Windows will generate an IP conflict error message. You can assign a static IP address to a PC by performing the following steps:

For Windows 98 and Millennium:

- A. Click **Start**, **Setting**, and **Control Panel**. Double-click **Network**.
- B. In *The following network components are installed* box, select the **TCP/IP**-> associated with your Ethernet adapter. If you only have one Ethernet adapter installed, you will only see one TCP/IP line with no association to an Ethernet adapter. Highlight it and click the **Properties** button.
- C. In the *TCP/IP properties* window, select the **IP address** tab, and select **Specify an IP address**. Enter a unique IP address that is not used by any other computer on the network connected to the System. You can only use an IP address in the ranges 192.168.0.2 to 192.168.0.99 and 192.168.0.151 to 192.168.0.254. Make sure that each IP address is unique for each PC or network device.
- D. Click the **Gateway** tab, and in the *New Gateway* prompt, enter **192.168.0.1**, which is the default IP address of the System. Click the **Add** button to accept the entry.
- E. Click the **DNS** tab, and make sure the **DNS Enabled** option is selected. Enter the Host and Domain names (e.g., John for Host and home for Domain). Enter the DNS entry provided by your ISP. If your ISP has not provided the DNS IP address, contact your ISP to get that information or go to its website for the information.
- F. Click the **OK** button in the *TCP/IP properties* window, and click **Close** or the **OK** button for the *Network* window.
- G. Restart the computer when asked.

### For Windows 2000:

- A. Click **Start**, **Settings**, and **Control Panel**. Double-click **Network and Dial-Up Connections**.
- B. Right-click the **Local Area Connection** that is associated with the Ethernet adapter you are using, and select the **Properties** option.
- C. In the *Components checked are used by this connection* box, highlight **Internet Protocol (TCP/IP)**, and click the **Properties** button. Select **Use the following IP address** option.
- D. Enter a unique IP address that is not used by any other computer on the network connected to the System. You can only use an IP address in the ranges 192.168.0.2 to 192.168.0.99 and 192.168.0.151 to 192.168.0.254.
- E. Enter the Subnet Mask, **255.255.255.0**.
- F. Enter the Default Gateway, **192.168.0.1** (System's default IP address).
- G. Toward the bottom of the window, select **Use the following DNS server addresses**, and enter the Preferred DNS server and Alternative DNS server (provided by your ISP). Contact your ISP or go on its website to find the information.
- H. Click the **OK** button in the *Internet Protocol (TCP/IP) Properties* window, and click the **OK** button in the *Local Area Connection Properties* window.
- I. Restart the computer if asked.

### For Windows XP:

The following instructions assume you are running Windows XP with the default interface. If you are using the Classic interface (where the icons and menus look like previous Windows versions), please follow the instructions for Windows 2000.

- A. Click **Start** and **Control Panel**.
- B. Click the **Network and Internet Connections** icon and then the **Network Connections** icon.
- C. Right-click the **Local Area Connection** that is associated with the Ethernet adapter you are using, and select the **Properties** option.
- D. In the *This connection uses the following items* box, highlight **Internet Protocol (TCP/IP)**. Click the **Properties** button.
- E. Enter a unique IP address that is not used by any other computer on the network connected to the System. You can only use an IP address in the ranges 192.168.0.2 to 192.168.0.99 and 192.168.0.151 to 192.168.0.254.
- F. Enter the Subnet Mask, **255.255.255.0**.
- G. Enter the Default Gateway, **192.168.0.1** (System's default IP address).
- H. Toward the bottom of the window, select **Use the following DNS server addresses**, and enter the Preferred DNS server and Alternative DNS server (provided by your ISP). Contact your ISP or go on its website to find the information.
- I. Click the **OK** button in the *Internet Protocol (TCP/IP) Properties* window. Click the **OK** button in the *Local Area Connection Properties* window.

**10. I want to test my Internet connection.**

- A. Check your TCP/IP settings.

**For Windows 98 and Millennium:**

Refer to Windows Help for details. Make sure **Obtain IP address automatically** is selected in the settings.

**For Windows 2000:**

1. Click **Start**, **Settings**, and **Control Panel**. Double-click **Network and Dial-Up Connections**.
2. Right-click the **Local Area Connection** that is associated with the Ethernet adapter you are using, and select the **Properties** option.
3. In the *Components checked are used by this connection* box, highlight **Internet Protocol (TCP/IP)**, and click the **Properties** button. Make sure that **Obtain an IP address automatically** and **Obtain DNS server address automatically** are selected.
4. Click the **OK** button in the *Internet Protocol (TCP/IP) Properties* window, and click the **OK** button in the *Local Area Connection Properties* window.
5. Restart the computer if asked.
6. Click the **OK** button in the *Internet Protocol (TCP/IP) Properties* window, and click the **OK** button in the *Local Area Connection Properties* window.
7. Restart the computer if asked.

**For Windows XP:**

The following instructions assume you are running Windows XP with the default interface. If you are using the Classic interface (where the icons and menus look like previous Windows versions), please follow the instructions for Windows 2000.

1. Click **Start** and **Control Panel**.
  2. Click the **Network and Internet Connections** icon and then the **Network Connections** icon.
  3. Right-click the **Local Area Connection** that is associated with the Ethernet adapter you are using, and select the **Properties** option.
  4. In the *This connection uses the following items* box, highlight **Internet Protocol (TCP/IP)**, and click the **Properties** button. Make sure that **Obtain an IP address automatically** and **Obtain DNS server address automatically** are selected.
- B. Open a command prompt.
- For Windows 98 and Millennium, click **Start** and **Run**. In the *Open* field, type **command**. Press the **Enter** key or click the **OK** button.
  - For Windows 2000 and XP, click **Start** and **Run**. In the *Open* field, type **cmd**. Press the **Enter** key or click the **OK** button.

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- C. In the command prompt, type **ping 192.168.0.1** and press the **Enter** key.
- If you get a reply, the computer is communicating with the System.
  - If you do NOT get a reply, check the cable, and make sure **Obtain an IP address automatically** is selected in the TCP/IP settings for your Ethernet adapter.
- D. In the command prompt, type **ping** followed by your Internet IP address and press the **Enter** key. The Internet IP Address can be found in the web interface of the IP Telephony System. For example, if your Internet IP address is 1.2.3.4, you would enter **ping 1.2.3.4** and press the **Enter** key.
- If you get a reply, the computer is connected to the System.
  - If you do NOT get a reply, try the ping command from a different computer to verify that your original computer is not the cause of the problem.
- E. In the command prompt, type **ping www.linksys.com** and press the **Enter** key.
- If you get a reply, the computer is connected to the Internet. If you cannot open a webpage, try the ping command from a different computer to verify that your original computer is not the cause of the problem.
  - If you do NOT get a reply, there may be a problem with the connection. Try the ping command from a different computer to verify that your original computer is not the cause of the problem.

### ***11. I am not getting an IP address on the Internet with my Internet connection.***

- A. Refer to “Problem #10, I want to test my Internet connection” to verify that you have connectivity.
- B. If you need to register the MAC address of your Ethernet adapter with your ISP, please see “Appendix E: Finding the MAC Address and IP Address for Your Ethernet Adapter.” If you need to clone the MAC address of your Ethernet adapter onto the System, see the Router - WAN Setup - MAC Clone Settings section of “Chapter 6: Using the Web-based Utility.”
- C. Make sure you are using the right Internet settings. Contact your ISP to see if your Internet connection type is DHCP, Static IP Address, or PPPoE (commonly used by DSL consumers). Refer to the Router - WAN Setup - Internet Connection Settings section of “Chapter 6: Using the Web-based Utility.”
- D. Make sure you use the right cable. Check to see if the LEDs for the Internet port are lit.
- E. Make sure the cable connecting from your cable or DSL modem is connected to the Internet port of the System or your network router. Verify that the Router - Status page of the System’s Web-based Utility shows a valid IP address from your ISP.
- F. Power off your network devices, including the System, cable/DSL modem, and router (if you have a separate router). Wait 30 seconds, and power on the cable/DSL modem first. Then power on the router (if used), System, and other network devices. Check the Router - Status tab of the System’s Web-based Utility to see if you get an IP address.

**12. I am not able to access the System's Web-based Utility Setup page.**

- A. Refer to "Problem #10, I want to test my Internet connection" to verify that your computer is properly connected to the System.
- B. Refer to "Appendix E: Finding the MAC Address and IP Address for Your Ethernet Adapter" to verify that your computer has an IP Address, Subnet Mask, Gateway, and DNS.
- C. Set a static IP address on your system; refer to "Problem #9: I need to set a static IP address on a PC."
- D. Refer to "Problem #17: I am a PPPoE user, and I need to remove the proxy settings or the dial-up pop-up window."

**13. I can't get my Virtual Private Network (VPN) to work through the System.**

VPNs that use IPSec with the ESP (Encapsulation Security Payload known as protocol 50) authentication will work fine. At least one IPSec session will work through the System; however, simultaneous IPSec sessions may be possible, depending on the specifics of your VPNs.

VPNs that use IPSec and AH (Authentication Header known as protocol 51) are incompatible with the System. AH has limitations due to occasional incompatibility with the NAT standard.

Change the IP address for the System to another subnet to avoid a conflict between the VPN IP address and your local IP address. For example, if your VPN server assigns an IP address 192.168.0.X (X is a number from 1 to 254) and your local LAN IP address is 192.168.0.X (X is the same number used in the VPN IP address), the System will have difficulties routing information to the right location. If you change the System's IP address to 192.168.2.1, that should solve the problem. Change the System's IP address through the Router - LAN Setup tab of the Web-based Utility. If you assigned a static IP address to any computer or network device on the network, you need to change its IP address accordingly to 192.168.2.Y (Y being any number from 1 to 254). Note that each IP address must be unique within the network.

Your VPN may require port 500/UDP packets to be passed to the computer that is connecting to the IPSec server. Refer to "Problem #15, I need to set up online game hosting or use other Internet applications" for details. Check the Linksys website at [www.linksys.com](http://www.linksys.com) for more information.

**14. I need to set up a server behind my System.**

To use a server like a web, ftp, or mail server, you need to know the respective port numbers they are using. For example, port 80 (HTTP) is used for web; port 21 (FTP) is used for FTP, and port 25 (SMTP outgoing) and port 110 (POP3 incoming) are used for the mail server. You can get more information by viewing the documentation provided with the server you installed. Follow these steps to set up port forwarding through the System's Web-based Utility. We will be setting up web, ftp, and mail servers.

- A. Access the System's Web-based Utility by going to **http://192.168.0.1** or the IP address of the System. Go to the **Router => Application** tab.
- B. Select **yes** from the *Enable* drop-down menu.

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- C. Enter any name you want to use for the service.
- D. Enter the port range of the service you are using. For example, if you have a web server, you would enter the range 80 (in the *Starting Port* field) to 80 (in the *Ending Port* field).
- E. Select the protocol you will be using, **TCP** or **UDP**, or select **Both**.
- F. Enter the IP address of the PC or network device that you want the port server to go to. For example, if the web server's Ethernet adapter IP address is 192.168.0.100, you would enter 100 in the field provided. Check "Appendix E: Finding the MAC Address and IP Address for Your Ethernet Adapter" for details on getting an IP address.
- G. Follow the instructions in steps B-F for the port services you want to use. Consider the examples below:

Enable	Service Name	Starting and Ending Ports	Protocol	IP Address
yes	Web server	80 to 80	Both	192.168.0.100
yes	FTP server	21 to 21	TCP	192.168.0.101
yes	SMTP (outgoing)	25 to 25	Both	192.168.0.102
yes	POP3 (incoming)	110 to 110	Both	192.168.0.102

When you have completed the configuration, click the **Submit All Changes** button.

### ***15. I need to set up online game hosting or use other Internet applications.***

If you want to play online games or use Internet applications, most will work without doing any port forwarding or DMZ hosting. There may be cases when you want to host an online game or Internet application. This would require you to set up the System to deliver incoming packets or data to a specific computer. This also applies to the Internet applications you are using. The best way to get the information on what port services to use is to go to the website of the online game or application you want to use. Follow these steps to set up online game hosting or use a certain Internet application:

- A. Access the System's Web-based Utility by going to **http://192.168.0.1** or the IP address of the System. Go to the **Router => Application** tab.
- B. Enter any name you want to use for the Application.
- C. Select **yes** from the *Enable* drop-down menu.
- D. Enter any name you want to use for the service.
- E. Enter the port range of the service you are using. For example, if you have a web server, you would enter the range 80 (in the *Starting Port* field) to 80 (in the *Ending Port* field).
- F. Select the protocol you will be using, **TCP** or **UDP**, or select **Both**.

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- G. Enter the IP address of the PC or network device that you want the port server to go to. For example, if the web server's Ethernet adapter IP address is 192.168.0.100, you would enter 100 in the field provided. Check "Appendix E: Finding the MAC Address and IP Address for Your Ethernet Adapter" for details on getting an IP address.
- H. Check the **Enabled** option for the port services you want to use. Consider the examples below:

Enable	Application	Starting and Ending Ports	Protocol	IP Address
yes	UT	7777 to 27900	Both	192.168.0.100
yes	Halfife	27015 to 27015	Both	192.168.0.105
yes	PC Anywhere	5631 to 5631	UDP	192.168.0.102
yes	VPN IPSEC	500 to 500	UDP	192.168.0.100

When you have completed the configuration, click the **Submit All Changes** button.

### **16. I can't get the Internet game, server, or application to work.**

If you are having difficulties getting any Internet game, server, or application to function properly, consider exposing one PC to the Internet using DeMilitarized Zone (DMZ) hosting. This option is available when an application requires too many ports or when you are not sure which port services to use. Make sure you disable all the forwarding entries if you want to successfully use DMZ hosting, since forwarding has priority over DMZ hosting. (In other words, data that enters the System will be checked first by the forwarding settings. If the port number that the data enters from does not have port forwarding, then the System will send the data to whichever PC or network device you set for DMZ hosting.) Follow these steps to set DMZ hosting:

- A. Access the System's Web-based Utility by going to **http://192.168.0.1** or the IP address of the System. Go to the **Router => Application** tab.
- B. Disable or remove the entries you have entered for forwarding. Keep this information in case you want to use it at a later time.
- C. Select **yes** from the *Enable DMZ* drop-down menu.
- D. Enter the Ethernet adapter's IP address of the computer you want exposed to the Internet. This will bypass the NAT security for that computer. Please refer to "Appendix E: Finding the MAC Address and IP Address for Your Ethernet Adapter" for details on getting an IP address.

Once completed with the configuration, click the **Submit All Changes** button.

**17. I am a PPPoE user, and I need to remove the proxy settings or the dial-up pop-up window.**

If you have proxy settings, you need to disable these on your computer. Because the System is the gateway Internet access, the computer does not need proxy settings to gain access. Follow these directions to verify that you do not have any proxy settings and that the browser you use is set to connect directly to the LAN.

For Microsoft Internet Explorer 5.0 or higher:

- A. Click **Start, Settings, and Control Panel**. Double-click **Internet Options**.
- B. Click the **Connections** tab.
- C. Click the **LAN settings** button and remove anything that is checked.
- D. Click the **OK** button to go back to the previous screen.
- E. Click the option **Never dial a connection**. This will remove any dial-up pop-ups for PPPoE users.

For Netscape 4.7 or higher:

- A. Start **Netscape Navigator**, and click **Edit, Preferences, Advanced, and Proxies**.
- B. Make sure you have **Direct connection to the Internet** selected on this screen.
- C. Close all the windows to finish.

**18. To start over, I need to set the System to its factory default settings.**

Refer to “Chapter 5: Using the Interactive Voice Response Menu” for instructions.

**19. I need to upgrade the firmware.**

Firmware upgrades are handled by your ITSP. Contact your service provider for more information.

**20. When I enter a URL or IP address, I get a time-out error or am prompted to retry.**

Go through this checklist until your problem is solved:

- Check if other PCs work. If they do, ensure that your workstation’s IP settings are correct (IP Address, Subnet Mask, Default Gateway, and DNS). Restart the computer that is having a problem.
- If the PCs are configured correctly, but still not working, check the System. Make sure that it is connected and powered on. Check its settings. (If you cannot connect to it, check the LAN and power connections.)
- If you have a separate router, check the router. Make sure that it is connected and powered on. Check its settings. (If you cannot connect to it, check the LAN and power connections.)
- Check your Internet connection (DSL/cable modem) to see if it is working correctly. You can remove the System and router (if used) to verify a direct connection.
- Manually configure the TCP/IP with a DNS address provided by your ISP.
- Make sure that your browser is set to connect directly and that any dial-up is disabled. For Internet Explorer, click **Tools, Internet Options**, and then the **Connection** tab. Make sure that Internet Explorer is set to **Never dial a connection**. For Netscape Navigator, click **Edit, Preferences, Advanced, and Proxy**. Make sure that Netscape Navigator is set to **Direct connection to the Internet**.

## Frequently Asked Questions

### ***How do I make a phone call?***

For calls within your system, pick up your phone and dial the extension number you want. For external calls, follow the instructions of your ITSP. You may be asked to dial a specific prefix for outside calls. For more information, contact your ITSP.

### ***Can I make calls if my Internet connection is down?***

When you make Internet phone calls, your high-speed Internet connection must be active. However, if you have maintained a traditional analog phone account, then you can still make calls through the Linksys Analog Telephone Adapter (model number: SPA3000); refer to the SPA3000 documentation for more information.

### ***How do I access my voicemail?***

Voicemail is handled by your ITSP. Contact your provider for more information.

### ***How do I check which Internet phones have registered with the System?***

Open the web browser on a networked computer. Enter <http://192.168.0.1/admin/voice/status>. The *PBX Status* screen lists the Internet phones that have registered with the System.

### ***What is the maximum number of IP addresses that the System will support?***

The System will support up to 253 IP addresses.

### ***Does the System support IPX or AppleTalk?***

No. TCP/IP is the only protocol standard for the Internet and has become the global standard for communications. IPX, a NetWare communications protocol used only to route messages from one node to another, and AppleTalk, a communications protocol used on Apple and Macintosh networks, can be used for LAN to LAN connections, but those protocols cannot connect from the Internet to the LAN.

### ***What is Network Address Translation and what is it used for?***

Network Address Translation (NAT) translates multiple IP addresses on the private LAN to one public address that is sent out to the Internet. This adds a level of security since the address of a PC connected to the private LAN is never transmitted on the Internet. Furthermore, NAT allows the System to be used with low-cost Internet accounts, such as DSL or cable modems, when only one TCP/IP address is provided by the ISP. The user may have many private addresses behind this single address provided by the ISP.

### ***Does the System support any operating system other than Windows 98, Millennium, 2000, or XP?***

Yes, but Linksys does not, at this time, provide technical support for setup, configuration or troubleshooting of any non-Windows operating systems.

### ***Does the System support ICQ send file?***

Yes, with the following fix: click **ICQ menu => preference => connections** tab=>, and check **I am behind a firewall or proxy**. Then set the firewall time-out to 80 seconds in the firewall setting. The Internet user can then send a file to a user behind the System.

### ***I set up an Unreal Tournament Server, but others on the LAN cannot join. What do I need to do?***

If you have a dedicated Unreal Tournament server running, you need to create a static IP for each of the LAN computers and forward ports 7777, 7778, 7779, 7780, 7781, and 27900 to the IP address of the server. You can also use a port forwarding range of 7777 to 27900. If you want to use the UT Server Admin, forward another port (8080 usually works well but is used for remote admin. You may have to disable this.), and then in the [UWeb.WebServer] section of the server.ini file, set the ListenPort to 8080 (to match the mapped port above) and ServerName to the IP assigned to the System from your ISP.

### ***Can multiple gamers on the LAN get on one game server and play simultaneously with just one public IP address?***

It depends on which network game or what kind of game server you are using. For example, Unreal Tournament supports multi-login with one public IP.

### ***How do I get Half-Life: Team Fortress to work with the System?***

The default client port for Half-Life is 27005. The computers on your LAN need to have "+clientport 2700x" added to the HL shortcut command line; the x would be 6, 7, 8, and on up. This lets multiple computers connect to the same server. One problem: Version 1.0.1.6 won't let multiple computers with the same CD key connect at the same time, even if on the same LAN (not a problem with 1.0.1.3). As far as hosting games, the HL server does not need to be in the DMZ. Just forward port 27015 to the local IP address of the server computer.

### ***How can I block corrupted FTP downloads?***

If you get corrupted files when you use your FTP client, try using another FTP program.

### ***The web page hangs; downloads are corrupt, or nothing but junk characters are being displayed on the screen. What do I need to do?***

Force your Ethernet adapter to 10Mbps or half duplex mode, and turn off the "Auto-negotiate" feature of your Ethernet adapter as a temporary measure. (Look at the Network Control Panel in your Ethernet adapter's Advanced Properties tab.) Make sure that your proxy setting is disabled in the browser. Check [www.linksys.com](http://www.linksys.com) for more information.

### ***How do I upgrade the System's firmware?***

Firmware upgrades are handled by your ITSP. Contact your service provider for more information.

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### ***Will the System function in a Macintosh environment?***

Yes, but the System's setup pages are accessible only through Internet Explorer 5.0 or Netscape Navigator 5.0 or higher for Macintosh.

### ***I am not able to get the web configuration screen for the System. What can I do?***

You may have to remove the proxy settings on your Internet browser, e.g., Netscape Navigator or Internet Explorer. Or remove the dial-up settings on your browser. Check with your browser documentation, and make sure that your browser is set to connect directly and that any dial-up is disabled. Make sure that your browser is set to connect directly and that any dial-up is disabled. For Internet Explorer, click **Tools, Internet Options**, and then the **Connection** tab. Make sure that Internet Explorer is set to **Never dial a connection**. For Netscape Navigator, click **Edit, Preferences, Advanced**, and **Proxy**. Make sure that Netscape Navigator is set to **Direct connection to the Internet**.

### ***What is DMZ hosting?***

Demilitarized Zone (DMZ) allows one IP address (computer) to be exposed to the Internet. Some applications require multiple TCP/IP ports to be open. It is recommended that you set your computer with a static IP if you want to use DMZ Hosting. To get the LAN IP address, see "Appendix E: Finding the MAC Address and IP Address for Your Ethernet Adapter."

### ***If DMZ is used, does the exposed user share the public IP with the System?***

No.

### ***Does the System pass PPTP packets or actively route PPTP sessions?***

The System allows PPTP packets to pass through.

### ***Is the System cross-platform compatible?***

Any platform that supports Ethernet and TCP/IP is compatible with the System.

### ***Which modems are compatible with the System?***

The System is compatible with virtually any cable or DSL modem that supports Ethernet.

### ***How can I check whether I have static or DHCP IP addresses?***

Ask your ISP to find out.

### ***How do I get mIRC to work with the System?***

Under the Router => Applications tab, set the Starting and Ending Ports to **113** for the computer on which you are using mIRC.

**If your questions are not addressed here, refer to the Linksys website, [www.linksys.com](http://www.linksys.com).**

# Appendix B: Configuring the Nighttime Auto-Attendant

## Description of the Auto-Attendant

The auto-attendant is an internal service within the System. It plays pre-recorded voice messages that offer the caller a menu of choices, so the auto-attendant can appropriately direct the call. After the caller has made a choice, the call is routed to the appropriate extension, so the caller is connected to the correct party or presented with another menu of choices.

There are three auto-attendants available, one for daytime, one for nighttime, and one for weekends/holidays. By default, the daytime auto-attendant is enabled, and the first message it plays (Prompt ID 1) is suitable for business hours. This appendix covers the steps required to configure the auto-attendant for nighttime hours.

## Instructions for Setting Up the Nighttime Auto-Attendant

You can save up to 10 customized greetings. The first four have default messages, which can be changed through the Interactive Voice Response Menu.

Prompt ID	Default Audio Message
1	"If you know your party's extension, you may enter it now."
2	"Your call has been forwarded."
3	"Not a valid extension, please try again."
4	"Goodbye."

If you want a caller to hear a different greeting during nighttime (non-business) hours, then you should record a new prompt, such as Prompt ID 5, using the Interactive Voice Response Menu and then configure the auto-attendant settings using the Web-based Utility. For example, Prompt ID 5 could say, "The company is currently closed. Our business hours are 9 AM to 5 PM, Monday to Friday."

The following instructions explain how to record Prompt ID 5 and configure the nighttime auto-attendant to use Prompt ID 5 as the initial greeting. You can also use these instructions to record additional prompts and further customize your auto-attendant, as long as you also update the AA script 2 code accordingly through the Web-based Utility (refer to "Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users").

## Recording a New Prompt

To record a new prompt, follow these instructions:

1. Using one of the analog telephones connected to the System, press \*\*\*\* (in other words, press the star key four times).
2. Wait until you hear “Linksys configuration menu. Please enter the option followed by the # (pound) key or hang up to exit.”
3. Press **72255#** to access the auto-attendant message settings.
4. You will hear, “Please enter the message number followed by the # key.” Press **5#**.
5. The Interactive Voice Response Menu will say, “Enter **1** to record. Enter **2** to review. Enter **3** to delete. Enter **\*** to exit.”
6. Press **1** and record your message.
7. After you have finished your message, press **#**.
8. After you record the message, you will hear, “To save, enter **1**. To review, enter **2**. To re-record, enter **3**. To exit, enter **\***.”

If you entered 1, the new message will be saved.

If you entered 2, you will hear the message played.

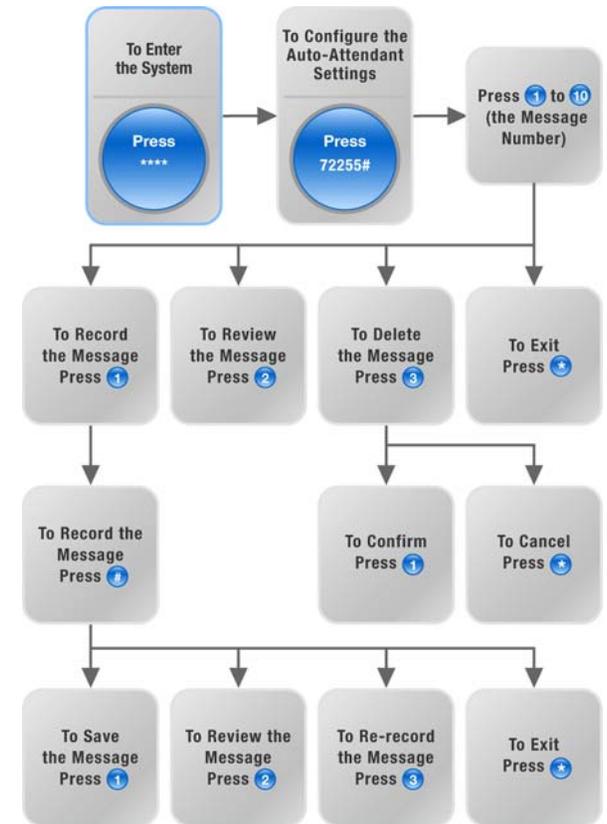
If you entered 3, you will be returned to step 7.

If you entered \*, you will be returned to the menu in step 5.

9. When you have finished recording Prompt ID 5, then hang up the telephone.

For more information about the Interactive Voice Response Menu, refer to “Chapter 5: Using the Voice Interactive Response Menu.” For status information about the auto-attendant messages or to configure advanced settings, such as dial plan rules, refer to “Chapter 6: Using the Web-based Utility.”

**To configure the nighttime auto-attendant, proceed to the next section.**



**Figure B-1: Auto-Attendant Message Options**



**NOTE:** If there is not enough memory left to record a new message, then you will hear, “Option failed” and be returned to step 4.



**NOTE:** If the message you want to save is longer than 15 seconds, then you will hear, “One moment, please.” This indicates that it will take several seconds to save the message. After the message has been saved, you can continue to use the Interactive Voice Response Menu.

## Configuring the Auto-Attendant Settings

To configure the nighttime auto-attendant, follow these instructions:

1. Launch Internet Explorer or Netscape Navigator on the administration computer.
2. Enter **<IP address of the System>/admin/voice/advanced** in the web browser's *Address* field. (Enter the IP address you assigned to the System when you installed it.)

Then press the **Enter** key.

3. The *Voice - Info* screen will appear. Click the **SIP** tab.
4. On the *SIP* screen, scroll down to the Auto Attendant Parameters.
5. For the *AA script 2* field, follow these instructions:
  - a. Copy the default *AA script 1* text to the Notepad (or other word processing program).

This is the default AA script 1 code:

```
<aa>
<form id="dir" type="menu">
  <audio src="prompt1" bargein="T"/>
  <noinput timeout="10" repeat="T"/>
  <nomatch repeat="F">
    <audio src="prompt3" bargein="T"/>
  </nomatch>
  <dialplan src="dp1"/>
  <match>
    <default>
      <audio src="prompt2"/>
      <xfer name="ext" target="$input"/>
    </default>
  </match>
</form>
</aa>
```

The screenshot shows the 'Auto Attendant Parameters' configuration page. It includes the following fields and values:

- AA Dial Plan 1: (<0:501>|1|<2:200>|<3:300>|5xx|8001)
- AA Dial Plan 2: (<0:501>|<2:200>|<3:300>|5xx|8001)
- AA script 1: <aa><form id="dir" type="menu"><audio src="prompt1" bargein="T"/><a
- AA script 2: <aa><form id="nt" type="menu"><audio src="prompt1" bargein="T"/><a
- AA script 3:
- DayTime AA: yes (dropdown)
- DayTime AA Script: 1 (dropdown)
- NightTime AA: yes (dropdown)
- NightTime AA Script: 2 (dropdown)
- Weekends/Holidays:
- Weekend/Holiday AA Script: 2 (dropdown)
- Day Time:
- DayTime Answer Delay: 12
- NightTime Answer Delay: 0
- Weekend/Holiday Answer Delay: 0

**Figure B-2: Voice - SIP Screen - Auto Attendant Parameters**

- b. Replace “dir” with “nt” (see the new text in boldface). Then add this line of code:  
**<audio src="prompt5" bargein="T"/>**

This is the AA script 2 code:

```
<aa>
<form id="nt" type="menu">
  <audio src="prompt5" bargein="T"/>
  <audio src="prompt1" bargein="T"/>
  <noinput timeout="10" repeat="T"/>
  <nomatch repeat="F">
    <audio src="prompt3" bargein="T"/>
  </nomatch>
  <dialplan src="dp1"/>
  <match>
    <default>
      <audio src="prompt2"/>
      <xfer name="ext" target="$input"/>
    </default>
  </match>
</form>
</aa>
```

- c. Copy the AA script 2 code from the Notepad, and paste it in the *AA script 2* field.
6. For the *DayTime* field, enter the daytime hours for the daytime auto-attendant in 24-hour format. The start and end times should be in this format:

start=hh:mm:ss;end=hh:mm:ss  
(hh for hours, mm for minutes, and ss for seconds)

For example, start=9:0:0;end=17:0:0 means the start time is 9 AM and the end time is 5 PM. The other hours (5 PM to 9 AM) are considered nighttime hours.

7. For the *NightTime AA* setting, select **yes**.
8. For the *NightTime AA Script* setting, select **2**.
9. Click the **Submit All Changes** button to save your new settings.

**Congratulations! You have set up the nighttime auto-attendant.**



**NOTE:** If you want to configure the Weekend/Holiday Auto-Attendant, refer to “Chapter 6: Using the Web-based Utility” for instructions.

# Appendix C: Dial Plan and Auto-Attendant Scripting for Advanced Users

## Overview

This appendix discusses the dial plan and auto-attendant (aa) features of the System. It also explains how to configure and write scripts for both features.

## Configuring Dial Plans

The System allows each phone line to be configured with a distinct dial plan. The dial plan specifies how to interpret digit sequences dialed by the caller and how to convert those sequences into an outbound dial string.

The Dial Plan settings contain the actual dial plan scripts for the various lines. Each plan contains a series of digit sequences, separated by this character, | . The collection of sequences is collected in parentheses, ' and '.

When a caller dials a series of digits, each sequence in the dial plan is tested as a possible match. The matching sequences form a set of candidate digit sequences. As more digits are entered by the caller, the candidates are eliminated until only one or none is valid.

The following table describes the entries to use for the dial plan.

**Table 1: Dial Plan Entries**

Dial Plan Entry	Function
*xx	Allows arbitrary 2-digit star code
[3469]11	Allows x11 sequences (for example, 311, 411, 611, 911)
0	Dials operator
00	Dials international operator
[2-9]xxxxxx	Dials US local number
1xxx[2-9]xxxxxx	Dials US 1 + 10-digit long distance number
xxxxxxxxxx.	Dials all other numbers, including international long distance



**NOTE:** If the default US dial plan does not meet your needs, visit [www.linksys.com/kb](http://www.linksys.com/kb) for additional dial plans, or use this appendix to write your own script.

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Entries include the following:

- Individual characters include 0 to 9, \*, #.
- The letter x matches any single number from 0 to 9.
- A subset of keys within brackets represents a range: [set].  
For example, [389] means 3 or 8 or 9
  - A numeric range is allowed within the brackets: [digit-digit].  
For example, [2-9] means 2, 3, 4, 5, 6, 7, 8, or 9.
  - A range can be combined with other keys.  
For example, [235-8\*] means, 2, 3, 5, 6, 7, 8, or \*.

These are the rules for the entries:

- Any key can be repeated zero or more times by adding a . (period) to the end.  
For example, 01. matches 0, 01, 011, 0111...
- A sub-sequence of keys, which can be empty, can be automatically replaced with a different sub-sequence using an angled bracket notation: <dialed-sub-sequence:transmitted-sub-sequence>.  
For example, 8:1650.xxxxxx matches 85551212 and transmits 16505551212.
- An outside line dial tone can be generated within a sequence by adding a , (comma) between digits.  
For example, 9,1xxxxxxxx sounds an outside line dial tone after the caller presses 9, until the 1 is pressed.
- A sequence can be barred or rejected by placing an ! (exclamation mark) at the end of the sequence.  
For example, 1900xxxxxx! automatically blocks all 900 area code numbers from being dialed.

Here are some examples of dial plans:

(1xxxxxxxx) - This dial plan accepts only US-style 1 + area code + local number, with no restrictions on the area code and number.

(1xxxxxxxx|<:1212>xxxxxx) - This dial plan allows US-style, 7-digit dialing and automatically inserts a 1 + 212 (local area code) in the transmitted number.

(<9,:>1xxxxxxxx|<8,:1212>xxxxxx) - This dial plan requires a caller to dial 8 as a prefix for local calls and 9 as a prefix for long distance. In either case, an outside line tone is played after the initial 8 or 9, and neither prefix is transmitted when the call is initiated.

(\*xx\[3469]11\0\00\[2-9]xxxxxx\1xxx[2-9]xxxxxx\xxxxxxxxxx.) - This dial plan permits these calls: arbitrary 2-digit star code calls, 311, 411, 611, 911, local operator calls, international operator calls, US local numbers, US 1 + 10-digit long distance number, and all other numbers.

## Configuring Dial Plans for the Auto-Attendant

You can define the dial rule in the dial plan setting and then do the translation in the auto-attendant script. In this case, the dial plan can be very simple, like (112131415xxx) or (xxxx|\*|#).

For example, the dial plan can be (<x:500x>|408555xxxx|xxxxx) or (<1:1002>|<2:21111>|<3:3333>|xxxxx). When the caller inputs some DTMF digits, the auto-attendant will parse them using the dial plan first, and then the parsing result will be fed into the auto-attendant script menu instruction. Each auto-attendant menu has a dial plan. You can define the dial rule in the settings, AA Dial Plan 1 and/or AA Dial Plan 2. Each auto-attendant dial plan setting has a matching ID, which can be used in auto-attendant XML scripting. For example, a user can specify dp1 to indicate AA Dial Plan 1.

**Table 2: Auto-Attendant Dial Plans**

Web-based Utility Setting	Matching ID in Auto-Attendant Script
AA Dial Plan 1	dp1
AA Dial Plan 2	dp2

## Configuring the Auto-Attendant

The auto-attendant (aa) is an internal service within the System. It plays pre-recorded voice messages that offer the caller a menu of choices, so the auto-attendant can appropriately direct the call. For example, a greeting could be, “Welcome to the abc company. For sales, press 1. For service, press 2. To speak to our operator, press 3.” (This custom greeting would have to be recorded using the Interactive Voice Response Menu.)

After the caller has made a choice, the call is routed to the appropriate extension, so the caller is connected to the correct party or presented with another menu of choices.

There are three auto-attendants available, one for daytime, one for nighttime, and one for weekends/holidays.

## Internal Dial Plan

When the auto-attendant is enabled, it parses and operates on user input (key presses or DTMF tones) following the rules specified in the auto-attendant dial plan of the System. These rules are specified by the AA Dial Plan parameters found on the *Voice - SIP* screen of the Web-based Utility.

## Parameters of the Auto-Attendant Feature

The following parameters are the minimum that must be configured.

- **Contact List.** This is a list of clients that the System alerts when there is an incoming call on the line. The auto-attendant must be included on this list. By default, the auto-attendant is the only client on this list, so the auto-attendant will pick up every call. You can decide to have the auto-attendant pick up a call if a number (or group) of clients did not pick up the call first. This parameter is configured on the *Voice - Line x* (x is 1-4) screen of the Web-based Utility.
- **AA Script.** The System lets you program the auto-attendant instructions using XML scripting grammar. These parameters, AA Scripts 1-3, are configured on the *Voice - SIP* screen of the Web-based Utility. Only one script is active at a time. Scripting is described in more detail below.
- **AA Dial Plan.** The auto-attendant parses the user input according to one of the two parameters, AA Dial Plan 1 or 2. The AA Script includes a reference to one of these two dial plan parameters through the dial plan instruction. These parameters are configured on the *Voice - SIP* screen of the Web-based Utility.
- **DayTime AA Script.** This defines which of the three scripts (AA script 1, 2, or 3) should be used for daytime hours. This parameter is configured on the *Voice - SIP* screen of the Web-based Utility.

## Prompts for the Auto-Attendant

The auto-attendant prompts are configured through the Interactive Voice Response Menu. To access these settings, follow these instructions:

1. Using one of the analog telephones connected to the System, press **\*\*\*\*** (in other words, press the star key four times).
2. Wait until you hear “Linksys configuration menu. Please enter the option followed by the # (pound) key or hang up to exit.”
3. Press **72255#** to access the auto-attendant message settings.

Refer to “Chapter 5: Using the Interactive Voice Response Menu” for more details.

The System can store up to 94.5 seconds of audio, excluding the default messages (Prompts 1-4). The maximum length of any message is one minute. The recorded messages will be encoded with G711U and saved in flash memory. Status information for these messages is found in the Auto-Attendant Prompt Status section of the *Voice - Info* screen of the Web-based Utility.

Each message is internally referred to as Prompt x, x being a number ranging from 1 to 10. You can customize the default prompts and add six additional prompts. When you reset the System to its factory default settings, customized messages will be erased, and prompts 1-4 will be restored to their default messages:

**Table 3: Default Auto-Attendant Prompts**

Prompt ID	Default Audio Message
1	"If you know your party's extension, you may enter it now."
2	"Your call has been forwarded."
3	"Not a valid extension, please try again."
4	"Goodbye."

### Customizing the Auto-Attendant

You have many parameters you can change to customize the auto-attendant. One of the most important parameters is the script, or set of instructions, that the auto-attendant executes when it is running. The next section will explain how to use XML scripting grammar.

#### The AA Scripts Parameters and XML Scripting

The System lets you use XML scripting grammar to define the auto-attendant instructions. You have a choice of three scripts, which are stored in the AA script 1-3 parameters on the *Voice - SIP* screen of the Web-based Utility. The instructions must be defined or encapsulated in a `<form>` structure. You may have multiple `<form>` structures within a script that the auto-attendant can transfer to, based on user input.

The XML scripting grammar supports two types of `<form>` structures, node and menu. The main difference between the two types is that within the node type, user input cannot be processed—only actions may be specified. The format of the node type is as follows:

```
<form id="form-id" type="node">
  <!--audio instruction (optional) -->
  <!--action instruction pair (mandatory) -->
</form>
```

In the menu type, user input can be processed. It is processed according to the dial plan statement associated with the menu and defines what action the auto-attendant executes when the user input matches the dial plan. The format of the menu type is as follows:

```
<form id="form-id" type="menu">
  <!-- dialplan instruction (mandatory)-->
  <!-- noinput instruction (optional) -->
  <!-- nomatch instruction (optional) -->
  <!-- match instruction (mandatory) -->
</form>
```

The complete set of XML instructions are described in the following table:

**Table 4: Auto-Attendant XML Instructions Set**

Instruction	Description	Syntax and Example(s)
dialplan	This determines the dialplan id of the current menu <form>. The auto-attendant will process the user input according to the dial plan and then will be dispatched to the match, nomatch, or noinput instruction.	<pre>&lt;dialplan src = "dp1"/&gt; "dp1" matches the AA Dial Plan 1 parameter found on the Voice - SIP screen of the Web-based Utility. "dp2" matches the AA Dial Plan 2 parameter found on the Voice - SIP screen of the Web-based Utility.</pre>
noinput	When specified, the auto-attendant will execute the specified audio and action instructions if the user does not input any digits in <timeout> seconds. If the repeat attribute is set to "T", then the auto-attendant will play the menu prompt after playing the prompt specified in the <noinput> audio instruction and ignore the action instruction; otherwise, the auto-attendant will execute the action instruction. By default, "repeat" is "F".	<pre>&lt;noinput timeout="5" repeat="T"&gt;   &lt;!--audio instruction (optional) --&gt;   &lt;!--action instruction (optional) --&gt; &lt;/noinput&gt;</pre>
nomatch	When specified, the nomatch instruction runs when the user input digits do not match anything in the dial plan. The auto-attendant will execute the specified audio and action instructions. If the repeat attribute is set to "T", the auto-attendant will play the menu prompt after playing the no input prompt and ignore the action instruction; otherwise, the auto-attendant will execute the action instruction. By default, "repeat" is "F".	<pre>&lt;nomatch repeat="F"&gt;   &lt;!--audio instruction (optional) --&gt;   &lt;!--action instruction (optional) --&gt; &lt;/nomatch&gt;</pre>

**Table 4: Auto-Attendant XML Instructions Set**

Instruction	Description	Syntax and Example(s)	
match	Upon a match between the user's input and the dial plan, the auto-attendant will transfer to the corresponding <case> and execute the corresponding audio and/or action instructions. If the auto-attendant cannot find a match in any of the <case> statements, it will perform the <default> case.	<pre> &lt;match &gt;   &lt;case input= "x"/&gt;     &lt;!--audio instruction (optional) --&gt;     &lt;!--action instruction (optional) --&gt;   &lt;/case&gt;   &lt;case input= "#"/&gt;     &lt;!--audio instruction (optional) --&gt;     &lt;!--action instruction (optional) --&gt;   &lt;/case&gt;   &lt;default&gt;     &lt;!--audio instruction (optional) --&gt;     &lt;!--action instruction (optional) --&gt;   &lt;/default&gt; &lt;/match&gt; </pre>	
Action Instructions	goto	The auto-attendant will transfer the caller from one <form> to the other <form>. All <form>s are identified by the attribute "id". The value in the id attribute must be unique; otherwise, the auto-attendant will select the last valid <form> as the transfer-to target.	<pre> &lt;goto link= "daytime"&gt; </pre> <p>"daytime" is the id of a &lt;form&gt; entry. Example: &lt;form id="daytime" type="menu"&gt;</p>
	xfer	The auto-attendant will perform a blind transfer of the caller to the target, and then it will end processing "target = \$input" is equivalent to the input value already passed by the dialplan. There is no significance to the name attribute.	<pre> &lt;xfer name= "Technical Support" target= "5000"/&gt; </pre>
	exit	When this action is reached, the auto-attendant will stop, and the call will be ended.	<pre> &lt;exit&gt; </pre>
audio	The auto-attendant will play the audio specified in the "src" attribute. This attribute must be prompt<n>, with <n> being a number in the range 1–10. When playing the audio, the auto-attendant will allow the caller to interrupt the current prompt by pressing digits if the bargein attribute is set to "T". The auto-attendant will ignore any digits from the caller if the bargein attribute is set to "F" (the default value).	<pre> &lt;audio src= "prompt1" bargein= "T"/&gt; </pre>	

Additional Parameters for Customization

Additional parameters are available on the *Voice - SIP* screen of the Web-based Utility and described in the following table:

**Table 5: Additional Auto-Attendant Configuration Parameters**

Parameter	Description	Type
AA Dial Plan 1/2	This describes the dial plan rule that the auto-attendant will use in a particular <form>.	Str256
AA Script 1/2/3	There are three parameters for storing auto-attendant scripts. One option is to use these as different treatments for daytime hours, nighttime hours, and weekends/holidays.	Str1024
Daytime AA Nighttime AA Weekend/Holiday AA	Each parameters controls whether the corresponding auto-attendant service will be enabled or disabled. This parameter can also be manually controlled through the Interactive Voice Response Menu (code: 79228). Refer to “Chapter 5: Using the Interactive Voice Response Menu.”	Bool
Daytime AA Script Nighttime AA Script Weekend/Holiday AA Script	This parameter specifies which script will be used for a particular auto-attendant treatment. The choices are 1 for the AA script 1 setting, 2 for the AA script 2 setting, and 3 for the AA script 3 setting.	Choice
DayTime Answer Delay NightTime Answer Delay Weekend/Holiday Delay	Each auto-attendant service has a corresponding delay setting, which is the number of seconds you want the auto-attendant to wait before answering. By default, the DayTime Answer Delay setting is 12 seconds (the operator is given 12 seconds to answer the phone). The NightTime and Weekend/Holiday Answer Delay settings are set to 0 (the auto-attendant will answer immediately).	Number
Weekends/Holidays	<p>This parameter defines the dates of weekends and holidays.</p> <p>Format: [wk=n1[,ni];][hd=mm/dd/yyyy mm/dd/yyyy-mm/dd/yyyy[, mm/dd/yyyy mm/dd/yyyy-mm/dd/yyyy];]</p> <p>The abbreviation wk stands for weekend. The value can be any number, 1 to 7, to represent Monday to Sunday. Up to four weekend days can be defined.</p> <p>The abbreviation hd stands for holiday. You do not have to include the year (yyyy) if you want the same date to apply every year.</p> <p>Example: wk=6,7;hd=1/1,7/4</p> <p>In this example, the weekend includes Saturday (6) and Sunday (7), and the holidays are January 1st and July 4th every year.</p>	Str512

**Table 5: Additional Auto-Attendant Configuration Parameters**

Parameter	Description	Type
DayTime	<p>This parameter defines the daytime hours. (The other hours are considered nighttime hours.)</p> <p>Format: start=hh:mm:ss;end=hh:mm:ss</p> <p>The abbreviation hh stands for hours and uses the 24-hour format. The abbreviation mm stands for minutes, and the abbreviation ss stands for seconds.</p> <p>Example: start=9:0:0;end=17:0:0</p> <p>In this example, the start time is 9 AM and the end time is 5 PM. The other hours (5 PM to 9 AM) are considered nighttime hours.</p> <p>If you do not enter start and end times, then the whole day (24 hours) is considered as daytime, so the nighttime auto-attendant will be disabled, even if the NightTime AA parameter is set to yes (enabled).</p>	Str64

### Example of a Typical Auto-Attendant Setup

This section will describe a typical auto-attendant. Here is the scenario:

An incoming call on Line 1 sequentially tries extensions 501, 503, and then 502. If the call is not answered, then the auto-attendant picks up the call. The auto-attendant announces prompts 5, 8, and 9 in sequence, and then waits up to 10 seconds for user input. If no input is received, then the call is forwarded to extension 501. If the user enters the following:

- an invalid extension - Prompt 3 is played. Afterwards, the auto-attendant starts over again.
- 1 - Prompts 7, 8, and then 9 are played. Afterwards, the auto-attendant starts over again.
- 2 - The auto-attendant begins executing the SUPPORT form and starts playing prompt 10. The significant differences between the 2 forms are as follows:
  - The dial plan has been changed to the AA Dial Plan 2 parameter. This includes the \* character (asterisk), which instructs the auto-attendant to resume with the “dt” form.
  - Prompt 10 is played every five seconds, until the user inputs an extension.
- All other valid extensions (defined here to be 0, 3, 5xx, 8001) are forwarded to the associated client.

The following table lists the parameters for this example.

**Table 6: Configuration Parameters for the Example**

Parameter	Value
(Line 1) Contact List	501,503,502,hunt=re;4;1,cfwd=aa
AA Dial Plan 1	(<0:501> 1 <2:200> <3:300> 5xx 8001)
AA Dial Plan 2	(<1:501> * <2:502> <3:503> 5xx)
AA script 1	See form id= "dt" below
AA script 2	See form id= "nt" below
DayTime AA	yes
DayTime AA Script	1
NightTime AA	yes
NightTime AA Script	2
Prompt 1-4	See the defaults in Table 3: Default Auto-Attendant Prompts.
Prompt 5	"Welcome to All Seasons Travel."
Prompt 6	"Thank you for calling All Seasons Travel. Presently we are closed."
Prompt 7	"We are open Monday through Friday 9 AM to 5:30 PM, Saturdays from 9 AM to noon, and we are closed on Sundays. Our address is 101 Main Street, Anytown, Anystate, USA."
Prompt 8	"If you know your party's three-digit extension, you may enter it now."
Prompt 9	"To reach our receptionist, press 0 at any time. For our company location, press 1. For sales, press 2. For travel support, press 3. Otherwise, please stay on the line for our receptionist."
Prompt 10	"Welcome to the All Seasons Travel support line. If this is regarding our Holiday Getaway Special, please press 1; otherwise, please stay on the line for one of our travel associates."

The following is the actual code for this example:

```
<aa>
  <form id="nt" type="menu">
    <audio src="prompt6" bargein="F"/>
    <audio src="prompt7" bargein="F"/>
    <audio src="prompt8" bargein="T"/>
    <noinput timeout="15" repeat="T"/>
    <nomatch repeat="F">
      <audio src="prompt3" bargein="T"/>
    </nomatch>
    <dialplan src="dp1"/>
    <match>
      <default>
        <audio src="prompt2"/>
        <xfer name="ext" target="$input"/>
      </default>
    </match>
  </form>
</aa>
```

```
<aa>
  <form id="dt" type="menu">
    <audio src="prompt5" bargein="T"/>
    <audio src="prompt8" bargein="T"/>
    <audio src="prompt9" bargein="T"/>
    <noinput timeout="10">
      <xfer name="rep" target="501"/>
    </noinput>
    <nomatch repeat="T">
      <audio src="prompt3" bargein="T"/>
    </nomatch>
    <dialplan src="dp1"/>
    <match>
      <case input="1">
        <audio src="prompt7" bargein="F"/>
        <audio src="prompt8" bargein="T"/>
        <audio src="prompt9" bargein="T"/>
      </case>
      <case input="2">
```

## IP Telephony System

```
        <audio src="prompt2" bargein="F"/>
        <goto link="SUPPORT"/>
    </case>
    <default>
        <audio src="prompt2" bargein="T"/>
        <xfer name="ext" target="$input"/>
    </default>
</match>
</form>
<form id="SUPPORT" type="menu">
    <dialplan src="dp2"/>
    <audio src="prompt10" bargein="T"/>
    <noinput timeout="5" repeat="T"/>
    <nomatch repeat="T">
        <audio src="prompt3" bargein="T"/>
    </nomatch>
    <match>
        <case input="*">
            <audio src="prompt2" bargein="F"/>
            <goto link="dt"/>
        </case>
        <default>
            <audio src="prompt2"/>
            <xfer name="ext" target="$input"/>
        </default>
    </match>
</form>
</aa>
```

# Appendix D: New Music for the Music-on-Hold Feature

## Overview

The System provides a default internal music file for music-on-hold. This is the music file heard by callers when their calls are placed on hold or in the parking lot (call park feature). You can use a TFTP transfer to replace the default file with your own music file.

This appendix describes how to convert your music file to the appropriate format and then configure the System.

## Before You Begin

Make sure you have the following for the computer you are using:

- TFTP server software
- a program to convert the music file (for example, MP3) to the raw G711u format
- its local IP address

You can use any TFTP server software and any music conversion program that supports the raw G711u format.

To find the local IP address of this computer, follow these instructions:

1. Click **Start**.
2. Click **Run**.
3. Enter **cmd** in the *Open* field, and click the **OK** button.
4. At the prompt, enter **ipconfig** and press the **Enter** key.
5. The local IP address of the computer will be displayed. Write it down; you will use it later.

## Instructions for Converting the Music File

The System only supports the raw G711u format. Follow these instructions to convert your music file to the supported format:

1. Open the music conversion program you use.
2. Convert the music file to this format:
  - Raw Type -  $\mu$ -Law
  - Frequency - 8 kHz
  - Bits per Sample - 8 Kps
  - Channels - Mono

The music file should be encoded in G711u format at 8000 samples/second. It should not contain any extra header information, and its maximum length is 65.536 seconds (524,288 bytes).

3. Save the converted music file to the TFTP filesystem root directory.

This file must be saved in the root directory so the TFTP server can upload it to the System later.

## Instructions for Configuring the IP Telephony System

Before you begin, make sure the TFTP server software is running on your computer.

1. To access the Web-based Utility of the System, launch Internet Explorer or Netscape Navigator, and enter the System's default local IP address, **192.168.0.1**, in the *Address* field. Press the **Enter** key.
2. You may be asked for your user name and password. Enter the user name. The default user name for administrative access is **admin**, and the default user name for user access is **user**. (These user names cannot be changed.) Then enter the password supplied by your Internet Telephony Service Provider (ITSP).
3. Click the **Voice** tab.
4. Click the **SIP** tab.
5. In the PBX Parameters section, enter **tftp://server\_IP\_address:port/path** in the *Internal Music URL* field.

The *server\_IP\_address* is the local IP address of the computer you are using. The port is the port number used by the TFTP server; by default it is **69**. The path indicates the location and name of the converted music file, which has been saved in the TFTP filesystem root directory.

## IP Telephony System

For example, if the computer's local IP address is 192.168.0.5, the directory is named rootdirectory, and the converted music file is named jazzmusic.dat, then `ftp://192.168.0.5:69/rootdirectory/jazzmusic.dat` should be entered in the *Internal Music URL* field. (The default port number, **69**, is used.)



**NOTE:** Keep the default, **imusic**, in the *Call Park MOH Server* field, so the call park feature will use the music file you have downloaded (imusic is the internal music file of the System).

6. Click the **Submit All Changes** button.
7. The System will reboot.
8. After its reboot, the System will download the file and store the samples in flash memory.

# Appendix E: Finding the MAC Address and IP Address for Your Ethernet Adapter

This section describes how to find the MAC address for your computer's Ethernet adapter so you can use the MAC address cloning feature of the Router. You can also find the IP address of your computer's Ethernet adapter. This IP address is used for the Router's forwarding and/or DMZ features. Follow the steps in this appendix to find the adapter's MAC or IP address in Windows 98, Me, 2000, or XP.

## Windows 98 or Me Instructions

1. Click **Start** and **Run**. In the *Open* field, enter **winipcfg**. Then press the **Enter** key or the **OK** button.
2. When the *IP Configuration* screen appears, select the Ethernet adapter you have connected to the Router via a CAT 5 Ethernet network cable.
3. Write down the Adapter Address as shown on your computer screen. This is the MAC address for your Ethernet adapter and is shown as a series of numbers and letters.

The MAC address/Adapter Address is what you will use for MAC address cloning or MAC filtering.

The example shows the Ethernet adapter's IP address as 192.168.1.100. Your computer may show something different.



**NOTE:** The MAC address is also called the Adapter Address.

## Windows 2000 or XP Instructions

1. Click **Start** and **Run**. In the *Open* field, enter **cmd**. Press the **Enter** key or click the **OK** button.
2. At the command prompt, enter **ipconfig /all**. Then press the **Enter** key.

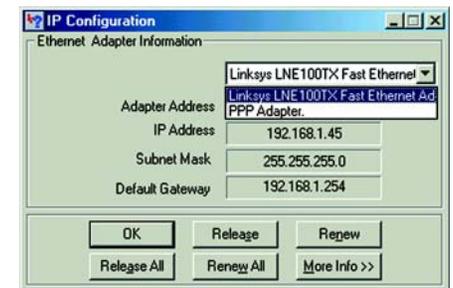


Figure E-1: IP Configuration Screen

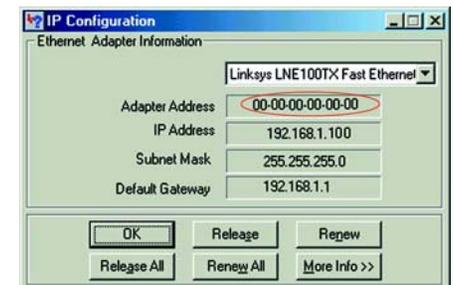


Figure E-2: MAC Address/Adapter Address

## IP Telephony System

- Write down the Physical Address as shown on your computer screen; it is the MAC address for your Ethernet adapter. This appears as a series of numbers and letters.

The MAC address/Physical Address is what you will use for MAC address cloning or MAC filtering.



**NOTE:** The MAC address is also called the Physical Address.

The example shows the Ethernet adapter's IP address as 192.168.1.100. Your computer may show something different.

## For the Router's Web-based Utility

For MAC address cloning, enter the 12-digit MAC address in the *Cloned MAC Address* field.

```
C:\>ipconfig /all
Windows 2000 IP Configuration

Host Name . . . . . :
Primary DNS Suffix . . . . . :
Node Type . . . . . : Hybrid
IP Routing Enabled. . . . . : No
WINS Proxy Enabled. . . . . : No

Ethernet adapter Local Area Connection:
   Connection-specific DNS Suffix  . :
   Description . . . . . : Linksys LNE100TX(c05) Fast Ethernet A
   Physical Address. . . . . : 00-00-00-00-00-00
   DHCP Enabled. . . . . : Yes
   Autoconfiguration Enabled . . . . : Yes
   IP Address. . . . . : 192.168.1.100
   Subnet Mask . . . . . : 255.255.255.0
   Default Gateway . . . . . : 192.168.1.1
   DHCP Server . . . . . : 192.168.1.1
   DNS Servers . . . . . : 192.168.1.1
   Primary WINS Server . . . . . : 192.168.1.1
   Secondary WINS Server . . . . . :
   Lease Obtained. . . . . : Monday, February 11, 2002 2:31:47 PM
   Lease Expires . . . . . : Tuesday, February 12, 2002 2:31:47 P
C:\>
```

Figure E-3: MAC Address/Physical Address

The screenshot shows a web-based utility interface for a router. The 'MAC Clone Settings' section is visible, with 'Enable MAC Clone Service' set to 'no' and 'Cloned MAC Address' field empty. Other sections include Internet Connection Settings, Static IP Settings, PPPoE Settings, Optional Settings, Remote Management, QoS Settings, and VLAN Settings.

Figure E-4: MAC Address Clone

# Appendix F: Windows Help

Almost all networking products require Microsoft Windows. Windows is the most used operating system in the world and comes with many features that help make networking easier. These features can be accessed through Windows Help and are described in this appendix.

## TCP/IP

Before a computer can communicate with the Router, TCP/IP must be enabled. TCP/IP is a set of instructions, or protocol, all PCs follow to communicate over a wired or wireless network. Your PCs will not be able to utilize networking without having TCP/IP enabled. Windows Help provides complete instructions on enabling TCP/IP.

## Shared Resources

If you wish to share printers, folders, or files over your network, Windows Help provides complete instructions on utilizing shared resources.

## Network Neighborhood/My Network Places

Other PCs on your network will appear under Network Neighborhood or My Network Places (depending upon the version of Windows you're running). Windows Help provides complete instructions on adding PCs to your network.

# Appendix G: Glossary

**Adapter** - A device that adds network functionality to your PC.

**Backbone** - The part of a network that connects most of the systems and networks together, and handles the most data.

**Bandwidth** - The transmission capacity of a given device or network.

**Bit** - A binary digit.

**Boot** - To start a device and cause it to start executing instructions.

**Bridge** - A device that connects different networks.

**Broadband** - An always-on, fast Internet connection.

**Browser** - An application program that provides a way to look at and interact with all the information on the World Wide Web.

**Buffer** - A shared or assigned memory area that is used to support and coordinate different computing and networking activities so one isn't held up by the other.

**Byte** - A unit of data that is usually eight bits long

**Cable Modem** - A device that connects a computer to the cable television network, which in turn connects to the Internet.

**CSMA/CA (Carrier Sense Multiple Access/Collision Avoidance)** - A method of data transfer that is used to prevent data collisions.

**Daisy Chain** - A method used to connect devices in a series, one after the other.

**Database** - A collection of data that is organized so that its contents can easily be accessed, managed, and updated.

**DDNS (Dynamic Domain Name System)** - Allows the hosting of a website, FTP server, or e-mail server with a fixed domain name (e.g., www.xyz.com) and a dynamic IP address.

**Default Gateway** - A device that forwards Internet traffic from your local area network.

## IP Telephony System

**DHCP (Dynamic Host Configuration Protocol)** - A networking protocol that allows administrators to assign temporary IP addresses to network computers by “leasing” an IP address to a user for a limited amount of time, instead of assigning permanent IP addresses.

**DMZ (Demilitarized Zone)** - Removes the Router's firewall protection from one PC, allowing it to be “seen” from the Internet.

**DNS (Domain Name Server)** - The IP address of your ISP's server, which translates the names of websites into IP addresses.

**Domain** - A specific name for a network of computers.

**Download** - To receive a file transmitted over a network.

**DSL (Digital Subscriber Line)** - An always-on broadband connection over traditional phone lines.

**Dynamic IP Address** - A temporary IP address assigned by a DHCP server.

**Encryption** - Encoding data transmitted in a network.

**Ethernet** - A networking protocol that specifies how data is placed on and retrieved from a common transmission medium.

**Firewall** - A set of related programs located at a network gateway server that protects the resources of a network from users from other networks.

**Firmware** - The programming code that runs a networking device.

**Fragmentation** - Breaking a packet into smaller units when transmitting over a network medium that cannot support the original size of the packet.

**FTP (File Transfer Protocol)** - A protocol used to transfer files over a TCP/IP network.

**Full Duplex** - The ability of a networking device to receive and transmit data simultaneously.

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

**Half Duplex** - Data transmission that can occur in two directions over a single line, but only one direction at a time.

**Hardware** - The physical aspect of computers, telecommunications, and other information technology devices.

## IP Telephony System

**HTTP (HyperText Transport Protocol)** - The communications protocol used to connect to servers on the World Wide Web.

**HTTPS (HyperText Transport Protocol Secure)** - The communications protocol used to connect to a secure server on the World Wide Web.

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

**IP Address** - The address used to identify a computer or device on a network.

**IPCONFIG** - A Windows 2000 and XP utility that displays the IP address for a particular networking device.

**IPSec (Internet Protocol Security)** - A VPN protocol used to implement secure exchange of packets at the IP layer.

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

**LAN** - The computers and networking products that make up your local network.

**MAC (Media Access Control) Address** - The unique address that a manufacturer assigns to each networking device.

**Mbps (MegaBits Per Second)** - One million bits per second; a unit of measurement for data transmission.

**Multicasting** - Sending data to a group of destinations at once.

**NAT (Network Address Translation)** - NAT technology translates IP addresses of a local area network to a different IP address for the Internet.

**NAT (Network Address Translation) Traversal** - A method of enabling specialized applications, such as Internet phone calls, video, and audio, to travel between your local network and the Internet. STUN is a specific type of NAT traversal.

**Network** - A series of computers or devices connected for the purpose of data sharing, storage, and/or transmission between users.

**NNTP (Network News Transfer Protocol)** - The protocol used to connect to Usenet groups on the Internet.

**Node** - A network junction or connection point, typically a computer or work station.

**Packet** - A unit of data sent over a network.

**Ping (Packet INternet Groper)** - An Internet utility used to determine whether a particular IP address is online.

## IP Telephony System

**POP3 (Post Office Protocol 3)** - A standard mail server commonly used on the Internet.

**Port** - The connection point on a computer or networking device used for plugging in cables or adapters.

**Power over Ethernet (PoE)** - A technology enabling an Ethernet network cable to deliver both data and power.

**PPPoE (Point to Point Protocol over Ethernet)** - A type of broadband connection that provides authentication (username and password) in addition to data transport.

**PPTP (Point-to-Point Tunneling Protocol)** - A VPN protocol that allows the Point to Point Protocol (PPP) to be tunneled through an IP network. This protocol is also used as a type of broadband connection in Europe.

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

**RJ-45 (Registered Jack-45)** - An Ethernet connector that holds up to eight wires.

**Router** - A networking device that connects multiple networks together.

**RTP (Real-time Transport Protocol)** - A protocol that enables specialized applications, such as Internet phone calls, video, and audio, to occur in real time.

**RTS (Request To Send)** - A networking method of coordinating large packets through the RTS Threshold setting.

**Server** - Any computer whose function in a network is to provide user access to files, printing, communications, and other services.

**SMTP (Simple Mail Transfer Protocol)** - The standard e-mail protocol on the Internet.

**SNMP (Simple Network Management Protocol)** - A widely used network monitoring and control protocol.

**Software** - Instructions for the computer. A series of instructions that performs a particular task is called a "program".

**SOHO (Small Office/Home Office)** - Market segment of professionals who work at home or in small offices.

**SPI (Stateful Packet Inspection) Firewall** - A technology that inspects incoming packets of information before allowing them to enter the network.

**Static IP Address** - A fixed address assigned to a computer or device that is connected to a network.

**Static Routing** - Forwarding data in a network via a fixed path.

## IP Telephony System

**STUN (Simple Traversal of UDP through NATs)** - A protocol that enables specialized applications, such as Internet phone calls, video, and audio, to travel between your local network and the Internet. STUN is a specific type of NAT traversal.

**Subnet Mask** - An address code that determines the size of the network.

**Switch** - 1. A data switch that connects computing devices to host computers, allowing a large number of devices to share a limited number of ports. 2. A device for making, breaking, or changing the connections in an electrical circuit.

**TCP (Transmission Control Protocol)** - A network protocol for transmitting data that requires acknowledgement from the recipient of data sent.

**TCP/IP (Transmission Control Protocol/Internet Protocol)** - A set of instructions PCs use to communicate over a network.

**Telnet** - A user command and TCP/IP protocol used for accessing remote PCs.

**TFTP (Trivial File Transfer Protocol)** - A version of the TCP/IP FTP protocol that has no directory or password capability.

**Throughput** - The amount of data moved successfully from one node to another in a given time period.

**Topology** - The physical layout of a network.

**TX Rate** - Transmission Rate.

**UDP (User Datagram Protocol)** - A network protocol for transmitting data that does not require acknowledgement from the recipient of the data that is sent.

**Upgrade** - To replace existing software or firmware with a newer version.

**Upload** - To transmit a file over a network.

**URL (Uniform Resource Locator)** - The address of a file located on the Internet.

**VPN (Virtual Private Network)** - A security measure to protect data as it leaves one network and goes to another over the Internet.

**WAN (Wide Area Network)**- The Internet.

**WINIPCFG** - A Windows 98 and Me utility that displays the IP address for a particular networking device.

# Appendix H: Specifications

<b>Model</b>	<b>SPA9000</b>
<b>Data Networking</b>	<b>MAC Address (IEEE 802.3) IPv4 - Internet Protocol v4 (RFC 791) upgradeable to v6 (RFC 1883) ARP - Address Resolution Protocol NS - A Record (RFC 1706), SRV Record (RFC 2782) DHCP Client - Dynamic Host Configuration Protocol (RFC 2131) DHCP Server - Dynamic Host Configuration Protocol (RFC 2131) PPPoE Client - Point to Point Protocol over Ethernet (RFC 2516) ICMP - Internet Control Message Protocol (RFC792) TCP - Transmission Control Protocol (RFC793) UDP - User Datagram Protocol (RFC768) RTP - Real Time Protocol (RFC 1889) (RFC 1890) RTCP - Real Time Control Protocol (RFC 1889) DiffServ (RFC 2475), Type of Service - TOS (RFC 791/1349) VLAN Tagging - 802.1p/q SNTP - Simple Network Time Protocol (RFC 2030) Upload Data Rate Limiting - Static and Automatic QoS - Voice Packet Prioritization over Other Packet Types Router or Bridge Mode of Operation MAC Address Cloning Port Forwarding</b>
<b>Voice Gateway</b>	<b>SIPv2 - Session Initiation Protocol Version 2 (RFC 3261, 3262, 3263, 3264) SIP Proxy Redundancy - Dynamic via DNS SRV, A Records Re-registration with Primary SIP Proxy Server SIP Support in Network Address Translation Networks - NAT (incl. STUN) Secure (Encrypted) Calling via Pre-Standard Implementation of Secure RTP Codec Name Assignment</b>

## IP Telephony System

### Voice Algorithms:

G.711 (A-law and  $\mu$ -law)

G.726 (16/24/32/40 kbps)

G.729 A

G.723.1 (6.3 kbps, 5.3 kbps)

Dynamic Payload Support

Adjustable Audio Frames Per Packet

DTMF: In-band & Out-of-Band (RFC 2833) (SIP INFO)

Flexible Dial Plan Support with Inter-Digit Timers

IP Address / URI Dialing Support

Call Progress Tone Generation

Jitter Buffer - Adaptive

Frame Loss Concealment

VAD - Voice Activity Detection w/ Silence Suppression

Attenuation / Gain Adjustments

MWI - Message Waiting Indicator Tones

VMWI - Via NOTIFY, SUBSCRIBE

Caller ID Support (Name & Number)

### Security

Password Protected System Reset to Factory Default

Password Protected Admin and User Access Authority

HTTPS with Factory Installed Client Certificate

HTTP Digest - Encrypted Authentication via MD5 (RFC 1321)

Up to 256-bit AES Encryption

### Provisioning, Administration & Maintenance

Web Browser Administration & Configuration via Integral Web Server

Telephone Key Pad Configuration of Select Networking Parameters via IVR

Automated Provisioning & Upgrade via HTTPS, HTTP, TFTP

Asynchronous Notification of Upgrade Availability via NOTIFY

Non-intrusive, In-Service Upgrades

Report Generation & Event Logging

## IP Telephony System

Stats in BYE Message  
Syslog & Debug Server Records - Per Line Configurable

Physical Interfaces    2 10/100BaseT RJ-45 Ethernet Port (IEEE 802.3) — 1 WAN, 1 LAN  
2 RJ-11 FXS Phone Ports - For Analog Circuit Telephone Device (Tip/Ring)

### Subscriber Line

Interface Circuit (SLIC) Ring Voltage: 40-55 Vrms Configurable  
Ring Frequency: 10 Hz - 40 Hz  
Ring Waveform: Trapezoidal and Sinusoidal  
Maximum Ringer Load: 3 REN  
On-hook/off-hook Characteristics:  
    On-hook Voltage (tip/ring): -50 V NOMINAL  
    Off-hook Current: 25 mA min.  
    Terminating Impedance: 8 Configurable Settings including North America 600 ohms, European CTR21

### Regulatory Compliance

FCC (Part 15 Class B), CE Mark, A-Tick/C-Tick

### Power Supply

Switching Type (100-240v) Automatic  
DC Input Voltage: +5 VDC at 2.0 A max.  
Power Consumption: 5 Watts  
Power Adapter: 100-240v - 50-60Hz (26-34VA) AC Input, 1.8m Cord

Indicator Lights/LEDs    Power, Ethernet, Phone 1/2 LEDs

### Documentation

Quick Installation  
User Guide  
Administration Guide - Service Providers Only  
Provisioning Guide - Service Providers Only

**IP Telephony System**

**ENVIRONMENTAL**

<b>Device Dimensions</b>	<b>3.98" x 3.98" x 1.1"</b>
<b>W x H x D</b>	<b>101 mm x 101 mm x 28 mm</b>
<b>Weight</b>	<b>5.3 oz. (0.15 kg)</b>
<b>Operating Temp.</b>	<b>32 to 113° F (0 to 45° C)</b>
<b>Storage Temp.</b>	<b>-13 to 185° F (-25 to 85° C)</b>
<b>Operating Humidity</b>	<b>10 to 90% Non-Condensing, Operating and Non-Operating</b>

# Appendix I: Warranty Information

**Contact your service provider if the warranty support issues are to be handled by them as per your service agreements.**

## LIMITED WARRANTY

Linksys warrants to You that, for a period of one year (the “Warranty Period”), your Linksys Product will be substantially free of defects in materials and workmanship under normal use. Your exclusive remedy and Linksys' entire liability under this warranty will be for Linksys at its option to repair or replace the Product or refund Your purchase price less any rebates. This limited warranty extends only to the original purchaser.

If the Product proves defective during the Warranty Period call Linksys Technical Support in order to obtain a Return Authorization Number, if applicable. BE SURE TO HAVE YOUR PROOF OF PURCHASE ON HAND WHEN CALLING. If You are requested to return the Product, mark the Return Authorization Number clearly on the outside of the package and include a copy of your original proof of purchase. RETURN REQUESTS CANNOT BE PROCESSED WITHOUT PROOF OF PURCHASE. You are responsible for shipping defective Products to Linksys. Linksys pays for UPS Ground shipping from Linksys back to You only. Customers located outside of the United States of America and Canada are responsible for all shipping and handling charges.

ALL IMPLIED WARRANTIES AND CONDITIONS OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE ARE LIMITED TO THE DURATION OF THE WARRANTY PERIOD. ALL OTHER EXPRESS OR IMPLIED CONDITIONS, REPRESENTATIONS AND WARRANTIES, INCLUDING ANY IMPLIED WARRANTY OF NON-INFRINGEMENT, ARE DISCLAIMED. Some jurisdictions do not allow limitations on how long an implied warranty lasts, so the above limitation may not apply to You. This warranty gives You specific legal rights, and You may also have other rights which vary by jurisdiction.

This warranty does not apply if the Product (a) has been altered, except by Linksys, (b) has not been installed, operated, repaired, or maintained in accordance with instructions supplied by Linksys, or (c) has been subjected to abnormal physical or electrical stress, misuse, negligence, or accident. In addition, due to the continual development of new techniques for intruding upon and attacking networks, Linksys does not warrant that the Product will be free of vulnerability to intrusion or attack.

TO THE EXTENT NOT PROHIBITED BY LAW, IN NO EVENT WILL LINKSYS BE LIABLE FOR ANY LOST DATA, REVENUE OR PROFIT, OR FOR SPECIAL, INDIRECT, CONSEQUENTIAL, INCIDENTAL OR PUNITIVE DAMAGES, REGARDLESS OF THE THEORY OF LIABILITY (INCLUDING NEGLIGENCE), ARISING OUT OF OR RELATED TO THE USE OF OR INABILITY TO USE THE PRODUCT (INCLUDING ANY SOFTWARE), EVEN IF LINKSYS HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES. IN NO EVENT WILL LINKSYS' LIABILITY EXCEED THE AMOUNT PAID BY YOU FOR THE PRODUCT. The foregoing limitations will apply even if any warranty or remedy provided under this Agreement fails of its essential purpose. Some jurisdictions do not allow the exclusion or limitation of incidental or consequential damages, so the above limitation or exclusion may not apply to You.

Please direct all inquiries to: Linksys, P.O. Box 18558, Irvine, CA 92623.

# Appendix J: Regulatory Information

## FCC Statement

This product has been tested and complies with the specifications for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used according to the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which is found by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna
- Increase the separation between the equipment or devices
- Connect the equipment to an outlet other than the receiver's
- Consult a dealer or an experienced radio/TV technician for assistance

## Safety Notices

**Caution:** To reduce the risk of fire, use only No.26 AWG or larger telecommunication line cord.

Do not use this product near water, for example, in a wet basement or near a swimming pool.

Avoid using this product during an electrical storm. There may be a remote risk of electric shock from lightning.

## Industry Canada (Canada)

This device complies with Canadian ICES-003 rule.

Cet appareil est conforme à la norme NMB-003 d'Industry Canada.

**User Information for Consumer Products Covered by EU Directive 2002/96/EC on Waste Electric and Electronic Equipment (WEEE)**

This document contains important information for users with regards to the proper disposal and recycling of Linksys products. Consumers are required to comply with this notice for all electronic products bearing the following symbol:



**English**

**Environmental Information for Customers in the European Union**

European Directive 2002/96/EC requires that the equipment bearing this symbol on the product and/or its packaging must not be disposed of with unsorted municipal waste. The symbol indicates that this product should be disposed of separately from regular household waste streams. It is your responsibility to dispose of this and other electric and electronic equipment via designated collection facilities appointed by the government or local authorities. Correct disposal and recycling will help prevent potential negative consequences to the environment and human health. For more detailed information about the disposal of your old equipment, please contact your local authorities, waste disposal service, or the shop where you purchased the product.

**Čeština/Czech**

**Informace o ochraně životního prostředí pro zákazníky v zemích Evropské unie**

Evropská směrnice 2002/96/ES zakazuje, aby zařízení označené tímto symbolem na produktu anebo na obalu bylo likvidováno s netříděným komunálním odpadem. Tento symbol udává, že daný produkt musí být likvidován odděleně od běžného komunálního odpadu. Odpovídáte za likvidaci tohoto produktu a dalších elektrických a elektronických zařízení prostřednictvím určených sběrných míst stanovených vládou nebo místními úřady. Správná likvidace a recyklace pomáhá předcházet potenciálním negativním dopadům na životní prostředí a lidské zdraví. Podrobnější informace o likvidaci starého vybavení si laskavě vyžádejte od místních úřadů, podniku zabývajícího se likvidací komunálních odpadů nebo obchodu, kde jste produkt zakoupili.

## Dansk/Danish

### Miljøinformation for kunder i EU

EU-direktiv 2002/96/EF kræver, at udstyr der bærer dette symbol på produktet og/eller emballagen ikke må bortskaffes som usorteret kommunalt affald. Symbolet betyder, at dette produkt skal bortskaffes adskilt fra det almindelige husholdningsaffald. Det er dit ansvar at bortskaffe dette og andet elektrisk og elektronisk udstyr via bestemte indsamlingssteder udpeget af staten eller de lokale myndigheder. Korrekt bortskaffelse og genvinding vil hjælpe med til at undgå mulige skader for miljøet og menneskers sundhed. Kontakt venligst de lokale myndigheder, renovationstjenesten eller den butik, hvor du har købt produktet, angående mere detaljeret information om bortskaffelse af dit gamle udstyr.

## Deutsch/German

### Umweltinformation für Kunden innerhalb der Europäischen Union

Die Europäische Richtlinie 2002/96/EC verlangt, dass technische Ausrüstung, die direkt am Gerät und/oder an der Verpackung mit diesem Symbol versehen ist nicht zusammen mit unsortiertem Gemeindeabfall entsorgt werden darf. Das Symbol weist darauf hin, dass das Produkt von regulärem Haushaltsmüll getrennt entsorgt werden sollte. Es liegt in Ihrer Verantwortung, dieses Gerät und andere elektrische und elektronische Geräte über die dafür zuständigen und von der Regierung oder örtlichen Behörden dazu bestimmten Sammelstellen zu entsorgen. Ordnungsgemäßes Entsorgen und Recyceln trägt dazu bei, potentielle negative Folgen für Umwelt und die menschliche Gesundheit zu vermeiden. Wenn Sie weitere Informationen zur Entsorgung Ihrer Altgeräte benötigen, wenden Sie sich bitte an die örtlichen Behörden oder städtischen Entsorgungsdienste oder an den Händler, bei dem Sie das Produkt erworben haben.

## Eesti/Estonian

### Keskkonnaalane informatsioon Euroopa Liidus asuvatele klientidele

Euroopa Liidu direktiivi 2002/96/EÜ nõuete kohaselt on seadmeid, millel on tootel või pakendil käesolev sümbol, keelatud kõrvaldada koos sorteerimata olmejäätmetega. See sümbol näitab, et toode tuleks kõrvaldada eraldi tavalistest olmejäätmevoogudest. Olete kohustatud kõrvaldama käesoleva ja ka muud elektri- ja elektroonikaseadmed riigi või kohalike ametiasutuste poolt ette nähtud kogumispunktide kaudu. Seadmete korrektne kõrvaldamine ja ringlussevõtt aitab vältida võimalikke negatiivseid tagajärgi keskkonnale ning inimeste tervisele. Vanade seadmete kõrvaldamise kohta täpsema informatsiooni saamiseks võtke palun ühendust kohalike ametiasutustega, jäätmekäitlusfirmaga või kauplusega, kust te toote ostsite.

## Español/Spanish

### Información medioambiental para clientes de la Unión Europea

La Directiva 2002/96/CE de la UE exige que los equipos que lleven este símbolo en el propio aparato y/o en su embalaje no deben eliminarse junto con otros residuos urbanos no seleccionados. El símbolo indica que el producto en cuestión debe separarse de los residuos domésticos convencionales con vistas a su eliminación. Es responsabilidad suya desechar este y cualesquiera otros aparatos eléctricos y electrónicos a través de los puntos de recogida que ponen a su disposición el gobierno y las autoridades locales. Al desechar y reciclar correctamente estos aparatos estará contribuyendo a evitar posibles consecuencias negativas para el medio ambiente y la salud de las personas. Si desea obtener información más detallada sobre la eliminación segura de su aparato usado, consulte a las autoridades locales, al servicio de recogida y eliminación de residuos de su zona o pregunte en la tienda donde adquirió el producto.

## Ελληνικά/Greek

### Στοιχεία περιβαλλοντικής προστασίας για πελάτες εντός της Ευρωπαϊκής Ένωσης

Η Κοινοτική Οδηγία 2002/96/EC απαιτεί ότι ο εξοπλισμός ο οποίος φέρει αυτό το σύμβολο στο προϊόν και/ή στη συσκευασία του δεν πρέπει να απορρίπτεται μαζί με τα μικτά κοινотικά απορρίμματα. Το σύμβολο υποδεικνύει ότι αυτό το προϊόν θα πρέπει να απορρίπτεται ξεχωριστά από τα συνήθη οικιακά απορρίμματα. Είστε υπεύθυνος για την απόρριψη του παρόντος και άλλου ηλεκτρικού και ηλεκτρονικού εξοπλισμού μέσω των καθορισμένων εγκαταστάσεων συγκέντρωσης απορριμμάτων οι οποίες παρέχονται από το κράτος ή τις αρμόδιες τοπικές αρχές. Η σωστή απόρριψη και ανακύκλωση συμβάλλει στην πρόληψη πιθανών αρνητικών συνεπειών για το περιβάλλον και την υγεία. Για περισσότερες πληροφορίες σχετικά με την απόρριψη του παλιού σας εξοπλισμού, παρακαλώ επικοινωνήστε με τις τοπικές αρχές, τις υπηρεσίες απόρριψης ή το κατάστημα από το οποίο αγοράσατε το προϊόν.

## Français/French

### Informations environnementales pour les clients de l'Union européenne

La directive européenne 2002/96/CE exige que l'équipement sur lequel est apposé ce symbole sur le produit et/ou son emballage ne soit pas jeté avec les autres ordures ménagères. Ce symbole indique que le produit doit être éliminé dans un circuit distinct de celui pour les déchets des ménages. Il est de votre responsabilité de jeter ce matériel ainsi que tout autre matériel électrique ou électronique par les moyens de collecte indiqués par le gouvernement et les pouvoirs publics des collectivités territoriales. L'élimination et le recyclage en bonne et due forme ont pour but de lutter contre l'impact néfaste potentiel de ce type de produits sur l'environnement et la santé publique. Pour plus d'informations sur le mode d'élimination de votre ancien équipement, veuillez prendre contact avec les pouvoirs publics locaux, le service de traitement des déchets, ou l'endroit où vous avez acheté le produit.

## Italiano/Italian

### Informazioni relative all'ambiente per i clienti residenti nell'Unione Europea

La direttiva europea 2002/96/EC richiede che le apparecchiature contrassegnate con questo simbolo sul prodotto e/o sull'imballaggio non siano smaltite insieme ai rifiuti urbani non differenziati. Il simbolo indica che questo prodotto non deve essere smaltito insieme ai normali rifiuti domestici. È responsabilità del proprietario smaltire sia questi prodotti sia le altre apparecchiature elettriche ed elettroniche mediante le specifiche strutture di raccolta indicate dal governo o dagli enti pubblici locali. Il corretto smaltimento ed il riciclaggio aiuteranno a prevenire conseguenze potenzialmente negative per l'ambiente e per la salute dell'essere umano. Per ricevere informazioni più dettagliate circa lo smaltimento delle vecchie apparecchiature in Vostro possesso, Vi invitiamo a contattare gli enti pubblici di competenza, il servizio di smaltimento rifiuti o il negozio nel quale avete acquistato il prodotto.

## Latviešu valoda/Latvian

### Ekoloģiska informācija klientiem Eiropas Savienības jurisdikcijā

Direktīvā 2002/96/EK ir prasība, ka aprīkojumu, kam pievienota zīme uz paša izstrādājuma vai uz tā iesaiņojuma, nedrīkst izmest nešķīrotā veidā kopā ar komunālajiem atkritumiem (tiem, ko rada vietēji iedzīvotāji un uzņēmumi). Šī zīme nozīmē to, ka šī ierīce ir jāizmet atkritumos tā, lai tā nenonāktu kopā ar parastiem mājsaimniecības atkritumiem. Jūsu pienākums ir šo un citas elektriskās un elektroniskās ierīces izmest atkritumos, izmantojot īpašus atkritumu savākšanas veidus un līdzekļus, ko nodrošina valsts un pašvaldību iestādes. Ja izmešana atkritumos un pārstrāde tiek veikta pareizi, tad mazinās iespējamais kaitējums dabai un cilvēku veselībai. Sīkākas ziņas par novecojuša aprīkojuma izmešanu atkritumos jūs varat saņemt vietējā pašvaldībā, atkritumu savākšanas dienestā, kā arī veikalā, kur iegādājāties šo izstrādājumu.

## Lietuvškai/Lithuanian

### Aplinkosaugos informacija, skirta Europos Sąjungos vartotojams

Europos direktyva 2002/96/EC numato, kad įrangos, kuri ir (arba) kurios pakuotė yra pažymėta šiuo simboliu, negalima šalinti kartu su nerūšiuotomis komunalinėmis atliekomis. Šis simbolis rodo, kad gaminį reikia šalinti atskirai nuo bendro buitinių atliekų srauto. Jūs privalote užtikrinti, kad ši ir kita elektros ar elektroninė įranga būtų šalinama per tam tikras nacionalinės ar vietinės valdžios nustatytas atliekų rinkimo sistemas. Tinkamai šalinant ir perdurbant atliekas, bus išvengta galimos žalos aplinkai ir žmonių sveikatai. Daugiau informacijos apie jūsų senos įrangos šalinimą gali pateikti vietinės valdžios institucijos, atliekų šalinimo tarnybos arba parduotuvės, kuriose įsigijote tą gaminį.

## Malti/Maltese

### Informazzjoni Ambjentali għal Kliġenti fl-Unjoni Ewropea

Id-Direttiva Ewropea 2002/96/KE titlob li t-tagħmir li jkun fih is-simbolu fuq il-prodott u/jew fuq l-ippakkjar ma jstax jintrema ma' skart municiġpali li ma għiex iſseparat. Is-simbolu jindika li dan il-prodott għandu jintrema separatament minn ma' l-iskart domestiku regolari. Hija responsabbiltà tiegħek li tarmi dan it-tagħmir u kull tagħmir iehor ta' l-elettriku u elettroniku permezz ta' faċilitajiet ta' għbir appuntati apposta mill-gvern jew mill-awtoritajiet lokali. Ir-rimi b'mod korrett u r-riciklagg jgħin jipprevjeni konsegwenzi negattivi potenzjali għall-ambjent u għas-saħħa tal-bniedem. Għal aktar informazzjoni dettaljata dwar ir-rimi tat-tagħmir antik tiegħek, jekk jogħġbok ikkuntattja lill-awtoritajiet lokali tiegħek, is-servizzi għar-rimi ta' l-iskart, jew il-hanut minn fejn xtrajt il-prodott.

## Magyar/Hungarian

### Környezetvédelmi információ az európai uniós vásárlók számára

A 2002/96/EC számú európai uniós irányelv megkívánja, hogy azokat a termékeket, amelyekben, és/vagy amelyek csomagolásán az alábbi címke megjelenik, tilos a többi szelektálatlan lakossági hulladékkal együtt kidobni. A címke azt jelöli, hogy az adott termék kidobásakor a szokványos háztartási hulladékelszállítási rendszerektől elkülönített eljárást kell alkalmazni. Az Ön felelőssége, hogy ezt, és más elektromos és elektronikus berendezéseit a kormányzati vagy a helyi hatóságok által kijelölt gyűjtőrendszeren keresztül számolja fel. A megfelelő hulladékfeldolgozás segít a környezetre és az emberi egészségre potenciálisan ártalmas negatív hatások megelőzésében. Ha elavult berendezéseinek felszámolásához további részletes információra van szüksége, kérjük, lépjen kapcsolatba a helyi hatóságokkal, a hulladékfeldolgozási szolgálattal, vagy azzal üzlettel, ahol a terméket vásárolta.

## Nederlands/Dutch

### Milieu-informatie voor klanten in de Europese Unie

De Europese Richtlijn 2002/96/EC schrijft voor dat apparatuur die is voorzien van dit symbool op het product of de verpakking, niet mag worden ingezameld met niet-gescheiden huishoudelijk afval. Dit symbool geeft aan dat het product apart moet worden ingezameld. U bent zelf verantwoordelijk voor de vernietiging van deze en andere elektrische en elektronische apparatuur via de daarvoor door de landelijke of plaatselijke overheid aangewezen inzamelingskanalen. De juiste vernietiging en recycling van deze apparatuur voorkomt mogelijke negatieve gevolgen voor het milieu en de gezondheid. Voor meer informatie over het vernietigen van uw oude apparatuur neemt u contact op met de plaatselijke autoriteiten of afvalverwerkingsdienst, of met de winkel waar u het product hebt aangeschaft.

## Norsk/Norwegian

### Miljøinformasjon for kunder i EU

EU-direktiv 2002/96/EF krever at utstyr med følgende symbol avbildet på produktet og/eller pakningen, ikke må kastes sammen med usortert avfall. Symbolet indikerer at dette produktet skal håndteres atskilt fra ordinær avfallsinnsamling for husholdningsavfall. Det er ditt ansvar å kvitte deg med dette produktet og annet elektrisk og elektronisk avfall via egne innsamlingsordninger slik myndighetene eller kommunene bestemmer. Korrekt avfallshåndtering og gjenvinning vil være med på å forhindre mulige negative konsekvenser for miljø og helse. For nærmere informasjon om håndtering av det kasserte utstyret ditt, kan du ta kontakt med kommunen, en innsamlingsstasjon for avfall eller butikken der du kjøpte produktet.

## Polski/Polish

### Informacja dla klientów w Unii Europejskiej o przepisach dotyczących ochrony środowiska

Dyrektywa Europejska 2002/96/EC wymaga, aby sprzęt oznaczony symbolem znajdującym się na produkcie i/lub jego opakowaniu nie był wyrzucany razem z innymi niesortowanymi odpadami komunalnymi. Symbol ten wskazuje, że produkt nie powinien być usuwany razem ze zwykłymi odpadami z gospodarstw domowych. Na Państwu spoczywa obowiązek wyrzucania tego i innych urządzeń elektrycznych oraz elektronicznych w punktach odbioru wyznaczonych przez władze krajowe lub lokalne. Pozbywanie się sprzętu we właściwy sposób i jego recykling pomogą zapobiec potencjalnie negatywnym konsekwencjom dla środowiska i zdrowia ludzkiego. W celu uzyskania szczegółowych informacji o usuwaniu starego sprzętu, prosimy zwrócić się do lokalnych władz, służb oczyszczania miasta lub sklepu, w którym produkt został nabyty.

## Português/Portuguese

### Informação ambiental para clientes da União Europeia

A Directiva Europeia 2002/96/CE exige que o equipamento que exibe este símbolo no produto e/ou na sua embalagem não seja eliminado junto com os resíduos municipais não separados. O símbolo indica que este produto deve ser eliminado separadamente dos resíduos domésticos regulares. É da sua responsabilidade eliminar este e qualquer outro equipamento eléctrico e electrónico através dos instalações de recolha designadas pelas autoridades governamentais ou locais. A eliminação e reciclagem correctas ajudarão a prevenir as consequências negativas para o ambiente e para a saúde humana. Para obter informações mais detalhadas sobre a forma de eliminar o seu equipamento antigo, contacte as autoridades locais, os serviços de eliminação de resíduos ou o estabelecimento comercial onde adquiriu o produto.

## Slovenčina/Slovak

### Informácie o ochrane životného prostredia pre zákazníkov v Európskej únii

Podľa európskej smernice 2002/96/ES zariadenie s týmto symbolom na produkte a/alebo jeho balení nesmie byť likvidované spolu s netriedeným komunálnym odpadom. Symbol znamená, že produkt by sa mal likvidovať oddelene od bežného odpadu z domácností. Je vašou povinnosťou likvidovať toto i ostatné elektrické a elektronické zariadenia prostredníctvom špecializovaných zberných zariadení určených vládou alebo miestnymi orgánmi. Správna likvidácia a recyklácia pomôže zabrániť prípadným negatívnym dopadom na životné prostredie a zdravie ľudí. Ak máte záujem o podrobnejšie informácie o likvidácii starého zariadenia, obráťte sa, prosím, na miestne orgány, organizácie zaoberajúce sa likvidáciou odpadov alebo obchod, v ktorom ste si produkt zakúpili.

## Slovenčina/Slovene

### Okoljske informacije za stranke v Evropski uniji

Evropska direktiva 2002/96/EC prepoveduje odlaganje opreme, označene s tem simbolom – na izdelku in/ali na embalaži – med običajne, nerazvrščene odpadke. Ta simbol opozarja, da je treba izdelek odvreči ločeno od preostalih gospodinjstvih odpadkov. Vaša odgovornost je, da to in preostalo električno in elektronsko opremo odnesete na posebna zbirališča, ki jih določijo državne ustanove ali lokalna uprava. S pravilnim odlaganjem in recikliranjem boste preprečili morebitne škodljive vplive na okolje in zdravje ljudi. Če želite izvedeti več o odlaganju stare opreme, se obrnite na lokalno upravo, odpad ali trgovino, kjer ste izdelek kupili.

**Suomi/Finnish**

**Ympäristöä koskevia tietoja EU-alueen asiakkaille**

EU-direktiivi 2002/96/EY edellyttää, että jos laitteistossa on tämä symboli itse tuotteessa ja/tai sen pakkauksessa, laitteistoa ei saa hävittää lajittelemattoman yhdyskuntajätteen mukana. Symboli merkitsee sitä, että tämä tuote on hävitettävä erillään tavallisesta kotitalousjätteestä. Sinun vastuullasi on hävittää tämä elektroniikkatuote ja muut vastaavat elektroniikkatuotteet viemällä tuote tai tuotteet viranomaisten määräämään keräyspisteeseen. Laitteiston oikea hävittäminen estää mahdolliset kielteiset vaikutukset ympäristöön ja ihmisten terveyteen. Lisätietoja vanhan laitteiston oikeasta hävitystavasta saa paikallisilta viranomaisilta, jätteenhävityspalvelusta tai siitä myymälästä, josta ostit tuotteen.

**Svenska/Swedish**

**Miljöinformation för kunder i Europeiska unionen**

Det europeiska direktivet 2002/96/EC kräver att utrustning med denna symbol på produkten och/eller förpackningen inte får kastas med osorterat kommunalt avfall. Symbolen visar att denna produkt bör kastas efter att den avskiljts från vanligt hushållsavfall. Det faller på ditt ansvar att kasta denna och annan elektrisk och elektronisk utrustning på fastställda insamlingsplatser utsedda av regeringen eller lokala myndigheter. Korrekt kassering och återvinning skyddar mot eventuella negativa konsekvenser för miljön och personhälsa. För mer detaljerad information om kassering av din gamla utrustning kontaktar du dina lokala myndigheter, avfallshanteringen eller butiken där du köpte produkten.

**For more information, visit [www.linksys.com](http://www.linksys.com).**

# Appendix K: Contact Information

## Internet Telephony Service Provider (ITSP)

For support, contact your ITSP.

### Linksys

Visit us online for information on the latest products and updates to your existing products at:

<http://www.linksys.com> or  
[www.linksys.com/support](http://www.linksys.com/support)