BLACK BOX
NETWORK SERVICES

VOE101

VoIP Analog Gateway

Getting Started Guide

CE

Important

This is a Class B device and is intended for use in a light industrial or residential environment. It is not intended nor approved for use in an industrial environment.

CUSTOMER SUPPORT INFORMATION Order **toll-free** in the U.S. 24 hours, 7 A.M. Monday to midnight Friday: **877-877-BBOX**
FREE technical support, 24 hours a day, 7 days a week: Call **724-746-5500** or fax **724-746-0746**
Mail order: **Black Box Corporation**, 1000 Park Drive, Lawrence, PA 15055-1018
Web site: www.blackbox.com • E-mail: info@blackbox.com

Radio and TV Interference

This equipment generates and uses radio frequency energy, and if not installed and used properly—that is, in strict accordance with the manufacturer's instructions—may cause interference to radio and television reception. This equipment has been tested and found to comply with the limits for a Class A computing device in accordance with the specifications in Subpart B of Part 15 of FCC rules, which are designed to provide reasonable protection from such interference in a commercial installation. However, there is no guarantee that interference will not occur in a particular installation. If the equipment causes interference to radio or television reception, which can be determined by disconnecting the cables, try to correct the interference by one or more of the following measures: moving the computing equipment away from the receiver, re-orienting the receiving antenna, and/or plugging the receiving equipment into a different AC outlet (such that the computing equipment and receiver are on different branches).

Industry Canada Notice

This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

This Declaration of Conformity means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction. Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. In some cases, the company's inside wiring associated with a single line individual service may be extended by means of a certified connector assembly (telephone extension cord). The customer should be aware that compliance with the above condition may not prevent degradation of service in some situations. Repairs to some certified equipment should be made by an authorized maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment. Users should ensure for their own protection that the ground connections of the power utility, telephone lines and internal metallic water pipe system, are connected together. This protection may be particularly important in rural areas.

FCC Compliance

EMC

- FCC Part 15, Class B
- EN55022, Class B
- EN55024

Safety

- EN60950

FCC Part 68 Compliance Statement

This equipment complies with Part 68 of FCC rules and the requirements adopted by ACTA. On the bottom side of this equipment is a label that contains—among other information—a product identifier in the format *US: AAAEQ##TXXXX*. If requested, this number must be provided to the telephone company.

The method used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA.

If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If trouble is experienced with this equipment, for repair or warranty information, please contact our company. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

CE Notice

We certify that the apparatus identified in this document conforms to the requirements of Council Directive 1999/5/EC on the approximation of the laws of the member states relating to Radio and Telecommunication Terminal Equipment and the mutual recognition of their conformity.

The safety advice in the documentation accompanying this product shall be obeyed. The conformity to the above directive is indicated by the **CE** sign on the device.

Trademarks Used In This Manual

All applied-for and registered trademarks are the property of their respective owners.

Normas Oficiales Mexicanas (NOM) Electrical Safety Statement

Instrucciones De Seguridad

1. Todas las instrucciones de seguridad y operación deberán ser leídas antes de que el aparato eléctrico sea operado.
2. Las instrucciones de seguridad y operación deberán ser guardadas para referencia futura.
3. Todas las advertencias en el aparato eléctrico y en sus instrucciones de operación deben ser respetadas.
4. Todas las instrucciones de operación y uso deben ser seguidas.
5. El aparato eléctrico no deberá ser usado cerca del agua—por ejemplo, cerca de la tina de baño, lavabo, sótano mojado o cerca de una alberca, etc.

6. El aparato eléctrico debe ser usado únicamente con carritos o pedestales que sean recomendados por el fabricante.

7. El aparato eléctrico debe ser montado a la pared o al techo sólo como sea recomendado por el fabricante.

8. Servicio—El usuario no debe intentar dar servicio al equipo eléctrico más allá a lo descrito en las instrucciones de operación. Todo otro servicio deberá ser referido a personal de servicio calificado.

9. El aparato eléctrico debe ser situado de tal manera que su posición no interfiera su uso. La colocación del aparato eléctrico sobre una cama, sofá, alfombra o superficie similar puede bloquea la ventilación, no se debe colocar en libreros o gabinetes que impidan el flujo de aire por los orificios de ventilación.

10. El equipo eléctrico deber ser situado fuera del alcance de fuentes de calor como radiadores, registros de calor, estufas u otros aparatos (incluyendo amplificadores) que producen calor.

11. El aparato eléctrico deberá ser connectado a una fuente de poder sólo del tipo descrito en el instructivo de operación, o como se indique en el aparato.

12. Precaución debe ser tomada de tal manera que la tierra fisica y la polarización del equipo no sea eliminada.

13. Los cables de la fuente de poder deben ser guiados de tal manera que no sean pisados ni pellizcados por objetos colocados sobre o contra ellos, poniendo particular atención a los contactos y receptáculos donde salen del aparato.

14. El equipo eléctrico debe ser limpiado únicamente de acuerdo a las recomendaciones del fabricante.

15. En caso de existir, una antena externa deberá ser localizada lejos de las lineas de energia.

16. El cable de corriente deberá ser desconectado del cuando el equipo no sea usado por un largo periodo de tiempo.

17. Cuidado debe ser tomado de tal manera que objectos liquidos no sean derramados sobre la cubierta u orificios de ventilación.

18. Servicio por personal calificado deberá ser provisto cuando:

   — A: El cable de poder o el contacto ha sido dañado; o

   — B: Objectos han caído o líquido ha sido derramado dentro del aparato; o

   — C: El aparato ha sido expuesto a la lluvia; o

   — D: El aparato parece no operar normalmente o muestra un cambio en su desempeño; o

   — E: El aparato ha sido tirado o su cubierta ha sido dañada.

Summary Table of Contents

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About this guide

This guide describes using the VOE101 VoIP Analog Gateway.

Audience

This guide is intended for the following users:

- VoIP telephony service providers
- Enterprise telecom and IT technicians

IMPORTANT

> Consumers of VoIP telephony service providers and employees of enterprises that received the Black Box VOE101 in conjunction with VoIP services are encouraged to contact their provider before making any configuration changes. Improper configuration may lead to a loss of services.

Structure

This guide contains the following chapters and appendices:

- Chapter 1 on page 19 provides information about VOE101 features and capabilities
- Chapter 2 on page 21 provides hardware installation procedures
- Chapter 3 on page 24 describes the Home section settings
- Chapter 4 on page 27 describes the LAN section settings
- Chapter 5 on page 34 describes the Telephony section settings
- Chapter 6 on page 50 describes the System section settings
- Chapter 7 on page 69 describes how to download and display the VOE101 user guide
- Chapter 8 on page 71 describes how to log out of the VOE101 management system
- Appendix A on page 73 contains compliance information for the VOE101
- Appendix B on page 75 contains specifications for the VOE101
- Appendix C on page 78 describes dialing plans and contains sample plans
- Appendix D on page 81 describes the calling features that can be accessed from phones attached to the VOE101
- Appendix E on page 83 explains how to configure the voice prompt function in the VOE101

For best results, read the contents of this guide *before* you install the VOE101.

Precautions

Notes, cautions, and warnings, which have the following meanings, are used throughout this guide to help you become aware of potential problems. ***Warnings*** are intended to prevent safety hazards that could result in personal injury. ***Cautions*** are intended to prevent situations that could result in property damage or impaired functioning.

Note A note presents additional information or interesting sidelights.

IMPORTANT

The alert symbol and IMPORTANT heading calls attention to important information.

CAUTION

The alert symbol and CAUTION heading indicate a potential hazard. Strictly follow the instructions to avoid property damage.

CAUTION

The shock hazard symbol and CAUTION heading indicate a potential electric shock hazard. Strictly follow the instructions to avoid property damage caused by electric shock.

WARNING

The alert symbol and WARNING heading indicate a potential safety hazard. Strictly follow the warning instructions to avoid personal injury.

WARNING

The shock hazard symbol and WARNING heading indicate a potential electric shock hazard. Strictly follow the warning instructions to avoid injury caused by electric shock.

Safety when working with electricity

WARNING

The VOE101 contains no user serviceable parts. The equipment shall be returned to Black Box for repairs, or repaired by qualified service personnel.

WARNING

The external power adapter shall be a listed Limited Power Source. Ensure that the power cable used with this devise meets all applicable standards for the country in which it is to be installed, and that it is connected to a wall outlet which has earth ground. The mains outlet that is utilized to power the devise shall be within 10 feet (3 meters) of the device, shall be easily accessible, and protected by a circuit breaker.

WARNING

Do not work on the system or connect or disconnect cables during periods of lightning activity.

In accordance with the requirements of council directive 2002/96/EC on Waste of Electrical and Electronic Equipment (WEEE), ensure that at end-of-life you separate this product from other waste and scrap and deliver to the WEEE collection system in your country for recycling.

General observations

- Clean the case with a soft slightly moist anti-static cloth
- Place the unit on a flat surface and ensure free air circulation
- Avoid exposing the unit to direct sunlight and other heat sources
- Protect the unit from moisture, vapors, and corrosive liquids

Typographical conventions used in this document

This section describes the typographical conventions and terms used in this guide.

General conventions

The procedures described in this manual use the following text conventions:

Table 1. General conventions

Convention	Meaning
Garamond blue type	Indicates a cross-reference hyperlink that points to a figure, graphic, table, or section heading. Clicking on the hyperlink jumps you to the reference. When you have finished reviewing the reference, click on the **Go to Previous View** button in the Adobe® Acrobat® Reader toolbar to return to your starting point.
Garamond bold type	Indicates the names of command buttons that execute an action.
< >	Angle brackets indicate function and keyboard keys, such as <SHIFT>, <CTRL>, <C>, and so on.

Chapter 1 **General information**

Chapter contents

VOE101 overview

The VOE101 VoIP Analog Gateway (see figure 1) provides transparent connectivity for analog phones and faxes to the world of Internet voice. Connecting to any analog phone, fax or PBX, the VOE101 product is an effective and flexible solution for accessing Internet-based telephone services and corporate intranet systems across established LAN and Internet connections like xDSL and cable modems.

Figure 1. VOE101

The VOE101 provides one RJ-45 Ethernet port and one FXS (RJ-11) analog phone port. Side panel LEDs quickly show at-a-glance the status of the system, WAN and phone ports.

A full suite of IP features (DHCP and NAT/PAT) are available on the VOE101 to provide easy interconnection to VoIP services and IP transport networks. VLAN tagging and prioritization enables voice traffic to be handled before data traffic by other devices on the network.

The web interface offers two levels of configuration access for the network operator and end user. The friendly web interface and product labeling (Phone, LAN, and WAN) to help ensure a trouble-free installation for the end user. Configuration and firmware can be downloaded from a TFTP server or HTTP server.

Chapter 2 **VOE101 installation**

Chapter contents

Installing the VOE101

1. The unit should be installed in a dry environment with at least 2 inches (5 cm) of clearance at the sides, front, and rear of the unit to allow air circulation for cooling.
2. Plug in the telephone (see figure 2).
3. Plug in the PC or LAN, or a LAN hub/switch.
4. Plug the power adapter into the power jack on the VOE101 (see figure 2). Connect the other end of the power cord to an appropriate AC power outlet.
5. Wait 30 seconds after powering the VOE101 on, then verify that the green *Power* LED is lit (see figure 2).
6. By default, the VOE101 will automatically request IP network settings from the LAN using DHCP. To determine the IP address of the VOE101, lift the handset off the attached analog phone and dial * * * *.

   Dial **1 0 0** #, listen to and record the IP address of the VOE101. (To manually set the IP address, see appendix E, "Voice prompt configuration" on page 83 for details).
7. Use a web browser to connect to the VOE101. The URL will be http://<ip address>. For example, if the VOE101 IP address was *10.10.10.2*, the URL would be *http://10.10.10.2*.

Note The default password for the VOE101 is "*root*".

IMPORTANT

Follow the directions of your voice service provider to set up voice services.

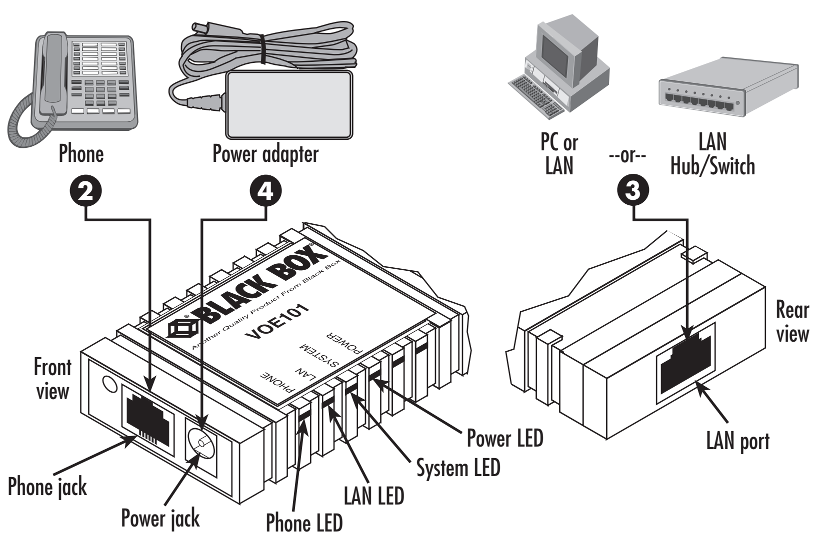


Figure 2. VOE101 installation diagram

Resetting the VOE101 to factory default

1. The VOE101 should be powered off.
2. Press the recessed button on the rear of the VOE101, then apply power to the unit.
3. Continue depressing the button while the LEDS are flashing, and only release the button after the LEDs have stopped.
4. You may now log in to the VOE101 using the password “root”.

Chapter 3 **Home**

Chapter contents

System information

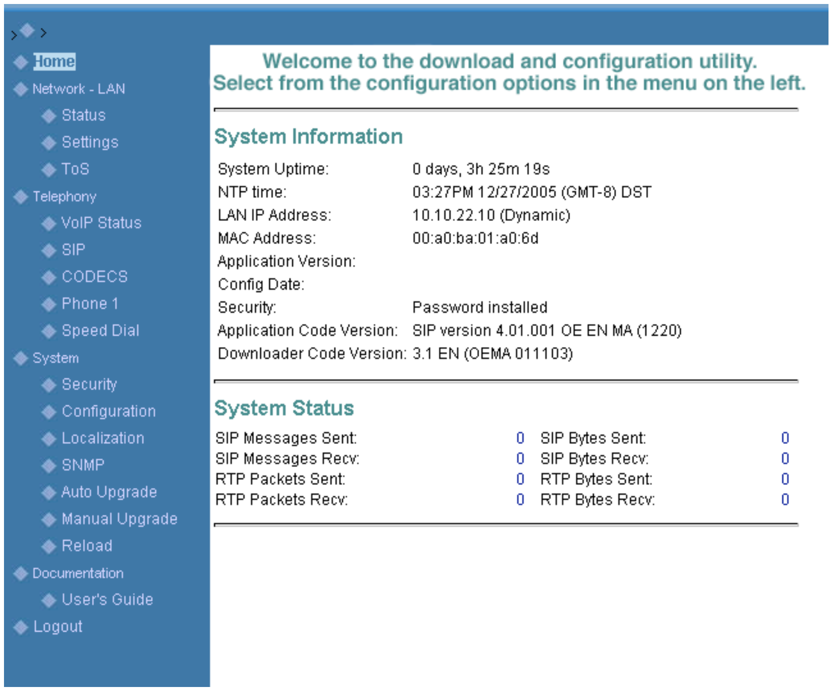


Figure 3. VOE101 VoIP download and configuration Home page

System Uptime

Shows how long the VOE101 has been operating since the last time it was reloaded (either by powering the unit off and then on again, or by selecting *System* > *Reload*).

LAN IP Address

The IP address of the VOE101. If *(Static)* is shown next to the address, it means the IP address was assigned to the VOE101 under the LAN Settings. (Dynamic) indicates the IP address was learned via DHCP.

MAC address

The media access control (MAC) address of the Ethernet interface in the VOE101.

Application Version

The VOE101 firmware version number.

Config Date

The date of the configuration file that was downloaded from an auto-upgrade server.

Security

Indicates that the VOE101 web interface utility has been secured with a password. To configure a password, see section “Set Security Password” on page 53.

Application Code Version
Shows the application code version being used.

Downloader Code Version
Shows the downloader code version being used.

System Status

Shows VoIP statistics for the period of time since the VOE101 was last reloaded (either by powering the unit off and then on again, or by selecting *System* > *Reload*).

SIP Messages Sent
Total number of VoIP SIP messages sent (including retransmissions).

SIP Messages Received
Total number of VoIP SIP messages received (including retransmissions).

SIP Bytes Sent
Total number of bytes of VoIP SIP messages sent (including retransmissions).

SIP Bytes Received
Total number of bytes of VoIP SIP messages received (including retransmissions).

RTP Packets Sent
Total number of VoIP RTP packets sent (including redundant packets).

RTP Packets Received
Total number of VoIP RTP packets received (including redundant packets).

RTP Bytes Sent
Total number of VoIP RTP bytes sent.

RTP Bytes Received
Total number of VoIP RTP bytes received.

Chapter 4 Network—LAN

Chapter contents

Status

Shows the status of key LAN network settings as configured under *LAN settings*

Interface Status

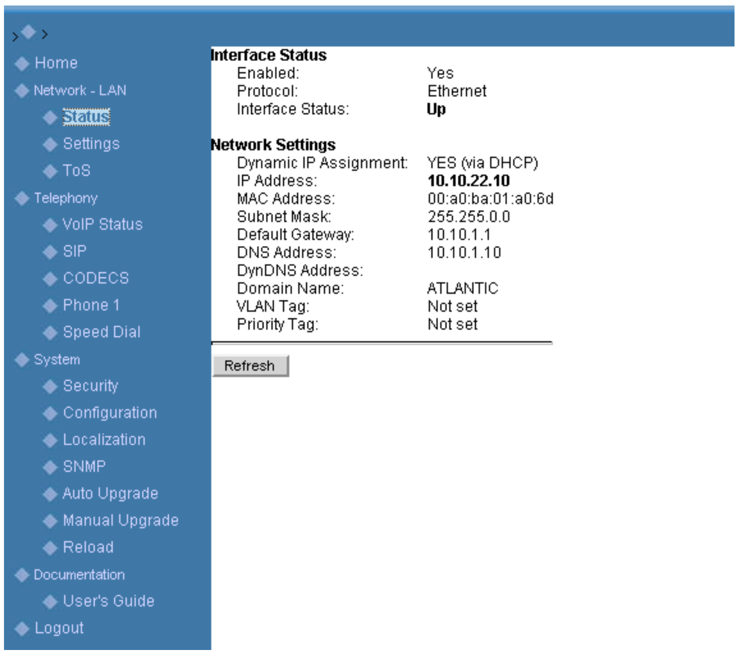


Figure 4. Internet Status window

Enabled

Yes indicates the LAN interface is enabled and ready to be used.

Service

Either *Routed* or *Bridged*, displays whether the VOE101's LAN interface connection is operating in a routed or bridged mode.

Interface Status

Either *Up* or *Down*.

Network Settings

These are the details of your LAN network settings.

Dynamic IP Assignment

Displays *Yes (via DHCP)* if you are using a dynamic IP address or *No* if a dynamic IP address is not being used.

IP address
The IP address of the VOE101 on the WAN interface.

MAC address
The MAC address of the WAN Ethernet interface in the VOE101.

Subnet Mask
The subnet mask is 32-bit number that filters a destination IP address to determine to which subnet it belongs. For example, a subnet mask of *255.255.0.0* for a network ID of *192.5.0.0* tells the switch to accept traffic destined for IP addresses that begin with *192.5*—all other packets are ignored.

Default Gateway
The IP address of the gateway. The gateway IP address can be retrieved automatically in DHCP mode or be set up manually with a fixed IP address.

Domain name
The network domain name of the VOE101.

DNS address
Refers to the address of your domain name server that was defined under LAN settings or that was learned dynamically through DHCP.

DynDNS address
The IP address of the dynamic DNS server that will be notified when the VOE101's dynamic IP address changes.

VLAN
VLAN tag value encoded in the LAN Ethernet header in all outgoing packets

Priority Tag
Priority tag value encoded in the LAN Ethernet header in outgoing packets.

Settings

The *Settings* window contains the following sections:

- Internet Configuration (see figure 5 on page 30)
- WAN PPPoE Configuration (see figure 6 on page 31)
- MAC Spoofing Configuration (see figure 8 on page 32)
- Internet VLAN Configuration (see figure 8 on page 32)

Note After configuring the sections, click the **Save Internet Settings** button (see figure 4 on page 28) to save the new configuration.

Internet Configuration

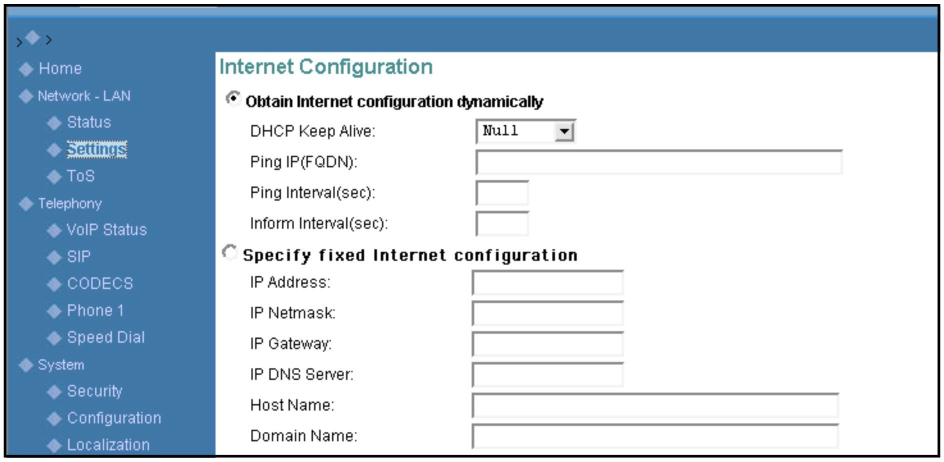


Figure 5. Internet Configuration section of the Settings window

Obtain LAN configuration dynamically

Select this option if appropriate. If you choose *Obtain LAN configuration dynamically*, the information is detected automatically through DHCP.

Specify fixed LAN configuration

Select this option if you will not be using DHCP. If you choose *Specify fixed LAN configuration*, you will have to enter the following information:

- IP address.
- IP of the netmask.
- IP of the gateway.
- IP of the DNS Server, if applicable.
- Host name (the name will identify the computer on the Internet, such as *VOE101.blackbox.com*).
- Domain name (the name that will identify one or more IP addresses). For example, the *blackbox.com* domain is used by Black Box. That domain can include multiple hostnames (such as **VOE101**.blackbox.com, **ftp**.blackbox.com, and so on) that point to individual computers on the Black Box network. In short, for the hostname URL *http://www.blackbox.com*, the domain name is *blackbox.com*.

WAN PPPoE Configuration

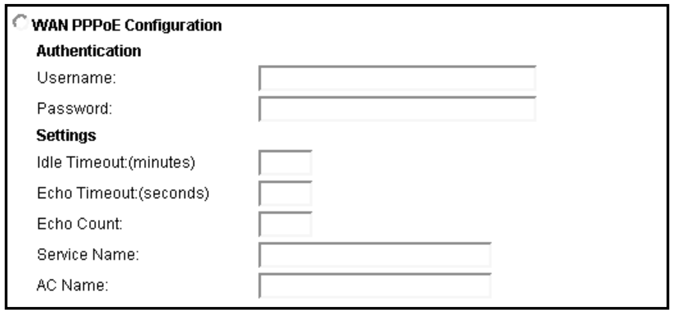


Figure 6. WAN PPPoE Configuration section of the Settings window

Enable PPPoE

Select *Yes* to enable PPPoE or *No* to disable PPPoE.

Authentication

Enter the username and password provided by your ISP.

Settings

Idle Timeout. Idle timeout before PPP connection is closed due to inactivity

Echo Timeout. The duration between sending PPP echo requests to server.

Echo Count. The number of unanswered PPP echo requests before the PPP connection is closed.

Service Name. PPPoE Service name

AC Name. PPPoE access concentrator (AC) name

Dynamic DNS

The VOE101 supports Dynamic DNS for use in environments where the IP address of LAN IP interface is not assigned statically (i.e. permanently) but instead is assigned dynamically using protocols like DHCP.

With a statically assigned IP addresses, DNS is used to establish a fixed relationship between an IP address a DNS name (example: 209.22.110.3 = joephone@blackbox.com). The DNS name allows the IP device, like a phone attached to an VOE101, to be found by the DNS name (joephone@blackbox.com) anywhere on the Internet.

Prior to Dynamic DNS, dynamically assigned IP address changes could not be automatically reflected in DNS. As a result, VoIP devices like the VOE101 could not always be located on the Internet by DNS name. This is especially true for cable modem and ADSL services that use DHCP to assign addresses to customers.

With Dynamic DNS, the VOE101 will inform the dynamic DNS server of its IP address when it receives a new or dynamically assigned IP address from the network. This allows the VOE101 to always be found using a fixed DNS name when the IP address changes. With Dynamic DNS cable modem and ADSL users can be found on the Internet using a DNS name.

The DNS server used for registration is operated by Dynamic Network Services, Inc., Dynamic DNS (DynDNS). You can find detailed information about the company and the services it offers on the webpage **www.dyndns.org**. The company offers different levels of service. The basic services are offered free of charge, while the more advanced services are fee-based.

Figure 7. Dynamic DNS Service window

Configuring Dynamic DNS

1. Select *System* > *Configuration*.
2. **Choose Sever**: To enable DynamicDNS, select the name of the DYNDNS service from the drop-down menu (see figure 7).
3. **Host Name**: The host name is the name of the VOE101 as registered on the DYNDNS service.
4. **Username**: The user name is the user name as registered on the Dynamic DNS service.
5. **Password**: The password is the password as registered on the Dynamic DNS service.

MAC Spoofing Configuration

Figure 8. MAC Spoofing Configuration section of the Settings window

WAN MAC Address (Spoofed)

Only available when the unit is using the VOE101 mode. The spoofed MAC address to be used by the device's LAN interfaces, the Ethernet address of the outgoing packets from the LAN interface would be replaced with this address. If blank, the LAN interfaces will use the hardware value of MAC

VLAN Configuration

The VOE101 can mark outgoing Ethernet frames on the LAN interface with VLAN and priority tags. Other devices on the LAN can use the tags to control how frames from the VOE101 are processed. All data leaving the VOE101 will be marked with the specific value unless overridden on the telephony VLAN configuration settings.

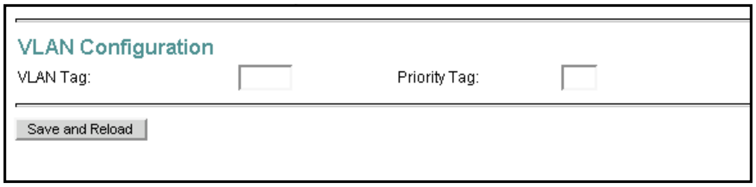


Figure 9. VLAN Configuration section of the Settings window

VLAN Tag (IEEE 802.1q)

IEEE 802.1q Ethernet VLAN tag for all outgoing packets on LAN Ethernet interface. The value should be between 0 and 4094.

Priority Tag (IEEE 802.1q)

IEEE 802.1q Ethernet Priority tag for all outgoing packets on LAN Ethernet interface. The value should be between 0 and 7.

Saving your work

When you are finished configuring the VLAN settings, click the **Save and Reload** button (see figure 9) to save all changes.

ToS/DiffServ

This sub-page is used to configure the Type-of-Service/Diffserv byte values which are to be used in the IP header of all transmitted SIP signaling packets and RTP packets. The ToS/DiffServ byte values are entered as two-digit hexadecimal values. If no special ToS/DiffServ value is to be used for a particular traffic type, enter **00** or leave the setting empty.

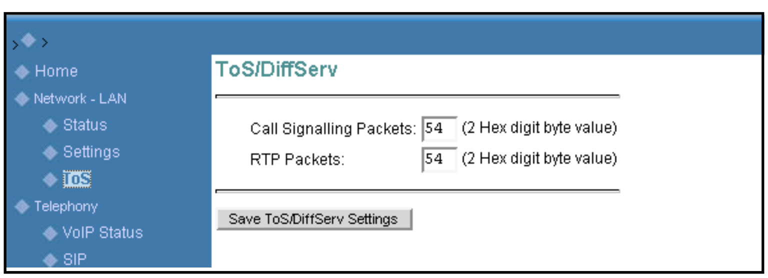


Figure 10. TOS/DiffServ window

Saving your work

When you are finished configuring ToS/DiffServ settings, click the **Save ToS/DiffServ Settings** button to save the changes.

Chapter 5 **Telephony**

Chapter contents

VoIP Status

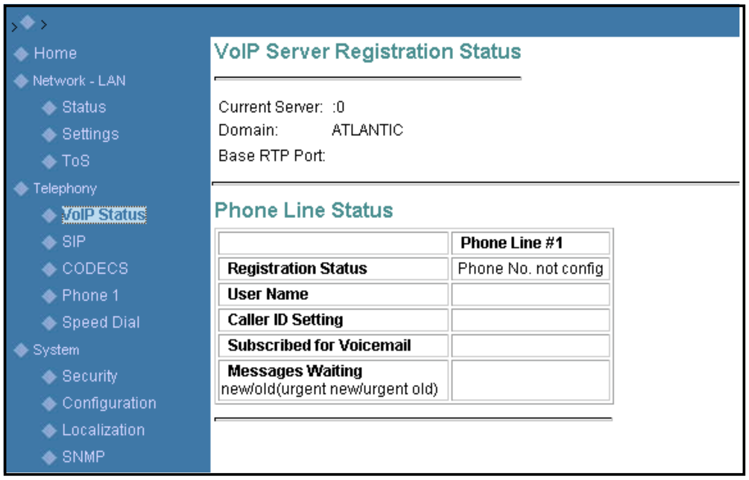


Figure 11. VoIP Status window

VoIP Server Registration Status

Current Server

Shows the current VoIP server that has been pre-defined or has been discovered using DNS service records (DNS-SRV).

Domain

The VoIP domain name is the domain name that is hosting the VoIP server.

Base RTP Port

Displays the base RTP port number for the RTP-RTCP port pair.

Phone Line Status

The *Phone Line Status* table shows the current operational status of the phone. It displays the VoIP registration status, the configured VoIP username, if the username is subscribed to voicemail and has messages waiting, and the caller ID setting. The messages waiting will indicate the number of new messages, old messages, new urgent messages and old urgent messages.

Registration Status

Shows the currently operational status of the phone. The following registration status values may be displayed:

- **Phone No. not config**—The phone number has not yet been configured for the phone line. Go to *Phone 1* to enter the phone configuration.
- **Online**—The phone line is online, registered with the SIP server and ready to send and receive phone calls using the SIP server.
- **Registering**—The VOE101 is in the process of registering with the VoIP server.
- **No Registration**—The VOE101 is not attempting to register with the SIP server because “Send Registration Request” is not checked on SIP sever setting screen.

- **Error: Not Authorized**—A SIP final status message status of 401 or 407 has been received from the SIP server.
- **Error: Forbidden**—SIP final status message status of 401 or 407 has been received from the SIP server.
- **Error 408: Request timeout**— A SIP final status message status of 408 has been received from the SIP server.
- **Response:** ***xxx***—A SIP final status message status of *xxx* has been received from the SIP server.

User Name

The authentication username that was specified under section "Phone 1" on page 44. This user name will be used to register with the VoIP server.

Caller ID Setting

The Caller ID name setting that was specified under section "Phone 1" on page 44. This is the display name that others will see as your caller id when you make a call.

Subscribed for Voicemail

Shows if the SIP Subscribe message was successful for this phone line. If "Yes", then the subscription process was successful. The number of voicemail messages in the queue will not be provided if the subscription for voice mail fails.

Messages waiting

After successful subscription to the SIP server, the SIP server/Internet Telephony service provider should send the number of voicemail messages that are queued up for the user using the SIP Notify message. The messages waiting shows the number of normal priority new messages, normal priority old (listened to) messages, number of urgent priority new voicemail messages and the number of old (listened to) urgent priority voicemail messages as reported to the VOE101 by the VoIP SIP server.

SIP

The *SIP* window contains the following sections:

- SIP Configuration (see figure 12 on page 39)
- SIP Extensions (see figure 14 on page 40)
- RTP Telephone Event Configuration (see figure 15 on page 40)
- VoIP VLAN Configuration (see figure 16 on page 41)
- SIP Parameters (see figure 17 on page 41)
- NAT Traversal (see figure 18 on page 42)

Note After configuring the sections, click the **Save SIP Settings** button (see figure 18 on page 42) to save the new configuration.

SIP Configuration

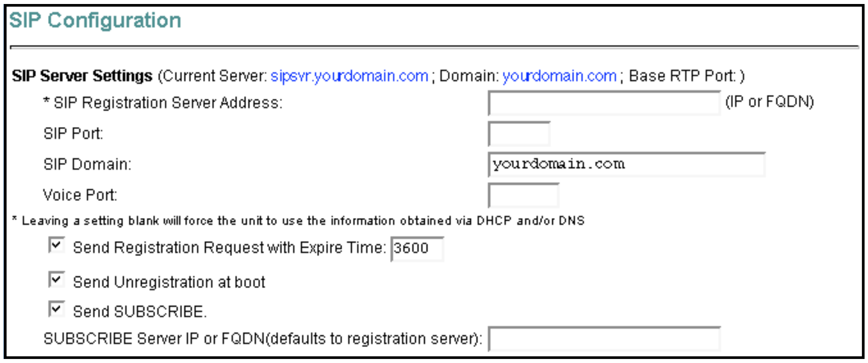


Figure 12. SIP Configuration section of the SIP window

SIP Server Settings

The VOE101 will automatically attempt to locate the VoIP server by using the domain name specified in the LAN interface or the server will be discovered via DHCP on the WAN interface. When found, the discovered server will be listed as the *Current Server*.

Enter the following information:

- Server address—The IP address or domain name hosting the VoIP SIP server
- Port—The UDP port of the VoIP SIP server. The default is *5060*.
- Domain name—The VoIP domain name (realm) is used for validation of each phone's username
- Send Registration Request with Expire Time—If selected, determines the amount of time (in seconds) that the SIP registration will be valid for.
- Unregistration—If checked, the VOE101 will send a SIP unregister at system reload before sending a SIP registration request.
- Send SUBSCRIBE—If checked, the VOE101 will send a SIP *subscribe* to the server specified. This box must be checked for the voicemail message counted and message waiting notification to work.
- SUBSCRIBE Server IP or FQDN—The IP address or fully qualified domain name of the subscription service. If not specified, the SIP subscribes to the SIP server.

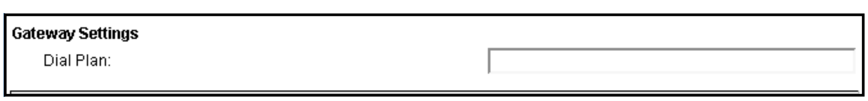


Figure 13. Gateway Settings section of the SIP window

Gateway Settings

Dial Plan. Refer to appendix C, "Dial plans" on page 78

SIP Extensions

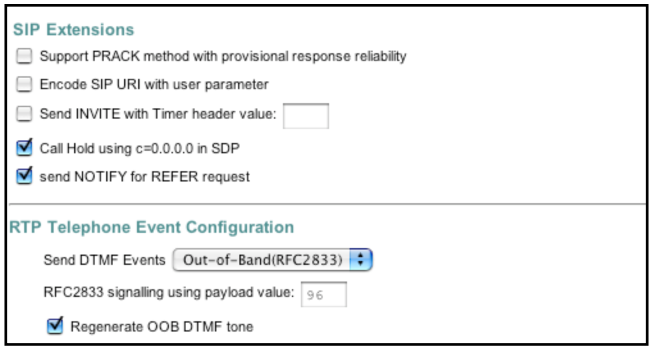


Figure 14. SIP Extensions section of the SIP window

Support PRACK method
Select to enable SIP provisional acknowledgement (PRACK) support as defined in RFC 3262.

Encode SIP URI with user parameter
Select to encode user=phone parameter in SIP URI.

Send INVITE with Timer header
Select to encode Timer header in all INVITE requests for ringing timeout.

Call Hold using C=0.0.0.0
When checked, calls will be held using the call hold method described in RFC 2543. If unchecked, the call hold would follow the RFC 3263 method.

Send NOTIFY
Send out SIP NOTIFY request to transferer for unattended and attended call transfer.

RTP Telephone Event Configuration

This sub-page allows configuration of the out-of-band signaling options for SIP. Select whether OOB telephone event signaling is to be done using the SIP INFO message, or to be done via RFC2833 RTP signaling. For additional information please refer RFC2833.

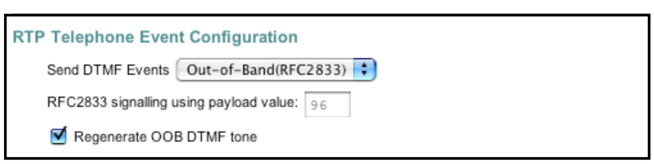


Figure 15. RTP Telephone Event Configuration section of the SIP window

VoIP VLAN Configuration

This sub-page allows configuration of specific VLAN tags that are to be applied to all SIP signalling and RTP packets used for VoIP calls. These VLAN settings will override any general VLAN settings applied to the interface.

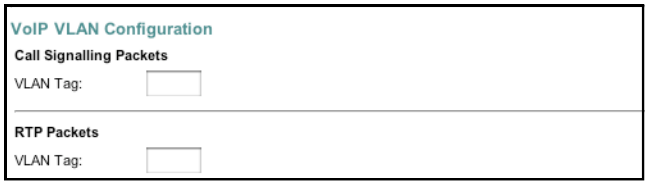


Figure 16. VoIP VLAN Configuration section of the SIP window

SIP Parameters

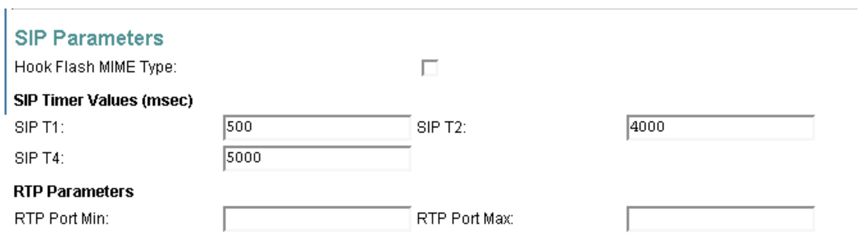


Figure 17. SIP Parameters section of the SIP window

Hook Flash MIME Type

This is the MIME Type to be used in a SIP INFO message used to signal hook flash event.

SIP Timer Values (milliseconds)

SIP T1. RFC 3261 T1 value (RTT estimate). Range: 0–64000 milliseconds (default is 5000 msec or 5 seconds).

SIP T2. RFC 3261 T2 value (maximum retransmit interval for non-INVITE requests and INVITE responses). Range: 0–64000 milliseconds (default is 4000 msec or 4 seconds).

SIP T4. RFC 3261 T4 value (maximum duration a message will remain in the network). Range: 0–64 seconds (default is 5000 msec or 5 seconds).

RTP Parameters

RTP Port Min and *RTP Port Max* define a range that contains at least four even-numbered ports (100–106, for example).

NAT Traversal

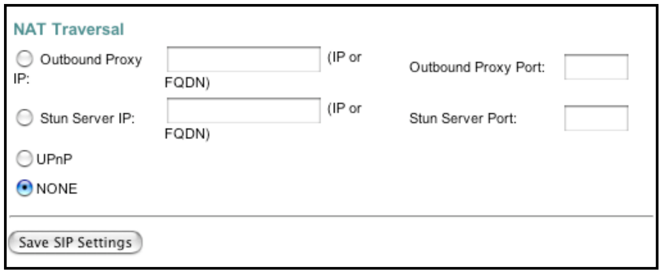


Figure 18. NAT Traversal section of the SIP window

Outbound Proxy IP

Type the fully qualified domain name for the outbound proxy server, or type the IP address provided by your service provider.

Outbound Proxy Port

Type the outbound proxy IP port number provided by your service provider.

Stun Server IP

Enter the fully qualified domain name for the stun server, or type the IP address provided by your service provider.

Stun Server Port

Type the stun server port number provided by your service provider.

UPnP

Universal plug-and-play method. This method works with NAT devices that support UPnP gateway.

NONE

Select this if you will not be using NAT traversal methods.

Saving your work

When you are finished configuring SIP settings, click the **Save SIP Settings** button (see figure 18) to save the changes.

Audio/CODEC Configuration

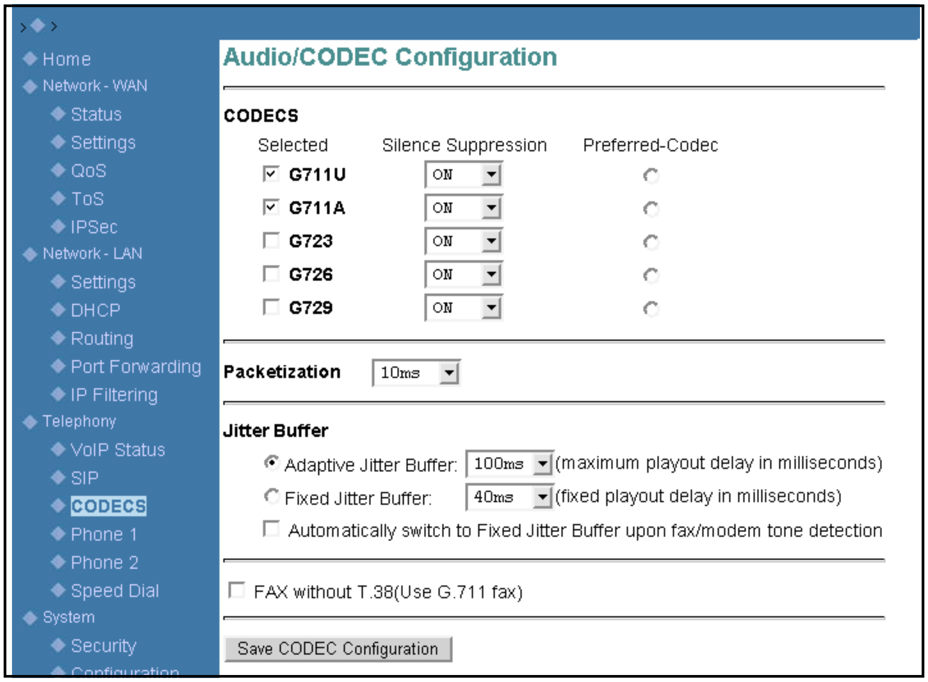


Figure 19. Audio/CODEC Configuration window

CODECS

- *Selected* column: Check the codecs that are acceptable to use
- *Silence Suppression* column: Specifies whether silence suppression should be turned on in the VOE101.
- *Preferred-Codec* column: Select the codec to be used as the first choice when encoding voice

Packetization

Configure the packet sending increment.

Jitter Buffer

Configure the timing of the voice buffering:

- Selection between adaptive or fixed jitter buffer. Default = ADAPTIVE.
- Set the adaptive jitter buffer maximum playout delay. Default = 100ms or Fixed jitter buffer playout delay. Default = 40ms
- Whether or not to automatically switch from an adaptive jitter buffer to a fixed jitter buffer upon fax/ modem tone detection

FAX without T.38 (Use G.711 fax)

When checked, T.38 is disabled and the VOE101 will not attempt T.38 session negotiation for fax transmissions initiated by the originator of the fax. The fax transmissions will be transported using G.711 fax pass-through. The selection of T.38 or G.711 fax pass-through for fax transmissions originated from the VOE101 will be determined by the receiving fax device. T.38 is enabled by default.

To disable T.38, go to *Telephony* > *CODECS* and select *FAX without T.38 (Use G.711 fax)* (see figure 19 on page 43).

Saving your work

When you are finished configuring CODEC settings, click the **Save CODEC Configuration** button (see figure 19 on page 43) to save the changes.

Phone 1

The *Phone 1* window contains the following sections:

- User Information (see figure 20)
- Supplementary Service Settings (see figure 21 on page 45)
- Dial Out Type (see figure 22 on page 47)
- Call Forward Settings (see figure 16 on page 41)
- Selective Call Forward Settings (see figure 17 on page 41)
- Incoming Call Block (see figure 18 on page 42)
- Distinctive Ring Settings (see figure 18 on page 42)
- HTTP Digest Setting (see figure 18 on page 42)

Note After configuring the sections, click the **Save** button (see figure 27 on page 49) to save the new configuration.

User Information

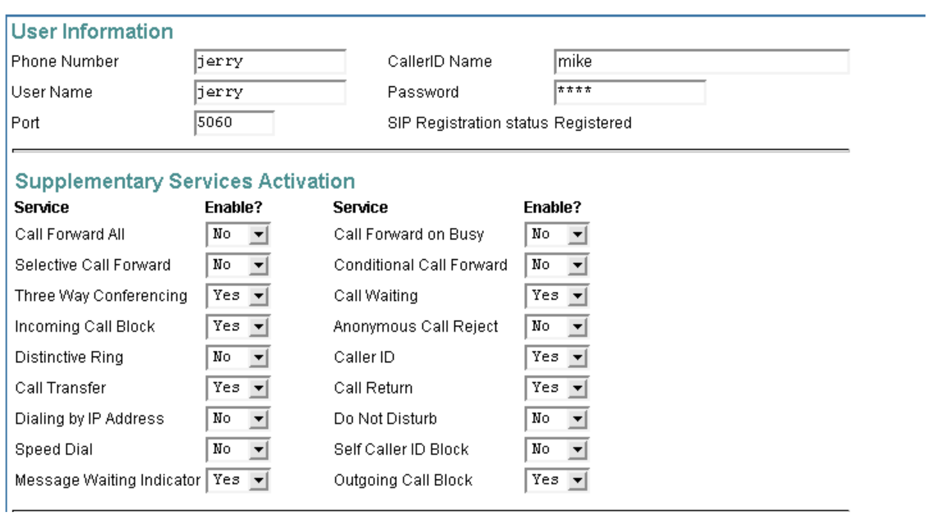


Figure 20. User Information section of Phone 1 window

Phone Number
Enter the telephone number or the user part of the SIP registration.

User Name
Enter the user name that will be used for validation of the VoIP SIP registration or call invitation.

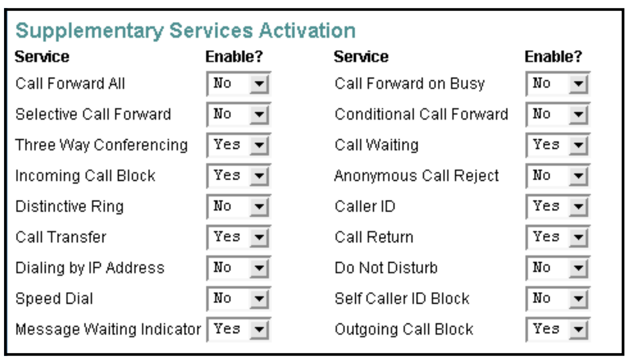
Port
Specify the signaling port.

CallerID Name
Enter the caller ID name.

Password
Enter the password.

Supplementary Service Activation

These settings enable or disable each of following calling features. Most features can also be enabled or disabled by using the telephone handset (see section “Supplementary Service Keys” on page 59 for details).


Supplementary Services Activation

Service	Enable?	Service	Enable?
Call Forward All	No	Call Forward on Busy	No
Selective Call Forward	No	Conditional Call Forward	No
Three Way Conferencing	Yes	Call Waiting	Yes
Incoming Call Block	Yes	Anonymous Call Reject	No
Distinctive Ring	No	Caller ID	Yes
Call Transfer	Yes	Call Return	Yes
Dialing by IP Address	No	Do Not Disturb	No
Speed Dial	No	Self Caller ID Block	No
Message Waiting Indicator	Yes	Outgoing Call Block	Yes

Figure 21. Supplementary Service Settings section of Phone 1 window

Call Forward All
Enable call forward all service—All received calls will be forwarded to the destination specified under the call forwarding settings.

Selective Call Forward
Enable call forward no answer service—All received calls that are not answered will be forwarded to the destination specified under the call forwarding settings.

Three-Way Conferencing
Enable three way conference service—This service enables you to add a third party to an existing two-way conversation, and hold a three-party conference call.

Incoming Call Block
Enable incoming call block service—Allows for selected inbound caller IDs to be blocked.

Distinctive Ring

Enable distinctive ringing service—This service allows additional telephone numbers to be added to an existing telephone line and when a caller dials one of these "distinctive ringing" numbers, the telephone will ring in a unique pattern to indicate which number is being dialed.

Call Transfer

Enable call transfer service—This service allows you to transfer calls to another number.

Dialing by IP Address

Enable IP dialing service—This service allows user IP addresses to be used to make calls.

Speed Dial

Enable speed dial service.

Message Waiting Indicator

Enable MWI service—The message-waiting indicator (MWI) is a common feature of telephone networks and uses an audible indication (such as a special dial tone) to indicate that a voice mail message is waiting.

Call Forward on Busy

Enable call forward on busy service.

Conditional Call Forward

Enable call forward selective service.

Call Waiting

Enable call waiting service.

Anonymous Call Reject

Enable block anonymous calls service—When enabled, calls from anonymous callers will be blocked.

Caller ID

Enable caller ID service.

Call Return

Enable call return service—When enabled, allows you to return a call to the last incoming call, whether the call was answered or not.

Do Not Disturb

Enable do not disturb service.

Self Caller ID Block

Enable blocking self caller ID shown in the outgoing message.

Outgoing Call Block

Enable blocking of outgoing calls based on the digit pattern specified under outgoing call block.

Dial Out Type

Figure 22. Dial Out Type section of Phone 1 window

Dial Out Type

Enable Hot-Line and Warm-Line services. To achieve this, one sequence in the dial plan must start with a pause, with a 0 delay for a Hot Line, and a non-zero delay for a Warm Line.

Hot Line Number

Input the number for Hot Line function—This number will be called immediately when the telephone goes off-hook.

Warm Line Number

Input the number for Warm Line function—The warm line function provides a delay period after the telephone goes off-hook for the user to dial a number different than that specified for the warm line. If the delay period has expired with no number being dialed, the warm line number will be dialed. The delay period is set under *System > Localization*.

Call Forward Settings

Figure 23. Call Forward Settings section of Phone 1 window

Cfwd All Dest

Input the destination for all call forwarding.

Cfwd Busy Dest

Input the destination for all busy call forwarding.

Selective Call Forward Settings

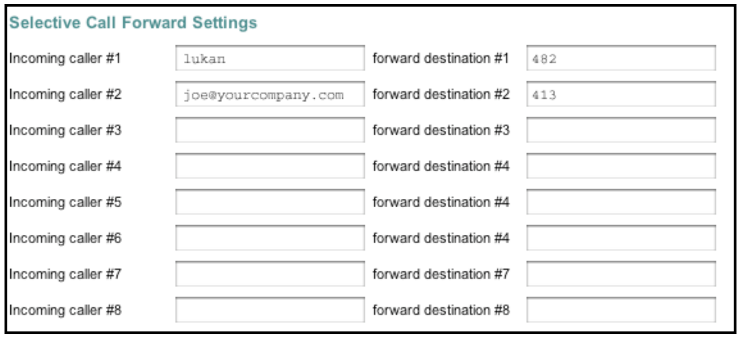


Figure 24. Selective Call Forward Settings section of Phone 1 window

Incoming caller #1–8
Up to 8 incoming calls can be selected for call forwarding.

Forward destination #1–8
Up to 8 destinations to which incoming calls can be forwarded.

Incoming Call Block

Figure 25. Incoming Call Block section of Phone 1 window

Block Caller ID
Specify a Caller ID for call block.

Outgoing Call Block

Outgoing call block allows for the blocking of an outbound call based on the digit pattern dialed. For example: outbound call blocking may be used to prevent an end user from dialing long distance or toll calls. Outbound call blocking uses the same digit matching process as the dial plan described in appendix C, "Dial plans" on page 78.

Figure 26. Incoming Call Block section of Phone 1 window

Digit Pattern
The *Digit Pattern* box is where you can specify the outgoing call block pattern.

Example: Specifying an outgoing call block pattern of *002|009|0204* will block all outbound calls to numbers 002, 009, or 0204.

HTTP Digest Setting

SIP INVITE must contain a valid Authorization header that is based on an Auth ID and a password using MD5 digest algorithm. The Auth ID must be specified in the username parameter in the Authorization header.

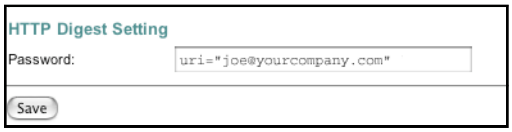


Figure 27. HTTP Digest Setting section of Phone 1 window

Saving your work

When you are finished configuring settings, click the **Save** button (see figure 27) to save the changes.

Speed Dial

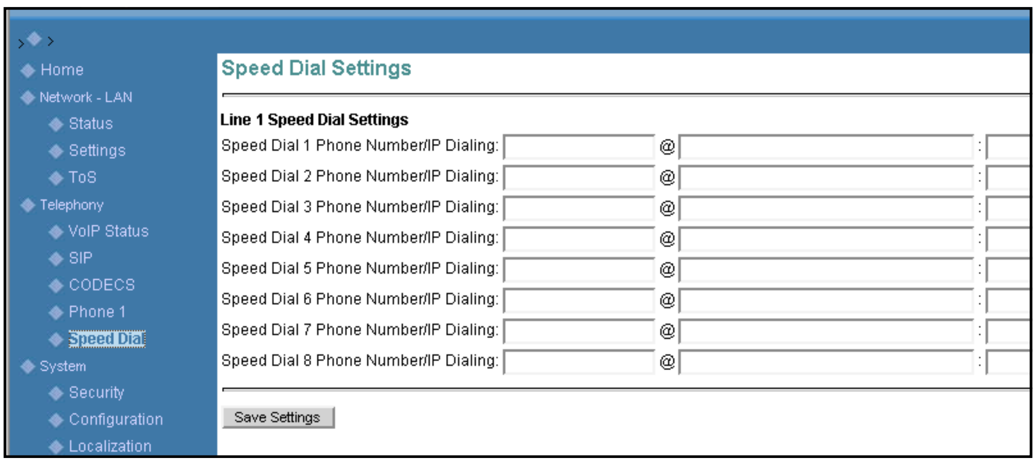


Figure 28. Speed Dial window

Line 1 Speed Dial Settings

Speed Dial Serv

Enable Speed Dial Service.

Speed Dial 1–8 Phone Number/IP Dialing

Target 1–8 phone number (or URL) assigned to speed dial.

Saving your work

When you are finished configuring settings, click the **Save Settings** button (see figure 28) to save the changes.

Chapter 6 **System**

Chapter contents

Set Security Password

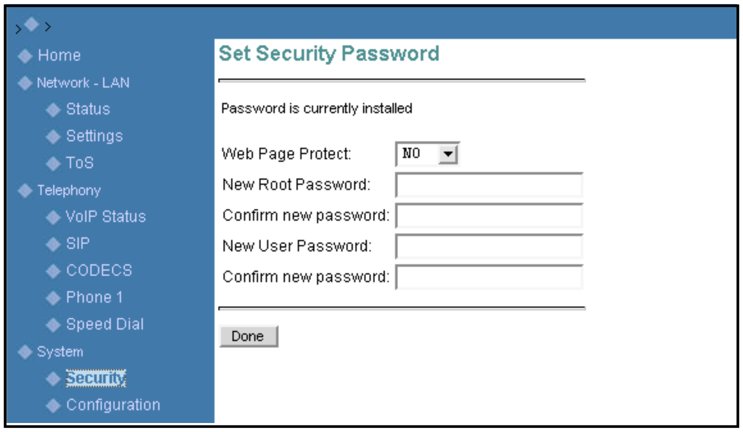


Figure 29. Set Security Password window

Two levels of system configuration are available: user level and system level. Access to each level is password controlled.

Web Page Protect

Enable or disable web access protection. When set to *YES*, a password will be required to access the VOE101 web configuration interface.

New Root Password

Type the administrator's password.

New User Password

Type the user's password.

Confirm new password

Re-enter the password for confirmation.

Saving your work

When you are finished configuring security settings, click the **Save Settings** button to save the changes.

Configuration

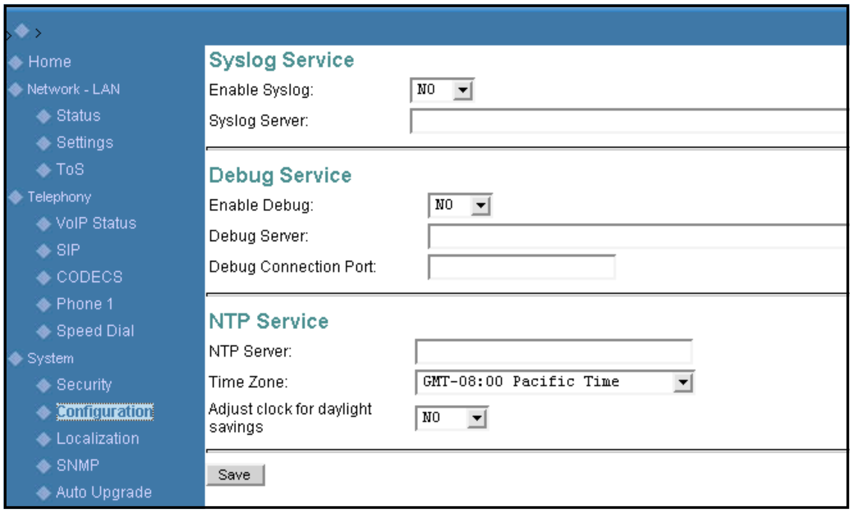


Figure 30. Configuration window

Syslog Service

Enable Syslog

Enable or disable system logging. During normal operations, Syslog should be turned off. A sample Syslog record of a complete call is shown below.

```
SYSLOG

11-09-2005     11:28:17       Local0.Info        209.49.110.185   Nov 09
08:28:34 syslog: [1234] On-Hook [7190]
11-09-2005     11:28:06       Local0.Info        209.49.110.185   Nov 09
08:28:23 syslog: [1234] Call-Out [7190]
11-09-2005     11:27:50       Local0.Info        209.49.110.185   Nov 09
08:28:07 syslog: [1234] Call-Out [190]
11-09-2005     11:16:32       Local0.Info        209.49.110.185   Nov 09
08:16:49 syslog: [1234] On-Hook [555]
11-09-2005     11:16:16       Local0.Info        209.49.110.185   Nov 09
08:16:33 syslog: [1234] Call-Out [7190]
11-09-2005     11:15:38       Mail.Emerg         209.49.110.185   Jan 01
00:00:00 syslog: DownloadConfig:No, DownloadImage:No
11-09-2005     11:15:36       Local0.Info        209.49.110.185   Jan 01
00:00:00 syslog: Check Autoupgrade
```

Syslog Server

Specify the syslog server IP address or DNS name. This feature specifies the server for logging VOE101 system information and critical events.

Debug Service

Enable Debug

Enable or disable System Debug.

Debug Server

The debug server IP address and port. This specifies the server for logging VOE101 debug information. Debug information can be sent to a syslog server. A sample record of debug output is shown below:

```
SYSLOG

07-07-2005 14:49:09 Local7.Debug 10.10.22.13 **EndPoint State: StateConnect ==> StateOnHook
   in src/common/endpoint.c: 333
07-07-2005 14:49:09 Local0.Info 10.10.22.13 Jul 07 11:49:09 syslog: [usersjphone] On-Hook
   [100]
07-07-2005 14:49:03 Local7.Debug 10.10.22.13 !!!! Handset 349b4 received event 52 in file src/
   common/fxs/ConnectFxs.c line 466
07-07-2005 14:48:59 Local7.Debug 10.10.22.13 The EnvetString is f and length is 1.
07-07-2005 14:48:59 Local7.Debug 10.10.22.13 **EndPoint State: StateWaiting ==> StateConnect
   in src/common/fxs/WaitingFxs.c: 116
07-07-2005 14:48:59 Local7.Debug 10.10.22.13 call 468bc ==>CallChangeState():: from
   CALL_ALERT to CALL_CONNECT in file src/common/fxs/WaitingFxs.c line 114
07-07-2005 14:48:59 Local7.Debug 10.10.22.13 APP rcv rsp 200 of req 32769
07-07-2005 14:48:59 Local7.Debug 10.10.22.13 call 468bc ==>CallChangeState():: from
   CALL_OUTGOING to CALL_ALERT in file src/common/fxs/WaitingFxs.c line 58
07-07-2005 14:48:59 Local7.Debug 10.10.22.13 APP rcv rsp 180 of req 32769
07-07-2005 14:48:58 Local7.Debug 10.10.22.13 !!!! Handset 349b4 received event 26 in file src/
   common/fxs/WaitingFxs.c line 212
07-07-2005 14:48:58 Local7.Debug 10.10.22.13 APP rcv rsp 100 of req 32769
07-07-2005 14:48:58 Local7.Debug 10.10.22.13 **EndPoint State: StateOffHook ==> StateWaiting
   in src/common/fxs/OffHookFxs.c: 418
07-07-2005 14:48:58 Local7.Debug 10.10.22.13 Dial numbers is 100
07-07-2005 14:48:58 Local0.Info 10.10.22.13 Jul 07 11:48:58 syslog: [usersjphone] Call-Out
   [100]
07-07-2005 14:48:58 Local7.Debug 10.10.22.13 call 0 ==>CallChangeState():: from CALL_IDLE to
   CALL_OUTGOING in file src/common/call.c line 80
07-07-2005 14:48:58 Local7.Debug 10.10.22.13 Codec[2]=8
07-07-2005 14:48:58 Local7.Debug 10.10.22.13 Codec[1]=18
07-07-2005 14:48:58 Local7.Debug 10.10.22.13 Codec[0]=0
07-07-2005 14:48:54 Local7.Debug 10.10.22.13 The EnvetString is 100 and length is 3.
07-07-2005 14:48:53 Local7.Debug 10.10.22.13 The EnvetString is 10 and length is 2.
07-07-2005 14:48:53 Local7.Debug 10.10.22.13 The EnvetString is 1 and length is 1.
07-07-2005 14:48:51 Local7.Debug 10.10.22.13 **EndPoint State: StateOnHook ==> StateOffHook
   in src/common/fxs/OnHookFxs.c: 55
```

Debug Connection Port

The port number of the debug server to be used for receiving debug messages from the VOE101. Use port 412 to send debug output to a syslog server.

NTP Service

The network time protocol (NTP) synchronizes timekeeping among a set of distributed time servers and clients. An NTP client synchronizes the local clock with some other time source, usually an NTP server. A list of public NTP servers is available at http://ntp.isc.org/bin/view/Servers/WebHome.

NTP Server

Specify the IP address of the NTP server.

Time Zone
Select the GMT time zone for your location. For help in determining the time zone, go to the *UTC/GMT Conversion* website at **http://www.dxing.com/utcgmt.htm**.

Adjust clock for daylight savings
Select *Yes* if you want the VOE101 to automatically compensate for daylight savings time.

Saving your work

When you are finished configuring settings, click the **Save** button to save the changes.

Service Access

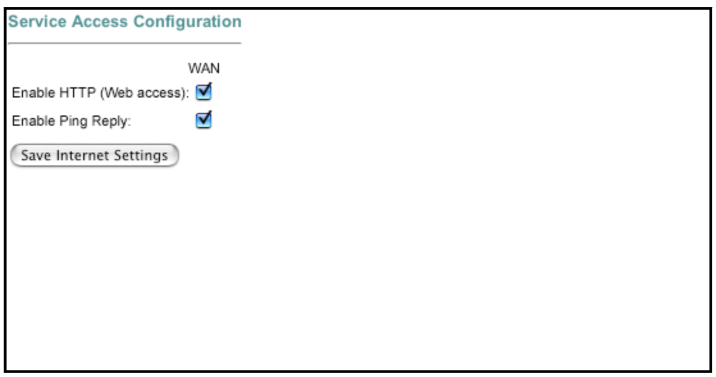


Figure 31. Service Access window

Enable HTTP

When checked access to the web configuration interface (HTTP) of the VOE101 through the WAN interface port is enabled.

Note If the *Enable HTTP* option is not checked, your WAN service provider will no longer be able to access the VOE101 to make configuration changes.

Enable Ping Reply

When checked, the VOE101 will reply to ping requests received on the WAN port. If the option is not checked, the VOE101 will not respond to ping requests received on the WAN port.

Saving your work

When you are finished configuring settings, click the **Save Service Access Settings** button to save the changes.

Localization

The *Localization* window contains the following sections:

- Call Progress Tones (see figure 32 on page 57)
- Distinctive Ring Setting (see figure 33 on page 58)
- Supplementary Service Keys (see figure 34 on page 60)

- Control Timer Values (see figure 36 on page 63)
- FXS Port (see figure 37 on page 64)
- Ring Setting (see figure 38 on page 64)
- FXS Port Polarity (see figure 39 on page 65)

Note After configuring the sections, click the **Save** button (see figure 39 on page 65) to save the new configuration.

Call Progress Tones

The dialtone, confirmation, busy, ringback and offhook warning call progress tones that are played to the handset can be set manually or the tones will be set automatically to the country specified under *Call ID method* if the checkbox, *Call Progress Tones synchronization* has been selected.

To set the call progress tones manually specify:

```
[frequency 1]@[Energy 1]+[frequency 2]@[Energy 2]#ON([ms]),OFF([ms]),R
```

Where:

- *Frequency 1* and *frequency 2* are the frequency of the tone to be played.
- *Energy 1* and *energy 2* are the energy level of the tone to be played in dBm.
- *ON(ms)* is the duration that the tone will be played in milliseconds.
- *OFF(ms)* is the duration of the pause between cycles of playing the tone.
- *R* indicates the tone should be repeated using the ON and OFF durations specified

To specify a 350-Hz and 480-Hz tone that will be played at 15 dB for one second and then pause for 0.5 seconds, you would specify:

```
350@-15+480@-15#ON(1000),OFF(500),R
```

The following example would repeat playing of a tone of 280 Hz at -8 dBm for 0.3 seconds with a pause of 0.1 second and a tone of 550 Hz for 0.8 seconds with a pause of 0.15 seconds.

```
280@-8+550@-20#ON(300),OFF(100),ON(800),OFF(150),R
```

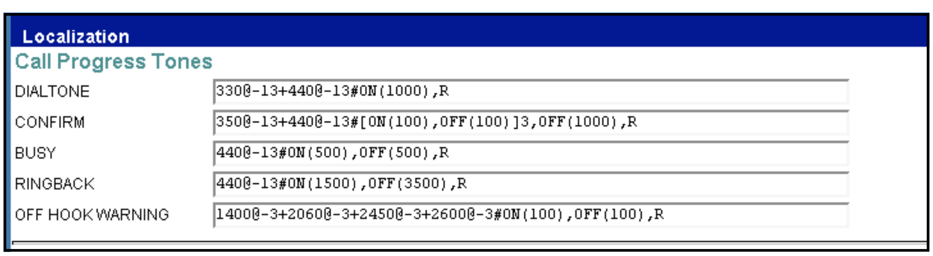


Figure 32. Call Progress Tones section of Localization window

Dial Tone

Played when prompting the user to enter a phone number.

Prompt Tone
Played when prompting the user to enter a call forward phone number.

Confirm Tone
This should be a brief tone to notify the user that the last input value has been accepted.

Holding Tone
Indicate to the local user that the far end has placed the call on hold.

Busy Tone
Played when a 486 RSC is received for an outbound call.

Ring Back Tone
Played for an outbound call when the far end is ringing.

Off Hook Warning
Played when the subscriber does not place the handset on the cradle properly.

Distinctive Ring Settings
The VOE101 Distinctive Ring Settings window (see figure 33) enables you to specify up to 8 sets of distinctive ring cadences.

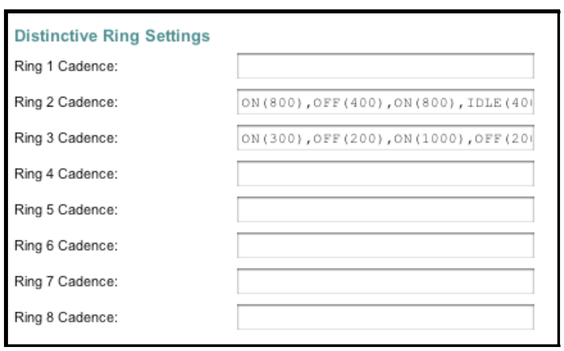


Figure 33. Distinctive Ring Setting section of Localization window

The following is a sample ring cadence pattern configuration:

```
Timeval ::= time in milliseconds
Repeatval ::= # of cycles to repeat
Tonename ::= "RING_0" | "RING_1" | "RING_2" | "RING_3" | "RING_4"
| "RING_5" | "RING_6" | "RING_7" | "RING_8" | "RING_9"
Idle ::= "IDLE"
Active ::= "ON" | "OFF"
Inactive ::= Idle "(" Timeval ")"
Active ::= Active "(" Timeval ")"
Sequence ::= Active | Active "," Sequence
Repetition ::= "[" Sequence "]" Repeatval
Repeat ::= "R"
Cycle ::= Sequence | Repetition
Fullsequence ::= Cycle | Cycle "," Fullsequence
Cadence ::= Fullsequence | Fullsequence "," Repeat | Fullsequence
"," Inactive "," Repeat
Ring ::= Cadence
```

Note The Bellcore standard ring cadence patterns are shown in table 2.

Table 2. Bellcore standard ring cadence patterns

Name	Value
RING_0	ON(2000), IDLE(4000), R
RING_1	ON(800), OFF(400), ON(800), IDLE(4000), R
RING_2	ON(400),OFF(200)]2,ON(800),IDLE(4000),R
RING_3	ON(300), OFF(200), ON(1000), OFF(200), ON(300), IDLE(4000), R
RING_4	ON(500)
RING_5	

Supplementary Service Keys

Supplementary Service Keys are key sequences that can enable and disable certain calling features from an analog phone handset attached to the VOE101.

Note The supplementary service keys must begin with a '*','#' character, or f (flash hook) and follow a 1 or 2 numeric digit(s).

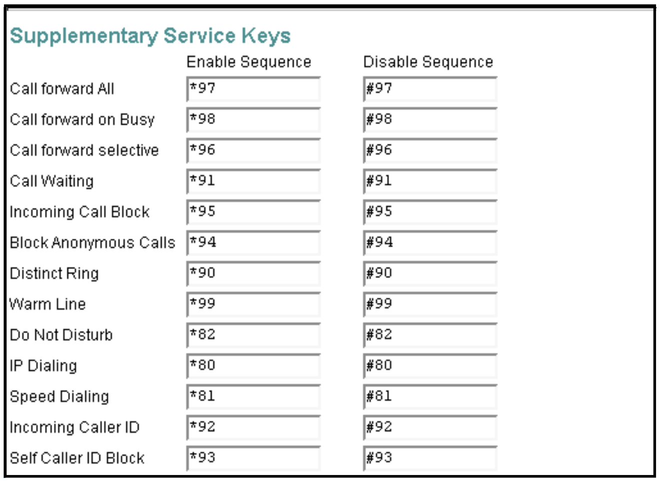


Figure 34. Supplementary Service Keys section of Localization Window

Call forward All
Enables or disables the unconditional call forwarding feature.

Call forward on Busy
Enables or disables the call forwarding on busy feature.

Call forward selective
Enables or disables the call forwarding on busy feature.

Call Waiting
Enables or disables the call waiting feature.

Incoming Call Block
Enables or disables the blocking of incoming calls based on the phone number of the caller feature.

Block Anonymous Calls
Enables or disables the blocking of incoming calls that are using the anonymous caller ID feature.

Distinctive Ring
Enables or disables the distinctive ring tones based on the phone number of the calling party feature.

Warm Line
Enables or disables the warm line calling feature.

Do Not Disturb
Enables or disables the do not disturb feature.

IP Dialing
Enables or disables dialing by IP address.

Speed Dialing
Enables or disables the speed dialing feature.

Income Caller ID
Enables or disables the display of incoming caller ID feature.

Self Caller ID Block
Enables or disables the blocking of transmission of caller ID feature for outgoing calls.

Calling Prefix Keys
Calling Prefix Keys and other configuration parameters define the behavior of the calling feature.

Note Following function key must be start by '*','#' character, or f (flash hook) and follow a 1 or 2 numeric digit(s).

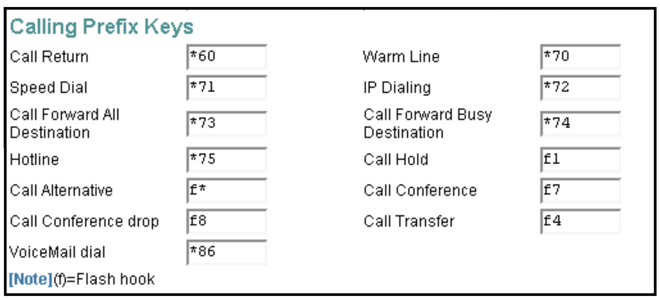


Figure 35. Calling Prefix Keys section of Localization Window

Call Return
Automatically places a call to the number of the last call received.

Warm Line
Configures the number to call automatically after a delay period once the telephone goes off hook. To configure the warm line number enter the calling prefix key (**70* is the factory default), the number to dial and then on hook the phone. You will hear a second dial tone after entering the calling prefix key. Before using warm-line calling it must be enabled using the activate sequence defined in the supplementary service table.

Example:

***7013019751000** and on-hook the phone. When the handset is lifted, *301951000* will be called if another number is not entered within the delay period specified under *System* > *Localization*.

Speed Dial
Configures speed dialing keys. To configure speed dialing numbers enter the calling prefix key (*71 is the factory default), the speed dial key, the number to dial and then on hook the phone. You will hear a second dial tone after entering the calling prefix key. Speed dial calling must be enabled using the activate sequence defined in the supplementary service table.

Example:

***7123019751000** and on-hook the phone. When **2** is pressed on the telephone key pad, *3019751000* will be called.

Call Forward All

Configures the number to unconditionally forward all calls to. To configure call forward all enter the calling prefix key (**73* is the factory default), the number to dial and then on hook the phone. You will hear a second dial tone after entering the calling prefix key. Before using, *Call Forward All* must be enabled using the activate sequence defined in the supplementary service table.

Example:

***733019751000** and on-hook the phone. All calls received will forward to *3019751000*.

IP Dialing

Allows direct calling using an IP address. To use calling by IP address enter the calling prefix key (**72* is the factory default), the * key, the IP address using the * key to between octets of the IP address, the * key and the port number. You will hear a second dial tone after entering the calling prefix key. Before using direct IP address dialing must be enabled using the activate sequence defined in the supplementary service table.

Example:

To place a call to IP address 192.168.1.20:5061 enter ***73*192*168*1*20*5060**.

Call Hold

Configures the key sequence to place a call on call on hold – The factory default is flash hook *1*.

Call Waiting (call alternative)

Configures the key sequence to switch between calls. The factory default is flash hook ***.

Call Conference

Configures the key sequence to conference two calls together. The factory default is flash hook *7*.

Call Conference Drop

Configures the key sequence to drop the last call that was added to the conference the conference. The factory default is flash hook *8*.

Call Transfer

Configures the key sequence for call transfer. The factory default is flash hook *4*.

Voicemail access

Configures the key sequence for call for voicemail access. The factory default is flash hook **86*. If the voicemail key sequence is entered on the phone a call will automatically be placed to the configured service provider using the configured authentication information.

Call Forward Busy Destination

Configures the number to forward calls to when busy. To configure call forward busy enter the calling prefix key (*74 is the factory default), the number to dial and then on hook the phone. You will hear a second dial tone after entering the calling prefix key. Before using, call forward busy must be enabled using the activate sequence defined in the supplementary service table.

Example:

***743019751000** and on-hook the phone. All calls received will forward to *3019751000* when the phone is busy.

Hotline

Configures the number to call when the telephone goes off hook. To configure the hotline number enter the calling prefix key (*75 is the factory default), the number to dial and then on hook the phone. You will hear a second dial tone after entering the calling prefix key. Before using hotline calling it must be enabled using the activate sequence defined in the supplementary service table.

Example:

***753019751000** and on-hook the phone. When the handset is lifted, *3019751000* will immediately be called.

Control Timer Values

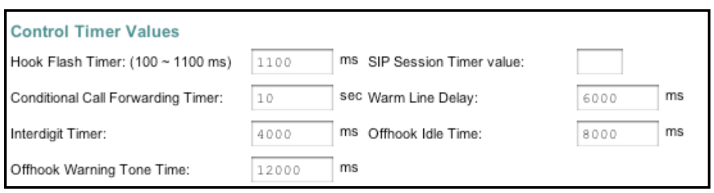


Figure 36. Control Timer Values section of Localization window

Hook Flash Timer

Maximum on-hook time before off-hook to qualify as hookflash. More than this value and the on-hook event is treated as on-hook (hanging up the call).

Minimum on-hook time before off-hook to qualify as hookflash. At less than this value, the on-hook event is ignored.

SIP Session Timer value

The amount of time the VOE101 will wait during an active call to send repeated re-invites on active calls to allow the SIP server to determine the status of a call.

Conditional Call Forwarding Timer

Specified a time period as a call forward condition. After the number of seconds specified, the conditional call forwarding process will be performed.

Warm Line Delay

Specify a time period as a delay time for warm line dialing. If warm line is enabled, then the VOE101 will wait this amount of time after the handset is off-hook before dialing the warm line phone number.

Interdigit Timer

The number of seconds the VOE101 will wait for the caller to input a subsequent digit of the dialed number. If the timer value is exceeded before the dial plan is matched (see section "Gateway Settings" on page 39), the busy tone will be played to the caller.

Offhook Idle Time

If the handset is off-hook with no dialing activity for longer than the time specified, then the busy tone will be played. The default is 6000 ms (6 seconds).

Offhook Warning tone time

If the handset is off-hook with no dialing activity for longer than the time specified, then the off-hook warning tone will be played. The default is 12000 ms (12 seconds). This value should be greater than the off-hook idle time (see section "Offhook Idle Time").

FXS Port

Choose the correct country for a proper impedance match.

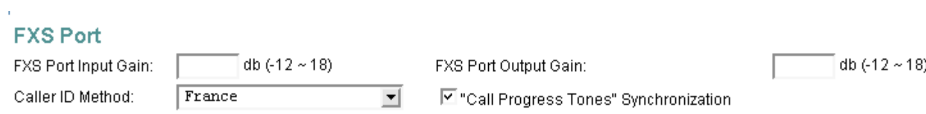


Figure 37. FXS Port Polarity Configuration section of Localization window

FXS port Input Gain

Adjust the input gain level for FXS port.

FXS port Output Gain

Adjust the output gain level for FXS port.

Caller ID Method

Specifies the country-specific Caller ID format.

"Call Progress Tones" Synchronization

When *"Call Progress Tones" Synchronization* is selected, the Caller ID presentation method value is used to automatically set the county-specific call progress tones.

Ring Setting

Figure 38. Ring Setting section of Localization window

Ring Waveform

Specify the ring tone waveform.

Ring Frequency
Specify the ring tone frequency.

Ring Voltage
Specify the ring tone voltage.

FXS Port Polarity Configuration

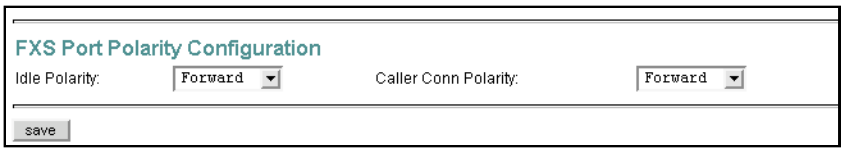


Figure 39. FXS Port Polarity section of Localization window

Idle Polarity
Polarity before call connected.

Caller Conn Polarity
Polarity after outbound call connected.

Saving your work

When you are finished configuring settings, click the **Save** button (see figure 39) to save the changes.

SNMP Configuration

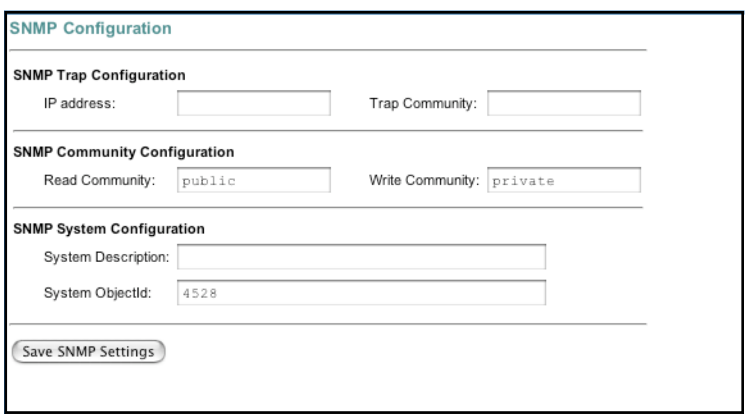


Figure 40. SNMP Configuration window

SNMP Trap Configuration

IP address
Trap host IP address.

Trap Community
The community name used by the SNMP manager to verify traps. The default value is *public*.

SNMP Community Configuration

Read Community

The community name used by the SNMP manager when reading SNMP data items from a client MIB. The default value is *public*.

Write Community

The community name used by the SNMP manager when setting SNMP data items in a client's MIB. The default value is *public*.

SNMP System Configuration

System Description

Description of the unit (e.g. "John's phone")

System Object Id

A vendor's enterprise ID

Saving your work

When you are finished configuring settings, click the **Save SNMP Settings** button to save the changes.

Auto Upgrade

The VOE101 family includes a configuration and firmware download manager server that allows for the updating of large numbers of VOE101s from a central location. By factory default, all VOE101 units are set with auto-update on and to access the Black Box auto-update server. Many of Black Box's carrier customers have chosen to setup their own auto-update server to provide service specific information for their end users. Black Box recommends that end users do not change the auto-update server or set "Enable auto-upgrade" to "NO" without consulting with their service provider.

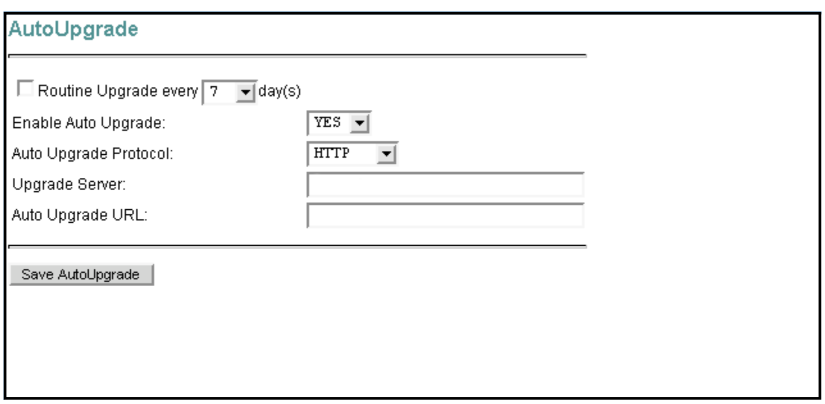


Figure 41. AutoUpgrade window

Routine Upgrade every xxx day(s)

If selected, the VOE101 will check for automatically check for updates at system reload time and on a periodic basis based on the number of days selected.

Enable Auto Upgrade

Enable or disable auto upgrade—If enabled, the VOE101 will automatically check the upgrade server for new system firmware and software upon reload, power cycle, or when the routine upgrade interval expires.

Auto Upgrade Protocol

Select the protocol for auto upgrade

Upgrade Server

Specify the auto upgrade server IP address

Auto Upgrade URL

Specify the auto upgrade server by URL. This field is dependent on the auto upgrade service package installation. The default value is **iadmgr**.

Saving your work

When you are finished configuring settings, click the **Save AutoUpgrade** button to save the changes.

Manual Upgrade

For both **HTTP** and **TFTP** methods, the device will reboot itself into the downloader mode if the main application is executing, and proceed with the ROM file download and permanent write of the application to the device's flash memory. During download of new firmware images, the LEDs on the VOE101 will flash sequentially. Typically, HTTP downloads take about 5–10 minutes. After the download is completed, the download status page will be displayed.

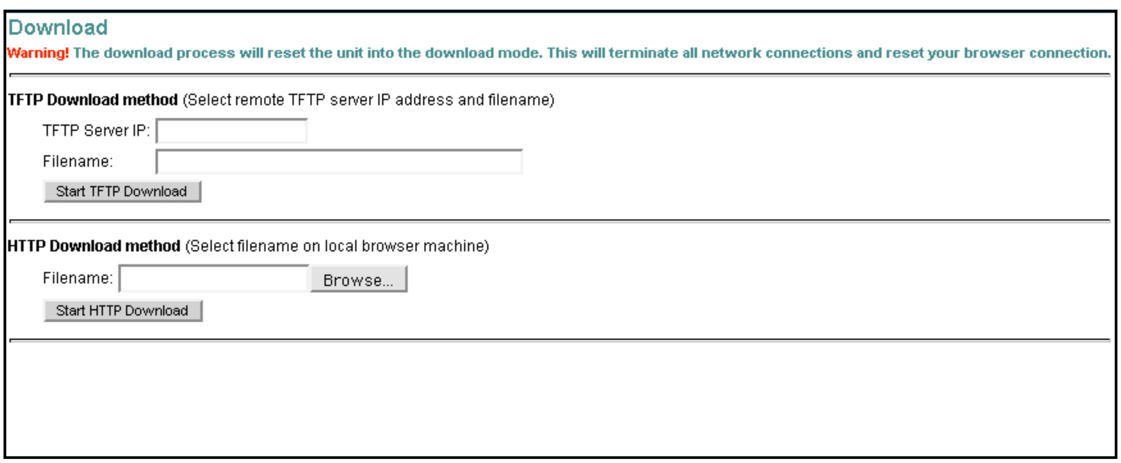


Figure 42. Manual Upgrade window

Reload

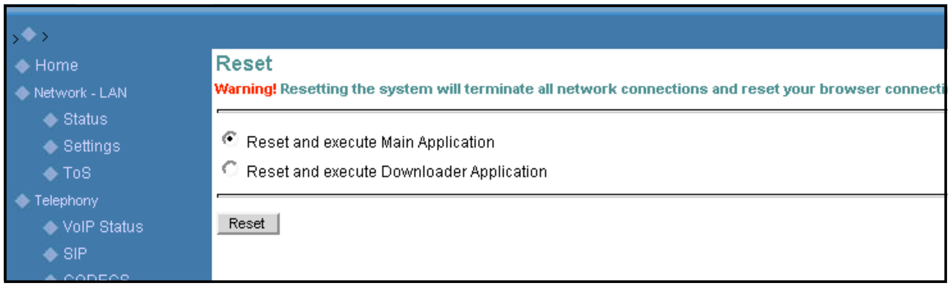


Figure 43. Reload window

CAUTION

Reloading the system will terminate all network connections and restart your browser connection.

Chose the *Reload and execute Main Application* option, for execution of the main application which you have configure, once you reload the system.

Chose the *Reload and execute Downloader Application* option, to being downloading, once you reload the system.

Chapter 7 Documentation

Chapter contents

Introduction

Clicking the **Documentation** link (see figure 44) connects to the Black Box website to display the most current version of the *VOE101 Getting Started Guide* in portable document format (PDF).

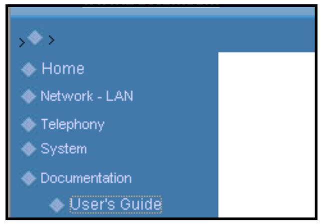


Figure 44. Documentation link

Chapter 8 Logout

Chapter contents

Introduction

Clicking *Logout* (see figure 45) displays *Logout—Sure to Logout?*.

If you want to exit from the VOE101 management utility, click the **Logout** button (see figure 45).

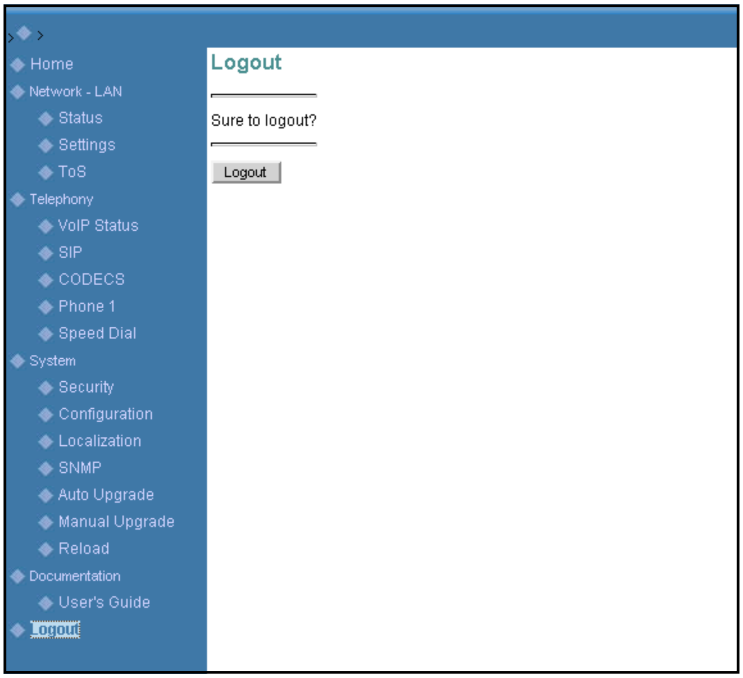


Figure 45. Logout window

You will be returned to the password verification page (see figure 46).

Figure 46. Password verification page

Appendix A **Compliance information**

Chapter contents

Compliance

EMC Compliance:

FCC Part 15, Class B

EN55022, Class B

EN55024

Safety Compliance

EN60950-1

Radio and TV Interference (FCC Part 15)

This equipment generates and uses radio frequency energy, and if not installed and used properly—that is, in strict accordance with the manufacturer's instructions—may cause interference to radio and television reception. This equipment has been tested and found to comply with the limits for a Class B computing device in accordance with the specifications in Subpart B of Part 15 of FCC rules, which are designed to provide reasonable protection from such interference in a commercial installation. However, there is no guarantee that interference will not occur in a particular installation. If the equipment causes interference to radio or television reception, which can be determined by disconnecting the cables, try to correct the interference by one or more of the following measures: moving the computing equipment away from the receiver, re-orienting the receiving antenna, and/or plugging the receiving equipment into a different AC outlet (such that the computing equipment and receiver are on different branches).

CE Notice (Declarations of Conformity)

We certify that the apparatus identified in this document conforms to the requirements of Council Directive 1999/5/EC on the approximation of the laws of the member states relating to Radio and Telecommunication Terminal Equipment and the mutual recognition of their conformity.

The safety advice in the documentation accompanying this product shall be obeyed. The conformity to the above directive is indicated by the **CE** sign on the device.

Appendix B **Specifications**

Chapter contents

Voice Connectivity

2-wire Loopstart, RJ-11/12

Short haul loop 1.1 km @3REN

Caller-ID Type-1/2 FSK and ITU V.23/Bell 202 generation

Connectivity

1 10/100 Full Duplex/Autosensing Ethernet RJ-45

Voice Processing (signalling dependent)

SIP

MGCP

- Packet Cable NCS 1.0
- IETF MGCP 1.0

Voice codes

- G.711 A-Law/μ-Law (64 kbps)
- G.726 (ADPCM 40, 32, 24, 16 kbps)
- G.723.1 (5.3 or 6.3 kbps)
- G.729ab (8 kbps)

G.168 echo cancellation

2 parallel voice connections

DTMF detection and generation

Carrier tone detection and generation

Silence suppression and comfort noise

Configurable dejitter buffer

DTFM in-band & out-of-band

Configurable transmit packet length

RTP/RTCP (RFC 1889)

Fax and Modem Support

G.711 transparent FAX

T.38 Fax relay (9.6 k, 14.4 k)

Voice Services/Features

Call forwarding

Call transfer

Call hold

Call waiting

3-way calling

IP Services

IPv4 ; RIPv1, v2 (RFC 1058 and 2453)

IP filtering

NAPT

NTP

DHCP client & server

PPPoE

ICMP redirect (RFC 792); Packet fragmentation

DiffServe/ToS set or queue per header bits

Management

Browser configuration interface

Voice prompt configuration

TFTP configuration & firmware loading

SNMP v2 agent (MIB II and private MIB)

Operating Environment

Operating temperature: 0–40°C (32–104°F)

Operating humidity: 5–80% (non condensing)

System

Power: 100–240 VAC (50/60 Hz)

Appendix C **Dial plans**

Chapter contents

Introduction

The MGCP and SIP code will allow provisioning (via web browser) of the dial plan. A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the SIP server or gatekeeper for resolution into a destination IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification.

The formal syntax of the dial plan is described by the following notation:

```
Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"
Timer ::= "T" | "t"
Letter ::= Digit | Timer | "#" | "*" | "A" | "a" | "B" | "b" | "C" | "c"| "D" | "d"
Range ::= "X" | "x" -- matches any digit
| "[" Letters "]" -- matches any of the specified letters
Letters::= Subrange | Subrange Letters
Subrange::= Letter -- matches the specified letter
| Digit "-" Digit -- matches any digit between first and last
Position::= Letter | Range
StringElement::= Position -- matches any occurrence of the position
| Position "." -- matches an arbitrary number of occurrences
including 0
String ::= StringElement | StringElement String
StringList::= String | String "|" StringList
DialPlan::= String | "(" StringList ")"
```

A dial plan, according to this syntax, is defined either by a (case insensitive) string or by a list of strings. Regardless of the above syntax a timer is only allowed if it appears in the last position in a string (12T3 is not valid). Each string is an alternate numbering scheme. The unit will process the dial plan by comparing the current dial string against the dial plan, if the result is under qualified (partial matches at least one entry) then it will do nothing further. If the result matches or is over-qualified (no further digits could possibly produce a match) then send the string to the gatekeeper and clear the dial string. The Timer T is activated when it is all that is required to produce a match. The period of timer T is 4 seconds. For example a dial plan of (xxxT|xxxxx) will match immediately if 5 digits are entered, it will also match after a 4 second pause when 3 digits are entered.

Sample Dial Plans

Simple Dial Plan

Allows dialing of 7-digit numbers (e.g. 5551234) or an operator on 0. Dial plan is (0T|xxxxxxx)

Non-dialed Line Dial Plan

As soon as handset is lifted the unit contacts the gatekeeper (used for systems where DTMF detection is done in-call). Dial plan is (x.) i.e. match against 0 (or more) digits. Note: the dot '.'

Complex Dial Plan

- Local operator on 0, long distance operator on 00
- 4-digit local extension number starting with 3, 4, or 5
- 7-digit local numbers are prefixed by an 8

- 2-digit star services (e.g. 69)
- 10-digit long distance prefixed by 91
- International numbers starting with 9011+variable number of digits.

Dial plan for this is:

```
(0T|00T|[3-5]xxx|8xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)
```

Appendix D **Calling Features**

Chapter contents

Introduction

The VOE101 family supports advanced calling features that can be turned on and off from phones attached to the VOE101 (see table 3).

Note Your telephony service provider must enable your service for these calling features to work.

Note F in table 3 refers to the *hook flash* event.

Table 3. Calling features

Feature	Keypad	Feature	Keypad
Call Hold	**F1**	Call Retrieve	**F***
Conference	**F7**	Conference **Drop**	**F8**
Call Transfer	**F4**		
Do not Disturb **ON**	***82**	Do not Disturb **OFF**	**#82**
Distinctive **ON**	***90**	Distinctive **OFF**	**#90**
Call Waiting **ON**	***91**	Call Waiting **OFF**	**#91**
Incoming Caller ID Display **ON**	***92**	Incoming Caller ID Display **OFF**	**#92**
Self Caller ID Block Service **ON**	***93**	Self Caller ID Block Service **OFF**	**#93**
Anonymous Call Reject **ON**	***94**	Anonymous Call Reject **OFF**	**#94**
Incoming Call Block **ON**	***95**	Incoming Call Block **OFF**	**#95**
Call Forward Selective **ON**	***96**	Call Forward Selective **OFF**	**#96**
Call Forward All **ON**	***97**	Call Forward All **OFF**	**#97**
Call Forward Busy **ON**	***98**	Call Forward Busy **OFF**	**#98**
Warm Line **ON**	***99**	Warm Line **OFF**	**#99**
IP Dialing **ON**	***80**	IP Dialing **OFF**	**#80**
Speed Dialing **ON**	***81**	Speed Dialing **OFF**	**#81**
Call Return	***60**		
Config Warm Line Number (*70*yyyyy* where *yyyyy* = number to call)	***70**		
Config Speed Dialing Number (*71*xyyyyy* where *x* = speed dial key and *yyyyy* = number to call)	***71**		
Config IP Dialing (*72*xxx***xxx***xxx***xxx***yyyy* where *xxx* = IP address and *yyyy* = optional port number)	***72**		
Set Call Forward Number (Wait for 3 short confirmation tones before hanging up)	***73**		
Access Voicemail	***86**		

Appendix E Voice prompt configuration

Chapter contents

Introduction

The VOE101 provides the ability to review and set the network configuration parameters using the handset of an attached analog telephone handset.

By default from the factory, DHCP is enabled and an IP address is not configured.

The VOE101 must be power cycled or reloaded after changing any of the network settings. Menu selection item *Network Status* will not reflect setting changes until after VOE101 is reloaded or power cycled.

Note Configuration of these settings can result in loss of connectivity to the VOE101 on the local LAN.

Accessing the voice prompt

Dial **** from the analog handset to reach the main menu.

Existing voice prompt configuration

On hook the analog phone.

Access Code	Main Menu Selection	Announcement/ Function	Voice Prompt	User input
* * * *	Main Menu	Plays main menu selections	Black Box VOE101 Configuration Main Menu	Enter selection code
100#	Network status	Plays DHCP setting, IP address, gateway IP address and IP network mask setting	100# Network status	None.
110#	DHCP setting	Enables or Disables DHCP	110# DHCP Settings	1# to enable DHCP 2# to disable DHCP or "#" to return to the main menu
120#	IP address setting	Sets IP address of the VOE101	120# Set IP Address	Use "*" to instead of ".", and "#" to end. Ex: 172*16*230*227# or "#" to return to the main menu
130#	Gateway setting	Sets the gateway router IP address	130# Set gateway router IP address	Use "*" to instant of ".", and "#" to end or "#" to return to the main menu
140#	Net mask setting	Set the IP network mask	140# Set IP network mask	Use "*" to instant of ".", and "#" to end or "#" to return to the main menu
150#	Reload	Immediately reloads the VOE101	None	None

BLACK BOX
NETWORK SERVICES

1000 Park Drive • Lawrence, PA 15055-1018 • 724-746-5500 • Fax 724-746-0746