User Manual

V^{2}IU 4350 Converged Network Appliance

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Introduction

The V^{2}IU 4350 Converged Network Appliance

The V^{2}IU 4350 is an intelligent, all-in-one networking solution for enterprises and service providers. It reduces costs by simplifying the deployment, management and security of converged voice, video and data networks. The 4350 provides the following important functions for converged networks:

T1 Wide Area Network (WAN) Access Router

The 4350 provides an integrated T1 CSU/DSU for small and medium office connectivity.

Note The 4350E is designed for use in Australia, New Zealand, Korea and Japan. The T1 port on the 4350E is functionally disabled through software. The T1 port should not be connected to the telco network.

- Fully integrated CSU/DSU
- T1 support
- Fractional T1 support
- Layer 2 protocol support for: HDLC, Cisco HDLC (cHDLC), PPP, MLPPP, MLPPPoFR, Frame Relay
- On-board RJ-48 connector for easy direct connection
- T1/E1 framer and transceiver
  - B8ZS/HDB3 zero suppression
  - Response to Inband Loop codes (ANSI)
  - Manual payload loop through the GUI
- External transmit clock input and receive clock output headers
- Timing: internal or external (loop times from the network)
- Provides long haul CSU or short haul DSU signaling

- Meets FCC part 68 protection requirements

Security

A stateful packet inspection firewall is used in combination with a VoIP application layer gateway to provide comprehensive “media-aware” security. The 4350 also supports IPSec for secure site-to-site networking.

VoIP

The 4350 resolves NAT/FW traversal problems for SIP, MGCP and H.323 traffic. It allows a single public IP address to be used for multiple VoIP clients.

Quality of Service

The 4350 maximizes WAN link utilization while optimizing voice quality using prioritization and shaping.

Call Quality Monitoring

Passive call quality monitoring for each SIP or MGCP voice call includes statistics needed to enforce SLAs and resolve networking problems that negatively affect call quality.

Future-proof Scalability

The 4350 is a powerful, flexible platform that can be deployed initially as a low-cost WAN access router and then licensed through software for more advanced VoIP features and increased call performance. It is the ideal platform for service providers offering DIA, hosted VoIP and managed security services or enterprises migrating to converged voice and data networks.

Feature Summary

- VoIP
  - SIP, MGCP (for voice) and H.323 (for video) application layer gateway enables a single public IP address to be used for multiple VoIP endpoints
- QoS
  - Class based queuing/prioritization

  - Diffserv marking and policing
  - Traffic shaping
  - VoIP call admission control prevents oversubscription of priority queue

- Security
  - Stateful packet inspection firewall
  - VoIP aware firewall dynamically provisions and closes UDP ports used for VoIP calls
  - IPSec: 3DES, SHA-1
  - NAT/PAT server hides enterprise LAN topology

- Passive Call Quality Monitoring
  - Per call statistics include mean opinion score (average and minimum), jitter, latency, packet loss and much more
  - Alarms for poor MOS scores
  - Active call count indicators

Front Panel LEDs

The LEDs display real-time information for key functions of the 4350. They are as follows:

	LED Label	Activity	Description
A	Power	Off	Power switch is off (or no power from the AC outlet)
		Green	Power is supplied to the unit
B	Status	Off	The unit could not boot up because of self test failure
		Green	Self test passed
		Flashing Green	Configuration is being written to permanent storage or an upgrade is in progress
C	T1	Off	The T1 is in an alarm state and not synchronized
		Green	T1 is in sync and no alarms are reported

Back Panel

The back panel of the 4350E contains the following connections:

	Name	Description
A	Power Connector	Accepts the plug from the supplied power cord to connect the unit to an AC power source
B	10/100 Mbps LAN Ports	4 x Local Area Network (LAN) ports to connect the IP phones or an Ethernet switch. Unit can also be configure through any of these ports using the web interface
C	USB Ports	2 x USB ports (Not Used)
D	Ethernet WAN Port	Can be used as an alternative to the T1 WAN ports. This port is typically used when connecting the 4350 to an existing T1/E1 WAN router, cable or xDSL modem.
E	Management Console Port	This port is used to establish a local console session with the 4350 using a VT100 terminal or emulation program. The cable required is a straight-through 8-wire cable. The serial port uses a baud rate of 9600, 8 data bits, 1 stop bit and no parity. This port is used for debug or local diagnostic purposes only. Primary configuration of the 4350 is performed from a web browser as covered in VOS User Manual
F	Erase Button	Press the Erase button twice in quick succession to reset the CLI password to the factory default. Press the Erase button three times in quick succession to reset the Polycom V2IU 4350 to the factory default. This will reset all passwords and erase all prior configurations.The default LAN address will now be 192.168.1.1.
G	T1/E1 WAN	T1/E1 port is used to connect to a data T1 line. The device at the far end of the line is a router or other device expecting TCP/IP data. Individual DS-0 channels on T1 are not used to carry uncompressed voice.

2

Getting Started

Physical Installation

The V^{2}IU 4350 is designed for desktop, rack or wall-mount installation. Please observe the following guidelines when installing the system:

- Never assume that the AC cord is disconnected from a power source. Always check first.
- Always connect the AC power cord to a properly grounded AC outlet to avoid damage to the system or injury.

Ensure that the physical location of the installation has adequate air circulation and meets the minimum operating conditions as provided in the environmental specifications for the system.

Warning Secure the power supply using a fastener or nearby shelf so that it does not hang from the power connector.

Required Tools and Materials

The following items are required:

- If the unit is to be mounted on the wall:
  - 1 Flat or Phillips screw driver
  - 2 round or flat head Phillips or slotted screws - 1 ½ inch long
  - 2 hollow wall anchors
- If the unit will be mounted in a shelf:
  - 1 Flat or Phillips screw driver

- If the T1/E1 port will be used to connect to WAN:
  - T1 cable to connect the T1/E1 port to a T1 line.

Warning To reduce the risk of fire, use only 26 AWG or larger wire (e.g. 24, 22, 20, etc.) to connect the T1 port on your unit to an RJ-45 jack.

- Ethernet cables to connect the LAN ports to LAN switches or to individual IP phones. They can also be used to connect the Ethernet WAN port to a WAN router, a Cable Modem or a DSL Modem, if T1/E1 port is not being used to connect to the WAN.

Desktop Installation

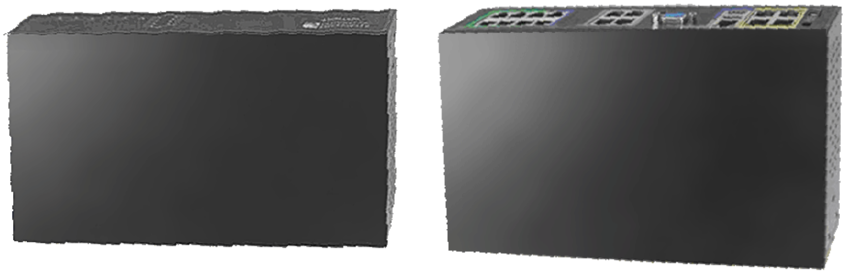
1. Remove the 4350 and accessories from the shipping container.
2. Place the 4350 on a flat, dry surface such as a desktop, shelf or tray.
3. Connect the power and network cables to the appropriate ports on the back of the system.

Caution To reduce the risk of fire, use only 26 AWG or larger wire (e.g. 24, 22, 20, etc.) to connect the T1 port on your unit to an RJ-45 jack.

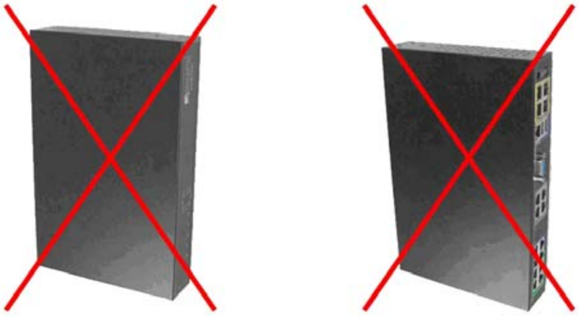
Wall-Mount Installation

1. The 4350 can be wall-mounted using the two mounting brackets on the bottom of the appliance. We recommend using two round or pan head screws.
2. Install two screws 5.9" (150 mm) horizontally apart on a wall or other vertical surface. The screws should protrude from the wall so that you can fit the appliance between the head of the screw and the wall. If you install the screws in drywall, use hollow wall anchors to ensure that the unit does not pull from the wall due to prolonged strain from the cable and power connectors.
3. Remove the 4350 and accessories from the shipping container.

4. Mount the 4350 on the wall as shown below.



Do not mount the 4350 on the wall as shown below.



1. Connect the power and network cables to the appropriate ports on the back of the system.

Warning Secure the power supply using a fastener or nearby shelf so that it does not hang from the power connector.

Caution To reduce the risk of fire, use only 26 AWG or larger wire (e.g. 24, 22, 20, etc.) to connect the T1 port on your unit to an RJ-45 jack.

Rack-Mount Installation

You can mount the 4350 in a shelf by using the rack-mount kit supplied with the product.

1. Attach the ear mounts to both sides of 4350 with the screws.
2. Attach the 4350 with the ear mounts to the shelf by screwing the ear mounts to the shelf with screws.

Connecting the Power and Cables

The 4350 comes with an AC power cord and power adapter for connecting the unit to the AC outlet.

Warning

Always connect the AC power cord to an AC outlet suitable for the power supply that came with the unit in order to reduce the risk of damage to it.

- Connect one end of the AC power cord to the power adapter and the other one to the AC outlet.
- Connect plug from the power adapter to the Power Connector on the 4350. Sometimes a little force is necessary to get the plug properly positioned

Caution

Secure the power adapter using a fastener or tie wrap to nearby shelf so that it does not hang from the power connector.

If you are connecting to the WAN using the T1, then connect the T1 cable to the T1/E1 port and the other end of the cable to the T1 line. Enable the T1 port from the Network Configuration Menu. In this case, the Ethernet WAN port will not be available.

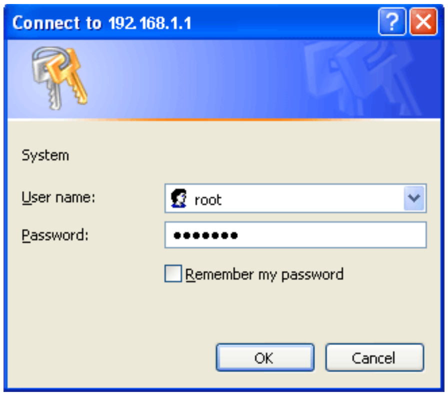
If you are connecting to a WAN router, cable modem or DSL modem, then connect the Ethernet cable to the Ethernet WAN port on 4350 and the other end to the WAN device.

Administration of the 4350

The 4350 is configured using a web browser such as Internet Explorer or Netscape Navigator. The 4350 is shipped with a pre-configured IP address for its LAN port of 192.168.1.1. To connect to the 4350, do the following:

1. Connect a PC using an IP address of 192.168.1.2 and subnet mask of 255.255.255.0 to LAN port 4 of the 4350.

2. Launch a web browser on the PC and enter the URL string: 192.168.1.1. Press **Return**.



Connect to 192.168.1.1

System

User name: root

Password: •••••••

□ Remember my password

3. Enter the username root and the password default to log into the system. The 4350 main configuration menu appears.

Note For secure management of your network, be sure to change the default userid and password as described under *Change the Administration Password*.

4. Configure the system using the information provided in Chapter 3.

3

Configuring the V^{2}IU 4350

The V^{2}IU 4350 is a flexible, easy to use converged network appliance that provides many critical networking functions for IP based voice, video and data. It can be installed in several different topologies:

- At the customer premise for IP Centrex and hosted video applications
- At the station side of enterprise IP PBXs
- At the trunk side of enterprise IP PBXs
- At the public/private IP address boundary for enterprise video applications

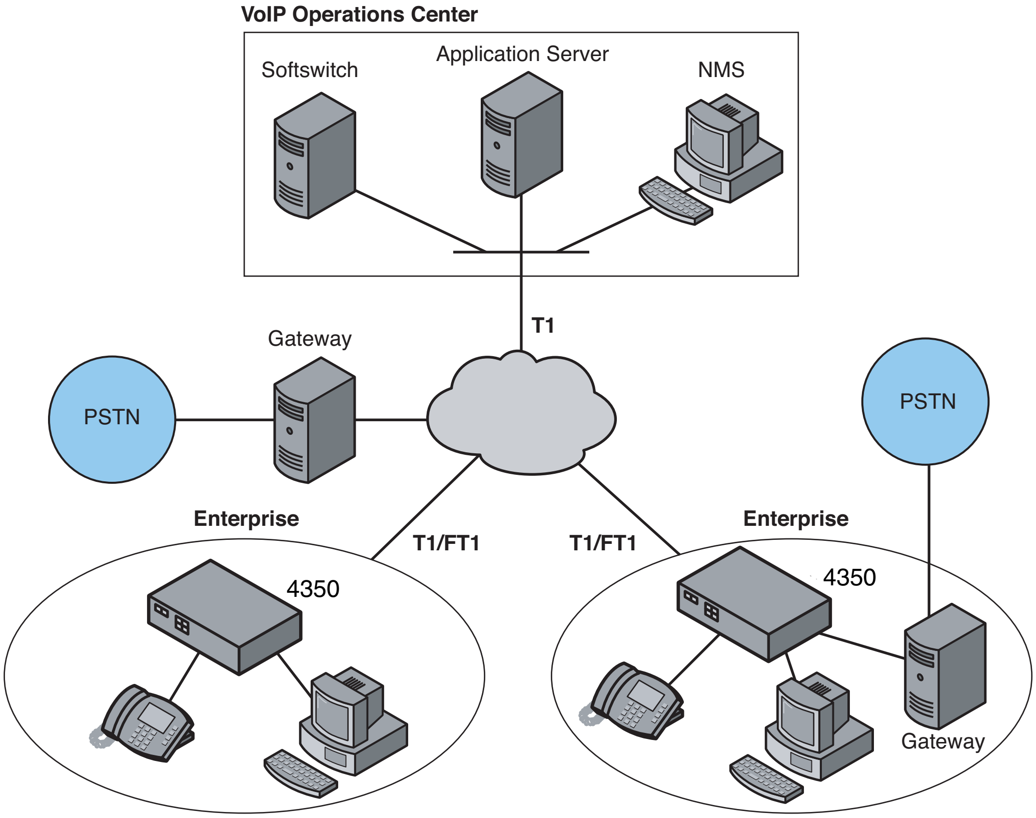
Most users will follow the steps provided in the “Configuring The Systems Settings” section of this manual to initially connect the 4350 into their IP network. The remainder of the configuration can be different based on the application, VoIP topology and presence of other networking equipment such as firewalls or DHCP servers. In general, however, the steps used to configure the 4350 are:

Step	Task
1	System configuration
2	VoIP configuration
3	Data networking configuration
4	Firewall configuration
5	Traffic management configuration

Some of the steps are optional depending on your particular application. We have provided configuration guidelines below for each of the application types supported by the 4350.

Configuration Guide For IP Centrex Applications

A typical 4350 installation for an IP Centrex application requires no external router or firewall. The 4350 WAN port is connected directly to the T1/E1 line and the LAN port(s) are connected directly to enterprise devices and/or Ethernet switches.



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VoIP signaling is performed in the service provider network via a softswitch and the 4350 acts as a proxy for the voice devices installed in the enterprise LAN. In this configuration a single public IP address is used to proxy for all of the IP phones and to route to multiple PC's installed on the LAN.

The 4350 performs the following functions in this application:

- WAN/LAN IP routing.
- Traffic shaping and priority queuing to guarantee high quality voice traffic. These mechanisms protect voice and data traffic from contending for the same network resources to guarantee low latency and the highest call quality possible for VoIP traffic. At the same time they ensure the best utilization of WAN bandwidth by enabling data traffic to burst up to full line rate in the absence of voice calls. Precedence is automatically given to traffic coming from IP phones and other devices using the 4350's Application Layer Gateway function.

- NAT/PAT translation for IP phones and PC's. This allows a single public IP address to be used on the WAN link to represent all of the private IP addresses assigned to the LAN IP phones and PC's.
- Static NAT entries. This enables the customer to use a WAN public IP address for data servers (web, mail, ftp, etc.) connected behind the 4350. These servers can then be configured with private IP addresses for additional security.
- A "VoIP" aware firewall. A full Layer 7 gateway for voice traffic and a stateful packet inspection firewall for data traffic.
- Call Admission Control (CAC). CAC uses a deterministic algorithm to decide when there are insufficient network resources available to adequately support new calls and then return the equivalent of a "fast busy" to new call requests.
- DHCP server and TFTP relay. These features are used to simplify and expedite the IP configuration of phones and PC's. This also includes VoIP signaling gateway information (MGCP, SIP, and H.323.
- Call quality monitoring (using MOS, jitter, latency, packet loss and much more) and test tools.

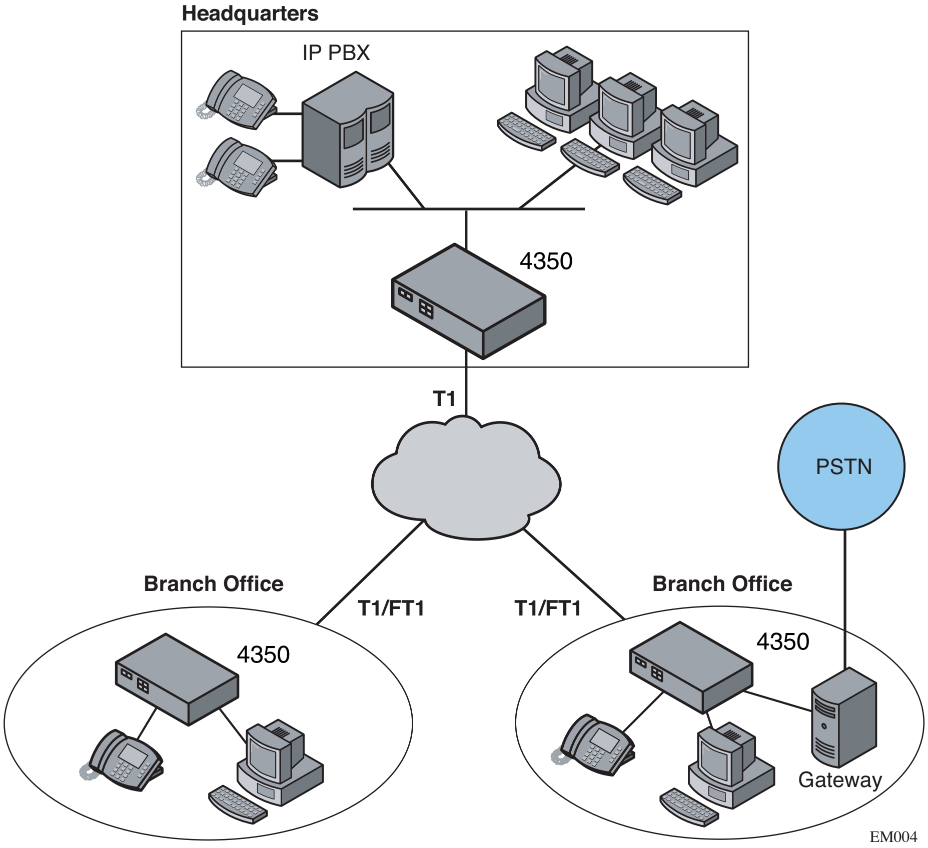
Configuration Outline

Task	Subtask	Configure For IP Centrex Application?
System Configuration	configure LAN/WAN interface	Yes
	set ethernet link rate	Optional
	enable the DHCP server	Optional but recommended
	configure SNMP	Optional
VoIP Configuration	enable the VoIP ALG	Yes
	configure a VoIP subnet route	Optional
Data Networking Configuration	dynamic NAT	Optional but recommended
	static NAT	Optional
	static IP routing	Optional
Firewall Configuration	enable the data firewall	Yes
	configure basic settings	Optional
	configure advanced settings	Optional
Traffic Management Configuration	enable traffic shaping	Yes

enable Call Admission Control	Optional

Configuration Guide For Station Side IP PBX Applications

Most private enterprise VoIP networks use an IP PBX at the corporate headquarters location to provide voice switching between headquarters, branch offices and the PSTN. The 4350 is used in these environments to securely connect branch office employees to the IP PBX installed in the corporate headquarters location.



The installation of an 4350 on the station side of an enterprise IP PBX is very similar to the IP Centrex application above. The branch office is connected to the corporate network using a private T1/E1 link connected directly to the WAN port of the 4350. The LAN port(s) of the 4350 are connected directly to enterprise devices and/or Ethernet switches.

The IP PBX in the corporate headquarters location performs VoIP signaling and the 4350 acts as a proxy for the voice devices installed at the branch office. Please note that in the configuration the 4350 located at the Headquarters location is acting as a WAN router only. The 4350s installed at the brand offices perform the following functions in this application:

- WAN/LAN IP routing.
- Traffic shaping and priority queuing to guarantee high quality voice traffic. These mechanisms protect voice and data traffic from contending for the same network resources to guarantee low latency and the highest call quality possible for VoIP traffic. At the same time they ensure the best utilization of WAN bandwidth by enabling data traffic to burst up to full line rate in the absence of voice calls. Precedence is automatically given to traffic coming from IP phones and other devices using the 4350's Application Layer Gateway function.
- NAT/PAT translation for IP phones and PC's. This allows a single IP address to be used on the WAN link to represent all of the private IP addresses assigned to the LAN IP phones and PC's.
- A "VoIP" aware firewall. A full layer 7 gateway for voice traffic and a stateful packet inspection firewall for data traffic.
- Call Admission Control (CAC). CAC uses a deterministic algorithm to decide when there are insufficient network resources available to adequately support new calls and then return the equivalent of a "fast busy" to new call requests.
- DHCP server and TFTP relay. These features are used to simplify and expedite the IP configuration of phones and PC's. This also includes VoIP signaling gateway information (MGCP, SIP, and H.323).
- Call quality monitoring and test tools.

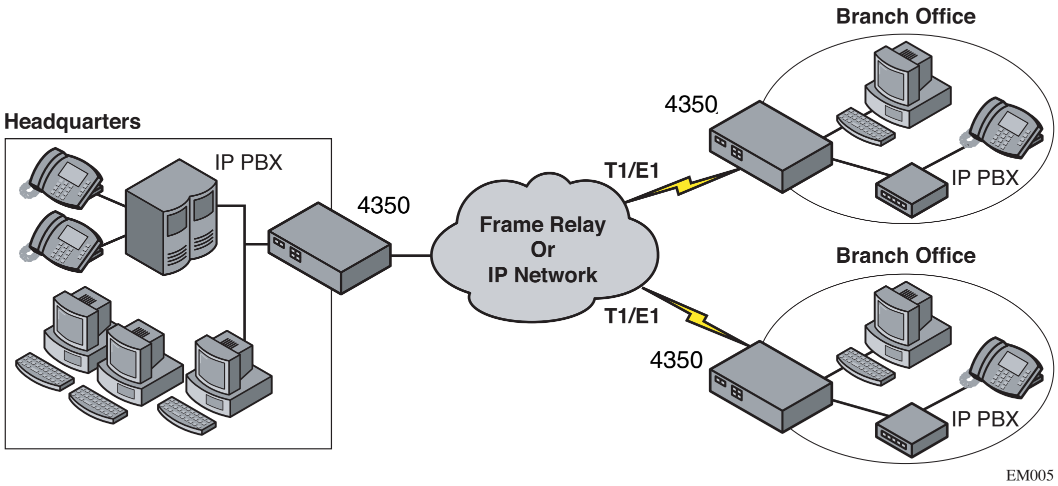
Configuration Outline

Task	Subtask	Configure For Station Side IP PBX Application?
System Configuration	configure LAN/WAN interface	Yes
	set ethernet link rate	Optional
	enable the DHCP server	Optional but recommended
	configure SNMP	Optional
VoIP Configuration	enable the VoIP ALG	Yes
	configure a VoIP subnet route	Optional

Data Networking Configuration	dynamic NAT	Optional but recommended
	static NAT	Optional
	static IP routing	Optional
Firewall Configuration	enable the data firewall	Yes
	configure basic settings	Optional
	configure advanced settings	Optional
Traffic Management Configuration	enable traffic shaping	Yes
	enable Call Admission Control	Optional

Configuration Guide For Trunk Side IP PBX Applications

Companies with existing IP-based WAN links for inter-office voice and data communications can use the 4350 as a traffic shaper to meet the stringent jitter, latency and packet loss requirements for toll quality voice. The 4350 is deployed at the edge of the WAN in both headquarters and branch office locations, as shown below.



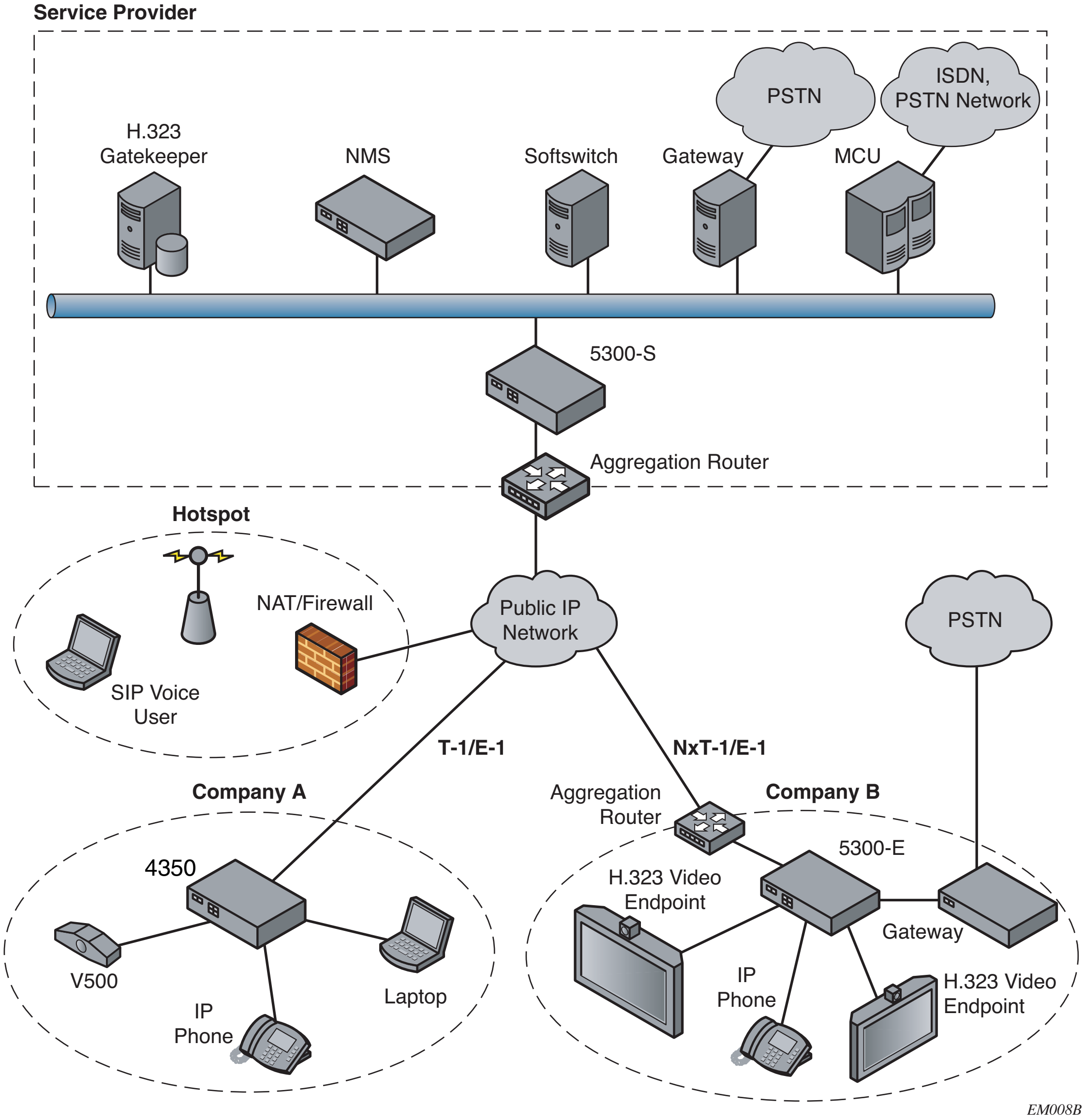
The 4350 performs WAN/LAN IP routing and traffic management functions in this application. In particular, it provides prioritization to ensure voice packets are not delayed or dropped while allowing data traffic to use all remaining bandwidth.

Configuration Outline

Task	Subtask	Configure For Trunk Side IP PBX Application?
System Configuration	configure LAN/WAN interface	Yes
	set ethernet link rate	Optional
	enable the DHCP server	Not required
	configure SNMP	Optional
VoIP Configuration	enable the VoIP ALG	Not required
	configure a VoIP subnet route	Not required
Data Networking Configuration	dynamic NAT	Not required
	static NAT	Not required
	static IP routing	Not required
Firewall Configuration	enable the data firewall	Not required
	configure basic settings	Not required
	configure advanced settings	Not required
Traffic Management Configuration	enable traffic shaping	Yes
	enable Call Admission Control	Not required

Configuration Guide For Hosted Video Applications

A typical 4350 installation for hosted video applications is depicted in the diagram below. In this scenario, the 4350s are used to connect all of the video endpoints to the Gatekeeper. The video endpoints should be configured to point to the LAN address of the 4350 as the Gatekeeper and the 4350 will proxy RAS and call setup messages to the Gatekeeper



The 4350 is installed at the customer premises and is used as a demarcation point for the video service by providing the following functions:

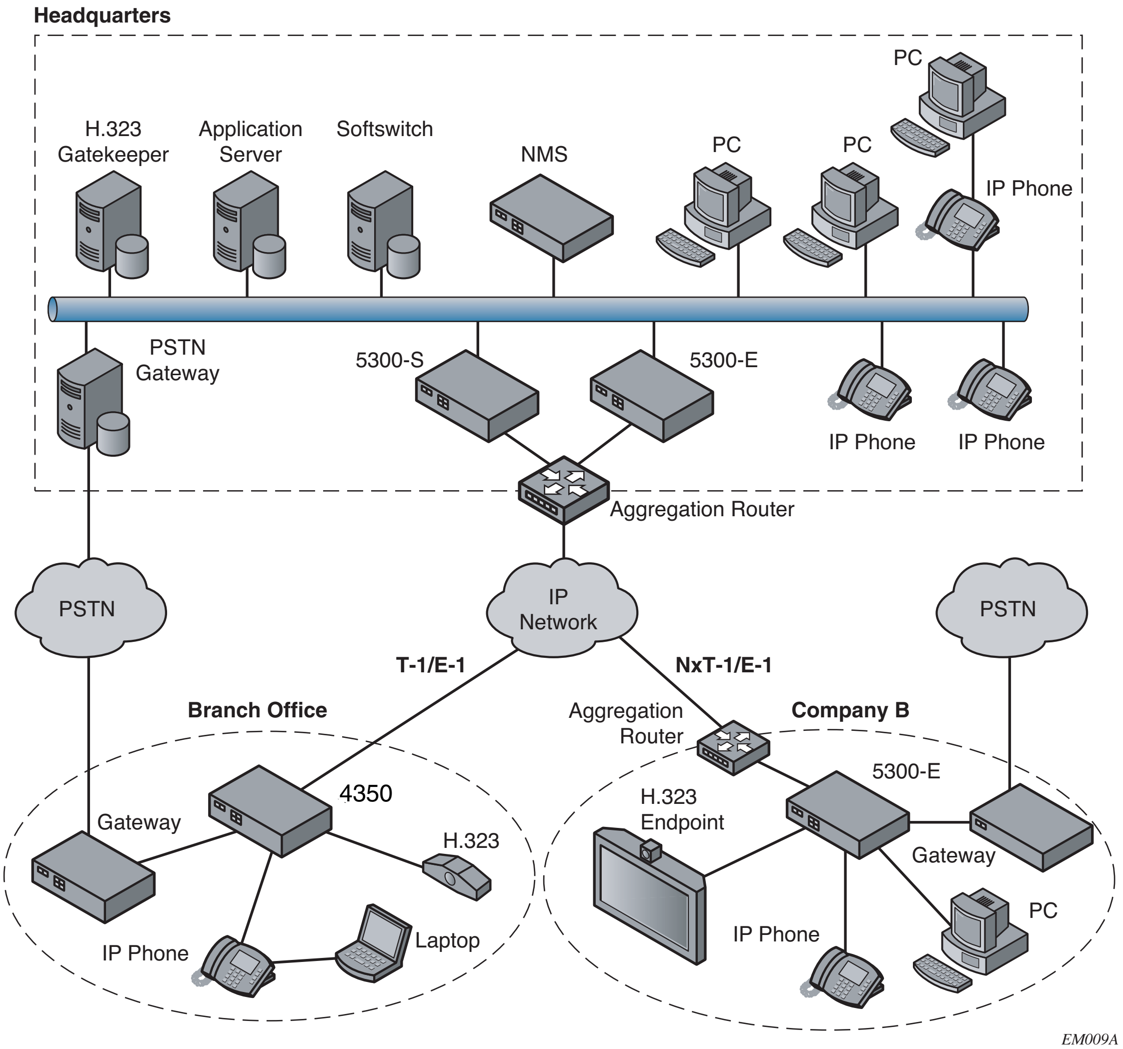
- WAN/LAN IP routing.

- Traffic shaping and priority queuing to guarantee high quality video traffic. These mechanisms protect video and data traffic from contending for the same network resources to guarantee low latency and the highest call quality possible for voice and video traffic. At the same time they ensure the best utilization of WAN bandwidth by enabling data traffic to burst up to full line rate in the absence of video calls. Precedence is automatically given to traffic coming from video endpoints and other devices using the 4350's Application Layer Gateway function.
- Video NAT/PAT translation for video endpoints and PC's. This allows a single IP address to be used on the WAN link to represent all of the private IP addresses assigned to the LAN video endpoints and PC's.
- A video aware firewall. A full layer 7 gateway for video traffic and a stateful packet inspection firewall for data traffic
- Call Admission Control (CAC). CAC uses a deterministic algorithm to decide when there are insufficient network resources available to adequately support new video calls and then return the equivalent of a "fast busy" to new call requests.

Task	Subtask	Configure For Hosted Video Applications?
System Configuration	configure LAN/WAN interface	Yes
	set ethernet link rate	Optional
	enable the DHCP server	Optional
	configure SNMP	Optional
VoIP Configuration	enable the VoIP ALG	Yes
	configure a VoIP subnet route	Optional
Data Networking Configuration	dynamic NAT	Optional but recommended
	static NAT	Optional
	static IP routing	Optional
Firewall Configuration	enable the data firewall	Yes
	configure basic settings	Optional
	configure advanced settings	Optional
Traffic Management Configuration	enable traffic shaping	Yes
	enable Call Admission Control	Optional

Configuration Guide For Enterprise Video Applications

A typical 4350 installation for enterprise video applications is depicted in the diagram below. In this scenario, the 4350s are used to connect all of the video endpoints to the Gatekeeper. The video endpoints should be configured to point to the LAN address of the 4350 as the Gatekeeper and the 4350 will proxy RAS and call setup messages to the Gatekeeper.



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The 4350 is installed at the private/public IP address boundary and provides the following functions:

- WAN/LAN IP routing.
- Traffic shaping and priority queuing to guarantee high quality video traffic. These mechanisms protect video and data traffic from contending for the same network resources to guarantee low latency and the highest

call quality possible for voice and video traffic. At the same time they ensure the best utilization of WAN bandwidth by enabling data traffic to burst up to full line rate in the absence of video calls. Precedence is automatically given to traffic coming from video endpoints and other devices using the 4350's Application Layer Gateway function.

- Video NAT/PAT translation for video endpoints and PC's. This allows a single IP address to be used on the WAN link to represent all of the private IP addresses assigned to the LAN video endpoints and PC's.
- A video aware firewall. A full layer 7 gateway for video traffic and a stateful packet inspection firewall for data traffic
- Call Admission Control (CAC). CAC uses a deterministic algorithm to decide when there are insufficient network resources available to adequately support new video calls and then return the equivalent of a "fast busy" to new call requests.

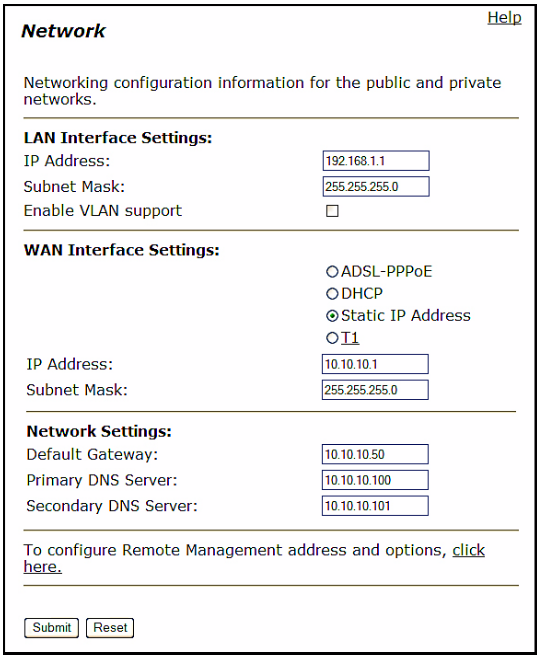
Task	Subtask	Configure For Hosted Video Applications?
System Configuration	configure LAN/WAN interface	Yes
	set ethernet link rate	Optional
	enable the DHCP server	Optional
	configure SNMP	Optional
VoIP Configuration	enable the VoIP ALG	Yes
	configure a VoIP subnet route	Optional
Data Networking Configuration	dynamic NAT	Optional but recommended
	static NAT	Optional
	static IP routing	Optional
Firewall Configuration	enable the data firewall	Yes
	configure basic settings	Optional
	configure advanced settings	Optional
Traffic Management Configuration	enable traffic shaping	Yes
	enable Call Admission Control	Optional

System Configuration

This section explains how to configure the 4350 to function in your IP network. You will configure the T1/E1 WAN interface, Ethernet interfaces, network addresses, DNS settings, default gateway, SNMP settings and change the administrative password.

1. Physically connect to the 4350 as described in Administration of the 4350 on page 2-4.

   A browser-based configuration GUI should appear, as shown here.



Help

Network

Networking configuration information for the public and private networks.

LAN Interface Settings:

IP Address:	192.168.1.1
Subnet Mask:	255.255.255.0
Enable VLAN support	□

WAN Interface Settings:

○ ADSL-PPPoE
○ DHCP
● Static IP Address
○ T1

IP Address:	10.10.10.1
Subnet Mask:	255.255.255.0

Network Settings:

Default Gateway:	10.10.10.50
Primary DNS Server:	10.10.10.100
Secondary DNS Server:	10.10.10.101

To configure Remote Management address and options, click here.

2. Select the Network entry in the Configuration Menu.

Configure the LAN Interface

The 4350 provides an integrated 4 port 10/100 Mbps ethernet switch that can be optionally configured to support 802.1q VLANs. Integrated VLAN support simplifies the integration of the 4350 with existing VLAN-based networks. The 4350 is able to receive 802.1q-tagged packets from a downstream VLAN switch and appropriately route and process them per its firewall rules. Packets received from the WAN are placed in the appropriate VLAN based on IP address routing.

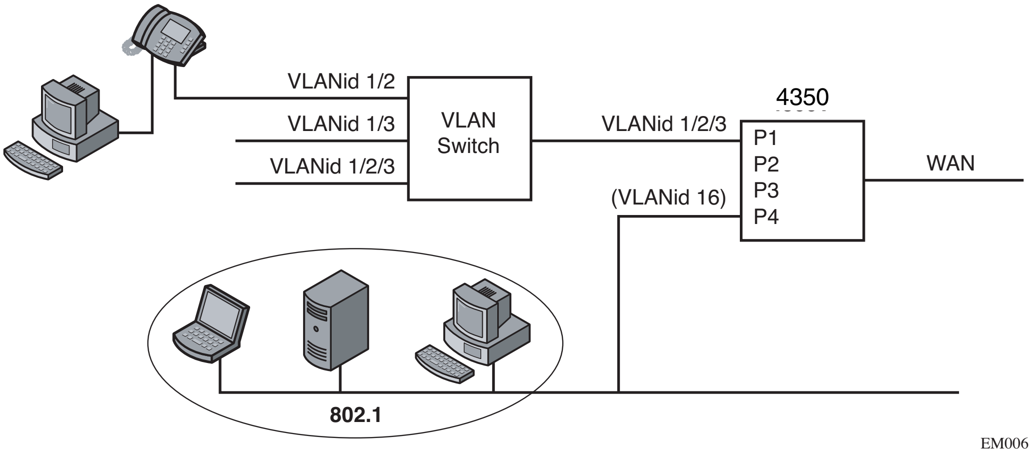
By default VLANs are not enabled and a single IP address is used for all 4 ethernet ports. The configuration of this address is as follows:

1. Enter the IP Address.
2. Enter the Subnet Mask (e.g. 255.255.255.0).

3. Press Submit.

Configuring VLANs in the 4350

As depicted in the diagram below, VLANs are used to connect the 4350 to an Ethernet switch that has been configured to use VLANs.



EM006

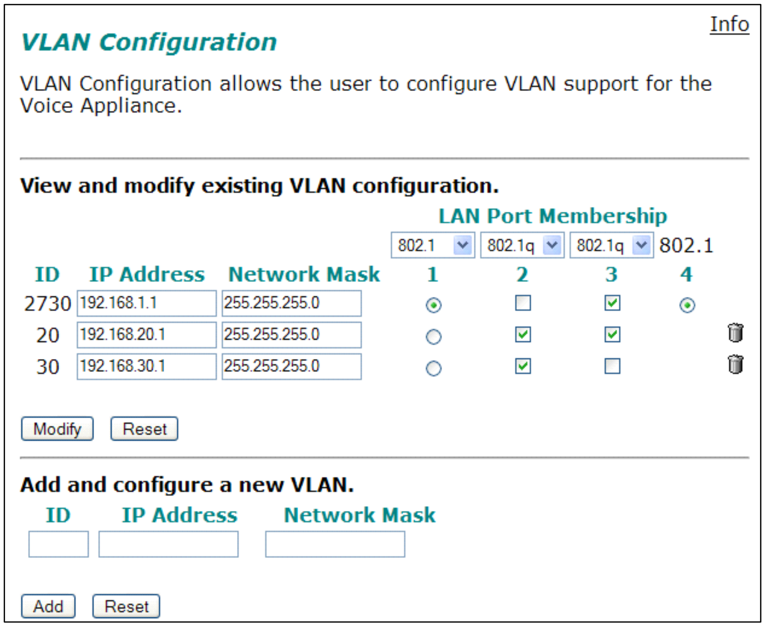
Typically, all VoIP devices are placed in the same VLAN while data devices are placed in a different VLAN. This is to ensure priority treatment of the VoIP traffic on the LAN. Note that the 4350 does not require VLANs to prioritize VoIP traffic; prioritization is determined by the VOS Application Layer Gateway, regardless of VLAN. Some important notes about VLANs:

- A physical LAN port will operate in either 802.1 or 802.1q mode, not both simultaneously
- The 4350 supports up to 16 VLANs
- A unique IP Subnet is assigned to each VLAN
- You can associate one or more VLANs to each LAN port operating in 802.1q mode
- Traffic within a VLAN is switched among all ports with membership
- Traffic between VLANs is routed by the 4350
- The 4350 ALG can only be assigned to one VLAN id
  - Only ALG traffic is prioritized over the WAN
  - Other non-VoIP traffic in the same VLAN will not receive priority treatment
- A DHCP server can be enabled/disabled per VLAN
- Cisco Discovery Protocol is not supported

- 802.1p is not currently supported

1. Select the Network link.
2. Select Enable VLAN support.
3. Press Submit.

Caution Be careful when changing a port from 802.1 to 802.1q mode. Any 802.1 devices connected to that port (such as your management PC!) will loose access to the 4350. Port 4 is only able to receive 802.1 frames, so a PC can always be connected to this port if the configuration of the other ports is unknown.



VLAN Configuration

VLAN Configuration allows the user to configure VLAN support for the Voice Appliance.

View and modify existing VLAN configuration.

			LAN Port Membership				
			802.1	802.1q	802.1q	802.1	
ID	IP Address	Network Mask	1	2	3	4	
2730	192.168.1.1	255.255.255.0	◉	☐	☑	◉	
20	192.168.20.1	255.255.255.0	○	☑	☑		🗑
30	192.168.30.1	255.255.255.0	○	☑	☐		🗑

Add and configure a new VLAN.

ID	IP Address	Network Mask

4. Select System.
5. Select VLAN Configuration.
6. Adjust LAN Port Membership drop-down boxes to specify 802.1 or 802.1q mode, as desired. Press Modify.

   If changing modes, the radio-buttons or checkboxes will change from one style to the other.
7. Under Add and configure a new VLAN enter a new VLAN ID, the 4350's IP address within this VLAN, and the Network Mask. Press Add.

   A new VLAN entry will be added to the VLAN Configuration above.
8. Depending on the mode of a physical port, assign it to one or more VLANs:

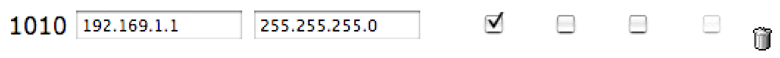
- 802.1 mode: Assign the port to any ONE VLAN.
- 802.1q mode: Assign the port to any number of VLANs

Perform steps 1 through 6 above for each VLAN you wish to create.

Modify an Existing VLAN Configuration

1. Select the Network link.
2. Select VLAN Settings.
3. Change the desired settings.
4. Press the Modify to modify the VLAN. The Reset button will restore the input area being modified to its previous value.

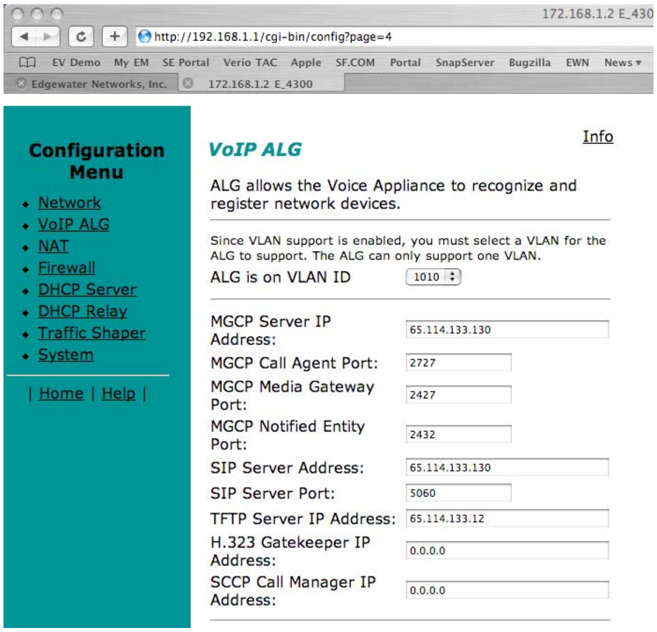
Delete an Existing VLAN Configuration



1010 192.169.1.1 255.255.255.0 ☑ □ □ □

1. Select the Network link.
2. Select VLAN Settings.
3. Press the trash can icon next to the VLAN you wish to delete.

Assign the 4350's ALG to your Priority VLAN



VoIP ALG

ALG allows the Voice Appliance to recognize and register network devices.

Since VLAN support is enabled, you must select a VLAN for the ALG to support. The ALG can only support one VLAN.

ALG is on VLAN ID: 1010

MGCP Server IP Address:	65.114.133.130
MGCP Call Agent Port:	2727
MGCP Media Gateway Port:	2427
MGCP Notified Entity Port:	2432
SIP Server Address:	65.114.133.130
SIP Server Port:	5060
TFTP Server IP Address:	65.114.133.12
H.323 Gatekeeper IP Address:	0.0.0.0
SCCP Call Manager IP Address:	0.0.0.0

Once you have completed your VLAN configuration you must assign the 4350 ALG to the VLAN containing your VoIP phones.

1. Select the VoIP ALG from the main configuration menu.
2. Use the drop down menu to assign the ALG to the VLAN ID containing your VoIP phones.
3. Press Submit.

Configure the WAN Interface

The 10/100 Ethernet WAN port is configured as follows:

1. Select ADSL-PPPoE if you want to connect to Internet using ADSL and your ISP has given PPPoE username and password. Press Submit. You will be prompted to enter username and password, enter these and press Submit again.
2. Select DHCP if you want to get WAN side IP address using DHCP server available in WAN side of the network. Press Submit.
3. Select Static IP address if you want to manually assign the IP address configuration to the ethernet WAN interface.
4. Enter the IP Address.
5. Enter the Subnet Mask (e.g. 255.255.255.0).

6. Enter the Default Gateway. This is usually the upstream router's IP address. Packets destined for IP networks not known to the 4350 are forwarded to the default gateway for handling.
7. Enter the Primary DNS Server. The DNS server is used by the 4350 to resolve domain names to IP addresses. The value entered into this field is provided to IP devices that use the 4350 as a DHCP server. The 4350 VoIP ALG also uses it if domain names are used instead of IP addresses to identify signaling and/or TFTP servers (see the section entitled "Configuring the VoIP ALG" for more details).
8. Enter the Secondary DNS Server. This server will be used in the event that the primary DNS server is not reachable.
9. Press Submit.

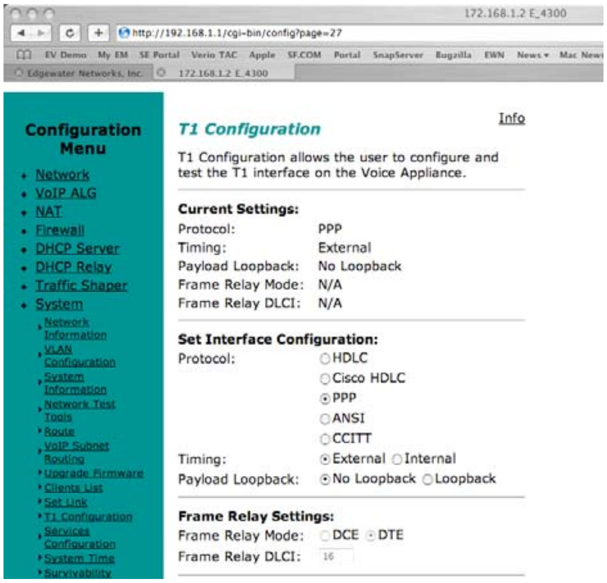
To enable the T1 interface:

1. Select Network.
2. Select the T1 radio button.
3. Select Submit.

To configure the T1 parameters:

1. Select Network.
2. Select the T1 link next to the radio button to proceed to the T1 Configuration page.

   The T1 Configuration menu will display, as shown here.



T1 Configuration

T1 Configuration allows the user to configure and test the T1 interface on the Voice Appliance.

Current Settings:

Protocol:	PPP
Timing:	External
Payload Loopback:	No Loopback
Frame Relay Mode:	N/A
Frame Relay DLCI:	N/A

Set Interface Configuration:

Protocol:	○ HDLC ○ Cisco HDLC ● PPP ○ ANSI ○ CCITT
Timing:	● External ○ Internal
Payload Loopback:	● No Loopback ○ Loopback

Frame Relay Settings:

Frame Relay Mode:	○ DCE ● DTE
Frame Relay DLCI:	16

The 4350 supports a wide range of T1/E1 Layer 2 configuration parameters. The specific values you will need must be supplied by the WAN provider.

Each of the 4350's configurable parameters are described below.

Protocol

Display and set the T1 Layer 2 protocol. Supported protocols are:

- HDLC
- Cisco HDLC
- PPP
- ANSI (Frame Relay)
- CCITT (Frame Relay)

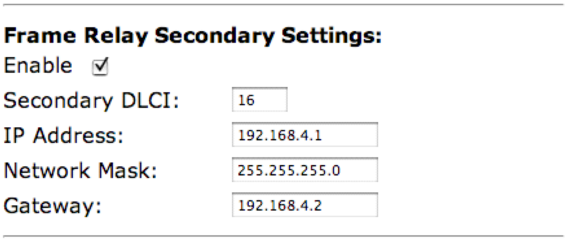
1. Select the desired T1 protocol.
2. Press Submit.

Frame Relay Mode and DLCI

When the Protocol is one of ANSI or CCITT, then additional Frame Relay configuration parameters are required.

The Frame Relay Mode is usually set to DTE for the customer premises.

The Frame Relay DLCI is set by the WAN provider and identifies the far-end device across the Frame Relay network. This DLCI can also be used to carry voice traffic only by enabling the Secondary DLCI for data.



Frame Relay Secondary Settings:

Enable ☑

Secondary DLCI: 16

IP Address: 192.168.4.1

Network Mask: 255.255.255.0

Gateway: 192.168.4.2

Most installations will use a single DLCI for both voice and data traffic. However, in instances where the network will provide a different quality of service based on DLCI number it is desirable to place all voice traffic on one DLCI and then configure a second DLCI for data. In this case, the Secondary DLCI is configured as follows:

1. Select Network.
2. Select the T1 link next to the radio button to proceed to the T1 configuration page.
3. Select Enable in the Frame Relay Secondary Settings section of the page.
4. Enter the Secondary DLCI, IP Address, Network Mask and Gateway for the data traffic using the Secondary DLCI.

Timing

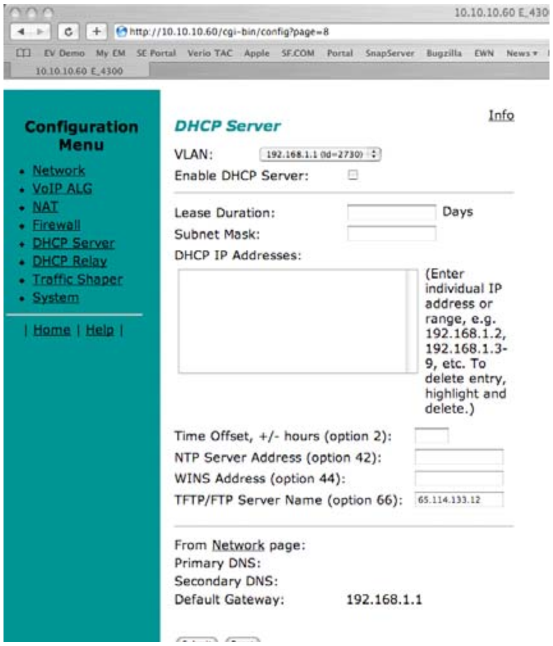
Display and set the clock timing source for the T1/E1 interface. The timing can be either derived from the network (External) or provided to the T1 interface by the V^{2}IU (Internal). With a carrier-provided T1, the timing is usually derived from the network (External, the default setting).

Warning Mismatched timing modes can result in WAN connectivity but with intermittent data loss.

Payload Loopback

Display and set the loopback setting. During T1 line testing the local interface can be set to Loopback to allow the network provider to verify connectivity and line quality. For normal operation the setting should always be No Loopback (the default setting).

Configure the DHCP Server



DHCP Server

VLAN: 192.168.1.1 (id=2730)

Enable DHCP Server: □

Lease Duration: Days

Subnet Mask:

DHCP IP Addresses: (Enter individual IP address or range, e.g. 192.168.1.2, 192.168.1.3-9, etc. To delete entry, highlight and delete.)

Time Offset, +/- hours (option 2):

NTP Server Address (option 42):

WINS Address (option 44):

TFTP/FTP Server Name (option 66): 65.114.133.12

From Network page:

Primary DNS:

Secondary DNS:

Default Gateway: 192.168.1.1

The 4350 can act as a DHCP server granting IP addresses to PCs, workstations, servers or voice devices (IP phones, IADs or softphones) connected to its LAN interfaces. DHCP is a protocol that enables IP devices to obtain temporary or permanent IP addresses (out of a pool) from centrally administered servers.

The user can configure blocks of IP addresses, a default gateway, DNS servers, NTP server address, Time offset from NTP value, WINS address and TFTP/FTP server name that can be served to the requesting IP devices.

In addition the 4350 will provide its LAN IP address in DHCP user options 150 and 151 for use by IP phones. Some IP phones use these values for configuration of their TFTP server and MGCP control server addresses.

Note The DHCP server in the 4350 should not be used if a DHCP server already exists in the same subnet as the 4350. Also, it is recommended that you assign static IP addresses for common-access devices such as network printers or fax machines.

You can also enable or disable the 4350 DHCP server on a per VLAN basis.

1. Select DHCP Server.
2. If you are using VLANs select the desired VLAN ID from the drop down menu.
3. The default value for the DHCP server is disabled. Click the top checkbox to enable or disable the internal DHCP server (default is disabled). If you are using VLANs select the desired VLAN ID.
4. Enter the Lease Duration.

   The lease duration is the amount of time in days that an IP device may use an assigned IP address before requesting that it be renewed. The default value is 7 days and the valid range of input is 1 to 30 days.
5. Enter the Subnet Mask.

   This is the subnet mask that will be sent via DHCP to the requesting IP devices.
6. Enter the DHCP IP Addresses.

   This is the pool of IP addresses that will be provided to the requesting IP devices. You can enter both individual IP addresses or a range of addresses using the following format:

   192.168.1.2 (single address)

   192.168.1.4-10 (address range 192.168.1.4 through 192.168.1.10)

Note The range format can only be used for class C addresses (those with a subnet mask of 255.255.255.0).

7. Enter the Time Offset (DHCP user option 2).
8. Set the time offset in hours from UTC for your local location. This value is optional; if supplied, it will be delivered to clients.
9. Enter the NTP Server Address (DHCP user option 42).

   This is the IP address of a Network Time Server. This value is optional; if supplied, it will be delivered to clients.
10. Enter the WINS Address.

Note If you are not using WINS leave this field blank.

The Windows Internal Naming Service (WINS) is a service that keeps a database of computer name-to-IP address mappings so that computer names used in Windows environments can be mapped to IP addresses. The WINS Address is the IP address of the WINS server in your network. This value will be delivered to clients.

1. Enter the TFTP/FTP Server Name (DHCP user option 66).

   Some IP phones use this setting to locate the TFTP or FTP servers which contain the phone software image used during boot. By default this option is the same as the TFTP server on the VoIP ALG page.

2. Primary and Secondary DNS

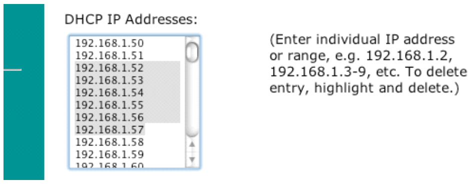
   The primary and secondary DNS values come from those set under the WAN interface configuration, see Configure the WAN interface. These values will be delivered to clients.

3. Default Gateway

   The default gateway is automatically set to the 4350's LAN address, see Configure the LAN interface. This value will be delivered to clients.

4. Press Submit.

Delete a DHCP IP Address



DHCP IP Addresses:

192.168.1.50
192.168.1.51
192.168.1.52
192.168.1.53
192.168.1.54
192.168.1.55
192.168.1.56
192.168.1.57
192.168.1.58
192.168.1.59
192.168.1.60

(Enter individual IP address or range, e.g. 192.168.1.2, 192.168.1.3-9, etc. To delete entry, highlight and delete.)

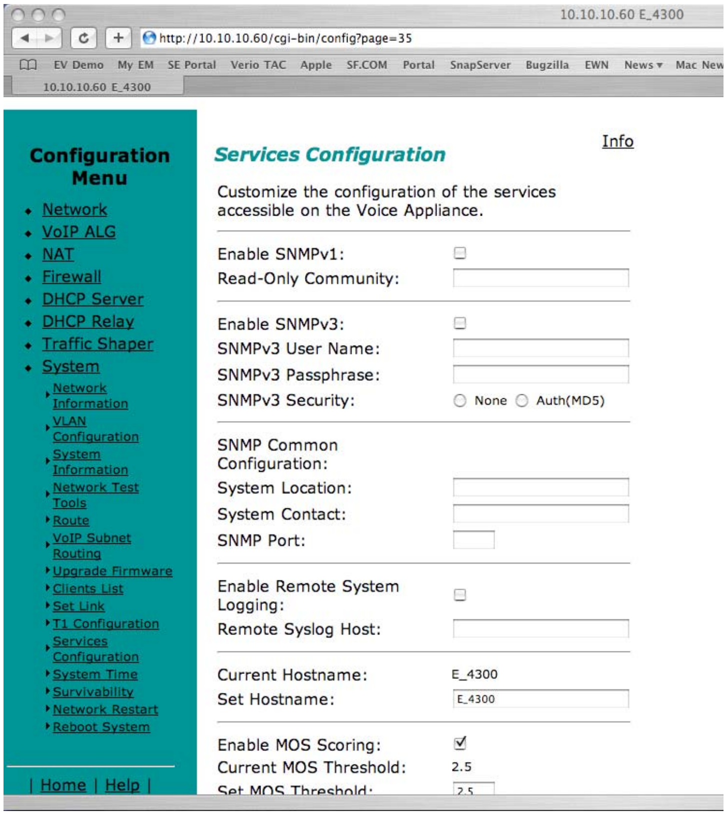
1. Select DHCP Server.
2. To delete an IP address or a range of IP addresses highlight an entry or range of entries in the DHCP IP Addresses list and press the Delete key on your keyboard.
3. Press Submit.

Disable The DHCP Server

4. Select DHCP Server.
5. Uncheck\ the Enable DHCP Server checkbox.
6. Press Submit.

Configure Hostname, SNMP and Remote Logging

The 4350 can be managed remotely by an SNMP network management system such as HP Openview. The 4350 supports SNMPv1 or SNMPv3 and MIB-II (RFC1213). All MIB-II variables are read only. The MIB variables sysContact and sysLocation are set by the web GUI.



Services Configuration

Customize the configuration of the services accessible on the Voice Appliance.

Enable SNMPv1: □
Read-Only Community:

Enable SNMPv3: □
SNMPv3 User Name:
SNMPv3 Passphrase:
SNMPv3 Security: ○ None ○ Auth(MD5)

SNMP Common Configuration:
System Location:
System Contact:
SNMP Port:

Enable Remote System Logging: □
Remote Syslog Host:

Current Hostname: E_4300
Set Hostname: E_4300

Enable MOS Scoring: ☑
Current MOS Threshold: 2.5
Set MOS Threshold: 2.5

Messages generated by the 4350 can be sent to a remote log server.

The configuration screen is reached through the Configuration Menu:

1. Select System.
2. Select System Overview.
3. Select Services Configuration.

Configure SNMP

1. Select the Enable SNMP v1 or v3 checkbox. If using SNMPv1 enter the Read-Only Community. If using SNMPv3 enter the User Name, Passphrase and Security method.

2. Enter the System Location.

   This is a comment string that can be used to indicate the physical location of the 4350. By default, no value is set.

3. Enter the System Contact.

   This is the administrative contact information for the 4350. By default, no value is set.

4. Enter the SNMP Port.

   This is the port that the 4350 uses for SNMP communications with the network management system. The default is 161.

5. Press Submit.

Disable SNMP

1. Select System.
2. Select System Overview.
3. Select Services Configuration.
4. Uncheck the Enable SNMP checkbox.
5. Press Submit.

Configure Remote System Logging

The 4350 can be configured to log system messages to an external syslog server.

1. Select the Enable Remote System Logging checkbox.
2. Enter the IP address of the Remote Syslog Host.

   By default messages are sent to the remote host on port 514. This port can be changed by using the syntax ADDRESS:PORT.

3. Press Submit.

Disable Remote System Logging

1. Select System.
2. Select System Overview.
3. Select Services Configuration.
4. Uncheck the Enable Remote System Logging checkbox.
5. Press Submit.

Configure a local Hostname

A locally configured hostname is useful for remote management. This name can appear as the identifier string for the 4350 on a system management console.

>> Enter a **host name** in the field provided.

Enable Mean Opinion Scoring (MOS)

The 4350 produces useful statistics on a per call basis that can be written to syslog. These include MOS, jitter, latency, packet loss and much more.

1. Select System.
2. Select System Overview.
3. Select Services Configuration.
4. Select Enable MOS.

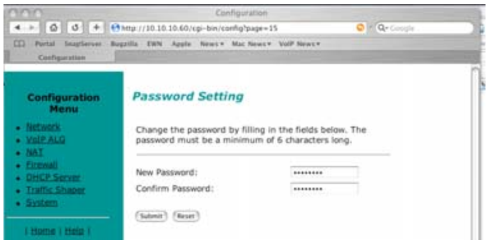
Set MOS Threshold

You can define a minimum MOS value in the 4350 such that a message will be sent to syslog when the measured MOS value drops below the minimum. This is useful when for monitoring a particular location for call quality problems and enables pro-active resolution of problems that negatively affect call quality.

1. Select System.
2. Select System Overview.
3. Select Services Configuration.
4. Enter the minimum MOS threshold in the Set MOS threshold field.
5. Press Submit

Change the Administration Password

We strongly recommend that you change the default password for the root administrative account using the following steps:



Password Setting

Change the password by filling in the fields below. The password must be a minimum of 6 characters long.

New Password: ********

Confirm Password: ********

Note

The new password must be between 6 and 20 characters in length. Any combination of alpha and numeric characters is accepted.

1. Enter the password you chose in step C again in the Confirm Password to ensure that there were no mistakes in the initial entry.
2. Press Submit.

Read-only User

This feature works by creating a new user with read-only access to the system. All information is displayed in a non-changeable form. Information changed in entry boxes cannot be submitted. In fact, most **Submit** and **OK** buttons are not visible.

Note: You must have administrator privileges and log in as an administrator to change read-only user.

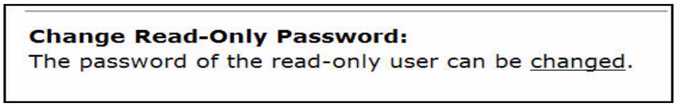
Enabling a Read-only User

To enable a read-only user, use the following steps:

1. Using the configuration graphical user interface, from the Configuration Menu on the left-hand side, click **Network**.

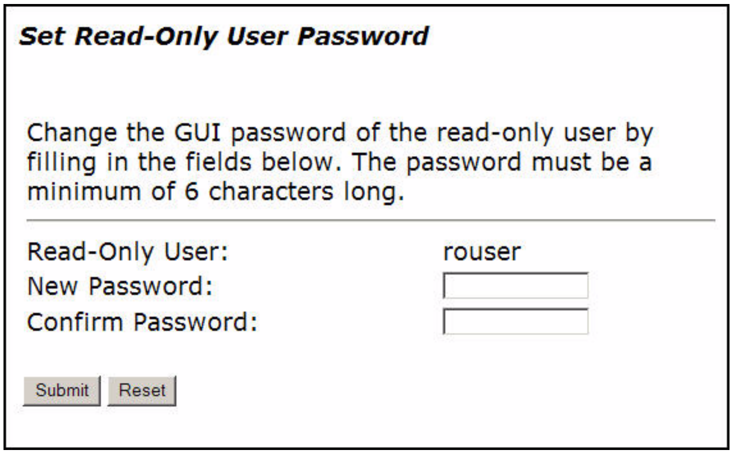
   Note: You must have administrator access and log in as an administrator to change read-only user.

2. Scroll down to the area of the screen shown below.



Change Read-Only Password:
The password of the read-only user can be changed.

3. Click changed. The following window screen appears:



Set Read-Only User Password

Change the GUI password of the read-only user by filling in the fields below. The password must be a minimum of 6 characters long.

Read-Only User:	rouser
New Password:	
Confirm Password:	

Note: All open web browsers must be closed when you change between administrative user “root” and read-only “rouser.”

4. Enter a new password. The password must be a minimum of six characters long.

4. Re-enter the new password to confirm it.

5. Click **Submit.**

Now when you access the system using this user name (rouser) and password, all fields are read-only.

Subinterfaces

The Subinterfaces feature allows a system administrator to assign additional IP addresses to interfaces. These are sometimes referred to as aliases or loopback interfaces. An additional address may be assigned to the system’s WAN interface to support, for example, another management IP address.

How Subinterfaces Works

A common use for subinterfaces is forwarding a public subnet. A subinterface may be created to support a subnet forwarded through the Polycom V²IU 4350. When forwarding a subnet through the Polycom V²IU 4350, it is necessary to assign an address for this subnet to the system to act as the subnet's gateway. To configure forwarding rules, use the **Forwarding Rules** submenu under the **Firewall** configuration link.

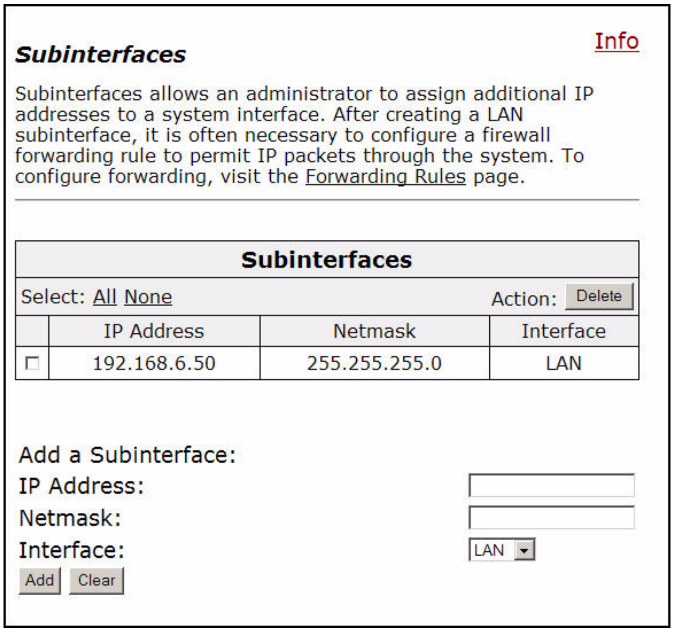
When applied to the WAN/Provider interface, these addresses are protected by the same firewall policy that is applied to the WAN/Provider address. Several other features in the system automatically create Subinterfaces. VRRP (if supported) and Static NAT automatically create Subinterfaces.

When viewing the Network Information page, Subinterfaces are designated in the Interface Information section with the device name and number, separated by a colon (for example, eth0:100).

Configuring Subinterfaces

To configure subinterfaces, use the following steps:

1. Using the configuration graphical user interface, from the Configuration Menu on the left-hand side, click **Network**.
2. Click **Subinterfaces**. The window shown below opens.



Subinterfaces

Info

Subinterfaces allows an administrator to assign additional IP addresses to a system interface. After creating a LAN subinterface, it is often necessary to configure a firewall forwarding rule to permit IP packets through the system. To configure forwarding, visit the Forwarding Rules page.

Subinterfaces

Select: All None — Action: Delete

	IP Address	Netmask	Interface
□	192.168.6.50	255.255.255.0	LAN

Add a Subinterface:

IP Address:	
Netmask:	
Interface:	LAN

3. On this screen, complete the following information:

- **IP Address** is the address to be assigned to the subinterface.
- **Netmask** is the network mask to use for the address. If several addresses are applied to an interface and these addresses are in a common network, they must use a common subnet. The system does not support supernetting.
- **Interface** is the port where the subinterfaces will be configured.

4. When you have finished entering this information, click **Add**. The following popup appears:
5. Click **OK**. The new subinterfaces entry appears on the Subinterfaces window in the list area.

ToS Byte Setting

Since the Internet itself has no direct knowledge of how to optimize the path for a particular application or user, the IP protocol provides a limited facility for upper layer protocols to convey hints to the Internet Layer about how the trade-offs should be made for the particular packet. This facility is the “Type of Service” or ToS facility.

ToS settings allow the service provider to prioritize time sensitive traffic, such as voice plus video to ensure minimized packet loss and delay through their network. When providing end-to-end QOS, it is important that the voice plus video traffic be placed in the correct queues to deliver a higher QOS than regular traffic. Regular traffic, that is not time sensitive, can be delayed with little or no indication to the user, while the slightest delay in voice plus video can cause auditable differences. The ToS byte setting helps prioritize traffic going to the WAN so a provider can prioritize the traffic correctly in its network.

Although the ToS facility has been a part of the IP specification since the beginning, it has been little used in the past. However, the Internet host specification now mandates that hosts use the ToS facility. Additionally, routing protocols (including OSPF and Integrated IS-IS) have been developed which can compute routes separately for each type of service. These new routing protocols make it practical for routers to consider the requested type of service when making routing decisions.

How the ToS Byte Setting Works

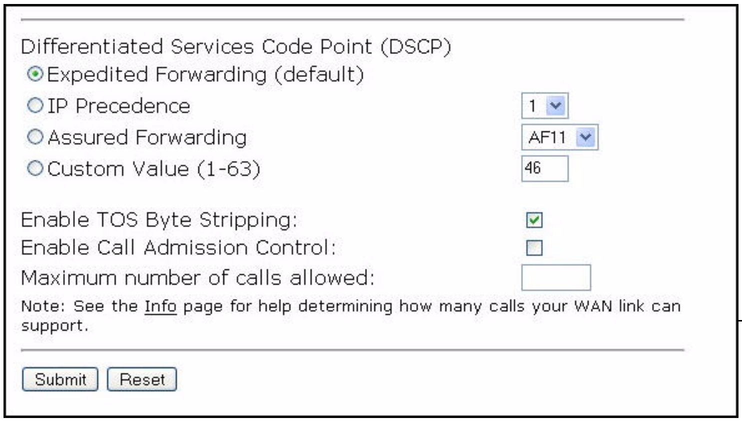
For all RTP traffic (voice and video), the Polycom V²IU 4350 marks the ToS byte in the IP header as “High Priority,” and strips (set to 0) the ToS byte for all other traffic. Unchecking the “Enable ToS Byte Stripping” option means that the ToS byte will not be stripped from non-RTP traffic, but will remain unchanged.

Note: For most situations, you should leave this setting as it is. Only change it if your provider indicates that you should do so.

Viewing or Changing the ToS Byte Setting

To view or change the ToS byte setting, use the following steps:

1. Using the configuration graphical user interface, from the Configuration Menu on the left-hand side, click **Traffic Shaper**.
2. Scroll down the area of the screen shown below.



Differentiated Services Code Point (DSCP)

● Expedited Forwarding (default)

○ IP Precedence 1

○ Assured Forwarding AF11

○ Custom Value (1-63) 46

Enable TOS Byte Stripping: ☑

Enable Call Admission Control: □

Maximum number of calls allowed:

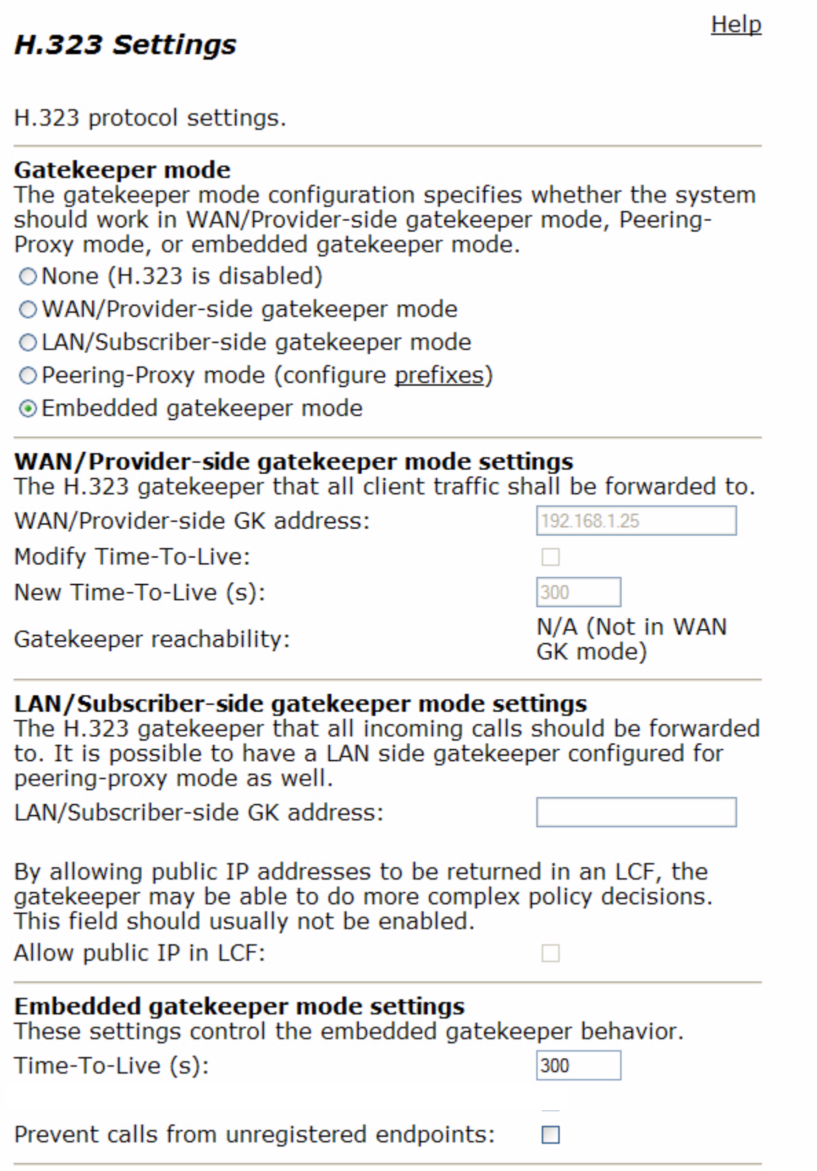
Note: See the Info page for help determining how many calls your WAN link can support.

3. For most situations, you should leave this setting as it is. Only change it if your provider indicates that you should do so. If your provider indicates that you need to change the ToS byte setting, that provider should also provide the other parameters required on this screen.

4. If you have changed the values, click **Submit** to activate the new settings.

H.323 Configuration

To access the H.323 Settings page, select **VoIP ALG > H.323** in the Configuration Menu.



H.323 Settings

Help

H.323 protocol settings.

Gatekeeper mode
The gatekeeper mode configuration specifies whether the system should work in WAN/Provider-side gatekeeper mode, Peering-Proxy mode, or embedded gatekeeper mode.

○ None (H.323 is disabled)
○ WAN/Provider-side gatekeeper mode
○ LAN/Subscriber-side gatekeeper mode
○ Peering-Proxy mode (configure prefixes)
● Embedded gatekeeper mode

WAN/Provider-side gatekeeper mode settings
The H.323 gatekeeper that all client traffic shall be forwarded to.

WAN/Provider-side GK address: 192.168.1.25
Modify Time-To-Live: □
New Time-To-Live (s): 300
Gatekeeper reachability: N/A (Not in WAN GK mode)

LAN/Subscriber-side gatekeeper mode settings
The H.323 gatekeeper that all incoming calls should be forwarded to. It is possible to have a LAN side gatekeeper configured for peering-proxy mode as well.

LAN/Subscriber-side GK address:

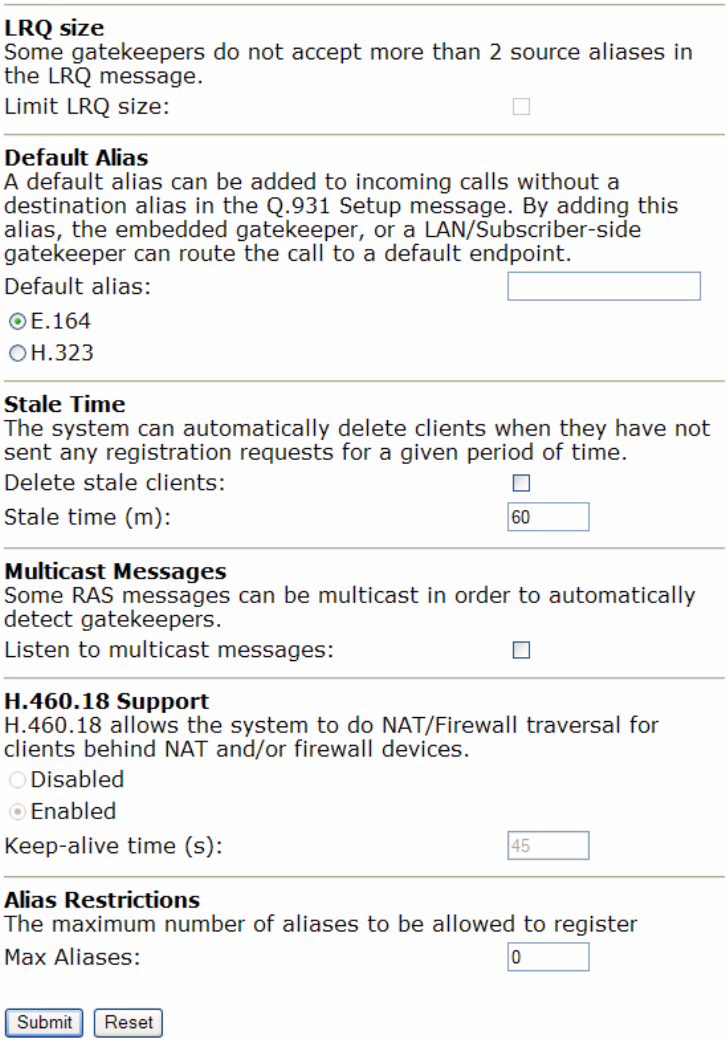
By allowing public IP addresses to be returned in an LCF, the gatekeeper may be able to do more complex policy decisions. This field should usually not be enabled.

Allow public IP in LCF: □

Embedded gatekeeper mode settings
These settings control the embedded gatekeeper behavior.

Time-To-Live (s): 300

Prevent calls from unregistered endpoints: □



LRQ size
Some gatekeepers do not accept more than 2 source aliases in the LRQ message.
Limit LRQ size: □

Default Alias
A default alias can be added to incoming calls without a destination alias in the Q.931 Setup message. By adding this alias, the embedded gatekeeper, or a LAN/Subscriber-side gatekeeper can route the call to a default endpoint.
Default alias:
● E.164
○ H.323

Stale Time
The system can automatically delete clients when they have not sent any registration requests for a given period of time.
Delete stale clients: □
Stale time (m): 60

Multicast Messages
Some RAS messages can be multicast in order to automatically detect gatekeepers.
Listen to multicast messages: □

H.460.18 Support
H.460.18 allows the system to do NAT/Firewall traversal for clients behind NAT and/or firewall devices.
○ Disabled
● Enabled
Keep-alive time (s): 45

Alias Restrictions
The maximum number of aliases to be allowed to register
Max Aliases: 0

The H.323 Settings page has the following areas:

- Gatekeeper Mode
- WAN/Provider-side gatekeeper mode settings
- LAN/Subscriber-side gatekeeper mode settings
- Embedded gatekeeper mode settings
- LRQ Size
- Default Alias
- Stale Time
- Multicast Messages

- H.460.18 Support
- Alias Restrictions

In the Gatekeeper mode area, select one of the following modes:

Item	Description
None	H.323 is disabled.
WAN/Provider-side gatekeeper mode	Specifies that the system will forward all client RAS messages to the gatekeeper. If this is selected, you must configure the settings in the WAN/Provider-side gatekeeper mode settings area.
LAN/Subscriber-side gatekeeper mode	Specifies that the system will act as a gatekeeper. If this option is selected, you must configure the settings in the LAN/Subscriber-side gatekeeper mode settings area.
Peering-Proxy mode	Allows calls to be forwarded to other endpoints based on the information sent from the endpoints. All the information about routing the call must be sent as part of the request or prefixes must be configured.
Embedded gatekeeper mode	Provides gatekeeper functions and accepts endpoint registrations. If this option is selected, you must configure the settings in the Embedded gatekeeper mode settings area.

If WAN/Provider-Side Gatekeeper mode is selected, you must configure the following parameters:

Item	Description
WAN/Provider-side GK address	Specifies the IP address of the gatekeeper
Modify Time-To-Live	Allows you to override the value for time-to-live returned by the gatekeeper before forwarding the response to the endpoint.
New Time-To-Live	Specifies how long an endpoint's registration should be valid.

If LAN/Subscriber-Side Gatekeeper mode is selected, you must configure the following parameters:

Item	Description
LAN/Subscriber-side GK address	Enter the IP address of the gatekeeper.
Allow public IP in LCF	Select the checkbox if the gatekeeper has been deployed with multiple outbound proxies and must decide which proxy to use based on the IP address returned in the LCF. This is an advanced configuration option and should usually not be selected.

If Embedded Gatekeeper is selected, you must configure the following parameters:

Item	Description
Time-to-Live(s)	Enter a time in seconds. This setting controls how long an endpoint's registration should be valid. At the end of this period the endpoint sends another registration request.
Prevent calls from unregistered endpoints:	Blocks unregistered LAN-side endpoints from making calls through the device.

In the LRQ Size area, you can limit the number of source aliases in a forwarded LRQ message to a maximum of two to allow interoperability with gatekeepers that cannot handle more than two source aliases.

In the Default Alias area, you can specify a default alias to be added to incoming calls without a destination message in the Q.931 Setup message. This alias allows the embedded gatekeeper or a LAN/Subscriber-side gatekeeper to route the call to a default endpoint. Enter a default alias and select one of the following types:

- E.164
- H.323

In the Stale Time area, you can arrange to delete clients that have not sent any registration requests for the specified interval. This area includes the following configurable parameters:

Item	Description
Delete stale clients	Select this checkbox to enable the stale timer feature.
Stale time (m)	Specify the length of the interval in minutes.

Some RAS messages can be multicast in order to automatically detect gatekeepers. In the Multicast Messages area, you can enable listening to multicast messages. This area includes the following configurable parameter:

Item	Description
Listen to multicast messages	Select this checkbox to enable listening to multicast messages.

In the H.460.18 Support area, you can configure H.460.18 support. This allows the system to do NAT/Firewall traversal for clients behind NAT or firewall devices. This area includes the following configurable parameters:

Item	Description
Disabled	Disables H.460.18 support.
Enabled	Enables H.460.18 support.
Keep-alive time(s)	Specifies the keep-alive time if H.460.18 support is enabled.

In the Alias Restrictions area, you can set a limit on the number of aliases that are allowed to register with the system. If this number is exceeded when a client tries to register, the registration is rejected. This area includes the following parameter:

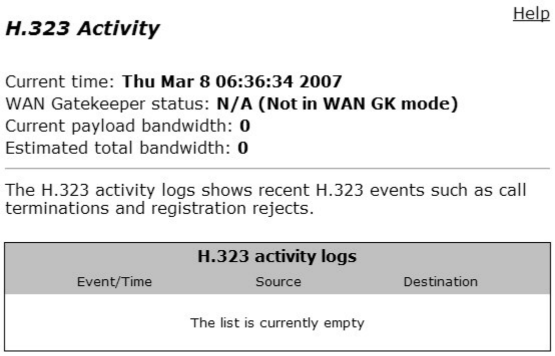
Item	Description
Max Aliases	Enter the maximum number of allowed aliases. If the value is set to 0, the maximum is not enforced.

The H.323 Settings page includes the following two buttons:

Item	Description
Submit	Applies the settings configured on this page.
Reset	Clears all fields and selections and allows you to enter new information.

H.323 Activity

To access the H.323 Activity page, select **VoIP ALG > H.323 Activity** in the Configuration Menu.



H.323 Activity

Current time: **Thu Mar 8 06:36:34 2007**
WAN Gatekeeper status: **N/A (Not in WAN GK mode)**
Current payload bandwidth: **0**
Estimated total bandwidth: **0**

The H.323 activity logs shows recent H.323 events such as call terminations and registration rejects.

H.323 activity logs

Event/Time	Source	Destination
The list is currently empty		

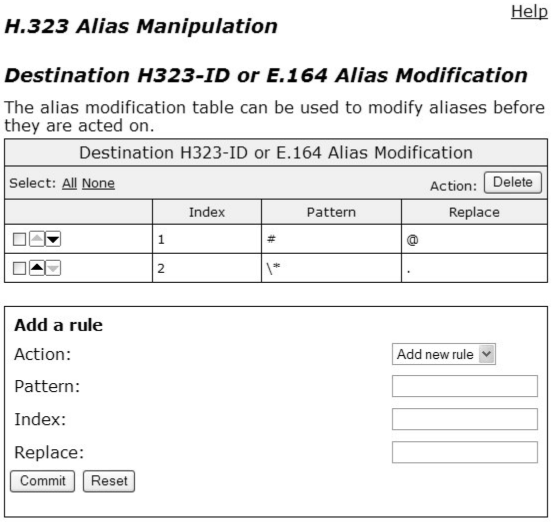
The H.323 Activity page is a read-only page that shows the following information:

- Current time
- WAN Gatekeeper status
- Current payload bandwidth
- Estimated total bandwidth
- Activity log of recent H.323 events

H.323 Alias Manipulation

Alias manipulation is performed immediately when a message (such as an ARQ, LRQ or a Setup) is received. Any matching pattern is replaced with the specified string, allowing you to replace characters or strings that are hard or impossible to dial on certain endpoints. Normal call look-up is performed following alias manipulation.

To access the H.323 Alias Manipulation page, select **VoIP ALG > H.323 >Alias Manipulation** in the Configuration Menu.



Help

H.323 Alias Manipulation

Destination H323-ID or E.164 Alias Modification

The alias modification table can be used to modify aliases before they are acted on.

Destination H323-ID or E.164 Alias Modification

Select: All None Action: Delete

	Index	Pattern	Replace
□ ▲▼	1	#	@
□ ▲▼	2	*	.

Add a rule

Action:	Add new rule
Pattern:	
Index:	
Replace:	

Commit Reset

This page includes the following areas:

Item	Description
Destination H323-ID or E.164 Alias Modification table	Lists alias manipulation rules. Rules are executed in the order in which they are listed. Use the arrows to move entries up and down, or use the Index field to specify where a new or edited rule falls in the list.
Add a rule	Allows you to add new prefixes to the Prefix Routing and Gatekeeping Neighboring table.

Item	Description
Action	Indicates whether the rule is to be added or edited.
Pattern	Specifies the pattern to be matched. See <l_link>“Regular Expressions” on page 39 for details on valid patterns.
Index	Determines the order in which the rule is scanned in the Destination H323-ID or E.164 Alias Modification table. To add a rule between two rules with consecutive indexes (n and m), use the higher index (m).
Replace	Specifies the string that will replace the matched pattern.

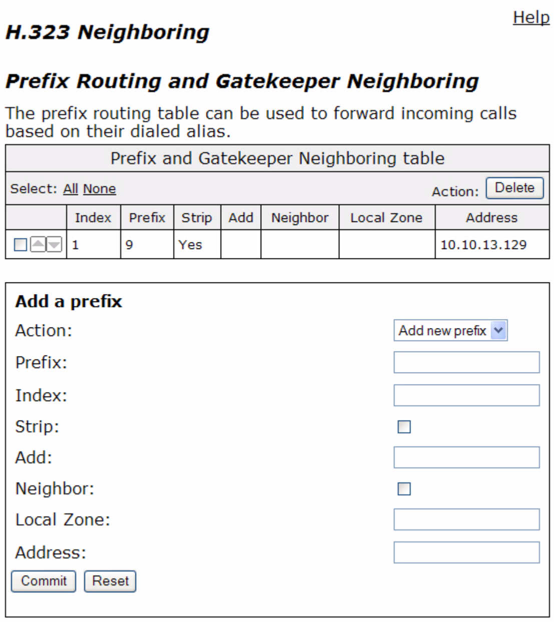
The H.323 Alias Manipulation page includes the following buttons:

Item	Description
Commit	Applies the settings configured on this page.
Reset	Clears all fields and selections and allows you to enter new information.

H.323 Neighboring

Neighboring and prefix routing can be used to route calls based on a matching prefix in the destination alias of the call. The call decision is made following alias manipulation and acts on the modified string, similar to other call lookup processes such as registered client look-up. Each prefix is associated with a domain name or IP address that is used in the event that the prefix matches.

To access the H.323 Neighboring page (formerly the Prefix Routing page), select **VoIP ALG > H.323 > Neighboring** in the Configuration Menu.



H.323 Neighboring

Prefix Routing and Gatekeeper Neighboring

The prefix routing table can be used to forward incoming calls based on their dialed alias.

Prefix and Gatekeeper Neighboring table

Select: All None

Action: Delete

	Index	Prefix	Strip	Add	Neighbor	Local Zone	Address
□ ▲ ▼	1	9	Yes				10.10.13.129

Add a prefix

Action:	Add new prefix
Prefix:	
Index:	
Strip:	□
Add:	
Neighbor:	□
Local Zone:	
Address:	

This page includes the following areas:

Item	Description
Prefix Routing and Gatekeeper Neighboring table	Lists rules for forwarding incoming calls based on their dialed alias. Rules are executed in the order in which they are listed. Use the arrows to move entries up and down, or use the Index field to specify where a new or edited rule falls in the list.
Add a prefix	Allows you to add new prefixes to the Prefix Routing and Gatekeeper Neighboring table.

Item	Description
Action	Indicates whether the rule is to be added or edited.
Prefix	Specifies the prefix pattern to be matched against the dialing string. See <l_link>“Regular Expressions” on page 39 for details on valid patterns.
Index	Determines the order in which the rule is scanned in the Prefix and Gatekeeper Neighboring table. To add a rule between two rules with consecutive indexes (n and m), use the higher index (m).
Strip	Indicates whether the matching prefix is stripped from the dialing string.
Add	Specifies a string to be prepended to the dialing string.
Neighbor	Determines whether a location request (LRQ) is sent when this prefix matches. • If enabled, the prefix becomes a neighboring statement. • If disabled, the incoming Q.931 Setup is forwarded to the given address without a preceding LRQ. This field is used for interoperability with other gatekeepers that may not accept a Setup without a preceding LRQ.
Local Zone	Provides compatibility with remote gatekeepers that are configured to accept LRQs only from sources that match its configured remote zone. If a gatekeeper is configured to accept requests only from a known source, enter the zone in this field.
Address	Specifies the IP address or domain name of the device to which the call is to be forwarded.

The H.323 Neighboring page includes the following buttons:

Item	Description
Commit	Applies the settings configured on this page.
Reset	Clears all fields and selections and allows you to enter new information.

Regular Expressions

Alias manipulation patterns and prefixes use regular expressions to match a string in the destination alias. A regular expression can be a string of literal characters to match or a set of special expressions.

Alias manipulation patterns can match a sub-string at any location and number of times within the alias. Prefixes are always searched from the left of the alias and cannot match a middle part or the end of the alias.

Regular expressions are listed in <l_link>Table 1 and <l_link>Table 2 lists some example expressions.

Table 1 Regular Expressions

Symbol	Description
.	Matches any single character.
[]	Matches any single character listed between the []. For example, [abc], [123]. If the characters are separated by a -, all characters between the two are matching, e.g. [a-z], [0-9]
()	Matches the literal string given, e.g. (abc)
|	Matches the block on either side of the |, e.g. a|b.
?	Matches 0 or 1 of the preceding block.
*	Matches 0 or more of the preceding block.
+	Matches 1 or more of the preceding block.
\	Escapes the special meaning of the next character.
{a}	Matches exactly 'a' numbers of the preceding block.
{a,}	Matches 'a' or more of the preceding block.
{a,b}	Matches between 'a' and 'b' (inclusive) of the preceding block.

Table 2 Example Regular Expressions

Expression	Description
100	Matches the string 100.
(555)?123	Matches 555123 or 123.
(408|555)	Matches 408 or 555.
555[0-9]{3}	Matches 555 followed by exactly 3 digits.
#	Matches the character '#'.
*	Matches the character '*'. Note that '*' by itself is a regular expression and must therefore be escaped with a '\' to match the character itself.

Forwarding Rules

Forwarding Rules allows a system administrator to forward data traffic for a subnet from one interface to another, overriding the Firewall’s default drop rules.

Allowing a subnet to be forwarded is commonly used when servers with public addresses are placed behind the system. Configuring the network in this way allows the system to manage and prioritize bandwidth, sharing it between the VoIP services and the servers.

How Forwarding Rules Works

When forwarding, one address from the forwarded range of addresses must be assigned to the rule's output interface. The Polycom V^{2}IU 4350 uses this address to act as a gateway router for the subnet. The address may be assigned using the Subinterfaces page.

Note: The subnet and forwarded addresses are not protected by the firewall.

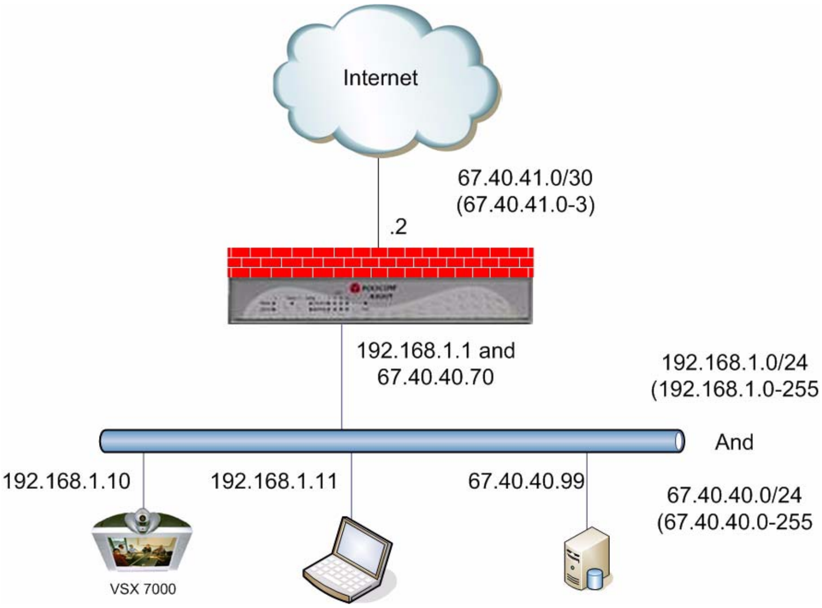
A similar method for forwarding traffic is provided by Proxy ARP. Proxy ARP is used to “bridge” addresses within a single subnet range from one interface to another. Often this is used to bridge and forward a public address to the protected side of the system without having to subnet the public address range. Proxy ARP does not require an additional gateway address on the system for the subnet, but does not allow port and protocol filtering for forwarded data.

Example

In this example:

- The ISP has supplied two separate subnets to the customer:
  - A small one (2 hosts) for the WAN link
  - A large one (254 hosts) for a bank of servers
- 67.40.41.2 is the WAN IP address for the Polycom V^{2}IU 4350
- NAT is a private IP range of 192.168.1.xxx using the WAN address for PCs and Phones
- On the LAN side of the Polycom V^{2}IU 4350 are the following:
  - Private IP subnet (192.168.1.xxx)
  - Public IP subnet (67.40.40.xxx)

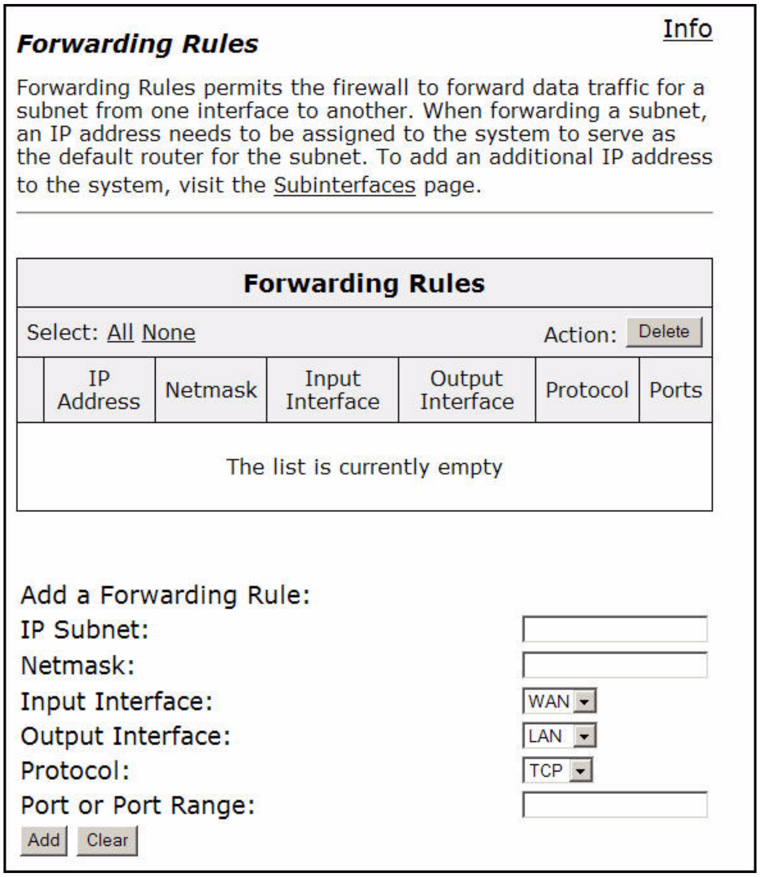
This is shown below.



Configuring Forwarding Rules

To configure address forwarding rules, use the following steps:

1. Using the configuration graphical user interface, from the Configuration Menu on the left-hand side, click **Firewall**.
2. Click **Forwarding Rules**. The window shown below opens.



Forwarding Rules Info

Forwarding Rules permits the firewall to forward data traffic for a subnet from one interface to another. When forwarding a subnet, an IP address needs to be assigned to the system to serve as the default router for the subnet. To add an additional IP address to the system, visit the Subinterfaces page.

Forwarding Rules

Select: All None Action: Delete

	IP Address	Netmask	Input Interface	Output Interface	Protocol	Ports
The list is currently empty						

Add a Forwarding Rule:

IP Subnet:	
Netmask:	
Input Interface:	WAN
Output Interface:	LAN
Protocol:	TCP
Port or Port Range:	

3. On this screen, complete the following information:

- **IP Subnet:** The subnet to be forward through the firewall from the Input Interface to the Output Interface.
- **Netmask:** The network mask to apply to the IP Subnet to create the range of IP addresses that are forwarded through the firewall.
- **Input Interface:** The interface where data is received that is destined for the forwarded subnet (destination address(es)).
- **Output Interface:** The interface where data is received that is sent from the forwarded subnet (source address(es)).
- **Protocol:** The following protocols are used:
  - UDP: for the specified network, allows the specified UDP port or port range to pass through the system
  - TCP: for the specified network, allows the specified TCP port or port range to pass through the system

  - Any: for the specified network, allows all ports and protocols through the system. No ports are required because not all protocols support the concept of ports.

- Port or Port Range: The port number or port range allowed through the system when UDP or TCP are selected. A port range is specified by separating the starting and ending ports with a colon ':' (for example, 22:80). The ports parameter is not supported when you select **Any** protocol because not all protocols support the concept of ports.

4. When you have finished entering this information, click **Add**.
5. Click **OK**. The new forwarding entry appears on the Forwarding Rules window in the list area.

Peering Proxy

H.323 prefixes can be used to route calls based on a matching prefix in the destination alias of the call. Each prefix is associated with a domain name or IP address to send the call to in case the prefix matches.

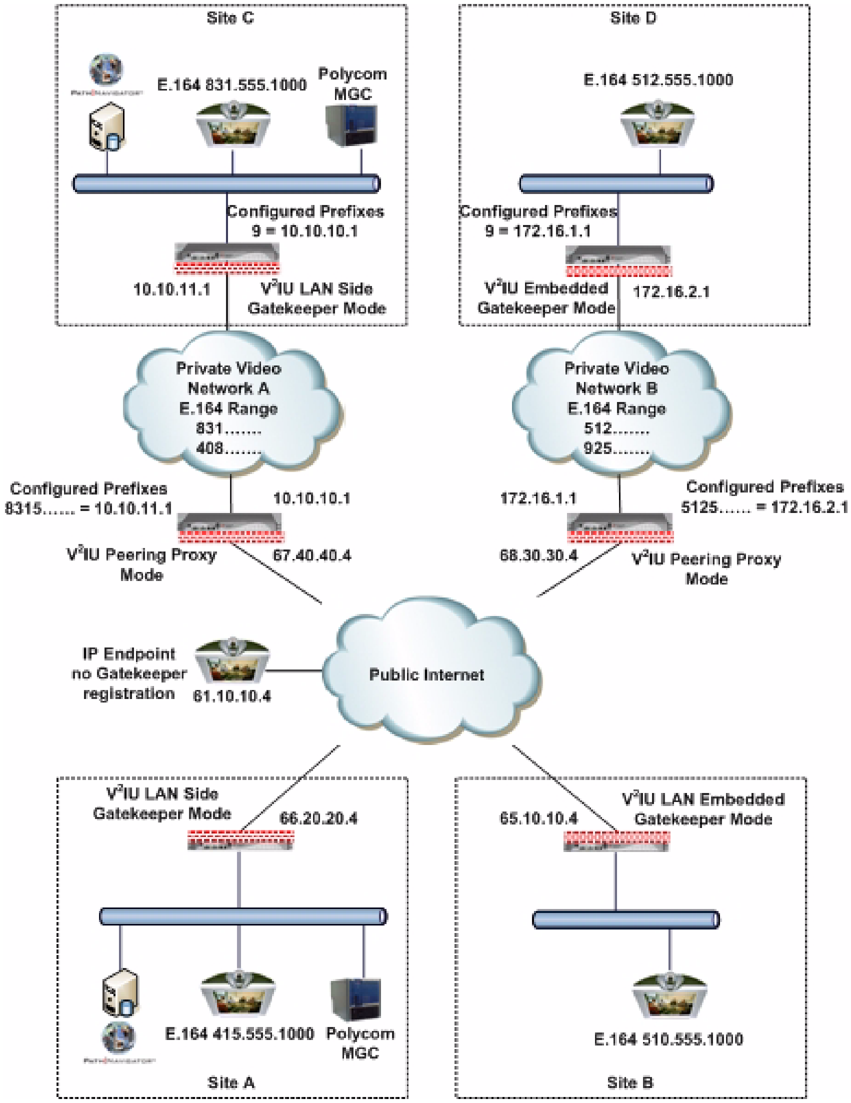
The prefixes are searched in order, that is, the first prefix is tried first, and then the next one on the list until the system finds a matching prefix. This means that if there are multiple matching prefixes, the first one is used.

How Peering Proxy Works

The Polycom V^{2}IU 4350 supports the concept of an H.323 Peering Proxy. This function provides advanced security layers or peering points within the network where a security layer is needed. Peering Proxy allows network providers to add internetworking connections between their “trusted” network and an unknown network. This topology hides their trusted network and the Stateful packet inspection Firewall provides the policies to ensure security. You can add Peering Proxies in series with one another to push the core H.323 networking infrastructure to meet individual security requirements.

The illustration below shows a sample diagram with dial plan and call flow examples. It is a snapshot of how the Peering Proxy can be deployed. Peering Proxy however, is not limited to this specific scenario, so contact your Polycom representative to discuss specific network requirements for full Peering Proxy support.

Note: A minimum configuration for Peering Proxy would be for inbound only prefixes, since there may be many endpoints to statically route calls to. There might also be a master gatekeeper to which all endpoints are registered. In this case, you would only need 1 prefix pointing to the master gatekeeper and let that gatekeeper signal the other endpoints directly.



In the example above, the Polycom V^{2}IU 4350 Peering Proxy is installed in "Private Video Network A and B," a peering point into this network. This network could have additional peering points to allow topology spreading of network resources. However, this example shows only a single point. Peering

Proxy provides an access point into this network and is responsible for the E.164 dial plan using NANP (North American Numbering Plans or NAP’s). The NAP’s in this case are 831 and 408.

Dial plan integrity is required to insure proper routing of prefix's. This means that if users are to dial into your network, they could be required to enter a “Prefix” on their V^{2}IU with a corresponding destination IP. If the user was to dial another user NOT destined to your network with the same beginning prefix, the prefix configured on this V^{2}IU would create a prefix match and the call would route incorrectly. The call routes to the destination defined in the prefix and not to the intended endpoint. The example shows “Private Video Network A's Peering Proxy” with an inbound prefix defined as 8315…… Any inbound call that matches 8315 with any 6 digits creates a prefix match and sends the call to 10.10.11.1. Refer to “Regular Expressions” in the Info button on the GUI interface for information on all the methods for defining prefixes.

Private Video Network A is one example of a V^{2}IU configured in “LAN Side Gatekeeper” mode with an ANNEX O dial method to dial “Off Net.” Internal “On Net” endpoints registered to the LAN Side Gatekeeper will dial E.164 only. This allows any location to place calls to any location with an ANNEX O dial plan, that is, E.164@WAN_IP or other V^{2}IU's deployed on the network. In this example a Peering Proxy has been deployed to allow dialing ingress and egress to the Public Internet. At each V^{2}IU location required to egress, the Public Internet requires a “Prefix” to be configured. This allows that location's endpoint to dial “Off Net” to the Public Internet. This prefix can be configured to any digit and may be part of the externally dialed E.164 in the E.164@WAN_IP, that is, to reach site A by dialing 4155551000@66.20.20.4 where the prefix is defined as 415* or 415……. In this example, a “9” was chosen. The prefix is then mapped to the LAN interface of the Peering Proxy 10.10.11.1. The dial string is now 94155551000@66.20.20.4 and a strip rule for the prefix is applied. This is needed to route the call at the destination correctly. If the Site C V^{2}IU does not strip the “9”, the destination V^{2}IU fails the call with a “No Registered Client” message (call failures can be viewed under the “H323 Activity” page in the GUI), since the “9” becomes part of the E.164. If you choose a prefix that matches the destination E.164, set Site A’s V^{2}IU to NOT strip matching prefixes.

NOTE: In this illustration E.164@WAN_IP was used as an example. Peering Proxy and all V^{2}IU's support user@host ANNEX O dialing methods, for example 123@1.1.1.1 or abc@1.1.1.1 or abc@abc.com with a DNS SRV record configured to point to an A record for the WAN IP of the V^{2}IU.

The following sections demonstrate the Dial Plan for ingress and egress calls to Private Video Network A as shown in the illustration.

Outbound from Site C to Site A

Site C dials an endpoint located at Site A: 94155551000@66.20.20.4. The PathNavigator receives the call and generates a Q.931setup to the V^{2}IU for that subnet. The V^{2}IU processes the Q.931 setup from the calling endpoint. The V^{2}IU looks for a prefix match. In this case, the “9” creates a match. The “Strip Matching Prefix” rule is applied, the “9” is stripped, and the call is routed to the Peering Proxy IP 10.10.10.1. The Peering Proxy applies the same rule set, in this case, NO matching prefix is found and ANNEX O dialing is applied. The call is now routed to Site A's V^{2}IU. The call is forwarded to the LAN Side PathNavigator where the registered client with the E.164 of 4155551000 is located and the call is gatekeeper routed to the called endpoint.

Inbound from Site A to Site C

Site A dials: 8315551000@67.40.40.4. (The destination IP is the Peering Proxy WAN IP address.) The Peering Proxy is configured with prefix 8315……and is mapped to the WAN IP of the V^{2}IU 10.10.11.1. As explained earlier, the prefix could be 831* or 83…… and so on, depending upon dial plan requirements. The PathNavigator receives the Q.931setup from the endpoint and forwards the call to the V^{2}IU for that subnet. The V^{2}IU receives the Q.931 setup from the calling endpoint. The V^{2}IU looks for a prefix match, finds NO matching prefix, and ANNEX O dialing is applied. The call is now routed to the Peering Proxy IP 67.40.40.4. The Peering Proxy receives the Q.931 setup and looks for a prefix match, in this case “8315” creates a match. The Peering Proxy now changes the destination IP to 10.10.11.1 and routes the call to Site A’s V^{2}IU. The Q.931 setup is forwarded to the LAN Side PathNavigator where the registered client with the E.164 of 8315551000 is located, and the call is gatekeeper routed to the called endpoint.

Outbound from Site C to Site D

Site C dials an endpoint located at Site D: 95125551000@68.30.30.4. The PathNavigator receives the call and generates a Q.931 setup to the V^{2}IU for that subnet. The V^{2}IU processes the Q.931 setup from the calling endpoint. The V^{2}IU looks for a prefix match, in this case the “9” creates a match. The “Strip Matching Prefix” rule is applied, the “9” is striped, and the call is routed to the Peering Proxy IP 10.10.10.1. The Peering Proxy applies the same rule set, in this case NO matching prefix is found, and ANNEX O dialing is applied. The call is now routed to the Peering Proxy for “Private Video Network B” IP 68.30.30.4. The Peering Proxy receives the Q.931 and looks for a prefix match. In this case, “5125” creates a match. The Peering Proxy now changes the destination IP to 172.16.2.1 and routes the call to Site D's V^{2}IU. The V^{2}IU is configured for Embedded Gatekeeper Mode. In this mode, the endpoint is directly registered and an E.164 registered client match is made. The call is then routed to the called endpoint.

Outbound from Site D to Site B

Site D dials an endpoint located at Site B: 95105551000@65.10.10.4. The V^{2}IU Embedded Gatekeeper is configured with a prefix of “9” to point to Peering Proxy 172.16.1.1. The V^{2}IU looks for a prefix match. In this case, the “9” creates a match. The “Strip Matching Prefix” rule is applied, the “9” is striped, and the call is routed to Peering Proxy IP 172.16.1.1. The Peering Proxy applies the same rule set. In this case NO matching prefix is found and ANNEX O dialing is applied. The call is now routed to Site B. The V^{2}IU is configured for

Embedded Gatekeeper Mode. In this mode, the endpoint is directly registered, an E.164 registered client match is made, and the call is routed to the called endpoint.

Outbound from Site C to Public IP Endpoint

Site C dials the public endpoint: 9@61.10.10.4. The PathNavigator receives the call and generates a Q.931 setup to the V^{2}IU for that subnet. The V^{2}IU receives the Call setup from the calling endpoint, and the V^{2}IU looks for a prefix match. In this case, the “9” creates a match. The “Strip Matching Prefix” rule is applied, the “9” is striped, and the call is routed to the Peering Proxy IP 10.10.10.1. The Peering Proxy applies the same rule set, in this case NO matching prefix is found, and direct IP dialing is applied.

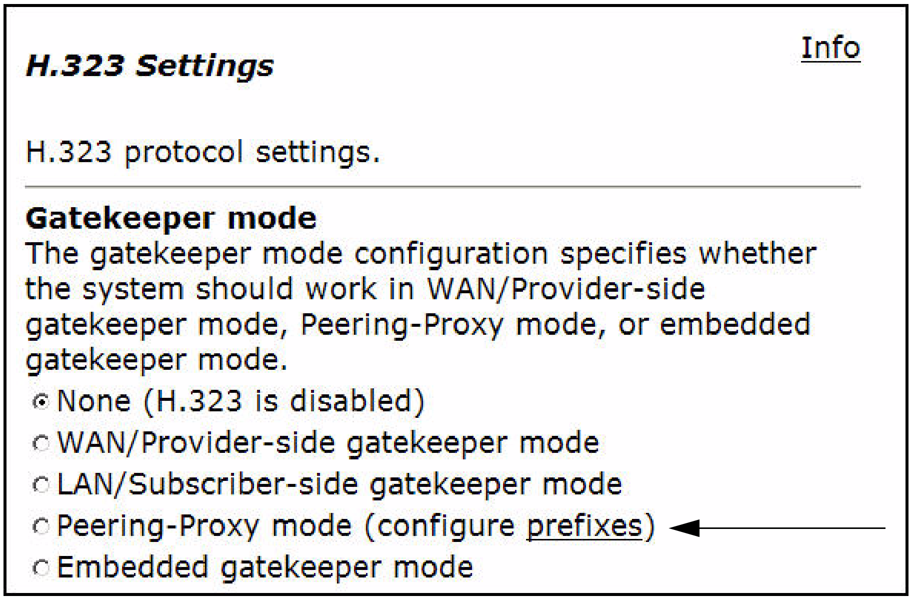
Inbound from Public IP Endpoint to Site C

Public IP endpoint is NOT registered to a gatekeeper and must dial an IP+EXT to reach Site A’s endpoint,. In this case, the IP address is 67.40.40.4 and EXT 8315551000. The Peering Proxy receives the call and looks for a prefix match. In this case “8315” creates a match. The Peering Proxy now changes the destination IP to 10.10.11.1 and routes the call to Site A’s V^{2}IU. The Q.931 setup is forwarded to the LAN Side PathNavigator where the registered client with the E.164 of 8315551000 is located, and the call is gatekeeper routed to the called endpoint.

Configuring Peering Proxy

To configure peering proxy, use the following steps:

1. Using the configuration graphical user interface, from the Configuration Menu on the left-hand side, click **VoIP ALG**.
2. Click **H.323**. The window shown below opens.



H.323 Settings

Info

H.323 protocol settings.

Gatekeeper mode

The gatekeeper mode configuration specifies whether the system should work in WAN/Provider-side gatekeeper mode, Peering-Proxy mode, or embedded gatekeeper mode.

● None (H.323 is disabled)

○ WAN/Provider-side gatekeeper mode

○ LAN/Subscriber-side gatekeeper mode

○ Peering-Proxy mode (configure prefixes)

○ Embedded gatekeeper mode

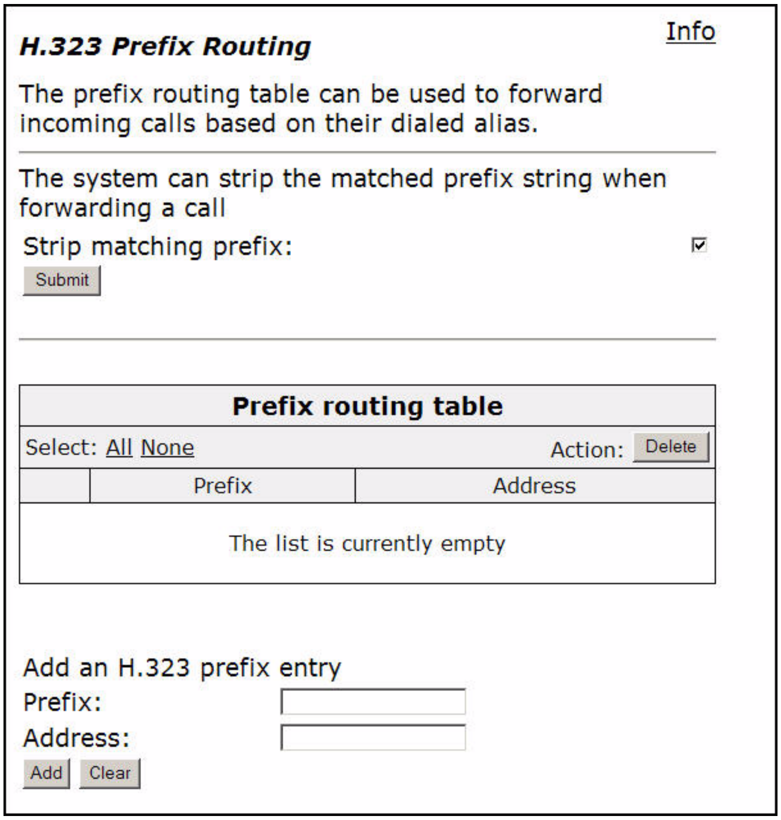
3. On this screen, check “Peering-Proxy mode”.
4. **Scroll** to the bottom of the window and click **Submit.**

Adding an H.323 Prefix Entry

You can add prefixes by entering the prefix string and the target address.

To add an H.323 prefix entry, use the following steps:

1. Using the configuration graphical user interface, from the Configuration Menu on the right-hand side, click **VoIP ALG**.
2. Click **H.323 Prefixes**. The window shown below opens.



H.323 Prefix Routing

The prefix routing table can be used to forward incoming calls based on their dialed alias.

The system can strip the matched prefix string when forwarding a call

Strip matching prefix: ☑

Prefix routing table

Select: All None   Action: Delete

	Prefix	Address
	The list is currently empty	

Add an H.323 prefix entry

Prefix: ____

Address: ____

The prefix routing table shows all currently configured prefixes. The prefixes are searched in the order they are entered. Each prefix can be moved up or down in the list. You can select and delete prefixes.

3. To strip a matching prefix, select the checkbox and click **Submit.**

   If you enable this, all matching prefixes are stripped from the destination alias before the call is forwarded.

4. To add an entry, enter the prefix and the address.

   The prefix string can be a regular expression as described above. The target address can be a domain name or an IP address.

5. Click **Add**. The new entry appears in the table.

Clients List Lock

Client List lockdown allows you to prevent new clients from registering. This is done as follows:

- Creating a client, as follows:

- Manually entering all clients that are allowed to use the system
- Running the system without the Client List lockdown feature until all desired clients have registered

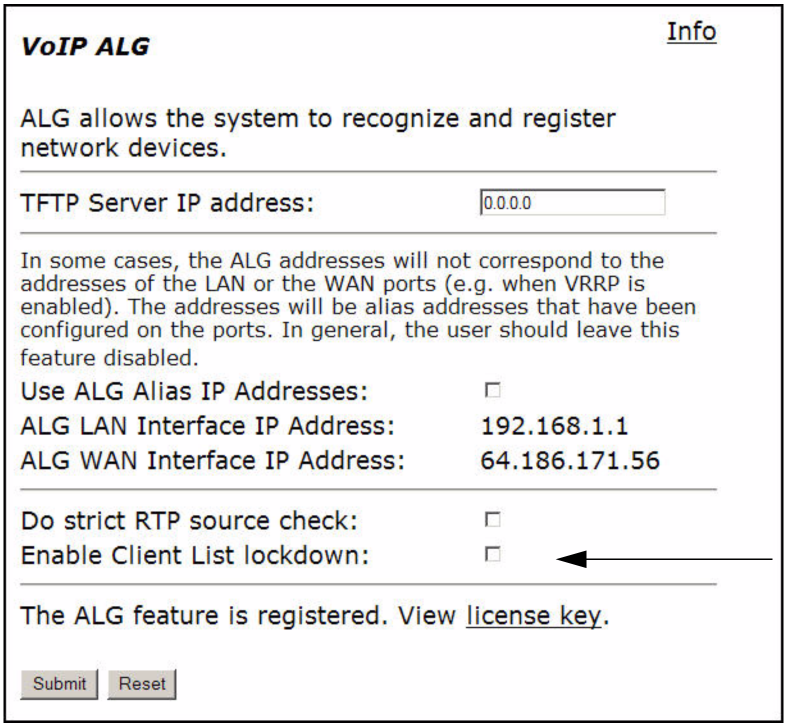
- Enabling this feature.

This feature is useful for lists involved with 911 usage.
When this feature is in effect, any message from an unauthorized SIP client will be rejected with a “403 Forbidden” response. MGCP messages will be discarded.

Enabling the Clients List Lock

To configure clients list lock, use the following steps:

6. Using the configuration graphical user interface, from the Configuration Menu on the left-hand side, click **VoIP ALG**. The following window appears.



VoIP ALG

Info

ALG allows the system to recognize and register network devices.

TFTP Server IP address: 0.0.0.0

In some cases, the ALG addresses will not correspond to the addresses of the LAN or the WAN ports (e.g. when VRRP is enabled). The addresses will be alias addresses that have been configured on the ports. In general, the user should leave this feature disabled.

Use ALG Alias IP Addresses: □
ALG LAN Interface IP Address: 192.168.1.1
ALG WAN Interface IP Address: 64.186.171.56

Do strict RTP source check: □
Enable Client List lockdown: □

The ALG feature is registered. View license key.

3. On this screen, check “Enable Client List lockdown”.
4. Scroll to the bottom of the window and click **Submit.**

H.323 Activity Monitor

The H.323 Activity Monitor shows any recent H.323 events that may be of interest to the administrator of the system. The information appears in three columns:

- Event/Time
- Source
- Destination

Following this information are a number of lines with event specific information such as call-id, duration, call-status, and so on.
Abnormal events have their event specific information listed in red.

Type of Events

The events that may currently be listed in the activity monitor are as follows:

- **Bandwidth change** - the endpoint requested a change of the bandwidth used for its call, only sent if the bandwidth management is enabled.
- **Call Setup** – Only sent if the call was 'successfully' established. A call is successfully established if the H.245 media negotiation connection was established.
- **Call Termination** – Sent when a call terminates. You can have a call termination event without a call setup event, for example, a failed call that doesn't reach the H.245 established state will not cause a call setup event, but only a call termination event.
- **Registration Reject** – Sent when a registration was rejected. This includes the authority that rejected the registration (our side or the gatekeeper (only in WAN GK mode) as well as a text reason for the rejection.
- **Gatekeeper reachability changed** (only in WAN GK mode). Gatekeeper status toggled from reachable to unreachable or vice versa.
- **Location Request** – Received a location request from a neighboring gatekeeper.
- **Location Confirm** – Sent, or forwarded, a location confirm to a previous request.
- **Location Reject** - Sent, or forwarded, a location reject to a previous request.

Call Status

The call status shows the last state of the call at the time of the event. Each call progresses through a number of states when being established. If a call fails, the call-status in the call termination event can help trouble-shoot the cause of the call failure. For example, if the call fails at the "Caller/Callee admission request received" state, there may be a problem communicating with the gatekeeper, whereas if the call fails at the "Attempting to establish outgoing Q.931 TCP connection" state, the remote endpoint may not be reachable. The following are call status messages:

- **"Caller admission request received"**

Received an admission request from the source endpoint and forwarded it to the gatekeeper.

- **"Caller admission response received"**

Received an admission response (either confirm or reject) from the gatekeeper and forwarded it to the source endpoint.

- **"Incoming Q.931 TCP connection established"**

Received an incoming Q.931 TCP connection from the source.

- **"Attempting to establish outgoing Q.931 TCP connection"**
Successfully resolved the destination of the call and attempting to establish an outgoing Q.931 TCP connection to the destination.

- **"Q.931 signaling received and forwarded"**
Both Q.931 TCP connections have been successfully established and Q.931 signaling has been received and forwarded.

- **"Callee admission request received"**
Received an admission request from the destination endpoint and forwarded it to the gatekeeper.

- **"Callee admission response received"**
Received an admission response (either confirm or reject) from the gatekeeper and forwarded it to the destination endpoint.

- **"Incoming H.245 TCP connection established"**
Received an incoming H.245 TCP connection from the source.

- **"Attempting to establish outgoing H.245 TCP connection"**
Attempting to establish an outgoing H.245 TCP connection to the destination.

- **"H.245 signaling received and forwarded"**
Both H.245 TCP connections have been successfully established and H.245 signaling has been received and forwarded. At this point, the call is considered established, even though no media channels have been opened up yet.

- **"Outgoing media channel established"**
An outgoing media channel (from the LAN/subscriber side to the WAN/provider side) has been opened.

- **"Incoming media channel established"**
An incoming media channel (from the WAN/provider side to the LAN/subscriber side) has been opened.

- **"Bidirectional media channels established"**
Media channels have been opened in both directions. This is a normal call where media is being sent in both directions.

Call Termination

The call termination cause may also give some information about why the call terminated or failed to be established.

- **"Out of system resources"**
The call could not be completed because the system was out of system resources.

- **"Client owning the call has been deleted"**
The call could not be completed because the client that made this call was deleted during the call setup.

- **"Connection to destination could not be established"**
A TCP connection to the destination could not be established.

- **"Connection refused by destination"**
The call could not be completed because the destination refused the incoming TCP connection.

- **"No route to destination"**
A TCP connection to the destination could not be established because the destination could not be reached. This could happen if there is no route to the destination or, if the destination is on the same subnet, the destination does not answer to ARP requests.

- **"Connection to destination timed out"**
The TCP connection attempt to the destination timed out before it could be established.

- **"Call ended by source"**
The call was gracefully terminated by H.323 signaling from the source. This usually indicates that the endpoint intended to terminate the call.

- **"Call ended by destination"**
The call was gracefully terminated by H.323 signaling from the destination. This usually indicates that the endpoint intended to terminate the call.

- **"Connection terminated by source"**
The call was terminated because the source terminated the TCP connection without prior call termination signaling.

- **"Connection terminated by destination"**
The call was terminated because the destination terminated the TCP connection without prior call termination signaling.

- **"No admission confirm received"**
The call could not be established because the admission response was not received from the gatekeeper.

- **"Cannot resolve destination"**
The call could not be established because the destination could not be resolved.

- **"At maximum bandwidth usage"**

The call could not be established because the system already is at the maximum allowed bandwidth.

- **"Received admission reject"**
The call was terminated because an admission reject was received from the gatekeeper.

- **"Received disengage request"**
The call was terminated because the endpoint requested to tear down the call.

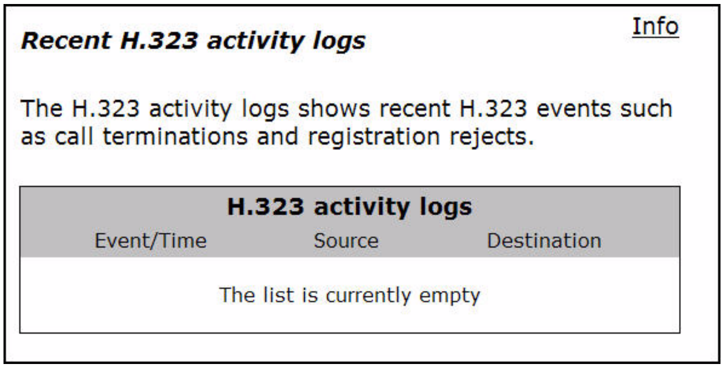
- **"Received invalid data"**
The call could not be established because the system received invalid data on the signaling channel.

- **"Cannot find client"**
The call could not be established because the called client could not be found.

Viewing the H.323 Activity Monitor

To configure the H.323 Activity Monitor, use the following steps:

5. Using the configuration graphical user interface, from the Configuration Menu on the left-hand side, click **VoIP**.
6. Click **H.323 Activity**. The window shown below opens.



Recent H.323 activity logs

The H.323 activity logs shows recent H.323 events such as call terminations and registration rejects.

H.323 activity logs		
Event/Time	Source	Destination
The list is currently empty		

2. On this screen, the event list contains three columns:
   - The Event/Time field - shows the type of event and the time that it occurred.
   - The Source field - shows the source of the event as an IP address and an alias (when available).
   - The Destination field - shows the destination of the event as an IP address and an alias (when available).

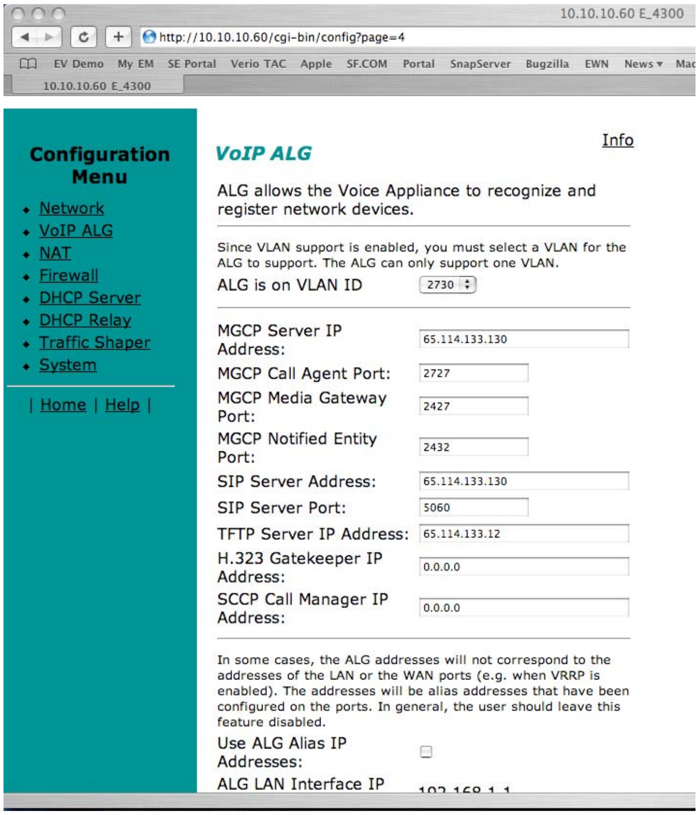
VoIP Configuration

The 4350 provides a VoIP application layer gateway (ALG) for the SIP, MGCP, and H.323 protocols. The ALG proxies the connection between the VoIP softswitch, IP PBX or gatekeeper and voice and video devices such as IP phones, IADs or softphones. By acting as a proxy the 4350 is able to provide several important functions for IP based voice and video:

- Provide NAT/PAT services for voice and video traffic. NAT/PAT for VoIP enables you to use a single public IP address on the WAN interface of the 4350 to represent multiple private IP addresses assigned to voice or video devices on the LAN. The NAT function maps both IP address and IP port number between the public and private addresses so that all signaling and VoIP media packets are translated. A single public IP address can support up to 253 voice and video devices.
- Provide security services for voice and video traffic.
  - NAT/PAT services hide enterprise LAN topology from hackers.
  - The ALG acts as a “voice and video aware” firewall and ensures only authenticated voice traffic enters the enterprise LAN. This is accomplished by the dynamic provisioning of signaling and media ports for authenticated voice devices. The implementation is stateful and open ports are closed automatically when no longer required to support the voice or video call.
- Enable mobility in the enterprise LAN for voice devices. This is useful, for example, when using WiFi or moving office locations. In these instances the IP address of the voice and video device may be changed.

Configure the VoIP ALG

In order to configure the VoIP ALG the 4350 must be told where to reach the signaling servers and TFTP server on behalf of the voice devices.



VoIP ALG

ALG allows the Voice Appliance to recognize and register network devices.

Since VLAN support is enabled, you must select a VLAN for the ALG to support. The ALG can only support one VLAN.

ALG is on VLAN ID: 2730

MGCP Server IP Address:	65.114.133.130
MGCP Call Agent Port:	2727
MGCP Media Gateway Port:	2427
MGCP Notified Entity Port:	2432
SIP Server Address:	65.114.133.130
SIP Server Port:	5060
TFTP Server IP Address:	65.114.133.12
H.323 Gatekeeper IP Address:	0.0.0.0
SCCP Call Manager IP Address:	0.0.0.0

In some cases, the ALG addresses will not correspond to the addresses of the LAN or the WAN ports (e.g. when VRRP is enabled). The addresses will be alias addresses that have been configured on the ports. In general, the user should leave this feature disabled.

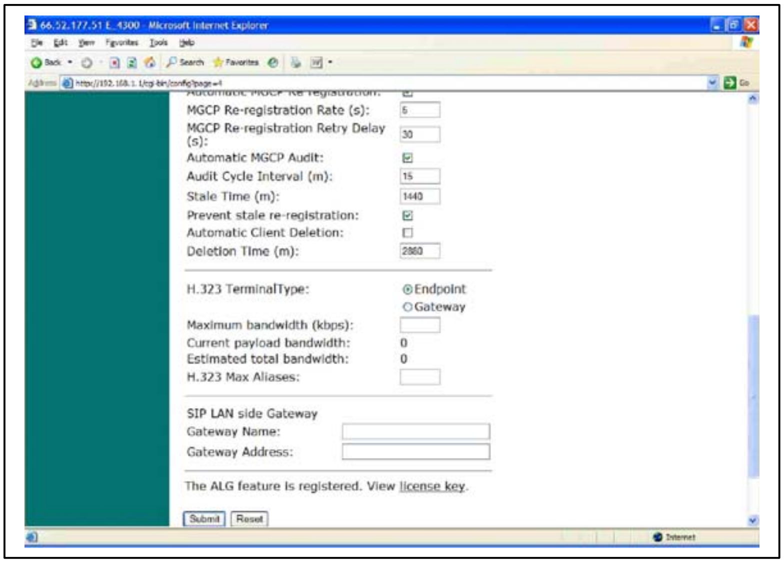
Use ALG Alias IP Addresses: □

ALG LAN Interface IP 192.168.1.1

1. Select VoIP ALG.
2. If using VLANs assign the ALG to a specific VLAN id using the drop down menu.
3. If you are using MGCP enter the MGCP Server IP Address, MGCP Media Gateway Port and MGCP Notified Entity Port.
4. If you are using SIP enter the SIP Server IP Address and SIP server port. The SIP server port is the port used by the SIP registrar. The default value is port 5060.
5. If you are using H.323 enter the H.323 Gatekeeper IP Address.
6. Enter the TFTP Server Address. This address is used to identify the TFTP server that contains the images used by IP phones at boot up. The 4350 performs a TFTP server relay function.

Note

It is not necessary to program in an FTP server address if your IP phones use the FTP protocol instead of TFTP to retrieve their images. A relay function is not needed for FTP as the 4350 will forward FTP traffic to the destination server as programmed in your IP phone.


66.52.177.51 E_4300 - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites
Address http://192.168.1.1/cgi-bin/config?page=4
Go
Internet

Automatic MGCP Re-registration: ☑
MGCP Re-registration Rate (s): 5
MGCP Re-registration Retry Delay (s): 30
Automatic MGCP Audit: ☑
Audit Cycle Interval (m): 15
Stale Time (m): 1440
Prevent stale re-registration: ☑
Automatic Client Deletion: □
Deletion Time (m): 2880

H.323 TerminalType: ● Endpoint ○ Gateway
Maximum bandwidth (kbps):
Current payload bandwidth: 0
Estimated total bandwidth: 0
H.323 Max Aliases:

SIP LAN side Gateway
Gateway Name:
Gateway Address:

The ALG feature is registered. View license key.

Submit Reset

7. Automatic MGCP Re-registration is used to re-register MGCP endpoints every time the network or system restarts. Enable this feature to automatically synchronize the softswitch and phones immediately after a restart. The default is Enabled.

8. The MGCP Re-registration Rate is used to set the number of MGCP RSIP messages to send per second to the Media Gateway Controller when re-registration is needed. If the MGCP Re-registration Rate needs to be changed, enter a value between 1 and 5. Generally, this value does not need to be modified. The default value is 5 msg/second.

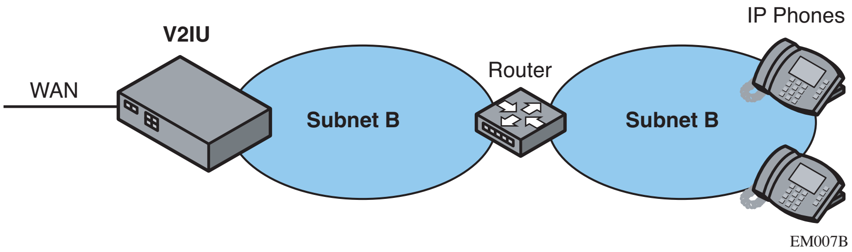
9. The system re-registers clients when it starts up. If any of these re-registration requests fail, the system will wait for the configured number of seconds and then retry the re-registration for the clients that failed. The system will make at most 10 re-registration requests for failed attempts. If the MGCP Re-registration Retry Delay needs to be changed, enter a value between 30 and 60 seconds. Generally, this value does not need to be modified. The default value is 30 seconds.

10. The H.323 TerminalType is used to specify the type of terminal that the Voice Appliance should use. It can be either endpoint or gateway. The Maximum Bandwidth specifies the bandwidth to allow for H.323 calls. The bandwidth is specified in kbps and if it is set to 0, bandwidth management is not enforced. Only calls with media traversing the 4350 is counted towards the bandwidth maximum.

11. The Current payload bandwidth calculates the current video traffic, without IP overhead, traversing the Appliance. The Estimated total bandwidth calculates the total video traffic, plus IP overhead, traversing the Appliance.
12. The H.323 Max Aliases limits the number of aliases that are allowed to register with the Voice Appliance. If this number is exceeded when a client tries to register, the registration will be rejected. If the value is set to 0, the maximum is not enforced.
13. The SIP LAN Side Gateway is used to configure a LAN side SIP gateway to which calls that are not for a registered phone can be sent. The name of the gateway is the name that is configured for the gateway in the soft-switch and the IP address is the address where the gateway can be reached.
14. Press Submit.

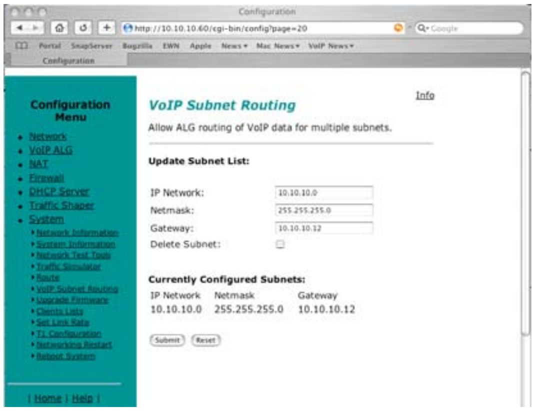
Configure VoIP Subnet Routing

It is not necessary to configure VoIP subnet routing if all of your voice and video devices are installed on the same IP subnet as the 4350. In some installations the voice and video devices are located in different subnets than the 4350 and connected via intermediate routers. In these instances it is necessary to configure a return path in the 4350 by specifying the intermediate router who knows how to reach the voice devices. This router must be reachable by the 4350.



Note

VoIP Subnet Routing is separate and independent from static data routes (see Static IP routing). VoIP subnet routes must be configured for each LAN subnet that contains devices making use of the 4350's Application Layer Gateway (ALG). These entries tell the ALG that the identified subnet is allowed to make use of its services and what router the ALG should use to reach that subnet.



VoIP Subnet Routing

Allow ALG routing of VoIP data for multiple subnets.

Update Subnet List:

IP Network: 10.10.10.0
Netmask: 255.255.255.0
Gateway: 10.10.10.12
Delete Subnet: □

Currently Configured Subnets:

IP Network	Netmask	Gateway
10.10.10.0	255.255.255.0	10.10.10.12

Enter a VoIP Subnet Route

1. Select System.
2. Select System Overview.
3. Select VoIP Subnet Routing.
4. Enter the IP Network (e.g. 10.10.12.0).

   This is the IP address of the remote subnet containing the voice devices.
5. Enter the **Netmask** (e.g. 255.255.255.0).

   This is the mask of the IP address of the subnet containing the voice devices.
6. Enter the **Gateway** (e.g. 10.10.10.2).

   This is the IP address of the intermediate router that knows the return path to the remote subnet from the 4350.
7. Press **Submit**.

Perform steps 1 through 7 for each remote subnet containing the voice devices.

Note

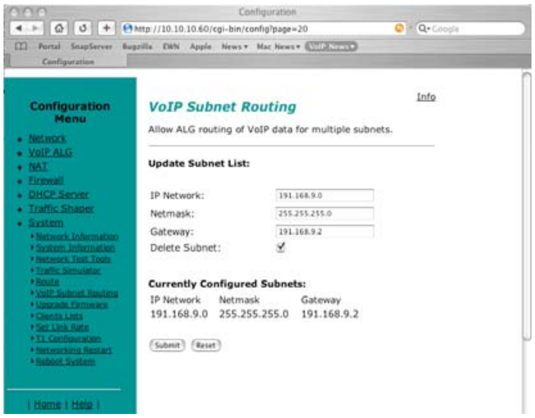
The 4350 is limited to a total of 20 different VoIP subnets.

Delete a VoIP Subnet Route

1. Select System.

2. Select System Overview.
3. Select VoIP Subnet Routing.
4. Enter the IP Network (e.g. 10.10.12.0).

   This is the IP address of the remote subnet containing the voice devices.



VoIP Subnet Routing

Allow ALG routing of VoIP data for multiple subnets.

Update Subnet List:

IP Network: 191.168.9.0
Netmask: 255.255.255.0
Gateway: 191.168.9.2
Delete Subnet: ☑

Currently Configured Subnets:

IP Network	Netmask	Gateway
191.168.9.0	255.255.255.0	191.168.9.2

5. Enter the Netmask (e.g. 255.255.255.0).

   This is the mask of the IP address of the subnet containing the voice devices.
6. Enter the Gateway (e.g. 10.10.10.2).
7. This is the IP address of the intermediate router that knows the return path to the remote subnet from the 4350.
8. Select the Delete Subnet checkbox.
9. Press Submit.

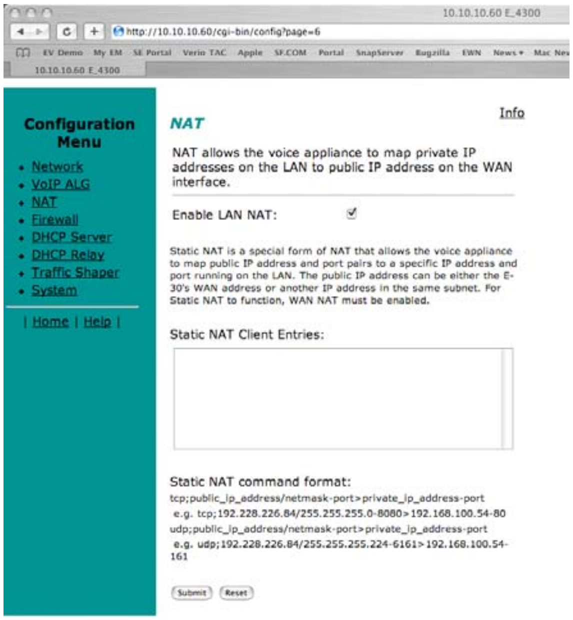
Perform steps 1 through 8 for each remote subnet that you wish to delete.

Configure IP Phones, IADs or Softphones

After configuring the 4350 VoIP ALG the voice devices must be configured to point to the LAN interface of the 4350 as their signaling gateway and optionally as their TFTP server (if they use the TFTP protocol to retrieve their software images). The steps required to setup these devices differ from vendor to vendor. Using the DHCP server included in the 4350 will significantly simplify the setup of these devices if they are able to obtain their IP configuration via DHCP. Please consult the applicable users guide of each device for detailed instructions.

Data Networking Configuration

The 4350 provides static IP routing and two types of Network Address Translation (NAT) functions for data traffic. This section describes the use and configuration of these features.



NAT

NAT allows the voice appliance to map private IP addresses on the LAN to public IP address on the WAN interface.

Enable LAN NAT: ☑

Static NAT is a special form of NAT that allows the voice appliance to map public IP address and port pairs to a specific IP address and port running on the LAN. The public IP address can be either the E-30's WAN address or another IP address in the same subnet. For Static NAT to function, WAN NAT must be enabled.

Static NAT Client Entries:

Static NAT command format:

```
tcp;public_ip_address/netmask-port>private_ip_address-port
 e.g. tcp;192.228.226.84/255.255.255.0-8080>192.168.100.54-80
udp;public_ip_address/netmask-port>private_ip_address-port
 e.g. udp;192.228.226.84/255.255.255.224-6161>192.168.100.54-
161
```

NAT for Data Traffic

NAT allows hosts on a private internal network (the LAN side of the 4350) to anonymously communicate with devices on an external network (the WAN side of the 4350). The 4350 with NAT enabled will re-write outbound packet headers using public IP addresses in place of private IP addresses so that the private IP addresses are not exposed to the external network. Additionally, the ports used by the IP addresses are also changed as they traverse the 4350. This is known as Port Address Translation (PAT) and provides an additional security measure. The 4350 maintains a table of these mappings so that return packets can be forwarded to the correct host on the private network.

The 4350 provides two types of NAT functions: dynamic NAT and static NAT. Dynamic NAT allows many private IP addresses to be mapped to a single public IP address (using different port numbers of the public IP address). Static NAT maps private IP addresses and port. For example, mapping a public IP address to a specific machine on the private network responsible for receiving email.

Note The 4350 ALG automatically handles NAT for voice devices.

Configure Dynamic NAT

Use Dynamic NAT when you have multiple PCs installed on the LAN side of the 4350 that require Internet or WAN access. Once Dynamic NAT is enabled the 4350 will automatically perform an address translation for all packets to/from the LAN side PCs.

1. From the Configuration Menu select NAT.
2. Use the Enable Lan NAT checkbox to enable or disable dynamic NAT.

   The default value for dynamic NAT is enabled.
3. Press Submit.

Configure Static NAT

Use Static NAT when a server or PC located in the private network needs to be accessible from the external network. Some examples include a corporate web server, a mail server or an FTP server. In these instances, the 4350 statically maps the public IP address of each server to the actual private IP address of the server.

Note In order for Static NAT to function dynamic NAT must be enabled.

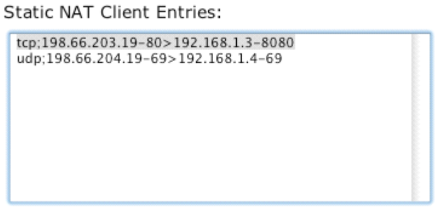
1. Select **NAT**.
2. Enter the public and private IP addresses and ports to be mapped in ***Static NAT Client Entries*** using the following format:

   ```
   Protocol;PublicIPAddress/netmask-port>PrivateIPAddress-port
   ```

   For example, the entry “tcp;198.66.203.19-80>192.168.1.3-8080” will map all web traffic destined to public IP address 198.66.203.19 to the private webserver 192.168.1.3 port 8080. The public IP address of 198.66.203.19 is automatically created as a “subinterface” or “secondary address” on the WAN interface of the 4350 so that external hosts can reach the web server.

   Each entry should be placed on a new line.
3. Press **Submit**.

Delete a Static NAT entry



Static NAT Client Entries:

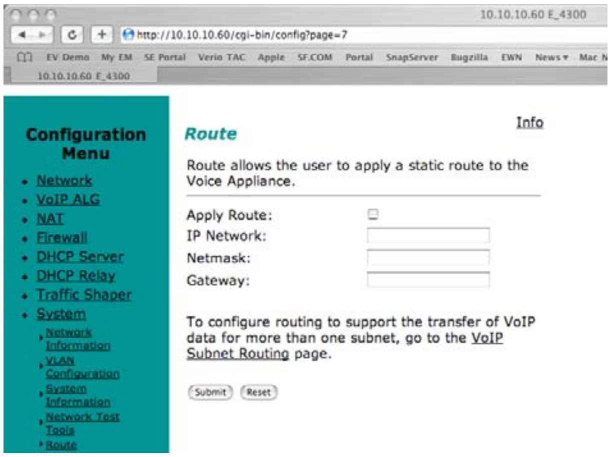
```
tcp;198.66.203.19-80>192.168.1.3-8080
udp;198.66.204.19-69>192.168.1.4-69
```

1. Select NAT.
2. To delete an IP address or a range of IP addresses highlight the entry in the Static NAT Client Entries list and press the Delete key on your keyboard.
3. Press Submit.

Static IP routing

In addition to locally connected IP networks the 4350 can forward traffic for a remote data network by configuring a static route entry. Any packets destined for the remote data network will be forwarded to the specified gateway address in the entry.

Configure the static route



Route

Route allows the user to apply a static route to the Voice Appliance.

Apply Route: □
IP Network:
Netmask:
Gateway:

To configure routing to support the transfer of VoIP data for more than one subnet, go to the VoIP Subnet Routing page.

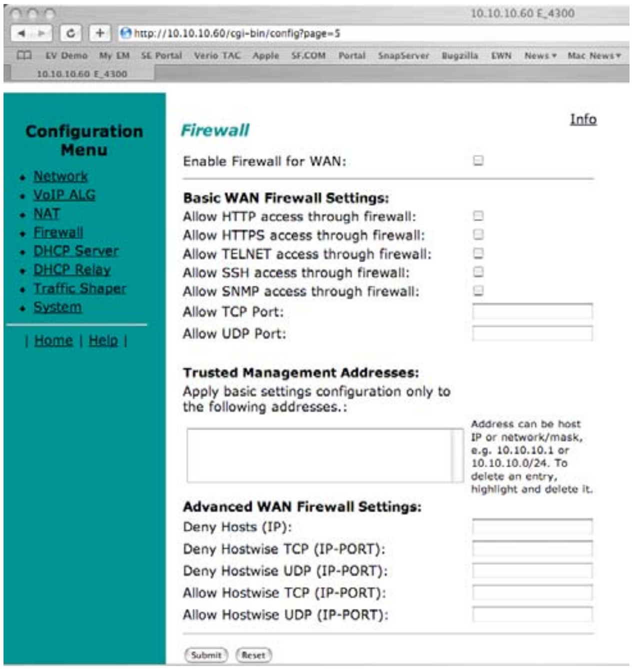
1. Select System.
2. Select System Overview.
3. Select Route.

4. Select the Apply Route checkbox.
5. Enter the IP Network address. This address is the remote data network you would like the 4350 to forward to the gateway. The hosts portion of the IP address should be set to “0”. For example, 10.10.20.0
6. Enter the Netmask of the remote data network. For example, 255.255.255.0
7. Enter the Gateway IP address of the interface that will receive all packets destined for the remote data network.
8. Press Submit.

Delete the static route

1. Select System.
2. Select System Overview.
3. Select Route.
4. Remove the check in the Apply Route checkbox.
5. Press Submit.

Firewall Configuration



Firewall

Enable Firewall for WAN: □

Basic WAN Firewall Settings:

Allow HTTP access through firewall: □

Allow HTTPS access through firewall: □

Allow TELNET access through firewall: □

Allow SSH access through firewall: □

Allow SNMP access through firewall: □

Allow TCP Port:

Allow UDP Port:

Trusted Management Addresses:

Apply basic settings configuration only to the following addresses.:

Address can be host IP or network/mask, e.g. 10.10.10.1 or 10.10.10.0/24. To delete an entry, highlight and delete it.

Advanced WAN Firewall Settings:

Deny Hosts (IP):

Deny Hostwise TCP (IP-PORT):

Deny Hostwise UDP (IP-PORT):

Allow Hostwise TCP (IP-PORT):

Allow Hostwise UDP (IP-PORT):

The 4350 uses a Stateful Packet Inspection (SPI) firewall to protect data devices installed behind the LAN interface. Voice devices are protected by the 4350 Application Layer Gateway (ALG) as described in ***VoIP Configuration***.

The firewall is enabled by default. The default behavior of the firewall is to:

- deny all traffic originating from the WAN
- allow all traffic originating from the LAN
- allow only return traffic for connections that originated from the LAN
- deny all traffic originating from the WAN to the 4350 itself
- allow all traffic originating from the LAN to the 4350

The default behavior can be modified using the basic and advanced settings fields on the firewall configuration page. We recommend that you use the 4350 firewall, however it can be disabled if the 4350 is installed behind an existing legacy firewall.

Enable or disable the firewall

1. Select **Fire**wall.
2. Use the Enable Firewall checkbox to either enable or disable the firewall.
3. Select Submi**t**.

Configure Basic settings

To allow or deny HTTP, Telnet and SSH traffic originating from the WAN to the 4350 simply use the checkboxes provided in the basic settings area of the firewall configuration page. By default, access from the WAN into the 4350 is disabled.

Warning Denying HTTP, Telnet or SSH traffic from the WAN may result in losing management connectivity to the 4350 if you are configuring the system remotely using the WAN link.

1. Select Firewall.
2. Use the three Allow access from WAN side checkboxes to enable or disable HTTP, Telnet, and/or SSH access from IP devices on the WAN side of the 4350.
3. Select Submit.

Configure Advanced Settings

A comprehensive security policy can be created using the advanced settings of the 4350 firewall. The policy actions that can be taken on any packet processed by the 4350 are summarized in the following table:

Action	Description	Input format
Allow TCP Port	Allows traffic with the specified TCP port to terminate on the 4350.	*Valid values range from 1 through 65535. *Multiple entries are separated by a space *Range value specified by the : character. For example, 25:50 means perform the action on ports 25 through 50
Allow UDP Port	Allows traffic with the specified UDP port to terminate on the 4350.	*Valid values range from 1 through 65535. *Multiple entries are separated by a space *Range value specified by the : character. For example: 25:50 means perform the action on ports 25 through 50
Deny Hosts (IP)	Denies all traffic with the source IP address matching the specified hosts	*Multiple entries are separated by a space *Classful IP addresses are assumed by default. For example: 192.168.3.1 uses a class c mask. Subnets can be specified using the / notation. E.g. 192.168.3.1/24
Deny Hostwise TCP (IP-Port)	Denies all traffic matching the specified TCP port numbers **and** the specified **source** IP addresses	*Multiple entries are separated by a space *Port are specified using a - character. For example: 192.168.3.1-23 for Telnet. *Port ranges are specified using a : character. For example: 192.168.3.1-23:50 means port 23 through 50 *Classful IP addresses are assumed by default. For example: 192.168.3.1 uses a class c mask. Subnets can be specified using the / notation. E.g. 192.168.3.1/24
Deny Hostwise UDP (IP-Port)	Denies all traffic matching the specified UDP port numbers **and** the specified **source** IP addresses	*Multiple entries are separated by a space *Port are specified using a - character. For example: 192.168.3.1-23 for Telnet. *Port ranges are specified using a : character. For example: 192.168.3.1-23:50 means port 23 through 50 *Classful IP addresses are assumed by default. For example: 192.168.3.1 uses a class c mask. Subnets can be specified using the / notation. E.g. 192.168.3.1/24
Allow Hostwise TCP (IP-Port)	Allows all traffic matching the specified TCP port numbers **and** the specified **source** IP addresses	*Multiple entries are separated by a space *Port are specified using a - character. For example: 192.168.3.1-23 for Telnet. *Port ranges are specified using a : character. For example: 192.168.3.1-23:50 means port 23 through 50 *Classful IP addresses are assumed by default. For example: 192.168.3.1 uses a class c mask. Subnets can be specified using the / notation. E.g. 192.168.3.1/24

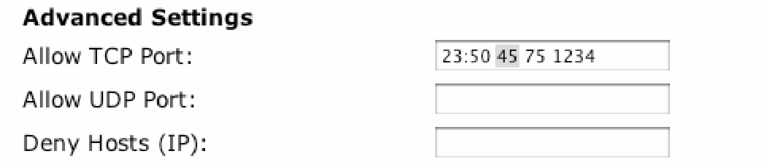
Allow Hostwise UDP (IP-Port)	Allows all traffic matching the specified UDP port numbers **and** the specified **source** IP addresses	*Multiple entries are separated by a space *Port are specified using a - character. For example: 192.168.3.1-23 for Telnet. *Port ranges are specified using a : character. For example: 192.168.3.1-23:50 means port 23 through 50 *Classful IP addresses are assumed by default. For example: 192.168.3.1 uses a class c mask. Subnets can be specified using the / notation. E.g. 192.168.3.1/24

If a given packet does not match any of the configured rules, it is dropped.

1. Select Firewall.
2. Enter the desired Advanced Settings using the table above as a guide.
3. Select Submit.

Remove Advanced Setting Entries

To remove an advanced firewall setting simply highlight the value in the entry box and delete it using the keyboard.



Advanced Settings

Allow TCP Port: 23:50 45 75 1234

Allow UDP Port:

Deny Hosts (IP):

1. Select Firewall.
2. Highlight the entry to be deleted in the Advanced Settings list and press the Delete key on your keyboard.
3. Press Submit.

Traffic Management Configuration

Traffic management is required to ensure high quality voice and video calls when voice, video, and data traffic share the same WAN link. Voice and video traffic must be prioritized for transmission over data traffic to meet the stringent jitter, latency and packet loss requirements for high quality voice and video. The 4350:

- Automatically prioritizes voice and video traffic over data traffic to ensure high quality voice and video calls.
- Maximizes WAN link utilization by allowing data traffic to burst up to full line rate in the absence of voice and video calls.

- Controls the data transfer rate of far-end WAN TCP devices to limit WAN link congestion.

- Supports network-based QoS applications by setting the TOS bits for all VoIP packets sent to the WAN and the LAN. TOS bits are used so that VoIP packets can be prioritized in the network by DiffServ enabled routers. The TOS bit value used by the 4350 is to "minimize delay and maximize throughput", or 0xb8 hexadecimal. This value is set for all VoIP packets processed by the 4350 and overwrites any specific TOS bit configuration set by VoIP endpoints.

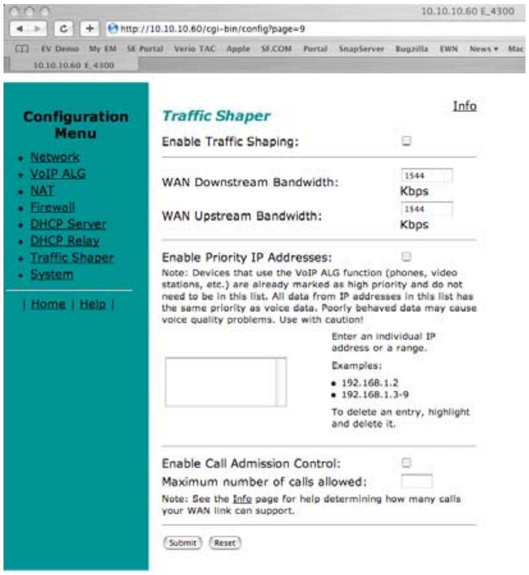
- Ensures that bandwidth allocated to new voice and video calls does not adversely affect the quality of existing active calls (Call Admission Control or CAC).

The 4350 combines sophisticated traffic management mechanisms including classification, prioritization, queuing, rate limiting and CAC to ensure high quality voice and video calls. Fortunately the system manages this complexity for you and configuring traffic management is very straightforward:

1. Enable traffic shaping.
2. Specify the upstream and downstream bandwidth of your WAN link.
3. Enable CAC.

Please follow the steps below to configure and enable traffic management.

Enable Traffic Shaping



Traffic Shaper

Enable Traffic Shaping: □

WAN Downstream Bandwidth: 1544 Kbps

WAN Upstream Bandwidth: 1544 Kbps

Enable Priority IP Addresses: □

Note: Devices that use the VoIP ALG function (phones, video stations, etc.) are already marked as high priority and do not need to be in this list. All data from IP addresses in this list has the same priority as voice data. Poorly behaved data may cause voice quality problems. Use with caution!

Enter an individual IP address or a range.

Examples:

- 192.168.1.2
- 192.168.1.3-9

To delete an entry, highlight and delete it.

Enable Call Admission Control: □

Maximum number of calls allowed:

Note: See the Info page for help determining how many calls your WAN link can support.

1. From the Configuration Menu, select Traffic Shaper.
2. Select the Enable traffic shaper checkbox.

3. Specify the upstream and downstream bandwidth of your WAN link
4. Enter the WAN Downstream Bandwidth in Kbps.
5. Enter the WAN Upstream Bandwidth in Kbps.

Note

For FT1/T1/E1 links the upstream and downstream bandwidths will always be the same value (the link is full-duplex).

Optionally enable priority IP addresses

VoIP traffic from devices that use the VoIP ALG function (phones, video stations, softphones on Pcs, etc.) are already marked as high priority and **do not** need to be manually configured in this list. This list is used to prioritize voice traffic from trunk interfaces of IP PBXs or other high priority devices that do not use the VoIP ALG function of the 4350.

>> Enter the IP address of other high priority devices in the priority IP Addresses box.

You can enter individual IP addresses or a range using by appending a "-"character to the last octet. For example, 10.10.10.2-5 would specify 10.10.10.2, 10.10.10.3, 10.10.10.4 and 10.10.10.5 as voice devices.

Warning

Care must be taken to ensure that the IP addresses entered do not include data devices such as PCs or workstations. Traffic from these devices will be placed in the priority voice queue internal to the 4350 and burst up to full line rate. This will starve actual voice devices by consuming priority bandwidth and result in dropped calls, busy signals & poor voice quality.

Enable CAC

The 4350 uses CAC to limit the number of active voice calls over the WAN link. This is necessary because a typical installation uses a ratio of 1:2 or 1:4 active voice calls to voice devices on the assumption that 50% or 25% of all users are on the phone at the same time. These ratios are guidelines only and at times the number of concurrent calls may exceed the amount of WAN bandwidth available to process the calls. In this instance existing phone calls will experience poor quality or be dropped all together. To prevent this from occurring a typical voice installation will set a threshold for the maximum number of concurrent voice calls supported by the WAN access link. New call requests in excess of this threshold will receive the equivalent of a "fast busy" and the WAN link will not become oversubscribed.

For IP Centrex installations the maximum number of concurrent voice calls is usually configured in the 4350 by enabling CAC. When the 4350 is deployed in IP PBX applications the maximum number of concurrent calls could be configured in the IP PBX. If the PBX is responsible for this setting you do not need to configure CAC in the 4350. Please check with your IT administrator to determine if this is the case.

Determining the maximum number of concurrent calls

The maximum number of concurrent calls that can be supported by the WAN access link is calculated using the following formula:

Max calls = (Maximum WAN upstream bandwidth * .85)/VoIP codec rate

where,

Maximum WAN upstream bandwidth = value entered in step D above (in Kbps)

VoIP codec rate = 85.6Kbps for G.711 voice devices or 29.6Kbps for G.729 voice devices.

The maximum WAN upstream bandwidth is multiplied by .85 in the formula above to reduce the total bandwidth available for voice calls by 15%. This reduction is necessary because the 4350 automatically reserves 15% of the total WAN bandwidth for low priority data traffic so that it is not starved completely. Starving data traffic completely would increase the number of retry attempts and exacerbate congestion on the link during periods of peak usage.

Examples

The maximum number of G.711 voice calls supported by a T1 (1.536 Kbps) WAN is calculated as follows:

(1.536*.85)/85.6 = 15.3 or 15 total voice calls.

The maximum number of G.711 voice calls supported by a 768Kbps fractional T1 WAN is calculated as follows:

(768*.85)/85.6 = 7.6 or 7 total voice calls

The maximum number of G.729 voice calls supported by a 256Kbps fractional T1 WAN is calculated as follows:

(256*.85)/29.6 = 7.4 or 7 total voice calls

After determining the maximum number of voice calls CAC is enabled as follows:

1. Select the Enable Call Admission Control checkbox.
2. Enter Maximum number of calls allowed as calculated above.
3. Press Submit.

A Closer Look at Traffic Management in the 4350

The traffic management mechanisms provided by the 4350 are designed to ensure high priority real-time voice and video traffic is processed before lower priority data traffic. At the same time, bandwidth not in use by voice and video traffic is made available so that data traffic can burst up to full line rate making efficient use of WAN bandwidth. Traffic management mechanisms are applied to traffic in both the upstream (LAN to WAN) and downstream (WAN to LAN) direction. Each direction is independent of the other and can support different size priority queues.

Classifying

High priority voice and video traffic generated by endpoint devices is automatically identified by the V^{2}IU's VoIP Application Layer Gateway. Other VoIP devices (not making use of the ALG) can be defined as high-priority by their IP address. The user configures these addresses into the priority list in the Traffic Shaper section of the 4350 web GUI.

As the 4350 processes packets they are identified as either high or low priority based on this configuration. Packets identified as high priority are marked as such in the TOS bits of their IP header, allowing prioritization by downstream routers. The TOS field is set to 12 hexadecimal "minimize delay and maximize throughput." This value overwrites any prior value.

Upstream Traffic Management

The 4350 appliance uses a combination of Class Based Queuing and simple classless queuing to send data in the upstream direction. The Class Based Queue (CBQ) consists of two priority classes (high and low), a scheduler to decide when packets need to be sent, and a traffic shaper to rate-limit by delaying packets before they are sent. Each of these is described in more detail below.

Priority classes

Voice and video traffic is placed in the high-priority queue and data traffic is placed in the low-priority queue. The IP header TOS field of packets in the high-priority queue is set to "minimize delay and maximize throughput".

Scheduler

High-priority data is polled before low priority data, thereby minimizing the latency for voice and video traffic. High-priority data is allowed to use up to 85% of the total WAN bandwidth. Although preferential treatment is given to high-priority data, 15% of the WAN link is always reserved so that low-priority data is not starved.

High priority data is polled before lower priority data to reduce overall latency for voice traffic.

Traffic shaper

To smooth bursts from high speed data links (typically from the LAN Ethernet heading to the WAN) the 4350 appliance uses a buffer that clocks data out at rates not exceeding automatically-calculated maximums. Low-priority data is clocked out at the WAN link's full rate LESS the bandwidth currently being used for high-priority (voice) data. High-priority data is clocked out at the WAN's full link rate. Any long-lasting burst condition in low-priority data will cause these packets to be delayed and, if necessary, dropped.

Downstream Traffic Management

Since the 4350 is the final transmitting device for WAN traffic in the upstream direction (LAN to WAN) it is easy to see how its QoS mechanisms can be applied to traffic it is transmitting to guarantee sufficient bandwidth for voice traffic. We have control over how packets are handed to the WAN interface.

In the downstream direction (WAN to LAN) we are installed at the receiving end of a service provider link and therefore have no control over the amount of voice or data traffic being sent to us over the WAN interface. How then can we still guarantee the quality of in-bound voice traffic when it is entirely possible for an FTP session, for example, to consume the vast majority of downstream bandwidth?

Fortunately this is possible by shaping on both the egress LAN and egress WAN ports of the 4350 appliance and leveraging the congestion avoidance mechanisms built into TCP. Essentially, data packets received by the 4350's WAN interface at a rate that exceeds the T1's bandwidth LESS the bandwidth used for active voice calls are delayed (then dropped if necessary) before being forwarding on to its LAN interface. Similarly, data traffic sent back to the 4350 for transmission to the WAN are also delayed (as described in the above section). This results in the WAN-based devices following the rules of TCP/IP congestion avoidance and slowing down their transmit rate. This technique is quite effective in practice, as end stations usually reduce their transmit rate before VoIP signaling has completed for new call setup.

For example consider the scenario where there are no voice calls over a WAN link and multiple FTP sessions are consuming all available bandwidth:

1. A new call request is received by the 4350 from the WAN.
2. All signaling messages for the call are classified as voice traffic and therefore prioritized for transmission to the LAN before servicing the inbound FTP data.
3. RTP traffic (the voice data within an ongoing VoIP session) is similarly classified as voice traffic and treated with priority.
4. FTP data is buffered (or dropped) by the 4350 and return data, including the FTP ACKs, are also delayed. This results in a throttling of the transmit rate by the (far-end) FTP hosts, reducing overall WAN bandwidth consumption.

Generally, excessive UDP traffic must be shaped in the service provider network, as UDP does not provide congestion avoidance mechanisms. The exception to this is in the case of RTP UDP-based messages for voice traffic.

Although RTP makes use of UDP the 4350 appliance is able to provide its own congestion avoidance mechanism for voice traffic using Call Admission Control (CAC).

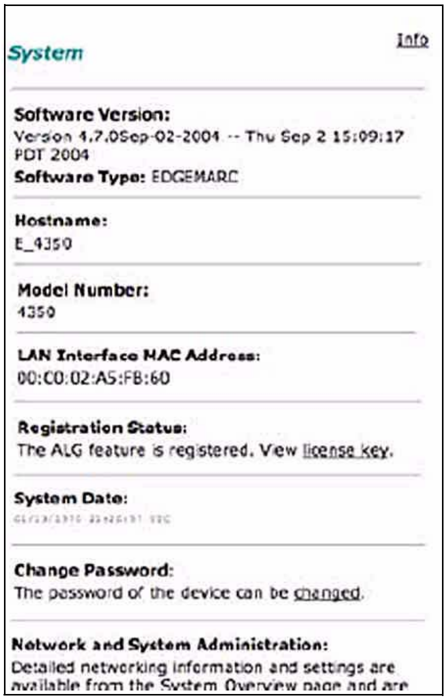
4

System Diagnostics

The V²IU 4350 provides a powerful set of diagnostic information, troubleshooting tools and utilities for system maintenance to network operators.

Viewing Software Version, Hardware Platform and the LAN MAC Address

The software version, hardware platform, and LAN MAC address are common pieces of information requested by technical support and are accessed directly through the System page of the 4350 web GUI.



System

Software Version:
Version 4.7.0Sep-02-2004 -- Thu Sep 2 15:09:17 PDT 2004
Software Type: EDGEMARC

Hostname:
E_4350

Model Number:
4350

LAN Interface MAC Address:
00:C0:02:A5:FB:60

Registration Status:
The ALG feature is registered. View license key.

System Date:
[illegible]

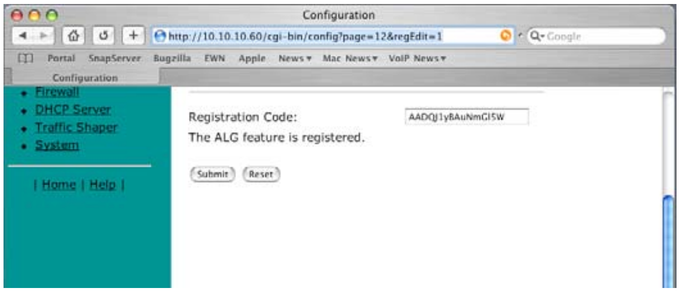
Change Password:
The password of the device can be changed.

Network and System Administration:
Detailed networking information and settings are available from the System Overview page and are

Viewing the ALG registration code

You will also find a link to the ALG registration code on the System page. The registration code enables the ALG and is pre-installed at the factory. If the registration code is inadvertently deleted you can re-enter the code using the following steps:

Enter the Registration Code



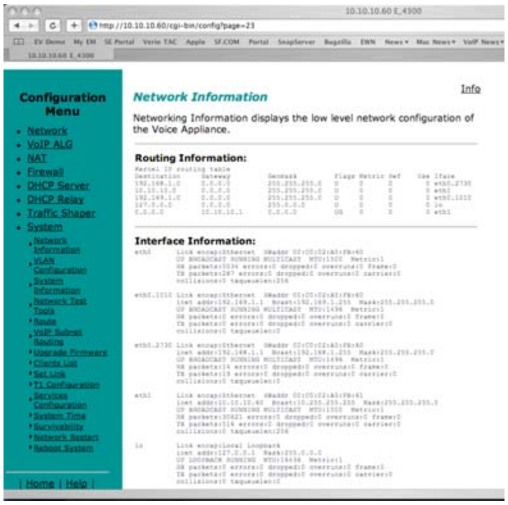
Registration Code: AADQJ1yBAuNmGI5W

The ALG feature is registered.

1. Select System.
2. Select registration code.
3. Select Edit Registration Code.
4. Enter the Registration Code.

   The registration code can be found on the sticker located on the bottom of the 4350.
5. Press Submit.

Viewing Networking Information



Network Information

Networking Information displays the low level network configuration of the Voice Appliance.

Routing Information:

```
Kernel IP routing table
Destination     Gateway         Genmask         Flags Metric Ref    Use Iface
192.168.1.0     0.0.0.0         255.255.255.0   U     0      0        0 eth0.2730
10.10.10.0      0.0.0.0         255.255.255.0   U     0      0        0 eth1
192.169.1.0     0.0.0.0         255.255.255.0   U     0      0        0 eth0.1010
127.0.0.0       0.0.0.0         255.0.0.0       U     0      0        0 lo
0.0.0.0         10.10.10.1      0.0.0.0         UG    0      0        0 eth1
```

Interface Information:

```
eth0      Link encap:Ethernet  HWaddr 00:C0:02:A3:FB:40
          UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
          RX packets:1034 errors:0 dropped:0 overruns:0 frame:0
          TX packets:287 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:256

eth0.1010 Link encap:Ethernet  HWaddr 00:C0:02:A3:FB:40
          inet addr:192.169.1.1  Bcast:192.169.1.255  Mask:255.255.255.0
          UP BROADCAST RUNNING MULTICAST  MTU:1496  Metric:1
          RX packets:0 errors:0 dropped:0 overruns:0 frame:0
          TX packets:0 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0

eth0.2730 Link encap:Ethernet  HWaddr 00:C0:02:A3:FB:40
          inet addr:192.168.1.1  Bcast:192.168.1.255  Mask:255.255.255.0
          UP BROADCAST RUNNING MULTICAST  MTU:1496  Metric:1
          RX packets:14 errors:0 dropped:0 overruns:0 frame:0
          TX packets:19 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0

eth1      Link encap:Ethernet  HWaddr 00:C0:02:A3:FB:41
          inet addr:10.10.10.60  Bcast:10.255.255.255  Mask:255.255.255.0
          UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
          RX packets:30821 errors:0 dropped:0 overruns:0 frame:0
          TX packets:314 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:256

lo        Link encap:Local Loopback
          inet addr:127.0.0.1  Mask:255.0.0.0
          UP LOOPBACK RUNNING  MTU:16436  Metric:1
          RX packets:0 errors:0 dropped:0 overruns:0 frame:0
          TX packets:0 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0
```

To view the networking configuration and status of the 4350 proceed to the Network Information page as follows:

1. Select System.
2. Select System Overview.
3. Select Network Information.

The following networking information is displayed:

Routing Information

The system routing table contains the static routes for hosts and networks that are configured on the 4350. If just the LAN and WAN IP addresses have been configured there will be four lines displayed:

- The private subnet will be associated with the LAN interface.
- A public subnet present for the WAN interface.
- An entry for the 4350 loopback interface
- The 4350's default gateway forwarding to the WAN interface

Additional lines may be displayed depending on the contents of the Route and VoIP Subnet Routing pages. Each of the entries on these pages will cause an additional entry in the routing table.

Link Status

Link Status displays the status of the ethernet interfaces. Ethernet autonegotiation is often unreliable, especially between different vendors or old and new networking equipment. Failure of autonegotiation is generally not a cause for concern. However, if the negotiated rates change intermittently or the link is reported as down or no link, the link rate may need to be set manually on the Set Link Rate page. Intermittent data and voice outages may be caused by link flapping when the two endpoints of the Ethernet cable cannot reach agreement using autonegotiation. If the link rate is set manually, ensure that the device at the far end of the connection can communicate at the desired rate. Incompatible rates can cause a loss of communication with the 4350.

Link status for the Ethernet ports is displayed via the LEDs adjacent to each physical port.

Interface Information

The specific status and configuration information for the system interfaces is displayed in the Interface Information section. HDLC0 shows the interface statistics for the T1/E1 WAN link. ETH0 shows the interface statistics for the internal LAN interface between the 4350 processor and the built-in LAN switch. Interface statistics for the external LAN ports are not displayed.

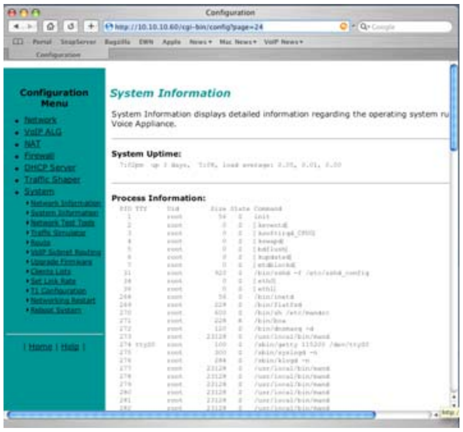
The interface statistics can point to areas of congestion in the network. If the errors statistic is a few percent or more of the total packets sent it may be an indication of excessive congestion on the network interface. If the congestion is not corrected the quality of voice calls will be affected. The topology of the network attached to the network interface with the errors should be examined and modified to better segment and isolate network traffic.

Viewing Advanced System Information

To view advanced system information for the 4350 proceed to the System Information page as follows:

1. Select System.
2. Select System Overview.
3. Select System Information.

The following system information is displayed:



System Information

System Information displays detailed information regarding the operating system ru
Voice Appliance.

System Uptime:

```
 7:02pm  up 3 days,  7:08, load average: 0.25, 0.01, 0.00
```

Process Information:

```
  PID TTY     Uid        Size State Command
    1         root         56   S   init
    2         root          0   S   [keventd]
    3         root          0   S   [ksoftirqd_CPU0]
    4         root          0   S   [kswapd]
    5         root          0   S   [bdflush]
    6         root          0   S   [kupdated]
    7         root          0   S   [mtdblockd]
   11         root        422   S   /bin/sshd -f /etc/sshd_config
   34         root          0   S   [eth0]
   39         root          0   S   [eth1]
  268         root         56   S   /bin/inetd
  269         root        228   S   /bin/flatfsd
  270         root        400   S   /bin/sh /etc/mandaemon
  271         root        228   R   /bin/boa
  272         root        120   S   /bin/dnsmasq -d
  273         root      23128   S   /usr/local/bin/mand
  274 ttyS0   root        100   S   /sbin/getty 115200 /dev/ttyS0
  275         root        300   S   /sbin/syslogd -n
  276         root        284   S   /sbin/klogd -n
  277         root      23128   S   /usr/local/bin/mand
  278         root      23128   S   /usr/local/bin/mand
  279         root      23128   S   /usr/local/bin/mand
  280         root      23128   S   /usr/local/bin/mand
  281         root      23128   S   /usr/local/bin/mand
  282         root      23128   S   /usr/local/bin/mand
```

System Uptime

System Uptime displays the current time, the amount of time elapsed since the last system reboot, and the system load averages for the past 1, 5, and 15 minutes. Uptime can help trace when a power outage may have interrupted service. Load averages that remain greater than 2 indicate excessive system loading. Partitioning voice traffic using a second system may be required.

Process Information

Displays the active processes in the 4350.

Memory Usage

Displays detailed memory allocation information that may be of use to technical support.

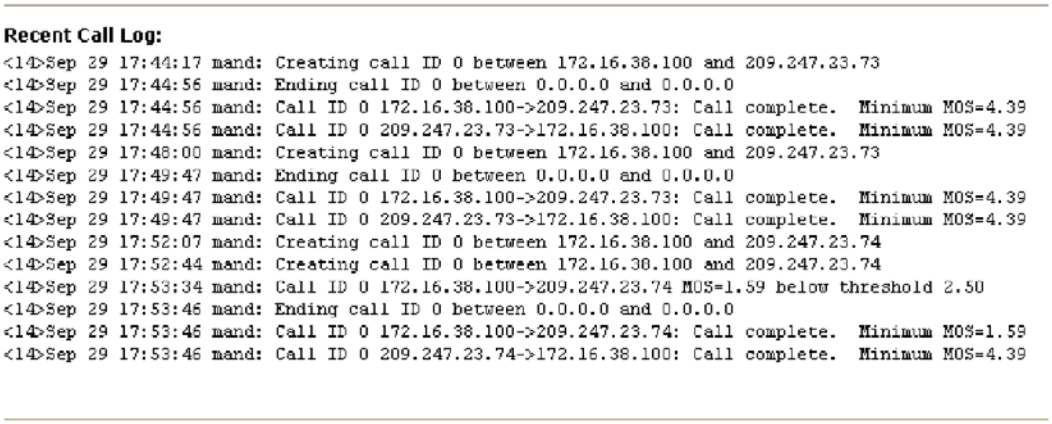
System Logging Messages

Displays information logged during system boot and normal operation. Logging messages may indicate unauthorized attempts to access the 4350, process restart messages, and excessive resource utilization messages.

Passive Voice Call Monitoring

The 4350 monitors live voice calls and performs objective speech quality assessment. This information enables the network operator to assess voice quality for the purposes of SLA tracking or problem isolation. Mean Opinion Score (MOS) results for RTP streams in both directions of a VoIP call are calculated at call completion. This information along with the IP addresses of the VoIP endpoints supporting the call are logged locally and optionally sent to an external syslog server (see **Enable Remote System Logging** **<<<find this heading>>>** for instructions on enabling logging to a remote syslog server). Additionally the 4350 will generate a real-time message for any MOS values calculated less than 2.5 (considered poor quality) during an active call.

Voice call quality information is found locally in the System Logging Messages section of the System Information page and a sample output is provided below.



Recent Call Log:

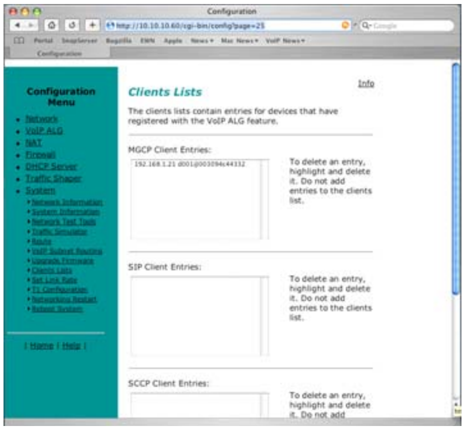
```
<14>Sep 29 17:44:17 mand: Creating call ID 0 between 172.16.38.100 and 209.247.23.73
<14>Sep 29 17:44:56 mand: Ending call ID 0 between 0.0.0.0 and 0.0.0.0
<14>Sep 29 17:44:56 mand: Call ID 0 172.16.38.100->209.247.23.73: Call complete.  Minimum MOS=4.39
<14>Sep 29 17:44:56 mand: Call ID 0 209.247.23.73->172.16.38.100: Call complete.  Minimum MOS=4.39
<14>Sep 29 17:48:00 mand: Creating call ID 0 between 172.16.38.100 and 209.247.23.73
<14>Sep 29 17:49:47 mand: Ending call ID 0 between 0.0.0.0 and 0.0.0.0
<14>Sep 29 17:49:47 mand: Call ID 0 172.16.38.100->209.247.23.73: Call complete.  Minimum MOS=4.39
<14>Sep 29 17:49:47 mand: Call ID 0 209.247.23.73->172.16.38.100: Call complete.  Minimum MOS=4.39
<14>Sep 29 17:52:07 mand: Creating call ID 0 between 172.16.38.100 and 209.247.23.74
<14>Sep 29 17:52:44 mand: Creating call ID 0 between 172.16.38.100 and 209.247.23.74
<14>Sep 29 17:53:34 mand: Call ID 0 172.16.38.100->209.247.23.74 MOS=1.59 below threshold 2.50
<14>Sep 29 17:53:46 mand: Ending call ID 0 between 0.0.0.0 and 0.0.0.0
<14>Sep 29 17:53:46 mand: Call ID 0 172.16.38.100->209.247.23.74: Call complete.  Minimum MOS=1.59
<14>Sep 29 17:53:46 mand: Call ID 0 209.247.23.74->172.16.38.100: Call complete.  Minimum MOS=4.39
```

Accessing Troubleshooting Tools

The 4350 provides convenient test tools to facilitate problem isolation and resolution. A network operator can use these tools to verify connectivity to/from the 4350 as well as trace datapaths to endpoints throughout the network.

Verify Registered Voice and Video Devices

The 4350 maintains a list of all registered voice and video devices called a clients list so that it can properly route voice and video calls. At startup, voice and video devices register their IP addresses with the 4350. The 4350 then registers on behalf of the voice and video devices by providing its own WAN IP address to the softswitch, gatekeeper, or IP PBX. If a user or network operator reconfigures the IP address of the voice or video device (such as an IP phone or IAD), it will re-register the new address with the 4350. In this instance voice and video calls may be routed improperly because the 4350 clients list contains out of date information.



Clients Lists

The clients lists contain entries for devices that have registered with the VoIP ALG feature.

MGCP Client Entries:

192.168.1.21 d001@003094c44332

To delete an entry, highlight and delete it. Do not add entries to the clients list.

SIP Client Entries:

To delete an entry, highlight and delete it. Do not add entries to the clients list.

SCCP Client Entries:

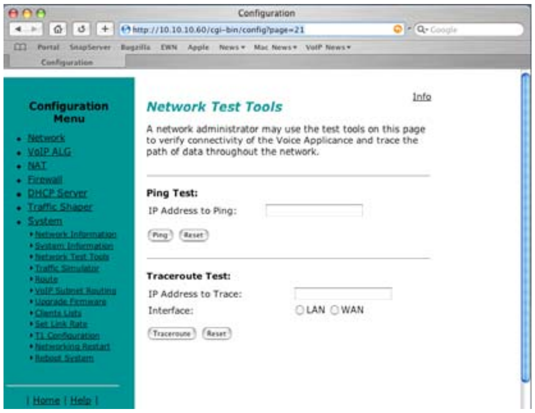
To delete an entry, highlight and delete it. Do not add

To update the clients list simply highlight and delete the stale entry using the following steps:

1. Select System.
2. Select System Overview.
3. Select Clients List.
4. Proceed to the appropriate signaling section, highlight the duplicate entry or entries and press the delete key on the keyboard
5. Press Submit.
6. Restart the VoIP ALG by following the instructions found in Restarting Networking Processes.

Performing a Ping Test

A ping test is the most common test used to verify basic connectivity to a networking device. Successful ping test results indicate that both physical and virtual path connections exist between the 4350 and the test IP address. Successful ping tests do not guarantee that all data traffic is allowed between the 4350 and the test IP address but is useful to verify basic reachability.



Network Test Tools

A network administrator may use the test tools on this page to verify connectivity of the Voice Applicance and trace the path of data throughout the network.

Ping Test:

IP Address to Ping:

Traceroute Test:

IP Address to Trace:

Interface: ○ LAN ○ WAN

The following steps are used to perform a ping test:

1. Select System.
2. Select System Overview.
3. Select Network Test Tools.
4. Enter the IP Address to Ping.
5. Press Ping.

The Network Test Tools page will be refreshed and the results of the ping test are displayed (this may take several seconds). The Reset button is used to clear the IP address entry used in step D above.

Performing a Traceroute Test

A traceroute test is used to track the progress of a packet through the network. The test can be used to verify that data destined for a WAN device reaches the remote IP address via the desired path. Similarly, internal network paths can be traced over the LAN to verify the local network topology. The following steps are used to perform a traceroute test:

1. Select System.
2. Select System Overview.
3. Select Network Test Tools.
4. Enter the IP address to Trace.
5. Select either the WAN or the LAN radio button
6. Press Traceroute.

The Network Test Tools page will be refreshed and the results of the traceroute test are displayed (this may take several seconds). The Reset button is used to clear the IP address entry used in step D above.

Restarting Networking Processes

In extreme circumstances while troubleshooting you may be asked to restart the networking processes including the VoIP ALG in the 4350 by technical support. Use the following steps to restart the networking processes:

1. Select System.
2. Select System Overview.
3. Select Networking Restart.
4. Press restart.

Warning Restarting network services will interrupt the system for up to a minute. All voice and data sessions currently in progress will be interrupted.

Rebooting the 4350

In extreme circumstances while troubleshooting you may be asked to reboot the 4350 by technical support. Please use the following steps to reboot the system:

1. Select System.
2. Select System Overview.
3. Select Rebooting System.
4. Press reboot.

Alternatively a reset can be performed locally by temporarily disconnecting the power cable from the 4350.

Warning Rebooting the system will interrupt services for a few minutes. All voice and data sessions currently in progress will be interrupted.

5

Saving and Restoring the V^{2}IU 4350 Configuration

The V^{2}IU 4350 stores all configuration information for the system in a series of individual files that reside in local flash memory. These files are read at boot time to determine the configuration identity of the 4350 and then stored in RAM as “running” state. As you configure the 4350 the *submit* command writes the configuration changes to both RAM and flash so that the files stored in flash are always up to date with the running state of the system.

The 4350 provides a utility that enables you to copy the individual configuration files stored in flash to a single, consolidated backup file. This single file can then be used as a backup for the entire system and restored at a later date if necessary. Multiple backup files with different system configurations can also be created and stored locally in the 4350 or on remote TFTP servers.

Note

No more than 2 backup files can be stored in the 4350’s flash due to size constraints. Also, it is recommended that you create a backup file after any configuration changes are made to the 4350. This is to prevent the loss of any configuration changes made since your last backup in the event that you must restore the system configuration.

Backup file operations are performed in the 4350 CLI using the ewn command.

The ewn Command

The syntax for the `ewn` command is as follows:

USAGE:

```
  ewn help|list
  ewn save|load|delete [file name]
  ewn upload|download [file name] [ip address]
```

where file name must use extension .conf1 or .conf2

The `ewn` command can be used with a local terminal connection or remotely using SSH.

1. Use a NULL modem cable to connect to serial port 1 of the 4350
2. Use a terminal emulator such as Hyperterminal set to a baud rate of 9600, 8, 1 and none (databits, stop bits and parity)

Alternatively you can connect to the 4350 remotely using SSH:

1. Logon as root
2. Enter the password

Once you are at the command prompt (#) you can create the backup file, store it to local flash, copy it to a remote TFTP server, copy it from a remote TFTP server, delete it, load it or list all available backup files.

Create a Backup File and Save in Local Flash

# ewn save <filename>

Saves the current running configuration.

Filename format (must use extension .conf1 or .conf2):
<filename1>.conf1

<filename2>.conf2

<filenameX> can be a combination of both letters and characters. For example, EWN30_041503.conf1 or location1_E30.conf2. Trying to use any other filename format will result in the error message: “EWN_ERROR_BAD_FILE_NAME”.

Warning The “.conf” extensions have special significance. If you save a configuration with <filename-new>.conf1, then any existing <filename-old>.conf1 will be overwritten with the new one.

Copy a Backup File to a Remote TFTP Server

```
# ewn upload <filename> <tftp server IP Address>
```

Copy a backup file from the 4350 to a TFTP server.

Download a Backup File from a Remote TFTP Server

```
# ewn download <filename> <tftp server IP Address>
```

Download a backup file from a TFTP server to the V²IU.

List the Available Backup Files

```
# ewn list
```

List all backup files stored in FLASH. If no file has been saved, the command will only return the # prompt.

Delete a Backup File

```
# ewn delete <filename>
```

Delete the backup file specified in the filename.

Load a Backup File so that it Becomes the Running Configuration

```
# ewn load <filename>
```

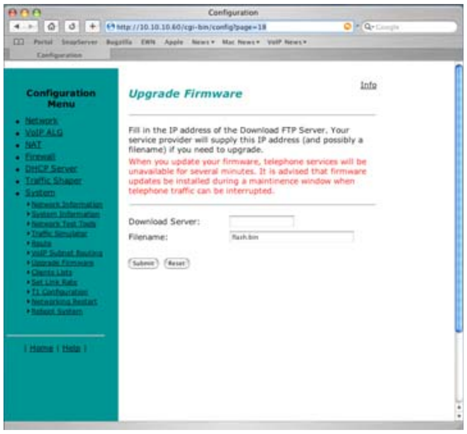
Loads the specified backup file into RAM and makes it the active running configuration.

Warning Issuing this command will automatically restart the 4350 and therefore interrupt any active voice calls and data sessions.

6

Upgrading the V^{2}IU 4350

This chapter describes how to upgrade your 4350 to the latest software release available from Polycom.



Upgrade Firmware

Fill in the IP address of the Download FTP Server. Your service provider will supply this IP address (and possibly a filename) if you need to upgrade.

When you update your firmware, telephone services will be unavailable for several minutes. It is advised that firmware updates be installed during a maintinence window when telephone traffic can be interrupted.

Download Server:

Filename: flash.bin

It is recommended that you reboot the 4350 prior to performing the upgrade. This is to make sure there is enough dynamic memory available to handle the upgrade process.

Warning When you update your software telephone services will be unavailable for several minutes. It is therefore advised that upgrades be performed during a maintenance window when telephone traffic can be interrupted.

Upgrade Procedure for Software Revision 1.3.11 or Later

Use this procedure if your 4350 is running software revision 1.3.11 or later. The software version can be found on the System page of the web GUI.

1. Select System.
2. Select System Overview.
3. Select Upgrade firmware.

4. Enter the Download Server address of ftp.support.polycom.com.
5. Enter the Filename: flash.bin
6. Press Submit.

   You can follow the progress of the upgrade by selecting the refresh the upgrade status link.

Warning Do not change the configuration or power off the device until the write is 100 percent complete. The 4350 may become unusable if the write is interrupted. The flash write can take up to 5 minutes depending on the speed of the download server.

   The system will automatically restart after the new image has been loaded.

7. Verify that the upgrade was successful by checking the software revision number found on the System page.

Appendix

Troubleshooting Tips

This section contains possible solutions to problems regarding the installation of the V^{2}IU 4350.

I am having trouble reaching the Internet through the 4350.

We recommend connecting a PC directly (or via a switch) to the LAN port of the 4350. The default LAN IP address of the 4350 is 192.168.1.1 so please be sure that the IP address of the PC is on the same network (such as 192.168.1.2). Once you have connected please verify that the IP configuration information in the *Network* page is correct. Some other items to try:

- Ping the WAN interface of the 4350 from the attached PC
- Ping the DNS server for your network. Sometimes connectivity problems occur when the domain name being used cannot be mapped to the proper IP address.
- Ping a well known address on the Internet.
- Ping the IP address of the remote softswitch or IP PBX.

I do not receive dial tone when going "off hook" or my phone will not register with the softswitch/IP PBX.

- Verify the configurations on the *VoIP ALG* page.
- Check that the ALG registration code is configured.

1. Select System.

2. Select registration code.

- Attempt to ping the softswitch using the ping tool in the web GUI.

3. Select System.

4. Select System Overview.

5. Select Network Test Tools.
6. Enter the softswitch address in the IP Address to Ping field.
7. Press Ping.

Specifications

WAN Ports	1xT1 CSU/DSU or 10/100 Ethernet
LAN Ports	4x10/100 Ethernet (switched)
Serial Ports	1xRS-232
Dimensions	Height (1.7”), Width (10”), Depth (7”)
Weight	2 lb
Power	12V 3A
Warranty	1 Year
Environmental	Operating Temperature: 5° to 40°C Humidity: 20% to 80%, non-condensing

Regulatory Notices

Important Safeguards

Read and understand the following instructions before using the system:

- Close supervision is necessary when the system is used by or near children. Do not leave unattended while in use.
- Only use electrical extension cords with a current rating at least equal to that of the system.
- Always disconnect the system from power before cleaning and servicing and when not in use.
- Do not spray liquids directly onto the system when cleaning. Always apply the liquid first to a static free cloth.
- Do not immerse the system in any liquid or place any liquids on it.
- Do not disassemble this system. To reduce the risk of shock and to maintain the warranty on the system, a qualified technician must perform service or repair work.
- Connect this appliance to a grounded outlet.
- Only connect the system to surge protected power outlets.
- Keep ventilation openings free of any obstructions.

SAVE THESE INSTRUCTIONS.

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END OF TERMS AND CONDITIONS

FCC PART 68 NOTICE TO USERS OF DIGITAL SERVICE

This equipment complies with Part 68 of the FCC Rules and the requirements adopted by ACTA. On the bottom surface of this equipment is a label that contains, among other information, a product identifier in the format US: EWRDENAN4300T. If requested, this number must be provided to the telephone company.

The following instructions are provided to ensure compliance with the Federal Communications Commission (FCC) Rules, Part 68.

(1) This device must only be connected to the T1 WAN network.

(2) Before connecting your unit, you must inform the telephone company of the following information:

Port ID	REN/SOC	FIC	USOC
T1 WAN	04DU9-DN, 04DU9-BN	6.0N	RJ48C

(3) If the unit appears to be malfunctioning, it should be disconnected from the telephone lines until you learn if your equipment or the telephone line is the source of the trouble. If your equipment needs repair, it should not be reconnected until it is repaired.

(4) If the telephone company finds that this equipment is exceeding tolerable parameters, the telephone company can temporarily disconnect service, although they will attempt to give you advance notice if possible.

(5) Under the FCC Rules, no customer is authorized to repair this equipment. This restriction applies regardless of whether the equipment is in or out of warranty.

(6) If the telephone company alters their equipment in a manner that will affect use of this device, they must give you advance warning so as to give you the opportunity for uninterrupted service. You will be advised of your right to file a complaint with the FCC.

(7) In the event of equipment malfunction, all repairs should be performed by our Company or an authorized agent. It is the responsibility of users requiring service to report the need for service to our Company or to one of our authorized agents.

INDUSTRY CANADA (IC) 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.

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. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be coordinated by a representative designated by the supplier. Any repairs or alterations made by a user to this equipment, or equipment malfunctions, may give the telephone communications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection, that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas”.

Caution

Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.”

WARRANTY AND REPAIR SERVICE CENTER:

The RAM Group

Kent McDonald

kent.macdonald@theramgroup.com

(403) 266-5840 x 100
This Class (B) digital apparatus complies with Canadian ICES-003.