

Reference Guide



SmartVoice Gateway Series


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IMPORTANT NOTE

To ensure your Gateway complies with Australian Standards:

- 1) The symbol  shall be marked next to the protective earthing terminal (not GND) and the terminal shall be permanently connected via building wire.
- 2) The colour of the protective earth conductor insulation shall be green-and-yellow.
- 3) The protective earthing conductor shall be permanently connected first prior to other connections (eg. telephone cables) and removed last after all other cables have been disconnected.
- 4) The protective earthing shall not rely on telecommunications network or a cable distribution system.

1. Introduction

Product Overview

The NetComm SmartVoice series of stand-alone VoIP Gateways carry both voice and facsimile over an IP network. They support the SIP industry standard call control protocol to be compatible with free registration services or VoIP service providers' systems and they work in two different modes: UA (User Agent) or Server. As a standard user agent, the gateways are compatible with all well-known Soft Switches and SIP proxy servers. With optional server software, the gateways can also be configured to establish a private VoIP network over the Internet without a 3rd party SIP Proxy Server.

Each of the SmartVoice series VoIP gateways have a different combination of FXS and FXO ports; FXS ports are used to connect directly to a telephone set and FXO ports are used to connect directly to PSTN lines. There are a number of SmartVoice Gateway models available with 4 ports, 8 ports, 16 ports and 32 ports combinations. The gateways can be seamlessly integrated into existing networks by connecting to a phone set, PBX, key telephone system, fax machine or PSTN line. With only a broadband connection such as an ADSL bridge/router, Cable Modem or leased line router, the gateways allow you to gain access to voice and fax services over an IP network thereby reducing the cost of international and long distance calls.

In addition, each gateway includes an in-built 4 port Ethernet switch which supports comprehensive Internet gateway functions to accommodate other PCs or IP devices to share the same broadband stream. The QoS function allows voice and data traffic to flow through at the highest priority, and with TOS/DiffServ bit enabled, it guarantees that voice packets will have the first priority to pass through a TOS/DiffServ enabled router.

With the support of DDNS, the gateways are reachable by a domain name where the IP address is dynamically assigned by the ISP, thereby allowing users to host a web site or mail server in a PPPoE or DHCP network.

Each gateway can be assigned with a fixed IP address or by DHCP, PPPoE, adopting the G.711, G.726, G.729A or G.723.1 voice compression format to save the network bandwidth while providing real-time and toll quality voice. In addition, in the event that the power supply fails or the Internet connection is lost, each gateway can divert the FXS lines to the PSTN network on the FXO ports so that users can still use their conventional PSTN line to make calls.



IMPORTANT SAFETY INSTRUCTIONS

- When using your telephone equipment, basic safety precautions should always be followed to reduce the risk of fire, electric shock and injury to persons, including the following:
 - Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electric shock from lightning.
 - Do not use the telephone to report a gas leak in the vicinity of the leak.
 - Use only the power cord indicated in this manual.

Items that you will need and things you will need to know

To correctly install and set up your VoIP Gateway you will need to ensure that the required broadband hardware and services are setup and configured in the appropriate way to integrate this equipment.

Broadband Service

- You will need access to a high speed broadband service via a RJ 45 (Ethernet) connection to a broadband modem.

VoIP Service

- You will need a VoIP account from your VoIP service provider. The quality of your voice service will depend on your service provider's network. (Note: With the SmartVoice gateway(s), you will be able to build your private VoIP network but with limited voice call reachability.)

Computer/Network

- You will need a PC or laptop with a network card and Web browsing application (for configuration only).

NetComm SmartVoice Gateway Features

WAN

- One 10/100Mbps auto-negotiation, auto-crossover RJ-45 Ethernet port
- Support static IP, PPPoE, Bigpond Cable and DHCP address assignment and dynamic DNS (DDNS)
- QoS: IP TOS (Type of Services) and DiffServ (Differentiated Services) for both SIP signaling and RTP
- NAT Traversal : Port Forwarding, STUN, UPnP and Outbound Proxy
- NTP: (Network Time Protocol RFC 1305), Accepts up to 3 Time Server
- Time Zone Support
- MAC Address Clone
- RTP Packet Summary : packet sent, packet received, packet loss for voice quality analysis

LAN

- Four 10/100Mbps auto-negotiation, auto-crossover RJ 45 Ethernet ports
- Supports router and bridge mode (NAT mode and Non-NAT mode)
- DHCP server

Advance Firewall and DoS Protection

- NAT (Network Address Translation) and PAT (Port Address Translation)
- DMZ, Virtual Server
- Traffic Filtering based on MAC address, IP address, TCP/UDP Port number and URL string pattern
- Prevents attacks based on TCP, UDP, IP and ICMP protocols
- Prevents attacks such as SYN Flood, IP Spoofing, Ping of Death, Tear Drop, etc.

Voice Features

- SIP (RFC3261) compatible
- Voice codecs : G.711 a /ulaw, G.726, G.729A, G.723.1
- CNG (Comfort Noise Generation)
- VAD (Voice Activity Detection)
- Silence suppression/detection
- G.165/G.168 echo cancellation
- Adjustable Jitter Buffer and programmable Gain Control
- In-Band DTMF, Out-Of-Band DTMF relay (RFC2833, SIP INFO)
- Multiple SIP Proxy server entries with failover mechanism
- Polarity reversal detection (FXO/PSTN) and generation (FXS)
- T.30 (G.III) / Real time T.38 / Secured T.38 FAX relay
- DTMF, FSK (Bellcore & ETSI) Caller ID detection and generation.
- Support Caller ID Restriction (CLIR)
- FXO hang up detection : Busy Tone cadence auto learning/detection*
- Support VoIP-to-PSTN (transit out) / PSTN-to-VoIP (transit in) applications*
- Digit Map for dial plan

- Speed Dial
- Local phone book for peer-to-peer calling
- E.164 Numbering & ENUM support
- Hot-Line, Warm-Line support
- Single Number / Account (reprehensive number) for multiple ports
- Group Hunting function
- Recordable greeting message
- Proprietary server and protocol enables small businesses to build private VoIP network; and use standard SIP network as same time – for dual network support
- Supplementary call features**:
 - o Call Hold, Call Waiting, Call Pickup
 - o Call Forward - Unconditional, Busy, No Answer
 - o Call Transfer - Unattended, Attended
 - o Three Way Calling (Media Server required)
- Failsafe mechanism :
 - o Network failure
 - o Service unavailable
 - o Power loss
- Analogue interface
 - o Connector : RJ-11
 - o Signaling protocol : Loop Start

Configuration & Maintenance

- Configuration methods:
 - o Web
 - o IVR
 - o Telnet
- Status reports:
 - o Port status
 - o Registration status
 - o Ping tests
 - o STUN/UPnP status
 - o Hardware / software information
- Firmware Upgrade through TFTP, FTP and proprietary image server
- Configuration Backup/Restore
- Reset button (with restore factory default function)
- Front Panel LED : voice ports, WAN, LAN1~4, Run, Power, Alarm
- Optional Auto Provisioning Server (APS) for mass deployment
- SNMP V1 / V2c (optional)

General Specifications

- Power Adaptor : AC 100V~240V 50/60Hz input, DC 12V output
- Temperature : Operation 0 °C ~ 45 °C
- Storage -25 °C ~ 75 °C
- Humidity : up to 90% RH, non-condensing
- 4-port Model Dimension (W/D/H) 4-port: 202 x 172 x 35 mm
- 8-port Model Dimension (W/D/H) 4-port: 287 x 161 x 38 mm
- 16-port Model Dimension (W/D/H) 4-port: 442 x 325 x 43 mm
- 32-port Model Dimension (W/D/H) 4-port: 442 x 325 x 43 mm
- 4-port Model Weight: 0.43Kg
- 8-port Model Weight: 1.34Kg
- 16-port Model Weight: 4.13Kg
- 32-port Model Weight: 4.13Kg

Note*: Features are only available in models which have FXO ports.

Note**: The availability of these features may depend on your VoIP service provider's network

SmartVoice Gateway Hardware Description

Front Panel (all models)



- Power Indicator: Green light indicates a normal power supply.
 - Run Indicator: Blinking green light indicates normal operation.
 - Alarm Indicator: When the system starts up, the red light will blink. It also indicates abnormal gateway behaviour.
 - P1 – PX stands for Port 1 – Port X (or Line 1 – Line X)
 - WAN stands for the WAN Port Indicator.
 - L1 – LX stands for the LAN Port Indicator.
- ✓ When starting up the system, the **Alarm**, **Run**, and **Power** indicators will light up. After approximately 15 seconds, the Alarm indicator will go off, the Run indicator will blink green, and the Power indicator will stay green under operational conditions. If the Alarm indicator continues to blink, it means the system is currently communicating with the ISP and has yet to obtain an IP address.
- ✓ When the **WAN** is connected, the WAN indicator will light up green and if data is being transmitted over the Internet, the indicator blinks green and orange.

Rear Panel (all models)

Phone ports are telephone ports (FXS) to be connected to telephone sets or PBX CO ports. Line ports (FXO) are to be connected to PSTN lines (telephone lines from a PSTN network).



NOTE: Do not connect FXS ports to each other. Also, do not connect any FXS port directly to a PSTN line or internal PBX. Doing so may damage VoIP gateway.

Restore the Gateway

To restore the gateway to its factory default settings (IP address, Username and Password only):

- (1) Disconnect the power plug.
- (2) Press reset (do not let go of the reset button).
- (3) Connect the power plug back into the socket (do not let go of the reset button).
- (4) Let go of the reset button after 6 seconds. The default IP address and username/password for web login will be restored.



NOTE: This reset action only resets the gateway IP and username/password settings. Other settings still remain in the gateway's flash memory. A complete hardware reset can be done through the gateway's IVR system and Web configuration interface.

2. Installation and Applications

Connecting your SmartVoice Gateway(s)

1. Connect the WAN port of your SmartVoice Gateway to your Broadband Modem using an ethernet cable.
2. Connect the power adapter to your SmartVoice Gateway and plug it into a powerpoint.



Please use the power adapter that comes with VoIP gateway. Using an adapter other than the one supplied with the VoIP gateway, may cause problems and will affect the warranty of the product.

3. Connect the Phone or PSTN lines as required to setup your SmartVoice Gateway. The way you setup the lines will depend on the model you have purchased. Refer to the following sections on both Network Connections and Telephone Connections for more information on the possible applications of your SmartVoice Gateway.

Once you have connected your gateway you can configure it using one of the two configuration options provided:

- 1) Web Interface
- 2) Telephone IVR (Interactive Voice Response)

The IVR provides basic query and configuration, while the Web interface provides a full configuration.

Network Connections

Possible network connections are divided into 3 basic modes as described below:

- Gateway can be assigned with a Public IP Address
- Gateway can be installed under the existing NAT
- Gateway can be assigned with a Public IP address and serve as an IP sharing router.

Gateway Assigned with a Public IP Address

The VoIP gateway will have a Public IP address for its Internet connection regardless of whether it is a static IP address, DHCP (using a Cable Modem), or PPPoE (Dialup / ADSL).

The following diagram shows a typical network connection in which broadband devices, such as an ADSL router or Cable modem, need to work in bridged mode so that the gateway is able to obtain a public IP address either through PPPoE, DHCP or static IP.



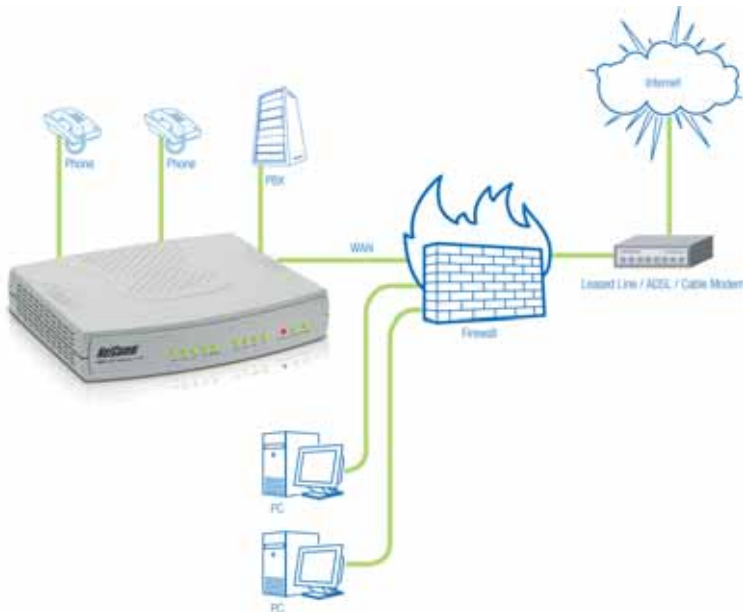
Gateway IP Settings	Need to be set up as static IP, DHCP, or PPPoE
NAT/STUN Settings	Unnecessary (Disabled)
DDNS Settings	Unnecessary (Disabled)

Note: The above diagram shows a 4-port SmartVoice gateway for illustration purposes. An 8, 16 or 32 port SmartVoice gateway could also be used depending upon your network requirements.

Gateway behind a NAT/Firewall Network

The VoIP gateway uses a private IP address and the Internet sharing function of other systems to connect to the Internet.

The following diagram shows a typical network connection in which broadband devices such as an ADSL router work in router (NAT) mode so that the gateway is behind a NAT/Firewall. In this case, the broadband router (NAT/Firewall) and the gateway need to be configured correctly in order to allow the gateway to communicate with the Internet and VoIP network properly, using different techniques to traverse the NAT/Firewall.



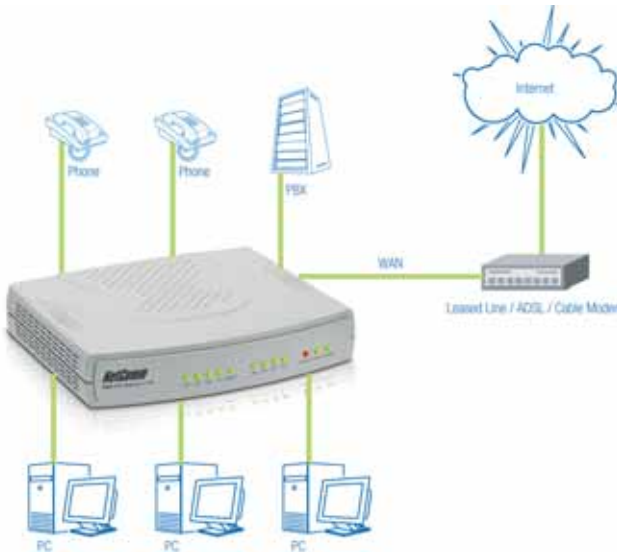
Note: The above diagram shows a 4-port SmartVoice gateway for illustration purposes. An 8, 16 or 32 port SmartVoice gateway could also be used depending upon your network requirements.

The table shown below lists possible solutions for different network scenarios.

LAN IP address of Internet sharing	Please avoid IP address 192.168.8.1-192.168.8.254 (You may need to change the settings of Internet sharing device or change the SmartVoice gateway LAN Port IP address)	
WAN IP of SmartVoice gateway	Set as a static private IP address or DHCP; when the WAN port is set as a static private IP, assign the LAN IP address of the Internet sharing to as default gateway IP of SmartVoice gateway; When the WAN IP is set as DHCP, please make sure you have DHCP server running in network to issue an IP to the SmartVoice gateway.	
NAT Traversal There are three ways to configure a SmartVoice gateway to traversing NAT in Internet sharing devices. For details, please refer to page 26.	NAT Public IP	<p>If the WAN of the IP sharing device has a static public IP address, then the "NAT IP/Domain" address is set as the Public IP address of the Internet sharing devices.</p> <p>If the WAN of the Internet sharing device uses a dynamic public IP address and you specify a domain name in "NAT IP/Domain" field, which has to comply with the DDNS settings.</p> <p>Note: you also need to forward specific ports on the NAT router to the WAN port of the SmartVoice gateway.</p>
	STUN client	Enable STUN client will avoid problems of configuring port forwarding in the Internet sharing device, but some NAT routers do not support STUN.
	UPnP Control point	Using UPnP to enable the SmartVoice gateway to pass through NAT of Internet sharing device. This function only works when the Internet sharing devices support UPnP and has it enabled.
DDNS Settings For details, please refer to page 27 for the DDNS Settings	The WAN of the Internet sharing device has a static public IP address.	(Unnecessary) Disabled
	The WAN of the Internet sharing device has a dynamic IP address.	Enabled: enter the registered URL (Uniform Resource Locator) and DDNS server info into corresponding areas.

Gateway Assigned with a Public IP Address and Serving as an Internet Sharing Device

The SmartVoice gateway will have a Public IP address, regardless of whether it is the public IP from a static setting, DHCP (using a Cable Modem), or PPPoE (to connect to your ADSL account), and can use the built-in Internet sharing functions to allow other PCs attached to SmartVoice gateway LAN ports to share the Internet connection. The network connection is shown in following diagram.



Gateway IP Settings	Needs to be set up as either in static IP, DHCP, or PPPoE
NAT/STUN Settings	Unnecessary (Disabled)
DDNS Settings	Unnecessary (Disabled)
DHCP service for PCs, please refer to page 24 for the gateway DHCP server settings	Default private IP addresses of the gateway LAN ranging from: 192.168.8.1-192.168.8.253 Subnet Mask: 255.255.255.0 Default Gateway: 192.168.8.254 PC can be set either in DHCP or static IP mode

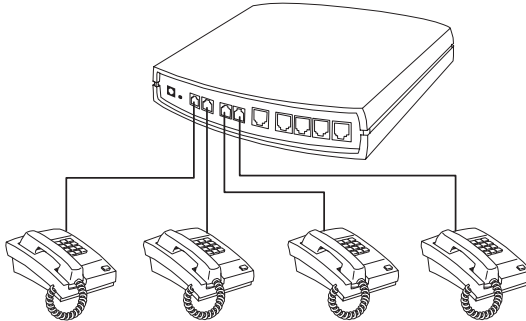
Note: The above diagram shows a 4-port SmartVoice gateway for illustration purposes. An 8, 16 or 32 port SmartVoice gateway could also be used depending upon your network requirements.

Telephone Connection

Example of SmartVoice Gateway using FXS interfaces

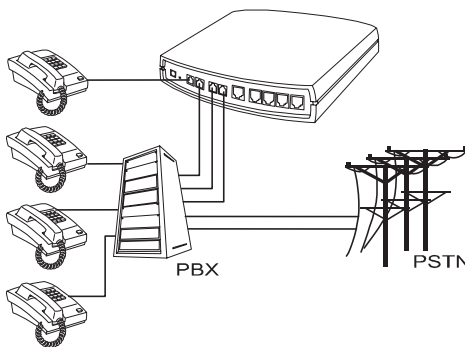
SmartVoice Gateway connecting directly to phone sets

After connecting telephone sets to Phone1-Phone4, users can make direct calls (Phone1-Phone4 are FXS interfaces). Each set acts as an independent extension line.



Integrating the SmartVoice Gateway with PBX

Phone1-Phone4 are FXS interfaces, and some of them can be connected to telephone sets for direct calls. Others can be connected to the PBX CO lines so that other extension lines can make VoIP calls.

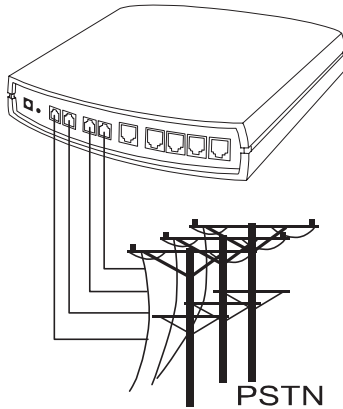


Note: The above diagrams show a 4-port SmartVoice gateway for illustration purposes. An 8, 16 or 32 port SmartVoice gateway could also be used depending upon your network requirements.

Example of SmartVoice Gateway using FXO interfaces

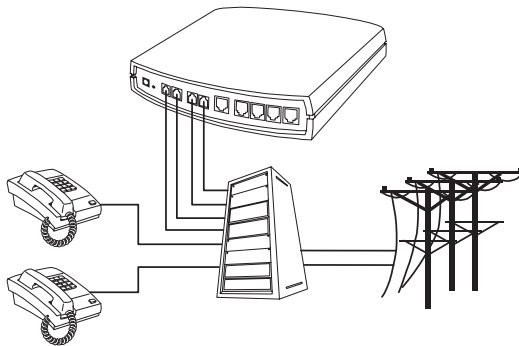
SmartVoice Gateway connecting directly to the Telephone Line of a PSTN

Line1-Line4 are FXO interfaces and can all be connected to a PSTN to serve as a bridge between the PSTN and VoIP network.



Integrating the SmartVoice Gateway with PBX

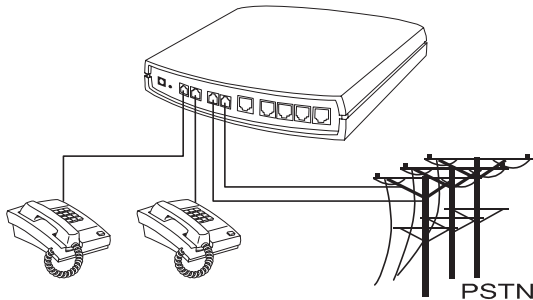
Line1-Line4 are FXO interfaces and can be connected with PBX extension lines (this is exclusively for an analog interface, and is not applicable for a digital interface).



Note: The above diagrams show a 4-port SmartVoice gateway for illustration purposes. An 8, 16 or 32 port SmartVoice gateway could also be used depending upon your network requirements.

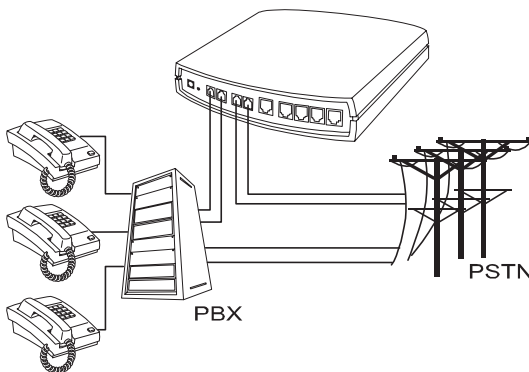
Example of SmartVoice Gateway using a combination of FXS and FXO ports

Phone1-Phone2 are FXS interfaces and can be directly connected to telephone sets for direct calls. Line1-Line2 are FXO interfaces and can be connected to PSTN lines to serve as a bridge between the PSTN and other VoIP telephones. The system also allows a call to be made from a traditional telephone line to connect with a user behind the SmartVoice Gateway.



Integrating the SmartVoice Gateway with PBX

Phone1-Phone2 are FXS interfaces and can be connected to PBX CO lines; Line1-Line2 are FXO interfaces and can be connected to a PSTN to act as a bridge between the PSTN and other VoIP telephones. The system also allows a call to be made from a traditional telephone line to connect with a user behind the SmartVoice Gateway.



Note: The above diagrams show a 4-port SmartVoice gateway for illustration purposes. An 8, 16 or 32 port SmartVoice gateway could also be used depending upon your network requirements.

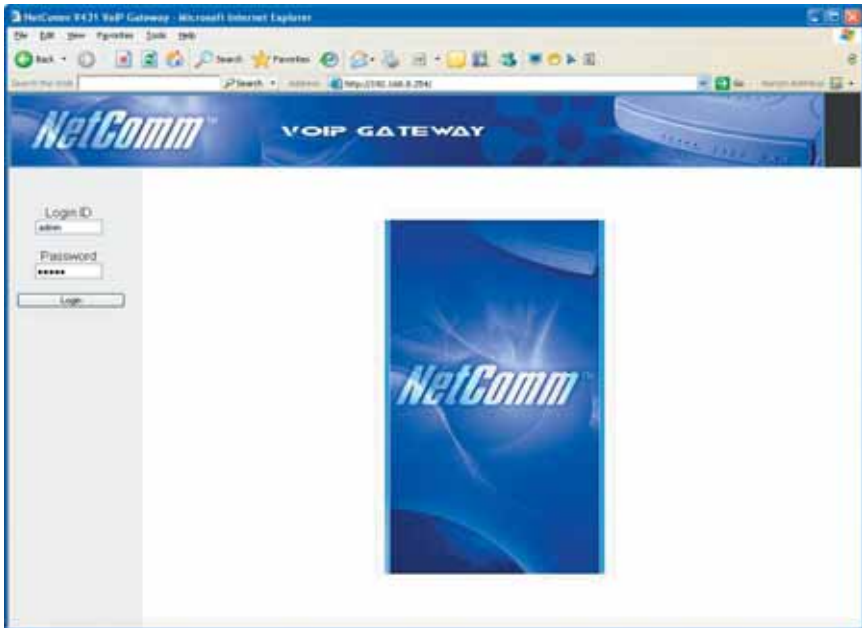
3. Configuring the Gateway via WEB Browser

The gateway also allows users to change gateway settings using a web browser. After opening a browser window, enter the gateway's IP address as the website address in order to enter the Web configuration screen as shown in the following diagram. (IE Browser used for example: Enter http://192.168.1.2.)

The factory default WAN IP address for Gateway is **192.168.1.2**. You can also enter "101" from the handset to inquire about the current WAN Port IP address. The factory default LAN Port IP address is **192.168.8.254**.

Instructions

- Open an Internet browser window.
- Enter the gateway's WAN Port IP address in the website address area (If the PC is connected to the LAN Port, enter the LAN Port IP address. The default is **192.168.8.254**)
- The following registration screen will appear (The factory default settings for both Login ID and Password are set to 'admin').



- After completing and confirming the settings, some of the settings will take effect immediately, but some settings e.g. network related settings will only take effect after the gateway is restarted. Please go to **System Operation** to save the settings before restarting the system.

The gateway doesn't allow multiple users to configure the gateway at same time. Once a user logs into the gateway, other users from different IP addresses cannot login at the same time. Please remember to logout or restart the system if you are not using the web configuration function.

Network Settings

The network settings are used to set the gateway's communication ports, IP configurations, DNS and DHCP server etc.

Network Settings (WAN)		
Current WAN IP Address 192.168.1.192		
Listen Port UDP [1 - 65535]	5060	
RTP Starting Port UDP [1 - 65500]	9000	
<input checked="" type="radio"/> DHCP		
<input type="radio"/> Static IP	IP Address	192.168.1.2
	Subnet mask	255.255.255.0
	Default Gateway IP	192.168.1.254
<input type="radio"/> PPPoE	PPPoE Account	
	PPPoE Password	*****
	Confirm Password	*****
<input type="radio"/> BigPond Cable	User Name	
	BigPond Cable Password	*****
	Confirm Password	*****
Domain Name Server Assignment <input checked="" type="radio"/> Auto <input type="radio"/> Manual		
Domain Name Server (Primary) IP 192.168.1.1		
Domain Name Server (Secondary) IP		
Factory Default MAC Address	XXXXXXXXXX <input type="button" value="Restore"/>	
Your MAC Address	XXXXXXXXXX <input type="button" value="Done"/>	
Current MAC Address		

- **Current WAN IP Address:** The IP address of the WAN port.
- **Listen Port UDP:** It is not necessary to change the protocol of the communication port used by the gateway, unless it conflicts with ports used by another device in your network.
- **RTP Starting Port UDP:** The initial value of the port number for transmitting voice data among gateway(s). Each line requires 2 ports. For example, the 400 series gateway requires 8 UDP ports and the 800 series gateway requires 16 UDP ports. It is not necessary to change these, unless it conflicts with ports used by another device in your network.

For example, if the starting port is 9000, then Line 1 is using ports 9000 and 9001, and Line 2 is using ports 9002 and 9003, and so forth.

IP Configuration (Setting WAN Port)

There are three methods of obtaining a WAN IP address:

1. Static IP
2. DHCP, means a Dynamic IP (Cable Modem)
3. PPPoE (Dialup ADSL)

Using the DHCP and PPPoE for obtaining an IP address may vary. If you are not familiar with the network connection, please contact your local ISP.

Setting Dynamic IP (DHCP)

DHCP <input checked="" type="radio"/>	<input type="text"/>
---------------------------------------	----------------------

Click “DHCP” to obtain a Dynamic IP address, and then click the “Accept” button at the bottom of the screen.

To save the settings click **System Operation** to select “Save Settings”, “Restart”, and then click the “Accept” button. Wait for a while (about 40 seconds), and the system will obtain the related IP value from the DHCP Server.



NOTE: After the system has obtained a new IP address, if using WAN Port to enter the Web Configuration Screen, a new IP address has to be used. The same applies to the following two settings.

Setting Static IP

Static IP <input checked="" type="radio"/>	IP address	<input type="text" value="192.168.1.2"/>
	Subnet mask	<input type="text" value="255.255.255.0"/>
	Default Gateway IP	<input type="text" value="192.168.1.254"/>

Select “Static IP” and enter the IP address, Subnet Mask and Default Gateway values. Then click the “Accept” button at the bottom of the screen.

Save the settings, and then restart the system. Wait for about 40 seconds for the system to restart.

ADSL PPPoE Settings

PPPoE <input checked="" type="radio"/>	PPPoE Account	<input type="text" value="PPPoE"/>
	PPPoE Password	<input type="text" value="****"/>
	Confirm Password	<input type="text" value="****"/>

Select “PPPoE” and enter the Account Number, Password and re-enter Password to confirm. Then click the “Accept” button at the bottom.

Save the settings, and then restart the system. The system will take about 40 seconds to restart.

BigPond

BigPond Cable <input type="radio"/>	User Name	<input type="text"/>
	BigPond Cable Password	<input type="text"/>
	Confirm Password	<input type="text"/>
	Login Server	<input type="text"/>

Click "BigPond Cable". Enter User Name and Password and then click the "Accept" button at the bottom.

DNS Settings

Domain Name Server Assignment	<input type="radio"/> Auto <input checked="" type="radio"/> Manual
Domain Name Server (Primary) IP	<input type="text" value="168.95.1.1"/>
Domain Name Server (Secondary) IP	<input type="text"/>

Domain Name Server (DNS): While a gateway is accessing another gateway or computer with a hostname, it will look up the IP address from the DNS provided by the ISP. The ISP whilst negotiating with PPPoE or DHCP usually assigns the DNS information. In the case that the DNS is not assigned automatically or WAN port is assigned with a static IP address, the DNS information must be assigned manually.

Auto

Gateway uses primary & secondary DNS addresses from ISP's DHCP server or PPPoE server.

Manual

Enter the primary & secondary DNS addresses manually. Please be sure the IP addresses are correct otherwise the gateway will not be able to access any hosts with their hostname.

Clone MAC

Factory Default MAC Address	<input type="text" value="000C2A050341"/>	<input type="button" value="Restore"/>
Your MAC Address	<input type="text" value="00001CD4AFOA"/>	<input type="button" value="Clone"/>
Current MAC Address	<input type="text" value="000C2A050341"/>	

Some Internet Service Providers (ISP) assign the bandwidth via the MAC (Media Access Control) address. You can click the "Clone" button to type in a MAC address which is recognized by ISP. It is only necessary to fill in the field if required by your ISP.

The "Your MAC Address" will be blank as you log in through WAN port.

LAN interface mode

LAN interface mode

Router
 Bridge

- **Router:** The system serves as a router.
- **Bridge:** The system serves as a bridge between WAN port and LAN port. (LAN default gateway will still be accessible for configuration).

Network Settings (LAN)

LAN IP / LAN default Gateway	<input type="text" value="192.168.8.254"/>	Subnet mask	<input type="text" value="255.255.255.0"/>
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- **Network Settings (LAN):** The gateway's LAN IP address and subnet mask value. Please note that the gateway is built under NAT. The gateway's LAN IP address cannot be in the same section as the Internet sharing devices, or else it is unable to make or receive calls. For example, if the LAN IP address of the Internet sharing device is 192.168.8.1, then the gateway's LAN IP address cannot be in the range between 192.168.8.1 ~ 192.168.8.254. If it is, please change the LAN IP address of the gateway (e.g. set the IP address to 192.168.99.254 instead).

DHCP Server			
Enable DHCP Server	<input checked="" type="checkbox"/>		
IP Pool Starting Address	<input type="text" value="192.168.8.1"/>	IP Pool ending Address	<input type="text" value="192.168.8.250"/>
Lease Time[1 - 9999hours]	<input type="text" value="1"/>		

- **Enable DHCP Server:** Enable or Disable DHCP server service of gateway.
- **IP Pool Starting Address:** The first IP address to be assigned to DHCP clients.
- **IP Pool ending Address:** The last IP address to be assigned to DHCP clients.
- **Lease Time:** The valid period of an assigned IP address.

Domain Name Server Assignment	<input checked="" type="radio"/> Auto <input type="radio"/> Manual		
Domain Name Server (Primary) IP	<input type="text"/>	Domain Name Server (Secondary) IP	<input type="text"/>

- **Domain Name Server Assignment:** The DNS information to be assigned to DHCP clients.
Auto : Assigns the same DNS information of WAN port to the DHCP clients.
Manual : Manually assign the DNS information for DHCP clients.

Port of Web Access from WAN[0 - 65535]	<input type="text" value="80"/>
--	---------------------------------

- **Port of Web Access from WAN:** Http port for WAN. To make this setting, the web configuration must be performed via the gateway's LAN port. The gateway always uses port 80 for Http connections via LAN ports of the gateway.



NOTE: For network security concern, Web Access for WAN port is disabled by default (port number "0" in this option means disable the Web Access). To enable it simply type in a valid port number in this field.

QoS Settings

WAN QoS

WAN QoS		
<input type="checkbox"/> QoS	Downstream Bandwidth	Full
	Upstream Bandwidth	Full
ToS / DiffServ Settings		
ToS IP Precedence	Signaling Precedence	3 (Flash)
	Voice Data Precedence	5 (CRITIC / ECP)
DiffServ (DSCP) <input type="radio"/>	Signaling Value	26 (Assured Forwarding Class 3 - Low Drop Precedence, AF31)
	Voice Data Value	46 (Expedited Forwarding, EF)

- QoS (Quality of Service):** Sets true bandwidth of your Internet connection to ensure sound quality during transmission. (When this function is enabled, the voice packet has the highest priority to ensure telecommunication quality while less bandwidth is assigned for data transmission). Some models of the VoIP gateway without this function can adjust the bandwidth automatically.



NOTE: The WAN QoS function only works as expected when you connect/configure the gateway as described on Page 13. If the gateway is behind an Internet sharing device as described in Page 14, QoS for VoIP traffic should be guaranteed by your network routers and not the VoIP gateway.

- ToS/DiffServ (Type of Service/DSCP):** The voice packet has the highest priority to ensure telecommunication quality; the larger the value you set, the higher priority you will get.



NOTE: Please contact your ISP when you configure these values.

NAT/DDNS (NAT Traversal)

NAT Traversal	
NAT Public IP <input checked="" type="checkbox"/>	NAT IP/Domain <input type="text" value="hostname.ddnsserve.com"/>
Enable STUN Client <input type="checkbox"/>	STUN Server IP / Domain <input type="text"/>
	STUN Server Port(1024 - 65535) <input type="text" value="3478"/>
Enable UPnP Control Point <input type="checkbox"/>	

If a gateway is set up behind an Internet sharing device, you can select either the NAT or STUN protocol.

- **NAT Public IP:** The IP address used by the gateway should be a private address. Furthermore, users must set the Virtual Server Mapping in the Internet sharing device (For example, a virtual server is usually defined as a Service Port, and all requests to this port will be redirected to this specified server's private IP address).

The default ports of the gateway are listed below:

- **Listen Port (UDP):** 5060
- **RTP Starting Port (UDP):** 9000~9015 (Listen Port used for telephone communication).
- **Port of Web Access from WAN (TCP):** the number you specified in this option in Network Settings page.



NOTE: You need to configure your Internet sharing device to forward the above ports to the WAN IP address of the gateway.

- **NAT IP/Domain:** Enter the NAT Server IP address (real public IP address of the Internet sharing device); or fill in a true URL (Uniform Resource Locator) when DDNS is used. Please refer to the DDNS settings.



NOTE: If setting a public IP in this field, it has to be a static public IP, otherwise VoIP communication may not be established properly. Please contact your ISP to check whether your Internet connection has static public IP addresses.

- **Enable STUN Client:** Using the STUN protocol prevents problems with setting the IP sharing function, but some NAT does not support this protocol.



NOTE: You can use the “Status - STUN Inquiry” page to detect type of NAT of your Internet sharing device. If the NAT type is “Symmetric NAT”, then the gateway is not able to traverse the NAT. It is not a flaw of the gateway design, but limitations of the STUN protocol.

- **STUN Server IP/Domain and Port:** Enter the STUN server IP address and Listen Port number. You can set 2 STUN servers separated by a semicolon.
- **Enable UPnP Control Point:** To enable the gateway's IP traffic to pass through an Internet sharing device. This function only works when the Internet sharing device supports UPnP and has it enabled.



NOTE: The “Status - Current Status” page will show the status of UPnP.

<input type="checkbox"/> Register to DDNS	
DynDNS DDNS Server	<input type="button" value="Default"/>
Server Address	<input type="text" value="members.dyndns.org"/>
Hostname	<input type="text" value="dyndns.org"/>
Login ID	<input type="text"/>
Password	<input type="password" value="....."/>
Confirm Password	<input type="password" value="....."/>
Behind NAT	<input type="checkbox"/> Yes
Custom	<input type="checkbox"/>

These settings are only necessary when the gateway is set up behind an Internet sharing device that uses a dynamic IP address and does not support DDNS.

You need to apply for an account with a DDNS service from **www.dyndns.org** before you type in the following information.

- **Server address:** Sets up the IP address or URL (Uniform Resource Locator) of the DDNS Server.
- **Hostname:** The URL of the system (or NAT) – apply from domain name registration providers (e.g. www.dyndns.org).
- **Login ID and Password:** The ID and password are used to log into the DDNS server.
- **Behind NAT:** Select only when the system is set up behind a NAT device.



NOTE: If the gateway is set up under NAT, then enter the hostname into the NAT IP/Domain that is the same with Hostname of DDNS.

Example:

NAT

NAT Traversal	
NAT Public IP <input checked="" type="checkbox"/>	NAT IP/Domain <input type="text" value="hostname.ddnserv.com"/>

DDNS

DynDNS DDNS Server	<input type="button" value="Default"/>
Server Address	<input type="text" value="members.dyndns.org"/>
Hostname	<input type="text" value="hostname.ddnserv.com"/>

Telephony Settings

Prefix Number Rules

- **Trunk Dial Out Verify/ Trunk Dial Out Replace:** The system will transfer the number for all transit out calls through the FXO port. For example: If you transit out with **01907123456**, the system will replace it with **190601 907123456**. If you transit out with **008621123456** the system will replace it with **190200 8621123456**.
- **Trunk Dial Out Deny:** The system will deny the call with the leading number filled in this column.



NOTE: This rule only applies to transit out calls of gateway FXO ports.

FXS Caller ID Generation	<input type="radio"/> Disable <input type="radio"/> DTMF <input checked="" type="radio"/> FSK
FXO Caller ID Detection	<input checked="" type="checkbox"/> Detection Level: <input type="text" value="B"/>
FXS Caller ID Type	<input checked="" type="radio"/> Bellcore <input type="radio"/> ETSI
Anonymous Caller ID (CLIR)	<input type="checkbox"/>
Anonymous Transit in W/O Caller ID	<input type="checkbox"/>
Trunk Incoming Prompt Voice	<input checked="" type="radio"/> Default Greeting <input type="radio"/> Recorded voice file <input type="radio"/> Dial Tone
FXO Hunting VoIP call in option	Caller Indicate Dial-Out FXO Hunting Default Dial-Out: <input type="text"/>
FXO Line VoIP call in option	Caller Indicate Dial-Out

- **FXS Caller ID Generation:** Select this option to enable the caller ID display function on FXS ports. FSK is used for Australian caller ID standards.
- **FXO Caller ID Detection:** Used to detect the Caller ID delivered from the PSTN to the FXO port. When enabled, the Caller ID detected on the FXO port will be sent to the SIP Proxy Server on transit in (dialing out) calls.
- **Detection Level:** If FXO can't detect Caller ID, try to adjust it until it can.



NOTE: You have to enable "Wait for Caller ID before FXO / Trunk pick up" to ensure Caller ID is detected correctly.

- **FSK Caller ID Type:** Bellcore is used in Australia caller ID standards.
- **Anonymous Caller ID (CLIR):** When enabled, the gateway will use "anonymous" as caller identification to place calls.
- **Anonymous Transit in W/O Caller ID:** When enabled, if the FXO can detect caller ID in calls from PSTN, the gateway will use the detected caller ID as caller identification for transit in calls; if FXO cannot detect caller ID, the gateway will use "anonymous" as caller identification for transit in calls. When it not checked, if the FXO can detect caller ID in calls from PSTN, the gateway will use the detected caller ID as caller identification for transit in calls; if FXO cannot detect caller ID, the gateway will use its port phone number as caller identification for transit in calls.
- **Trunk Incoming Prompt Voice:** Select the greeting (must use the IVR 132 function to record a voice file) when FXO receives an inbound call (transit in).
- **FXO Hunting VoIP call in option:**

Caller Indicate Dial-Out: When users make calls to FXO ports from Internet, they will be required to dial a PSTN number after the calls connects to FXO port.

Default Dial-Out: When users make calls to FXO ports from Internet, the gateway will dial to PSTN with the number filled in "FXO Hunting Default Dial-Out".

- **FXO Hunting Default Dial-Out:**

This will take effect as **FXO Hunting VoIP call in option** is set to **Default Dial-Out**.

- **FXO Line VoIP call in options:**

The usage of this option is as same as **FXO Hunting VoIP call in option**. The Default Dial-Out refers to individual lines rather than a hunting group. The individual FXO line default dial out number can be set in Line configuration section as below.

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) (0 - 60 s)	Dial-Out Prefix	FXO Line Default Dial-Out	FXS Group	Enable FAX
1	<input checked="" type="checkbox"/>	FXS	<input type="checkbox"/>	<input type="text"/>	0			<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
2	<input checked="" type="checkbox"/>	FXS	<input type="checkbox"/>	<input type="text"/>	0			<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
3	<input checked="" type="checkbox"/>	FXS	<input type="checkbox"/>	<input type="text"/>	0			<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
4	<input checked="" type="checkbox"/>	FXO	<input type="checkbox"/>	<input type="text"/>	0	<input type="text"/>	<input type="text"/>		<input checked="" type="checkbox"/>

- **Enable:** To enable a line; if some lines are not used, disable them (Pause Function) to avoid unnecessary waiting when an incoming call is diverting to this line.

Hotline Functions

- **FXS port:** When the user picks up the phone, the gateway automatically dials your assigned hotline number. When in hotline mode, other phone numbers cannot be dialed.
- **FXO port:** When receiving a call from an outside line, the gateway will divert the call to the assigned hotline number.
- **Hot Line No.:** Enter the hot line number for automatic dialing function.
- **Warm Line:** When the Warm Line function is in use, user can dial a number. Otherwise the system will divert incoming calls from an outside line to the Hot Line Number after a set wait time.

Example:

Assume the assigned hotline for the SmartVoice Gateway line 3 is 701 and the Warm Line (Hot Line Delay) is 5 seconds. If no extension number is dialed within 5 seconds, the call will be automatically diverted to the assigned hotline (ext 701). The system allows users to record a voice prompt (e.g. "Please enter an extension number or wait for the operator to connect you") to use in this situation.

Assume the assigned hotline for the SmartVoice Gateway line 4 is 702 and the wait time is 0 second. When P4 receives a call from an outside line, it will be automatically diverted to extension 702.

- **Dial-out Prefix:** It is the number dialed automatically by the system when the FXO interface diverts a call to the PSTN by VoIP.
- **FXO Line Default Dial-Out:** Default number that FXO will dial out when it receive an incoming call from VoIP.

Example:

If PBX extension needs to dial "0" to make a PSTN call, and the FXO ports are connected to PBX extension. In this case, the Dial-out prefix should be set to "0". If the PBX requires some delay time before capturing a line, then the trunk prefix should be set as "0," so that after dialing a 0, it will pause for 1 second before dialing the destination number. Each comma represents a 1 second delay. If more delay time is required, simply add more commas. Please note that if a Dial-out prefix is set, the line won't be able to dial any PBX extension line (the FXS interface does not have a trunk prefix function).

- **Group Hunting:** Select group hunting when there is an incoming call and the gateway will automatically assign an unassigned call according to the Hunting Priority. If Line 2 does not want to be set as an assigned line to receive any inbound calls, the function can be disabled. Users can also use the Up or Down key to adjust hunting priority (No setting is required for the FXO interface).
- **Enable FAX:** Enable this line to detect if there is a FAX tone to transfer the codec.

Trunk Hunting Order	First Idle ▾
Transit in Busy Tone Limit [0 - 60 s]	3
Ring Time Limit [10 - 600 s]	90
Enable End of Digit Tone	<input type="checkbox"/>
VoIP Call Out Notification	<input type="checkbox"/>
Force Calling Thru PSTN Code	
Support One Way Media	<input checked="" type="checkbox"/>

Description:

- **Trunk Hunting Order:** To set FXO dial-out mode when there is an incoming call dialed FXO representative number. (This option only appears in the gateway which has more one FXO ports)
First Idle: The gateway will assign each unassigned call from first FXO line.
Sequential: The gateway will automatically assign the first unassigned call to the first FXO line. The second FXO line will dial the second unassigned call out. Each line will be used.
- **Transit in Busy Tone Limit:** The duration the VoIP gateway plays a busy tone before FXO port goes on-hook. Used to notify the caller from PSTN that this call is finished.
- **Ring Time Limit (10 - 600secs):** The timeout to cancel a call when no one answers.
- **Enable End of Digit Tone:** The VoIP gateway will play a “Beep-Beep” tone to notify that the call is in progress.
- **VoIP Calling Notification:** The gateway will play a tone to notify the call is through VoIP.
- **Force Calling Thru PSTN code:** Dial the code to get a PSTN line for dial out. For example: If you specify “33” in this option and would like dial “23456789” through PSTN, just dial “33 23456789”
- **Support One Way Media:** If it is disabled, the system will send RTP immediately when the connection with Proxy is set up. The default is enabled. If communicating with other Gateway has problem, please disable this function.

FXS Group Hunting / Ring Priority			
Hunting / Ring	Hunting	Sequential Ring Time [1 - 100 s]	
1	Line 1		
2	Line 2		
3	Line 3		

- **Hunting/Ring:** It is able to set FXS group hunting using simultaneous ring or sequential ring.
- **Sequential Ring Time:** To set the ring time of each port, when sequential ring is chosen.
- **Hunting Priority:** it can be adjusted using the Up or Down arrows.

SIP Settings

All Call through OutBound Proxy	<input type="checkbox"/>		
OutBound Proxy IP / Domain	<input type="text"/>	OutBound Proxy Port [1 - 65535]	<input type="text" value="5060"/>

- All Call through OutBound Proxy:** An outbound proxy server handles SIP call signaling as a standard SIP proxy server would. Furthermore, it receives and transmits phone conversation traffic (media) in between two communication parties. This option tells the gateway to send and receive all SIP packets to the destined outbound proxy server rather than the remote gateway. This helps VoIP calls to pass through any NAT protected network without additional settings or techniques. Please make sure your VoIP service provider supports outbound proxy services before you enable it.

Session Expiration [0=disable, 10 - 1800]	<input type="text" value="0"/>
Session Refresh Request	<input checked="" type="radio"/> UPDATE <input type="radio"/> re-INVITE
Session Refresher	<input checked="" type="radio"/> UAS <input type="radio"/> UAC

- Session Expiration:** It is to avoid the billing of abnormal dropping the call because of Internet. The default is disabled.
- Session Refresh Request:** to send the packet of UPDATE or re-INVITE to
- Session Refresher:** It is the gateway's role in Session Timer. UAS is an originator, and UAC is a replier.

E.164

International Call Prefix Digit	<input type="text"/>
Country Code	(Other) <input type="text"/>
Long Distance Call Prefix Digit	<input type="text"/>
Area Code	<input type="text"/>

E.164 Numbering	To Invite Proxy	<input type="checkbox"/>
	Transform to Transit Out	<input type="checkbox"/>
ENUM Header Exception	<input type="text" value="070"/>	

- **International Call Prefix Digit:** Enter the International call prefix.
- **Country Code:** Select the desired country code.
- **Long Distance Call Prefix Digit:** The long-distance prefix digit for making a long-distance call.
- **Area Code:** Please enter the area code.
- **E.164 Numbering:** To invite Proxy to follow the E.164 rule. It depends on the Proxy.



NOTE: All settings in this section are specific to your VoIP network. Please ask your VoIP service provider whether they require these settings.

<input type="checkbox"/> Enable Support of SIP Proxy Server / Soft Switch			
<input checked="" type="checkbox"/> Enable SIP Proxy 1			
Proxy Server IP / Domain	<input type="text" value="192.168.1.1"/>	Proxy Server Port [1 - 65535]	<input type="text" value="5060"/>
Proxy Server Realm	<input type="text"/>	TTL (Registration Interval) [10 - 7200 s]	<input type="text" value="600"/>
SIP Domain	<input type="text"/>	Use Domain to Register	<input type="checkbox"/>
<input type="checkbox"/> Enable SIP Proxy 2			
Proxy Server IP / Domain	<input type="text" value="192.168.1.1"/>	Proxy Server Port [1 - 65535]	<input type="text" value="5060"/>
Proxy Server Realm	<input type="text"/>	TTL (Registration Interval) [10 - 7200 s]	<input type="text" value="600"/>
SIP Domain	<input type="text"/>	Use Domain to Register	<input type="checkbox"/>
Bind Proxy Interval for NAT [0 - 180 s]	<input type="text" value="0"/>		
Initial Unregister	<input type="checkbox"/>		

- **Enable Support of SIP Proxy Server / Soft Switch:** Enable the functions to inter-work with Proxy Server / Soft Switch. When SIP Proxy 1 and 2 are enabled, the system will register to SIP Proxy 2 after all lines are failed to register to SIP Proxy 1. SIP Proxy 2 is a backup system.
- **Proxy Server IP/Domain:** Enter the Proxy Server IP address or URL (Uniform Resource Locator). You can set 3 redundant Proxy spread by semicolon.
EX: 61.123.231.1;12.34.56.78;proxy.sip.sip
- **Proxy Server Port:** Enter the Proxy Server **listen** port number. (The default value is 5060)

- **Proxy Server Realm:** This is used for gateway SIP account authentication in a SIP server. In most cases, the gateway can automatically detect your SIP server realm. So you can leave this option as blank. However, if your SIP server requires you to use a specific realm you can manually type it in here. **If you fail to make a call, please contact your VoIP service provider.**
- **TTL:** Enter the desired time interval at which the gateway will report to your Proxy Server.
- **SIP Domain/Use Domain to Register:** Enter the correct SIP domain to avoid registration failure (it is not necessary to set this with some Proxy Servers). If you enable “Uses Domain to Register” the VoIP Gateway will register to Proxy with the domain name you filed. Alternatively, the VoIP Gateway will register to a Proxy with the IP it resolves. **If you fail to make a call, please contact your VoIP service provider.**
- **Bind Proxy Interval for NAT:** This function is able to keep the binding is existed when VoIP Gateway is behind NAT and SIP Proxy is not able to keep the binding.
- **Initial Unregister:** After rebooting, it is unregistered first and then do the general register process.

FXS/ FXO Representative number registers to Proxy:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
FXS Representative Number		9966	<input checked="" type="checkbox"/>		9966	*****	*****
FXO Representative Number		9977	<input checked="" type="checkbox"/>		9977	*****	*****

Assuming that your registered ID and password are individual, the settings should be as above:

- **FXS Representative Number:** Register all FXS ports as a hunting group.
- **FXO Representative Number:** Register all FXO ports as a hunting group.
All the grouped FXO ports will be hunted automatically. It is available when FXO registers to Proxy.
- **Register:** Register to Proxy if ticked.

Hunting pattern of grouped FXS/FXO ports will be based on the hunting policy you specified on page 33.



NOTE: Please contact your VoIP provider about whether they can provide reprehensive numbers to allow you to use them for more than one FXS and FXO ports. FXS and FXO reprehensive numbers should be different.

Each line registers to Proxy independently:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
1	FXS	6701 <input type="button" value="Auto"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	6701	*****	*****
2	FXS	6702	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	6702	*****	*****
3	FXS	6703	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	6703	*****	*****
4	FXO	6704	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	6704	*****	*****

- **Number:** VoIP phone number
- **User ID/Account:** VoIP account Authentication ID or account name
- **Password:** password for VoIP account authentication
- **Invite with ID / Account:** VoIP gateway can be invited to a VoIP trunk gateway w/o registering with a Proxy. Please contact your VoIP provider.

As there are various Proxy Server providers, NetComm has designed the gateway to be compatible with as many SIP VoIP networks as possible, and according to RFC standards. If any registration problem occurs, please consult your VoIP provider.



NOTE: When you register with a Proxy Server, dialing principles may vary with different Proxy Servers, especially when dialing through a FXO port. Please consult your VoIP Service Provider for details.

Private Network

Users can establish a private network by using the Phone Book Manager Service. The Phone Book Manager Service is different from using the Proxy. The gateway is able to register with the Phone Book Manager Service and the SIP Proxy at the same time.

Private Network	
Phone Book Manager Service	
Server Settings	
Enable Phone Book Manager Server	<input type="checkbox"/> Clients List
Share Phone Book to Clients	<input type="checkbox"/> TTL (Expire time: mins) [1 - 60] <input type="text" value="1"/>
Client Settings	
Register to Phone Book Manager	<input type="checkbox"/>
Gateway Name for Phone Book Manager	<input type="text" value=""/>
Phone Book Manager IP/Domain	<input type="text" value="203.44.167.33"/> Phone Book Manager Server Listen Port [1 - 65535] <input type="text" value="1690"/>
Password Settings	
Phone Book Manager Login Password	<input type="password" value="*****"/> Confirm Password <input type="password" value="*****"/>

Server Settings

- **Enable Phone Book Manager Server:** It allows other Gateway users to register the IP address and Gateway Number in this Phone book manager server. It is recommended to use a static IP for the Phone Book Manager Server.
- **Share Local Phone Book to clients:** While this option is enabled and the gateway is performing as a Phone Book Manager Server, this gateway will append its local Phone Book entries to the Server for other clients to lookup.
- **TTL (Time to Live):** If a Gateway system that is controlled by the Phone Book Manager Server does not report back within the deadline set by TTL, the system will be excluded from the user's list. Each Gateway should report to the Phone Book Manager Server once every 30 seconds.

Client Settings

- **Register to Phone Book Manager:** Used to register to the Phone Book Manager. If the gateway is the Phone Manager Server, it has to enable this function to communicate with other clients.
- **Gateway Name for Phone Book Manager:** The alias registered with the Phone Book Manager Server.
- **Phone Book Manager IP/Domain:** Enter the IP address for the Phone Book Manager Server. It supports URL (Uniform Resource Locator).
- **Phone Book Manager Server Listen Port:** The protocol communication port for transmitting signals between the Phone Book Manager and other Gateway systems.
- **Phone Book Manager Login Password:** Enter the password for Phone Book Manager Service. If the gateway works as a Phone Book Manager **client**, the Login password is the one which is used to register in a Phone Book Manager server. If the gateway works as Phone Book Manager **server**, the Login password is the one which is used for checking against Phone Book Manager clients when they registering in the server.



NOTE: When the gateway is configured as Phone Book Manager server, it can also be configured as a client at the same time by enabling Register to Phone Book Manager option. This means the gateway also shares its Private Gateway Number to other clients which register on it.

Private Network Numbers		
Gateway Number		20347852
Line	Type	Number
1	FxS	901
2	FxS	902
3	FxS	903
4	FxO	904

Private Network Numbers

- **Gateway Number:** A self-defined phone number for the gateway that the gateway will use as its phone number to register on a Phone Book Manager server.
- **Line:** A self-defined extension number which allows remote gateways to reach a specific extension line.

Example:

Assume a gateway uses “5700” as its gateway number and registers on a Phone Book Manager server which your gateway registers on. To make a call to the gateway, simply dial 5700 which will cause one of the FXS ports of the gateway to ring (the ringing port depends on hunting policy of the gateway). If the gateway has configured its line extensions as “901”~“904”, then to reach the extension line “902” of the gateway you just simply dial 5700902.

Calling Features

Calling Features								
Line	Type	Do Not Disturb	Unconditional Forward	Busy Forward	No Answer Forward	Call Hold	Call Transfer	Call Waiting
FXS Representative Number			<input type="checkbox"/>	<input type="checkbox"/>	(NR)	(NR)	(NR)	(NR)
FXO Representative Number			<input type="checkbox"/>	<input type="checkbox"/>	(NR)	(NR)	(NR)	(NR)
1	FXS	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/> After(10 - 60)	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
2	FXS	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/> After(10 - 60)	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
3	FXS	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/> After(10 - 60)	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
4	FXO	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	(NR)	(NR)	(NR)	(NR)

- **Do Not Disturb:** It will only be able to call out when it is enabled. All incoming calls will be restricted.
- **Unconditional Forward:** All incoming calls will be forwarded to the “Forwarding Number” automatically. If it forwards to FXO, it only make FXO hook off, not make FXO dial out.
- **Busy Forward:** Forward the incoming call to “Forwarding Number” when the port is busy.
- **No Answer Forward:** Forward the incoming call to “Forwarding Number” after ring timeout expires without answer.
- **Call Hold:** Enable the call hold feature on the specific FXS port.



NOTE: Call Hold must be checked; Call Transfer or Call Waiting is active.

- **Call Transfer:** Enable the call transfer feature on the specific FXS port.
- **Call Waiting:** Enable the call waiting feature on the specific FXS port.

Calling Feature Instructions:

- **Call Hold:** Call will be put on hold after FLASH button pressed on the phone set. The gateway will play on-hold music, which is provided by your VoIP network, to the remote end.
- **Call Transfer:** Call will be put on hold after FLASH button pressed on local phone set (gateway plays on-hold music to the remote end). Meanwhile, the local user can dial out to another number after dial tone is observed. After the handset is replaced back on-hook, the call on hold will then be transferred to the new call regardless of the status of the new call. If the wrong number is dialed for the new call, just press the FLASH button to get back the call on hold. In another case, if the local user doesn't hang up the phone after the new call is set up, press the FLASH button to switch between the first call and the new call. Please be informed that the PBX between phone sets and the gateway must support FLASH features in order to use this function. If a phone set is connecting directly to the FXS port of the gateway and not functioning to FLASH, please adjust the settings in “Flash Detect Time” in “Advanced Options” section.



NOTE: The availability of the above features also depends on your VoIP network. Please also check with your service provider on these services.

Advanced Options

Administrator's Name	<input type="text"/>		
Administrator's Password	<input type="password"/>	Confirm Password	<input type="password"/>
Web UI Login ID	<input type="text"/>		
Web UI / IVR Password	<input type="password"/>	Confirm Password	<input type="password"/>
Web UI auto logout [30 - 300 s]	<input type="text" value="60"/>		



NOTE: There are two levels to enter Web. Administrator is able to change all settings. Web UI only changes some settings.

- **Administrator's name and Password:** Enter administrator name and password, which has the highest level of control of the gateway.
- **Web UI Login ID and Web UI/IVR Password:** Enter login ID and password when you log into the Web interface/IVR of the gateway as a normal user.
- **Web UI auto log out:** If a user does not act within the effective time range when logging into the web interface, the user will be disconnected from the web page to allow others to login.

Dial Wait Timeout [1 - 60 s]	<input type="text" value="10"/>	Inter Digits Timeout [1 - 60 s]	<input type="text" value="4"/>
Minimum DTMF ON Length [40 - 500 ms]	<input type="text" value="80"/>	Minimum DTMF OFF Length [40 - 500 ms]	<input type="text" value="80"/>
DTMF Detection Sensitivity	(less) <input type="radio"/> 1 <input type="radio"/> 2 <input checked="" type="radio"/> 3 <input type="radio"/> 4 <input type="radio"/> 5 (more)		
Enable Out-of-Band DTMF	<input checked="" type="checkbox"/> Enable Hook Flash Event <input checked="" type="checkbox"/> RFC 2833 Payload Type 101 <input type="checkbox"/> SIP Info		
Use Second CPT after SIP registered	<input checked="" type="checkbox"/>		
Line Settings (Gain, Flash Time, Enable Polarity Reversal)			

- **Dial Wait Timeout:** Use it to set the waiting time for the user's first key pressing when dialing a number. The user will hear a busy tone if he/she does not press the first key within the set time frame.
- **Inter Digits Timeout:** Set the waiting time between each key pressing. The entered numbers will be dialed after the timeout.
- **Minimum DTMF ON Length (Dial on)/ Minimum DTMF OFF Length (Dial off - between tones):** Used to set dial tone when a call is being diverted to another extension.
- **DTMF Detection Sensitivity:** Used to adjust the sensitivity of the telephone keys.
- **Enable Out-of-Band DTMF:** To send DTMF keys (0~9, *, #,) follow the RFC2833 rules or via SIP Info.



NOTE: Out-of-Band DTMF transport method may vary with different VoIP networks, please contact your VoIP provider for their preferred method.

NOTE: The 800 series gateway has three extra options compared to the 400 series gateway: FXO Dial Type, FXO Impedance and FX Impedance. Use the default settings for these options as they have been configured for the Australian telephone network.

- **Payload Type:** payload type of RFC2833.
- **Uses Second CPT for VoIP Call:** This function is usually applied when the user selects VoIP as the primary path for outgoing calls and PSTN as the backup. By enabling this function, the gateway will generate a different set of tones to inform the user that VoIP is in service. Should VoIP fail and fallback to PSTN, the user will hear PSTN tones instead of the second set CPT. (for CPT related settings, please refer to Trunk Management -> CPT/Cadence Settings).

Line Settings

Line	Extension Number	Type	Listening Volume	Speaking Volume	Tone Volume	Flash Time	Enable Polarity Reversal	PSTN Answer Detection	PSTN Ring Off Length [1000 - 20000 ms]
1	701	FXS	0 ▾ All	0 ▾ All	5 ▾ All	0.5 ▾ All	<input type="checkbox"/>		
2	702	FXS	0 ▾	0 ▾	5 ▾	0.5 ▾	<input type="checkbox"/>		
3	703	FXO	0 ▾	0 ▾	5 ▾	0.5 ▾	<input type="checkbox"/>	Disable ▾	2000
4	704	FXO	0 ▾	0 ▾	5 ▾	0.5 ▾	<input type="checkbox"/>	Disable ▾	2000

- **Listening Volume:** Adjusts the hearing volume.
- **Speaking Volume:** Adjusts the speaking volume.
- **Tone Volume:** This setting will be applied to all tones generated by the VoIP gateway including Dial Tone, Busy Tone, and so on.
- **Flash Time:**
 - FXS:* Used to adjust the detection period of flash signal from the phone set connected to the FXS port. For example, if pressing the FLASH key disconnects a call, increase the "Flash Detect Time" to fix this issue.
 - FXO:* Used to set the time frame that FXO generates a FLASH signal.
- **Enable Polarity Reversal:**
 - FXS:* As the remote site answers this call or goes on hook the FXS port will reverse the polarity.
 - FXO:* This option forces the gateway to detect the reversal of polarity on FXO port as the primary signal to drop a call. Some telephone switches or PBX systems reverse the line polarity to inform the remote end to drop an ongoing call. Please consult with the telephone service provider for availability of this feature.
- **PSTN Answer Detection:** This is only used with specific VoIP providers.
- **PSTN Ring Off Length:** This is for PSTN making out if the call rings off before picking up.

Codec Settings					
Preferred Codec Type	G.729A (8kps)				
Jitter Buffer (80 - 1200 ms)	120				
Silence Detection / Suppression	<input type="checkbox"/>		Echo Cancellation <input checked="" type="checkbox"/>		
Codec	<input checked="" type="checkbox"/> G.711 u-law	<input checked="" type="checkbox"/> G.723.1	<input checked="" type="checkbox"/> G.726	<input checked="" type="checkbox"/> G.729A	<input checked="" type="checkbox"/> G.711 a-law
Packet Interval (ms)	20	30	20	20	20
Approximate Bandwidth Required (kbps)	85.6	20.8	53.6	29.6	85.6

- **Preferred Codec Type:** Since different voice codecs have different compression ratios, so the sound quality and occupied bandwidths will also be different. It is recommended that you use the default provided (G.729) because it is popular and will provide better sound quality.
- **Jitter Buffer:** Adjusts the jitter to receive a packet. If the jitter range is too wide, it will delay voice transmission.
- **Silence Detection/Suppression:** If one side of a connection is not speaking, the system will stop sending voice data (packet) to decrease bandwidth usage.
- **Echo Cancellation:** Prevents voice quality caused by echo interference.
- **Packet Interval:** Defines how long the VoIP gateway sends a RTP packet to the other party of a VoIP conversation. The smaller the value, the more bandwidth usage. The larger the value, the more voice delay.
- **Approximate Bandwidth Require:** The bandwidth required varies with codec format and packet time.

Fax Settings

FAX Settings	
T.38 <input checked="" type="radio"/>	<input checked="" type="radio"/> UDP <input type="radio"/> TCP <input checked="" type="checkbox"/> Enable High Quality
T.30 <input type="radio"/>	FAX Codec: <input type="text" value="G.726 (32kbps 30)"/> Approximate Bandwidth Required (kbps): <input type="text" value="53.6"/> FAX Frame Count Per Packet (2 - 8): <input type="text" value="2"/> FAX Jitter Buffer (50 - 1200 ms): <input type="text" value="300"/>
FAX Tone Detection Sensitivity (secs) <input type="radio"/> 1 <input type="radio"/> 2 <input checked="" type="radio"/> 3 <input type="radio"/> 4 <input type="radio"/> 5 (more)	

- **T.38:** The T.38 protocol is used for better and faster facsimile transmission. So it is recommended to enable this function to gain better fax quality. When this function is enabled, please select UDP or TCP. If selecting TCP and some gateways cannot use the Fax function, please select UDP instead.



NOTE: When a FAX tone is detected in a call, the gateway will automatically switch from voice mode to fax mode. So fax settings will be temporarily applied to a specific port which detects fax tones, instead of its voice settings.

- **Enable High Quality:** The system sends the same FAX frame twice to get a high quality of the FAX. It requires more bandwidth.
- **T.30:** The system uses T.30 as the protocol for fax transmission. The parameter settings are the same as for voice transmission. However, enabling the T.30 will consume more network resources and will affect transmission quality.
- **FAX Detect sensitivity:** Used to adjust the sensitivity of detection as to whether a phone call is a FAX or not.



NOTE: When you send fax over an IP network it needs your network to support fax over IP function (either T.38 or T.30). Please consult your VoIP provider for this setting.

Drop Inactive Call	
Silence Detection Threshold [0 - 60 db]	<input type="text" value="0"/> (0 : Disable)
Drop Silent Call Timeout [30 - 3600 s]	<input type="text" value="120"/>

This is used as a standard to determine whether or not to hang up the phone. The system will hang up the phone automatically to avoid keeping the line engaged if the detected volume is below the Silence Detection Threshold and the time exceeds the Drop Silent Call Timeout.

Drop Inactive Call

- **Silence Detection Threshold:** The volume below the threshold is used as a standard to determine whether or not to hang up the phone.
- **Drop Silent Call Timeout:** If the detected volume is below the threshold and the time exceeds the silence detection interval, the system will hang up the phone automatically to avoid keeping the line engaged.



NOTE: Please be careful with these setting. Improper values might cause unexpected automatic disconnection of a call. Default values are recommended.

Digit Map

There are 50 sets of leading digit entries to choose voice routing interface – Auto select, PSTN or VoIP.

Default Call Route

- **Default Call Route:** The default call route can be Auto, VoIP, PSTN and Deny.

Auto (VoIP first): The call route is VoIP first, and the next is PSTN.

VoIP: The call route is VoIP only.

PSTN: The call route is PSTN only.

Deny: The call will be deny if the dial-out number is not in the table.

#	Enable	Leading Digits	Total Digit Count [0=disable, 1- 40]	Route
1	<input checked="" type="checkbox"/>	<input type="text" value="03"/>	<input type="text" value="10"/>	<input type="text" value="VoIP"/>
2	<input checked="" type="checkbox"/>	<input type="text" value="02"/>	<input type="text" value="10"/>	<input type="text" value="PSTN"/>
3	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="10"/>	<input type="text" value="Auto (VoIP first)"/>
4	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="10"/>	<input type="text" value="Auto (VoIP first)"/>

- **Enable:** Enable detection of this entry.
- **Leading Digits:** The leading digits for the VoIP gateway to scan while the user is dialing.
- **Total Digit Count:** Total number of digits that VoIP gateway should accept. 0 means that the VoIP gateway scans leading digits only and disregards total digit count.
- **Route:** The interface that calls should go through if above conditions are satisfied.

Example:

For the above setting, if a user dialed 0394978867, which matches Digit Map rule 1, then the gateway will route this call through VoIP. If 0294278766 is dialed, then the gateway will route this call through PSTN.

Phone Book

The system can set up and store 100 phone numbers in the phone book and provide an IP address query when calling other Gateway(s). If no Phone book manager is set within a Gateway group, then all Gateway systems have to set up phone data for each VoIP gateway to communicate with each other.

Local Phone Book				
1 - 5			6 - 10	
#	Gateway Name	Gateway Number	IP / Domain Name	Port
1	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text" value="5060"/>
2	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text" value="5060"/>

- **Gateway Name:** Enter the Gateway's code or an easy-to-remember name.
- **Gateway Number:** Enter the desired Gateway number.
- **IP/Domain Name:** Enter the IP address or URL (Uniform Resource Locator) of the Gateway.
- **Port:** Enter the Gateway's port.

Speed Dial

This system can set up 100 numbers for speed dialing. Setting methods are as follows:

	Speed Dial 1 - 10	11 - 20
#	Speed Dial Code("?" = single digit ; "%" = wildcard)	Number To Dial
Speed Dial 1	<input type="text" value="55"/>	<input type="text" value="32568791"/>
Speed Dial 2	<input type="text" value="3??"/>	<input type="text" value="5213??"/>
Speed Dial 3	<input type="text" value="00%"/>	<input type="text" value="856%"/>

Method 1- Single mapping: Fill a short code into the "Speed Dial Code" column, and enter the desired phone number into the "Number To Dial" column.

For example, pick up the handset and dial **55#** and the system will dial 32568791.

Method 2- Multi mapping: Fill the prefix code into the "Speed Dial Code" column and the format to transfer into the "Number To Dial" column.

For example, pick up the handset and dial **301#**, and the system will dial **521301**.

If the user dials **00 1657987456321#**, the system will DIAL **856 1657987456321**



NOTE: The gateway uses the "#" key as a delimiter to detect end of dialed number input.

Caller Filter

<input checked="" type="radio"/> Allow <input type="radio"/> Deny		
Enable	Filter IP address	Subnet mask
<input checked="" type="checkbox"/>	<input type="text" value="61.23.45.67"/>	<input type="text" value="255.255.255.0"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>

This function is used to allow or deny the SIP Invite message from the Proxy list ONLY.

- **Filter IP Address:** Enter the IP address that you would like to allow/deny.
- **Subnet mask:** Enter the subnet mask that is applied to the IP that entered in the **Filter IP address** field.

Transit Call Control

If you wish to restrict a general user (one who is not required to enter the PIN code) to local calls only and prohibit him/her from making long-distance calls started with a prefix "0", do the following steps:

1. Enable the Outbound Call Control function,
2. Set the PIN code for Outbound Level 5 to blank,
3. Set the Long-Distance Control Table to correspond with the Outbound Level 5 to prohibit making any call with the prefix "0".



NOTE: Transit Call Control is effective when it cooperate with Long-Distance Control Table.

Inbound Call Control	<input type="checkbox"/>	Outbound Call Control	<input type="checkbox"/>
----------------------	--------------------------	-----------------------	--------------------------

- **Inbound Call Control:** Determines when users make a phone call from a PSTN to Gateway FXO whether or not they check the inbound PIN code while placing the phone call to VoIP network only effective for incoming calls calling from a PSTN trunk.
- **Outbound Call Control:** Determines when users utilize the Gateway FXO interface to divert to PSTN network whether or not they check the outbound PIN code only effective for outgoing calls being diverted to a PSTN Trunk.

Transit Call Control			
Inbound Call Control	<input type="checkbox"/>	Outbound Call Control	<input type="checkbox"/>
#	PIN Code	Enable	Privileges
1	<input type="text"/>	<input type="checkbox"/>	0 ▾
2	<input type="text"/>	<input type="checkbox"/>	0 ▾
3	<input type="text"/>	<input type="checkbox"/>	0 ▾
4	<input type="text"/>	<input type="checkbox"/>	0 ▾
5	<input type="text"/>	<input type="checkbox"/>	0 ▾

- **PIN Code:** Enter the PIN code (4-6 digits or leave blank. A blank indicates no PIN code is required at this level. Generally, the PIN at level 5 can remain blank to simplify the phone number.)
- **Enable:** Enables the PIN code at each level.
- **Privileges:** The level is divided into 0~5 (The levels are in descending order; 0 stands for the highest authority and 5 stands for the lowest.)

Dialing principle to enable PIN code

***Inbound Call Control PIN Code * Outbound Call Control PIN Code * Phone number**

Using * to the separate PIN code and the phone number based on actual settings. The PIN can also be omitted.

The actual operation is shown below:

- If making a call from Gateway A to Gateway B, and diverting to PSTN (local call) when Gateway A and Gateway B both have Outbound Call Control disabled, dial as follows:

Gateway B Number + Local Phone Number

For example: **5168 35020311#**

- When Gateway A disables Inbound Call Control and Gateway B enables Outbound Call Control, then dial as follows:

*** Gateway B PIN Code * Gateway B Number Local Phone Number**

For example: ***123456* 5168 35020311#** (Gateway B PIN Code: 123456)

- When Gateway A enables Inbound Call Control and Gateway B disables Outbound Call Control, then: If using an extension line at Gateway A to make a local call to a remote end (at Gateway B site) through Gateway B, then dial

Gateway B Number + Local Phone Number

For example: **5168 35020311#**

- If using a trunk line to dial to Gateway A, and then making a local call to a remote end (at Gateway B site) through Gateway B, then dial

*** Gateway A PIN Code * Gateway B Number + Local Phone Number**

- When the instruction "Enter phone number" is given, then dial

For example: ***111111* 5168 35020311#** (Gateway A PIN Code: 111111)

- If using a trunk line to dial to Gateway A and then to the extension line of Gateway B, dial

*** Gateway A PIN Code * Gateway B Number**

When the instruction "Enter phone number" is given.

For example: ***111111* 5168#**

- When Gateway A has Inbound Call Control enabled and Gateway B has Outbound Call Control enabled, then if using an extension line at Gateway A to make a phone call to a remote end through Gateway B, dial:

*** Gateway B PIN Code * Gateway B Number Local Phone Number**

For example: ***123456* 5168 35020311#** (Gateway B PIN Code: 123456)

- If using a trunk line to dial to Gateway A, and then dialing a local call to a remote site through Gateway B, please dial

*** Gateway A PIN Code * Gateway B PIN Code * Gateway B Number Local Phone Number**

For example: ***111111* 123456* 5168 35020311#** (Gateway A PIN Code 111111 Gateway B PIN Code 123456)

Long-Distance Control Table

This table controls the level of authority of an outgoing (transit out) call that is dialed through FXO and diverted to PSTN, as below:

Long Distance Control Table						
#	0	1	2	3	4	5
1	0204					
2						
3						
4						
5						

Descriptions:

- Digit strings in this table are prefixes that the gateway will check on dialed numbers in transit out calls.
- This table is used to prohibit dialing any numbers started with specified prefixes.
- If Level 0 (the highest level) is set to prohibit dialing any number started with prefix 0204, then any level below 0 (including Levels 1 to 5) are also prohibited.
- If Level 1 is set to prohibit dialing any number with prefix 0, then any level below 1 (including Levels 2 to 5) are also prohibited. Since Level 0 is not restricted to any prefix, therefore at level 0 users can dial a number with the prefix 0.

Principle: Downward Restriction —If users at a higher level cannot dial a number with a certain prefix, then users at lower levels also cannot dial a number with the same prefix.

Long Distance Exception Table

This table handles any exceptions to the long-distance call table.

Long Distance Exception Table						
#	0	1	2	3	4	5
1	020488988					
2						
3						
4						
5						

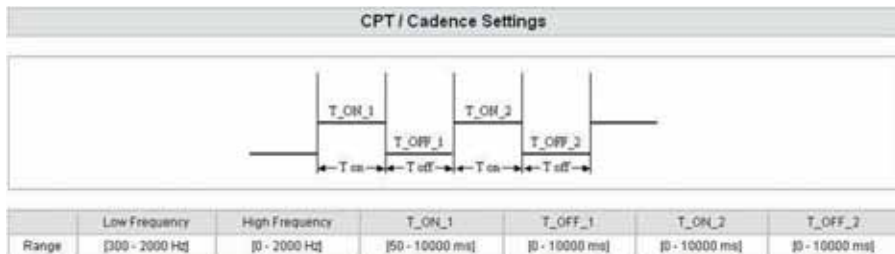
Instruction:

- According to the Long Distance Control Table, users at Level 0 are prohibited from dialing a number with the prefix 0204. But, if the number 020488988 is set in the Exception Table as above, then users can dial this number.

Principle: Upward Opening — If the users at a lower level can dial a number with a certain prefix, then the users at higher levels can also dial a number with the same prefix.

CPT/Cadence Settings

CPT/Cadence setting parameters serve as the basis of an FXO interface to determine whether or not a PSTN-call receiving party has hung up the phone. If the following parameters differ from the parameters of the actual assigned lines, it could cause the FXO to continue to engage a line.



Busy Tone Cadence Measurement

Busy Tone Cadence Measurement

	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	Auto Learning	
BTC # 1	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="200 / 200"/> <input type="button" value="Setting"/>
BTC # 2	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="200 / 200"/> <input type="button" value="Setting"/>
BTC # 3	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="200 / 200"/> <input type="button" value="Setting"/>
BTC # 4	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="200 / 200"/> <input type="button" value="Setting"/>
BTC # 5	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="200 / 200"/> <input type="button" value="Setting"/>

BTC Detection Sensitivity (less) 1 2 3 4 5 (more)

- **Busy Tone Cadence Measurement and auto learning:** Provides a solution of FXO integrated with PSTN or PBX. FXO will learn the busy tone automatically.
- **BTC Detection Sensitivity:** The more sensitivity, the more quickly the system will cut off the call. If the system often cuts off an un-finished call, select less sensitivity.

CPT parameters Table

The CPT has 3 sets of parameter tables. You can adjust it based on local PSTN or PBX.

#1 Enable Detect	Setting 1					
Tone Type	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	<input type="text" value="350"/>	<input type="text" value="440"/>	<input type="text" value="3000"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>
Congestion Tone	<input type="text" value="480"/>	<input type="text" value="820"/>	<input type="text" value="250"/>	<input type="text" value="250"/>	<input type="text" value="0"/>	<input type="text" value="0"/>
Busy Tone	<input type="text" value="480"/>	<input type="text" value="820"/>	<input type="text" value="500"/>	<input type="text" value="500"/>	<input type="text" value="0"/>	<input type="text" value="0"/>
Ring Back Tone	<input type="text" value="440"/>	<input type="text" value="480"/>	<input type="text" value="1000"/>	<input type="text" value="2000"/>	<input type="text" value="0"/>	<input type="text" value="0"/>

Moreover, users can use CPT Auto Detect to detect CPT parameters. Instructions are shown in the following section. The method to detect CPT is as follows:



NOTE: To cope with different local PSTN and different PBX models, the system provides auto busy tone cadence learning to prevent the FXO from engaging a line. However, if the line of the receiving party is engaged and his/her PSTN uses a voice prompt to replace the traditional beep sound, then the system would not be able to detect a busy tone. Silence detection should then be used to determine whether or not to end the call.

CPT Auto Detect

- 2 PSTN phone numbers or 2 PBX extension lines are needed.
- Connect one of the phone sets to a FXO port.
- The line of “Dial Number” must be on-hook. Set the outgoing phone number the same as the phone line that is in use as above, and click the “Accept” button to start detection.
- If detection is successful, the parameters obtained will automatically be inserted into the CPT parameter.
- Save settings and restart the system.

Detailed description is given below:

Click “CPT auto detect” at the bottom of Trunk CPT Settings.

Disable UDT	UDT Setting	# 1 Disable	# 2 Disable	# 3 Disable	# 4 Disable
Congestion Tone / Busy Tone	Low Frequency	0	0	0	0
	High Frequency	0	0	0	0

[Use CPT auto detect](#)

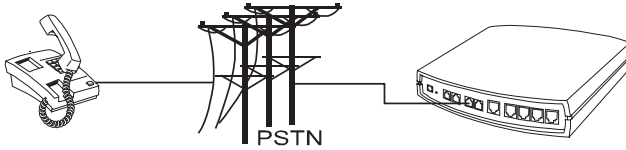
If CPT auto detect is used, the function would halt every operation of the Gateway. Select “**I am sure of it**”, and then click the “Accept” button. Wait for 15 seconds, and then you will enter CPT auto detect window.

Uses CPT auto detect

The function would halt every operation of the Gateway

I am sure of it

Direct Connection to PSTN



Note: The above diagrams show a 4-port SmartVoice gateway for illustration purposes. An 8, 16 or 32 port SmartVoice gateway could also be used depending upon your network requirements.

CPT Auto Detect

System Operation

Tone	Detect Time
<input checked="" type="checkbox"/> Dial Tone	9 (5 - 15secs)
<input checked="" type="checkbox"/> Reorder Tone	9 (5 - 15secs)
<input checked="" type="checkbox"/> Busy Tone	9 (5 - 15secs)
Detect Channel(3 - 4)	3
Phone Number	36008914
Dial Number	36008913

Save detect value to CPT group 1

- Connect one of the trunk lines to the Gateway FXO Port.
- The line of "Dial Number" (36008913) must be on-hook.
- **Detect Channel:** Enter 3 (if the trunk line is connected to P3, and uses P3 for outgoing detection).
- **Phone Number:** Enter the number of the FXO line.
- **Dial Number:** Enter the number of the end to be tested—36008913.
- **CPT group:** Enter the group to save your settings to once the test has been completed.
- Finally, click the "Accept" button.

Detection in Progress: during detection, the following windows will appear.

Current operation

Detecting Dial Tone...

Current operation

Detecting Reorder Tone...

Once detection of a busy tone is in progress, the system will dial the number to be tested (in this case 36008913). After it rings pick up the phone and enter"#", then hang up. The system will then detect a busy tone automatically.

Current operation

Detecting Busy Tone...
Please hook-off after the extension ring up, then press [#] before hook-off.

After detecting it will be as below:

Detect again	Detection Value					
Tone	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	400	0	3000	0	0	0
Congestion Tone	400	0	250	250	0	0
Busy Tone	400	0	250	250	0	0

#2	Current Value					
Tone	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	400	0	3000	0	0	0
Congestion Tone	400	0	250	250	0	0
Busy Tone	400	0	250	250	0	0

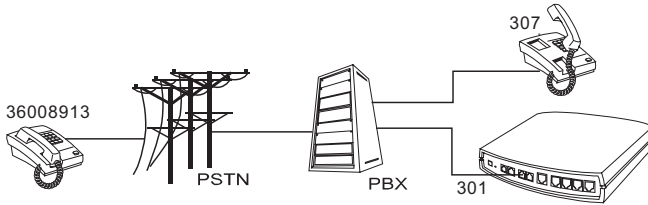
If you sure overwrite Current Value, please click Accept

Accept

Connected to a PBX Extension Line

If the gateway is connected to a PBX extension line, then the busy tone of both the PBX and the PSTN must be detected.

Connecting to a PBX extension line and detecting the busy tone of the PBX

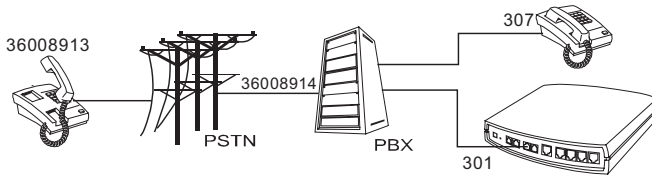


Note: The above diagrams show a 4-port SmartVoice gateway for illustration purposes. An 8, 16 or 32 port SmartVoice gateway could also be used depending upon your network requirements.

CPT Auto Detect	
System Operations	
Tone	Detect Time
<input checked="" type="checkbox"/> Dial Tone	9 (5 - 15secs)
<input checked="" type="checkbox"/> Reorder Tone	9 (5 - 15secs)
<input checked="" type="checkbox"/> Busy Tone	9 (5 - 15secs)
Detect Channel(3 - 4)	3
Phone Number	301
Dial Number	307
Save detect value to CPT group	1

- Connect one of the PBX extension lines to the Gateway FXO Port.
- **Detect Channel:** Enter 4 (if the trunk line is connected to P4, and uses P4 for outgoing detection).
- **Phone Number:** Set the number of the FXO line –to detect Reorder Tone.
- **Dial Number:** Enter the number of the end to be tested—307.
- **CPT group:** Enter the group to save your settings to once the test has been completed.
- Finally, click the “Accept” button.

Connected to a PBX extension line to detect the busy tone of PSTN



Note: The above diagrams show a 4-port SmartVoice gateway for illustration purposes. An 8, 16 or 32 port SmartVoice gateway could also be used depending upon your network requirements.

CPT Auto Detect	
System Operation	
Tone	Detect Time
<input checked="" type="checkbox"/> Dial Tone	9 (5 - 15secs)
<input checked="" type="checkbox"/> Reorder Tone	9 (5 - 15secs)
<input checked="" type="checkbox"/> Busy Tone	9 (5 - 15secs)
Detect Channel(3 - 4)	3
Phone Number	0,36008914
Dial Number	0,36008913
Save detect value to CPT group: 1	

- Connect one of the PBX extension lines to the gateway's FXO Port.
- **Detect Channel:** Enter 3 (if the trunk line is connected to P3, and uses P3 for outgoing detection).
- **Phone Number:** Set the number of FXO line –to detect Reorder Tone.
- **Dial Number:** Enter the trunk prefix of a PBX, comma, and the number of the end to be tested -- 0,36008913. (Assuming that an extension line is set to dial 0 to make outgoing calls).
- **CPT group:** Enter the group to save your settings to once the test has been completed. If connecting the gateway to a PBX extension line, please do not set the detected busy tone of the PBX and the PSTN in the same set, otherwise the value detected the first time will be overwritten.
- Finally, click the "Accept" button.

Filling in the CPT Table

Fill in the table after the detection is completed as below, where the values are the frequency and On-and-Off ratio detected. Please click the “Accept” button. If connecting the Gateway to a PBX extension line, please do not set the detected busy tone of the PBX and the PSTN in the same set, otherwise the value detected the first time will be overwritten.

Detect again	Detection Value					
Tone	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	400	0	3000	0	0	0
Congestion Tone	400	0	250	250	0	0
Busy Tone	400	0	250	250	0	0

#2	Current Value					
Tone	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	400	0	3000	0	0	0
Congestion Tone	400	0	250	250	0	0
Busy Tone	400	0	250	250	0	0

If you sure overwrite Current Value, please click Accept

Save Settings

Save the settings and restart the system after the test is completed. Then click the “Accept” button. The Gateway will use the new parameter to detect whether a call has ended.

<input type="checkbox"/> Save Settings	Save all configurations.
Be sure to save all settings before restart.	
<input type="checkbox"/> Restart	Restart the Gateway right away. All calls will be DROPPED when Restart.

Current Status and System Information

These two pages show that the status of VoIP Gateway. There are Port Status, Server Registration Status, WAN Port Information, LAN Port Information and Hardware.

Current Status							
Port Status							
No	Type	Extension Number	Line Status	Calls	Dialed Number	Proxy Register	UPnP on RTP
1	FXS	5004	Idle	0		Successful	
2	FXS	702	Idle	0		Disabled	
3	FXS	703	Idle	0		Disabled	
4	FXO	704	Idle	0		Disabled	
SIP Proxy Hunting Number Registration				FXS Disabled (3 days 19:44:34), FXO Disabled (3 days 19:44:34)			
Server Registration Status							
DDNS Registration				Disabled (3 days 19:44:34)			
Phone Book Manager Registration				Disabled (3 days 19:44:34)			
STUN Registration				Disabled (3 days 19:44:34)			
UPnP Negotiation				Disabled (3 days 19:44:34)			
System Information							
WAN Port Information							
Factory Default MAC Address				00 0C 2A 05 4E CC			
IP Address				192.168.1.2			
Subnet Mask				255.255.255.0			
Default Gateway				192.168.1.254			
DNS				168.95.1.1			
LAN Port Information							
MAC Address				00 0C 2A 05 4E CD			
IP Address				192.168.8.254			
Subnet Mask				255.255.255.0			
DHCP Server							
DHCP Server				Enabled			
IP Pool Range				192.168.8.1 - 192.168.8.250			
Lease Time				1 hour(s)			
DNS				168.95.1.1			
Hardware							
Hardware Platform				PBX3S10			
Driver				0.9.4 Build(14) 28/Dec/2005 13:51:39			
DSP				482			

- **Port Status:** It includes if each port registers to Proxy successfully, the lasted dialed number, how many calls each port had since the system is start, etc.
- **Server Registration Status:** It shows the registration status of DDNS, Phone Book Manager, STUN and UPnP.
- **WAN Port Information:** It shows IP address, subnet mask, default gateway and DNS server. If you use PPPoE to obtain IP, you can know if the IP is obtained through this.
- **LAN Port Information:** It shows LAN port IP, subnet mask, and the status of DHCP server.
- **Hardware:** It shows the hardware platform.

RTP Packet Summary

Displays the information of the last finished call. It contains peer IP, peer port, packets sent, packet received and packet lost. Press the **Refresh** button to get the latest RTP Packet Summary. Voice quality can be degraded significantly if packet lost exceeds 5%. To avoid high packet loss, you need either to increase your broadband bandwidth or configure your QoS in your broadband router.

RTP Packet Monitor

Line 1	G.711 64kbps	Send Packet Count	0	Receive Packet Count	0	Packet Lost Count	N/A
The last packet's source IP				The last packet's source Port		0	

Line 2	G.711 64kbps	Send Packet Count	0	Receive Packet Count	0	Packet Lost Count	N/A
The last packet's source IP				The last packet's source Port		0	

Line 3	G.711 64kbps	Send Packet Count	0	Receive Packet Count	0	Packet Lost Count	0
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STUN Inquiry

Using “STUN Inquiry” to detect your IP sharing device’s NAT type and communication between STUN server and client (built-in SmartVoice gateway).

STUN Inquiry	
NAT Type	Open
STUN Server	1.2.3.4
STUN Server Port(1024 - 65535)	3478



NOTE: If detected NAT type is “Symmetric NAT”, then the gateway is not able to traverse the NAT of your Internet sharing devices. This is not a flaw of the gateway’s design, but nature of STUN protocol design.

Ping Test

Use "ping" to identify if the remote peer is reachable. Fill in remote IP address and click "Test" will start the test.

Ping Test	
IP address	<input type="text" value="192.168.8.59"/>
Request Count[1 - 100]	<input type="text" value="4"/>
PingTestPacketSize[56 - 5600bytes]	<input type="text" value="56"/>

Port Filtering

Port filtering enables you to control all data that can be transmitted in routers; **principles of filtering--When the port used at the source end is within the limited scope, it will be filtered without transmission.**

Enable Port Filtering

- **Enable port filtering:** whether to enable this function or not.

Port Range	TCP / UDP	Remark
0 - 0	Both ▾	
0 - 0	Both ▾	
0 - 0	Both ▾	

- **Port Range:** Set the range of port to be filtered, suppose it is 80 and when use protocol is Both or TCP, all computers will be unable to use the services of http (port 80)— will be unable to browse normal Web pages.



NOTE: If you filter port 80, it will also block you from web access to the gateway. So be careful with these settings.

- **TCP/UDP:** Choose to either filter TCP or UDP, or choose to filter both.
- **Remark:** Remark field, you can write comments by yourself.

IP Filtering

IP Filtering is to limit internal users from accessing the Internet.

Enable IP Filtering <input type="checkbox"/>		
IP	TCP / UDP	Remark
<input type="text"/>	Both ▼	<input type="text"/>
<input type="text"/>	Both ▼	<input type="text"/>
<input type="text"/>	Both ▼	<input type="text"/>

- **IP:** Input the IP address that you want to filter; the limited IP address will be unable to transmit the data to the Internet.
- **TCP/UDP:** Choose to either filter TCP or UDP, or choose to filter both.
- **Remark:** Remark filed, you can write comments by yourself.

MAC Filtering

MAC (Media Access Control) address filtering is to filter the transmission of data by network card physical address.

Enable MAC Filtering <input type="checkbox"/>	
MAC	Remark
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>

MAC: input MAC that will be limited to accessing Internet.

Virtual Server

Enabling the users on Internet to access the WWW, FTP and other services under your NAT. When remote user are accessing Web or FTP servers through WAN end IP address, it will be routed to the server at the internal LAN end and be routed to the server at the internal LAN end as appropriate in accordance with the externally required services.

Enable Virtual Server <input type="checkbox"/>				
WAN Port Range	TCP / UDP	LAN Host IP Address	Server Port Range	Remark
0 - 0	Both ▾		0 - 0	
0 - 0	Both ▾		0 - 0	

- **WAN Port Range:** Input the port on the WAN side.
- **TCP/UDP:** Select the communication protocols used by the server—TCP or UDP.
- **LAN Host IP Address:** Input IP address that provides various services servers.
- **Server Port Range:** Input the port used by the LAN host.

DMZ

Lets the server on the LAN to be directly exposed to the Internet for accessing data. Either this function or the virtual server can be selected for use.

URL Filter

URL filter is used to deny device from LAN accessing specific web sites. The system will block the URL that contains the string.

Enable	URL string pattern to be blocked	IP Range
<input type="checkbox"/>	<input type="text"/>	0.0.0.0 - 0.0.0.0
<input type="checkbox"/>	<input type="text"/>	0.0.0.0 - 0.0.0.0
<input type="checkbox"/>	<input type="text"/>	0.0.0.0 - 0.0.0.0

Special Applications

Provide multiple connections for special applications.

Enable	Name	Incoming Type	Incoming Port Range	Trigger Type	Trigger Port Range
<input type="checkbox"/>	MSN Gaming Zone	BOTH	2300-2400,28800-29000	TCP	47624 - 47624
<input type="checkbox"/>	Quick Time	TCP	6970-6999	TCP	554 - 554
<input type="checkbox"/>	ICU II	BOTH	2000-2038,2050-2051,2069,2085,3010	TCP	2019 - 2019
<input type="checkbox"/>	PC-to-Phone	TCP	12120,12122,24150-24220	TCP	12053 - 12053
<input type="checkbox"/>		TCP		TCP	0 - 0
<input type="checkbox"/>		TCP		TCP	0 - 0
<input type="checkbox"/>		TCP		TCP	0 - 0
<input type="checkbox"/>		TCP		TCP	0 - 0

- **Name:** The name of the special application.
- **Incoming Type:** The protocol used by the special application.
- **Incoming Port range:** Port range on the WAN side that will be used to access the application.
- **Trigger Type:** The protocol used to trigger the application.
- **Trigger Port Range:** Port range used to trigger the application.

DoS Prevention Settings

Enable DoS Prevention	<input checked="" type="checkbox"/>
Enable DoS Prevention on LAN	<input checked="" type="checkbox"/>
Whole System Flood	<input checked="" type="checkbox"/> SYN <input type="text" value="50"/> (Packets/Second) [50 - 500]
	<input checked="" type="checkbox"/> FIN <input type="text" value="50"/> (Packets/Second) [50 - 500]
	<input type="checkbox"/> UDP <input type="text" value="68"/> (Packets/Second)
	<input checked="" type="checkbox"/> ICMP <input type="text" value="50"/> (Packets/Second) [50 - 500]
Per-Source IP Flood	<input checked="" type="checkbox"/> SYN <input type="text" value="30"/> (Packets/Second) [30 - 300]
	<input checked="" type="checkbox"/> FIN <input type="text" value="30"/> (Packets/Second) [30 - 300]
	<input type="checkbox"/> UDP <input type="text" value="68"/> (Packets/Second)
	<input checked="" type="checkbox"/> ICMP <input type="text" value="30"/> (Packets/Second) [30 - 300]
TCP / UDP Port Scan	<input type="checkbox"/> TCP / UDP Port Scan Level <input type="text" value="Low"/>
TCP Scan	<input type="checkbox"/>
TCP SYN with Data	<input type="checkbox"/>
TCP Land	<input type="checkbox"/>
UDP Echo Chargen	<input type="checkbox"/>
UDP Bomb	<input type="checkbox"/>
UDP Land	<input type="checkbox"/>
Ping of Death	<input checked="" type="checkbox"/>
ICMP Smurf	<input checked="" type="checkbox"/>
ICMP Land	<input checked="" type="checkbox"/>
IP Spoof	<input type="checkbox"/>
Tear Drop	<input type="checkbox"/>

Source Blocking	
Enable Source IP Blocking	<input type="checkbox"/>
Blocking Time [2 - 600 s]	<input type="text" value="120"/>

- Enable DoS Prevention: To prevent DoS from WAN.
- Enable DoS Prevention on LAN: To prevent DoS from LAN.
- Packet/Second: If the same type of packets received in one second is more than the specified value, then it will be treated as attacks. And VoIP gateway will block the IP if you checked "Enable Source IP Blocking".

Source Blocking	
Enable Source IP Blocking	<input type="checkbox"/>
Blocking Time [2 - 600 s]	<input type="text" value="120"/>

- **Enable Source IP Blocking:** Block the IP.
- **Blocking Time:** The time to block the IP.

NTP (Network Time Protocol)

NTP						
	Year	Month	Day	Hour	Minute	Second
Gateway Time	<input type="text" value="2006"/>	<input type="text" value="2"/>	<input type="text" value="23"/>	<input type="text" value="13"/>	<input type="text" value="27"/>	<input type="text" value="9"/>
Time Zone	+ <input type="text" value="11"/> : <input type="text" value="00"/>					
#	Time Server					
1	<input type="text" value="ntp.ucsd.edu"/>					
2	<input type="text" value="ntp.univ-lyon1.fr"/>					
3	<input type="text" value="time.nuri.net"/>					

Time Zone: Set the Time Zone where VoIP gateway resides.

Time Server #1~#3: Set the Time Server where VoIP gateway should sync up during start up. (NTP protocol)

Backup and Restore (Configuration)

You can backup settings to a file and restore settings from that file. You also can restore all settings back to default by selecting **Restore Default Configurations** and click **Restore**.



NOTE: It needs to Save Settings and Restart, and all settings will back up default settings or have new setting that you upload.

Backup Configurations	
Configuration File	<input type="button" value="Backup"/>
Configuration Template File	<input type="button" value="Backup"/>

- **Configuration File:** Backup the all settings.
- **Configuration Template File:** Backup the settings as template file for editing.

Restore Configurations	
<input checked="" type="radio"/> Upload Configuration File	<input type="text"/> <input type="button" value="....."/>
<input type="radio"/> Restore Default Configurations	
<input type="button" value="Restore"/>	

Provision Settings

Options in this section are only required for VoIP networks in which provisioning system has implemented.

Fill in the parameters needed of Provision Server from your service provider.

Enable Auto Provisioning	<input type="checkbox"/>		
Provision Server Address	<input type="text"/>		
Port [1 - 65535]	<input type="text" value="10101"/>		
Packet Format	Proprietary ▾		
Connect Provision Server During Start Up	<input checked="" type="checkbox"/>		
Connect Provision Server Periodically	<input checked="" type="checkbox"/>	Auto Provision Interval [60 - 604800 s]	<input type="text" value="10800"/> Random Offset
			[1 - 1800s] <input type="text" value="600"/>
Provision Retry Times [0=always, 1 - 99] [0 - 99]	<input type="text" value="10"/>	Retry Interval [30 - 120 s]	<input type="text" value="30"/>
Suspend Service	<input type="checkbox"/>		

System Operations (Save Settings)

Some settings are effective by **Restart**. Remember to save all settings by **Save Settings** before to restart.

<input type="checkbox"/> Save Settings	Save all configurations.
Be sure to save all settings before restart.	
<input type="checkbox"/> Restart	Restart the Gateway right away. All calls will be DROPPED when Restart.

- **Save Settings:** Save settings after completing. The new settings will take effect after the system is restarted. Please select “Save Settings”.
- **Restart:** If it is necessary to restart the system, please select “Restart” and click the “Accept” button.

Software Upgrade

Gateway provides software upgrade function from a remote end. Please consult your service provider for all following details.

Software Upgrade	
To Save Current Settings, Save Settings	
Current Software Version No. [1.2.33.6-92-79]	
Upgrade Server	<input type="radio"/> TFTP <input type="radio"/> FTP <input checked="" type="radio"/> Image Server
Software Upgrade Server IP	<input type="text" value="192.168.1.161"/>
Software Upgrade Server Port [1 - 65535]	<input type="text" value="6001"/>
User Name	<input type="text"/>
Password	<input type="text"/>
Directory	<input type="text"/>

- **Upgrade Server:** Choose the server type.
- **Software Upgrade Server IP:** Enter the software server IP address.
- **Software Upgrade Server Port:** Enter the port that server uses. TFTP is 69, FTP is 21 and SmartVoice gateway upgrade server is 6001.
- **User Name/ Password:** The account to access FTP server.
- **Directory:** The path of TFTP or FTP.

Logout

Gateway only allows one user to login at a time, so whenever a change is made, please save the settings, restart the system, or logout to avoid the situation where other users cannot login to change settings.

Logout

To save settings, click [Here](#)

4. Configuring the Gateway via IVR

Preparation

- Install the Gateway according to the instructions as described in Chapter 2.
- If a static IP is used, confirm the desired IP settings of the WAN Port (IP address, Subnet Mask, and Default gateway). Please contact your local Internet Service Provider (ISP) if you have any questions.
- If using dialup ADSL (PPPoE) for network connection, confirm your broadband account name and password.
- If users wish to build a Gateway under the NAT, Gateway WAN Port IP address and LAN Port should not use the same range. This is to avoid phone failures.

Configuration Methods

The gateway provides two configuration options:

- 1) Web Interface
- 2) Telephone IVR (Interactive Voice Response)

The IVR provides basic query and configuration, while the Web interface provides a full configuration.

IVR (Interactive Voice Response)

The gateway provides a convenient IVR function. Users only need to pick up a handset and enter the function code for the query and setting. A PC is not required.



NOTE: After finishing the settings, make sure the new settings are saved. This is so that the new settings will take effect after the system is restarted.

Instructions

FXS Port:

1. Connected the Gateway to the telephones as required.
2. To access the gateway via IVR, pick up the handset and press *****[password]#**. Convert your password to a numerical sequence. The character to number conversion can be acquired from PPPoE Character Conversion Table on page 86.
3. After entering a correct IVR password, you will hear a indication tone which is beep tones about approximate 1 second apart, now the system is in IVR mode. Enter function codes to check or set the gateway configurations.

For example: The factory default IVR access password is "admin". Enter *****4144534954#**, you are now in IVR setting mode, enter a function code to check or configure the gateway. If your password is "1234", then enter ****1234#** to enter IVR setting mode.

FXO Port:

1. Connect the Gateway to the PSTN line as required.
2. To access the gateway via IVR, dial the phone number of the FXO Port using an external line. You will hear the instruction "enter value".
3. Enter the IVR password. The factory default code is "admin". Enter *****4144534954#** as above. You are now in the IVR setting mode. Refer to the IVR Function Table over the page for the query codes.
4. Once the first setting or query has been completed, you will hear the tone indication again. Then use the same procedure to make a second query or setting.
5. Dial 509 to save the settings and wait for about 3 seconds for the confirmation tone. To exit IVR mode, simply hang up the phone.

Example:

- enter the password *****4144534954#** (You are now in IVR mode)
- enter **101** (to query IP address of the gateway's WAN port)
- the system responds with an IP address
- you can continue with more settings or queries: enter **111** (to set IP address)
- enter **192*168*1*2** (IP number).

Inquire Current Gateway's WAN Port IP address

After entering IVR mode, dial **101**. The system will repeat the current WAN Port IP address. If the system does not repeat the IP address, it indicates that the Gateway fails to get an IP from DHCP or PPPoE. Please check that the cable connection, account number, and password are correct.

Save Settings

After entering IVR mode, dial **509** (Save Settings). Wait for about 3 seconds and after hearing a confirmation tone "1", hang up the phone. Please reboot the Gateway to enable the new settings.

IVR Functions Table

Function Code	Description	Example
111/101	WAN Port IP address Set/Query	Use in conjunction with function code 114 , select 1 for a Static IP function.
112/102	WAN Port Subnet Mask Set/Query	
113/103	WAN Port Default Gateway Set/Query	
114/104	Current Network IP Access Set/Query (1: Static IP, 2.DHCP, 3.PPPoE)	
115/105	DNS IP address Set/Query	
116/106	Phone book manager IP address Set/Query	Must use 116/106, 117/107 in conjunction with each other.
117/107	Set/Query whether or not to use Public Telephone Book (0: Disable 1:Enable)	
199/099	Set/Query whether or not this Gateway acts as the phone book manager (0: Disable 1: Enable)	
066	Query the connection to Phone book manager	
118	Restart	
121	Setting PPPoE Account	Use in conjunction with function code 114 , select 3 for a PPPoE function
122	Setting PPPoE Password	
123	Setting NAT router public IP address	Must use 123 and 124 in conjunction with each other.
124	Uses NAT router port forwarding (0: Disable 1: Enable)	
151/141	Register to Proxy Server Set/Query (0: Disable 1: Enable)	
152/142	Proxy Server IP address Set/Query	
153/143	Proxy Server Port Set/Query	
125	Set Proxy Server account	
126	Set Proxy Server password	
154/144	Uses STUN Set/Query (0: Disable 1: Enable)	
155/145	STUN IP address Set/Query	
156/146	STUN Port Set/Query	
311/301	LAN Port IP address Set/Query	
312/302	LAN Port Subnet Mask Set/Query	
131/132	Play/Record greeting message	
133	Saving greeting message	
211/201	Set/Query International Prefix code	Prefix dialed before making an international call e.g. 0011.
212/202	Set/Query Country Code	Setting country code, e.g. 61
213/203	Set/Query Area Prefix Code (Long-Distance Prefix Code)	Prefix dialed before making a long-distance call e.g. 0.
214/204	Set/Query Area Code	eg. "02" for NSW.

Function Code	Description	Example
215/205	Set/Query Gateway Telephone Number (Representative Number)	
216/206	Set/Query the extension number of Line 1.	
217/207	Set/Query the gateway web configuration interface port number	
109	Restoring factory default IP address configuration	A static IP address for WAN Port IP: 192.168.1.2 Mask: 255.255.255.0 Gateway: 192.168.1.254
409	Restoring factory default settings	
509	Save settings	
209	Soft Upgrade	

IP Configuration Settings (WAN port)

Static IP Settings



NOTE: Complete static public/private IP settings should include a static IP (Option 1 under 114), IP address (111), Subnet Mask (112), and Default Gateway (113). Please contact your local Internet Service Provider (ISP) if you have any questions.

Function	Command
Select a Static IP	<ul style="list-style-type: none"> After entering IVR mode, dial 114. After hearing "Enter value", dial 1 (to select static IP)
IP address Settings	<ul style="list-style-type: none"> After entering IVR mode, dial 111. After hearing "Enter value", enter your IP address, followed by "#". <p>Example: If the IP address is 192.168.1.200, dial 192*168*1*200#.</p>
Subnet Mask Settings	<ul style="list-style-type: none"> After entering IVR mode, dial 112. After hearing "Enter value", enter your subnet mask, followed by "#". <p>Example: If the mask value is 255.255.255.0, dial 255*255*255*0#.</p>
Default Gateway Settings	<ul style="list-style-type: none"> After entering IVR mode, dial 113. After hearing "Enter value", enter your default gateway's IP address, followed by "#". <p>Example: If the Default Gateway is 192.168.1.254, dial 192*168*1*254#.</p>
Save Settings and Restart	<ul style="list-style-type: none"> To save settings, dial 509 (Save Settings). The system will save the settings. Please restart the system. Wait for about 40 seconds for the system to restart, and then enter 101 to check if the IP address is retained. If the IP address is not repeated, it indicates that Gateway has not been properly connected; please check if the cable connection is correct.

Dynamic IP (DHCP) Settings

After entering IVR mode, dial **114**.

After hearing "Enter value", dial 2 (to select DHCP).

Saving settings –press **509** (Save Settings). Please restart the system. After the system is restarted, press **101** to check if the IP address is retained.



NOTE: If the system does not repeat the IP address, it indicates that the Gateway failed to communicate with a DHCP server. Please check your DHCP server or consult your ISP if they issue an IP address to you via DHCP protocol.

ADSL PPPoE Settings



NOTE: Complete PPPoE settings should include: Select PPPoE (Option 3 of 114), PPPoE account (121) and PPPoE password (122).

Please contact your local Internet Service Provider (ISP) if you have any questions.

Select PPPoE

- After entering IVR mode, dial **114**.
- After hearing "**Enter value**", dial **3** (to select PPPoE).

PPPoE Account Settings

- After entering IVR mode, dial **121**.
- After hearing "**Enter value**", enter the account number, followed by "#".

Example: If the account is "**0284943122@isp.com.au**", please enter **02080409040301 020271 495956 72435553#**.

Please note that it is necessary to enter two digits for each character/number; for example, enter "**01**" for "**1**" and "**11**" for "**A**".

It is recommended that you use the Web Interface to configure your PPPoE account details to make your life easier. Refer to the PPPoE Character Conversion table on the following page for key mappings.

PPPoE Password Setting

- After entering IVR mode, dial **122** after hearing "**enter value**" followed by "#".

Example: If the password is "**3ttixike**", please enter "**03 60 60 49 64 49 51 45#**".

Save Settings and Restart

To save settings, dial **509** (Save Settings). The system will save the settings. Please restart the system. Wait for about 40 seconds for the system to restart, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, it indicates that the Gateway has not been properly connected. Please check that the cable connection, account, or password are correct.

Recorded Voice File

- The gateway allows users to record their incoming call greeting message, when calling via FXO (transit-in calls).
- After entering IVR mode, dial **132**. After hearing "**Enter value**", record the incoming call greeting message. To end recording, simply hang up.
- To listen to the recorded message, press **131**. Press **133** to save the message.
- To use (activate) recorded greeting voice for transit-in calls, choose "**Recorded voice file**" option in "**Trunk Incoming Prompt Voice**".

PPPoE Character Conversion Table

Number	Input Key	Upper Case Letter	Input Key	Lower Case Letter	Input Key	Symbol	Input Key
0	00	A	11	A	41	@	71
1	01	B	12	B	42	•	72
2	02	C	13	C	43	!	73
3	03	D	14	D	44	"	74
4	04	E	15	E	45	\$	75
5	05	F	16	F	46	%	76
6	06	G	17	G	47	&	77
7	07	H	18	H	48	'	78
8	08	I	19	I	49	(79
9	09	J	20	J	50)	80
		K	21	K	51	+	81
		L	22	L	52	,	82
		M	23	M	53	-	83
		N	24	N	54	/	84
		O	25	O	55	:	85
		P	26	P	56	;	86
		Q	27	Q	57	<	87
		R	28	R	58	=	88
		S	29	S	59	>	89
		T	30	T	60	?	90
		U	31	U	61	[91
		V	32	V	62	\	92
		W	33	W	63]	93
		X	34	X	64	^	94
		Y	35	Y	65	_	95
		Z	36	Z	66	{	96
							97
						}	98

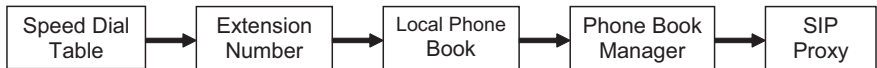
5. Dialing and dialed number routing principle

Instruction

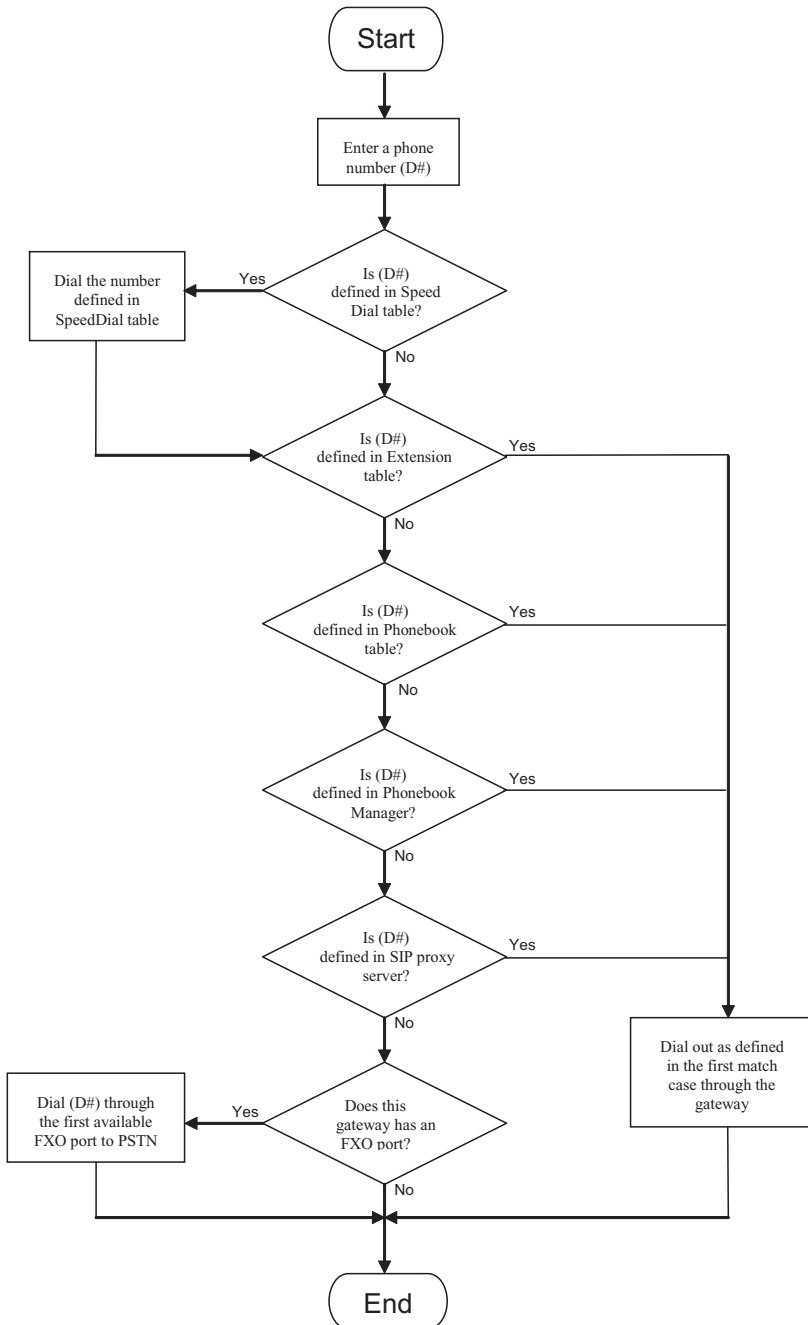
- After a phone number is entered, dial # to call out immediately or, wait until the “Inter DTMF Timeout” expires (defined in “Advanced Options”, default=4 seconds).
- If the phone number fits the setting of Digit Map, the gateway dials out the phone number through the assigned interface automatically.
- The phone number should have at least 2 digits (not including * and #).

Dialed Number Processing Flow

To maintain maximum flexibility, the number dialed will be looked up from several tables defined in the gateway. Once no match can be found, it will look up again from the registered SIP Proxy Server. The look up flow is shown below:



A complete flow chart is on the next page.



Appendix A: Glossary

10BASE-T	A designation for the type of wiring used by Ethernet networks with a data rate of 10 Mbps. Also known as Category 3 (CAT 3) wiring. See also data rate, Ethernet.
100BASE-T	A designation for the type of wiring used by Ethernet networks with a data rate of 100 Mbps. Also known as Category 5 (CAT 5) wiring. See also data rate, Ethernet.
ADSL	Asymmetric Digital Subscriber Line. The most commonly deployed type of DSL for home users. The term asymmetrical refers to its unequal data rates for downloading and uploading (the download rate is higher than the upload rate). The asymmetrical rates benefit home users because they typically download much more data from the Internet than they upload.
analog	Of data, having a form is analogous to the data's original waveform. The voice component in DSL is an analog signal. See also digital.
ATM	Asynchronous Transfer Mode A standard for high-speed transmission of data, text, voice, and video, widely used within the Internet. ATM data rates range from 45 Mbps to 2.5 Gbps. See also data rate.
authenticate	To verify a user's identity, such as by prompting for a password.
binary	The "base two" system of numbers, that uses only two digits, 0 and 1, to represent all numbers. In binary, the number 1 is written as 1, 2 as 10, 3 as 11, 4 as 100, etc. Although expressed as decimal numbers for convenience, IP addresses in actual use are binary numbers; e.g., the IP address 209.191.4.240 is 11010001.10111111.00000100.11110000 in binary. See also bit, IP address, network mask.
bit	Short for "binary digit," a bit is a number that can have two values, 0 or 1. See also binary.
bps	bits per second
bridging	Passing data from your network to your ISP and vice versa using the hardware addresses of the devices at each location. Bridging contrasts with routing, which can add more intelligence to data transfers by using network addresses instead. The My ADSL Modem can perform both routing and bridging. Typically, when both functions are enabled, the device routes IP data and bridges all other types of data. See also routing.
broadband	A telecommunications technology that can send different types of data over the same medium. DSL is a broadband technology.
Broadcast	To send data to all computers on a network.
CO	Central Office A circuit switch that terminates all the local access lines in a particular geographic serving area; a physical building where the local switching equipment is found. xDSL lines running from a subscriber's home connect at their serving central office.
DHCP	Dynamic Host Configuration Protocol DHCP automates address assignment and management. When a computer connects to the LAN, DHCP assigns it an IP address from a shared pool of IP addresses; after a specified time limit, DHCP returns the address to the pool.

DHCP relay	Dynamic Host Configuration Protocol relay. A DHCP relay is a computer that forwards DHCP data between computers that request IP addresses and the DHCP server that assigns the addresses. Each of the My ADSL Modem's interfaces can be configured as a DHCP relay. See DHCP.
DHCP server	Dynamic Host Configuration Protocol server. A DHCP server is a computer that is responsible for assigning IP addresses to the computers on a LAN. See DHCP.
digital	Of data, having a form based on discrete values expressed as binary numbers (0's and 1's). The data component in DSL is a digital signal. See also analog.
DNS	Domain Name System. The DNS maps domain names into IP addresses. DNS information is distributed hierarchically throughout the Internet among computers called DNS servers. When you start to access a web site, a DNS server looks up the requested domain name to find its corresponding IP address. If the DNS server cannot find the IP address, it communicates with higher-level DNS servers to determine the IP address. See also domain name.
domain name	A domain name is a user-friendly name used in place of its associated IP address. For example, www.globespan.net is the domain name associated with IP address 209.191.4.240. Domain names must be unique; their assignment is controlled by the Internet Corporation for Assigned Names and Numbers (ICANN). Domain names are a key element of URLs, which identify a specific file at a web site, e.g., http://www.netcomm.com.au. See also DNS.
download	To transfer data in the downstream direction, i.e., from the Internet to the user.
DSL	Digital Subscriber Line A technology that allows both digital data and analog voice signals to travel over existing copper telephone lines.
Ethernet	Local area network traffic will be carried by standard Category 5 cable referred to as Ethernet. The most commonly installed computer network technology, usually using twisted pair wiring. Ethernet data rates are 10 Mbps and 100 Mbps. See also BASE-T,100BASE-T, twisted pair.
Filtering	To screen out selected types of data, based on filtering rules. Filtering can be applied in one direction (upstream or downstream), or in both directions.
filtering rule	A rule that specifies what kinds of data a routing device will accept and/or reject. Filtering rules are defined to operate on an interface (or multiple interfaces) and in a particular direction (upstream, downstream, or both).
Firewall	Any method of protecting a computer or LAN connected to the Internet from intrusion or attack from the outside. Some firewall protection can be provided by packet filtering and Network Address Translation services.
FTP	File Transfer Protocol - A program used to transfer files between computers connected to the Internet. Common uses include uploading new or updated files to a web server, and downloading files from a web server.
GGP	Gateway to Gateway Protocol. An Internet protocol that specifies how gateway routers communicate with each other.
Gbps	Abbreviation for Gigabits (GIG-uh-bits) per second, or one billion bits per second. Internet data rates are often expressed in Gbps.
GRE	Generic Routing Encapsulation. TCP/IP protocol suite, transport layer encapsulation protocol.

hop	When you send data through the Internet, it is sent first from your computer to a router, and then from one router to another until it finally reaches a router that is directly connected to the recipient. Each individual "leg" of the data's journey is called a hop.
hop count	The number of hops that data has taken on its route to its destination. Alternatively, the maximum number of hops that a packet is allowed to take before being discarded. See also TTL.
host	A device (usually a computer) connected to a network.
HTTP	Hyper-Text Transfer Protocol HTTP is the main protocol used to transfer data from web sites so that it can be displayed by web browsers. See also web browser
ICMP	Internet Control Message Protocol An Internet protocol used to report errors and other network-related information. The ping command makes use of ICMP.
IGMP	Internet Group Management Protocol An Internet protocol that enables a computer to share information about its membership in multicast groups with adjacent routers. A multicast group of computers is one whose members have designated as interested in receiving specific content from the others. Multicasting to an IGMP group can be used to simultaneously update the address books of a group of mobile computer users or to send company newsletters to a distribution list.
in-line filter	See Microfilter
Internet	The global collection of interconnected networks used for both private and business communications.
intranet	A private, company-internal network that looks like part of the Internet (users access information using web browsers), but is accessible only by employees.
IP	See TCP/IP.
IP address	Internet Protocol address The address of a host (computer) on the Internet, consisting of four numbers, each from 0 to 255, separated by periods, e.g., 209.191.4.240. An IP address consists of a network ID that identifies the particular network the host belongs to, and a host ID uniquely identifying the host itself on that network. A network mask is used to define the network ID and the host ID. Because IP addresses are difficult to remember, they usually have an associated domain name that can be specified instead. See also domain name, network mask.
ISP	Internet Service Provider A company that provides Internet access to its customers, usually for a fee.
LAN	Local Area Network A network limited to a small geographic area, such as a home, office, or small building.
LED	Light Emitting Diode An electronic light-emitting device. The indicator lights on the front of the My ADSL Modem are LEDs.
MAC address	Media Access Control address The permanent hardware address of a device, assigned by its manufacturer. MAC addresses are expressed as six pairs of characters.
mask	See network mask.
Mbps	Abbreviation for Megabits per second, or one million bits per second. Network data rates are often expressed in Mbps.

Microfilter	In splitterless deployments, a microfilter is a device that removes the data frequencies in the DSL signal, so that telephone users do not experience interference (noise) from the data signals. Microfilter types include in-line (installs between phone and jack) and wall-mount (telephone jack with built-in microfilter). See also splitterless.
NAT	Network Address Translation A service performed by many routers that translates your network's publicly known IP address into a Private IP address for each computer on your LAN. Only your router and your LAN know these addresses; the outside world sees only the public IP address when talking to a computer on your LAN.
NAT rule	A defined method for translating between public and private IP addresses on your LAN.
network	A group of computers that are connected together, allowing them to communicate with each other and share resources, such as software, files, etc.A network can be small, such as a LAN, or very large, such as the Internet.
network mask	A network mask is a sequence of bits applied to an IP address to select the network ID while ignoring the host ID. Bits set to 1 mean "select this bit" while bits set to 0 mean "ignore this bit." For example, if the network mask 255.255.255.0 is applied to the IP address 100.10.50.1, the network ID is 100.10.50, and the host ID is 1. See also binary, IP address, subnet
NIC	Network Interface Card An adapter card that plugs into your computer and provides the physical interface to your network cabling, which for Ethernet NICs is typically an RJ-45 connector. See Ethernet, RJ-45.
Pass-through Line	The line that connects the NB9/NB9W to a POTS line may be referred to as a pass-through line
packet	Data transmitted on a network consists of units called packets. Each packet contains a payload (the data), plus overhead information such as where it came from (source address) and where it should go (destination address).
ping	Packet Internet (or Inter-Network) Groper A program used to verify whether the host associated with an IP address is online. It can also be used to reveal the IP address for a given domain name.
port	A physical access point to a device such as a computer or router, through which data flows into and out of the device.
POTS	Plain Old Telephone Service Traditional analog telephone service using copper telephone lines. Pronounced pots. See also PSTN.
POTS splitter	See splitter.
PPP	Point-to-Point Protocol A protocol for serial data transmission that is used to carry IP (and other protocol) data between your ISP and your computer. The WAN interface on the My ADSL Modem uses two forms of PPP called PPPoA and PPPoE. See also PPPoA, PPPoE.
PPPoA	Point-to-Point Protocol over ATM One of the two types of PPP interfaces you can define for a Virtual Circuit (VC), the other type being PPPoE. You can define only one PPPoA interface per VC.
PPPoE	Point-to-Point Protocol over Ethernet One of the two types of PPP interfaces you can define for a Virtual Circuit (VC), the other type being PPPoA. You can define one or more PPPoE interfaces per VC.

protocol	A set of rules governing the transmission of data. In order for a data transmission to work, both ends of the connection have to follow the rules of the protocol.
remote	In a physically separate location. For example, an employee away on travel who logs in to the company's intranet is a remote user.
RIP	Routing Information Protocol The original TCP/IP routing protocol. There are two versions of RIP: version and version II.
RJ-11	Registered Jack Standard-11 The standard plug used to connect telephones, fax machines, modems, etc. to a telephone jack. It is a 6-pin connector usually containing four wires.
RJ-45	Registered Jack Standard-45 The 8-pin plug used in transmitting data over phone lines. Ethernet cabling usually uses this type of connector.
routing	Forwarding data between your network and the Internet on the most efficient route, based on the data's destination IP address and current network conditions. A device that performs routing is called a router.
rule	See filtering rule, NAT rule.
SDNS	Secondary Domain Name System (server) A DNS server that can be used if the primary DSN server is not available. See DNS.
SNMP	Simple Network Management Protocol The TCP/IP protocol used for network management.
splitter	A device that splits off the voice component of the DSL signal to a separate line, so that data and telephone service each have their own wiring and jacks. The splitter is installed by your telephone company where the DSL line enters your home. The CO also contains splitters that separate the voice and data signals, sending voice to the PSTN and data on high-speed lines to the Internet. See also CO, PSTN, splitterless, microfilter.
splitterless	A type of DSL installation where no splitter is installed, saving the cost of a service call by the telephone company. Instead, each jack in the home carries both voice and data, requiring a microfilter for each telephone to prevent interference from the data signal. ADSL is usually splitterless; if you are unsure if your installation has a splitter, ask your DSL provider. See also splitter, microfilter.
subnet	A subnet is a portion of a network. The subnet is distinguished from the larger network by a subnet mask which selects some of the computers of the network and excludes all others. The subnet's computers remain physically connected to the rest of the parent network, but they are treated as though they were on a separate network. See also network mask.
subnet mask	A mask that defines a subnet. See also network mask.
TCP	See TCP/IP.
TCP/IP	Transmission Control Protocol/Internet Protocol The basic protocols used on the Internet. TCP is responsible for dividing data up into packets for delivery and reassembling them at the destination, while IP is responsible for delivering the packets from source to destination. When TCP and IP are bundled with higher-level applications such as HTTP, FTP, Telnet, etc., TCP/IP refers to this whole suite of protocols.
Telnet	An interactive, character-based program used to access a remote computer. While HTTP (the web protocol) and FTP only allow you to download files from a remote computer, Telnet allows you to log into and use a computer from a remote location.

TFTP	Trivial File Transfer Protocol. A protocol for file transfers, TFTP is easier to use than File Transfer Protocol (FTP) but not as capable or secure.
TTL	Time To Live A field in an IP packet that limits the life span of that packet. Originally meant as a time duration, the TTL is usually represented instead as a maximum hop count; each router that receives a packet decrements this field by one. When the TTL reaches zero, the packet is discarded.
twisted pair	The ordinary copper telephone wiring long used by telephone companies. It contains one or more wire pairs twisted together to reduce inductance and noise. Each telephone line uses one pair. In homes, it is most often installed with two pairs. For Ethernet LANs, a higher grade called Category 3 (CAT 3) is used for 10BASE-T networks, and an even higher grade called Category 5 (CAT 5) is used for 100BASE-T networks. See also 10BASE-T, 100BASE-T, Ethernet.
upstream	The direction of data transmission from the user to the Internet.
USB	Universal Serial Bus A serial interface that lets you connect devices such as printers, scanners, etc. to your computer by simply plugging them in. The My ADSL Modem is equipped with a USB interface for connecting to a stand-alone PC.
VC	Virtual Circuit A connection from your ADSL router to your ISP.
VCI	Virtual Circuit Identifier Together with the Virtual Path Identifier (VPI), the VCI uniquely identifies a VC. Your ISP will tell you the VCI for each VC they provide. See also VC.
VPI	Virtual Path Identifier Together with the Virtual Circuit Identifier (VCI), the VPI uniquely identifies a VC. Your ISP will tell you the VPI for each VC they provide. See also VC.
VSP	VoIP Service provider
WAN	Wide Area Network Any network spread over a large geographical area, such as a country or continent. With respect to the My ADSL Modem, WAN refers to the Internet.
Web browser	A software program that uses Hyper-Text Transfer Protocol (HTTP) to download information from (and upload to) web sites, and displays the information, which may consist of text, graphic images, audio, or video, to the user. Web browsers use Hyper-Text Transfer Protocol (HTTP). Popular web browsers include Netscape Navigator and Microsoft Internet Explorer. See also HTTP, web site, WWW.
Web page	A web site file typically containing text, graphics and hyperlinks (cross-references) to the other pages on that web site, as well as to pages on other web sites. When a user accesses a web site, the first page that is displayed is called the Home page. See also hyperlink, web site.
Web site	A computer on the Internet that distributes information to (and gets information from) remote users through web browsers. A web site typically consists of web pages that contain text, graphics, and hyperlinks. See also hyperlink, web page.
WLAN	Wireless Local Area Network
WWW	World Wide Web Also called (the) Web. Collective term for all web sites anywhere in the world that can be accessed via the Internet.POTS A telephone line used for a standard phone-line and service will be referred to as POTS (=Plain Old Telephone Service)

Appendix B: Cable Information

This cable information is provided for your reference only. Please ensure you only connect the appropriate cable into the correct socket on either this product or your computer.

If you are unsure about which cable to use or which socket to connect it to, please refer to the hardware installation section in this manual. If you are still not sure about cable connections, please contact a professional computer technician or NetComm for further advice.

RJ-45 Network Ports

RJ-45 Network Ports can connect any networking devices that use a standard LAN interface, such as a Hub/Switch Hub or Router. Use unshielded twisted-pair (UTP) or shield twisted-pair (STP) cable to connect the networking device to the RJ-45 Ethernet port. Depending on the type of connection, 10Mbps or 100Mbps, use the following Ethernet cable, as prescribed.

10Mbps: Use EIA/TIA-568-100-Category 3, 4 or 5 cable.

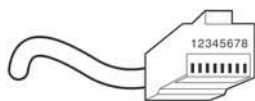
100Mbps: Use EIA/TIA-568-100-Category 5 cable.

Note: To prevent loss of signal, make sure that the length of any twisted-pair connection does not exceed 100 metres.



RJ-45 Connector Pin Assignment	Normal Assignment
1	Input Receive Data +
2	Input Receive Data -
3	Output Transmit Data +
6	Output Transmit Data -
4,5,7,8	Not used

Figure 1



RJ-45 plug attached to cable

Figure 2

Straight and crossover cable configuration

There are two types of the wiring: Straight-Through Cables and Crossover Cables. Category 5 UTP/STP cable has eight wires inside the sheath. The wires form four pairs. Straight-Through Cables has same pinouts at both ends while Crossover Cables has a different pin arrangement at each end.

In a straight-through cable, wires 1,2,3,4,5,6,7 and 8 at one end of the cable are still wires 1~8 at the other end. In a crossover cable, the wires of 1,2,3,6 are reversed so that wire 1 become 3 at the other end of the cable, 2 becomes 6, and so forth.

To determine which wire is wire 1, hold the RJ-45 cable tip with the spring clip facing towards the ground and the end pointing away from you. The copper wires exposed upwards to your view. The first wire on the far left is wire 1. You can also refer to the illustrations and charts of the internal wiring on the following page.

Straight-Through Cabling



Figure 3

Wire	Becomes
1	1
2	2
3	3
6	6

Cross-Over Cabling



Figure 4

Wire	Becomes
1	3
2	6
3	1
6	2

Note: To prevent loss of signal, make sure that the length of any twisted-pair connection does not exceed 100 metres.

Appendix C: Registration and Warranty Information

All NetComm Limited ("NetComm") products have a standard 12 month warranty from date of purchase against defects in manufacturing and that the products will operate in accordance with the specifications outlined in the User Guide. However some products have an extended warranty option (please refer to your packaging). To be eligible for the extended warranty you must supply the requested warranty information to NetComm within 30 days of the original purchase by registering on-line via the NetComm web site at:

www.netcomm.com.au

Contact Information

If you have any technical difficulties with your product, please do not hesitate to contact NetComm's Customer Support Department.

Email: support@netcomm.com.au
Fax: (+612) 9424-2010
Web: www.netcomm.com.au

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Customer Information

ACA (Australian Communications Authority) requires you to be aware of the following information and warnings:

- (1) This unit shall be connected to the Telecommunication Network through a line cord which meets the requirements of the ACA TS008 Standard.
- (2) This equipment has been tested and found to comply with the Standards for C-Tick and or A-Tick as set by the ACA. These standards are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio noise and, if not installed and used in accordance with the instructions detailed within this manual, may cause interference to radio communications. However, there is no guarantee that interference will not occur with the installation of this product in your home or office. If this equipment does cause some degree of interference to radio or television reception, which can be determined by turning the equipment off and on, we encourage the user to try to correct the interference by one or more of the following measures:
 - Change the direction or relocate the receiving antenna.
 - Increase the separation between this equipment and the receiver.
 - Connect the equipment to an alternate power outlet on a different power circuit from that to which the receiver/TV is connected.
 - Consult an experienced radio/TV technician for help.
- (3) The power supply that is provided with this unit is only intended for use with this product. Do not use this power supply with any other product or do not use any other power supply that is not approved for use with this product by NetComm. Failure to do so may cause damage to this product, fire or result in personal injury.

Product Warranty

The warranty is granted on the following conditions:

1. This warranty extends to the original purchaser (you) and is not transferable;
2. This warranty shall not apply to software programs, batteries, power supplies, cables or other accessories supplied in or with the product;
3. The customer complies with all of the terms of any relevant agreement with NetComm and any other reasonable requirements of NetComm including producing such evidence of purchase as NetComm may require;
4. The cost of transporting product to and from NetComm's nominated premises is your responsibility; and,
5. NetComm does not have any liability or responsibility under this warranty where any cost, loss, injury or damage of any kind, whether direct, indirect, consequential, incidental or otherwise arises out of events beyond NetComm's reasonable control. This includes but is not limited to: acts of God, war, riot, embargoes, acts of civil or military authorities, fire, floods, electricity outages, lightning, power surges, or shortages of materials or labour.
6. The customer is responsible for the security of their computer and network at all times. Security features may be disabled within the factory default settings. NetComm recommends that you enable these features to enhance your security.

The warranty is automatically voided if:

1. You, or someone else, use the product, or attempts to use it, other than as specified by NetComm;
2. The fault or defect in your product is the result of a voltage surge subjected to the product either by the way of power supply or communication line, whether caused by thunderstorm activity or any other cause(s);
3. The fault is the result of accidental damage or damage in transit, including but not limited to liquid spillage;
4. Your product has been used for any purposes other than that for which it is sold, or in any way other than in strict accordance with the user manual supplied;
5. Your product has been repaired or modified or attempted to be repaired or modified, other than by a qualified person at a service centre authorised by NetComm; and,
6. The serial number has been defaced or altered in any way or if the serial number plate has been removed.

Limitations of Warranty

The Trade Practices Act 1974 and corresponding State and Territory Fair Trading Acts or legalisation of another Government ("the relevant acts") in certain circumstances imply mandatory conditions and warranties which cannot be excluded. This warranty is in addition to and not in replacement for such conditions and warranties.

To the extent permitted by the Relevant Acts, in relation to your product and any other materials provided with the product ("the Goods") the liability of NetComm under the Relevant Acts is limited at the option of NetComm to:

- Replacement of the Goods; or
- Repair of the Goods; or
- Payment of the cost of replacing the Goods; or
- Payment of the cost of having the Goods repaired.



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