



WELCOME TO THE WONDERFUL WORLD OF VOIP



Congratulations on your purchase of a NetComm VoIP ATA. Whether this is your first VoIP device or you are upgrading from another device we're glad to have you as part of the NetComm family. The NetComm V110/V210P/V211/V220 ATA's are true next generation technology and in addition to the fantastic savings you're going to experience you'll also get fantastic features such as routing, T.38 Fax and 3 way conferencing.

This **Quick Start Guide** contains everything you need to get you up and running for everyday use but if you need to delve into any of the more advance configuration features available on your unit you'll find a comprehensive **User Guide** on the **CD** that accompanies the ATA.

Should you run into any problems with your unit (and let's face it, we all need a little help from time to time) you can find the answers to many questions in the support section of our website www.netcomm.com.au/Support/

So without any further delay it's time to get you more familiar with your ATA so you can start using VoIP.

***IMPORTANT NOTICE:** Please ensure that the Broadband modem/router to be used with this ATA has been designed to handle VoIP traffic. Most new broadband modem/router will be able to handle VoIP without any issues and do not need extra configuration to handle VoIP service. However, old broadband modem/router might need to be configured correctly to support it.*

Please check with the broadband modem/router manufacturers for more details.

VOIP ATA FEATURES

Network Protocol	Tone
• SIP v1 (RFC2543), v2(RFC3261)	• Ring Tone
• IP/TCP/UDP/RTP/RTCP	• Ring Back Tone
• IP/ICMP/ARP/RARP/SNTP	• Dial Tone
• TFTP Client/DHCP	• Busy Tone
• Client/ PPPoE Client	• Programming Tone
• Telnet/HTTP Server	
• DNS Client	
• NAT/DHCP Server	
Codec	Phone Function
• G.711: 64k bit/s (PCM)	• Volume Adjustment
• G.723.1: 6.3k / 5.3k bit/s	• Speed dial key
• G.726: 16k / 24k / 32k / 40k bit/s (ADPCM)	• Phone book
• G.729A: 8k bit/s (CS-ACELP)	• Flash
• G.729B: adds VAD & CNG to G.729	
Voice Quality	Call Function
• VAD: Voice activity detection	• Call Hold
• CNG: Comfort noise generator	• Call Waiting
• LEC: Line echo canceller	• Call Forward
• Packet Loss Compensation	• Caller ID
• Adaptive Jitter Buffer	• 3-way conference
IP Assignment	Auto Provisioning
• Static IP	• HTTP
• DHCP	• FTP
• PPPoE	• TFTP
DTMF Function	NAT Traversal
• In-Band DTMF	• STUN
• Out-of Band DTMF	
• SIP Info	
SIP Server	Configuration
• Registrar Server	• Web Browser
• Outbound Proxy	• IVR/Keypad

Firmware Upgrade	Security
<ul style="list-style-type: none">• TFTP	<ul style="list-style-type: none">• HTTP 1.1 basic/digest authentication for Web setup
<ul style="list-style-type: none">• HTTP	<ul style="list-style-type: none">• MD5 for SIP authentication (RFC2069/ RFC 2617)
Interface	Modem & Fax modes
1 WAN port interface	<ul style="list-style-type: none">• G.711 fax/modem pass-through with fax/modem detection
1 LAN port interface	<ul style="list-style-type: none">• T.38 support
1 PSTN port interface (FXO) (Optional)	
1 VOIP port interface (FXS)	

PACKAGE CONTENTS

Your NetComm ATA package contains the following items:

- V2xx ATA
- Quick Installation Guide
- User Guide and Software on CD
- RJ-45 Straight-through Ethernet Cable
- DC12 volts/0.6 A Power Supply Unit
- One RJ-11 telephone cable (V210P / V211 only)

MINIMUM SYSTEM REQUIREMENTS

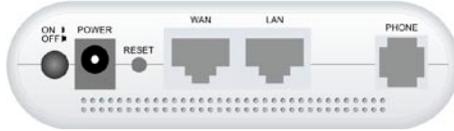
Before continuing with the installation of your Netcomm V2xx ATA, please confirm that you comply with the minimum system requirements:

- Broadband (ADSL or Cable) connection
- PC with TCP/IP networking protocol installed
- Windows XP/Vista PC for Easy Setup Utility
- VoIP account with a VSP (VoIP Service Provider)

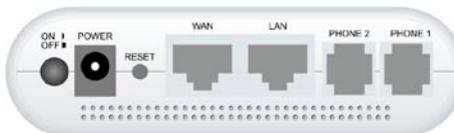
GETTING TO KNOW YOUR NETCOMM V2XX ATA

It is recommended that you take a moment to acquaint yourself with the indicator lights, ports and default settings of the V2xx ATA prior to commencing with installation.

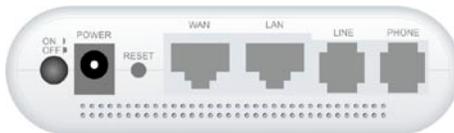
Ports and Buttons



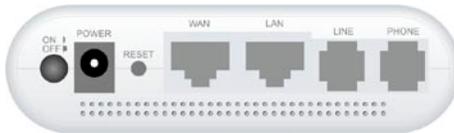
V110 (WAN + 1 LAN + 1 FXS)



V220 (WAN + 1 LAN + 2 FXS)



V211 (WAN + 1 LAN + 1 FXO + 1 FXS)



V210P (WAN + 1 LAN + 1 PSTN pass through + 1 FXS)

Port / Button	Description
Power	Connect the supplied power adapter to the ATA. The power requirement is DC12 volts/0.6 A <i>Note: Using a power supply with a different voltage rating will damage this product</i>
Power Switch	Power on/off the NetComm ATA.
WAN Port	Connect to Broadband devices, such as an ADSL or Cable modem.
LAN Port	Connect to Ethernet network devices, such as a PC, switch, or router. Depending on the connection, you may need a cross over cable or a straight through cable
RESET	The Reset button will set the NetComm ATA to its factory default setting and reset the unit. You may need to reset the NetComm ATA into its factory defaults if you loose the ability to enter the NetComm ATA via the web interface, or following a software upgrade.
LINE Socket	Connect a telephone cable between the NetComm ATA line jack and a wall jack.
PHONE Socket	Connect a standard telephone handset to the NetComm ATA phone jack using a telephone cable.

LED Indicators



LED	Description
PWR	This LED illuminates to indicate the system is powered.
SIP	This LED illuminates when the NetComm ATA is registered successfully to the SIP server.
ETH	This LED illuminates when a connection is established to WAN/LAN port and flashes when WAN/LAN port is sending/receiving data.

Default Settings

LAN (Management)	
Static IP Address:	192.168.22.1
Subnet Mask:	255.255.255.0
Default Gateway:	blank
WAN (Internet)	
WAN mode:	DHCP
Modem Access	
Username:	admin
Password:	admin

Many configuration changes made using the web interface will only become active after clicking the Save & Reboot button on the Save Savings / Reboot page in the web configuration page.

Note: Certain voice parameters do not require a Save & Reboot to take effect. These Voice Parameters will take effect on the next voice call after the Voice Parameter is entered and submitted. However, if Save & Reboot is not done, then these Voice Parameters will not be saved over a power cycle. The Voice Parameters that can be changed "on the fly" are noted in the respective sections.

Restoring Factory Defaults

The reset button will reset the ATA to its factory default configuration. Occasions may present themselves where you need to restore the factory default settings on your ATA. Typical situations are:

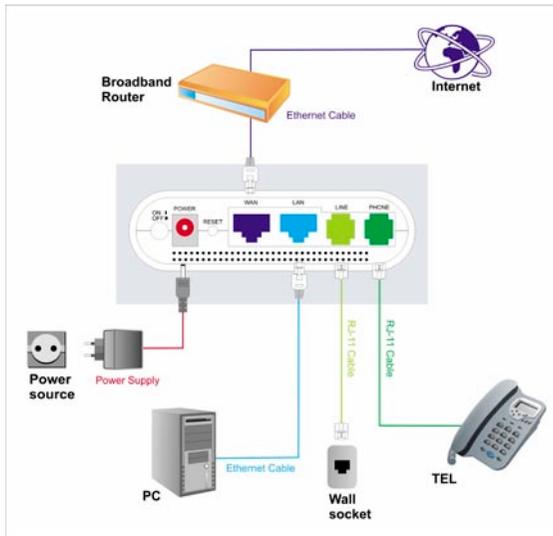
- You have lost your password and unable to login to the ATA;
- You have purchased the router from someone else and need to reconfigure the device.
- You are asked to perform a factory reset by a member of the excellent NetComm Support Staff.

In order to restore your router to its factory default settings, please follow these steps:

- Ensure that the ATA is powered on (for at least 20 seconds).
- Use a paper clip or a pencil tip to depress the reset button for ten seconds and release. At this point, the reset is in progress. Do not power off the unit at this point.
- After the ATA reboots, the default settings are now restored. This entire process takes several minutes to complete.
- Once you have reset the ATA to its default settings you will be able to access the device's web configuration using <http://192.168.22.1> with username "admin" and password "admin".

HARDWARE INSTALLATION

1. Connect the WAN port to your broadband router using the Ethernet cable supplied. If you would like your PC to be connected to the network through the V100/V210P/V211/V220, you can connect another Ethernet cable (not supplied) from your PC to the LAN Ethernet port. This enables your PC and the ATA to share one Ethernet port on your router/switch.



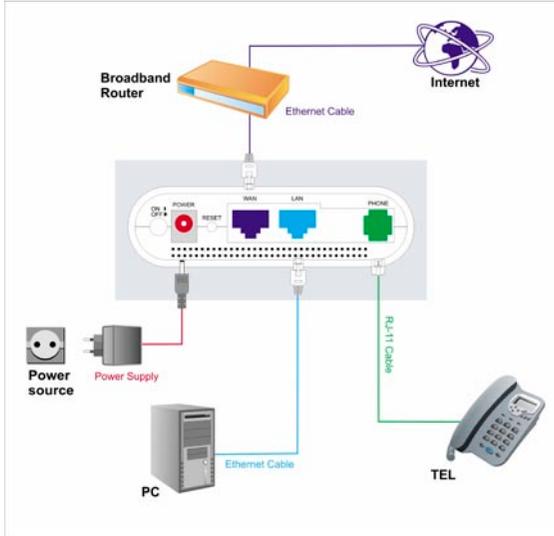
V211 and V210P

2. Connect your telephone handset to the Phone port. For V220, you can connect an additional handset to Phone 2 port.

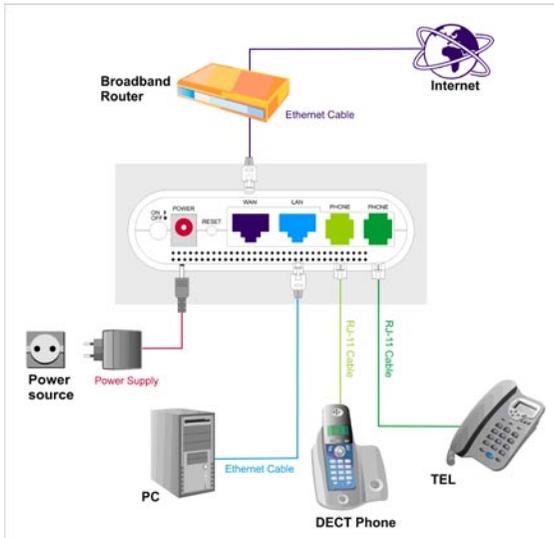
Notes: A second VoIP account is needed unless your VoIP provider supports multiple connections from the one account and both ports need to be configured individually (even when using one account that supports multiple connections).

3. For V210P or V211 unit, connect the Line port to your PSTN wall socket.
4. Connect the AC power adaptor and press the power button to turn the ATA on. The ATA will boot up and in a few minutes, the Power and Ethernet Lights will light up.

Note: Only use the power adaptor supplied with the NetComm ATA. A different adapter may damage the product.



V110



V220

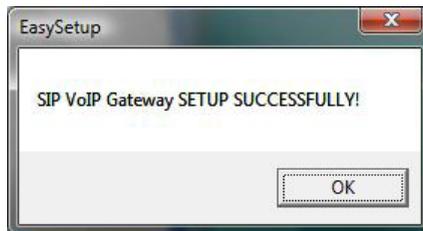
SOFTWARE CONFIGURATION

Easy Setup Utility (for Windows Only)

1. For easy configuration, insert the included CD into your CD-ROM drive. The CD should auto-start. If it does not start, click on Start -> Run and type in D:\Easysetup\vpES.exe (where D is the drive letter of your CD-ROM drive).
2. The default Internet access is DHCP Client Mode. Click on WAN Configuration to change the internet access type.
3. Enter the SIP Proxy Domain, Proxy Server, Outbound Proxy, Username, Auth. ID and Auth. Password provided by your VSP.
4. And then click on Setup, the software will start to configure the NetComm ATA. Follow the instructions of the Easy Setup utility which will guide you to complete the configuration.



5. After a few minutes, the NetComm ATA should report that the setup has been successful.



Note: The easy setup utility can not be used to retrieve details. The utility will initialize the unit back to default settings when executed.

Configuration via a web browser

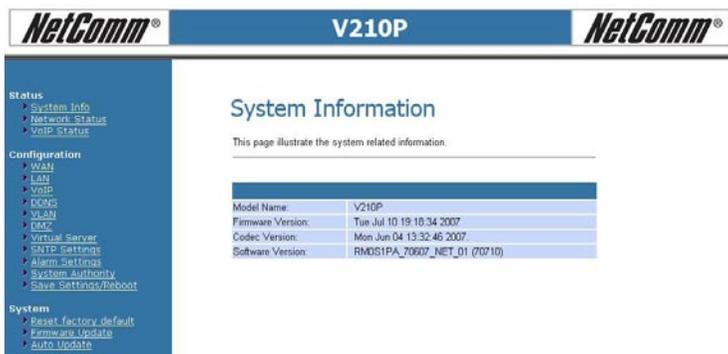
If you do not have a Windows PC available, or you want to perform more complex configuration, you can configure the NetComm ATA using a web browser.

Accessing the web configuration

1. Open your Web browser (e.g. Internet Explorer, Firefox).
2. Enter the LAN port default IP address of the NetComm ATA (e.g. <http://192.168.22.1>) in the address bar and press Enter.
3. At the login screen, enter the gateway username and password. The default login username is admin, and the default login password is admin.



The image shows a web browser window displaying the login page for a NetComm V210P device. The page has a blue header with the text "Login V210P". Below the header, there is a light blue background with the instruction "Enter your username and password to login" and the device model "V210P". There are two input fields: "Username" and "Password". At the bottom of the form, there are two buttons: "Login" and "Clear".



The image shows the main configuration page of the NetComm V210P web interface. The page has a blue header with the NetComm logo on the left and right, and "V210P" in the center. On the left side, there is a navigation menu with the following categories: "Status" (System Info, Network Status, VoIP Status), "Configuration" (Web, LAN, VoIP, DNS, SIP, DMZ, Virtual Server, SIP Settings, Alarm Settings, System Authnity, Save Settings/Reboot), and "System" (Reset factory default, Firmware Update, Auto Update). The main content area is titled "System Information" and contains the text "This page illustrate the system related information:". Below this text is a table with the following data:

Model Name:	V210P
Firmware Version:	Tue Jul 10 19:18:34 2007
Codec Version:	Mon Jun 04 13:32:46 2007
Software Version:	RMDSIPA_70607_NET_01 (70710)

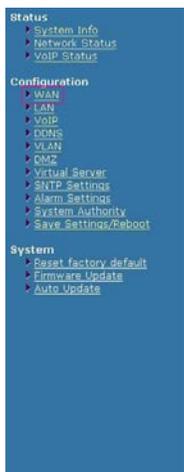
WAN Configuration

There are three types of WAN connection that you can use

- Static IP, to specify the IP address manually. Mostly used when you connect the V2xx ATA to a broadband router with out a DHCP server.
- Dynamic IP, to connect to a service which has a DHCP server on the network. Mostly used when using the V2xx ATA with an Optus cable modem or connecting the V2xx ATA to another router / gateway.
- PPPoE Client, to connect to a network with PPPoE service. Mostly used to connect the V2xx ATA to a bridge ADSL modem.

Static IP Configuration

1. Click on WAN on the left menu.
2. Change the IP Type to DHCP.
3. Enter the required information, IP Address, Subnet Mask, Gateway, DNS Server1, DNS Server2 and Host Name in the respective fields.
4. After that, click the Submit button.



WAN Settings

You can configure the WAN settings in this page.

LAN Mode: Bridge NAT

WAN Setting	
IP Type:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input type="radio"/> PPPoE
IP:	192.168.1.5
Mask:	255.255.255.0
Gateway:	192.168.1.1
DNS Server1:	192.168.1.1
DNS Server2:	4.2.2.2
MAC:	000296127411
Host Name:	
PPPoE Setting	
User Name:	
Password:	
Service Name:	
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

Option	Description
LAN Mode	Configures the VoIP Gateway as a router or bridge.
IP Type	Fixed IP address allows the user to manually assign an IP address. DHCP mode allows the VoIP Gateway to automatically obtain an IP address from a DHCP enabled server on your network. PPPoE allows the VoIP Gateway to use a PPPoE connection. In this mode, enter the PPPoE Username and Password in the fields under PPPoE settings.
IP	The unique address of the VoIP Gateway on your network.
Mask	The subnet mask of your network.
Gateway	The default gateway address through which the VoIP Gateway communicates with the Internet. This is usually your modem / router.
DNS Server 1	The Domain Name Service that the NetComm ATA uses to resolve domain names to IP addresses. This is usually your modem / Router or the primary DNS server specified by your Internet Service Provider (ISP).
DNS Server 2	The Domain Name Service that the NetComm ATA uses to resolve domain names to IP addresses. This is usually your modem / router. DNS 2 can be a different address to DNS 1 to provide a backup DNS server if DNS 1 fails.
MAC	The Ethernet MAC address of the WAN interface.
Host Name	The host name of the VoIP Gateway (Optional)

5. Click on Save Settings/Reboot on the left menu and then click the Save & Reboot button to save your settings.

Save Settings / Reboot

You have to save settings & reboot to effect them.

Save Settings and reboot:

Save & Reboot

You can press the reboot button to restart the system.

Reboot system without saving settings:

Reboot Only

6. The NetComm ATA will reboot automatically to make your changes effective. Please wait while it reboots.

Information

This page inform user important information.

System will reboot automatically to effect those changes and please wait for a moment while rebooting....

DHCP Mode

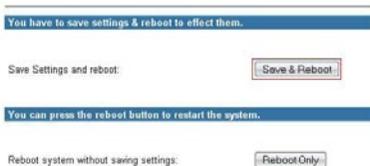
1. Click on WAN on the left menu.
2. Change the IP Type to DHCP.
3. After that, click the Submit button.



Option	Description
DNS Server 1	The Domain Name Service that the NetComm ATA uses to resolve domain names to IP addresses. This is usually your modem / Router or the primary DNS server specified by your Internet Service Provider (ISP). (Optional)
DNS Server 2	The Domain Name Service that the NetComm ATA uses to resolve domain names to IP addresses. This is usually your modem / router. DNS 2 can be a different address to DNS 1 to provide a backup DNS server if DNS 1 fails. (Optional)
MAC	The Ethernet MAC address of the WAN interface. (Optional)
Host Name	The host name of the VoIP Gateway. (Optional)

4. Click on Save Settings/Reboot on the left menu and then click the Save & Reboot button to save your settings.

Save Settings / Reboot



5. The NetComm ATA will reboot automatically to make your changes effective. Please wait while it reboots.

Information

This page inform user important information.

System will reboot automatically to effect those changes and please wait for a moment while rebooting....

PPPoE Client Mode

1. Click on WAN on the left menu.
2. Change the IP Type to DHCP.
3. Enter the username and password supplied by your Internet Service Provider.
4. After that, click the Submit button.



WAN Settings

You can configure the WAN settings in this page.

LAN Mode: Bridge NAT

WAN Setting	
IP Type:	<input type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input checked="" type="radio"/> PPPoE
IP:	192.168.1.5
Mask:	255.255.255.0
Gateway:	192.168.1.1
DNS Server1:	192.168.1.1
DNS Server2:	4.2.2.2
MAC:	000286127411
Host Name:	
PPPoE Setting	
User Name:	username@ispname.cc
Password:	password
Service Name:	
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

Option	Description
User Name	Your PPPoE username. This is normally specified by your Internet Service Provider (ISP).
Password	Password for the above PPPoE account.
Service Name	The name of your service provider. (Optional)

5. Click on Save Settings/Reboot on the left menu and then click the Save & Reboot button to save your settings.

Save Settings / Reboot

You have to save settings & reboot to effect them.

Save Settings and reboot:

You can press the reboot button to restart the system.

Reboot system without saving settings:

6. The NetComm ATA will reboot automatically to make your changes effective. Please wait while it reboots.

Information

This page inform user important information.

System will reboot automatically to effect those changes and please wait for a moment while rebooting...

VoIP Configuration

1. Click on VoIP on the left menu.
2. Click on SIP Service Provider.



VoIP Configuration

You can configure the VoIP settings, please click the hyperlink.



3. Enter the information supplied by your VoIP service provider and click Submit.

SIP Service Provider

You can set information of service domains in this page.

SIP Service Provider	
Active:	<input checked="" type="radio"/> On <input type="radio"/> Off
SIP Proxy Domain:	<input type="text"/>
Proxy Server:	<input type="text"/>
Outbound Proxy:	<input type="text"/>
Display Name:	<input type="text"/>
User Name:	<input type="text"/>
Auth. ID:	<input type="text"/>
Auth. Password:	<input type="text"/>
Subscribe for MWI:	<input type="radio"/> On <input checked="" type="radio"/> Off
Status:	Not Registered
SIP Expire Time:	<input type="text" value="60"/> (15~86400 sec)
Use DNS SRV:	<input type="radio"/> On <input checked="" type="radio"/> Off

Field	Description
Active	If set to on, the NetComm ATA will attempt to register with your VoIP service provider using this account.
SIP Proxy Domain	The SIP Proxy Domain is specified by your VSP.
Proxy Server	The host name of IP address of the SIP server. The proxy server is responsible for forwarding requests received from the NetComm ATA to your VoIP service provider. The Proxy Server is specified by your VoIP service provider.
Outbound Proxy	The host name of IP address of the Outbound Proxy server. The Outbound proxy server is responsible for forwarding requests received from the NetComm ATA to your VoIP service provider. The Proxy Server is specified by your VoIP service provider.
Display Name	The name or number the called party sees when you call them.
User Name	Your VoIP number. This number is assigned by your VoIP service provider.
Auth. Name	The authentication username assigned by your VoIP service provider. This can be different to your VoIP number.
Auth. Password	The authentication password provided by your VoIP service provider.
Status	Indicates if the NetComm ATA is registered to your VSP. If "Registered" is shown, you can make and receive calls via your VoIP service provider. If the NetComm ATA reports "not registered", check the SIP Service Provider settings above and try again.
SIP Expiry Time	This setting specifies a recommendation to the SIP server on the re-registration interval. Increasing this time will send less registration traffic, but can cause other issues.

- Click on Save Settings/Reboot on the left menu and then click the Save & Reboot button to save your settings.

Save Settings / Reboot

You have to save settings & reboot to effect them.

Save Settings and reboot:

You can press the reboot button to restart the system.

Reboot system without saving settings:

- The NetComm ATA will reboot automatically to make your changes effective. Please wait while it reboots.

Information

This page inform user important information.

System will reboot automatically to effect those changes and please wait for a moment while rebooting...

- Once the device has rebooted, check if the SIP LED is illuminated to indicate successful registration with your VoIP Service Provider. If not, please check your settings.

ADVANCED CONFIGURATION

Your NetComm ATA has many advanced features that you may want or need to use in the future. Let's start by taking a look at the menus in the web interface.

1. Open your Web browser (Internet Explorer, Firefox etc).
2. Enter the LAN port default IP address of the NetComm ATA (e.g. <http://192.168.22.1>) in the address bar and press Enter.
3. At the login screen, enter the gateway username and password. The default login username is admin, and the default login password is admin.

Login V210P

Enter your username and password to login

V210P

Username

Password

NetComm® **V210P** **NetComm®**

Status

- System Info
- Network Status
- VoIP Status

Configuration

- WAN
- LAN
- VoIP
- DDNS
- VLAN
- DMZ
- Virtual Server
- SNTP Settings
- Alarm Settings
- System Authority
- Save Settings/Reboot

System

- Reset factory default
- Firmware Update
- Auto Update

System Information

This page illustrate the system related information.

Model Name:	V210P
Firmware Version:	Tue Jul 10 19:18:34 2007
Codec Version:	Mon Jun 04 13:32:46 2007
Software Version:	RMOS1PA_70607_NET_01 (70710)

4. On the Home Page, click on VoIP on the left menu to view the VoIP Gateway Configuration page.

Many configuration changes made using the web interface will only become active after clicking the Save & Reboot button on the Save Savings / Reboot page.

Note: Certain voice parameters do not require a Save & Reboot to take effect. These Voice Parameters will take effect on the next voice call after the Voice Parameter is entered and submitted. However, if Save & Reboot is not done, then these Voice Parameters will not be saved over a power cycle. The Voice Parameters that can be changed "on the fly" are noted in the respective sections.

Status Page

System Information Page

This page illustrate the system related information

System Information

This page illustrate the system related information.

Model Name:	V210P
Firmware Version:	Tue Jul 10 19:18:34 2007
Codec Version:	Mon Jun 04 13:32:46 2007
Software Version:	RMDS1PA_70607_NET_01 (70710)

Network Status Page

You can view the current Network setting in this page.

Network Status

This page shows current status of network interfaces of the system.

WAN Status	
Type:	Fixed IP Client
IP:	172.17.1.69
Mask:	255.255.0.0
Gateway:	172.17.1.1
DNS Server 1:	0.0.0.0
DNS Server 2:	0.0.0.0

LAN Status	
Type:	DHCP Server
IP:	192.168.22.1
Mask:	255.255.255.0
DNS Server 1:	0.0.0.0
DNS Server 2:	0.0.0.0

VoIP Status Page

The page shows current status of VoIP SIP Service provider.

VoIP Status

The page shows current status of VoIP SIP Service provider.

VoIP SIP Provider Status	
Domain Server:	192.168.10.100
Display Name:	1234
User Name:	1234
Status:	Registered

Configuration

WAN

You can configure the WAN settings in this page.

WAN Settings

You can configure the WAN settings in this page.

LAN Mode: Bridge NAT

WAN Setting	
IP Type:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input type="radio"/> PPPoE
IP:	<input type="text" value="172.17.1.69"/>
Mask:	<input type="text" value="255.255.0.0"/>
Gateway:	<input type="text" value="172.17.1.1"/>
DNS Server1:	<input type="text" value="0.0.0.0"/>
DNS Server2:	<input type="text" value="0.0.0.0"/>
MAC:	<input type="text" value="000296127411"/>
Host Name:	<input type="text" value="VOIP_TA1S1P"/>

PPPoE Setting	
User Name:	<input type="text"/>
Password:	<input type="text"/>
Service Name:	<input type="text"/>

The LAN mode allows the VoIP Gateway to be configured as a router or a bridge.

- Bridge: The system serves as a bridge between WAN port and LAN port.
- NAT: The system serves as a NAT router, allowing multiple computers and the VoIP gateway to share a single public IP address.

The IP Type determines how the VoIP Gateway obtains its public IP address. Fixed IP allows you to manually enter a fixed IP address, mask and gateway address. DHCP client allows the VoIP Gateway to discover an IP address automatically. You may need to refer to your current network environment to configure the VoIP Gateway properly.

The PPPoE setting is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please enter the Username and the Password correctly.

LAN

You can configure the LAN settings/DHCP server in this page.

LAN Settings

You can configure the LAN settings in this page.

LAN Setting	
IP:	<input type="text" value="192.168.22.1"/>
Mask:	<input type="text" value="255.255.255.0"/>
MAC:	<input type="text" value="000296127412"/>
DHCP Server	
DHCP Server:	<input checked="" type="radio"/> On <input type="radio"/> Off
Start IP:	<input type="text" value="150"/>
End IP:	<input type="text" value="200"/>
Lease Time:	<input type="text" value="1"/> : <input type="text" value="0"/> (dd:hh)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

Options	Description
IP	The unique address of the VoIP Gateway on your local network.
Mask	The subnet mask of your local network.
MAC	The Ethernet MAC address of the WAN interface.
DHCP Server	Enable or disable DHCP server.
Start IP	The start of the IP address given by the DHCP server when the DHCP server is enabled.
End IP	The end of the IP address given by the DHCP server when the DHCP server is enabled.
Lease Time	The lease time for the DHCP address.

VoIP

The VoIP Gateway Configuration page sets parameters for all VoIP application.

VoIP Configuration

You can configure the VoIP settings, please click the hyperlink.

SIP Settings

[SIP Service Provider](#)

[Port Settings](#)

[Codec Settings](#)

[Codec ID Settings](#)

[DTMF Settings](#)

[RPort Settings](#)

[QoS Settings](#)

Phone Book

[Phone Book](#)

Phone Settings

[Call Forward](#)

[Volume Settings](#)

[DND Settings](#)

[Caller ID Settings](#)

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Others

[Auto Configuration Settings](#)

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[Advanced Settings](#)

The VoIP Gateway Configuration page is divided into four general categories: SIP Setting, Phone Book, Phone Setting, and Others.

SIP Setting

In the SIP settings sub menu you can setup the SIP Service Provider, Port Settings, Codec Settings, Codec ID Settings, DTMF Settings, RPort Setting and QoS.

- [SIP Setting](#)
- [SIP Service Provider](#)
- [Port Setting](#)
- [Codec Settings](#)
- [Codec ID Settings](#)
- [DTMF Settings](#)
- [RPort Settings](#)
- [QoS](#)

SIP Service Provider

In the SIP Service Provider page you will need to enter the account and the related information for your SIP account. Please refer to your VSP (Voice Service Provider provider for more information. You can register two SIP accounts in the V220.

SIP Service Provider

You can set information of service domains in this page.

SIP Service Provider	
Active:	<input checked="" type="radio"/> On <input type="radio"/> Off
SIP Proxy Domain:	<input type="text"/>
Proxy Server:	<input type="text"/>
Outbound Proxy:	<input type="text"/>
Display Name:	<input type="text"/>
User Name:	<input type="text"/>
Auth. ID:	<input type="text"/>
Auth. Password:	<input type="text"/>
Subscribe for MWI:	<input type="radio"/> On <input checked="" type="radio"/> Off
Status:	Not Registered
SIP Expire Time:	<input type="text" value="60"/> (15~86400 sec)
Use DNS SRV:	<input type="radio"/> On <input checked="" type="radio"/> Off
<input type="button" value="Submit"/> <input type="button" value="Reset"/> <input type="button" value="Back"/>	

Options	Description
Active	If set to on, the VoIP gateway will attempt to register with your VoIP service provider using this account.
SIP Proxy Domain	The SIP Proxy Domain is specified by your VSP.
Proxy Server	The host name of IP address of the SIP server. The proxy server is responsible for forwarding requests received from the VoIP Gateway to your VoIP service provider. The Proxy Server is specified by your VoIP service provider.
Outbound Proxy	The host name of IP address of the Outbound Proxy server. The Outbound proxy server is responsible for forwarding requests received from the VoIP Gateway to your VoIP service provider. The Proxy Server is specified by your VoIP service provider.
Display Name	The name or number the called party sees when you call them.

User Name	Your VoIP number. This number is assigned by your VoIP service provider.
Auth Name	The authentication username assigned by your VoIP service provider. This can be different to your VoIP number.
Auth Password	The authentication password provided by your VoIP service provider.
Subscribe for MWI	When enabled the Subscribe for Message Waiting Indication will be sent periodically.
Status	Indicates if the VoIP Gateway is registered to your VSP. If “Registered” is shown, you can make and receive calls via your VoIP service provider. If the VoIP gateway reports “not registered”, check the SIP Service Provider settings above and try again.
SIP Expiry Time	This setting specifies a recommendation to the SIP server on the re-registration interval. Increasing this time will send less registration traffic, but can cause other issues.
Use DNS SRV	When DNS SRV is switched on, the VoIP Gateway will look up DNS SRV records and register with the server with the highest priority. DNS SRV can provide redundancy if the primary SIP server becomes unreachable. Please check with your VSP to find out if they support DNS SRV.

Port Setting

You can configure the SIP and RTP port number in this page. Please leave as default unless otherwise specified by your VSP.

Port Settings

You can set the port number in this page.

SIP Port:	<input type="text" value="5060"/>	(10~65533)
RTP Port:	<input type="text" value="60000"/>	(10~65533)

Options	Description
SIP Port	The port on the SIP server that receives requests from the ATA. Default: 5060.
RTP Port	The base port to receive RTP packets. Default: 60000.

Codec Settings

You can set the Codec Priority, RTP Packet Length, and VAD (Voice Activity Detection) settings in this page. Please leave as default unless otherwise specified by your VSP.

Codec Settings

You can set the codec settings in this page.

Codec Priority	
Codec Priority 1:	G.729
Codec Priority 2:	G.711 a-law
Codec Priority 3:	G.711 u-law
Codec Priority 4:	G.723
Codec Priority 5:	G.726 - 16
Codec Priority 6:	G.726 - 24
Codec Priority 7:	G.726 - 32
Codec Priority 8:	G.726 - 40
Codec Priority 9:	GSM

RTP Packet Length	
G.711 & G.729:	20 ms
G.723:	30 ms

G.723 5.3K	
G.723 5.3K:	<input type="radio"/> On <input checked="" type="radio"/> Off

Voice VAD	
Voice VAD:	<input type="radio"/> On <input checked="" type="radio"/> Off

Codec ID Setting

You can set the value of Codec ID in this page. Please leave as default unless otherwise specified by your VoIP Service Provider.

Codec ID Settings

You can set the value of Codec ID in this page.

Codec Type	ID	Default Value
G726-16 ID:	23 (95~255)	<input checked="" type="checkbox"/> 23
G726-24 ID:	22 (95~255)	<input checked="" type="checkbox"/> 22
G726-32 ID:	2 (95~255)	<input checked="" type="checkbox"/> 2
G726-40 ID:	21 (95~255)	<input checked="" type="checkbox"/> 21
RFC 2833 ID:	101 (95~255)	<input checked="" type="checkbox"/> 101

DTMF Setting

You can set the Out of Band DTMF and Send DTMF SIP Info Enable/Disable in this page. To change this setting, please follow your VoIP Service Provider's information. When you finished the setting, please click the Submit button.

DTMF Settings

You can set the DTMF setting in this page.

<input checked="" type="radio"/> RFC 2833
<input type="radio"/> Inband DTMF
<input type="radio"/> Send DTMF SIP Info

- **RFC 2833:** Click this button to send mid-call DTMF tones in RTP packets separately using RFC2833, i.e., dynamic negotiation of RTP payload for DTMF digits will be done.
- **Inband DTMF (IN AUDIO):** Click this button to send mid-call DTMF tones in RTP packets with the same payload as voice, i.e., dynamic payload negotiation for DTMF digits will not be done.
- **Send DTMF SIP Info:** This field is configurable when RFC 2833 is selected as the DTMF relay mechanism. Specify the payload number that needs to be used for DTMF information negotiated in SDP during SIP signaling.

RPort Function

You can enable or disable the RPort feature in this page. To change this setting, please follow the recommendation of your VSP.

RPort Settings

You can enable/disable the RPort setting in this page.

RPort: On Off

QoS

You can configure the Hold by RFC and Voice/SIP QoS in this page.

The QoS setting is to set the voice packets priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still needs to cooperate with the others Internet devices.

QoS Settings

You can set other settings in this page.

Hold by RFC:	<input type="radio"/> On <input checked="" type="radio"/> Off
Voice QoS (Diff-Serv):	<input type="text" value="40"/> (0-63)
SIP QoS (Diff-Serv):	<input type="text" value="40"/> (0-63)

Phone Book Configuration

The Phone Book contains Speed Dial Settings where you can setup Speed Dial number. If you want to use Speed Dial you just dial the speed dial number then press “#”.

The Phone Book has a maximum of 140 speed dial entries. You can add and delete the entry from this page.

To add a phone number into the Speed Dial list, enter the position, the name, the number (Speed Dial Number), and the phone number (by URL type) and then click the “Add Phone” button.

If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.

If you want to delete all phone numbers, you can click “Delete All” button.

Phone Book

You can add/delete items in current phone book.

Phone Book Page: ▾

Position	Name	Number	URL	Select
0				<input type="checkbox"/>
1				<input type="checkbox"/>
2				<input type="checkbox"/>
3				<input type="checkbox"/>
4				<input type="checkbox"/>
5				<input type="checkbox"/>
6				<input type="checkbox"/>
7				<input type="checkbox"/>
8				<input type="checkbox"/>
9				<input type="checkbox"/>

Add New Phone

Position:
 Name:
 Number:
 URL:

Phone Book Page	
Book Page	Default page is Page1. There are total 14 pages from Page 1 to Page 14
Phone	Show the phone number by sequence. There are total 140 phone numbers from Phone 0 to Phone 139 can be set
Name	Display the Name for the Speed Dial Number
Number	Display the Speed Dial Number that you configured
URL	Display the URL or number that you configured
Select	Select the item.
Delete Selected [Button]	Delete selected item
Delete All [Button]	Delete all items
Reset [Button]	Reset selected item

Add New Phone	
Position	Enter the phone number from 0 to 139
Name	Enter the Speed Dial Name
Number	Enter the Speed Dial Number
URL	Enter the URL, VoIP Phone Number, Remote WAN IP Address of VoIP Gateway
Add Phone [Button]	Add the new Phone which you configured
Reset [Button]	Reset configured items

Examples:

Position	Name	Number	URL	Select
0	test iptel	000	test@iptel.org	<input type="checkbox"/>
1	ip address 10.32	001	192.168.10.32	<input type="checkbox"/>
2	ip address 10.32 5062	002	192.168.10.32:5062	<input type="checkbox"/>
3	john	003	88888888	<input type="checkbox"/>
4	Voip user	004	voipuser	<input type="checkbox"/>
5	home	005	0019998887777	<input type="checkbox"/>

Example 1: Position: 0, Name: “test iptel”, Number: 000, URL: test@iptel.org

Position	Name	Number	URL	Select
0	test iptel	000	test@iptel.org	<input type="checkbox"/>

When the user dials the Number 000, it will dial to the VoIP User test who registers to the SIP Server iptel.org.

Be noted that you need also register to the SIP Server iptel.org. If you register to the different SIP Server, please make sure that the SIP Server allows you dial to iptel.org.

Example 2: Position: 1, Name: “ip address 10.32”, Number: 001, URL: 192.168.10.32

1	ip address 10.32	001	192.168.10.32	<input type="checkbox"/>
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When the user dials the Number 001, it will dial to the VoIP Device whose IP Address is 192.168.10.32.

Example 3: Position: 2, Name: “ip address 10.32 5062”, Number: 002, URL: 192.168.10.132:5062

2	ip address 10.32 5062	002	192.168.10.32:5062	<input type="checkbox"/>
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When the user dials the Number 002, it will dial to the VoIP Device whose IP Address is 192.168.10.132 with the port 5062.

Example 4: Position: 3, Name: “john”, Number: 003, URL: 88888888

3	john	003	88888888	<input type="checkbox"/>
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When the user dials the Number 003, it will dial to the VoIP User whose phone number is 88888888.

Example 5: Position: 4, Name: “Voip user”, Number: 004, URL: voipuser

4	Voip user	004	voipuser	
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When the user dials the Number 004, it will dial to the VoIP User whose phone number is voipuser.

Example 6: Position: 5, Name: “home”, Number: 005, URL: 0019998887777

5	home	005	0019998887777	
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When the user dials the Name [Speed Dial Number] 005, it will dial to the PSTN phone number 000019998887777 by VoIP OUT.

Note: please make sure that if your VoIP Service Provider support the VoIP OUT.

If your VoIP Service Provider supports the VoIP OUT, please follow the instructions of your VoIP Service Provider to dial the PSTN phone number by VoIP OUT.

For example: dialing as per the VoIP Server Provider recommended dialing sequence, 00 + country code + telephone number (e.g. 00 1 999 888 7777).

Example 7: When dial a VoIP Phone Number which isn't configured in the Name [Speed Dial Number] list, it'll dial out the VoIP Phone Number.

Phone Setting

Phone Setting contains Call Forward, Volume Settings, DND Settings, Auto Answer, Caller ID, Dial Plan Settings, Flash Time Settings, Call Waiting Settings, T.38(FAX) Settings and Hot line Settings functions.

Phone Settings
Call Forward
Volume Settings
DND Settings
Auto Answer
Caller ID
Dial Plan Settings
Flash Time Settings
Call Waiting Settings
T.38(FAX) Settings
Hot line Settings

Call Forward function

You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by enabling the respective mode.

Call Forward Settings

You can set the forward number of your phone in this page.

All Forward: Off On
 Busy Forward: Off On
 No Answer Forward: Off On

	Name	URL
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd Time Out:	<input type="text" value="5"/> (2~8 Ring)	
<input type="button" value="Submit"/> <input type="button" value="Reset"/> <input type="button" value="Back"/>		

Option	Description
All Forward	All incoming call will forward to the URL/number you configured.
Busy Forward	If you are on the phone, the new incoming call will forward to the URL/number you configured.
No Answer Forward	If you can not answer the phone after a specific ring you configured, the incoming call will forward to the URL/number you configured.
Off	Disable call forward.
IP/ON	Enable call forward for URL/number.
PSTN (Optional)	Enable call forward for PSTN phone number. Only for V211.
All Fwd No.	The URL/number you configured will be forward to for All Forward

Busy Fwd No.	The URL/number you configured will be forward to for Busy Forward
No Answer Fwd No.	The URL/number you configured will be forward to for No Answer Forward
Name	Display the name of URL/number that you configured
URL	Enter the URL, VoIP Phone Number, Remote WAN IP Address of VoIP Gateway which you want forward to.
No Answer Fwd Time Out	You can set the Time Out time for system to start to forward the call to the number you configured for No Answer Forward

Example 1: All Forward: IP, Name.: 7777, URL/Number: 7777

All Forward: Off IP PSTN
 Busy Forward: Off IP
 No Answer Forward: Off IP PSTN

	Name	URL/Number
All Fwd No.:	7777	7777
Busy Fwd No.:		
No Answer Fwd No.:		

All incoming call will forward to the VoIP phone number 7777.

Example 2: All Forward: IP, Name: 192.168.10.36, URL/Number: 192.168.10.36

All Forward: Off IP PSTN
 Busy Forward: Off IP
 No Answer Forward: Off IP PSTN

	Name	URL/Number
All Fwd No.:	192.168.10.36	192.168.10.36
Busy Fwd No.:		
No Answer Fwd No.:		

All incoming call will forward to the VoIP IP Gateway's WAN IP Address 192.168.10.36.

Example 3: All Forward: PSTN, Name.: 88888888, URL/Number: 88888888

All Forward: Off IP PSTN
 Busy Forward: Off IP
 No Answer Forward: Off IP PSTN

	Name	URL/Number
All Fwd No.:	88888888	88888888
Busy Fwd No.:		
No Answer Fwd No.:		

All incoming call will forward to the PSTN phone number 88888888.

Example 4: All Forward: IP, Name.: 7777, URL/Number: 7777

All Forward: Off IP PSTN
 Busy Forward: Off IP
 No Answer Forward: Off IP PSTN

	Name	URL/Number
All Fwd No.:		
Busy Fwd No.:	7777	7777
No Answer Fwd No.:		

If you are on the phone, the new incoming call will forward to the VoIP phone number 7777.

Example 5: All Forward: IP, Name: 192.168.10.36, URL/Number: 192.168.10.36

All Forward: Off IP PSTN
 Busy Forward: Off IP
 No Answer Forward: Off IP PSTN

	Name	URL/Number
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	192.168.10.36	192.168.10.36
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

If you are on the phone, the new incoming call will forward to the VoIP IP Gateway's WAN IP Address 192.168.10.36.

Example 6: All Forward: IP, Name.: 7777, URL/Number: 7777

All Forward: Off IP PSTN
 Busy Forward: Off IP
 No Answer Forward: Off IP PSTN

	Name	URL/Number
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	7777	7777
No Answer Fwd Time Out:	3 (2~8 Ring)	

If you can not answer the phone after 3 rings, the incoming call will forward to the VoIP phone number 7777.

Example 7: All Forward: IP, Name: 192.168.10.36, URL/Number: 192.168.10.36

All Forward: Off IP PSTN
 Busy Forward: Off IP
 No Answer Forward: Off IP PSTN

	Name	URL/Number
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	192.168.10.36	192.168.10.36
No Answer Fwd Time Out:	3 (2~8 Ring)	

If you can not answer the phone after 3 rings, the incoming call will forward to the VoIP IP Gateway's WAN IP Address 192.168.10.36.

Example 8: All Forward: PSTN, Name.: 88888888, URL/Number: 88888888

All Forward: Off IP PSTN
 Busy Forward: Off IP
 No Answer Forward: Off IP PSTN

	Name	URL/Number
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	88888888	88888888
No Answer Fwd Time Out:	3 (2~8 Ring)	

If you can not answer the phone after 3 rings, the incoming call will forward to the PSTN phone number 88888888.

Volume Setting function

You can setup the Handset Volume, PSTN-Out Volume, Handset Gain and the PSTN-In Gain.

Volume Settings

You can set the volume of your phone in this page.

Handset Volume:	<input type="text" value="10"/>	(0~12)
Handset Gain:	<input type="text" value="10"/>	(0~15)

Options	Description
Handset Volume:	to set the volume for on your handset.
PSTN-Out Volume:	(V211 only) is to set the PSTN volume out.
Handset Gain:	to set the volume sent out to the other party handset.
PSTN-In Gain:	(V211 Only) is to set the volume sent in to your handset from PSTN.

DND Setting function

The do not disturb feature allows the device to automatically return busy during a specified time period. Incoming calls may return busy or go to voice mail dependant on the setup of your VSP account.

DND Settings

You can set the do not disturb period of your phone in this page.

DND Always: On Off

DND Period:	<input type="radio"/> On <input checked="" type="radio"/> Off
From:	<input type="text" value="00"/> : <input type="text" value="00"/> (hh:mm)
To:	<input type="text" value="00"/> : <input type="text" value="00"/> (hh:mm)

Options	Description
DND Always	When enabled, incoming calls will be blocked and the caller will hear the busy tone.
DND Period	When enabled, incoming calls will be blocked and the caller will hear the busy tone when the call is placed during the time period specified below. If the "From" time is larger than the "To" time, the block time will from day 1 to day 2.
From	Enter the start time of the time period. (24 hours format, hh:mm)
To	Enter the end time of the time period. (24 hours format, hh:mm)

Auto Answer function (Only applicable for V211)

You can set the Auto Answer function to make the device answer the incoming call.

If the call arrives from VoIP, then the VoIP Gateway can let the user redial the call to a PSTN phone number. If the call is coming from the PSTN, then the VoIP Gateway can let the user redial to a VoIP Phone number.

Once the VoIP Gateway receives the specified number of rings in the Auto Answer Counter and the caller has authenticated by entering the correct PIN code as configured, the caller can make a call on the other network. For example, you can call in on the VoIP network and make a PSTN call out on the FXO port or you can call in on the PSTN line (FXO port) and make a VoIP call.

In order to make a call, simply enter the desired number or speed dial number followed by the '#' key.

If you call in via PSTN and make a VoIP toll bypass call, you can terminate the call by simply hanging up. If you call in via the VoIP network and make a PSTN toll bypass call, the call is terminated by ending the VoIP call.

Auto Answer

You could enable/disable the auto answer in this page.

Auto Answer:	<input type="radio"/> On <input checked="" type="radio"/> Off
Auto Answer Counter:	<input type="text" value="03"/> (0-6)
PIN Code Enabled:	<input type="radio"/> On <input checked="" type="radio"/> Off
PIN Code:	<input type="text"/>
<input type="button" value="Submit"/> <input type="button" value="Reset"/> <input type="button" value="Back"/>	

Options	Description
Auto Answer	When enabled, the gateway will automatically answer the call.
Auto Answer Counter	Set the number of rings before the gateway answers the call.
PIN Code Enabled	Turn on to make the gateway prompt for a pin code before placing the call.
PIN Code	Enter the pin code required to be entered by the caller to make a call.

Example 1: Auto Answer: On, Auto Answer Counter: 3

Auto Answer

You could enable/disable the auto answer in this page.

Auto Answer:	<input checked="" type="radio"/> On <input type="radio"/> Off
Auto Answer Counter:	<input type="text" value="3"/> (0-8)
PIN Code Enabled:	<input type="radio"/> On <input checked="" type="radio"/> Off
PIN Code:	<input type="text"/>
<input type="button" value="Submit"/> <input type="button" value="Reset"/> <input type="button" value="Back"/>	

How to Use – PSTN to VoIP Call:

1. Call in via PSTN.
2. When you hear the dial tone indicating that the VoIP Gateway is receiving 3 rings and expecting a number, dial the VoIP phone number to which you want to call, then press # (optional) to make a PSTN to VoIP call.

How to Use – VoIP to PSTN Call:

1. Call in via VoIP.
2. When you hear the dial tone indicating that the VoIP Gateway is receiving 3 rings and expecting a number, dial the VoIP phone number to which you want to call, then press # (optional) to make a VoIP to PSTN call.

Example 2: Auto Answer: On, Auto Answer Counter: 3, PIN Code Enabled: On, PIN Code: 1234

Auto Answer

You could enable/disable the auto answer in this page.

Auto Answer:	<input checked="" type="radio"/> On <input type="radio"/> Off
Auto Answer Counter:	<input type="text" value="3"/> (0-8)
PIN Code Enabled:	<input checked="" type="radio"/> On <input type="radio"/> Off
PIN Code:	<input type="text" value="1234"/>
<input type="button" value="Submit"/> <input type="button" value="Reset"/> <input type="button" value="Back"/>	

How to Use – PSTN to VoIP Call:

1. Call in via PSTN.
2. When you hear the continued BEEP BEEP indicating that the VoIP Gateway is asking you to enter the PIN Code:
3. Enter the correct PIN Code 1234, then press #.
4. When you hear the dial tone indicating that the VoIP Gateway is expecting a number, dial the VoIP phone number to which you want to call, then press # (optional) to make a PSTN to VoIP call.

How to Use – VoIP to PSTN Call:

1. Call in via VoIP.
2. When you hear the continued BEEP BEEP indicating that the VoIP Gateway is asking you to enter the PIN Code:
3. Enter the correct PIN Code 1234, then press #.
4. When you hear the dial tone indicating that the VoIP Gateway is expecting a number, dial the PSTN phone number to which you want to call, then press # (optional) to make a VoIP to PSTN call.

Caller ID function

You can set the device to show Caller ID in your PSTN Phone or IP Phone. In Australia, Caller ID is sent using FSK after the 1st ring.

Caller ID Settings

You can enable/disable the caller ID setting in this page.

Caller ID:	Caller ID after 1st Ring (FSK) ▾
Single Caller ID:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CID Without Time:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CID Type 2:	<input type="radio"/> Yes <input checked="" type="radio"/> No

Options	Description
Single Caller ID	Enable to detect Single Caller ID
CID Without Time	Enable to detect the Caller ID without time.
CID Type 2	Enable to detect the Caller ID Type 2.

Dial Plan function

Dial plans can make your dialing easier. If the dialed number meets the prefix, the ATA will modify the dialed number before dialing it. If the dialed number does not meet the prefix, the ATA will dial the number without any modification.

Dial Plan Settings

You can set the dial plan in this page.

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 1:	<input type="text"/> + <input type="text"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 2:	<input type="text"/> + <input type="text"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 3:	<input type="text"/> + <input type="text"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 4:	<input type="text"/> + <input type="text"/>
Dial now:	<input type="text"/>
Auto Dial Time:	5 (3~9 sec)
Use # as send key:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Use * for IP dialing:	<input checked="" type="radio"/> Yes <input type="radio"/> No

Options	Description
Drop Prefix	If set to yes, the ATA will remove the prefix from the number before passing it onto the VSP.
Replace rule1 Replace rule2 Replace rule3 Replace rule4	There are total 4 replace rules for use. +: or xxx: Define the length of digits.
Auto Dial Time	Determines the time the ATA will wait before making the call after the last digit has been dialed.

Symbol explain:

x or X	0,1,2,3,4,5,6,7,8,9 (Wildcard)
+	Or

Example 1: Drop prefix: No, Replace rule 1: 002, 8613+8662

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 1:	002 + 8613+8662

When the number 8613 has been dialed, the prefix 002 will be added and the real phone number [002+8613+xxx] will be dialed out.

For example, when you dial the number 86315555 and the prefix 002 will be added and the real phone number 00286135555 will be dialed out.

When the number 8662 has been dialed, the prefix 002 will be added and the real phone number [002+8662+xxx] will be dialed out.

For example, when you dial the number 86625555 and the prefix 002 will be added and the real phone number 00286625555 will be dialed out.

Example 2: Drop prefix: Yes, Replace rule 2: 006, 002+003+004+005+007+009

Drop prefix :	<input checked="" type="radio"/> Yes <input type="radio"/> No
Replace rule 2:	006 + 002+003+004+005+007+009

When the number 002 has been dialed, the digits 002 will be replaced to 006 and the whole digits [006+xxx] will be dialed out.

For example, when you dial the number 0025555 and the digits 002 will be replaced to 006 and then the real phone number 0065555 will be dialed out.

When the number 003 has been dialed, the digits 003 will be replaced to 006 and the real phone number [006+xxx] will be dialed out.

For example, when you dial the number 0035555 and the digits 003 will be replaced to 006 and then the real phone number 0065555 will be dialed out.

When the number 004 has been dialed, the digits 004 will be replaced to 006 and the real phone number [006+xxx] will be dialed out.

For example, when you dial the number 0045555 and the digits 004 will be replaced to 006 and then the real phone number 0065555 will be dialed out.

When the number 005 has been dialed, the digits 005 will be replaced to 006 and the real phone number [006+xxx] will be dialed out.

For example, when you dial the number 0055555 and the digits 005 will be replaced to 006 and then real phone number digits 0065555 will be dialed out.

When the number 007 has been dialed, the digits 007 will be replaced to 006 and the real phone number [006+xxx] will be dialed out.

For example, when you dial the number 0075555 and the digits 007 will be replaced to 006 and then the real phone number 0065555 will be dialed out.

When the number 009 has been dialed, the digits 009 will be replaced to 006 and the real phone number [006+xxx] will be dialed out.

For example, when you dial the number 0095555 and the digits 009 will be replaced to 006 and then the real phone number 0065555 will be dialed out.

Example 3: Drop prefix: No, Replace rule 3: 009, 12

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 3:	009 + 12

When the number 12 has been dialed, the prefix 009 will be added and the whole digits [009+12+xxx] will be dialed out.

For example, when you dial the number 125555 and the prefix 009 will be added and the real phone number 009125555 will be dialed out.

Example 4: Drop prefix: No, Replace rule 4: 007, 5xxx+35xx+21xx

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 4:	007 + 5xxx+35xx+21xx

When the number 5xxx has been dialed, the prefix 007 will be added and the whole digits [007+5xxx] will be dialed out. Be note that the range of xxx is from 000 to 999.

For example, when you dial the number 5000 and the prefix 007 will be added and the real phone number 0075000 will be dialed out.

For example, when you dial the number 5999 and the prefix 007 will be added and the real phone number 0075999 will be dialed out.

When the number 35xx has been dialed, the prefix 007 will be added and the whole digits [007+35xx] will be dialed out. Be note that the range of xx is from 00 to 99.

For example, when you dial the number 3500 and the prefix 007 will be added and the real phone number 0073500 will be dialed out.

For example, when you dial the number 3599 and the prefix 007 will be added and the real phone number 0073599 will be dialed out.

When the number 21xx has been dialed, the prefix 007 will be added and the whole digits [007+21xx] will be dialed out. Be note that the range of xx is from 00 to 99.

For example, when you dial the number 2100 and the prefix 007 will be added and the real phone number 0072100 will be dialed out.

For example, when you dial the number 2199 and the prefix 007 will be added and the real phone number 0072199 will be dialed out.

When the number 534 have been dialed, the prefix 007 will not be added and the real phone number 534 will be dialed out due to 534 (3 digits) is not in the rule 5xxx (4 digits).

When the number 358822 have been dialed, the prefix 007 will not be added and the real phone number 358822 will be dialed out due to 358822 (6 digits) is not in the rule 35xx (4 digits).

Auto Dial Time function

Auto Dial Time: (3~9 sec)

This value determines the time the ATA waits after the last digit is entered, before placing the call. Alternatively, you can press # to immediately dial the number.

Flash Time Settings function

When you use the PSTN phone and you need to press the Hook to simulate Flash key (Switch to the other phone line or HOLD), this function is to set the amount of time you need to press the Hook to represent the Flash function.

Flash Time Settings

You can set the flash time in this page.

Max Flash Time: x 10 ms (4~255)

Call Waiting Setting

You can enable/disable the call waiting setting in this page.

When a new incoming call arrives while you are on an existing call, you can push the Flash button to switch to the new call. You can then use the Flash button to switch between the two calls.

To end the first call, hang up the phone. The phone will now ring, pick it up to talk to the second caller. To end this call, simply hang up.

Call Waiting Settings

You can enable/disable the call waiting setting in this page.

Call Waiting: On Off

T.38 (FAX) Setting

You can enable/disable the T.38 FAX function in this page.

T.38 (FAX) Settings

You can enable/disable the FAX function in this page.

T.38 (FAX): On Off

T.38 Port: (1024~65533)

Options	Description
T.38 (FAX)	Switch on to enable T.38 fax support.
T.38 Port	Default is 60000.

T.38 support

Fax Pass-through

In fax pass-through mode, UDPTL packets are not used. Fax communication between the two fax machines is carried out entirely in-band over a voice call (over RTP). The ATA is aware that the call in progress is a fax call and not voice call. If during a voice call, the CED/CNG fax tones are recognized, then the VoIP Gateway will change the voice codec to G.711, and if necessary, turn off echo cancellation (EC) and voice activity detection (VAD) and adjust the jitter and reorder buffers to fix the network delay for the duration of the call.

T.38 support mode

T.38 provides an ITU-T standards-based method and protocol for fax. In this mode, the ATA will establish a normal voice call and switch to T.38 fax based on the detection of fax tones. This will cause the ATA to renegotiate the session parameters with new T.38 parameters. The rest of the fax signaling and data is then encapsulated and sent in IFP packets. The ITU-T T.38 defines the behavior for both Internet Aware Fax Devices (IAF, network aware fax machine) and gateways connected to G3FE (Group 3 Fax equipment). This ATA supports both kinds of behaviors.

Hot line Settings

The hot line function causes the ATA to dial the hot line number specified immediately after the phone is picked up.

Hot line Settings

You can set the hot line in this page.

Use Hot Line : Enable Disable

Hot line number:

Options	Description
Use Hot Line	Select enable to enable the Hot Line function.
Hot Line Number	Enter the URL, VoIP Phone Number or Remote WAN IP Address of VoIP device which you want to use for Hot Line.

Example 1: Use Hot Line: Enable, Hot line number: 2468013579

Use Hot Line : Enable Disable

Hot line number: 2468013579

Every time when you pick up the phone, it will dial to the VoIP Phone Number 2468013579 automatically.

Example 2: Use Hot Line: Enable, Hot line number: 192.168.10.63

Use Hot Line : Enable Disable

Hot line number: 192.168.10.63

Every time you pick up the phone, it will establish a call to the SIP device at the IP address 192.168.10.63.

Others

The others menu contains the following sub menus :

Others
Auto Configuration Settings
FXS Impedence Settings
STUN Settings
MAC Clone Settings
Tones Settings
Advanced Settings

Auto Configuration Settings

You can enable/disable the auto configuration/provisioning setting in this page.

The ATA allows for provisioning and remote upgrade. Provisioning is achieved through configuration profiles transferred to the device via TFTP, HTTP or FTP. The ATA can be configured to update its VoIP Configuration from a remote profile on power up or reboot.

Auto Configuration Settings

You can enable/disable the auto configuration setting in this page.

Auto Configuration: Off TFTP FTP HTTP

TFTP Server:	<input type="text"/>	
HTTP Server:	<input type="text"/>	Exp. 60.35.187.30
HTTP File Path:	<input type="text"/>	Exp. /download/
FTP Server:	<input type="text"/>	Exp. 60.35.17.1
FTP Username:	<input type="text"/>	
FTP Password:	<input type="text"/>	
FTP File Path:	<input type="text"/>	Exp. /file/load

Options	Description
Auto Configuration	Enable Auto Configuration. (Default:off)
TFTP Server	Enter IP or Host Name of TFTP Server.
HTTP Server	Enter IP or Host Name of HTTP Server.
HTTP Path	Enter the path where the provisioning file is located.
FTP Server	Enter IP or Host Name of FTP Server.
FTP Username	Enter Username which provided by FTP Server.
FTP Password	Enter Password which provided by FTP Server.
File Path	Enter the path where the provisioning file is located.

Example 1: Auto Configuration for HTTP Server

Auto Configuration: HTTP, HTTP Server: 192.168.10.100, HTTP Path: /

Auto Configuration: Off TFTP FTP HTTP

TFTP Server:	<input type="text"/>
HTTP Server:	192.168.10.100
HTTP Path:	/
FTP Server:	<input type="text"/>
FTP Username:	<input type="text"/>
FTP Password:	<input type="text"/>
File Path:	<input type="text"/>

Every time when you power on the ATA it will update its VoIP configuration to the latest one from Auto Provisioning Server (HTTP Server) automatically.

Example 2: Auto Configuration for TFTP Server

Auto Configuration: TFTP, TFTP Server: 192.168.10.100

Auto Configuration: Off TFTP FTP HTTP

TFTP Server:	192.168.10.100
HTTP Server:	<input type="text"/>
HTTP Path:	<input type="text"/>
FTP Server:	<input type="text"/>
FTP Username:	<input type="text"/>
FTP Password:	<input type="text"/>
File Path:	<input type="text"/>

Every time when you power on the ATA, it will update its VoIP configuration to the latest one from Auto Provisioning Server (TFTP Server) automatically.

Example 3: Auto Configuration for FTP Server

Auto Configuration: FTP, FTP Server: 192.168.10.100, FTP Username: 1234, FTP Password: 1234, FTP Path: /

Auto Configuration: Off TFTP FTP HTTP

TFTP Server:	<input type="text"/>
HTTP Server:	<input type="text"/>
HTTP Path:	<input type="text"/>
FTP Server:	192.168.10.100
FTP Username:	1234
FTP Password:	••••
File Path:	/

Every time when you power on the VoIP Gateway, it will update its VoIP configuration to the latest one from Auto Provisioning Server (FTP Server) automatically.

FXO & FXS Impedance Setting

You can select the FXO & FXS Impedance Setting for a different country. Please leave as default unless otherwise instructed.

FXS Impedance Settings

You can select the FXS impedance of the analog telephone by different country in this page.

Options	Description
FXO Port	Select Impedance for the FXO port. (only on V211)
FXS Port	Select Impedance for the FXS port.

STUN Setting

You can enable/disable the STUN feature and configure the STUN Server IP address in this page. This function can help your ATA to work properly behind a NAT device. To change these settings please follow your VoIP Service Provider's information.

STUN Settings

You can set the IP of STUN server in this page.

Options	Description
STUN	When enabled the NetComm ATA uses STUN (Simple Transversal of UDP through NAT) if the unit is behind a NAT enabled router and the router has no ALG for SIP, or NONE to disable STUN (VoIP Gateway is not to use STUN for NAT traversal). VoIP Gateway also supports a proprietary implementation of NAT traversal where the Service provider is expected to provide some relay support. If NONE is selected, then based on the responses received, the VoIP Gateway will dynamically determine if the SIP Server supports the proprietary implementation. Note: Even when STUN is enabled, the VoIP Gateway does an automatic detection of the presence of SIP ALG and disables the use of STUN. This is to avoid some media problems arising out of the behavior of some ALGs when STUN is used at the user end.
STUN Server	Enter the IP address or host name of the STUN server.
STUN Port	Enter the port number on which the STUN server listens for requests from the ATA. (default : 3478)

MAC Clone Settings

Some ISPs have a restriction that allows only 1 host locked by its MAC address to talk on the internet. If you change network cards, you have to call them up to change the MAC. The NetComm ATA can clone the computer's MAC that was originally set up in this configuration.

MAC Clone Settings

You can enable/disable the MAC clone setting in this page.

MAC Clone: On Off

Options	Description
MAC Clone	Default is Off (disable). When it was On (enable). The VoIP Gateway clones the computer's MAC that was originally set up for such an ISP.

Advanced Settings

Advanced Settings

You can change advanced setting in this page.

ICMP Not Echo:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Send Anonymous CID:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Management from WAN:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Billing Signal:	Disabled
CPC Delay:	2 (2-5 Seconds)
CPC Duration:	0 x 10 ms (0-120)
Send Flash event:	Disabled
SIP Encrypt:	Disabled
PPPoE retry period:	5 Seconds
System Log Server:	
System Log Type:	None

Options	Description
ICMP Not Echo	When switched to yes, the ATA will not echo the ICMP request.
Send Anonymous CID	Send Anonymous as the Caller ID to display when you make a VoIP call.
Management from WAN	Switch to yes to enable remote management.
Billing Signal	To enable billing signal. Leave as default unless instructed by your VoIP Service Provider.
CPC Delay	Default is 2. The VoIP Gateway will send the CPC after the delay time which you configure.
CPC Duration	When VoIP Gateway is the called party, CPC duration is the "voltage drop" duration, before it plays dial tone again.
Send Flash event	Default is Disable. There are two types of Flash event: DTMF Event and SIP Info.
SIP Encrypt	Default is Disable. There are four types of SIP Encrypt: INFINET, AVS, WALKERSUN1 and WALKERSUN2.
PPPoE retry period	Time before the PPPoE client retries the connection.
System Log Server	IP address of the System log server
System Log Type	Type of system log generated.

DDNS

Dynamic DNS (DDNS) allows the ATA to register your IP address with a DDNS provider each time the IP address changes. This allows you to discover the ATA from a remote location when the ATA is used with a dynamically assign IP address service.

DDNS Settings

You can set the configuration of DDNS in this page.

DDNS: On Off

Host Name:	<input type="text" value="netcommleest.dyndns.org"/>
User Name:	<input type="text" value="netcommleest"/>
Password:	<input type="password" value="XXXXXXXXXXXX"/>
E-mail Address:	<input type="text"/>

DDNS Server:	<input type="text"/>
DDNS Server List:	<input type="text" value="members.dyndns.org"/> ▼
Type:	<input type="text" value="dyndns"/> ▼
Wild Card:	<input type="text" value="on"/> ▼

BACKMX:	<input type="radio"/> On <input checked="" type="radio"/> Off
Off Line:	<input type="radio"/> On <input checked="" type="radio"/> Off

Example:

1. Configure the WAN connection to PPPoE Client and make sure the ATA has the WAN IP Address (Public IP Address).

WAN Settings

You can configure the WAN settings in this page.

LAN Mode: Bridge NAT

WAN Setting	
IP Type:	<input type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input checked="" type="radio"/> PPPoE
IP:	<input type="text" value="202.128.118.246"/>
Mask:	<input type="text" value="255.255.255.0"/>
Gateway:	<input type="text" value="202.128.113.243"/>
DNS Server1:	<input type="text" value="0.0.0.0"/>
DNS Server2:	<input type="text" value="0.0.0.0"/>
MAC:	<input type="text" value="000296127411"/>
Host Name:	<input type="text"/>

PPPoE Setting	
User Name:	<input type="text" value="netcomm@netcomm.ne"/>
Password:	<input type="password" value="XXXXXXXX"/>
Service Name:	<input type="text"/>

2. Enable DDNS and configure the DDNS settings including Host Name, User Name and Password

DDNS Settings

You can set the configuration of DDNS in this page.

DDNS: On Off

Host Name:	<input type="text" value="netcommtest.dyndns.org"/>
User Name:	<input type="text" value="netcommtest"/>
Password:	<input type="password" value="XXXXXXXXXX"/>
E-mail Address:	<input type="text"/>

DDNS Server:	<input type="text"/>
DDNS Server List:	<input type="text" value="members.dyndns.org"/>
Type:	<input type="text" value="dyndns"/>
Wild Card:	<input type="text" value="on"/>

BACKMX:	<input type="radio"/> On <input checked="" type="radio"/> Off
Off Line:	<input type="radio"/> On <input checked="" type="radio"/> Off

3. If configured correctly, you should be able to visit the home page of the ATA by entering the DDNS Host Name as follow.



VLAN Settings

You can set the VLAN settings in this page.

VLAN Settings

You can set the VLAN settings in this page.

VLAN Packets:	<input type="radio"/> On	<input checked="" type="radio"/> Off
VID (802.1Q/TAG):	<input type="text" value="136"/>	(2 ~ 4094)
User Priority (802.1P):	<input type="text" value="0"/>	(0 ~ 7)
CFI:	<input type="text" value="1"/>	(0 ~ 1)

Options	Description
VLAN Packets	When enabled the ATA will receive VLAN Packets function.
VID (802.1Q/TAG)	Configure the Virtual LAN ID (VLAN ID or VID) for VLAN Server. The VLAN Identifier is a 12-bit field. It uniquely identifies the VLAN to which the frame belongs. The field can have a value between 2 and 4094. (Default 136).
User Priority (802.1P)	Default is 0. Configure user priority. Also known as user priority, this 3-bit field refers to the IEEE 802.1p priority. The field indicates the frame priority level which can be used for the prioritization of traffic. The field can represent 8 levels (0 through 7).
CFI	The Canonical Format Indicator is a 1-bit field. If the value of this field is 1, the MAC address is in non-canonical format. If the value is 0, the MAC address is in canonical format.

DMZ

A DMZ Host PC is set up 'between' your (private) LAN and the (public) WAN to allow access from the outside world to a specified and isolated zone on your network. It is most commonly used to provide access to a Web server or Game server without exposing the rest of your computers to the Internet. Turn on the DMZ and enter the IP address of the DMZ computer and click Submit. The computer with that IP address can then serve web pages or games to the outside world, while the rest of your network remains private.

DMZ Settings

You can configure your demilitarized zone setting in this page.

DMZ: On Off

DMZ Host IP:

Virtual Server

Virtual Servers are used for port forwarding from the WAN to LAN networks. If any local PCs need to be mapped to the UDP/TCP port on WAN side, you can enter the mappings here. There can be up to 24 different Virtual Server configurations.

Virtual Server Settings

You can set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP (Control) [TCP 21], FTP (Data) [TCP 20], E-mail (POP3) [TCP 110], E-mail (SMTP) [TCP 25], DNS [UDP 53] and Telnet [TCP 23].

Virtual Server Page:

Num	Enable	Protocol	In Port	Ex Port	Server IP	Select
0	<input type="checkbox"/>					<input type="checkbox"/>
1	<input type="checkbox"/>					<input type="checkbox"/>
2	<input type="checkbox"/>					<input type="checkbox"/>
3	<input type="checkbox"/>					<input type="checkbox"/>
4	<input type="checkbox"/>					<input type="checkbox"/>
5	<input type="checkbox"/>					<input type="checkbox"/>
6	<input type="checkbox"/>					<input type="checkbox"/>
7	<input type="checkbox"/>					<input type="checkbox"/>

Add Virtual Server

Num: (0-23)
 Server IP:
 Protocol:
 Internal Port: External Port:

Options	Description
Virtual Server Page	Default page is Page1. There are 3 pages, Page 1 to Page 3
Num	Show the number by sequence. There are a total of 24 numbers from mapping 0 to mapping 23 that can be set.
Enable	Check to enable the port forwarding protocol.
Protocol	TCP or UDP.
In Port (Internal Port)	Internal port number on the LAN side.
Ex Port (External Port)	External port number on the WAN side.
Server IP	Display the private network IP address for the particular server.
Select	Select the item of the Virtual Server
Enable Selected [Button]	Enable selected item
Delete Selected [Button]	Delete selected item
Delete All [Button]	Delete all items
Reset [Button]	Reset selected item

Add Virtual Server	
Num	Enter the number corresponding to the Virtual Server configuration.
Server IP	Enter the private network IP address for the particular server.
Protocol	Select TCP or UDP.
Internal Port	Enter the port number of the Private Network (LAN or internal network). In most cases, the private port number is same as public port number. This port number cannot be seen from the WAN side.
External Port	Enter the port number of the Public Network (WAN or external network).
Add Server [Button]	Add the new Server which you configured
Reset [Button]	Reset configured items

Example 1 (FTP Server):

Num: 0, Server IP: 10.0.0.150, Protocol: TCP, Internal Port: 21, External Port: 21

Add Virtual Server

Num: (0-23)

Server IP:

Protocol:

Internal Port: External Port:

Virtual Server Page:

Num	Enable	Protocol	In Port	Ex Port	Server IP	Select
0	<input checked="" type="checkbox"/>	TCP	21	21	10.0.0.150	<input type="checkbox"/>

Other people can visit your FTP Server by entering the WAN IP Address of the ATA and then the NetComm ATA will re-directly it to your LAN IP 10.0.0.150.

Well Known TCP/UDP Ports			
Port	Protocol	UDP	TCP
20	File Transfer Protocol (FTP) Data		X
21	FTP Commands		X
23	Telnet		X
25	SMTP		X
43	Whois		X
53	Domain Name System (DNS)	X	X
69	Trivial File Transfer Protocol (TFTP)	X	
70	Gopher		X
79	Finger		X
80	HTTP		X
110	POP3		X
111	SUN Remote Procedure Call (RPC)	X	
115	SFTP		X
119	Network News Transfer Protocol (NNTP)		X
123	Network Time Protocol (NTP)		X
144	News	X	X
161	Simple Network Management Protocol (SNMP)	X	
162	SNMP traps	X	
179	Border Gateway Protocol (BGP)		X
443	Secure HTTP (HTTPS)		X
513	rlogin		X
514	rexec		X
517	talk	X	X
518	ntalk	X	X
520	Routing Information Protocol (RIP)	X	
1701	Layer 2 Tunneling Protocol (L2TP)	X	
2000	Open Windows	X	X
2049	Network File System (NFS)		X
6000	X11	X	X

SNTP Settings

Simple Network Time Protocol (SNTP) can be used to sync the time on the NetComm ATA. If SNTP is enabled, your ATA clock is synchronized with an Internet time server once a day. However, if you don't have a continuous Internet connection through a cable modem or DSL modem, the automatic synchronization might not always occur.

SNTP Settings

You can set the SNTP servers in this page.

SNTP: On Off

Primary Server:	<input type="text" value="time.windows.com"/>
Secondary Server:	<input type="text" value="208.184.49.9"/>
Time Zone:	GMT <input type="button" value="+"/> <input type="button" value="-"/> 10 <input type="button" value=":"/> 00 (hh:mm)
Sync. Time:	<input type="text" value="1"/> <input type="text" value="0"/> <input type="text" value="0"/> (dd:hh:mm)

Options	Description
Primary Server	Enter the primary NTP server name or IP address.
Secondary Server	Enter the secondary NTP server name or IP address. The secondary server will be used if the primary server becomes unreachable.
Time Zone	Enter the offset from GMT for your time zone.
Sync	Reports the synchronized time.

If time synchronization fails, it might be for one of the following reasons:

- You are not connected to the Internet. Establish an Internet connection before you attempt to synchronize your clock.
- Your personal or network firewall prevents clock synchronization. Most corporate and organizational firewalls will block time synchronization
- The time shown on your ATA is too different from the current time on the Internet time server. Internet time servers might not synchronize your clock if your ATA's time is off by more than 15 hours.

Alarm Settings

Provide the alarm function. The alarm will sound when it reached the Alarm Time that you configured.

Alarm Settings

You can set the alarm time in this page.

Alarm: ON OFF

Alarm Time: (hh:mm)

Current time: 2007-07-25 14:37

Options	Description
Alarm	Default is OFF (Disable). When it was ON (Enable), It will enable the Alarm function.
Alarm Time	Default is 0:0 (hh:mm). Set the Alarm Time. (24 hours format, hh:mm)
Current time	The current time of the NetComm ATA.

Example 1: Alarm: ON, Alarm Time: 8:1(hh:mm)

Alarm: ON OFF

Alarm Time: (hh:mm)

Current time: 2005-01-01 08:00

The alarm will sound when it reached the current time 08:01.

System Authority

In the System Authority page you can change the NetComm ATA's login name and password.

System Authority

You can change the login username/password in this page.

New username:	<input type="text"/>
New password:	<input type="password"/>
Confirmed password:	<input type="password"/>

Save Settings/Reboot

If modifications are made to the Netcomm ATA's settings, they will be lost if they are not saved to the ATA non-volatile memory. Click Save & Reboot button to make any configuration changes permanent. After you click the Save & Reboot button, the ATA will automatically restart and the new setting will take effect.

If you only want to reboot the NetComm ATA, Click the Reboot Only button. After you click the Reboot Only button, the ATA will automatically restart.

Save Settings / Reboot

You have to save settings & reboot to effect them.

Save Settings and reboot:

You could press the reboot button to restart the system.

Reboot system without saving settings:

System

Reset factory default Page

You can restore the NetComm ATA settings to factory default by clicking the restore button. The ATA will restore to default and automatically restart.

Reset to Factory Default

You can click the restore button to restore the factory settings.

Restore default settings:

Firmware Update Page

To update the NetComm ATA firmware, click browse to locate the new firmware or enter the location and name of the firmware in the File Location field. Click Update, to upgrade the firmware to the new version.

Firmware Update

You can update the newest firmware in this page.

File Location:

Auto Update Page

The NetComm ATA can check and update the firmware automatically either when powering up or via a schedule.

Auto Update Settings

You can set auto update settings in this page.

Update via: Off TFTP FTP HTTP

TFTP Server:	<input type="text"/>	
HTTP Server:	<input type="text"/>	Exp. 60.35.187.30
HTTP File Path:	<input type="text"/>	Exp. /download/
FTP Server:	<input type="text"/>	Exp. 60.35.17.1
FTP Username:	<input type="text"/>	
FTP Password:	<input type="text"/>	
FTP File Path:	<input type="text"/>	Exp. /file/load
Check new firmware:	<input type="radio"/> Power ON <input checked="" type="radio"/> Scheduling	
Scheduling (Date):	14	(1~30 days)
Scheduling (Time):	AM 00:00- 05:59 ▼	
Automatic Update:	<input checked="" type="radio"/> Notify only <input type="radio"/> Automatic (Scheduling)	
Firmware File Prefix:	<input type="text" value="TA1S1P"/>	
Next update time:	<input type="text"/>	

Options	Description
Update via	Enable or disables the Auto Update function. To enable, select the type of auto update server – TFTP, FTP or HTTP.
TFTP Server	The address of the TFTP.
HTTP Server	The address of the HTTP server.
HTTP File Path	The path where the file is located.
FTP Server	The address of the FTP server.
FTP Username	The username for the FTP server.
FTP Password	The password for the FTP server.
FTP File Path	The path where the file is located.
Check new firmware	Determines when the NetComm ATA will check for new firmware. When power ON is selected the ATA will check for updates when powering up. If scheduling is selected, the VoIP gateway will check for updates using the schedule set below.
Scheduling (Date)	The VoIP Gateway will check if there is new firmware on the TFTP/FTP/HTTP Server periodically. The range of the Scheduling Date is 1 - 30. Default is 14.
Scheduling (Time)	Default is AM 00:00- 05:59. The VoIP Gateway will check if there is new firmware on the TFTP/FTP/HTTP Server periodically. There are four Scheduling Time: AM 00:00- 05:59, AM 06:00- 11:59, PM 12:00- 17:59, PM 18:00- 23:59

Automatic Update	Notify only: When newer firmware is available, the VoIP Gateway will only notify you by a "BEEP BEEP BEEP" you when you pick up the phone. It will not automatically upgrade. Automatic (Scheduling): When there is a newer firmware, it will update the firmware automatically.
Firmware File Prefix	The file prefix of the firmware
Next update time	Displays the next update or check time.

Notify Only Auto Update Procedure:

1. Power on the VoIP Gateway and it will check if there is any newer firmware available on the Server. When there is a newer firmware, it will only notify you by "BEEP BEEP BEEP" after you pick up the phone.
2. Please press #190# and then hang up the phone to unlock the special key on keypad.
3. Please press #160# and then hang up the phone to ask NetComm ATA to update the firmware immediately.
4. It takes around 7 minutes for updating the new firmware.

AUTO UPDATE PROCEDURES:

1. Power on the NetComm ATA and reaches Next update time. It will check if there is any update firmware is newer one on the Server. When there is a newer firmware, it'll update the firmware automatically. It will take around 7 minutes to update the firmware.

Be noted:

1. If the NetComm ATA is powered off and has passed the Next update time, it will not update the firmware after you power on the NetComm ATA. It'll only update when the NetComm ATA is power on and reaches Next update time.
2. If you are on the phone and have a conversation to others by VoIP and the Next update time is passing, it will update the firmware immediately after you hang up the phone.

IVR Interface for VoIP Gateway

You can use the PSTN phone to configure the NetComm ATA. Please follow the instruction below to configure it.

Group	IVR Action	IVR Menu Choice	Parameter(s)	Notes:
Function	Reboot	#195#	None	After you hear "Option Successful," hang-up. The system will reboot automatically.
Function	Factory Reset	#198#	None	System will automatically Reboot. WARNING: ALL "User-Changeable" NON DEFAULT SETTINGS WILL BE LOST! This will include network and service provider data.
Info	Check IP Address	#120#	None	IVR will report the LAN port IP address
Info	Check IP Type	#121#	None	IVR will report the WAN Port DHCP is enabled or disabled.
Info	Check the Phone Number	#122#	None	IVR will report the VoIP number in use.
Info	Check Network Mask	#123#	None	IVR will report the WAN Port network mask
Info	Check Primary DNS Server Setting	#125#	None	IVR will announce the current setting in the Primary DNS field.
Info	Check IP Address	#126#	None	IVR will report the WAN port IP address
Info	Check Firmware Version	#128#	None	IVR will announce the version of the firmware running on the VoIP Gateway.
Setting	Set DHCP client	#111#	None	The system will change to DHCP client type
Setting	Set Static IP Address	#112xxx*xxx*xxx*xxx#	Enter IP address using numbers on the telephone key pad. Use the *(star) key when entering a decimal point.	DHCP will be disabled and system will change to the Static IP type.
Setting	Set Network Mask	#113xxx*xxx*xxx*xxx#		Must set Static IP first.
Setting	Set Gateway IP	#114xxx*xxx*xxx*xxx#		Must set Static IP first.
Setting	Set Primary DNS Server	#115xxx*xxx*xxx*xxx#		Must set Static IP first.
Setting	Set Codec	#130+[1-8]#	1:G.711 u-Law, 2: G.711 a-Law, 3:G.723.1, 4: G.729a, 5: G.726 16K, 6:G.726 24K, 7: G.726 32K, 8: G.726 40K,	You can set the codec you want to the first priority.
	Set Handset Gain	#131+[00~15]#	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 6
	Set Handset Volume	#132+[00~12]#	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10

How to make a phone call

Dial a VoIP Phone call

When your VoIP Gateway is configured properly, you can make a phone call via VoIP.

If you want to make a phone call, you can dial the phone number and press “#” button to start to dial the phone number.

Dial a PSTN Phone call (V210P/V211)

By default outgoing calls should be placed with the VoIP service. If you want to make a phone call via the PSTN pass-through port, you can press “0*” and dial the phone number.

Note: On loss of power, the V210P/V211 will switch all calls from the FXS port through to the PSTN pass-through.

The VoIP Gateway also provides some functions that list as below:

Call Features

The NetComm ATA also provides some functions that list as below:

Blind Transfer

This feature allows a user (transferor) to transfer an existing call to another telephone number (transfer target) without connecting to the transfer target number.

How to Use:

1. During an existing call, perform a hook flash to put the other party on hold and get a dial tone.
2. When you hear the dial tone, press #510# on your telephone dial-pad.
3. When you hear the dial tone indicating that the VoIP Gateway is expecting a number, dial the phone number to which you want to transfer the other party, then press # (optional) and then hang up the phone.

Attendant Transfer

This feature allows a user to transfer an existing call to another telephone number after first consulting with the dialed party (transfer target) before hanging up.

How to Use:

1. During an existing call, perform a hook flash to put the other party on hold and get a dial tone.
2. When you hear the dial tone, press #511# on your telephone dial-pad.
3. When you hear the dial tone, dial the telephone number to which the existing party is to be transferred, then press # (optional).
4. When the target transfer answers the phone, you may consult with the target transfer, and then hang up your phone to transfer the call to the target transfer.

3-Way Conferencing

How to Use:

1. Dial the first number.
2. During connection to the first party, perform a hook flash to put the first party on hold.
2. When you hear the dial tone, press #512# on your telephone dial-pad.
3. When you hear the recall dial tone, dial another number and talk with the second person.
4. To conference with both callers at the same time, perform a hook flash.
5. To transfer the second call to first call, perform a hook flash after entering into conferencing mode.

Note: If you hang up during conferencing, it will transfer the first call to the second call.

Call Waiting

How to Use:

1. When a new call is coming while you are talking, you can push the Flash button or perform a hook flash to switch to the new call.
 2. You can push the Flash button to switch between the two calls.
- or
1. Dial the first number to make a conversation.
 2. During connection to the first party, push the Flash button or perform a hook flash to put the first party on hold.
 3. When you hear the dial tone, dial another number and talk with the second person.
 4. You can push the Flash button or perform a hook flash to switch between the two calls.

Call Hold

How to Use:

1. When a new call is coming in while you are on an existing call, you can push the Flash button or perform a hook flash to hold the current call for a while, then push the Hold key again to keep talking.
2. You can push the Flash button to switch between the two calls.

COMMON PROBLEM AND SOLUTIONS

No power to the ATA.

- Please ensure that the power cable is connected correctly and
- The power switch is turn on. (Both at the power point and the power switch at the back of the unit).

Note: Only use the power adapter supplied with the NetComm ATA. A different adapter may damage the product.

ETH light is off.

- Please ensure that the ATA is connected to your broadband modem/router using the correct cable and your computer is connected to the ATA.

No Dial Tone and the SIP light is off. Please check the following:

- The WAN port of the unit is configured correctly according to your network structure and
- The SIP is configured correctly according to your VSP (VoIP Service Provider). You have to ensure that all details are entered correctly.

There is no sound coming from the phone handset.

- Please ensure that you phone handset is connected to the correct port (phone port) and the cable is not damage.

When making a call, the call does not go through and always come with busy tone.

- Check that the SIP light is on.
- Try to call a different number.
- If your broadband mode/router is not VoIP aware, you must set up port forwarding to allow the voice traffic to go through the modem/router. There are certain ports that you need to specify, 5060-5061 (UDP) and 60000-60008 (UDP). Please consult your modem/router user guide for more information on how to configure port forwarding.

The ATA never receives call and all incoming calls go directly to voice mail.

- Check that the SIP light is on.
- Try to call from a different number.
- If your broadband mode/router is not VoIP aware, you must set up port forwarding to allow the voice traffic to go through the modem/router. There are certain ports that you need to specify, 5060-5061 (UDP) and 60000-60008 (UDP). Please consult your modem/router user guide for more information on how to configure port forwarding.
- Ensure that your VoIP account supports incoming call from a PSTN number. Depending on your VSP, not all VoIP account supports incoming calls from a PSTN/mobile number.

APPENDIX A: GLOSSARY

This glossary defines acronyms and keywords used in this document.

ATA	Analog Telephone Adaptor
Codec	The format by which audio or video streams are compressed for transmission over networks.
CPC	CPC (Calling Party Control) is a signal sent from most modern electronic COs to indicate that the Calling Party has hung up.
DTMF	Dual-tone multifrequency. DTMF is the system that is used in interactive voice-response menu systems such as the menu system for accessing voicemail messages. The DTMF system allows the user to interact with the menu by pressing keys on the key pad.
FoIP	Fax over Internet Protocol
FXO	Foreign Exchange Office
FXS	Foreign Exchange Subscriber
IP	Internet Protocol. A data-oriented protocol used for communicating data across a network. IP is the most common protocol used on the internet.
IP address	A unique number that devices use in order to identify and communicate with each other on a computer network using the IP standard.
MWI	Message Waiting Indicator. An indicator that there is a voicemail message for the owner of an account.
PSTN	Public Switch Telephone Network. The traditional land-line phone network.
RTP	Real-time Transport Protocol. A protocol for delivering the media portion of a data transmission over an IP network. SRTP is another media protocol.
Signaling	In a VoIP phone call, the information in a call that deals with establishing and controlling the connection, and managing the network. The non-signaling portion of the call is the Media.
SIP	Session Initiation Protocol. The signaling protocol followed by VoIP Gateway for handling phone calls.
SIP account	An account that provides the user the ability to make VoIP phone calls. The account encapsulates the rules and functions the user can access.
SIP address	The address used to connect to a SIP endpoint. In other words, the "phone number" used in a VoIP phone call. For example, sip:test@domainA.com.
STUN	Simple Transversal of UDP through NAT
TCP	Transmission Control Protocol. A transport protocol for delivering data over an IP network. Other transport protocols are TLS and UDP.
TLS	Transport Layer Security. A transport protocol for delivering data over an IP network. TLS is a secure transport protocol, which means that all the data being transmitted (signaling and media) is encrypted. Other transport protocols are TCP and UDP.
UA	User Agent
UDP	User Datagram Protocol. A transport protocol for delivering data over an IP network. Other transport protocols are TCP and TLS.
URI	URI Uniform Resource Identifier. A name or address that identifies a location on the world wide web. A SIP address is a type of URI.
URL	Uniform Resource Locator. A URI that both identifies a name or address and indicates how to locate it.
VoIP	Voice over Internet Protocol. A variation of IP used for sending voice data over the internet, in other words, used for making phone calls over the internet.
VoIP Service Provider	A business that provides a VoIP service, allowing a user to connect to the internet in order to make VoIP phone calls using VoIP Gateway. The VoIP service provider sets up a SIP account for the user.

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, powersurges, 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 authorized 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.



Did you know your phones will disrupt
your ADSL2+ connection...

Even If You Have Older ADSL Filters?

Due to faster ADSL speeds, an inline ADSL2+ Microfilter stops your ADSL connection being disrupted by telephones connected to the same line. **Older filters cannot handle these speeds.** A high-quality Microfilter from NetComm will ensure you have a stable broadband Internet connection with no reduction in the quality of your telephone service.

Having one easy to install Microfilter for every telephone type device, including dial up modems and Foxtel, is essential to reducing voice and data service problems and interruptions.

NetComm EM1550 ADSL2+ Microfilters conform to relevant ACA requirements and Telstra specifications and will:

- Help you obtain faster technical support from your ISP
- Minimise unexplained disruption of your ADSL connection
- Work with all current ADSL connection speeds
- Microfilters are also splitters which allow you to connect the line between phone and modem

Grab your NetComm Microfilters today, either:

Visit a retailer closest to you...

Harvey Norman, Officeworks or Harris Technology.

Order direct...

Visit www.netcomm.com.au/ADSL/EM1550.php

Phone (02) 9424-2055 (Quote Ref No - 1550)

Product Warranty

NetComm products have a standard 12 months warranty from date of purchase. However some products have an extended warranty option, via registering your product online at the NetComm website www.netcomm.com.au. Refer to the User Guide for complete product warranty conditions, limitations of warranty and other legal and regulatory information.

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

www.netcomm.com.au

Note: NetComm Technical Support for this product only covers the basic installation and features outlined in the Quick Start Guide. For further information regarding the advanced features of this product, please refer to the configuring sections in the User Guide or contact a Network Specialist.

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