TT11 VoIP Router 1FXS/1FXO TA

User Guide

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1.0 INTRODUCTION

Voice over IP is a technology that allows anyone to make a telephone call over the Internet. This is a quick user guide for the TT11 VoIP Router 1FXS/1FXO Telephone Adaptor. It is intended to help you configure this device and have it ready to run within a few minutes. Please follow the user guide carefully as troubleshooting the TA can be very difficult and time consuming.

Before Installation

This product can be set up using a web browser, such as Internet Explorer.

If you purchased this product to make a VoIP call, you must have either an Ethernet-based Cable or a DSL modem with an active connection to the Internet.

2.0 PACKAGE CONTENT

The following materials are included in the package. Please check the package to ensure that all the materials are listed below. Contact TraiTel immediately if an item is missing.

TT11 VoIP Router TA

Ethernet cable

Phone cable

Quick Installation Guide

Power Adapter (12V DC)

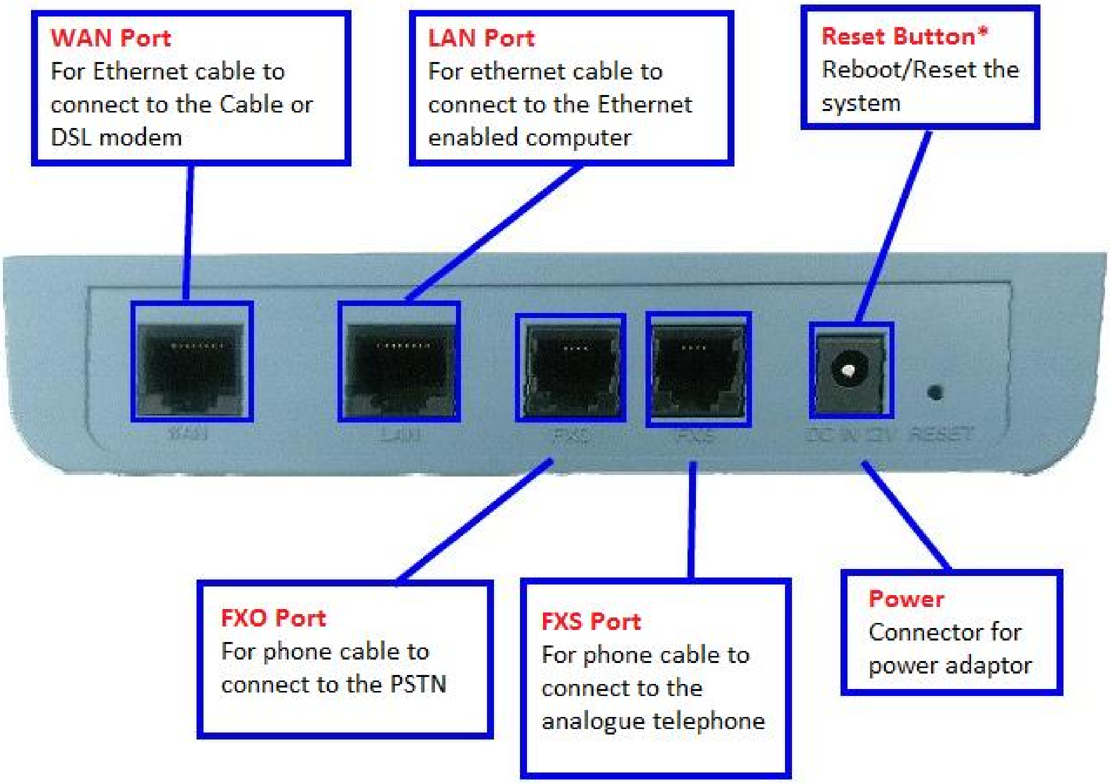
3.0 SUMMARY OF LED & CONNECTOR DESCRIPTION

3.1 The Front LEDs

Item	Name	Colour	Status	Description
1	Power	Green	On	System Power on
			Off	System Power off
2	Run	Green	On	System crash
			Off	System crash
			Flash	System running
3	SIP Reg	Green	On	SIP Registration Ok
			Off	SIP Registration Fail
4	FXS	Green	On	FXS Channel is in use
			Off	FXS Channel is not in use
			Flash	FXS Channel is ringing
5	FXO	Green	On	FXO Channel is in use
			Off	FXO Channel is not in use
6	LAN	Green	On	LAN port is connected
			Off	LAN port is not connected
			Flash	Packet transmit in LAN port
7	WAN	Green	On	WAN port is connected
			Off	WAN port is not connected
			Flash	Packet transmit in WAN port

3.2 The Rear Ports

Reset Button Mode*		
Mode	**Situation A**	**Situation B**
	Press and hold for less than 3 seconds	Press and hold for more than 3 seconds
Result	Reboot the system	Restore to default setting

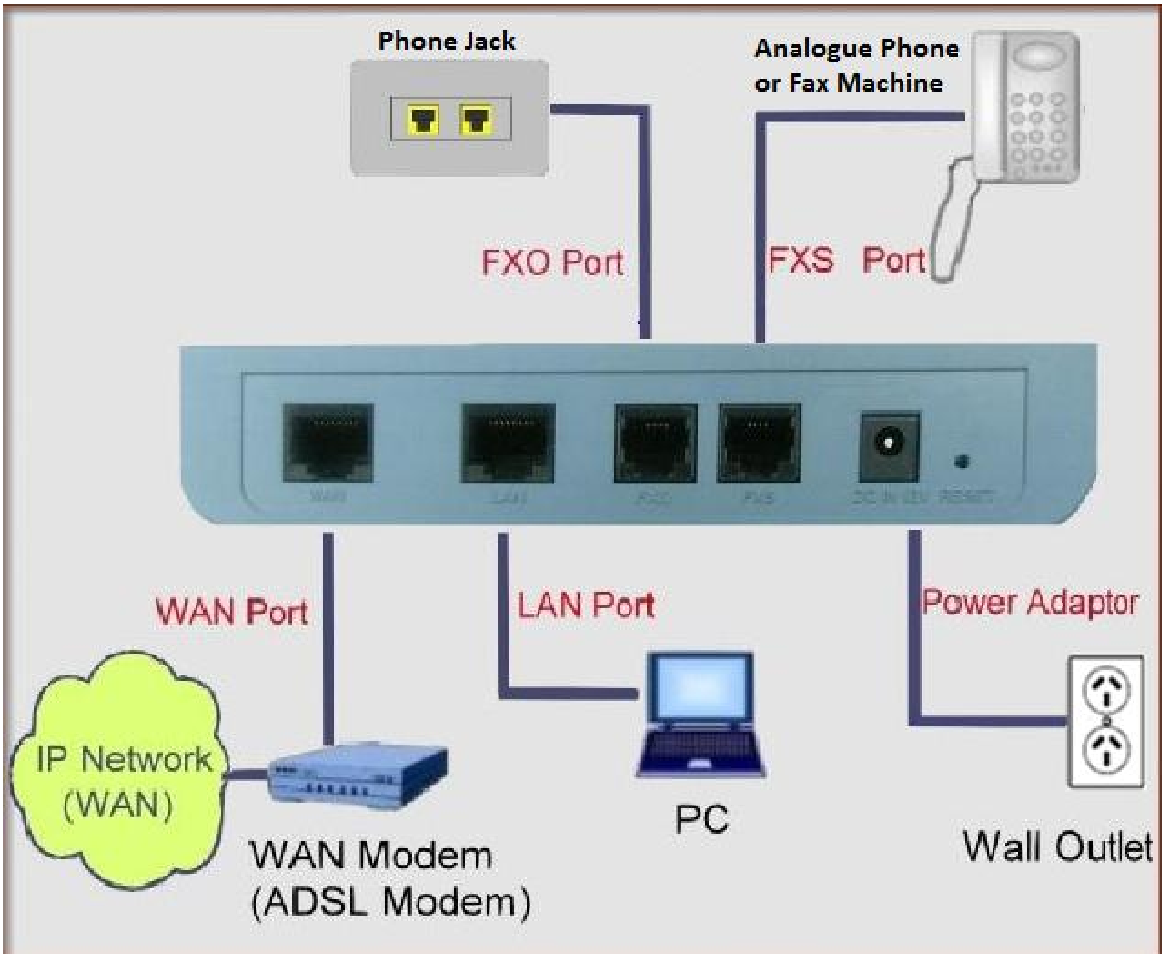


4.0 INSTALLATION

Connecting the TT11 VoIP Router 1FXS/1FXO TA to Your Network

- **A.** Connect the power adapter to this jack. The power LED will light to indicate proper operation.
- **B.** Connect an Ethernet cable to the WAN port when connecting to the Internet access device such as a Cable or DSL modem. The WAN LED will indicate the proper connection.
- **C.** Connect an Ethernet cable to the LAN port when connecting to a PC or an office/home network. The LAN LED will indicate the proper connection.
- **D.** Connect the phone cable to the FXO port when connecting to the phone jack.
- **E.** Connect the phone cable to the FXS port when connecting an analogue telephone.

When you have connected all the cables to the VoIP TA, it will look similar to this:



5.0 USING THE CONFIGURATION MENU

The configuration menu of TT11 VoIP Router 1FXS/1FXO TA can be accessed using a convenient and user-friendly web browser.

If you have an account with TraiTel, please log in at www.traitel.co.nz, select “Voice” then “Configure SIP device”. If you have multiple SIP devices, you **must** create a unique sub-account for each device. Sub-accounts can be created by going to “Settings” then “Sub-Accounts”.

Once you have logged in to your TraiTel account, please continue here.

5.1 Accessing Configuration Menu

- Open the web browser (i.e. Internet Explorer, Netscape…)
- Enter the **IP Address** of the router, which by default is 192.168.15.1 followed by :9999

- The default username and password are “admin” and “admin”.
- Click **OK**

5.2 Main Menu

Once you have logged on to the TT11 VoIP Router TA through the web browser, you can begin the set up according to your requirements. On the configuration main menu, the left navigation panel links you to the set up pages directly. They include:

- **Status**
- **System** (Provision, Firmware, Restart, Backup/Restore, User Management)
- **Networks**
- **VoIP** (SIP Settings, Line Settings, IAX settings, Call Control, Dialling Plan, CDR)
- **Advanced** (Remote Access, Virtual Server, Firewall, MAC Filter, IP Filter, Port

  Filter) The following sections provide an overview of the settings.

5.3 Status – Device Information and Line Status

Status		
Device Information		
System Up-Time	0 Hour 10 Min	
System Current Time	Tue May 11 12:18:01 2010	
Build Time	Mon Sep 1 17:12:36 2008	
Firmware Version	01.00.00 (Sep 1 2008)	
WAN MAC Address	00:D0:E9:01:EF:4B	
LAN MAC Address	00:D0:E9:01:EF:4C	
Register Status		
	SIP Register Status	
	IAX Register Status	
Line Status :	Line 1	Line 2
	IDLE	IDLE

System Up-Time Records system up-time.

System Current Time Shows the system current time. See the Time Zone section for more information.

Build Time Shows the build time of the software.

Firmware Version Shows the firmware version.

WAN MAC Address Shows the WAN MAC address.

LAN MAC Address Shows the LAN MAC address.

SIP Register Status A hotlink to SIP settings status.

IAX Register Status A hotlink to IAX settings status.

Line 1 Status Shows line 1 register status (FXS).

Line 2 Status Shows line 2 register status (FXO).

5.4 Status – LAN

LAN	
IP Address	192.168.15.1
Subnet Mask	255.255.255.0
DHCP Server	Enabled
DHCP Server Start IP	192.168.15.100
DHCP Server End IP	192.168.15.200

IP Address Shows the LAN port IP address.

Subnet Mask Shows the LAN port subnet mask.

DHCP Server Shows the DHCP server status — Enable or Disable

DHCP Server Start IP Shows the start IP address that DHCP server distributes.

DHCP Server End IP Shows the end IP address that the DHCP server distributes.

5.5 Status – WAN and DNS

WAN	
Connection Type	DHCP
IP Address	192.168.10.6
Subnet Mask	255.255.255.0
Default Gateway	192.168.10.1
DHCP Server IP	192.168.10.1
DNS	
DNS Servers IP	192.168.10.1

Connection Type Shows the network connection type of WAN port.

IP Address Shows the WAN port IP address.

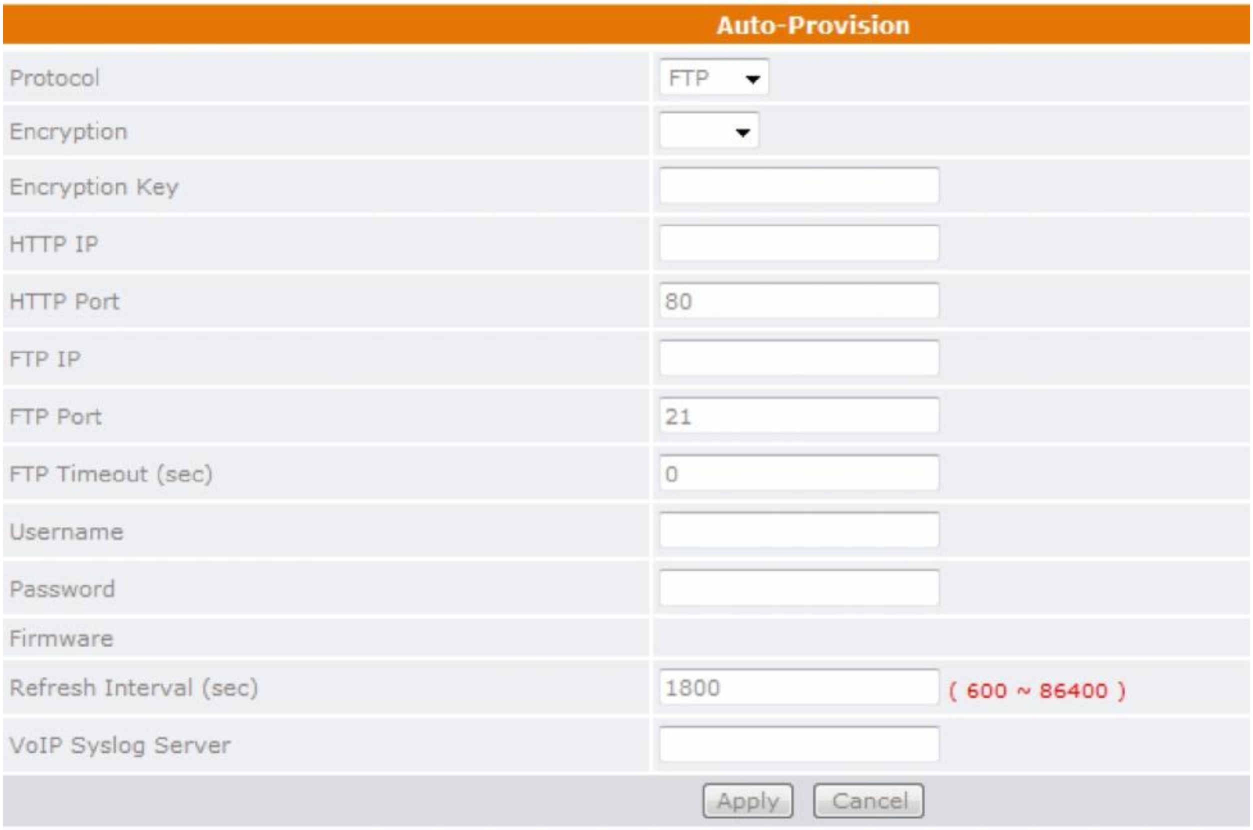
Subnet Mask Shows the WAN port subnet mask

Default Gateway Shows the IP address of default gateway.

DHCP Server IP Shows the DHCP server IP address.

DNS Servers IP Shows the DNS server IP address.

5.6 System – Provision



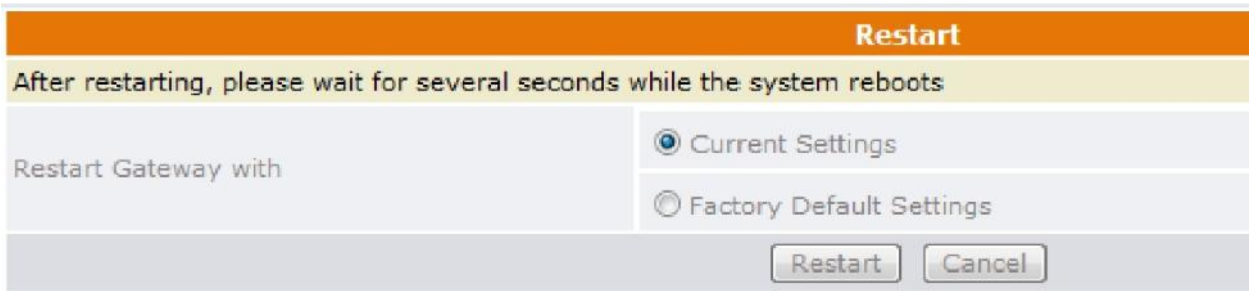
There are 2 kinds of protocol for auto-provision. Please select FTP or HTTP. Click **Apply** to update the modification.

IP, Port, Username, Password, Firmware	Auto-provisioning is an advanced feature of the TT11VoIP Router TA. For further information on using this function, please contact your ISP.

5.7 System – Firmware

Firmware Type	Select the firmware type.
Firmware Filename	Select the saved firmware filename from your computer's folder.
Restore Factory Default Setting	Enable or disable the function of **Restore Factory Default Setting**.
Status	Show status

5.8 System – Restart



Click Restart to update the modification.

Restart Gateway with	Select restart this gateway with **Current Setting** or **Factory Default Settings**.

5.9 System – Backup/Restore

Backup Configuration Can save the backup configuration file into your computer. (“dialplan” for dialling plan or “xconfig” for others)

Restore Configuration Restores the saved configuration file.

5.10 System – User Management

Click **Apply** to update the modification.

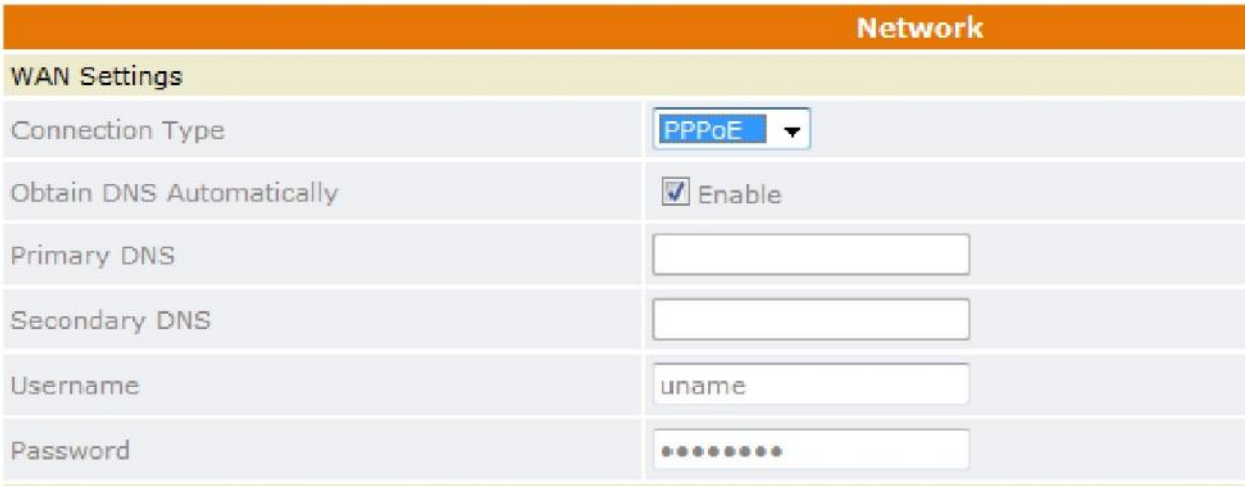
Admin ID Enter the admin ID.

Admin Password Enter or change the admin password.

Guest ID Enter the guest ID.

Guest Password Enter or change the guest password.

5.11 Networks – PPPoE WAN Setting



Select PPPoE as network connection type if your ISP uses PPPoE. Most DSL users use PPPoE.

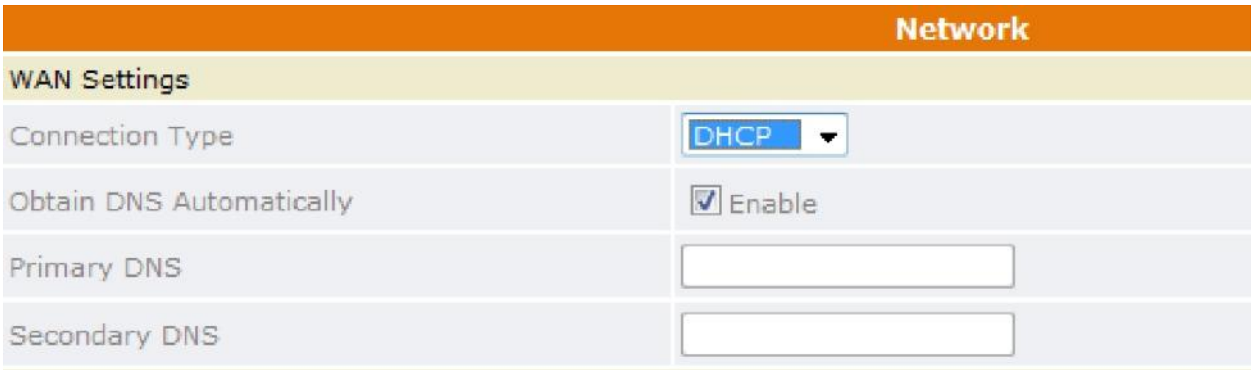
Obtain DNS Automatically	Enable this to obtain DNS automatically.
Primary DNS	Enter the primary DNS server IP address.
Secondary DNS	Enter the secondary DNS server IP address.
Username	PPPoE ID/username proved by your ISP.
Password	PPPoE password.

5.12 Networks – Static WAN Setting

Select Static as the network connection type if all the Wide Area Network IP addresses are provided to you by your ISP.

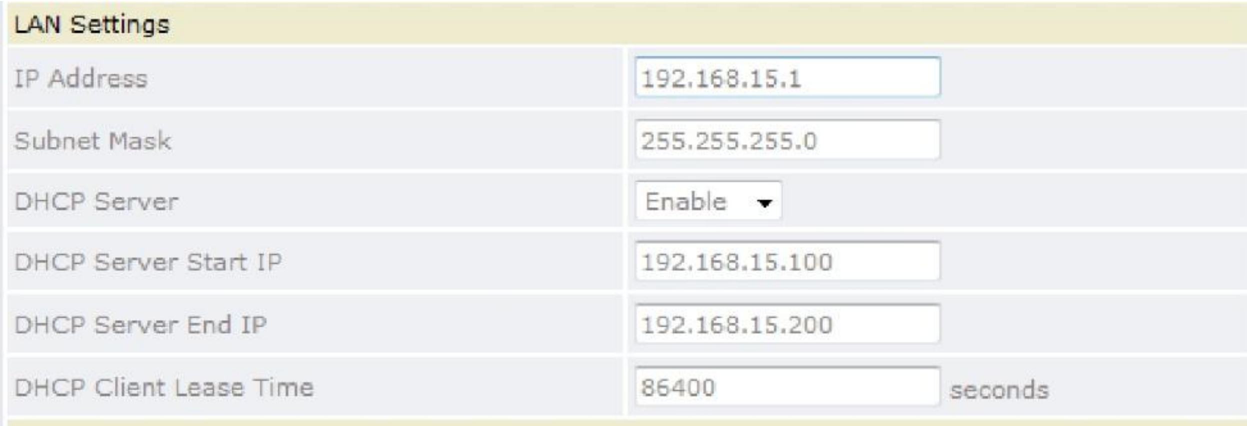
IP Address	Enter the IP address assigned to you by your ISP.
Subnet Mask	Enter the subnet mask address.
Default Gateway	Enter the default gateway IP address.
Primary DNS	Enter the primary DNS server IP address.
Secondary DNS	Enter the secondary DNS server IP address.

5.13 Networks – DHCP WAN Setting



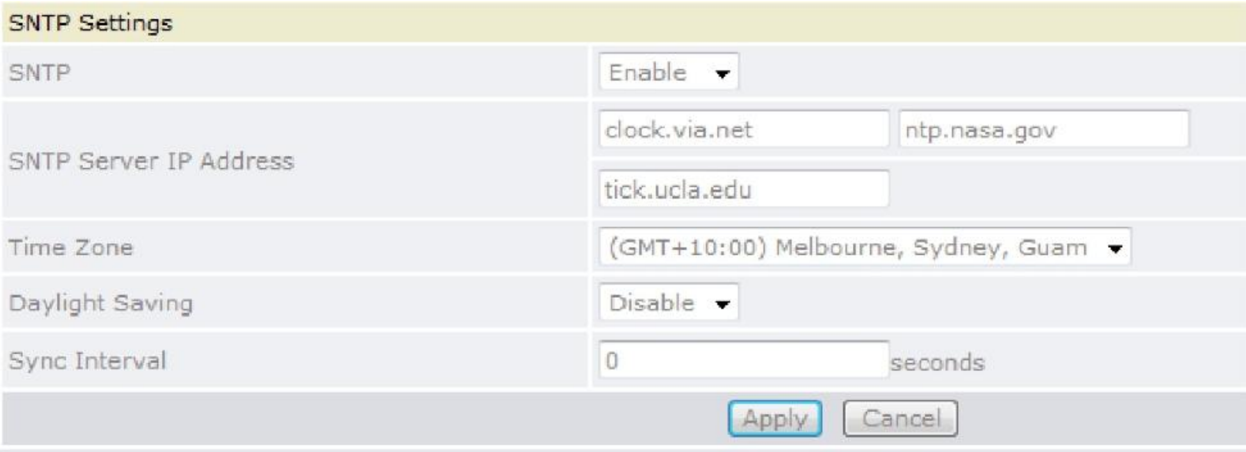
Select DHCP as network connection type that allows the network administrator to distribute IP addresses when this gateway is plugged into a different place in the network.

5.14 Networks – LAN Setting



IP Address	LAN port IP address.
Subnet Mask	LAN port Subnet Mask.
DHCP Server	Enable or disable LAN side DHCP server.
DHCP Server Start IP	The starting IP address for the DHCP server's IP assignment if this function is enabled.
DHCP Server End IP	The end IP address for the DHCP server's IP assignment if this function is enabled.
DHCP Client Lease Time	The time period for the DHCP server to expire the IP that is assigned.

5.15 Networks – SNTP Setting



Click **Apply** to update the modification.

SNTP	Enable/Disable SNTP.
SNTP Server IP Address	Enter the SNTP server IP address. The SNTP server allows the gateway to synchronise the local time with the remote server.
Time Zone	Choose your time zone.
Daylight Saving	Enable or disable daylight saving.
Sync Interval	Show the periodic interval the gateway waits before it resynchronises the gateway's time with that of the specified SNTP server.

5.16 VoIP – SIP Settings

Click **Apply** to update the modification.

Display Name	The name displayed on the LCD. Note that TraiTel does not send this information to the caller. The username provided under the VoIP configuration goes here.
Number	The number in the URI displayed on the LCD. Note that Traitel does not send this number to the caller. The username provided under the VoIP configuration goes here.
Username	Username to log in to the SIP server. The username provided under the VoIP configuration goes here.
Password	User password to log in to the SIP server. Your TraiTel account main password goes here.
Register Status	The current status of the SIP registration.

Local Port	The local SIP port of the TA (typically 5060).
Outbound Proxy Server	Outbound proxy server IP address.
Outbound Proxy Port	Port number of Outbound Proxy server.
Send Messages via Outbound Proxy	Enable/Disable send messages via outbound proxy. When this feature is enabled, all messages will be sent through the outbound proxy.
Registrar Server	SIP Registrar server IP address.
Registrar Port	Port number of SIP Registrar server.
Register Fail Retry Time	The periodic interval at which the device retries the SIP registration after a failure of the SIP registration.
Session Type	Select Invite or Update as the Session Timer method.
Session Refresher	Select UAC (User Agent Client) or UAS (User Agent Server) as the Session Timer refresher.
Session Expires	The time interval in which the TA periodically refreshes SIP sessions by sending repeated Invite or Update requests, depending on the session type.
Register Expires	The time after which the registration on the SIP Registrar expires. The TA must send SIP Register to keep the registration at half the setting time.
UDP Timeout	Timeout for SIP requests (100 ~ 3000ms). Set "0" to disable this feature and follow the RFC 3261 rule.
UDP Retry Times	The number of times to send SIP requests. Set to "0" to disable this feature and follow the RFC 3261 rule.
Enable PRACK	A SIP method which acknowledges provisional responses like 180 ringing. Enable for a more reliable connection.
Anonymous	1. If Disable is selected, the full URI and name are sent to the receiver's device when the user makes a call. 2. When Full URI is selected, "Anonymous" is sent to the receiver's device. It may display anonymous or nothing on the receiver's device.

3. When Display name is selected, only the display name is replaced by "Anonymous" when the user makes a phone call. It may display anonymous or nothing on the receiver's device.

Anonymous Reject — Select Enable to reject anonymous calls.

Redundancy Package — Enable/Disable the B2B service which is a special platform that provides advanced VoIP services. Please contact your service provider for details.

5.17 VoIP – Line Settings

Dial Button — Allows the user to dial numbers using the web interface.

Polarity Reversal — Enable/Disable the polarity reversal signal.

TX Gain — Set a specific send audio level .

RX Volume — Set a specific receive audio level.

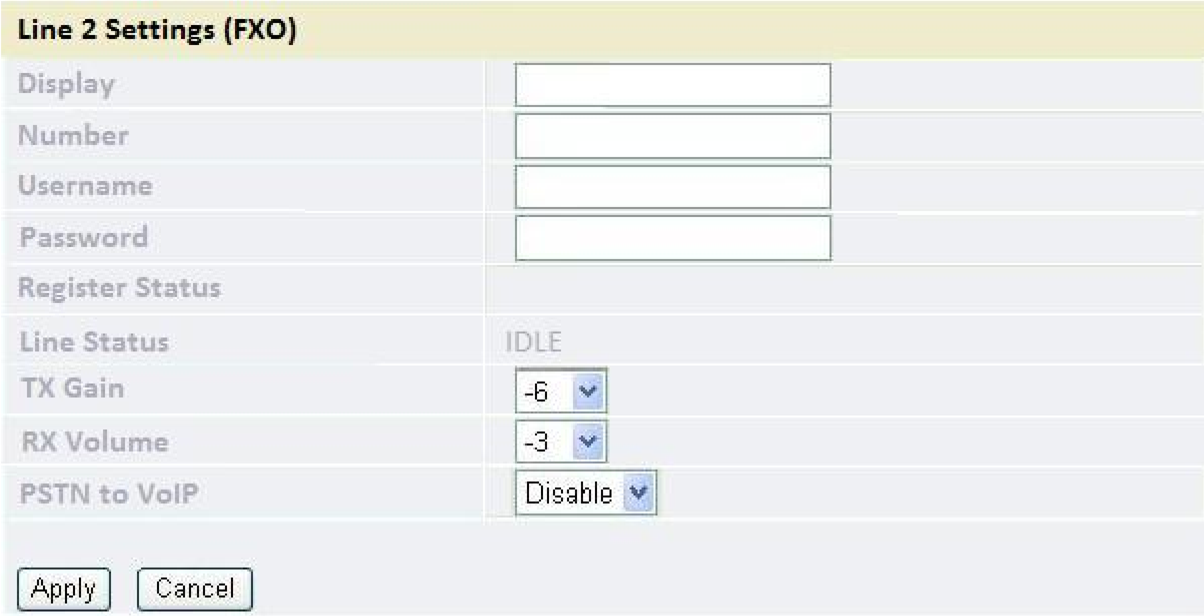
Do Not Disturb — Select Enable to reject incoming calls or Disable to disable the function.

Call Forward — Call Forward allows you to forward incoming calls to a pre-designated telephone number. It includes **No Answer, Busy** and **Unconditional**. Please enter the IP address, URI or number to be registered with the SIP server.

Call Forward Timer — The time elapsed before the call is considered not answered and is forwarded to a designated number. The default is 10 seconds.

Call Waiting — Call Waiting ensures that all important calls get to you. For example, if you are on the phone when another person tries to call you, an audible beep will inform you that someone is waiting on the other line. You can decide whether you want to put the current caller on hold and take the incoming call.

T38 Enable/Disable T.38 support, normally for fax users. Choose “Talk after fax end” to resume the phone call after the fax process is finished.



Display Name displayed on the LCD for the caller.

Number The number in the URI displayed on the LCD for the caller.

Username Username to log in to the SIP server.

Password Password to log in to the SIP server.

Register Status Shows the registration status in the Register Server.

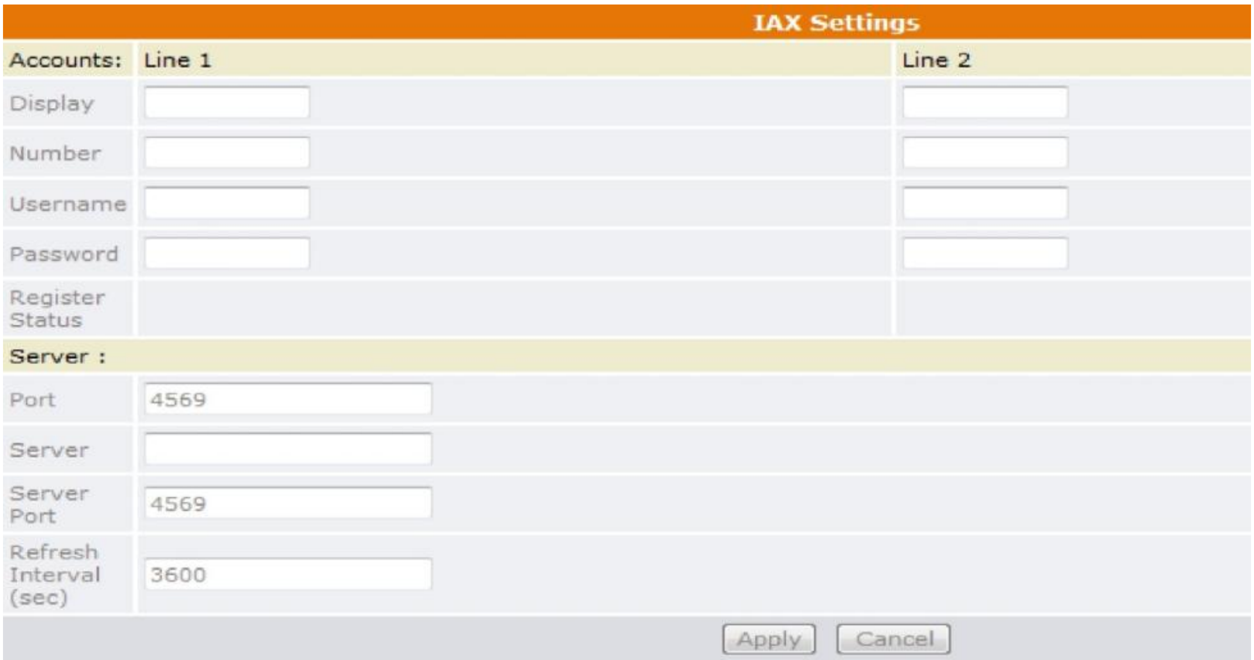
Line Status Shows the line status.

TX Gain Set a specific transmit level

RX Volume Set a specific receive level.

PSTN to VoIP Enable/Disable the PSTN to VoIP feature. When enabled, any PSTN call will transfer to a VoIP call. When disabled, any PSTN call will ring the analogue phone connected to the FXS port.

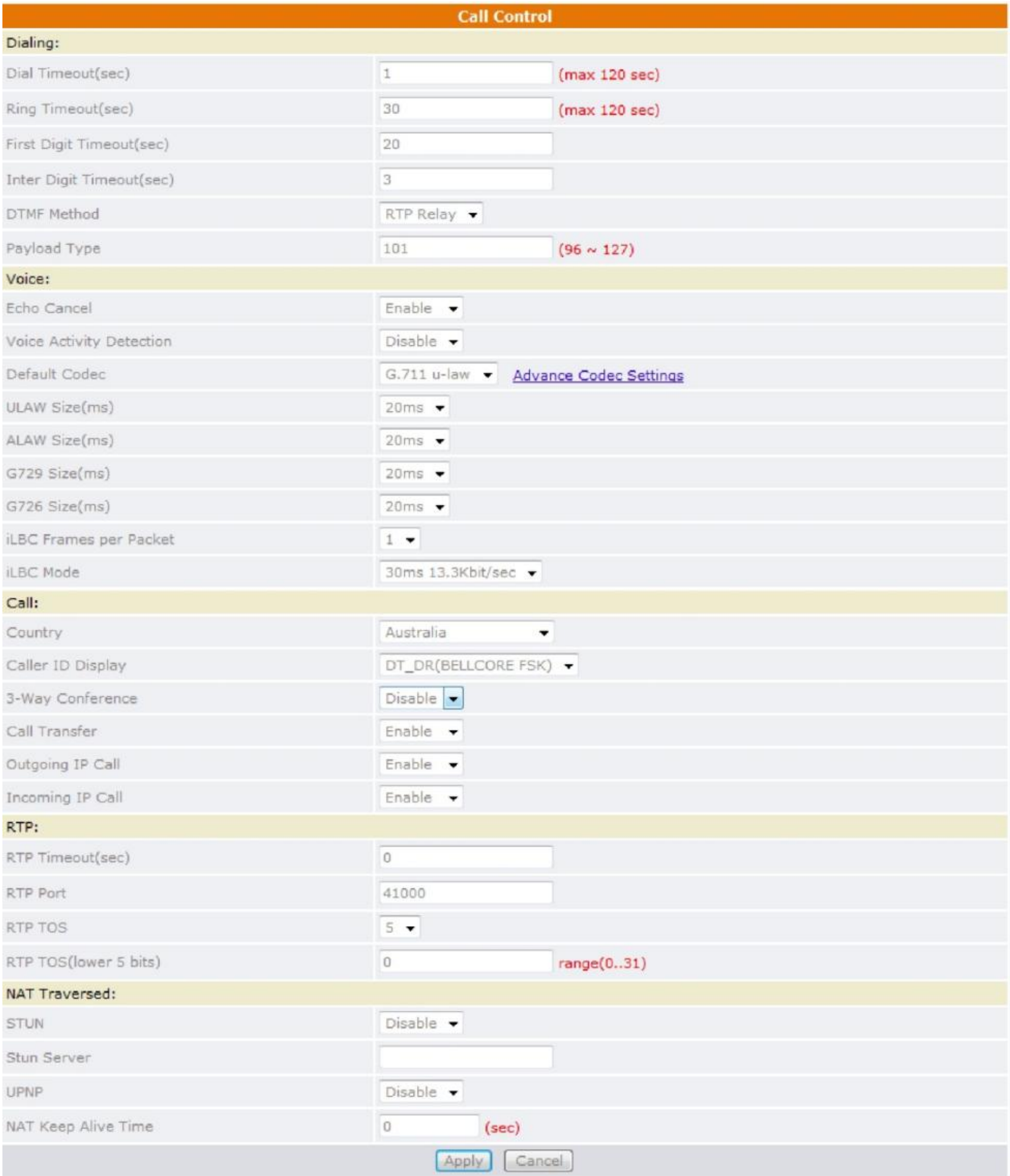
5.18 VoIP – IAX Settings



IAX- Inter Asterisk exchange protocol is a proprietary protocol of Asterisk by Digium. It is a simple protocol like SIP. Yet, it can pass through any kind of NAT due to the way that it's been designed.

Display Name	Name displayed on the LCD for the caller.
Number	The number in the URI displayed on the LCD for the caller.
Username	User name to log into the IAX server.
Password	User password to log into the IAX server.
Register Status	The current status of the IAX Registration
Port	TT11 VoIP Router TA supports IAX2 protocol. Normally IAX2 uses Port 4569.
Server	The Asterisk server's IP address
Server Port	The port number for the Asterisk server. Default is 4569
Refresh Interval	The time interval at which the phone periodically refreshes IAX sessions.

5.19 VoIP – Call Control



Click **Apply** to update the modification.

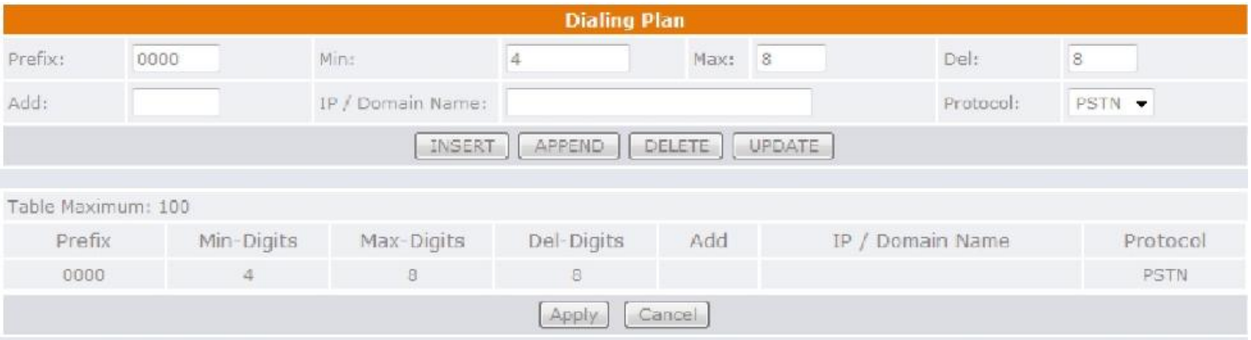
Dial Timeout

The TT11 VoIP Router TA will automatically cancel a call out if the caller does not pick up the phone within the set time.

Ring Timeout	The TT11 VoIP Router will automatically reject a call if the call is not picked up within the set time.
First Digit Timeout	Specifies the maximum duration for the first digit to be keyed in.
Inter Digit Timeout	Specifies the maximum duration between successive digits before the dialled in numbers are sent out.
DTMF Method	Please choose RTP Relay, Voice or SIP Info.
Payload Type	The Payload type for the DTMF method "RTP Relay".
Echo Cancel	The algorithm for cancelling echo within the voice stream. Enabling this function is recommended.
Voice Activity Detection	The voice activity detection (VAD) is a component of the DSP software that examines a caller's incoming signal and determines if the signal contains sufficient energy and is likely to be speech rather than a click.
Default Codec	Default voice codec.
ULAW, ALAW, G.729, G.726 Size	One RTP packet is sent out on every specified time cycle.
iLBC Frames per Packet	Selectable amount of frames for each RTP packet.
iLBC Mode	One RTP packet is sent out on every specified time cycle.
Country	Defines the user location to fit local Telco system requirements.
Caller ID Display	Select the method of Caller ID generation.
3-Way Conference	Enable/Disable 3 way conference. Hook-flash to start.
Call Transfer	Enable to transfer the call after it hangs up from a 3 way conference. The TT11 must be the initiator of the 3 way conference call.
Outgoing IP Call	Enable/Disable outgoing IP call.
Incoming IP Call	Enable/Disable incoming IP call.

RTP Timeout (sec)	The timer for terminating the SIP session if the gateway is aware of the absence of an RTP stream.
RTP Port	Initial port number for sending RTP packets.
RTP TOS	Type of service value for Quality of Service.
RTP TOS (lower 5 bits)	The lower 5 bits of the TOS field (0~31).
STUN	Enable/Disable STUN (Simple Traversal of UDP through NAT). This function is used for NAT traversal.
Stun Server	The IP address or host name of the STUN server.
UPNP	Enable/Disable UPnP (Universal Plug & Play). This function is used for NAT traversal.
NAT Keep Alive Time	The time interval that the IP phone sends the keep-alive packet in order to ensure that NAT works properly.

5.20 VoIP – Dialling Plan



Local dialling plan allows users to dial out to a VoIP Device using a pre-defined number. Users do not have to change their dialling habit. Click **Apply** to update the modification.

Prefix	Numbers defined here are used as the beginning digits of the dialling pattern. Maximum input length is 6 digits.
Min.	Minimum digits user can key in.
Max.	Maximum digits user can key in.
Del.	Number of digits defined in this field will be removed from the dialling pattern. For example, if we dialled 81352109378 and the delete digit is 2, then the actual dialled number is 352109378. First 2 digits are removed. Maximum digit delete is 3 digits.
Add	Numbers in this field are added at the beginning of the dialling pattern. For example, if 001 is in this field, the number dialled is 001+the rest of the numbers. The input length is limited to 6 digits.
IP / Domain Name	The IP address or domain name of the remote side VoIP device. When the prefix number is matched, this call will go to the VoIP device with this IP address or domain name.
Protocol	Choose the dialling plan for SIP or IAX.
[Insert]	Insert a record where the current record is located (Current record is in a different colour).
[Append]	Add a new record to the bottom of the list.
[Delete]	Delete the selected record.
[Update]	Modify the value of the selected record.

5.21 VoIP – CDR

Please click the **Refresh** button to see the updated CDR.

5.22 Advanced – Remote Access

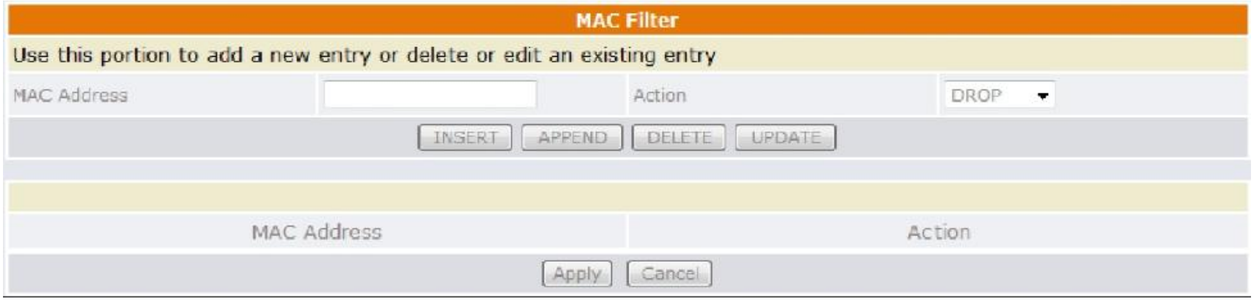
To temporarily permit remote administration of the gateway (i.e. from outside your LAN). Click **Apply** to update the modification.

5.23 Advanced – Virtual Server

Click **Apply** to update the modification.

Service Type	Select the service you wish to configure.
Protocol	Automatic when you choose service type.
Start Port	Enter the public start port number to configure.
End Port	Enter the public end port number to configure.
IP Address	Enter the IP address of a specific internal server to which requests from the specified port are forwarded.
DMZ	Enable/Disable the DMZ function.
DMZ IP Address	Enter the IP address of a specific internal server which needs to be accessible from the external network.

5.24 Advanced – Firewall – MAC Filter



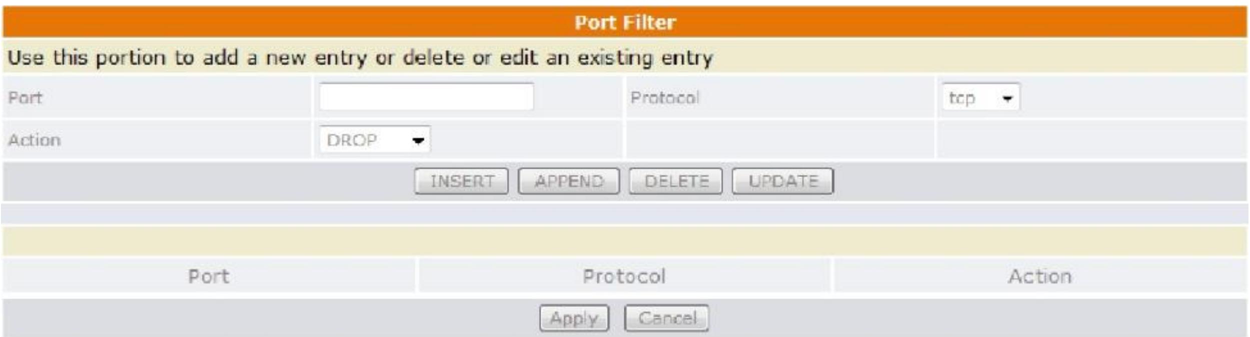
A MAC (Media Access Control) address is the unique networks hardware identifier for each PC on your network's interface (i.e. its Network Interface Card or Ethernet Card). Using your gateway's MAC address filter function, you can configure the switch to only accept traffic from specified machines, or else to block specific machines from accessing your LAN.

There is no pre-defined MAC address filter rule; you can add the filter rules to meet your requirements. Click **Apply** to update the modification.

5.25 Advanced – Firewall – IP Filter

Using your gateway's IP filter function, you can configure the switch to only accept traffic from specified machines, or else to block specific machines from accessing your LAN. Click **Apply** to update the modification.

5.26 Advanced – Firewall – Port Filter



Using your gateway's port filter function, you can configure the switch to only accept traffic from specified machines, or else to block specific machines from accessing your LAN. Click **Apply** to update the modification.

6.0 Troubleshooting

If your TT11 VoIP Router TA is not functioning properly, you can refer to this chapter first for sample troubleshooting before contacting TraiTel. This can save you time and effort but if the issues are not covered here please call or send an email to customer-service@traitel.com.au.

ISSUE	RECOMMENDED ACTION
I forgot my TT11 VoIP Router TA log in and/or password.	1. Try the default log in and password, please refer to Chapter 5.1 2. Restore the TA to its factory default settings by holding the Reset button on the back for 6 seconds or more.
None of the LEDs are on when I turn on the TT11 VoIP Router TA.	1. Check if the power cord is connected properly. 2. Check if there is AC power coming from the power outlet.
Why can't I ping any PC on the LAN?	1. Check the LAN LED on the front panel. The LED should be lit on the port that has a PC connected. If it is off, check the cables between your TT11 and the PC. 2. Verify the IP address and subnet mask are consistent between the router and the workstations.
Why can't I dial my friend's SIP number?	1. Check the Registrar server domain name/IP address and outbound proxy domain name/IP address (under SIP settings in the configuration menu). Ensure you have the correct name or IP address. 2. Check the LED display on the front panel of the TT11 to see if the SIP Reg LED is lit. If it is not lit, use a web browser and access the configuration menu. Make sure that the Registrar server domain name/IP address is correct. 3. Check the Register status under SIP account settings in the configuration menu (from the web browser). If your status is unregistered, it means you do not have a SIP account. Contact Your SIP service provider to get an account.
Why isn't my firmware updating?	1. Your TT11 VoIP Router TA automatically detects new firmware when you plug in the power. If a new version is available the TA will automatically update the firmware. 2. Check the FTP address is correct.

	3. Check with TraiTel if the firmware filename is correct.
Why do I get "Can't Upgrade Now" on the screen when I click [Apply] in the configuration menu?	Make sure you exit the Setting mode (phonebook, menu, speed dial...) before you click [Apply] in the configuration menu.

Appendix: Product Specification

Protocol
- IETF SIP (RFC3261)
- IAX2

Network Interface
- RJ-45 x2, 10/100BaseT (WAN*1 & LAN*1)

Phone Interface
- RJ-11 x1, connect to analogue phone
- RJ-11 x1, connect to PSTN

Call Features
- Call Transfer (attended)
- Call Forward (Busy/No Answer/Unconditional)
- Call Hold/Retrieve
- Call Waiting
- ID Display
- Anonymous Call
- Anonymous Call Blocking

Fax Support
- G.711 pass-through
- T.38

Codec
- G.711 μ-law
- G.711a-law
- G.723.1 (5.3k) (option)
- G.723.1 (6.3k) (option)
- G.726
- G.729a
- iLBC

DTMF
- In-band DTMF
- Out-of-band DTMF (RFC2833)
- SIP INFO

SIP Server Support
- Registrar Server (set from web)
- Outbound Proxy (set from web)

Security
HTTP 1.1 basic/digest authentication for web set up
MD5 for SIP authentication (RFC 2069/RFC 2617)

Dial Methods
- Direct IP call without SIP proxy
- Dial number via SIP server
- Dial URI via SIP server

Router
- Virtual Server
- Firewall
- Remote
- Access NAT
- DHCP Server
- DMZ

Voice Quality
- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generation)
- AEC (Acoustic Echo Cancellation) G.168
- Jitter buffer

QoS
- ToS field

Dial Signal
- DTMF dialling

Tone
- DTMF Ring
- Tone
- Ring Back Tone (local and remote)
- Dial Tone
- Busy Tone

IP Assignment
- Static IP
- DHCP
- PPPoE

NAT Traversal
- UPnP
- STUN

TCP/IP
- IP/TCP/UDP/DHCP/RTP/ICMP/
- HTTP/SNTP/FTP/DNS

Configuration
- Web browser
- Auto-provisioning system

Firmware Upgrade
- Web-based Interface
- Auto-provisioning system

Power
- Input AC 100-120V/220-240V 50/60Hz
- Output DC 12V

Environmental
- Operating temperature 0 ~ 40°C
- Storage temperature: -20 ~ 60°C
- Operating humidity: 20% ~ 80%

Physical Dimensions
- Size: 157(l) x 121(w) x 33(h) mm
- Weight: 260g
- Colour: Blue/White

Certification Compliance
- FCC Part 15 Class B
- CE Class B
- VCCI Class B
- EN60950