



# **TT11 VoIP Router 1FXS/1FXO TA**

## **User Guide**

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## 1.0 INTRODUCTION

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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

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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

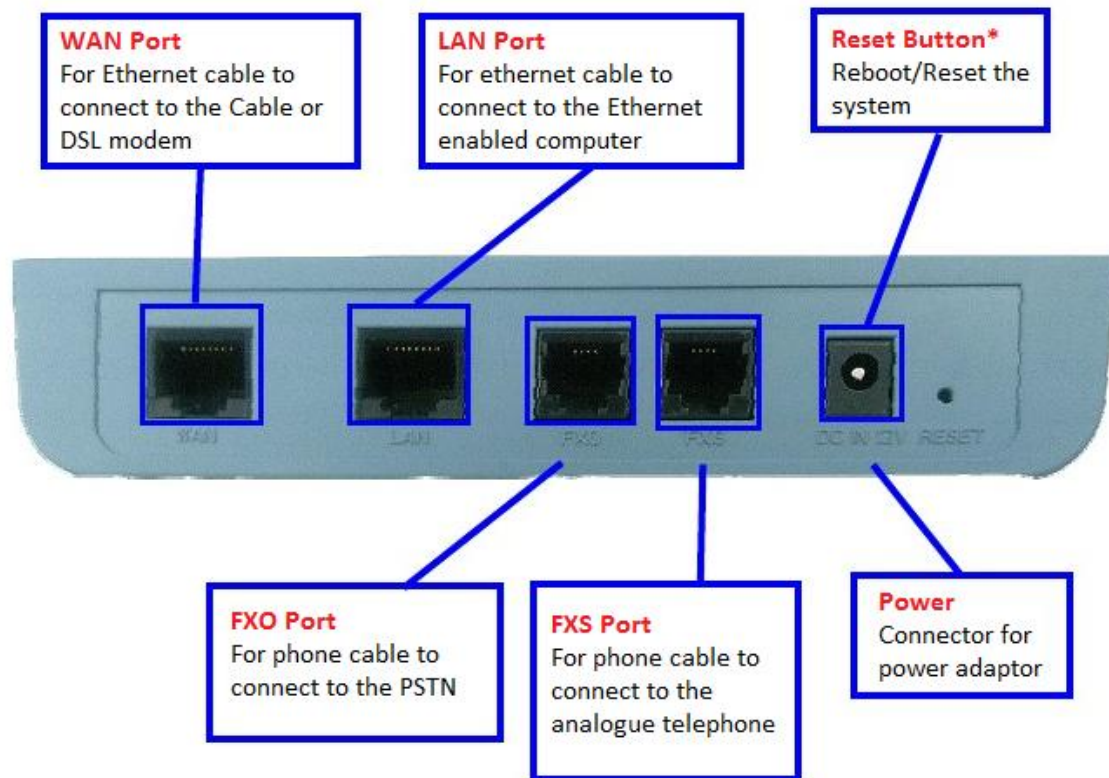
### 3.1 The Front LEDs

Item	Name	Color	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



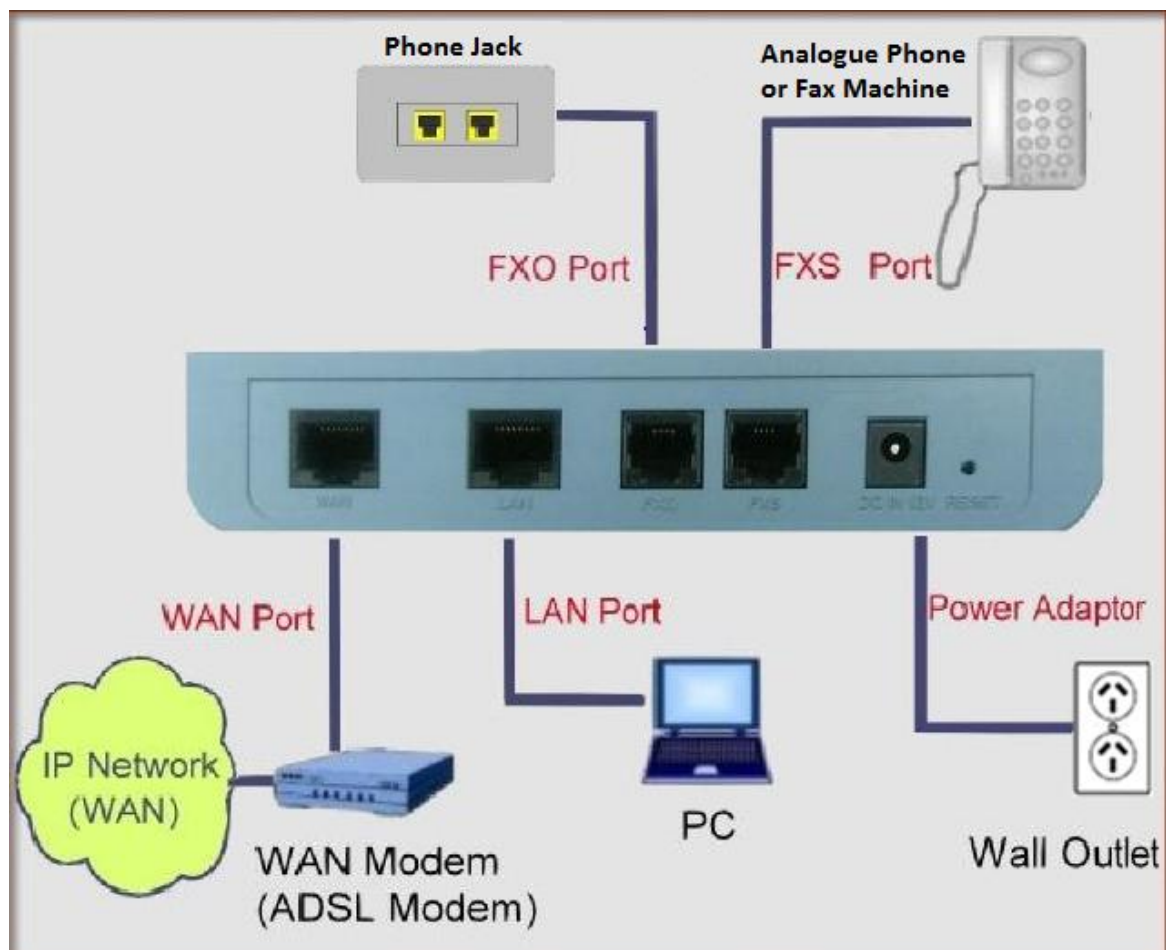
## 4.0 INSTALLATION

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### 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 analog telephone.

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



## 5.0 USING THE CONFIGURATION MENU

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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.com](http://www.traitel.com), 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, Dialing 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		
	▶	<a href="#">SIP Register Status</a>
	▶	<a href="#">IAX Register Status</a>
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

<b>IP Address</b>	Shows the LAN port IP address.
<b>Subnet Mask</b>	Shows the LAN port subnet mask.
<b>DHCP Server</b>	Shows the DHCP server status — Enable or Disable
<b>DHCP Server Start IP</b>	Shows the start IP address that DHCP server distributes.
<b>DHCP Server End IP</b>	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

<b>Connection Type</b>	Shows the network connection type of WAN port.
<b>IP Address</b>	Shows the WAN port IP address.
<b>Subnet Mask</b>	Shows the WAN port subnet mask
<b>Default Gateway</b>	Shows the IP address of default gateway.
<b>DHCP Server IP</b>	Shows the DHCP server IP address.
<b>DNS Servers IP</b>	Shows the DNS server IP address.

## 5.6 System – Provision

Auto-Provision	
Protocol	FTP ▼
Encryption	▼
Encryption Key	<input type="text"/>
HTTP IP	<input type="text"/>
HTTP Port	80
FTP IP	<input type="text"/>
FTP Port	21
FTP Timeout (sec)	0
Username	<input type="text"/>
Password	<input type="text"/>
Firmware	<input type="text"/>
Refresh Interval (sec)	1800 ( 600 ~ 86400 )
VoIP Syslog Server	<input type="text"/>
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>	

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 Download	
Firmware Type	image.flash ▼
Firmware Filename	<input type="text"/> <input type="button" value="Browse..."/>
Restore Factory Default Settings	Disable ▼
Status	
<input type="button" value="Download"/> <input type="button" value="Cancel"/>	

### 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

Restart	
After restarting, please wait for several seconds while the system reboots	
Restart Gateway with	<input checked="" type="radio"/> Current Settings
	<input type="radio"/> Factory Default Settings
<input type="button" value="Restart"/> <input type="button" value="Cancel"/>	

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

The screenshot shows a web interface titled "Backup / Restore". It contains two main sections: "Backup Configuration" and "Restore Configuration". The "Backup Configuration" section has a "Backup" button. The "Restore Configuration" section has a "Configuration File" input field, a "Browse..." button, and a "Restore" button. A warning message states: "Restore will overwrite the current configuration and restart the device. If you want to keep the current configuration, please use 'Backup' first to save the current configuration."

### Backup Configuration

Can save the backup configuration file into your computer. ("dialplan" for dialing plan or "xconfig" for others)

### Restore Configuration

Restores the saved configuration file.

## 5.10 System – User Management

The screenshot shows a web interface titled "User Management". It contains four input fields: "Admin ID" (with value "admin"), "Admin Password" (with masked characters "\*\*\*\*\*" and a "Change" button), "Guest ID" (with value "user"), and "Guest Password" (with masked characters "\*\*\*\*" and a "Change" button). Below these fields is a link "Restore guest's default setting". At the bottom are "Apply" and "Cancel" buttons.

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

Network	
WAN Settings	
Connection Type	PPPoE ▼
Obtain DNS Automatically	<input checked="" type="checkbox"/> Enable
Primary DNS	<input type="text"/>
Secondary DNS	<input type="text"/>
Username	uname
Password	••••••••

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

Network	
WAN Settings	
Connection Type	<b>STATIC</b> ▼
IP Address	192.168.12.1
Subnet Mask	255.255.255.0
Default Gateway	192.168.12.1
Obtain DNS Automatically	<input type="checkbox"/> Enable
Primary DNS	
Secondary DNS	

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

Network	
WAN Settings	
Connection Type	<b>DHCP</b> ▼
Obtain DNS Automatically	<input checked="" type="checkbox"/> Enable
Primary DNS	
Secondary DNS	

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

LAN Settings	
IP Address	<input type="text" value="192.168.15.1"/>
Subnet Mask	<input type="text" value="255.255.255.0"/>
DHCP Server	<input type="text" value="Enable"/>
DHCP Server Start IP	<input type="text" value="192.168.15.100"/>
DHCP Server End IP	<input type="text" value="192.168.15.200"/>
DHCP Client Lease Time	<input type="text" value="86400"/> seconds

### 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

SNTP Settings	
SNTP	Enable ▼
SNTP Server IP Address	clock.via.net ntp.nasa.gov
	tick.ucla.edu
Time Zone	(GMT+10:00) Melbourne, Sydney, Guam ▼
Daylight Saving	Disable ▼
Sync Interval	0 seconds
<div>Apply Cancel</div>	

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 synchronize 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 resynchronizes the gateway's time with that of the specified SNTP server.

## 5.16 VoIP – SIP Settings

SIP Settings		
Accounts:	Line 1	Line 2
Display Name	<input type="text" value="callmedemo.7"/>	<input type="text"/>
Number	<input type="text" value="callmedemo.7"/>	<input type="text"/>
Username	<input type="text" value="callmedemo.7"/>	<input type="text"/>
Password	<input type="password" value="*****"/>	<input type="password"/>
Register Status	Register OK	
Register:		
Local Port	<input type="text" value="5060"/>	
Outbound Proxy Server	<input type="text" value="test.sip.traitel.com.au"/>	
Outbound Proxy Port	<input type="text" value="5060"/>	
Send Messages via Outbound Proxy	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	
Registrar Server	<input type="text" value="test.sip.traitel.com.au"/>	
Registrar Port	<input type="text" value="5060"/>	
Register Fail Retry Time	<input type="text" value="300"/> ( 10 ~ 600 sec)	
Session	<input type="text" value="SIP/2.0"/>	

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.

<b>Local Port</b>	The local SIP port of the TA (typically 5060).
<b>Outbound Proxy Server</b>	Outbound proxy server IP address.
<b>Outbound Proxy Port</b>	Port number of Outbound Proxy server.
<b>Send Messages via Outbound Proxy</b>	Enable/Disable send messages via outbound proxy. When this feature is enabled, all messages will be sent through the outbound proxy.
<b>Registrar Server</b>	SIP Registrar server IP address.
<b>Registrar Port</b>	Port number of SIP Registrar server.
<b>Register Fail Retry Time</b>	The periodic interval at which the device retries the SIP registration after a failure of the SIP registration.
<b>Session Type</b>	Select Invite or Update as the Session Timer method.
<b>Session Refresher</b>	Select UAC (User Agent Client) or UAS (User Agent Server) as the Session Timer refresher.
<b>Session Expires</b>	The time interval in which the TA periodically refreshes SIP sessions by sending repeated Invite or Update requests, depending on the session type.
<b>Register Expires</b>	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.
<b>UDP Timeout</b>	Timeout for SIP requests (100 ~ 3000ms). Set "0" to disable this feature and follow the RFC 3261 rule.
<b>UDP Retry Times</b>	The number of times to send SIP requests. Set to "0" to disable this feature and follow the RFC 3261 rule.
<b>Enable PRACK</b>	A SIP method which acknowledges provisional responses like 180 ringing. Enable for a more reliable connection.
<b>Anonymous</b>	<ol style="list-style-type: none"> <li>1. If Disable is selected, the full URI and name are sent to the receiver's device when the user makes a call.</li> <li>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.</li> </ol>

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.

Session Type	INVITE ▼
Session Refresher	▼
Session Expires	1800 sec
Register Expires	3600 sec
UDP Time Out	1500 ( 100 ~ 3000 msec)
UDP Retry Times	3 ( 1 ~ 6 )
Enable PRACK	Disable ▼
Anonymous	Disable ▼
Anonymous Reject	Disable ▼
Redundancy Package	Disable ▼

## 5.17 VoIP – Line Settings

Line Settings		
Line 1 Settings ( FXS )		
DIAL1	96962118	
Polarity Reversal	Disable ▼	
TX Gain	-4 ▼	
RX Volume	-4 ▼	
Do Not Disturb	Disable ▼	
Call Forward	<input type="checkbox"/> No Answer	
	<input type="checkbox"/> Busy	
	<input type="checkbox"/> Unconditional	
Call Forward Timer(sec)	10 (default 10)	
Call Waiting	Enable ▼	
T38	Disable ▼	

### 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.

Line 2 Settings (FXO)	
Display	<input type="text"/>
Number	<input type="text"/>
Username	<input type="text"/>
Password	<input type="text"/>
Register Status	
Line Status	IDLE
TX Gain	-6 ▼
RX Volume	-3 ▼
PSTN to VoIP	Disable ▼
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>	

#### 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 analog phone connected to the FXS port.

## 5.18 VoIP – IAX Settings

IAX Settings		
Accounts:	Line 1	Line 2
Display	<input type="text"/>	<input type="text"/>
Number	<input type="text"/>	<input type="text"/>
Username	<input type="text"/>	<input type="text"/>
Password	<input type="text"/>	<input type="text"/>
Register Status		
Server :		
Port	<input type="text" value="4569"/>	
Server	<input type="text"/>	
Server Port	<input type="text" value="4569"/>	
Refresh Interval (sec)	<input type="text" value="3600"/>	
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>		

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

Call Control	
<b>Dialing:</b>	
Dial Timeout(sec)	1 (max 120 sec)
Ring Timeout(sec)	30 (max 120 sec)
First Digit Timeout(sec)	20
Inter Digit Timeout(sec)	3
DTMF Method	RTP Relay
Payload Type	101 (96 ~ 127)
<b>Voice:</b>	
Echo Cancel	Enable
Voice Activity Detection	Disable
Default Codec	G.711 u-law <a href="#">Advance Codec Settings</a>
ULAW Size(ms)	20ms
ALAW Size(ms)	20ms
G729 Size(ms)	20ms
G726 Size(ms)	20ms
iLBC Frames per Packet	1
iLBC Mode	30ms 13.3Kbit/sec
<b>Call:</b>	
Country	Australia
Caller ID Display	DT_DR(BELLCORE FSK)
3-Way Conference	Disable
Call Transfer	Enable
Outgoing IP Call	Enable
Incoming IP Call	Enable
<b>RTP:</b>	
RTP Timeout(sec)	0
RTP Port	41000
RTP TOS	5
RTP TOS(lower 5 bits)	0 range(0..31)
<b>NAT Traversed:</b>	
STUN	Disable
Stun Server	
UPNP	Disable
NAT Keep Alive Time	0 (sec)
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>	

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.



<b>Ring Timeout</b>	The TT11 VoIP Router will automatically reject a call if the call is not picked up within the set time.
<b>First Digit Timeout</b>	Specifies the maximum duration for the first digit to be keyed in.
<b>Inter Digit Timeout</b>	Specifies the maximum duration between successive digits before the dialed in numbers are sent out.
<b>DTMF Method</b>	Please choose RTP Relay, Voice or SIP Info.
<b>Payload Type</b>	The Payload type for the DTMF method "RTP Relay".
<b>Echo Cancel</b>	The algorithm for cancelling echo within the voice stream. Enabling this function is recommended.
<b>Voice Activity Detection</b>	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.
<b>Default Codec</b>	Default voice codec.
<b>ULAW, ALAW, G.729, G.726 Size</b>	One RTP packet is sent out on every specified time cycle.
<b>iLBC Frames per Packet</b>	Selectable amount of frames for each RTP packet.
<b>iLBC Mode</b>	One RTP packet is sent out on every specified time cycle.
<b>Country</b>	Defines the user location to fit local Telco system requirements.
<b>Caller ID Display</b>	Select the method of Caller ID generation.
<b>3-Way Conference</b>	Enable/Disable 3 way conference. Hook-flash to start.
<b>Call Transfer</b>	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.
<b>Outgoing IP Call</b>	Enable/Disable outgoing IP call.
<b>Incoming IP Call</b>	Enable/Disable incoming IP call.

<b>RTP Timeout (sec)</b>	The timer for terminating the SIP session if the gateway is aware of the absence of an RTP stream.
<b>RTP Port</b>	Initial port number for sending RTP packets.
<b>RTP TOS</b>	Type of service value for Quality of Service.
<b>RTP TOS (lower 5 bits)</b>	The lower 5 bits of the TOS field (0~31).
<b>STUN</b>	Enable/Disable STUN (Simple Traversal of UDP through NAT). This function is used for NAT traversal.
<b>Stun Server</b>	The IP address or host name of the STUN server.
<b>UPNP</b>	Enable/Disable UPnP (Universal Plug & Play). This function is used for NAT traversal.
<b>NAT Keep Alive Time</b>	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

Dialing Plan							
Prefix:	<input type="text" value="0000"/>	Min:	<input type="text" value="4"/>	Max:	<input type="text" value="8"/>	Del:	<input type="text" value="8"/>
Add:	<input type="text"/>	IP / Domain Name:	<input type="text"/>		Protocol:	<input type="text" value="PSTN"/>	
<div>INSERT   APPEND   DELETE   UPDATE</div>							
Table Maximum: 100							
Prefix	Min-Digits	Max-Digits	Del-Digits	Add	IP / Domain Name	Protocol	
0000	4	8	8			PSTN	
<div>Apply   Cancel</div>							

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

**Prefix** Numbers defined here are used as the beginning digits of the dialing 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 dialing pattern. For example, if we dialed 81352109378 and the delete digit is 2, then the actual dialed 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 dialing pattern. For example, if 001 is in this field, the number dialed 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 dialing plan for SIP or IAX.

**[Insert]** Insert a record where the current record is located (Current record is in a different color).

**[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

Call Record							
CDR = 0							
Seq	Caller	Callee	State	Start	Ring	Talk	End
<div>Refresh</div>							

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

## 5.22 Advanced – Remote Access

Remote Access	
Telnet Access	Enable
Web Access	Enable
Accept ICMP Requests	Enable
<div>Apply Cancel</div>	

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

Virtual Server			
Use this portion to add a new entry or delete or edit an existing entry			
Service type	User Defined	Protocol	tcp
Start Port		End Port	
IP Address			
<div>INSERT APPEND DELETE UPDATE</div>			
Type	Port Start	Port End	IP Address
DMZ Setting			
DMZ	<input type="checkbox"/> Enable	DMZ IP Address	
<div>Apply Cancel</div>			

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

MAC Filter	
Use this portion to add a new entry or delete or edit an existing entry	
MAC Address	Action
<input type="text"/>	DROP
<input type="button" value="INSERT"/> <input type="button" value="APPEND"/> <input type="button" value="DELETE"/> <input type="button" value="UPDATE"/>	
MAC Address	Action
<input type="text"/>	<input type="text"/>
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>	

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

IP Filter	
Use this portion to add a new entry or delete or edit an existing entry	
IP Address	Action
<input type="text"/>	DROP
<input type="button" value="INSERT"/> <input type="button" value="APPEND"/> <input type="button" value="DELETE"/> <input type="button" value="UPDATE"/>	
IP Address	Action
<input type="text"/>	<input type="text"/>
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>	

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

Port Filter		
Use this portion to add a new entry or delete or edit an existing entry		
Port	Protocol	
<input type="text"/>	tcp	
Action		
DROP		
<input type="button" value="INSERT"/> <input type="button" value="APPEND"/> <input type="button" value="DELETE"/> <input type="button" value="UPDATE"/>		
Port	Protocol	Action
<input type="text"/>	<input type="text"/>	<input type="text"/>
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>		

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

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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](mailto:customer-service@traitel.com).

ISSUE	RECOMMENDED ACTION
<b>I forgot my TT11 VoIP Router TA log in and/or password.</b>	<ol style="list-style-type: none"><li>1. Try the default log in and password, please refer to Chapter 5.1</li><li>2. Restore the TA to its factory default settings by holding the Reset button on the back for 6 seconds or more.</li></ol>
<b>None of the LEDs are on when I turn on the TT11 VoIP Router TA.</b>	<ol style="list-style-type: none"><li>1. Check if the power cord is connected properly.</li><li>2. Check if there is AC power coming from the power outlet.</li></ol>
<b>Why can't I ping any PC on the LAN?</b>	<ol style="list-style-type: none"><li>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.</li><li>2. Verify the IP address and subnet mask are consistent between the router and the workstations.</li></ol>
<b>Why can't I dial my friend's SIP number?</b>	<ol style="list-style-type: none"><li>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.</li><li>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.</li><li>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.</li></ol>
<b>Why isn't my firmware updating?</b>	<ol style="list-style-type: none"><li>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.</li><li>2. Check the FTP address is correct.</li></ol>

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

# Appendix: Product Specification

<b>Protocol</b> <ul style="list-style-type: none"><li>IETF SIP (RFC3261)</li><li>IAX2</li></ul>	<b>Security</b> <p>HTTP 1.1 basic/digest authentication for web set up</p> <p>MD5 for SIP authentication (RFC 2069/RFC 2617)</p>	<b>NAT Traversal</b> <ul style="list-style-type: none"><li>UPnP</li><li>STUN</li></ul>
<b>Network Interface</b> <ul style="list-style-type: none"><li>RJ-45 x2, 10/100BaseT (WAN*1 &amp; LAN*1)</li></ul>	<b>Dial Methods</b> <ul style="list-style-type: none"><li>Direct IP call without SIP proxy</li><li>Dial number via SIP server</li><li>Dial URI via SIP server</li></ul>	<b>TCP/IP</b> <ul style="list-style-type: none"><li>IP/TCP/UDP/DHCP/RTP/ICMP/</li><li>HTTP/SNTP/FTP/DNS</li></ul>
<b>Phone Interface</b> <ul style="list-style-type: none"><li>RJ-11 x1, connect to analogue phone</li><li>RJ-11 x1, connect to PSTN</li></ul>	<b>Router</b> <ul style="list-style-type: none"><li>Virtual Server</li><li>Firewall</li><li>Remote</li><li>Access NAT</li><li>DHCP Server</li><li>DMZ</li></ul>	<b>Configuration</b> <ul style="list-style-type: none"><li>Web browser</li><li>Auto-provisioning system</li></ul>
<b>Call Features</b> <ul style="list-style-type: none"><li>Call Transfer (attended)</li><li>Call Forward (Busy/No Answer/Unconditional)</li><li>Call Hold/Retrieve</li><li>Call Waiting</li><li>ID Display</li><li>Anonymous Call</li><li>Anonymous Call Blocking</li></ul>	<b>Voice Quality</b> <ul style="list-style-type: none"><li>VAD (Voice Activity Detection)</li><li>CNG (Comfort Noise Generation)</li><li>AEC (Acoustic Echo Cancellation) G.168</li><li>Jitter buffer</li></ul>	<b>Firmware Upgrade</b> <ul style="list-style-type: none"><li>Web-based Interface</li><li>Auto-provisioning system</li></ul>
<b>Fax Support</b> <ul style="list-style-type: none"><li>G.711 pass-through</li><li>T.38</li></ul>	<b>QoS</b> <ul style="list-style-type: none"><li>ToS field</li></ul>	<b>Power</b> <ul style="list-style-type: none"><li>Input AC 100-120V/220-240V 50/60Hz</li><li>Output DC 12V</li></ul>
<b>Codec</b> <ul style="list-style-type: none"><li>G.711 <math>\mu</math>-law</li><li>G.711a-law</li><li>G.723.1 (5.3k) (option)</li><li>G.723.1 (6.3k) (option)</li><li>G.726</li><li>G.729a</li><li>iLBC</li></ul>	<b>Dial Signal</b> <ul style="list-style-type: none"><li>DTMF dialing</li></ul>	<b>Environmental</b> <ul style="list-style-type: none"><li>Operating temperature 0 ~ 40°C</li><li>Storage temperature: -20 ~ 60°C</li><li>Operating humidity: 20% ~ 80%</li></ul>
<b>DTMF</b> <ul style="list-style-type: none"><li>In-band DTMF</li><li>Out-of-band DTMF (RFC2833)</li><li>SIP INFO</li></ul>	<b>Tone</b> <ul style="list-style-type: none"><li>DTMF Ring</li><li>Tone</li><li>Ring Back Tone (local and remote)</li><li>Dial Tone</li><li>Busy Tone</li></ul>	<b>Physical Dimensions</b> <ul style="list-style-type: none"><li>Size: 157(l) x 121(w) x 33(h) mm</li><li>Weight: 260g</li><li>Color: Blue/White</li></ul>
<b>SIP Server Support</b> <ul style="list-style-type: none"><li>Registrar Server (set from web)</li><li>Outbound Proxy (set from web)</li></ul>	<b>IP Assignment</b> <ul style="list-style-type: none"><li>Static IP</li><li>DHCP</li><li>PPPoE</li></ul>	<b>Certification Compliance</b> <ul style="list-style-type: none"><li>FCC Part 15 Class B</li><li>CE Class B</li><li>VCCI Class B</li><li>EN60950</li></ul>