



# **TT11 VoIP router 2FXS 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 2FXS 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 Adaptor (12V DC)**

## 3.0 SUMMARY OF LED & CONNECTOR DESCRIPTION

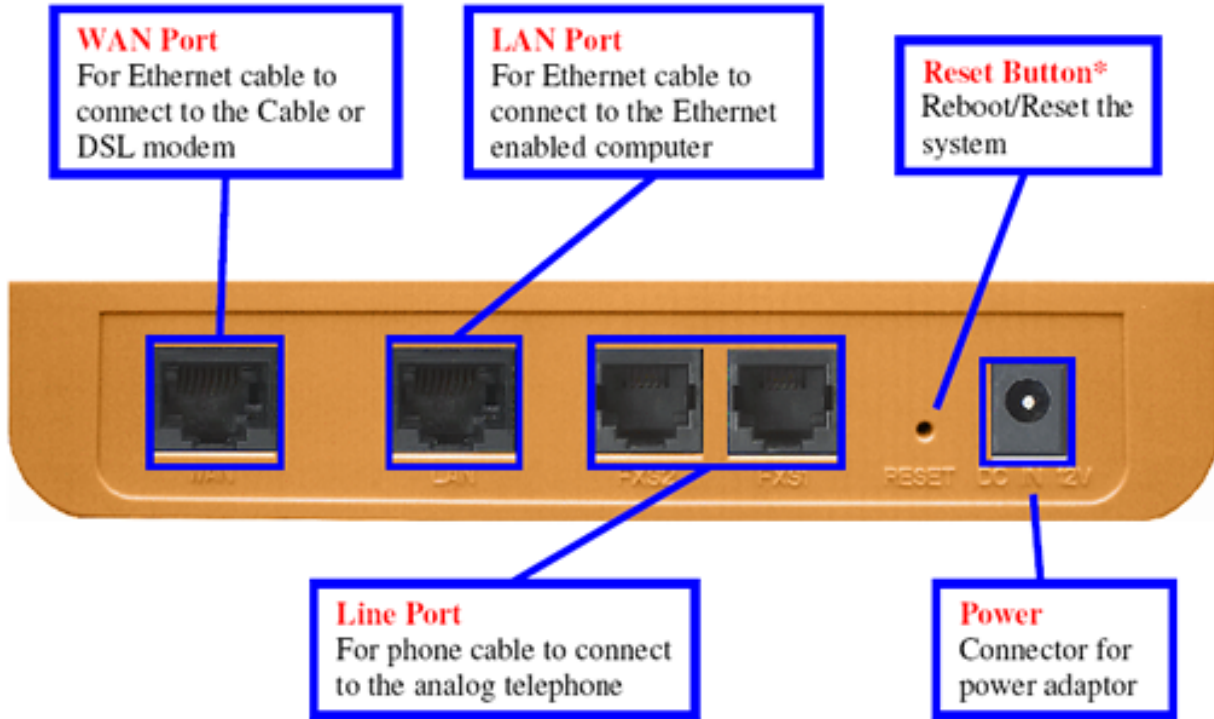
### 3.1 The Front LEDs

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	FXS	Green	On	FXS Channel is in use
			Off	FXS Channel is not in use
			Flash	FXS Channel is ringing
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*		
	Situation A	Situation B
Mode	Press and hold for less than 3 seconds	Press and hold for more than 3 seconds
Result	Reboot the system	Restore to default setting



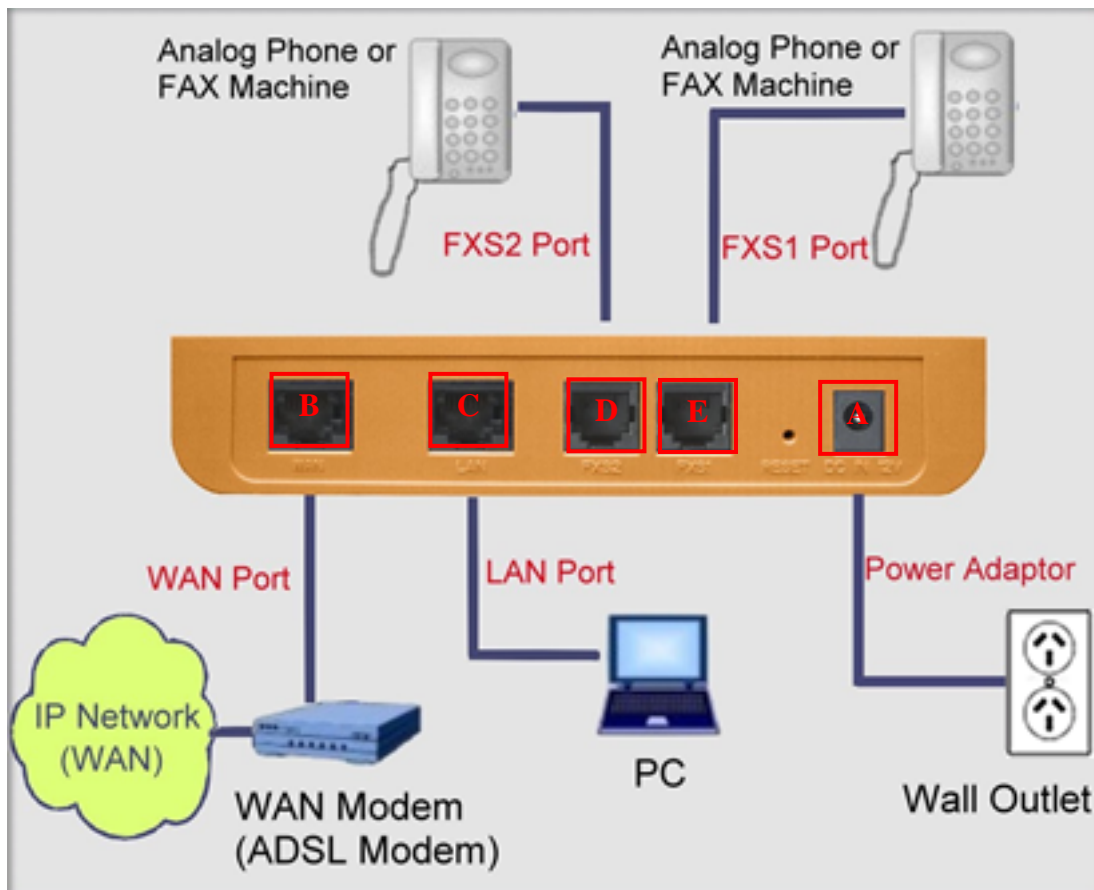
## 4.0 INSTALLATION

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### Connecting the TT11 VoIP Router 2FXS TA to Your Network

- A.** Connect the power adaptor to this jack. The Power LED will light to indicate proper operation.
- B.** Connect the Ethernet cable to the WAN Port when connecting to the Internet access device such as Cable or a DSL modem. The WAN LED will indicate the proper connection.
- C.** Connect the 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 available FXS Port when connecting to an analog telephone.
- E.** Connect the phone cable to the available FXS Port when connecting to 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 2FXS TA can be accessed using a convenient and user-friendly web browser.

If you have an account with TraiTel, please log in by going to [www.traitel.com](http://www.traitel.com) and go to “Voice” followed by “Hardware” then “Configure SIP device”. If you have multiple SIP devices, it is **highly** recommended that you create a unique sub account for each SIP device. Sub accounts can be created by going to “Your Account” main menu, followed by “Sharing and Security”.

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

### 5.1 Accessing Configuration Menu

- Open the web browser (ie. 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



The screenshot displays the web interface for the TRAI TEL Model TT11. The top left features the TRAI TEL logo and the tagline "Maximizing Solutions for a Smarter Future". The top right corner identifies the device as "Model TT11". A left-hand navigation menu lists the following categories: Status, System, Network, VoIP, and Advanced. The main content area is titled "Status" and contains the following sections:

- 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
- 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
- WAN**

Connection Type	DHCP
IP Address	192.168.10.2
Subnet Mask	255.255.255.0

Once you have logged on to the TT11 VoIP Router TA through the web browser, you can begin the setup according to your requirements. On the configuration main menu, the left navigation panel links you to the setup 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		
<b>Device Information</b>		
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	
<b>Register Status</b>		
	<a href="#">▶ SIP Register Status</a>	
	<a href="#">▶ IAX Register Status</a>	
<b>Line Status :</b>	<b>Line 1</b>	<b>Line 2</b>
	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 firmware.

#### 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 for detail of the status.

#### IAX Register Status

A hotlink to IAX Settings for detail of the status.

#### Line 1 Status

Shows line 1 register status. (FXS)

#### Line 2 Status

Shows line 2 register status. (FXS)

## 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 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

Auto-Provision	
Protocol	FTP ▼
Encryption	▼
Encryption Key	<input type="text"/>
HTTP IP	<input type="text"/>
HTTP Port	80 <input type="text"/>
FTP IP	<input type="text"/>
FTP Port	21 <input type="text"/>
FTP Timeout (sec)	0 <input type="text"/>
Username	<input type="text"/>
Password	<input type="text"/>
Firmware	<input type="text"/>
Refresh Interval (sec)	1800 <input type="text"/> ( 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 system is an advanced feature of TT11 VoIP 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 folder of your computer.

### 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". At the top, an orange header contains the title. Below it, a light blue bar contains the text: "This form allows you to backup the configuration settings to your computer, or restore the configuration from your computer." The main content is divided into two sections. The "Backup Configuration" section has a dropdown menu set to "Settings" and a "Backup" button. The "Restore Configuration" section has a text input field for the "Configuration File" with a "Browse..." button, a warning message: "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.", and a "Restore" button.

### Backup Configuration

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

### Restore Configuration

Can restore the configuration file from computer that has saved.

## 5.10 System – User Management

The screenshot shows a web interface titled "User Management". It features four rows of input fields: "Admin ID" with the value "admin", "Admin Password" with masked characters and a "Change" button, "Guest ID" with the value "user", and "Guest Password" with masked characters and a "Change" button. Below these fields is a yellow bar with the text "Restore quest's default setting" in blue. At the bottom, there 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 of 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 provided by your ISP.

### **Password**

PPPoE password.

## 5.12 Networks – Static of WAN Setting

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

Select Static as network connection type if all Wide Area Network IP is 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 of WAN Setting

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

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 to be a DHCP server for LAN side.

### 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 ending IP address for the DHCP server's IP assignment if this function is enabled.

### DHCP Client Lease Time

The time period for DHCP server to expire the IP that is assigned to.

## 5.15 Networks – SNTP Setting

SNTP Settings	
SNTP	<input type="text" value="Enable"/>
SNTP Server IP Address	<input type="text" value="clock.via.net"/> <input type="text" value="ntp.nasa.gov"/>
	<input type="text" value="tick.ucla.edu"/>
Time Zone	<input type="text" value="(GMT+10:00) Melbourne, Sydney, Guam"/>
Daylight Saving	<input type="text" value="Disable"/>
Sync Interval	<input type="text" value="0"/> seconds

Click Apply to update the modification.

### SNTP

Enable/Disable SNTP.

### SNTP Server IP Address

Enter the SNTP server IP address. SNTP server allows this gateway to synchronize the local time with remote server.

### Time Zone

Choose your time zone.

### Daylight Saving

Enable or disable the function of 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	callmedemo.7	
Number	callmedemo.7	
Username	callmedemo.7	
Password	••••••••	
Register Status	Register OK	
<b>Register:</b>		
Local Port	5060	
Outbound Proxy Server	test.sip.traitel.com.au	
Outbound Proxy Port	5060	
Send Messages via Outbound Proxy	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	
Registrar Server	test.sip.traitel.com.au	
Registrar Port	5060	
Register Fail Retry Time	300	( 10 ~ 600 sec)
Session	300	

Click Apply to update the modification.

### Display

Name displayed on the LCD for the caller. Note that TraiTel does not send this display to the caller. The user name provided under VoIP hardware configuration goes here.

<b>Number</b>	The number in the URI displayed on the LCD for the caller. Note that TraiTel does not send this number to the caller. The user name provided under VoIP hardware configuration goes here.
<b>User Name</b>	User name to log into the SIP server. The user name provided under VoIP hardware configuration goes here.
<b>Password</b>	User password to log into the SIP server. Your TraiTel account's main password goes here.
<b>Register Status</b>	The current status of the SIP Registration.
<b>Local Port</b>	Local SIP port number of TA.
<b>Outbound Proxy Server</b>	Outbound Proxy server IP address.
<b>Outbound Proxy Port</b>	Port number of Outbound Proxy Server.
<b>Send Message via Outbound Proxy</b>	Enable/Disable send message via Outbound Proxy. When enable this feature, all message will be sent through the Outbound Proxy.
<b>Registrar Server</b>	SIP Register Server IP address.
<b>Registrar Port</b>	Port number of SIP Register Server.
<b>Register Fail Retry Time</b>	The periodic interval at which the device retries the SIP registration after a failure of SIP registration is occurred.
<b>Session Type</b>	Select Invite or Update to be the method for Session Timer.
<b>Session Refresher</b>	Select UAC (User agent client) or UAS (User agent server) to be the refresher for Session Timer.
<b>Session Expires</b>	The time interval in which the TA periodically refreshes SIP sessions by sending repeated INVITE or Update request, depending on session type.
<b>Register Expires</b>	The time after which the registration on SIP Registrar expires. The phone must send SIP REGISTER to keep the registration at half of the setting time.
<b>UDP Timeout</b>	Timeout for SIP requests (100 ~ 3000 ms). Set "0" to disable the feature and follow the rule in RFC 3261.

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 ▼

### UDP Retry Times

The number of times to send SIP requests. Set “0” to disable the feature and follow the rule in RFC 3261.

### Enable PRACK

A SIP method which is applied to the condition of acknowledging to the provisional responses like 180 Ringing. Select Enable for a more reliable connection.

### Anonymous

1. If DISABLE is selected, full URI and name are sent to the receiver’s phone when the user makes a phone call. The URI and name of the caller are displayed on the receiver’s phone.
2. When Full URI is selected, it uses “Anonymous” as its display name and URI when the user makes a phone call. It may display “Anonymous” or nothing on the receiver’s phone.
3. When Display Name is selected, only display name is replaced by “Anonymous” when the user makes a phone call. It may display “Anonymous” or nothing on the receiver’s phone.

### Anonymous Reject

Select Enable to reject anonymous calls.

### Redundancy Package

Enable/Disable redundancy package.

### B2B Service

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

## 5.17 VoIP – Line Settings

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

### Polarity Reversal

Enable / Disable of generation a signal of polarity reverse

### Dial Button

Allows the user to dial numbers using the web interface.

### TX Gain

Set a specific sound intensity for transmitting sound

### RX Volume

Set a specific volume intensity for receiving sound.

### Do Not Disturb

Enable to reject incoming calls or 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 IP address, URI or number registered with SIP server.

### Call Forward Timer

The time elapsed before the call is considered not answered and 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 or not 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.

## 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		
<b>Server :</b>		
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.

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

## 5.19 VoIP – Call Control

Call Control	
<b>Dialing:</b>	
Dial Timeout(sec)	1 <input type="text"/> (max 120 sec)
Ring Timeout(sec)	30 <input type="text"/> (max 120 sec)
First Digit Timeout(sec)	20 <input type="text"/>
Inter Digit Timeout(sec)	3 <input type="text"/>
DTMF Method	RTP Relay <input type="text"/>
Payload Type	101 <input type="text"/> (96 ~ 127)
<b>Voice:</b>	
Echo Cancel	Enable <input type="text"/>
Voice Activity Detection	Disable <input type="text"/>
Default Codec	G.711 u-law <input type="text"/> <a href="#">Advance Codec Settings</a>
ULAW Size(ms)	20ms <input type="text"/>
ALAW Size(ms)	20ms <input type="text"/>
G729 Size(ms)	20ms <input type="text"/>
G726 Size(ms)	20ms <input type="text"/>
iLBC Frames per Packet	1 <input type="text"/>
iLBC Mode	30ms 13.3Kbit/sec <input type="text"/>
<b>Call:</b>	
Country	Australia <input type="text"/>
Caller ID Display	DT_DR(BELLCORE FSK) <input type="text"/>
3-Way Conference	Disable <input type="text"/>
Call Transfer	Enable <input type="text"/>
Outgoing IP Call	Enable <input type="text"/>
Incoming IP Call	Enable <input type="text"/>
<b>RTP:</b>	
RTP Timeout(sec)	0 <input type="text"/>
RTP Port	41000 <input type="text"/>
RTP TOS	5 <input type="text"/>
RTP TOS(lower 5 bits)	0 <input type="text"/> range(0..31)
<b>NAT Traversed:</b>	
STUN	Disable <input type="text"/>
Stun Server	<input type="text"/>
UPNP	Disable <input type="text"/>
NAT Keep Alive Time	0 <input type="text"/> (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 callee does not pick up the phone within the set amount of time.

<b>Ring Timeout</b>	The TT11 VoIP Router TA will automatically reject a call in if the call is not picked up within the set amount of time.
<b>First Digit Timeout</b>	Specifying the maximum duration for the first digit to be keyed in.
<b>Inter Digit Timeout</b>	Specifying the maximum duration between successive digits before the dialled 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 the function is recommended.
<b>Voice Activity Detection</b>	The voice activity detection (VAD) is a component in the DSP software that examines a caller's incoming signal and determines if the signal contains significant energy and is likely to be speech rather than a click.
<b>Default Codec</b>	Default voice codec.
<b>ULAW size, ALAW size G.729 size, 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>	Define user location to fit the local Telco system requirements.
<b>Caller ID Display</b>	Selectable of the method for Caller ID generation
<b>3-Way Conference</b>	Enable / Disable 3-Way conference. Press “Hook Flash” to start this function.
<b>Call Transfer (At the end of Conference)</b>	Enable /Disable to transfer call after it hangs up from a 3-way conference. The TA 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 (ms)</b>	The timer for terminating the SIP session if the gateway is aware of the absence of 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 five 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 STUN server.
<b>UPNP</b>	Enable / Disable UPnP (Universal Plug and Play). This function is used for NAT traversal.
<b>NAT Keep Alive Time</b>	The time interval that the IP phone always sends the keep-alive packet in order to ensure NAT works properly.

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

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 is used to compare with 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 digit 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 delete digit 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 marked as 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
<input type="button" value="Refresh"/>							

Please click the Refresh button if you want to see the updated CDR.

## 5.22 Advanced – Remote Access

Remote Access	
Telnet Access	Enable ▾
Web Access	Enable ▾
Accept ICMP Requests	Enable ▾
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>	

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	<input type="text"/>	End Port	<input type="text"/>
IP Address	<input type="text"/>		
<input type="button" value="INSERT"/> <input type="button" value="APPEND"/> <input type="button" value="DELETE"/> <input type="button" value="UPDATE"/>			
Type	Port Start	Port End	IP Address
DMZ Setting			
DMZ	<input type="checkbox"/> Enable	DMZ IP Address	<input type="text"/>
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>			

Click Apply to update the modification.

### Service Select

Select the service you wish to configure.

### Protocol

Automatic when you choose service select

### Start Port

Enter the public start port number you wish to configure.

### End Port

Enter the public end port number you wish to configure.

### IP Address

Enter the IP address of a specific internal server to which requests from the specified port is forwarded.

### DMZ

Enable / Disable the function of DMZ

### 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="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="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="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 your time and effort but if the symptoms persist, please give us a call or send an email to [customer-service@traitel.com.au](mailto:customer-service@traitel.com.au).

QUESTION	RECOMMENDED ACTION
<b>I forgot my TT11 VoIP Router TA login and/or password</b>	<ol style="list-style-type: none"><li>1. Try the default login 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 of this TA 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 power cord is connected properly.</li><li>2. Check if there is proper 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 lighting on for the port that has a PC connected. If it is off, check the cables between your TT11 VoIP Router TA 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 Registrar Server Domain Name/IP address and Outbound Proxy Domain Name/IP Address (under SIP Settings in Configuration Menu). Make sure you have the right Name or IP Address.</li><li>2. Check the LED display on the front panel of TA to see if SIP Reg LED is lighting. If it is not lighted, 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 web browser). If your status is unregistered, it means you do not have a SIP account. Contact</li></ol>

	your SIP service provider to get an account.
<b>Why isn't my firmware updating?</b>	<ol style="list-style-type: none"> <li>1. Your TT11 VoIP Router 2FXS TA automatically detects for new firmware when you plug the power. If a new version is available the TA will automatically update the firmware.</li> <li>2. Check if the FTP address is correct.</li> <li>3. Check with TraiTel if firmware filename is correct.</li> </ol>
<b>Why do I get "Can't Upgrade Now" screen when I click [Apply] in the configuration menu?</b>	<ol style="list-style-type: none"> <li>1. Make sure you exit setting mode (phonebook, menu, speed dial...) before you click [Apply] in the configuration menu.</li> </ol>

# Appendix: Product Specification

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## Protocol

- IETF SIP (RFC3261)
- IAX2

## Network Interface

- RJ-45 x 2, 10/100BaseT (WAN\*1 & LAN\*1)

## Phone Interface

- RJ-11 x 2, connect to analog phone

## Call Features

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

## FAX Support

- G.711 pass- through
- T.38

## Codec

- G.711 $\mu$ -law
- G711a-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(RFC 2833)
- SIP INFO

## SIP Server Support

- Registrar Server (Setting from web)
- Outbound Proxy (Setting from web)

## Security

- HTTP 1.1 basic/digest authentication for Web setup
- 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 50/60 Hz
- 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
- Color: Blue / White

## Certification Compliance

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