VoIP Analog Telephone Adapter

VIP-158

User's manual

Version 1.00

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The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

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Revision

User's Manual for PLANET VoIP Analog Telephone Adapter:

Model: ATA

Rev: 1.00 (2007, October)

Part No. EM-VIP_158

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Chapter 1 Introduction

Overview

Based on the flexible VoIP technology platform, the PLANET Analog Telephone Adaptor products are standards-based SIP (RFC 3261) communication devices, which are widely deployed by VoIP providers of the emerging VoIP managed voice services. It also supports the most popular local services market to ensure the compatibility and IP devices cost-effectively.

Easy-to-install and simple-to-use, it eliminates the time and effort associated with complicated installation procedures. The VIP-158 supports Web-based configuration, TFTP Auto-Provisioning and TFTP Auto-Firmware Upgrade.
The VIP-158 offers and enhanced traditional the telephony communication services to home users via the existing broadband connection in the Internet or corporation network with very low cost.

With the VIP-158, SOHO users are able to save the installation cost and extend their past investments of telephones, conference and speakerphones. The VIP-158 can be the bridge between traditional analog systems and IP network with an extremely affordable investment.

Product Features

- Feature-rich telephone service over home Internet / Intranet connection
- Cost-effective, easy-to-use solution for Analog Telephone Adapter
- Web-based utility and machine configuration
- Remote administrator authentication

VoIP Features

- SIP 2.0 (RFC3261) compliant
- G.729a and G.711 voice codec
- DHCP / PPPoE / Fixed IP allocation
- SIP proxy / Peer-to-Peer communications
- VAD / CNG / Echo Cancellation / DTMF tone detection and regeneration

Package Content

The contents of your product should contain the following items:

VoIP Telephone Adapter

Power adapter

Quick Installation Guide

User's Manual CD

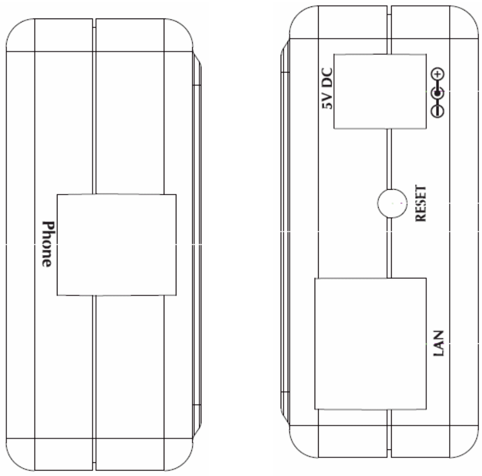
RJ-11 cable x 1

Physical Details

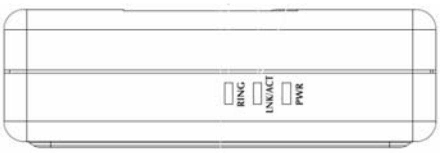
The following figure illustrates the each panel of SIP ATA.

VIP-158: SIP Analog Telephone Adapter (1 x RJ-45, 1 x RJ-11)

Front Panel of VIP-158



Left / Right Panel of VIP-158



Top Panel of VIP-158

Physical Interface & Button

1	**RESET**	Reset to the factory default setting
2	**5V DC**	5V DC Power input outlet
3	**LAN**	RJ-45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** CAT-5 cable.
4	**Phone**	RJ-11 connector, connected directly to the analog phone.

Note

Machine default IP is http://192.168.0.1. Press **RESET** button on rear panel over 5 seconds will reset the VoIP Phone Adapter to factory default value. (Except speed dial and call forward settings)

LED Display

PWR	Power is supplied to the device.
LNK/ACT	**OFF**: the device is disconnected to LAN. **ON**: the device is connected to LAN.
RING	**OFF**: the phone is idle. **ON**: the phone is in use (off-hook). **Blinking**: the phone is ringing.

Chapter 2

Preparations & Installation

Physical Installation Requirement

This chapter illustrates basic installation of ATA analog Phone Adapter (“**ATA**” in the following term)

- Network cables. Use standard 10/100Base-TX network (UTP) cables with RJ45 connectors.
- TCP/IP protocol must be installed on all PCs.

For Internet Access, an Internet Access account with an ISP, and either of a DSL or Cable modem

Administration Interface

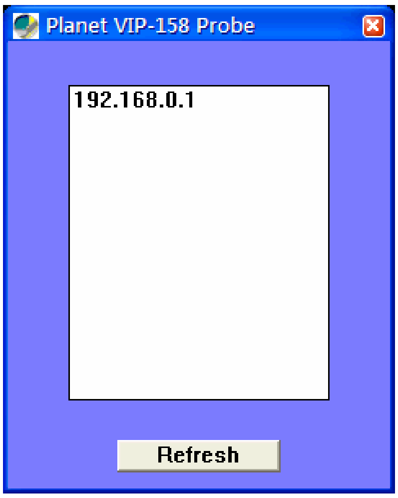
PLANET ATA provides GUI (Web based, Graphical User Interface) and utility for machine management and administration.

Utility quickly search access

Using for soft utility to search SIP ATA from current network. The utility not only easy-to-use and provides user more convenience for configuration access, at the some time If you forget this IP address can also found that via the utility.

Copy this utility tool in your laptop or desktop computer first. And, this utility tool can only be executed in Windows series of operating systems.

Click the icon for windows desktop to start searching ATA in the network.



Select “**Refresh**” and you will get the results as above choose the device you want to configuration, click this IP address of ATA to browse the web page.

Web configuration access

To start ATA web configuration, you must have one of these web browsers installed on computer for management

- Microsoft Internet Explorer 6.00 or higher with Java support

Default IP address of ATA is **192.168.0.1**. You may now open your web browser, and insert http://192.168.0.1 in the address bar of web browser to logon ATA web configuration page.

ATA will prompt for logon username/password, please enter: ***root*** / ***null*** **(no password)** to continue machine administration.

Note

```
Please locate your PC in the same network segment
(192.168.0.x) of ATA. If you're not familiar with TCP/IP,
please refer to related chapter on user's manual CD or
consult your network administrator for proper network
configurations.
```

Chapter 3

Network Service Configurations

Configuring and monitoring your ATA from web browser

The ATA integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard web browser, you can configure and check machine status from anywhere around the world.

Overview on the web interface of ATA

With web graphical user interface, you may have:

- More comprehensive setting feels than traditional command line interface.
- Provides user input data fields, check boxes, and for changing machine configuration settings
- Displays machine running configuration

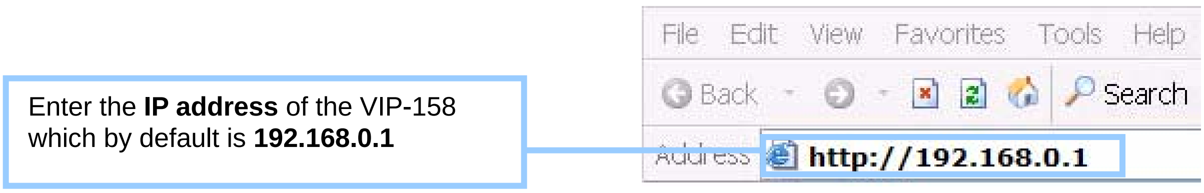
To start ATA web configuration, you must have one of these web browsers installed on computer for management

- Microsoft Internet Explorer 6.00 or higher with Java support

Manipulation of ATA via web browser

Log on ATA via web browser

After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to logon Phone Adapter web configuration page.



Phone Adapter will prompt for logon username/password: ***root*** / ***null (no password)***

ATA login prompt screen

Parameter Description

IP Address	LAN IP Address of the ATA	
	Default : 192.168.0.1	
Subnet Mask	LAN mask of the ATA	
	Default : 255.255.255.0	
Default Gateway	Gateway of the ATA	
	Default : 192.168.0.254	
DNS	DNS server of the ATA	
	Default : Null	
PPPoE Username	PPPoE user name of the ISP provider	
	Default : Null	
PPPoE Password	PPPoE pass word of the ISP provider	
	Default : Null	
PPPoE IP	System will get the IP address when your PPPoE is connection	
	Default : 0.0.0.0	
PPPoE DNS	System will get the DNS when your PPoE is connection	
	Default : 0.0.0.0	
Connect Type	Connect Type of the ATA	
	Default : STATIC	
MAC	Show the MAC address of the Ethernet interface.	
	Default : 00304Fxxxxxx	
Version	Show the firmware version	
	Default : 1.0.0	

Connection Type Description

SIATIC IP	The ISP will assign IP Address, and related information.
DHCP client	In most circumstances, it is no need to configure the DHCP settings.
PPPoE	The ISP will assign PPPoE username / password for Internet access.

After confirming the modification you've done, please click on the **SUBMIT** button to apply settings effective and the ATA will be reload page automatic by itsely, that you must to afresh enter the final modification IP address for logon web management.

Network	
IP Address	192.168.0.1
Subnet Mask	255.255.255.0
Default Gateway	192.168.0.254
DNS	0.0.0.0
PPPoE Username	
PPPoE Password	
PPPoE IP	0.0.0.0
PPPoE DNS	0.0.0.0
Connection Type	STATIC
MAC	00304F123455
Version	1.0.0

Note

Please be noticed that the Utility Tool is only designed for the LAN environment settgin. If the “**Connect Type**” is “**PPPoE**”, the Utility Tool can **NOT** find the device.

Chapter 4

VoIP Telephone Adapter Configurations

SIP Account Settings

In SIP you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the ATA. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.

SIP	
Server	192.168.0.50
Server Port	5060
SIP Port	5060
SIP Name	100
Username	100
Password	•••
Register Expires	1800
Register Alive	0
Registrar	100
STUN Enable	Disable
STUN Server	
Register Status	OK
Phone Status	IDLE

Besides, if you want to have the VoIP communication to get across the NAT, you should select the "**STUN Enable**" item to be "**Enable**". Also, you should input the IP address of the "**STUN Server**".

Parameter Description

Server	SIP Proxy Server IP address or Domain Name **Default : Null**
Server Port	Port number of SIP register Server **Default : 5060**
SIP Port	Port number of VIP-158 **Default : 5060**
SIP Name	This SIP account **Default : Null**
Username	User name of the SIP account to log into the SIP server **Default : Null**
Password	User password of the SIP account to log into the SIP server **Default : Null**
Register Expires	**Set the time re-registration** **Default : 1800 second**
NAT Keep Alive	Set the time of keeping the NAT alive **Default : 0 second**
Registrar	SIP Registration Server IP address or Domain Name **Default : Null**
STUN Enable	Set the STUN (Simple Traversal of UDP through NAT) function **Default : Disable**
STUN Server	**STUN Server IP address** **Default : Null**
Register Status	Show the registration status in Register Server **Default : IDLE**
Phone Status	Show the analog phone **Default : IDLE**

Call Control Settings

User can adjust the items of the “Call Control” when in VoIP communication. And, basically system will use the following default setting values if user does not want to change them.

Parameter Description

RTP Port	Initial port number for sending RTP packets. **Default : 4100**
RTP TOS (0 - 255)	Type of Service value for Quality of Service. **Default : 160**
Default Codec	Set the default voice codec to be “G.711 u-law”, “G.711 a-law” or “G.729A”. **Default : G7.11 u-law**
Packet Size	The size of one RTP packet (which is sent out on every specified time cycle, from 10ms to 60ms). **Default : 20ms**
Jitter Buffer	Set the jitter buffer delay time (from 10ms to 60ms) **Default : 60ms**
VAD	Set the VAD (voice activity detection) function **Default : Disable**
LEC	Set the LEC (Line Echo Cancellation) function **Default : Enable**
DTMF Command	Set the DTMF command to be “INBAND”, ”RFC2833” or “SIP-Info” **Default : INBAND**
DTMF Payload Type (96-127)	Set the DTMF Payload Type (from 96 to 127). **Default : 101**
Speaker Volume	Set the speaker volume of the analog phone in dB unit. **Default : - 8**
Mic. Gain	Set the microphone gain of the analog phone in dB unit. **Default : - 8**
Tone Volume	Set the tone volume of the analog phone in dB unit. **Default : - 5**
Tone County	Set the tone rule specification with the associated country name (CCITT, USA, BT, and Brazil). **Default : USA**

Call Control	
RTP Port	41000
RTP TOS (0~255)	160
Default Codec	G.711 u-law
Packet Size	20ms
Jitter Buffer	60ms
VAD	Disable
LEC	Enable
DTMF Command	RFC2833
DTMF Payload Type (96~127)	101
Speaker Volume	-8
Mic. Gain	-8
Tone Volume	-5
Tone Country	USA

Provision Settings

The provision feature is for user to easily use ATA by plug & play without the complicated setting procedures. (Auto provisioning system is an advanced feature of SIP ATA. For further information on using this function, please contact your ISP.)

1.) If the ISP supports the auto-provisioning system, please select the “**TFTP Provision**” item to be “**Enable**”. Otherwise, select “**Disable**”.

2.) Also, if user has got the TFTP server IP address from the ISP or network administrator, please input it in the blank of “**TFTP Server IP**”.

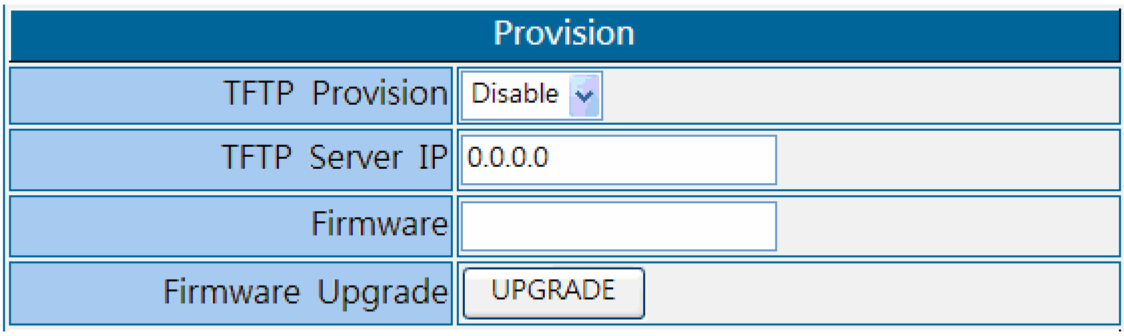
3.) If user knows the firmware filename, please fill it in blank of “**Firmware**”.

4.) After above three steps, user should select “**SUBMIT**” button to deliver the new setting values to device before the firmware upgrading.

5.) After step 4, user can upgrade the firmware by selecting the “**UPGRADE**” button.

Parameter Descriptio

TFTP Provision	Set the auto-provisioning functionality. **Default : Disable**
TFTP Server IP	Shows the IP address of the TFTP server. **Default : 0.0.0.0**
Firmware	Shows the filename of the firmware. **Default : Null**
Firmware Upgrade	Button for processing the functionality of "**Function Upgrade**".

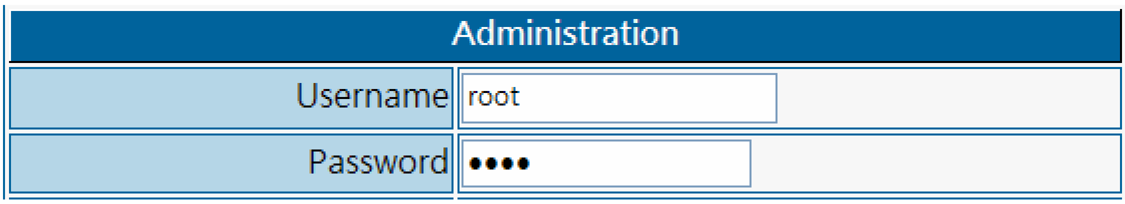


Administrator Setting

User can change the "Username" & "Password" of the login stage.

Parameter Descriptio

Username	Enter of change the username for web logging in. **Default : root**
Password	Enter of change the password for web logging in. **Default : No password (null)**



Making the setting effective

1.) Aftersetting the above all, user should select “**SUMBIT**” button to deliver the new setting.

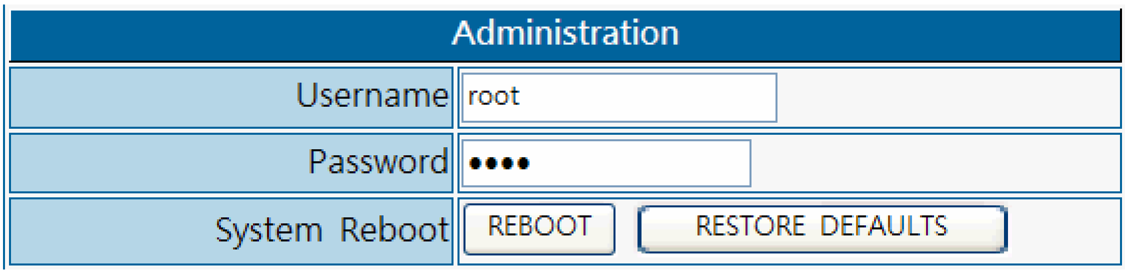
2.) From previous step, if users do not want to deliver the new setting values to the device, user can select “**RESET**” button to return the original settings.

3.) After step 1, select “**REBOOT**” button to make the settings effective.

4.) Besides, if user wants to use the default settings, please just select the “**RESTORE DEFAULTS**” button.

Parameter Descripti

REBOOT	For rebooting the system of the device with the current settings of the device
RESTORE DEFAULT	For restoring the factory default settings of the devices
SUBMIT	For delivering the setting values to the device
RESET	For resetting the setting values to the original values (when getting into this web page this time)

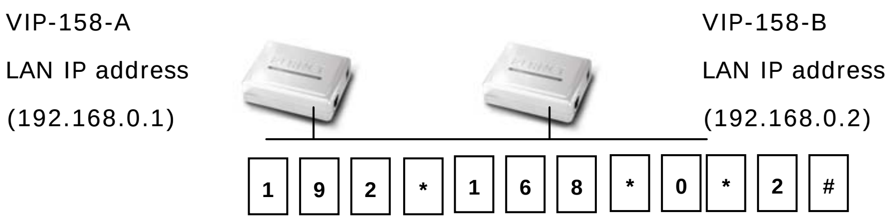


Appendix A Voice communication samples

There are several ways to make calls to desired destination in ATA. In this section, we'll lead you step by step to establish your first voice communication via keypad and web browsers operations.

Peer to peer (P2P) mode

Assuming there are two VIP-158 in the network the IP address are 192.168.0.1 and 192.168.0.2



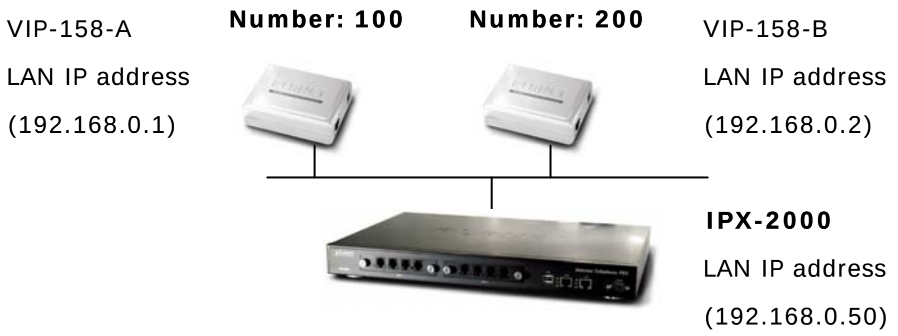
STEP :

Pick up telephone handset of VIP-158-A and dial "**192.168.0.2#**". Then the phone of VIP-158-B should ring. You can do the same thing to the VIP-158-B.

ⓘ Hint

- If the IP address of the remote calling party is known, you may directly make calls by preset number via its IP address and end with "**#**".
- If the Telephone Adapter is installed behind a NAT/firewall/ IP sharing device, please make sure the NAT device support SIP applications before making calls.

SIP Proxy mode

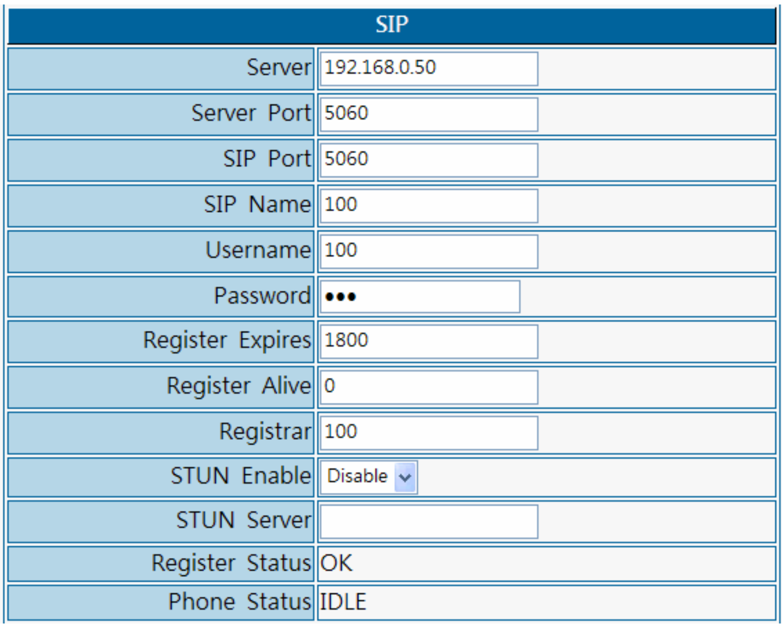


STEP 1:

Log in IPX-2000 and create two testing accounts/password: **100** / **123** (for VIP-158-A), and **200** / **123** (for VIP-158-B) for the voice calls.

STEP 2:

Please log in VIP-158-A via web browser, find to the **SIP** item. In the setting page, please insert the account/password information obtained from your service provider (in this sample, we're using PLANET IPX-2000 as the IP PBX server for SIP account, call authentications), and then the sample configuration screen is shown below:



SIP	
Server	192.168.0.50
Server Port	5060
SIP Port	5060
SIP Name	100
Username	100
Password	•••
Register Expires	1800
Register Alive	0
Registrar	100
STUN Enable	Disable
STUN Server	
Register Status	OK
Phone Status	IDLE

STEP 3:

Repeat the same configuration steps on VIP-158-B, and check the machine registration status, make sure the registrations are completed.

STEP 4:

To verify the VoIP communication, please pick up the telephone. Dial the destination number to make call between SIP clients. For example, VIP-158-A (with number 100) with keypad number 200 to VIP-158-B, or reversely makes calls from SIP client (VIP-158-B) to the number 100 (VIP-158-A).

Appendix B Frequently Asked Questions List

If your SIP ATA is not functioning properly, you can refer to this chapter first for sample troubleshooting before contacting your dealer. This can save your time and effort but if the symptoms persist, please consult your dealer.

Q: I forget my ATA login username and / or password

A:

1.) Restore VIP-158 to its factory default settings by pressing the "Reset" button which is at the back panel of the device for 5 seconds or more.

Q: Non of the LEDs are on when I turn on the SIP ATA

A:

1.) Check if power cord is connected properly.
2.) Check if there is proper AC power coming from the power outlet.

Q: Why can't I dial my friend's SIP number?

A:

1.) Check SIP Server Domain Name/IP address. Make sure you have the right Name or IP address.
2.) Check the web browser and access the configuration menu. Make sure that the SIP Server Domain Name/IP Address is correct.
3.) Check the register status under SIP Account Settings in the configuration menu (from web browser). If your status is "Idle" (unregistered), it means you do not have a SIP account. Contact your SIP service provider to get an account.

Q: Why isn't my firmware updating?

A:

1.) The VIP-158 can automatically detect the new firmware just after you plug in the power. If new version is available the ATA will automatically update the firmware.
2.) Check if TFTP server IP address is correct.
3.) Check with your supplier if firmware filename is correct.

Appendix C VIP-158 Specifications

Product	SIP Telephone Adapter
Model	VIP-158
Hardware	
LAN	1 x 10/100Mbps RJ-45 port
FXS	1 x RJ-11 connection
Protocols and Standard	
Standard	SIP 2.0 (RFC3261), STUN (RFC 3489), UPnP, MD5 for SIP authentication (RFC 2069 / RFC 2617)
Voice codec	G.729a, G.711
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) G.168: Line echo canceller (LEC) Jitter Buffer DTMF Detection and Generation In-Band and Out-of-Band (RFC 2833), (SIP INFO)
Protocols	TCP/IP, UDP, DHCP, RTP, HTTP, ICMP, ARP, DNS, TFTP, PPP, PPPoE
Network and Configuration	
Access Mode	Static IP, DHCP, PPPoE
Management	Web, Utility, Auto-provisioning (TFTP)
Dimension (W x D x H)	73 x 55 x 24 mm
Operating Environment	0~40 degree C, 10~95% humidity
Power Requirement	5V DC
EMC/EMI	CE, FCC Class B