TA1610 FXO VoIP Gateway

User Manual

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In accordance with the requirements of council directive 2002/96/EC on Waste of Electrical and Electronic Equipment (WEEE), ensure that at end-of-life you separate this product from other waste and scrap and deliver to the WEEE collection system in your country for recycling.

Contents

About This Guide

Yeastar TA1610 Analog VoIP Gateways are cutting-edge products that connect legacy telephones, fax machines and PBX systems with IP telephony networks and IP-based PBX systems. Featuring rich functionalities and easy configuration, TA1610 is ideal for small and medium enterprises that wish to integrate a traditional phone system into IP-based system. TA1610 helps them to preserve previous investment on legacy telephone system and reduce communication costs significantly with the true benefits of VoIP.

Audience

This manual will help you learn how to operate and manage your TA1610 FXO Analog VoIP Gateway. In this guide, we describe every detail on the functionality and configuration of TA1610. We begin by assuming that you are interested in TA1610 and familiar with networking and other IT disciplines.

Safety when working with electricity

- Do not open the device when the device is powered on.
- Do not work on the device, connect or disconnect cables when lightning strikes.
- Switch off the power before plugging or unplugging the cables.
- Disconnect all telecommunication network connectors and cable distribution system connectors before power off the TA1610.

Features Highlights

- 16 FXO ports
- Fully compliant with SIP and IAX2
- Flexible calling rules
- Configurable VoIP Server templates
- Codec: G711 a/u-law, G722, G723, G726, G729A/B, GSM, ADPCM
- Echo Cancellation: ITU-T G.168 LEC
- Web-based GUI for easy configuration and management
- Excellent interoperability with a wide range of IP equipment

Getting Started

In this chapter, we guide you through the basic steps to start with a new TA1610:

- Accessing Web GUI
- Web Configuration Panel
- Application Description

Accessing Web GUI

The TA1610 attempts to contact a DHCP server in your network to obtain valid network settings (e.g., the IP address, subnet mask, default gateway address and DNS address) by default.

Please enable DHCP Server in your network to obtain the TA1610 IP address.

How to check TA1610 IP address:

1. Pick up the analog phone, then access the voice menu prompt by dialing “*******”.
2. Dial "**1**" to check the IP address.
3. Dial "**2**" for web access address.

After entering the IP address in the web browser, users will see a log-in screen.

Check the default settings below:

Username: admin

Password: password

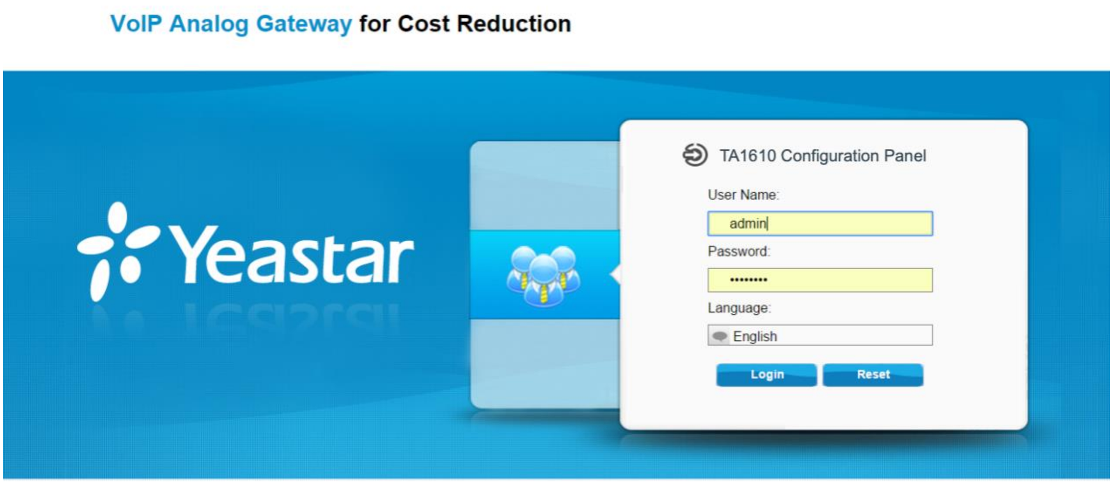


Figure 2-1 TA1610 Login page

Web Configuration Panel

There are 4 main sections on the Web Configuration Panel for users to check the TA1610's status and configure it.

- **Status:** check System Status, Port Status, Trunk Status, Network Status and check call logs, system logs.

- **System:** configure Network Settings, Security related Settings, System Date and Time, Password, Backup and Restore, etc.

- **Gateway:** configure FXO ports, gateway settings and SIP settings, etc.

- **Logout:** log out TA1610.

Note:

After saving the changes, remember to click the "Apply changes" button on the upper right corner of the Web GUI to make the changes take effect.

Application Description

Connect IPPBX and TA FXO Gateway

Yeastar TA FXO gateway is a solution to extend FXO ports for your IPPBX.

Two modes are available for you to connect IPPBX and TA FXO gateway, we call them VoIP mode and SPS (Service Provider SIP)/SPX (Service Provider IAX) mode.

Three modes are available for you to connect your SIP server and TA1610 gateway. We call them SIP Account Mode, VoIP Mode and SPS (Service Provider SIP) Mode. You can choose any one of the 3 modes to connect your SIP server and TA1610. SPS Mode is recommended.

Account Mode:

Create one SIP account on TA1610, and take the SIP account to register one SIP trunk on your SIP server. Then TA1610 and your SIP server are connected by the account.

- **Calls from SIP to PSTN**

  1) Create one outbound route on your SIP sever, and select the SIP trunk you have registered just now.
  2) Configure a “IP->Port” route on TA1610, choose the SIP account in the field “Call Source”, and choose a PSTN trunk or PSTN trunk group in the field “Call Destination”.
  3) Make a call from your SIP Server and the call should match the outbound route dial rules.

- **Calls from PSTN to SIP**

  1) Create an inbound route on your SIP server, and select the SIP trunk you have registered just now.
  2) Configure a “Port->IP” route on TA1610, choose a PSTN trunk or PSTN trunk group in the field “Call Source”, and choose the SIP account in the filed “Call Destination”.
  3) When a call comes to PSTN trunk on TA1610, the call will be routed to the destination of the SIP server inbound route.

- **Register SIP account on IP phone**

  With account mode, you can directly take the SIP account to register on your SIP phone or softphone; then make calls from softphone though PSTN trunk on TA1610 and receive incoming calls on your SIP phone or softphone. In this way, you don’t have to set up any SIP server.

VoIP Mode

Take a SIP account from your SIP server, and register it on TA1610 as a VoIP trunk. In this way, TA1610 and your SIP server are connected.

- **Calls from SIP to PSTN**

1) Configure a IP-> Port route on TA1610; choose the VoIP trunk in the field "Call Source", and choose PSTN trunk in the field "Call Destination". Enable Two-stage Dialing on the route.
2) Make a call from your SIP server, dial the SIP account number which is registered on TA1610. You will hear a dial tone, then dial the external number out through PSTN trunk.

- **Calls from PSTN to SIP**

1) Configure a Port->IP route on TA1610, choose PSTN trunk in the field "Call Source", and choose the SIP trunk in the field "Call Destination".
2) When an incoming call reaches PSTN trunk on TA1610, you will hear a dial tone, then dial an extension number of the SIP server.

SPS Mode (Recommended)

Create a Service Provider SIP trunk on TA1610 to connect to your SIP server. Add another Service Provider SIP trunk on your SIP server, connecting to TA1610.

- **Calls from SIP to PSTN**

1) Create one outbound route on your SIP sever, and select the SIP trunk you have created just now.
2) Configure a IP->Port route on TA1610, choose the SPS trunk in the field "Call Source", and choose PSTN trunk in the field "Call Destination".
3) Make a call from your SIP Server and the call should match the outbound route dial rules.

- **Calls from PSTN to SIP**

1) Configure a Port->IP route on TA1610, choose PSTN trunk in the field "Call Source", and choose the SPS trunk in the field "Call Destination".
2) Create one inbound route on your SIP server and select the SIP trunk created just now.
3) When an incoming call reaches PSTN trunk on TA41/810, You will hear a dial tone, then dial an extension number of the SIP Server, it will be routed to the destination of the SIP server inbound route.

Note: if you want the call to go directly to the destination number of your SIP server, you don't have to create an inbound route on SIP server, instead set a Hotline number on TA1610 route.

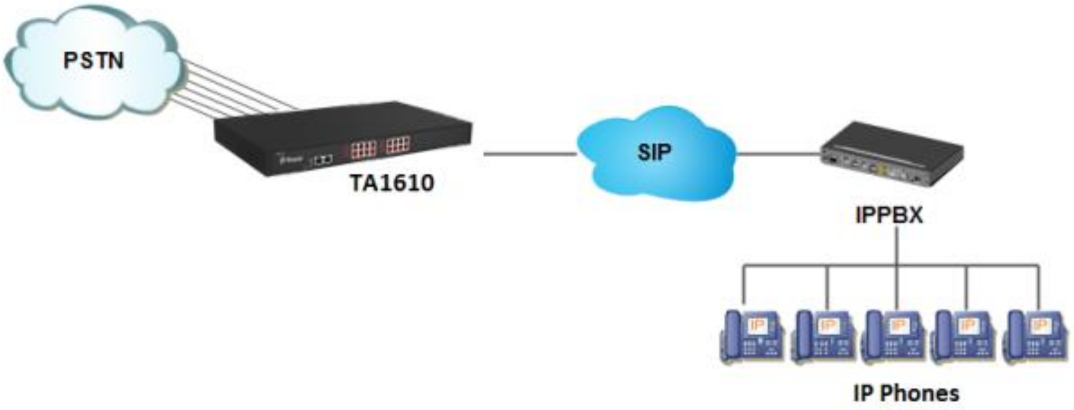


Figure 2-2 Connect IPPBX and TA FXO Gateway

For incoming calls from the PSTN to TA1610, TA1610 will forward the call to a configured SIP extension or to an inbound destination of IPPBX like IVR.

Connect TA FXO Gateway and FXS Gateway

TA FXO gateway can be connected to a FXS gateway using SPS/SPX Mode. Imagine this, the FXO gateway is set up in Site A, and the FXS gateway in Site B. People in Site B can make and receive calls using the local PSTN lines (which is connnected to Site A's provoider). With this solution, you can call a local number using a local PSTN line wherever you are.

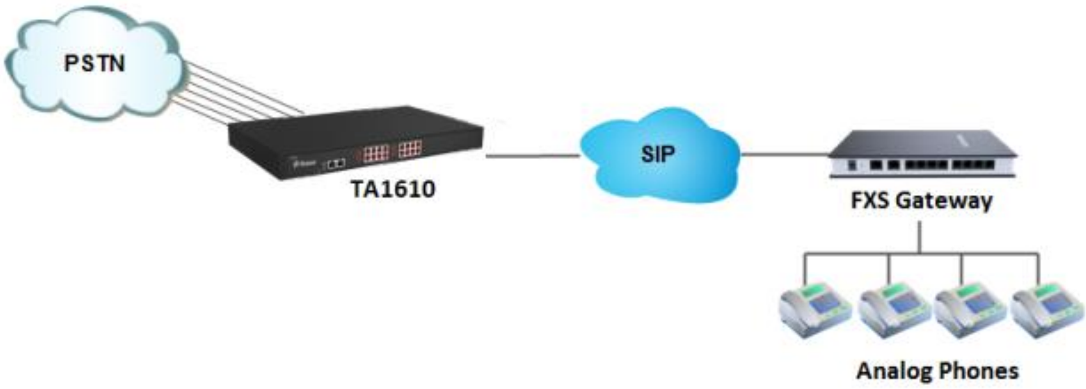


Figure 2-3 Connect TA FXO Gateway and FXS Gateway

FXO Port Settings

This chapter explains how to configure FXO port on TA1610, go to **Gateway→ Port List→ Port List** page to configure the FXO ports.

- FXO Port Settings
- Port Group

FXO Port Settings

Click "Edit" button to configure the FXO port.

1) General Settings

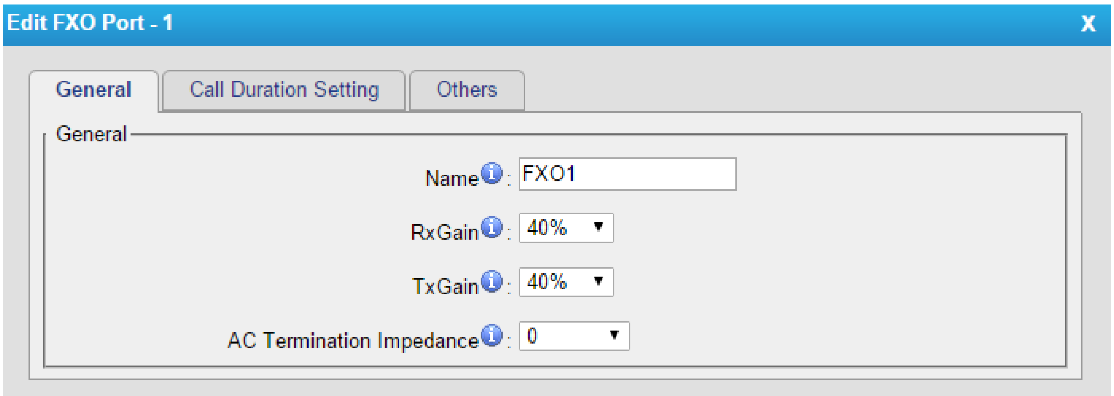


Figure 3-1 FXO Port General Settings

Table 3-1 Description of FXO Port General Settings

Items	Description
Name	The trunk Name.
RX Gain	The receive volume. The default setting is 40%.
TX Gain	The transmit volume. The default setting is 40%.

AC Termination Impedance	Select the impedance of the analog line connected to the FXO port. Here is the impedance value for the settings: 0 - 600 Ohm (North American) 1 - 900 Ohm 2 - 270 Ohm + (750 Ohm || 150nF) and 275 Ohm + (780 Ohm || 150nF) 3 - 220 Ohm + (820 Ohm || 120nF) and 220 Ohm + (820 Ohm || 115nF) 4 - 370 Ohm + (620 Ohm || 310nF) 5 - 320 Ohm + (1050 Ohm || 230nF) 6 - 370 Ohm + (820 Ohm || 110nF) 7 - 275 Ohm + (78 Ohm || 150 nF) 8 - 120 Ohm + (820 Ohm || 110 nF) 9 - 350 Ohm + (1000 Ohm || 210nF) 10 - 0 Ohm + (900 Ohm || 30nF) 11 - 600 Ohm + 2.16 uF 12 - 900 Ohm + 1 uF 13 - 900 Ohm + 2.16 uF 14 - 600 Ohm + 1 uF 15 - Global complex impedance

2) Call Duration Settings

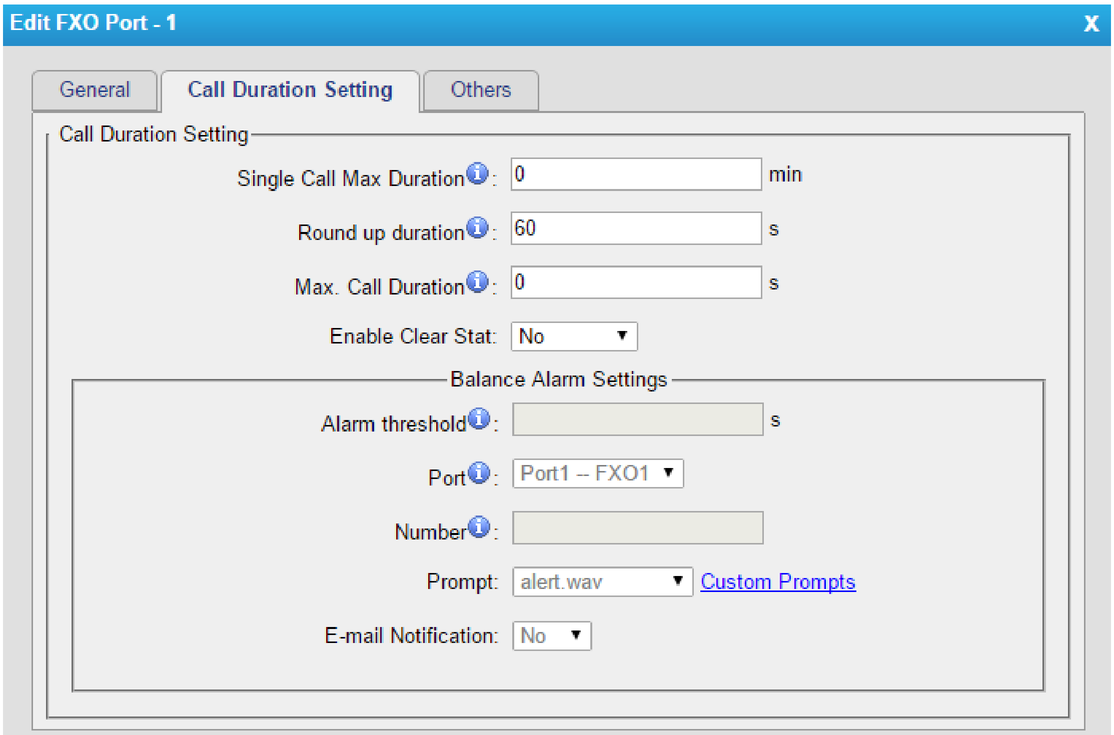


Figure 3-2 FXO Port Call Duration Setting

Table 3-2 Description of FXO Port Call Duration Settings

Items	Description
Single CallMax Duration(min)	Configure the duration of each call, it's 0 by default, which means no limit.
Round up Duration	Once the value of Billing Unit is changed, the "Round Up Duration" will be cleared, "Call Duration" will also change accordingly.
Max. Call Duration(min)	Defines the maximum number of billing unit called within a month through the trunk. To disable this limitation set the value at 0.
Enable Clear Stat.	The date to clean the duration status each month.
Balance Alarm Settings	When Max. Call Duration(min) is configured a 0 (no limit), this feature is disabled.
Alarm threshold(min)	Cofigure the time duration when TA1610 will send

	the alarm message. The value must be less than “Max Call Duration”.
Port	Choose the port to dial the alarm call.
Number	The number to receive the alarm call.
Prompt	The prompt played during the alarm call,you can customize the prompts as your wish.
E-mail	The email address to receive the alarm email. Note: please make sure SMTP test is successful in “Email settings” page before configuring this.

3) Other Settings

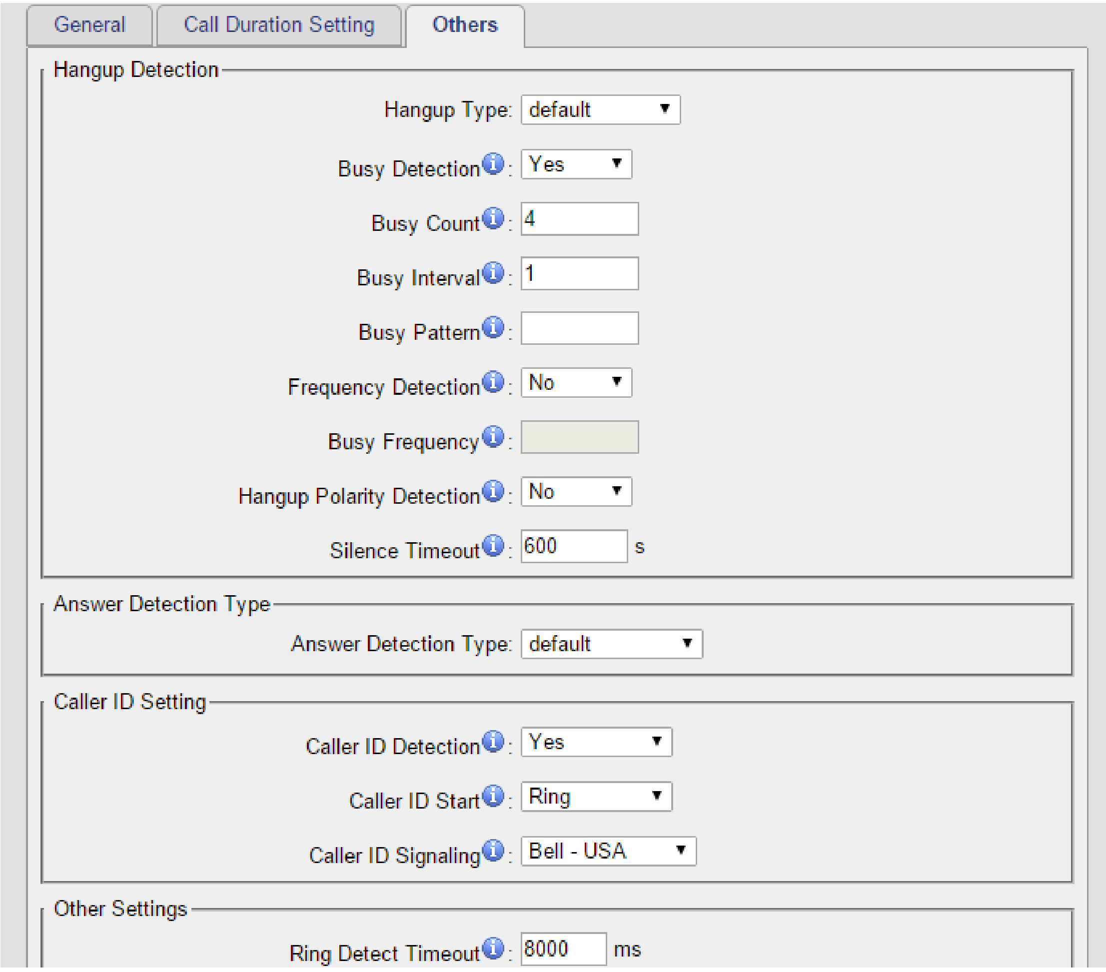


Figure 3-3 FXO Port Other Settings

Table 3-3 Description of FXO Port Other Settings

Hangup Detection	
Hangup Type	Select which kind of hangup type will be used to detect the call and hang up.
Busy Detection	Enable or disable Busy Detection. It is used for detecting far end hangup or busy signal.
Busy Count	If Busy Detection is enabled, it is also possible to specify how many busy tones to wait for before hanging up. The default is 4, but better results can be achieved if this setting is set as 6 or 8. Higher value requires more time for detection, but lower the probability that a false detection may occur.
Busy Interval	Set the busy detection interval.
Busy Pattern	If Busy Detection is enabled, you need to specify the cadence of the busy signal. If a busy pattern is not specified, the system will accept any repeating sound-silence pattern as a busy signal. If a busy pattern is specified, then the system will check the length of the sound and the silence patterns, which will further reduce the chance of a false positive.
Frequency Detection	Enable or disable Frequency Detection, it is used for frequency detection.
Busy Frequency	If Frequency Detection is enabled, you must specify the local frequency.
Hangup Polarity Detection	Enable or disable Polarity Detection. The call will be considered as “hang up” on a polarity reversal.
Silence Timeout	Define the ring out value for this port.
Answer Detection Type	
Answer Detection Type	Answer Detection settings are configured for accurate billing. Select which type to detect the call as answered. 1) Default.TA1610 will start to charge once you grab the PSTN trunk to call out, whether the call is answered or not.

	2) Polarity Detection: If the PSTN line supports polarity, you can choose "Polarity detection". When the callee answers the call, the provider will send a polarity signal, and then TA1610 starts to bill.
Custom Ring Tone	Enable or disable Custom Ring Tone. If the custom ring tone is enabled, you need to configure the following settings according to the ringback signal.
Max Ring Duration	Max duration of the ring tone.
Max Ring Interval Duration	Max pause between the two ring tones.
Min Ring Detection	Enable Min Ring Detection, which is useful for complex situations, like when jitter or noise occurs on the PSTN line. Generally it is disabled.
Min Ring Duration	Min duration of the received tone.
Min Ring Interval Duration	Min pause between the two received tones.
Caller ID Setting	
Caller ID Detection	Enable or disable caller ID detection.
Caller ID Start	This option allows one to define the start of a caller ID signal. Ring: start to detect when a ring is received Polarity: start to detect when a polarity reversal is started Before Ring: start to detect before a ring tone
Caller ID Signaling	This option defines the type of caller ID signaling to use. Bell-USA: US standard V23-UK: UK standard V23-Japan: Japanese standard V23-Japan Pure: Japanese standard

	DTMF: DTMF signal Please check with your PSTN service provider to configure Caller ID Settings. If you don't know how to configure, please contact Yeastar support.
Other Settings	
Ring Detect Timeout	There should be a timeout to determine if there is a hang up before the line is answered. Range from 3000 to 8000. Default is 8000 ms.

Port Group

Port group is a feature that allows you to define specific PSTN trunks to a group. A trunk group can be used in a route. When a call is coming or going through the route, an available trunk would be selected in the trunk group. There are two ring strategies supported for Port Group:

- Round-Robin: select the next available port in line.
- Least Used: select the port that is least used.

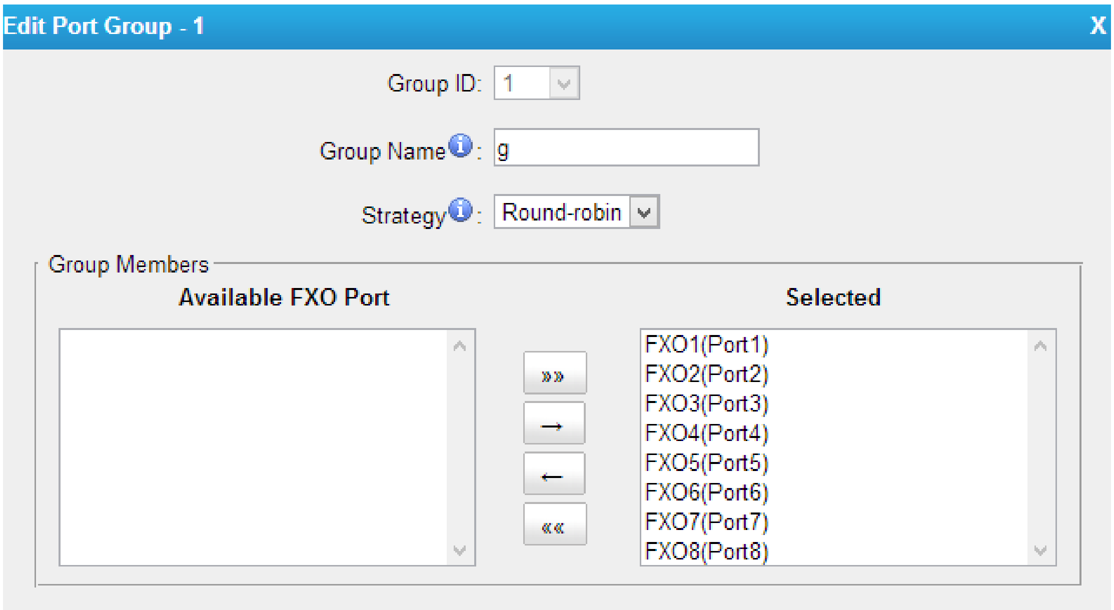


Figure 3-4 Port Group

VoIP Settings

To integrate with other IPPBX, we need to configure the VoIP settings in TA FXO Gateway to set up VoIP trunk (SIP and IAX). In this chapter, we introduce the following settings:

- VoIP Trunk
- Trunk Group
- SIP Settings
- IAX Settings

VoIP Trunk

There are 3 types of trunks listed in this page, Account, Trunk and Service Provider.

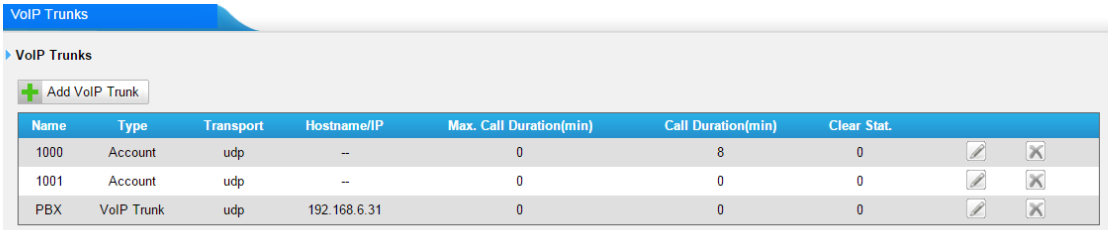


Figure 4-1 VoIP Trunk

1) Account

It's an SIP account created in TA1610 so that the other devices can register SIP trunk at their side using these information.

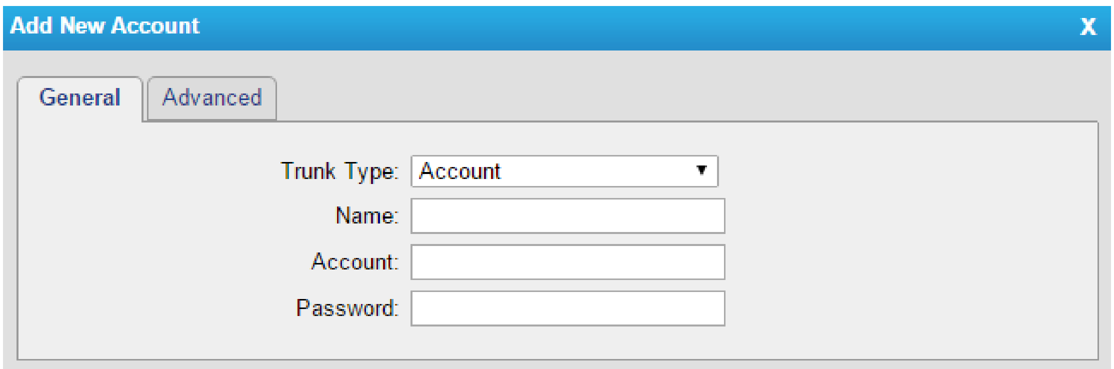


Figure 4-2 Account

Table 4-1 Description of Account Settings

Items	Description
Trunk Type	Choose the type of trunk, “Account”.
Name	Define the name.
Account	Define the Account number.
Password	Set a password for this account.

2) VoIP Trunk

It’s a SIP trunk configured in TA1610 to register to the SIP provider, please make sure this trunk works properly in advance with provider before configuring TA1610.

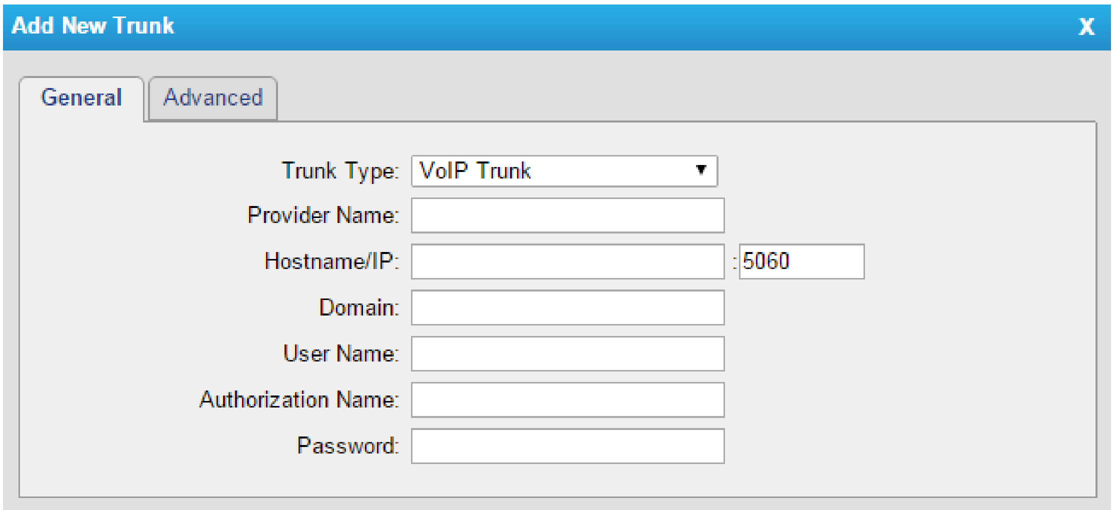


Figure 4-3 VoIP Trunk Settings

Table 4-2 Description of VoIP Trunk Settings

Items	Description
Trunk Type	Choose the type of trunk, “VoIP Trunk”.
Provider Name	A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. E.g. “yeastar”.
Hostname/IP	Service provider’s hostname or IP address. **Note:** 5060 is the standard port number used by SIP

	protocol. Don't change this part if it is not required.
Domain	VoIP provider's server domain name or IP address.
User Name	User name of SIP account provided from the SIP Server provider.
Authorization Name	Authorization Name of SIP account provided from the SIP Server provider.
Password	Password of the SIP account.

3) Service Provider

This is service provider trunk (peer to peer mode) which authorized using IP address only.

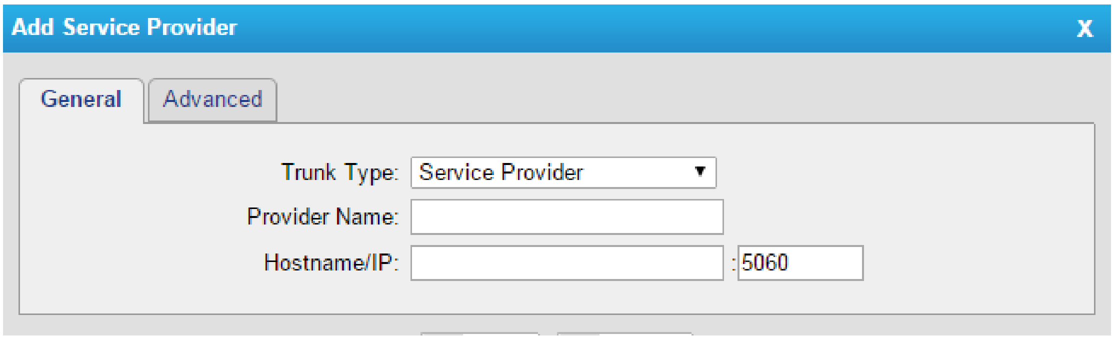


Figure 4-4 Service Provider Trunk Settings

Table 4-3 Description of Service Provider Trunk Settings

Items	Description
Trunk Type	Choose the type of trunk, "Service Provider".
Provider Name	A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. E.g. "yeastar".
Hostname/IP	Service provider's hostname or IP address. **Note:** 5060 is the standard port number used by SIP protocol. Don't change this part if it is not required.

Trunk Group

Trunk group is a feature that allows you to define specific SIP trunks to a group. A trunk group can be used in a route. When a call is coming or going through the route, an available trunk would be selected in the trunk group.

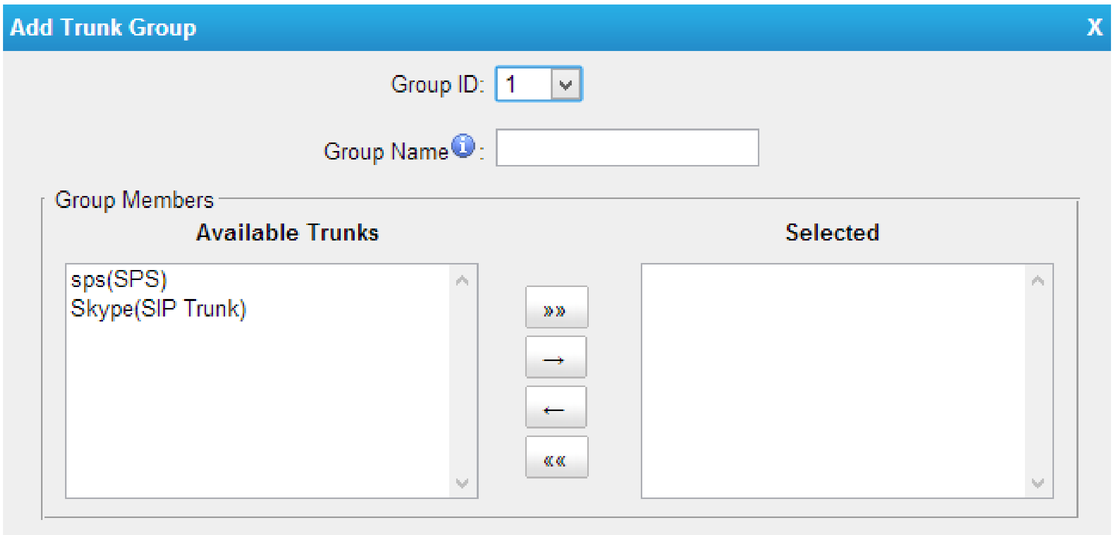


Figure 4-5 Trunk Group

SIP Settings

It is wise to leave the default setting as provided on this page. However, for a few fields, you need to change them to suit your situation.

1) General

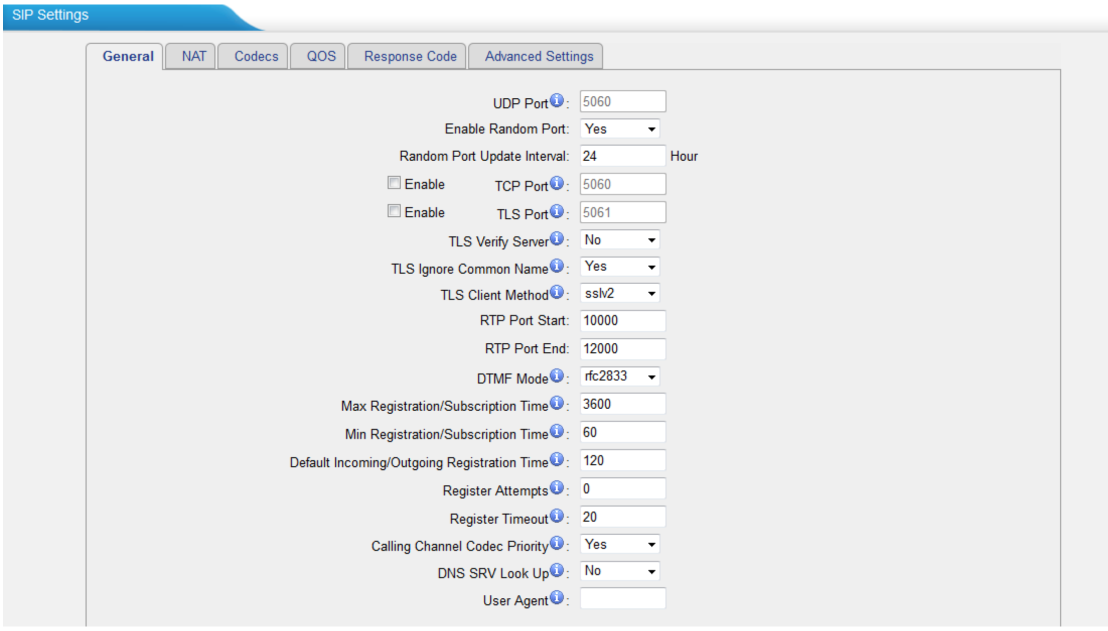


Figure 4-6 SIP General Settings

Table 4-4 Description of SIP General Settings

Items	Description
UDP Port	Port used for SIP registrations. The default is 5060.
Enable Random Port	Enable or Disable Random SIP port.
Random Port Update Interval	Set the Random Port Update Interval.
TCP Port	Port used for SIP registrations. The default is 5060.
TLS Port	Port used for SIP registrations. The default is 5061.
TLS Verify Server	When using TA FXO Gateway as a TLS client, whether or not to verify server’s certificate. It is “No” by default.
TLS Verify Client	When using TA FXO Gateway as a TLS server, whether or not to verify client’s certificate. It is “No” by default.
TLS Ignore Common Name	Set this parameter as “No”, then common name must be the same with IP or domain name.
TLS Client Method	When using TA FXO Gateway as TLS client, specify the protocol for outbound TLS connections. You can select

	it as tlsv1, sslv2 or sslv3.
RTP Port Start	Beginning of the RTP port range.
RTP Port End	End of the RTP port range.
DTMF Mode	Set the default mode for sending DTMF. Default setting: rfc2833
Max Registration/Subscription Time	Maximum duration (in seconds) of a SIP registration. The default is 3600 seconds.
Min Registration/Subscription Time	Minimum duration (in seconds) of a SIP registration. The default is 60 seconds.
Default Incoming/Outgoing Registration Time	Default Incoming/Outgoing Registration Time: the default duration (in seconds) of incoming/outgoing registration.
Register Attempts	The number of SIP REGISTER messages to send to a SIP Registrar before giving up. The default is 0 (no limit).
Register Timeout	Number of seconds to wait for a response from a SIP Registrar before classifying the register has timed out. The default is 20 seconds.
Calling Channel Codec Priority	Once enabled, when dialing out via SIP/SPS trunks, the codec of calling channel will be selected preferentially. If not, TA FXO Gateway will follow the priority order in your SIP/SPS trunks.
Video Support	Support SIP video or no. The default is yes.
Max Bit Rate	Configure the max bit rate for video stream. The default: 384kb/s.
DNS SRV Look Up	Please enable this option when your SIP trunk contains more than one IP address.
User Agent	To change the user agent parameter of asterisk, the default is “TA FXO Gateway”; you can change it if needed.

2) NAT

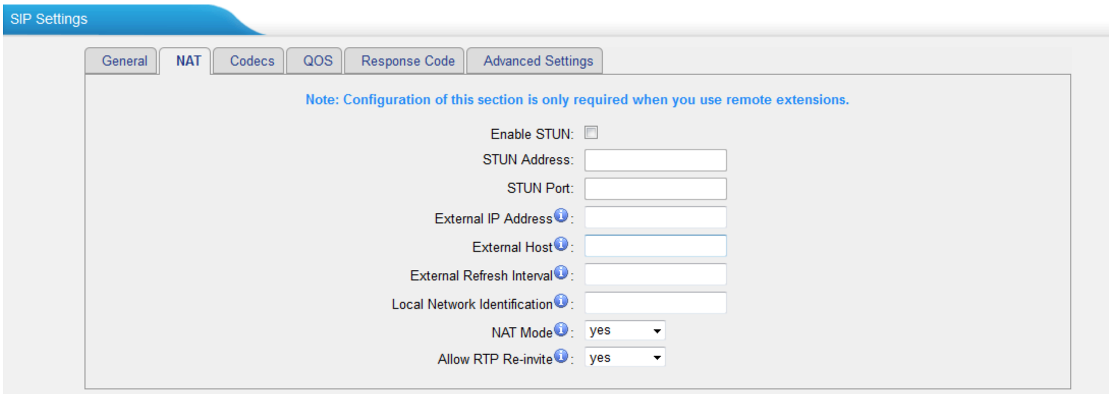


Figure 4-7 NAT Settings

Table 4-5 Description of SIP NAT Settings

Items	Description
Enable STUN	STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.
STUN Address	The STUN server allows clients to find out their public address, the type of NAT they are behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between the client and the VOIP provider and so establish a call.
External IP Address	The IP address that will be associated with outbound SIP messages if the system is in a NAT environment.
External Host	Alternatively you can specify an external host, and the system will perform DNS queries periodically. This setting is only required when your public IP address is not static. It is recommended that a static public IP address is used with this system. Please contact your ISP for more information.
External Refresh Interval	Used to identify the local network using a network number/subnet mask pair when the system is behind a NAT or firewall. Some examples of this are as follows:

	"192.168.0.0/255.255.0.0": All RFC 1918 addresses are local networks; "10.0.0.0/255.0.0.0": Also RFC1918; "172.16.0.0/12":Another RFC1918 with CIDR notation; "169.254.0.0/255.255.0.0": Zero conf local network. Please refer to RFC1918 for more information.
NAT Mode	Global NAT configuration for the system; the options for this setting are as follows: Yes = Use NAT. Ignore address information in the SIP/SDP headers and reply to the sender's IP address/port. No = Use NAT mode only according to RFC3581. Never = Never attempt NAT mode or RFC3581 support. Route = Use NAT but do not include rport in headers.
Allow RTP Reinvite	By default, the system will route media steams from SIP endpoints through itself. Enabling this option causes the system to attempt to negotiate the endpoints to route packets to each other directly, bypassing the system. It is not always possible for the system to negotiate endpoint-to-endpoint media routing.

3) Codecs

We can choose the allowed codec in TA1610, a codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet. For more information about codec, you can refer to this page: http://en.wikipedia.org/wiki/List_of_codecs

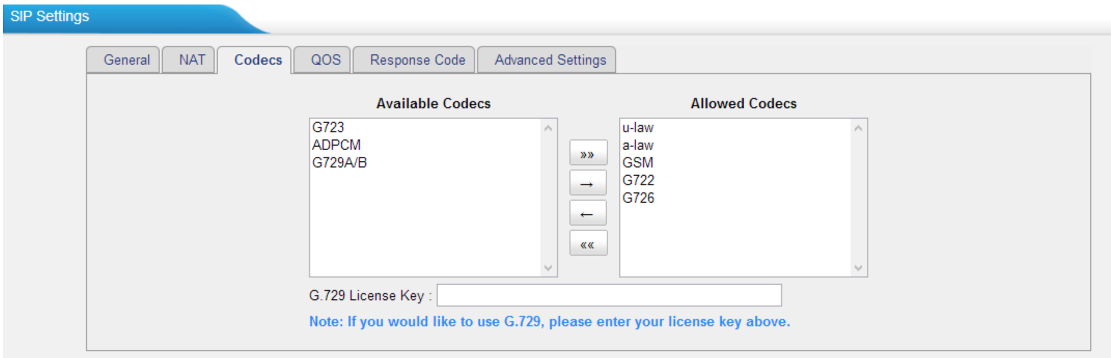


Figure 4-8 Codecs

If you want to use codec G729, we recommend buying a license key and input it here.

4) Qos

QoS (Quality of Service) is a major issue in VoIP implementations. The issue is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due interference from other lower priority traffic. When the network capacity is insufficient, QoS could provide priority to users by setting the value.

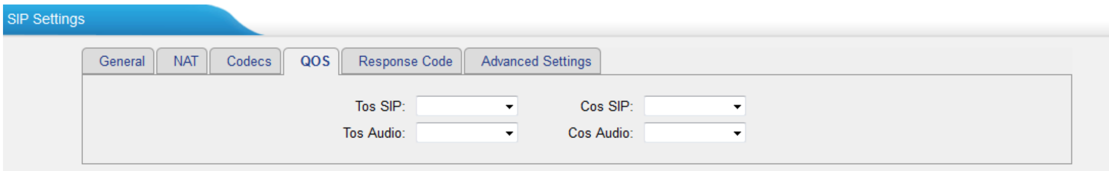


Figure 4-9 Qos

Note: It's recommended that you configure the QoS in your router or switch instead of TA FXO Gateway side.

5) Response Code

You can change the response code on TA FXO Gateway to the one you want before sending it to the VoIP server. It helps the VoIP server understands better the exact call status, like busy, no response and others.

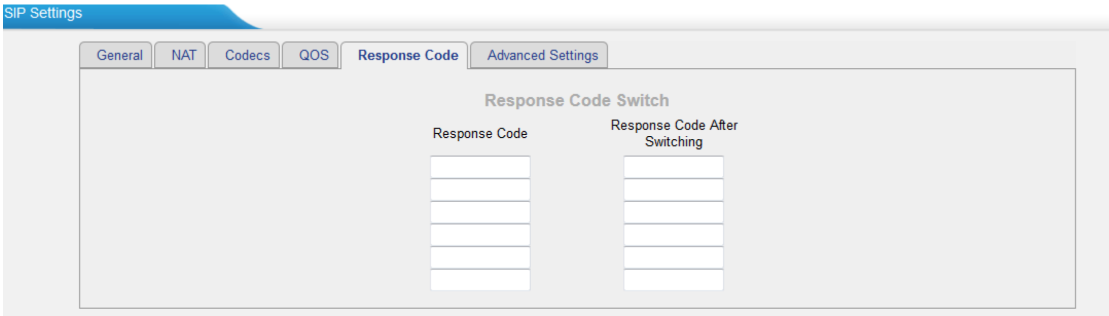


Figure 4-10 Response Code

Note: we don't' recommend configuing this if you are not familiar with the code of call status from the VoIP server.

6) Advanced Settings

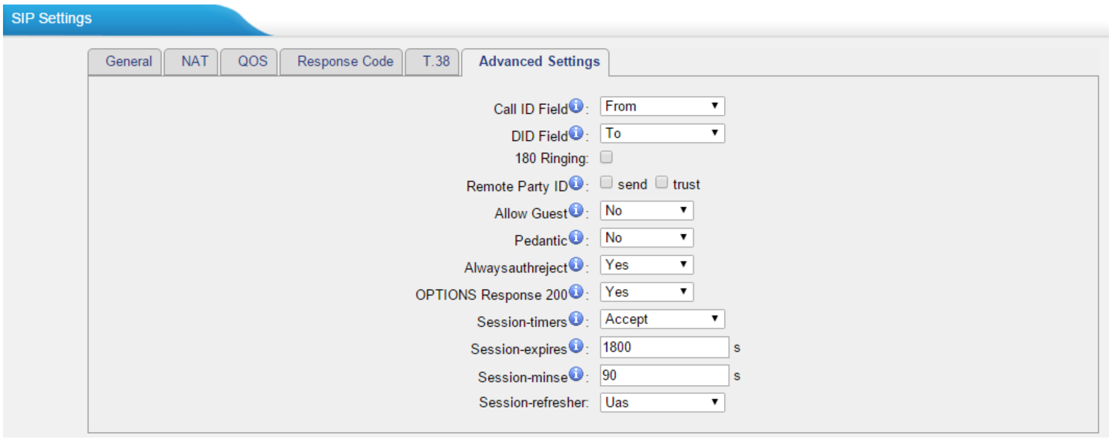


Figure 4-11 SIP Advanced Settings

Table 4-6 Description of SIP Advanced Settings

Items	Description
Call ID Field	Where to get the caller ID in SIP packet.
DID Field	Where to get the DID in SIP packet.
180 Ringing	It is set when the telecom provider needs. Usually it is not needed.
Remote Party ID	Whether to send Remote-Party-ID on SIP header or not. Default: no.
Allow Guest	Whether to allow anonymous registration extension or not. Default: no. It’s recommended that it is disabled for security reason.
Pedantic	Enable pedantic parameter. Default: no.
Alwaysauthreject	If enabled, when TA FXO Gateway rejects “Register” or “Invite” packets, TA FXO Gateway always respond the packets using “SIP404 NOT FOUND”. It’s recommended that it is enabled for security reason.
OPTIONS Response 200	If set to yes, the response to an OPTIONS is always 200OK.

Session-timers	Enable session-timer mode, default: yes. If you find the call is cut off every 15 minutes every time, please disable this.
Session-expires	The max refresh interval
Session-minse	The min refresh interval, which mustn't be shorter than 90s.
Session-refresher	Choose the session-refresher, the default is Uas.

IAX Settings

IAX is the Internal Asterisk Exchange protocol, you can connect to TA FXO Gateway or register IAX trunk to another IAX server. It's supported by the asterisk-based IPPBX.

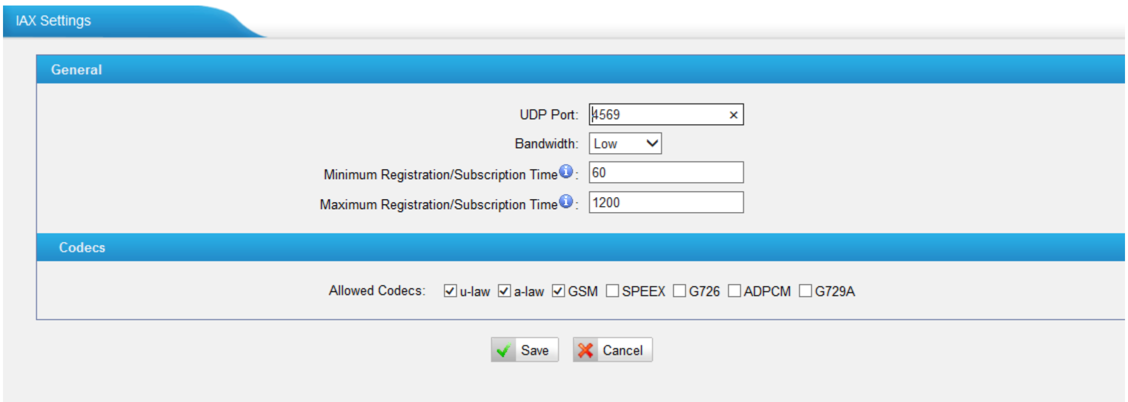


Figure 4-12 IAX Settings

Table 4-7 Description of IAX Settings

Items	Description
UDP Port	Port used for IAX2 registrations. Default is 4569.
Bandwidth	Low/medium/high with this option you can control which codec to be used.
Minimum Registration	Minimum duration (in seconds) of an IAX2 registration.

Time/Subscription Time	Default is 60 seconds
Maximum Registration Time/Subscription Time	Maximum duration (in seconds) of an IAX2 registration. Default is 1200 seconds.
Codecs	Enable the codec you want for IAX communication.

Routes Settings

After connecting Yeastar TA1610 gateway with the VoIP server, you need to configure the routes settings on TA1610 to route the calls through the gateway. In this chapter, we introduce the following sections:

- IP->Port
- Port->IP/Port
- Blocklist
- Callback Settings

IP->Port

Configure IP->Port routes to control calls from your SIP server to TA1610 FXO ports.

Click “Edit” to check the route details, there are two modes for you.

1) Simple Mode

Choose “Yes” for Simple Mode, the simple mode configuration page appears as below.

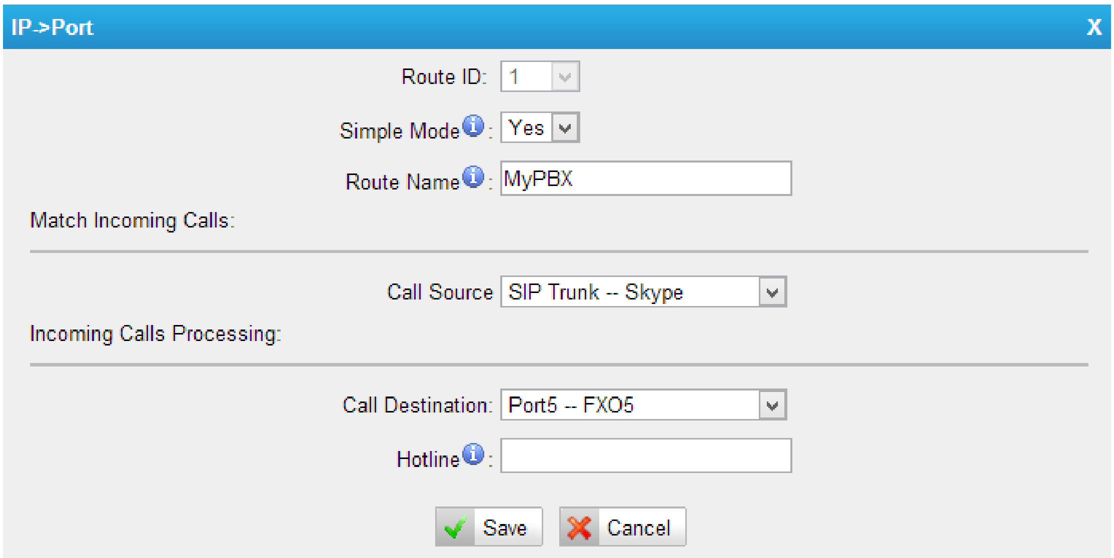


Figure 5-1 Simple Mode Route

Table 5-1 Description of Simple Mode Route

Items	Description
Route Name	Define the route name.
Call Source	Choose the trunk or trunk group for the incoming calls.
Call Destination	Choose the trunk or trunk group to route the incoming calls to.
Hotline	Dial the number directly, The dial pattern is ignored.

2) Detail Mode

Choose “No” for Simple Mode, you will see the detailed configuration page as the following picture shows. Detailed settings for **Match Incoming Calls** and **Handle Matched Incoming Calls** are provided in Detailed Mode.

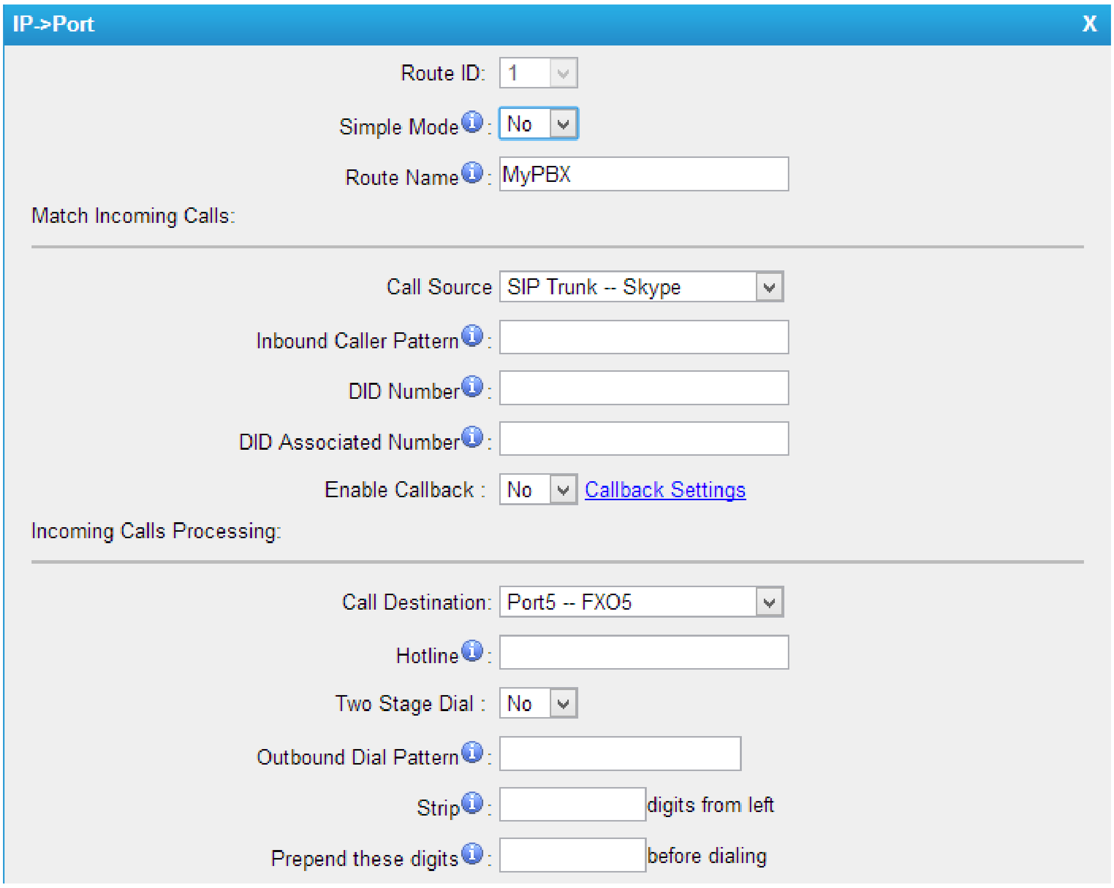


Figure 5-2 Detailed Mode Route

Table 5-2 Description of Match Incoming Calls Settings

Items	Description
Call Source	Choose the trunk or trunk group for the incoming calls.
Inbound Caller Pattern	Match the prefix of caller ID for incoming calls.
DID Number	Define the expected DID Number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. You can also use pattern matching to match a range of numbers.
DID Associated Number	Define the extension for DID number. You can input number and "-"in this field, and the format can be xxx or xxx-xxx. The count of the number must be only one or equal the count of the DID number.

Table 4-13 Description of Handle Matched Incoming Calls Settings

Items	Description
Call Destination	Choose the trunk or trunk group to route the incoming calls to.
Hotline	Direct number to the SIP Server. The parameter is ignored if a SIP Account is selected on this route.
Two-stage Dialing	Enable or Disable Two-stage Dialing.
Outbound Dial Pattern	Outbound calls that match this dial pattern will use this outbound route.
Strip	Allows the user to specify the number of digits that will be stripped from the front of the phone number before the call is placed. For example, if users must press 0 before dialing a phone number, one digit should be stripped from the dial string before the call is placed.
Prepend	These digits will be prepended to the phone number before the call is placed. For example, if a trunk requires 10-digit dialing, but users are more comfortable with 7-digit dialing, this field could be used to prepend a 3-digit area code to all 7-digit phone numbers before calls are placed.

Port->IP/Port

Port->IP/Port routes are used to control incoming calls to PSTN trunks on TA1610 and route the calls to your SIP server or another PSTN trunk on TA1610.

Click “Edit” to check the route details, there are two modes for you.

1) Simple Mode

Choose “Yes” for Simple Mode, the simple mode configuration page appears as below.

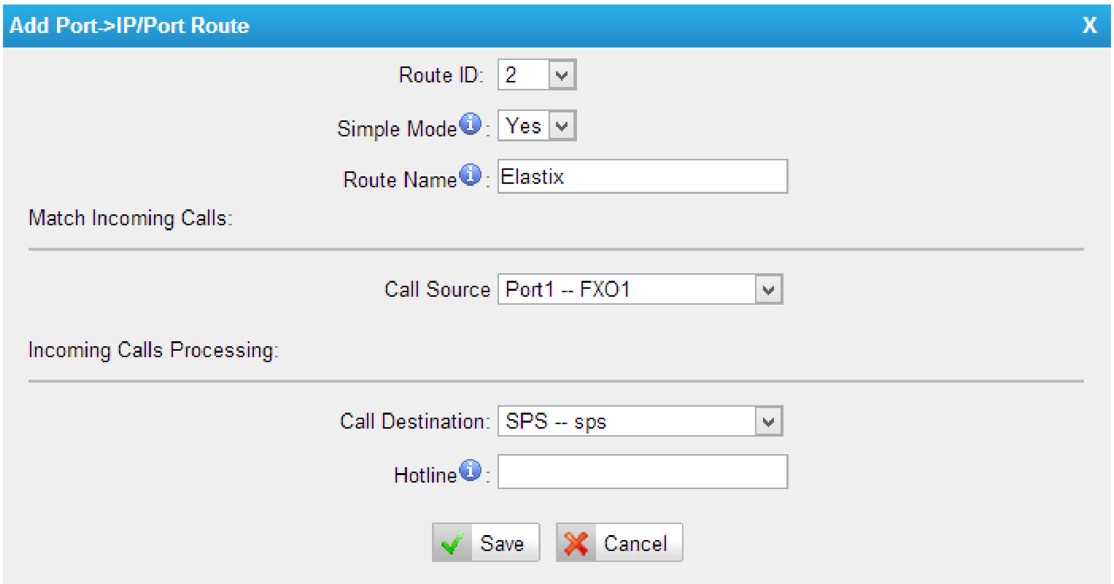


Figure 5-3 Simple Mode Route

Table 5-3 Description of Simple Mode Route

Items	Description
Route Name	Define the route name.
Call Source	Choose the trunk or trunk group for the incoming calls.
Call Destination	Choose the trunk or trunk group to route the incoming calls to.
Hotline	Dial the number directly, The dial pattern is ignored.

2) Detail Mode

Choose “No” for Simple Mode, you will see the detailed configuration page as the following picture shows. Detailed settings for **Match Incoming Calls** and **Handle Matched Incoming Calls** are provided in Detailed Mode.

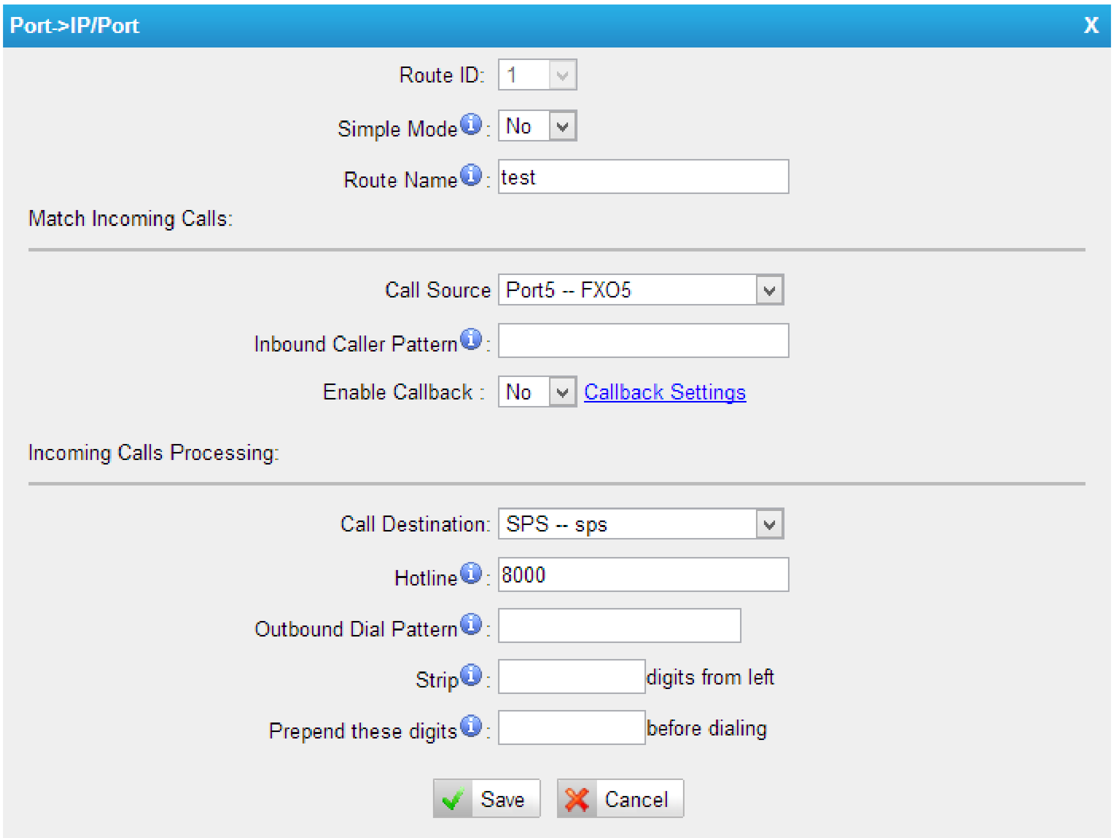


Figure 5-4 Detailed Mode Route

Table 5-4 Description of Match Incoming Calls Settings

Items	Description
Call Source	Choose the trunk or trunk group for the incoming calls.
Inbound Caller Pattern	Match the prefix of caller ID for incoming calls.
Enable Callback	Wether to enable callback feature.

Table 5-5 Description of Handle Matched Incoming Calls Settings

Items	Description
Call Destination	Choose the trunk or trunk group to route the incoming calls to.
Hotline	Direct number to the SIP Server. The parameter is ignored if a SIP Account is selected on this route.
Outbound Dial Pattern	Outbound calls that match this dial pattern will use this outbound route.
Strip	Allows the user to specify the number of digits that will be stripped from the front of the phone number before the call is placed. For example, if users must press 0 before dialing a phone number, one digit should be stripped from the dial string before the call is placed.
Prepend	These digits will be prepended to the phone number before the call is placed. For example, if a trunk requires 10-digit dialing, but users are more comfortable with 7-digit dialing, this field could be used to prepend a 3-digit area code to all 7-digit phone numbers before calls are placed.

Blocklist

Blocklist is used to block an incoming or outgoing call. If the number of incoming or outgoing call is listed in the number blocklist, the caller will hear the following prompt: “The number you have dialed is not in service. Please check the number and try again”. The system will then disconnect the call.

You can add a number with the type: inbound, outbound or both.

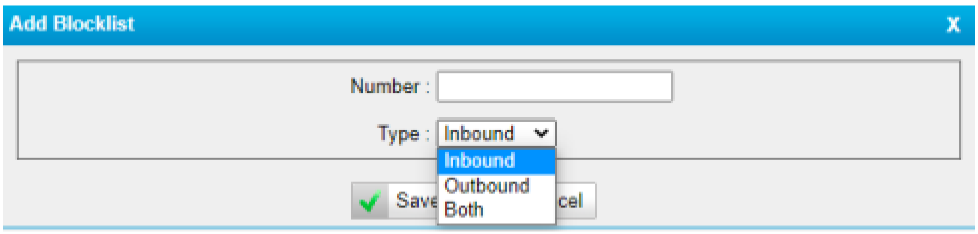


Figure 5-5 Blocklist

Callback Settings

1) If you'd like to use callback feature, please make sure it's enabled on the IP->Port or Port->IP/Port route setting panel.
2) No callback rules needed to be set if the trunk supports call back with the caller ID directly.
3) Add Callback numbers, then callback will work for the added callback numbers. Tick "Allow All Numbers", callback feature will work for all numbers.

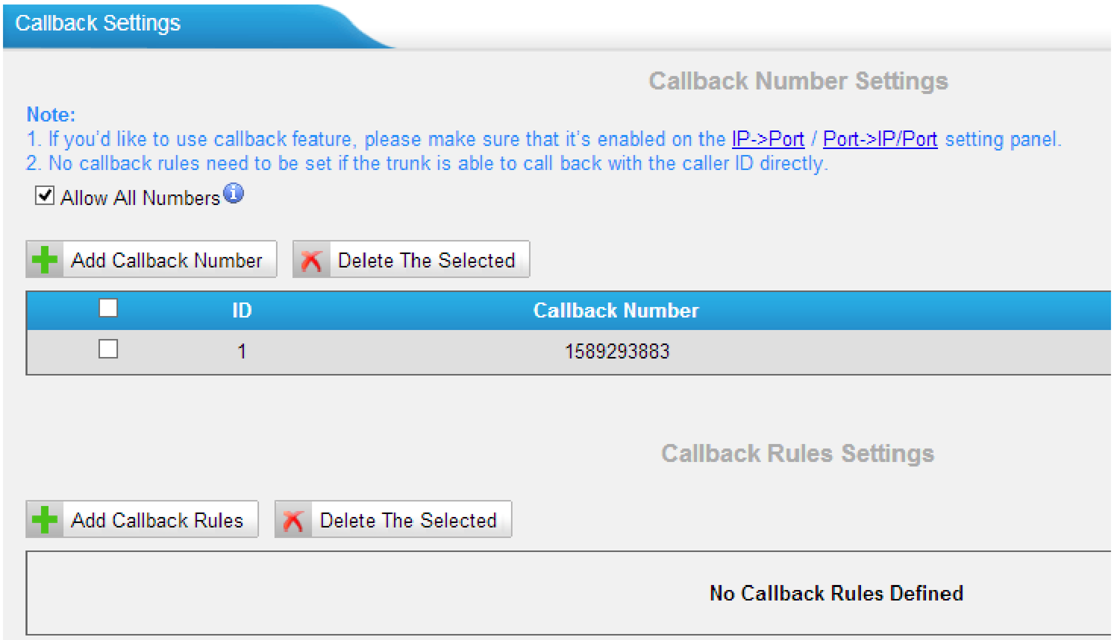


Figure 5-6 Callback Settings

Gateway Settings

This chapter explains Gateway settings, which can be applied globally to TA1610. The gateway settings can be configured under **Gateway→ Gateway Settings**.

- General Preferences

General Preferences

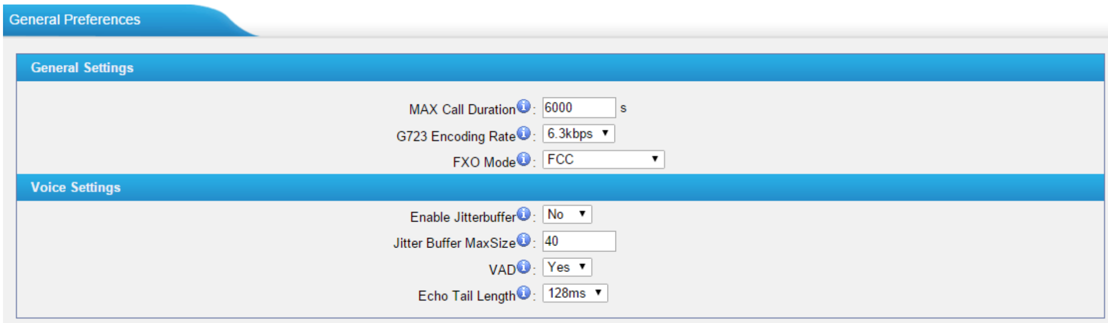


Figure 6-1 General Preferences

Table 6-1 General Preferences

General Settings	
MAX Call Duration	The absolute maximum amount of time permitted for a call. A setting of 0 disables the timeout.
G723 Encoding Rate	Set the G723 encoding rate.
FXO Mode	Select country to set the On Hook Speed, Ringer Impedance, Ringer Threshold, Current Limiting, TIP/RING voltage adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country's analog line characteristics. The default setting is "FCC".
Voice Settings	
Enable Jitterbuffer	Forces the use of a jitter buffer on the received side of a SIP channel. The call quality will be improved if this

	option is enabled.
Jitter Buffer MaxSize	Max length of the jitter buffer in milliseconds. Default: 40.
VAD	Voice Activity Detection.
Echo Tail Length	In some cases, the echo canceller doesn't train quickly enough and there is echo at the beginning of the call which then quickly fades out.

Audio Settings

This chapter explains prompt settings on TA1610.

- Custom Prompts

Custom Prompts

We can upload the prompts in this page; you can also download it and save it as a backup.

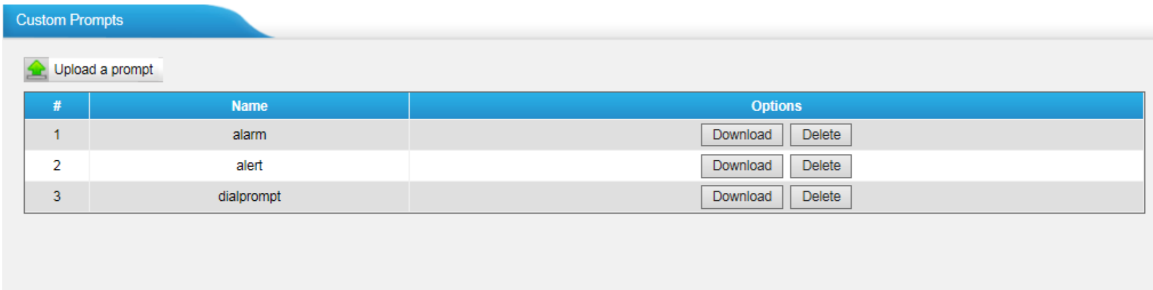


Figure 7-1 Custom Prompts

The administrator can upload prompts by doing the following:

1) Click “Upload Prompt”.

2) Click “Browse” to choose the desired prompt.

3) Click “Upload” to upload the selected prompt.

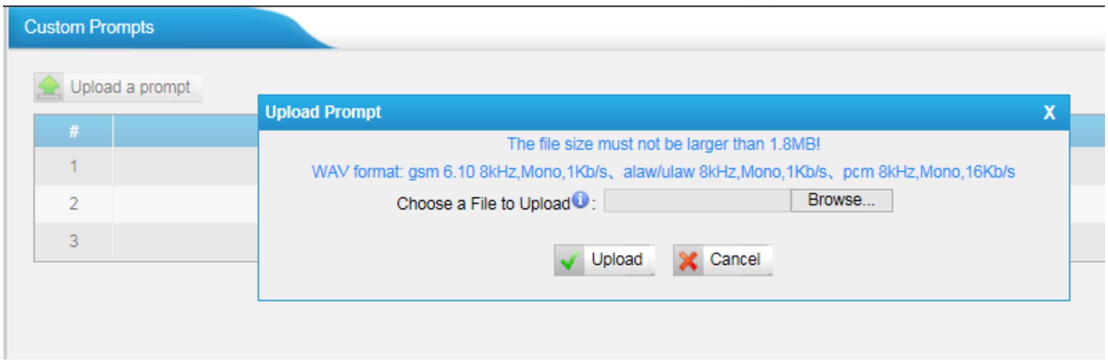


Figure 7-2 Upload A Prompt

Note: The file size must not be larger than 1.8 MB, and the file must be WAV format:

GSM 6.10 8 kHz, Mono, 1 Kb/s;

Alaw/Ulaw 8 kHz, Mono, 1 Kb/s;

PCM 8 kHz, Mono, 16 Kb/s.

Advanced Settings

This chapter explains SIP settings and Distinctive Ringtones.

- Tone Zone Settings
- DTMF Settings

Tone Zone Settings

Advanced ring tones for all the FXO ports can be configured on this page. There are pre-grogrammed tone zone settings for some countries and regions. Users can simply find and select thier country to get tone zone settings for the gateway.

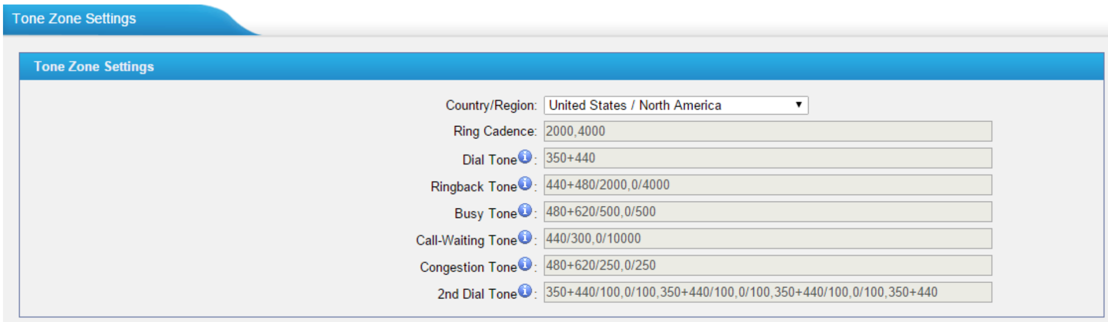


Figure 8-1 Tone Zone Settings

Users may also configure the tone zone according to the national standard by selecting "User custom for Tone Zone". Please refer to the document below and configure the tone zone settings on TA FXO Gateway:

http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf

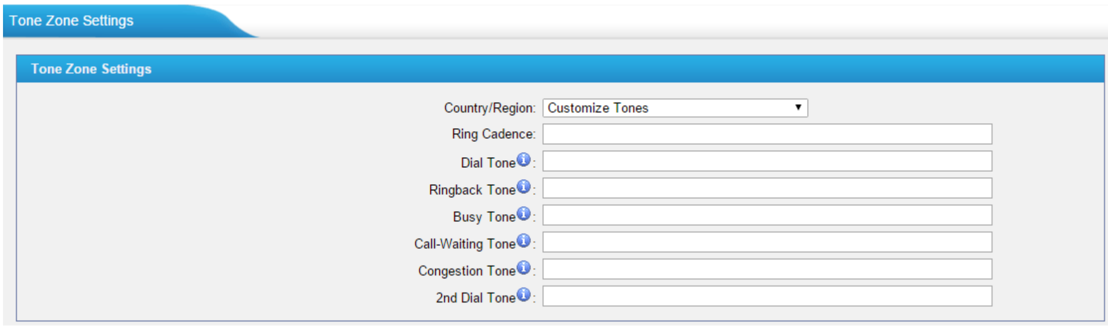


Figure 8-2 Customize Tones

Table 8-1 Description of Tone Zone Settings

Items	Description
Country/Region	Choose the country to get pre-programmed tone zone settings or choose "User custom for Tone Zone" to configure the settings manually.
Ring Cadence	Configuration option for all FXO ports ring cadence for all incoming calls.
Dial Tone	Prompt tone of off-hook dial tone.
Ringback Tone	The tone sent to caller when ringing is on.
Busy Tone	Used for busy line prompt.
Call-Waiting Tone	Used for notification in call waiting.
Congestion Tone	Used to indicate that an invalid code has been dialed, or that all circuits (trunks) are busy and/or the call is unroutable.
2nd Dial Tone	Used for the second stage dial tone.

DTMF Settings

DTMF signal sent from TA1610 to the receiver can be set on this page.

Digit Length and Dial Pause Between Digit: 100.100 (ms)

Default Digit Volume: -10,-10 (dB)

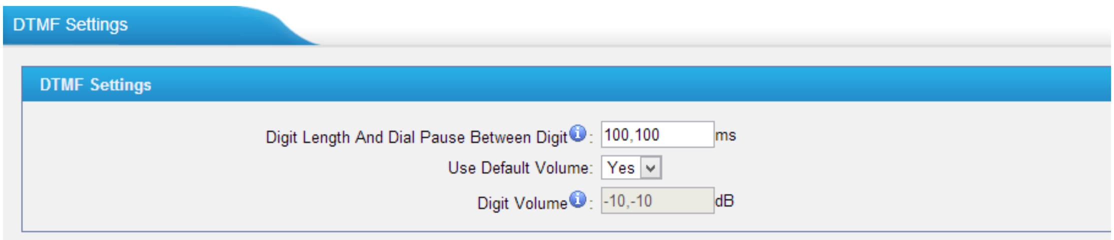


Figure 8-3 DTMF Settings

Network Preferences

This chapter explains network settings on TA1610. Click the main menu  on the top of the Web GUI to check the network settings.

- LAN Settings
- Service
- VLAN Settings
- VPN Settings
- DDNS Settings
- Static Route

LAN Settings

After successfully logging in the TA1610 Web GUI for the first time, users could go **System→Network Preferences→LAN Settings** to configure the network for TA1610.

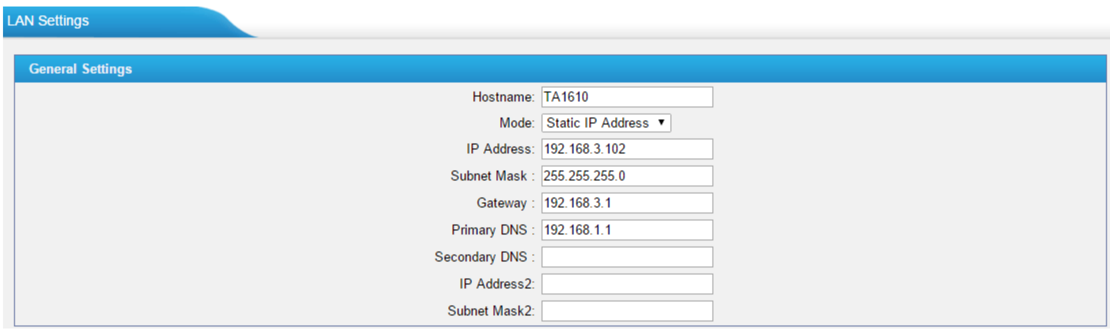


Figure 9-1 LAN Settings

Table 9-1 LAN Settings

Items	Description
Hostname	Set the host name for TA1610.
Mode	Choose the network mode: • Static IP Address • DHCP • PPPoE
IP Address	Set the IP Address for TA1610.
Subnet Mask	Set the subnet mask for TA1610.
Gateway	Set the gateway for TA1610.
Primary DNS	Set the primary DNS for TA1610.
Secondary DNS	Set the secondary DNS for TA1610.
IP Address2	Set the second IP Address for TA1610.
Subnet Mask2	Set the second subnet mask for TA1610.

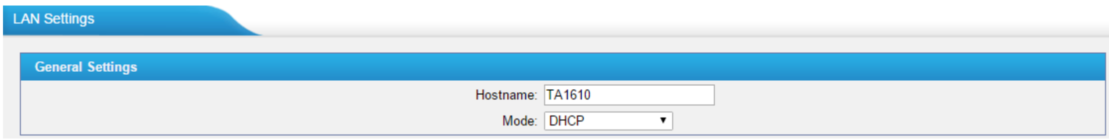


Figure 9-2 DHCP Mode

Select DHCP mode to get network automatically from the local network.

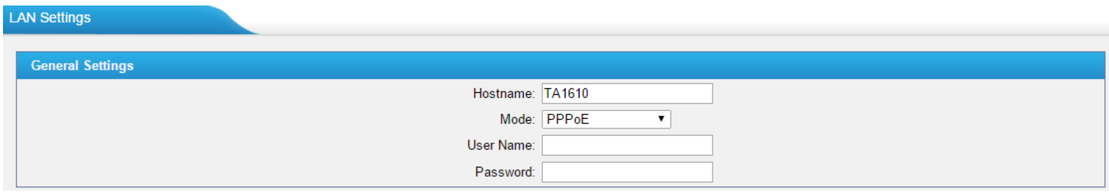


Figure 9-3 PPPoE

Fill in user name and password to access the Internet via PPPoE.

WAN Settings

WAN port is disabled by default. If you want to use dual network, you should enable WAN port, and configure the network settings and static routes to route network traffic through the proper Ethernet interface.

Note: TA gateway does not act as a router to route the network traffic from WAN port to LAN port.

Go to **System→Network Preferences→WAN Settings** to configure the WAN port of TA1610.

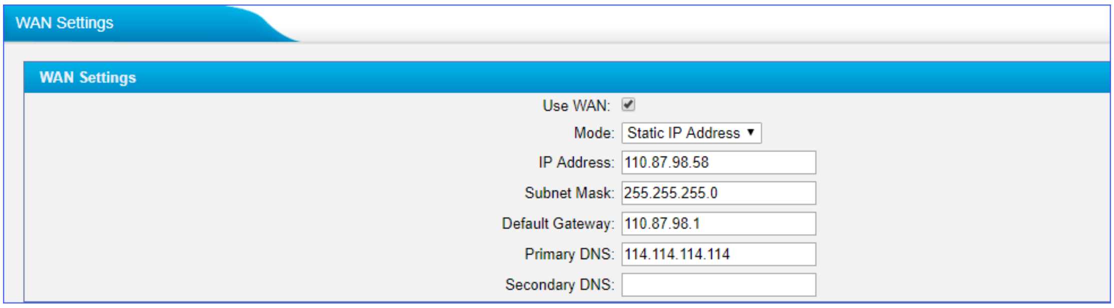


Figure 9-4 WAN Settings

Table 9-2 WAN Settings

Items	Description
Use WAN	Whether to enable WAN port or not. Note: If WAN port is enabled, the default network interface is WAN.
Mode	Choose the network mode: • Static IP Address • DHCP • PPPoE
IP Address	Set the IP Address of the public network.
Subnet Mask	Set the subnet mask of the public network.

Default Gateway	Set the gateway of the public network.
Primary DNS	Set the primary DNS of the public network.
Secondary DNS	Set the secondary DNS of the public network.

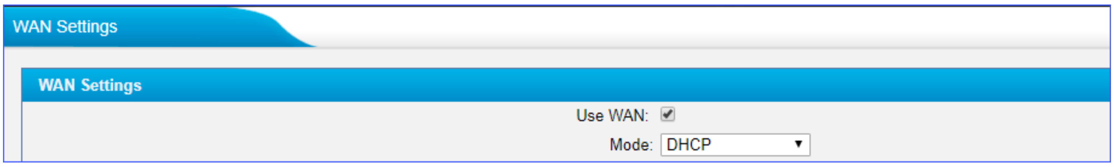


Figure 9-5 DHCP Mode

Select DHCP mode to get network automatically from the local network.

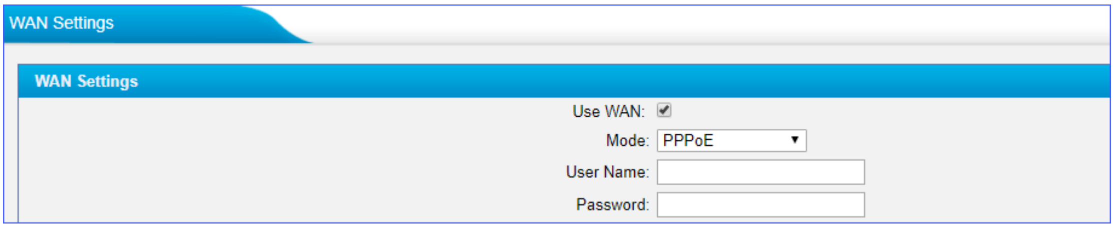


Figure 9-6 PPPoE

Fill in user name and password to access the Internet via PPPoE.

Service

The administrator can manage all the access methods on TA on the "Service" page.

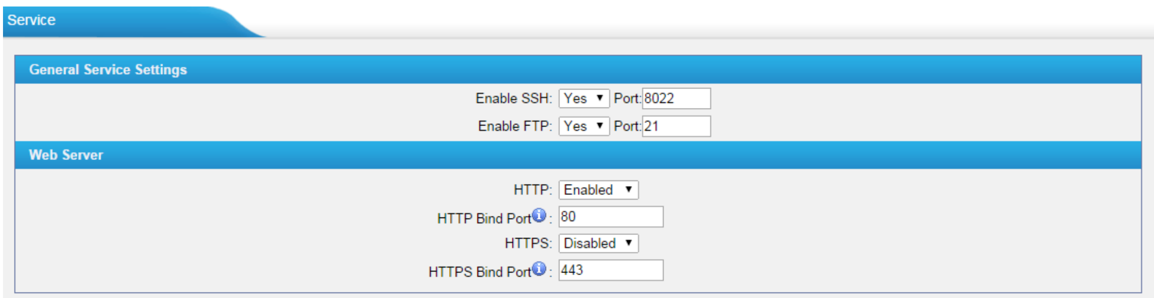


Figure 9-7 Service Settings

Table 9-3 Description of Service Settings

Items	Description
SSH	By using SSH, you can log in to TA1610 and run commands. It's disabled by default. We don't recommend enabling it if not needed. The default port for SSH is 8022.
FTP	FTP access; The default port is 21.
HTTP	HTTP web access; The default port is 80.
HTTPS	HTTPS web access, it is disabled by default, and you can enable it to get safer web access.

VLAN Settings

VLAN (Virtual Local Area Network) is a group of hosts with a common set of requirements, which communicate as if they were attached to the same broadcast domain, regardless of their physical location.

A VLAN is a broadcast domain created by switches. This means the VLAN is configured on switches, layer 3 switches. Note that some of the switches don't support VLAN.

Note:

TA1610 acts as a VLAN client, a 3-layer switch is needed.

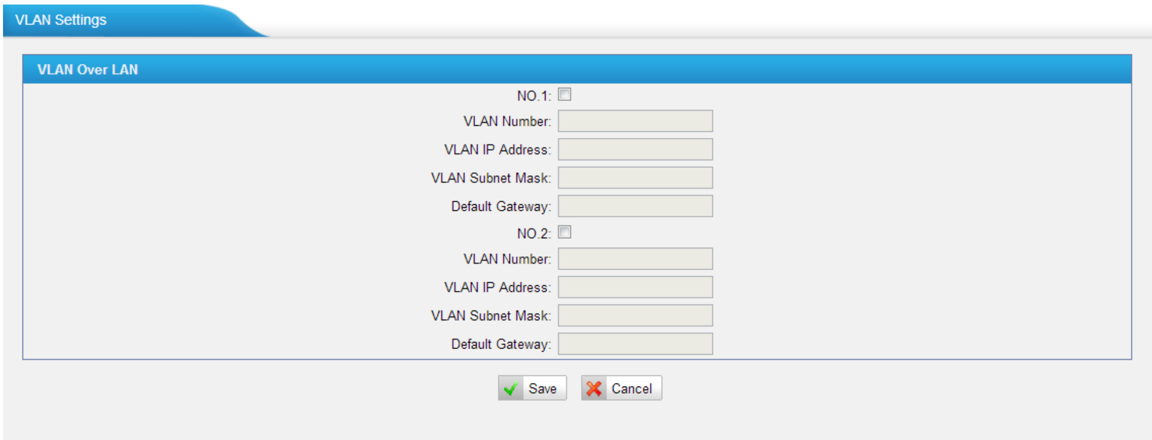


Figure 9-8 VLAN Settings

Please follow the steps below to set up VLAN on TA1610.

Step1. Create VLANs on your switch.

Step2. Allocate a VLAN ID and IP address for TA1610.

Step3. Configure VLAN settings page on TA1610.

VPN Settings

A virtual private network (VPN) is a method of computer networking typically using the public internet that allows users to privately share information between remote locations, or between a remote location and a business' home network. A VPN can provide secure information transport by authenticating users, and encrypting data to prevent unauthorized persons from reading the information transmitted. The VPN can be used to send any kind of network traffic securely. TA1610 supports OpenVPN.

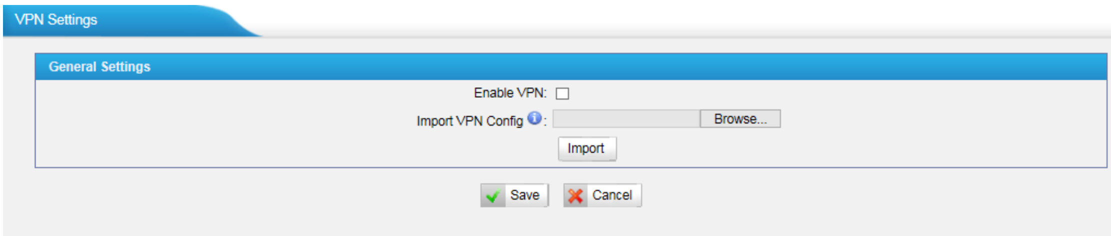


Figure 9-9 VPN Settings

- **Enable VPN**

  Enable VPN feature.

- **Import VPN Config**

  Import configuration file of OpenVPN.

Notes:

1. Uncomment “user” and “group” in the “config” file. You can get the config package from the OpenVPN provider.
2. TA1610 works as VPN client mode only.

DDNS Settings

DDNS(Dynamic DNS) is a method/protocol/network service that provides the capability for a networked device, such as a router or computer system using the Internet Protocol Suite, to notify a Domain Name System (DNS) name server to change, in real time, the active DNS configuration of its configured hostnames, addresses or other information.

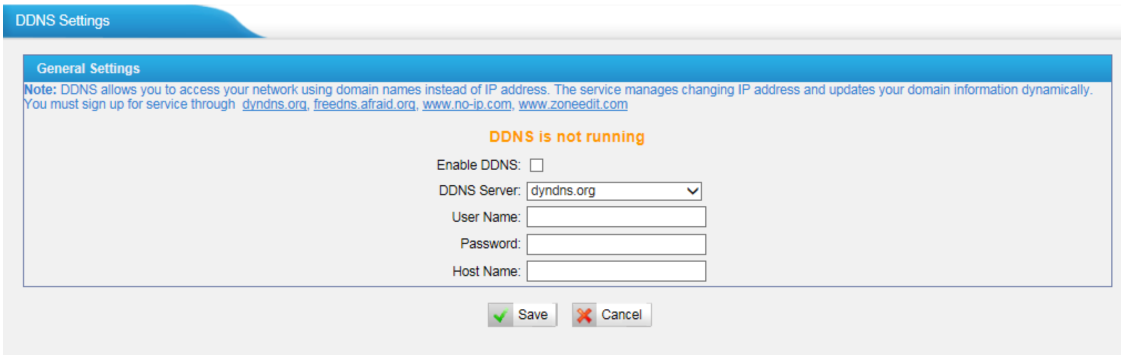


Figure 9-10 DDNS Settings

Table 9-4 Description of DDNS Settings

Items	Description
DDNS Server	Select the DDNS server you sign up for service.
User Name	User name the DDNS server provides you.
Password	User account's password.
Host Name	The host name you have got from the DDNS server

Note: DDNS allows you to access your network using domain names instead of IP address. The service manages changing IP address and updates your domain information dynamically. You must sign up for service through dyndns.org, freedns.afraid.org, www.no-ip.com, www.zoneedit.com.

Static Route

TA FXO Gateway will have more than one Internet connection in some situations but it has only one default gateway. You will need to set some Static Route for TA FXO Gateway to force it to go out through different gateway when accessing to different internet.

The default gateway priority of TA FXO Gateway from high to low is VPN/VLAN →LAN port.

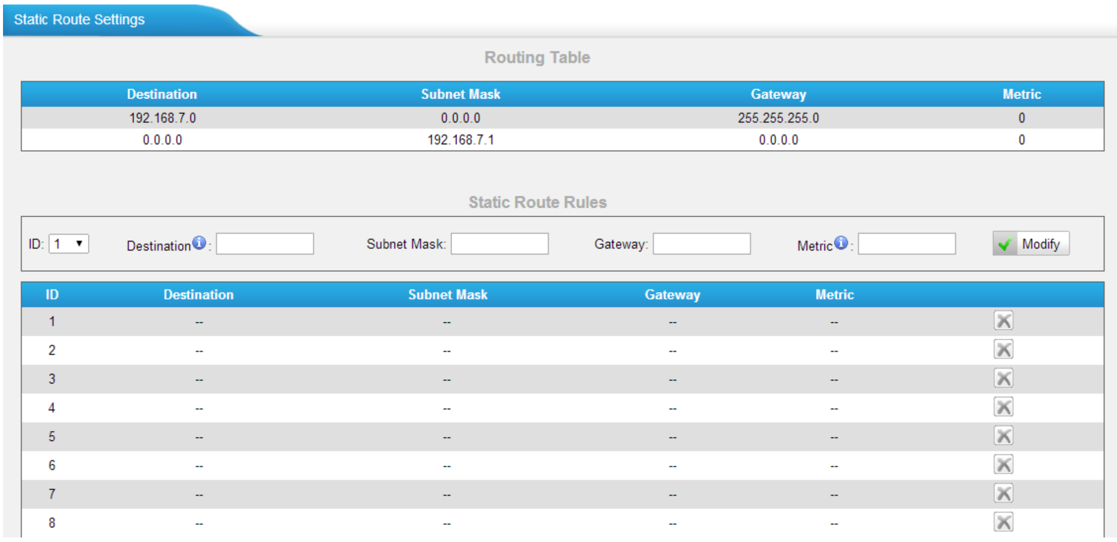


Figure 9-11 Static Route

1) Route Table

The current route rules of TA FXO Gateway.

2) Static Route Rules

You can add new static route rules here.

Table 9-5 Description of Static Route Settings

Items	Description
Destination	The destination network to be accessed to by TA FXO Gateway.
Subnet Mask	Specify the destination network portion.
Gateway	Define which gateway TA FXO Gateway will go through when accessing the destination network.
Metric	The cost of a route is calculated by using what are called routing metric. Routing metrics are assigned to routes by routing protocols to provide measurable statistic which can be used to judge how useful (how low cost) a route is.
Interface	Define which internet port to go through.

Remote Management

Yeastar Remote Management provides an affordable, low maintenance solution to easily deploy Yeastar VoIP PBX and TA VoIP gateways across multiple locations, reducing complexity and providing deep visibility and control.

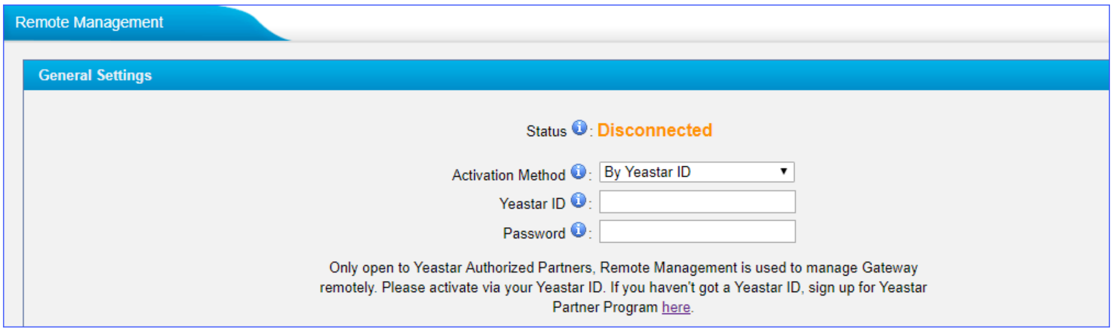


Figure 9-12 Remote Management

Yeastar Remote Management Quick Start

Add TA gateway to Yeastar Remote Management as follows.

For more information about Yeastar Remote Management, refer to Yeastar Remote Management User Guide.

You can add TA gateway to Yeastar Remote Management either by Yeastar ID or aut hentication code. Choose either method to connect TA gateway to Yeastar Remote Management according to your needs.

- **Connect to Remote Management by Yeastar ID**

  If you can access the TA gateway, you can enter your Yeastar ID in the TA gateway, the device will be added to Yeastar Remote Management.

  1. Log in the web interface of TA gateway, and go to System>Network Preferences> Remote Management.
  2. Select "**By Yeastar ID**" from the drop-down menu of **Activation Method**, enter your Yeastar ID and password.

3. Click "**Save**", the TA gateway will be added to Remote Management. "Connected" displays if the device is added successfully.

- **Connect to Remote Management by Authentication Code**

If you cannot access the TA gateway, you can add the TA gateway to Yeastar Remote Management by authentication code.

1. Log in the web interface of Yeastar Remote Management, go to Device> My device. Click "**Add**", enter a device name to help you identify the device, and click "**Add**" to add the device.
2. Optional. Verify serial number and MAC address.
   - Select this option: You need to fill in the serial number and MAC address of the device. The generated authentication code can be applied to the device only.
   - Unselect this option: Any S-Series PBX can access to Remote Management by the generated authentication code. If the device is added to Remote Management successfully, the related serial number and MAC address will be added to device information automatically.
3. Click "**Add**", the system will generate an unique authentication code.
4. Log in the web interface of TA gateway, go to System>Network Preferences>Remote Management.
5. Select "**By Authentication Code**" from the drop-down menu of **Activation Method**, and enter the authentication code.
6. Click "**Save**", the device will be added to Remote Management successfully.

   "Connected" displays if the device is added successfully.

Security Center

This chapter describes how to secure your TA1610. It is strongly recommended that users configure firewall and other security options on TA1610 to prevent the attack fraud and the system failure or calls loss.

- Security Center
- Alert Settings
- AMI Settings
- Certificates
- Firewall Rules
- IP Blocklist

Security Center

All the security settings including Firewall, Service, Port Settings in TA1610 are displayed in Security Center. Users could rapidly check and configure the relevant security settings here.

1) Firewall

In the "Firewall" tab, users could check firewall configuration and alert settings. By clicking the relevant button, you can enter the configuration page directly.

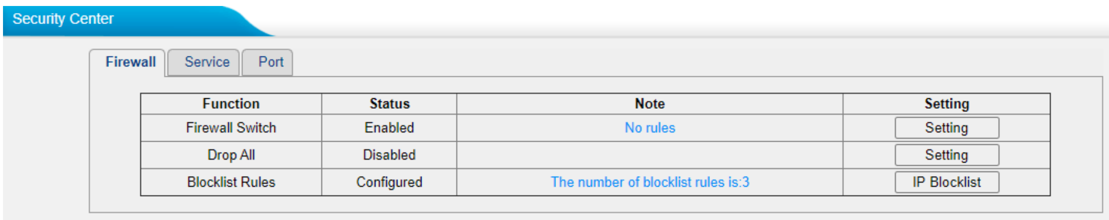
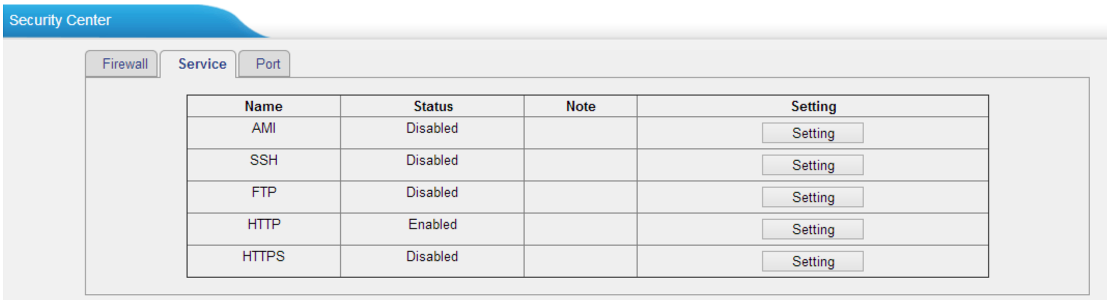


Figure 10-1 Security Center—Firewall

2) Service

In “Service” tab, you can check AMI/SSH status. For AMI/SSH, you can enter the according page by clicking the button in “Setting” column.

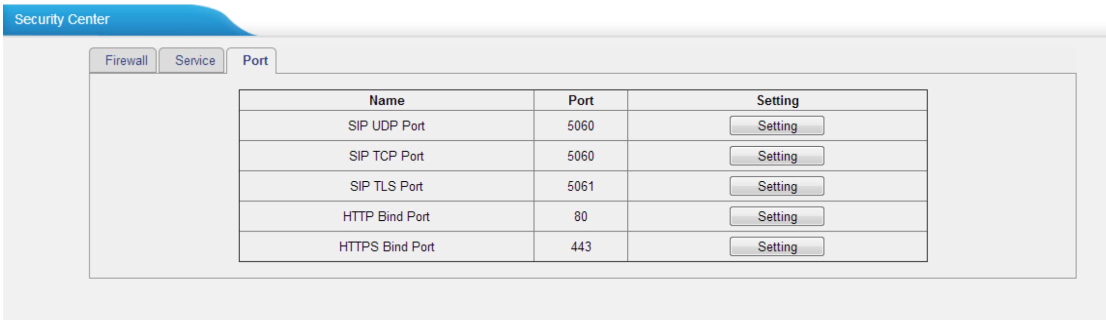


Name	Status	Note	Setting
AMI	Disabled		Setting
SSH	Disabled		Setting
FTP	Disabled		Setting
HTTP	Enabled		Setting
HTTPS	Disabled		Setting

Figure 10-2 Security Center—Service

3) Port

In “Port” tab, you can check SIP port and HTTP port. You can also enter the relevant page by clicking the button in “Setting” column.



Name	Port	Setting
SIP UDP Port	5060	Setting
SIP TCP Port	5060	Setting
SIP TLS Port	5061	Setting
HTTP Bind Port	80	Setting
HTTPS Bind Port	443	Setting

Figure 10-3 Security Center—Port

Alert Settings

If the device is under attack, the system will alert users via call or E-mail.

The attack modes include IP attack and Web Login.

- **IPATTACK**

  When the system is attacked by IP address, the firewall will add the IP to auto IP Blocklist and notify the user if it matches the protection rule.

- **WEBLOGIN**

  Web Login Alert Notification: entering the wrong password consecutively for five times when logging in TA FXO Gateway Web interface will be deemed as an attack, the system will limit the IP login within 10 minutes and notify the user.

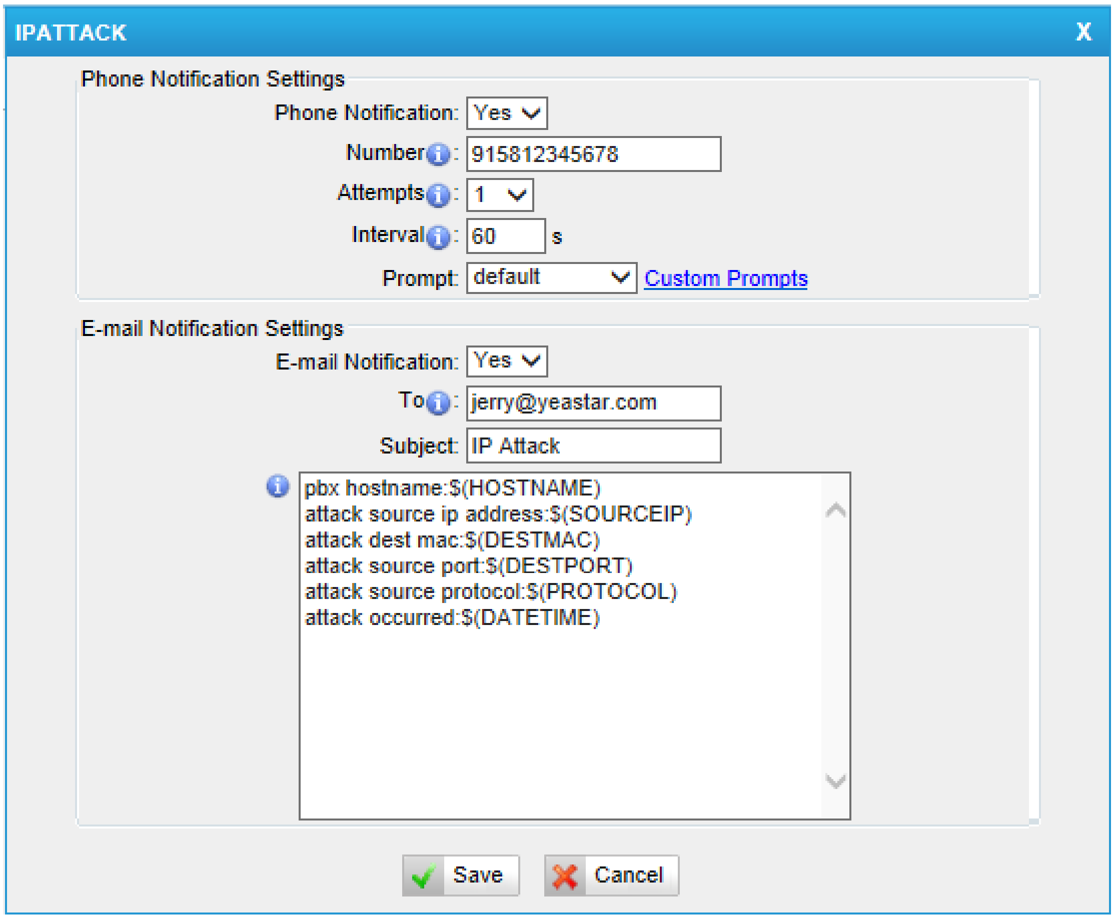


Figure 10-4 Alert Settings

Table 10-1 Description of Alert Settings

Phone Notification Settings	
PHONE Notification	Whether to enable phone notification or not.
Number	The numbers could be set for alert notification; users can setup multiple extension and outbound phone numbers. Please separate them by ";".

	Example: "500;9911", if the extension has configured Follow Me Settings, the call would go to the forwarded number directly.
Attempts	The attempts to dial a phone number when there is no answer.
Interval	The interval between each attempt to dial the phone number. Must be longer than 3 seconds, the default value is 60 seconds.
Prompt	Users will hear the prompt while receiving the phone notification.
Email Notification Settings	
E-mail Notification	Whether to enable E-mail Notification or not.
Recipient's Name	The recipients for the alert notification, and multiple email addresses are allowed, please separate them by ";". E.g. jerry@yeastar.com;jason@yeastar.com,456@sina.com
Subject	The subject of the alert email.
Email Content	Text content supports predefined variables. Variable names and corresponding instructions are as follows: gateway hostname:$(HOSTNAME) attack source ip address:$(SOURCEIP) attack dest mac:$(DESTMAC) attack source port:$(DESTPORT) attack source protocol:$(PROTOCOL) attack occurred:$(DATETIME)

AMI Settings

The Asterisk Manager Interface (AMI) is a system monitoring and management interface provided by Asterisk. It allows live monitoring of events that occur in the system, as well enabling you to request that Asterisk perform some action. The actions that are available are wide-ranging and include things such as returning status information and originating new calls. Many interesting applications have been developed on top of Asterisk that take advantage of the AMI as their primary interface to Asterisk.

There are two main types of messages on the Asterisk Manager Interface: manager events and manager actions.

The 3rd party software can work with TA1610 using AMI interface. It is disabled by default. If necessary, you can enable it.

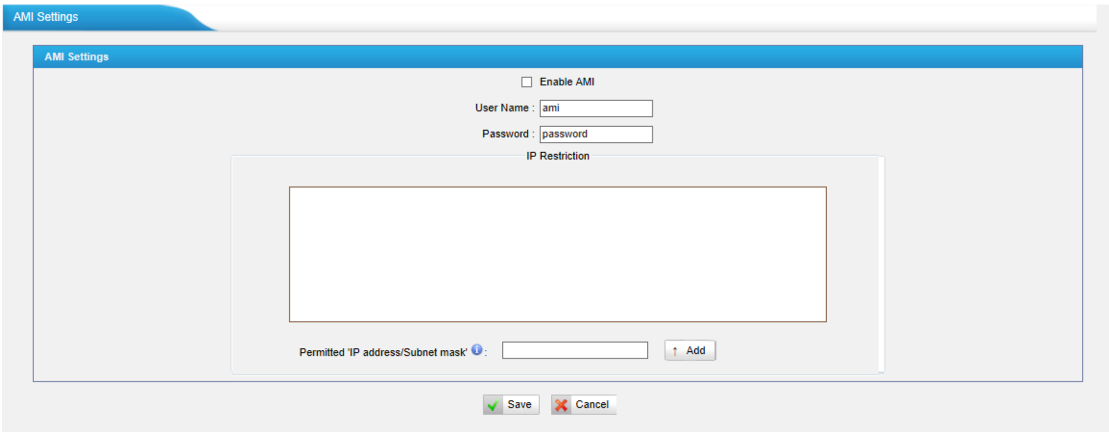


Figure 10-5 AMI Settings

- **User Name, Password & Port**

  After enabling AMI, you can use this username and password to log in TA1610. The default port is 5038.

- **Permitted "IP address/Subnet mask"**

  You can set which IP is allowed to log in TA1610 AMI interface.

Certificates

TA1610 supports TLS transport, you can configure FXO port with TLS transport. To use TLS, you should upload certificates first.

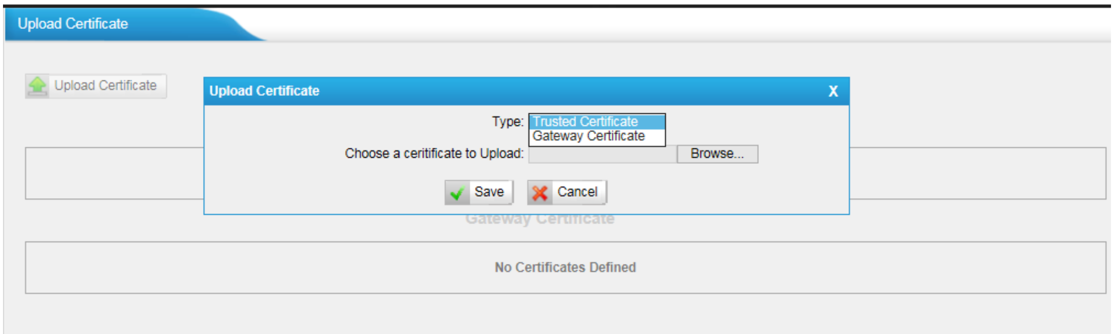


Figure 10-6 Upload Certificate

- **Trusted Certificate**

  This certificate is a CA certificate. When selecting “TLS Verify Client” as “Yes”, you should upload a CA. The relevant VoIP provider should also have this certificate.

- **Gateway Certificate**

  This certificate is server certificate. No matter selecting “TLS Verify Client” as “Yes” or “NO”, you should upload this certificate to TA1610. If the VoIP provider enables “TLS Verify server”, you should also upload the relevant CA certificate on the VoIP provider.

Firewall Rules

Firewalls are used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially intranets. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.

Figure 10-7 Firewall Settings

1) General Settings

Table 10-2 Description of Firewall General Settings

Items	Description
Enable Firewall	Enable the firewall to protect the device.
Disable Ping	Enable this item to drop net ping from remote hosts.
Drop All	When you enable “Drop All” feature, the system will drop all packets or connection from other hosts if there are no other rules defined. To avoid locking the devices, at least one “TCP” accept common rule must be created for port used for SSH access, port used for HTTP access and port sued for CGI access.

2) Common Rules

There is no default rule; you can create one as required.

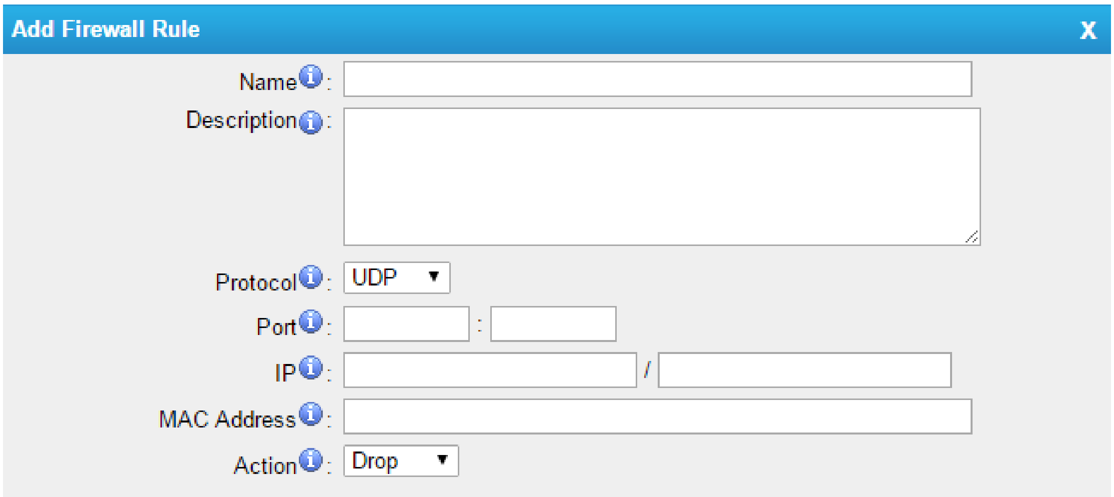


Figure 10-8 Common Rules

Table 10-3 Description of Common Rules

Items	Description
Name	A name for this rule, e.g. “HTTP”.
Description	Simple description for this rule. E.g. accept the specific host to access the Web interface for configuration.
Protocol	The protocols for this rule.
Port	Initial port should be on the left and end port should be on the right. The end port must be equal to or greater than start port.
IP	The IP address for this rule. The format of IP address is: IP/mask E.g. 192.168.5.100/255.255.255.255 for IP 192.168.5.100 E.g. 192.168.5.0/255.255.255.0 for IP from 192.168.5.0to 192.168.5.255.
MAC Address	The format of MAC Address is XX:XX:XX:XX:XX:XX, X means 0~9 or A~F in hex, the A~F are not case sensitive.
Action	Accept: Accept the access from remote hosts.

	Drop: Drop the access from remote hosts. Ignore: Ignore the access.

Note: the MAC address will be changed when it's a remote device, so it will not be working to filter using MAC for remote devices.

3) Auto Defense

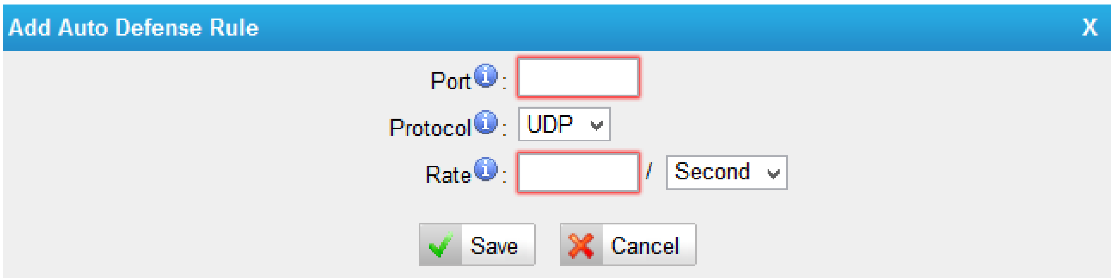


Figure 10-9 Auto Defense

Table 10-4 Description of Auto Defense

Items	Description
Port	The port you want to auto defense, for example, 8022.
Protocol	Select the protocol. You can select UDP or TCP.
Rate	The maximum packets or connections can be handled per unit time. For example, if you configure it as below: Port: 8022 Protocol: TCP Rate: 10/min Then, it means maximum 10 TCP connections can be handled in 1 minute. The 11th connection will be dropped.

IP Blocklist

You can set some packets accept speed rules here. When an IP address, which hasn't been accepted in common rules, sends packets faster than the allowed speed, it will be set as a black IP address and be blocked automatically.

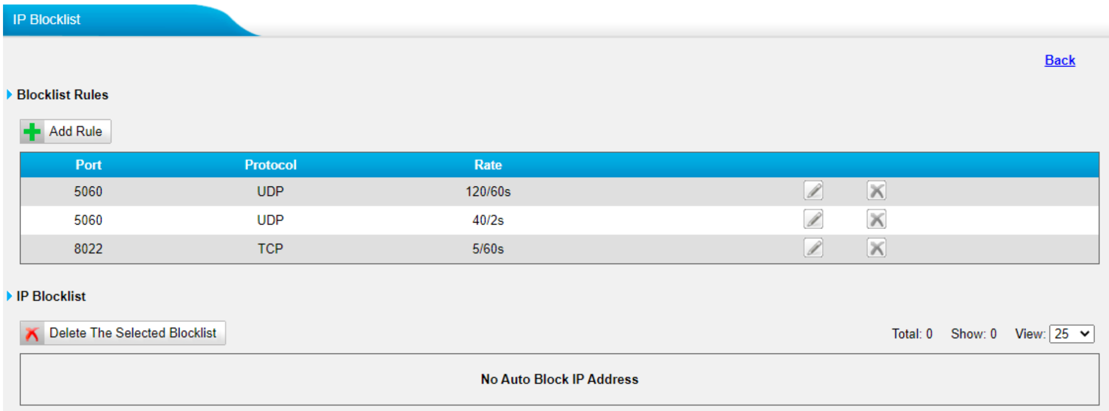


Figure 10-10 IP Blocklist Settings Page

1) Blocklist rules

We can add the rules for IP Blocklist rate as demanded.

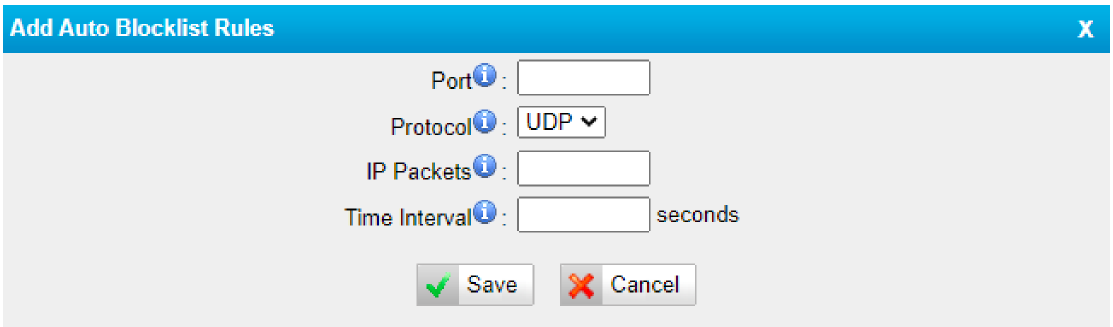


Figure 10-11 Add Blocklist Rule

Table 10-5 Description of Auto Blocklist Rules

Items	Description
Port	Auto defense port
Protocol	Auto defense protocol. TCP or UDP.
IP Packets	Allowed IP packets number in the specific time interval.
Time interval	The time interval to receive IP packets. For example, IP packets 90, time interval 60 means 90 IP packets are allowed in 60 seconds.

2) IP Blocklist

The blocked IP address will display here, you can edit or delete it as youwish.

System Preferences

This chapter describes system maintenance settings including the followings:

- Password Settings
- Date and Time
- Auto Provision Settings
- Firmware Update
- Backup and Restore
- Reset and Reboot

Password Settings

It is highly recommended to change the system's password after first login. Go to **System→System Preferences→Password Settings** to change the password.

1. Enter the old password first.
2. Enter a new password and retype the new password to confirm. The password complexity will be detected, which will help users to set a strong password and make TA1610 safer. A strong password is comprised of letters, numbers and characters.
3. Save the changes, the user will be automatically logged out.
4. Log in TA1610 using the new password.

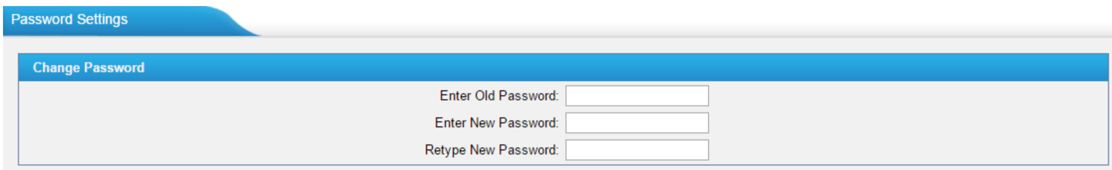


Figure 11-1 Password Settings

Date and Time

Please adjust the time of TA1610 (including the time zone) consistent with your local time. Go to **System→System Preferences→Date and Time** to configure the system date and time.

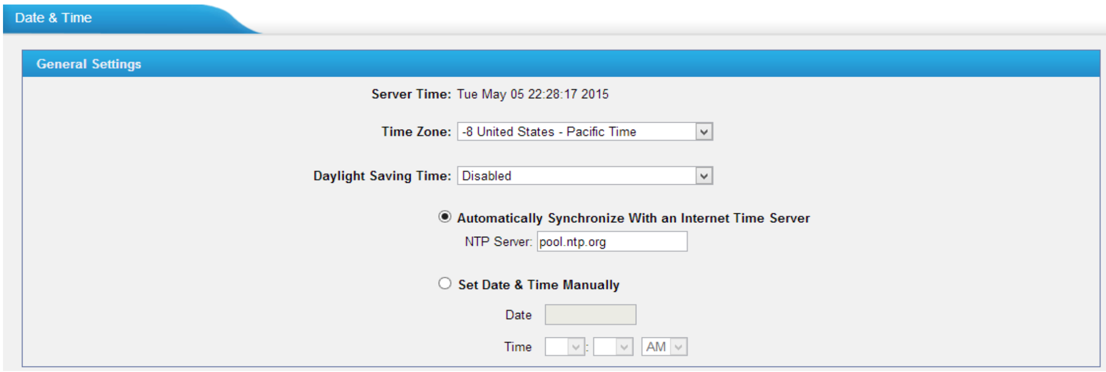


Figure 11-2 Date and Time

- **Time Zone**

  Select your current and correct time zone on TA1610.

- **Daylight Saving Time**

  The option is disabled by default. Enable it when necessary.

- **Automatically Synchronize with an Internet Time Server**
  TA1610 will adjust its internal clock to a central network server. Please note the TA1610 should be able to access to the Internet if you choose this method.
- **Set Date & Time Manually**
  Enter the time using the numbers on your keyboard.

Note: you have to reboot the system to make the changes take effect.

Auto Provision Settings

Three methods are supported for Auto Provision: PNP, DHCP and you can manually configure a server URL to get the configuration file from the server. Go to **System→System Preferences→Auto Provision Settings** to configure.

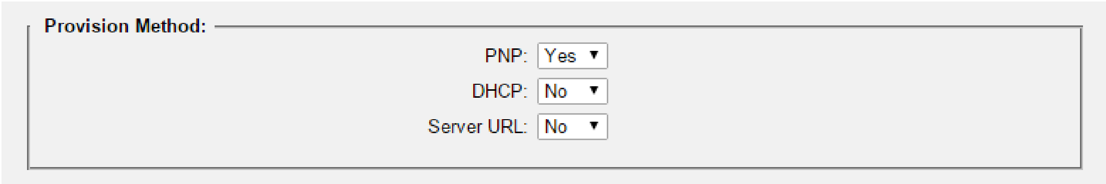


Figure 11-3 Auto Provision Methods

PNP and **DHCP** modes work along with MyPBX "TA Provisioning".Firstly, users need to configure TA1610 on MyPBX "TA Provisioning" page. Then TA1610 will find and get the configuration file from MyPBX during boots up.

In **PNP** mode, you just need to place the TA1610 in the same IP range network with MyPBX, then you can find the TA1610 and provision it on MyPBX ”TA Provisioning” page.

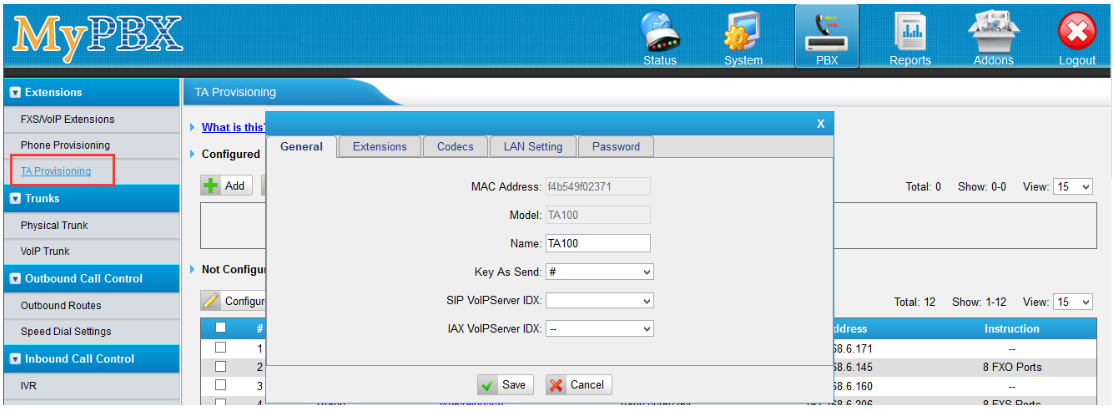


Figure 11-4 MyPBX TA Provisioning

If you use **DHCP** mode to do auto provision, you should enable DHCP Server on MyPBX to make it as a DHCP server. (System→Network Preferences→DHCP Server).

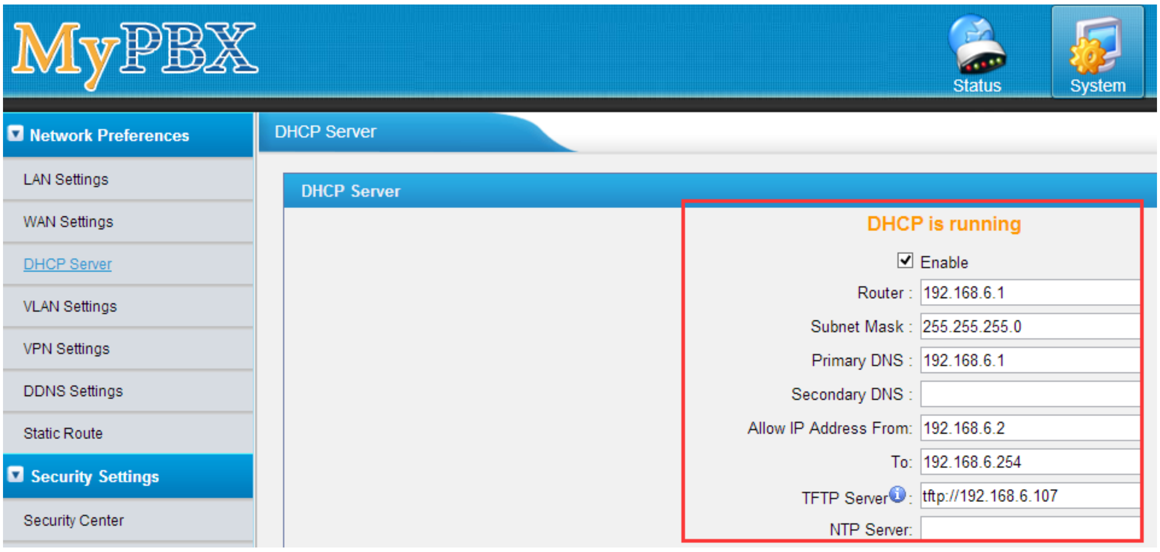


Figure 11-5 Set MyPBX as a DHCP Server

Then select DHCP mode on LAN settings page to make TA1610 as a DHCP client.

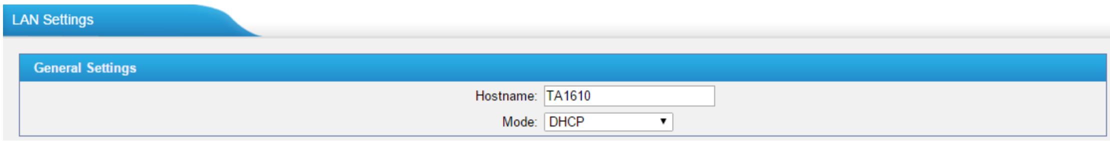


Figure 11-6 Set TA1610 as a DHCP Client

Another way to do auto provision is to download configuration file from the configured server URL. Fill in the URL, user name, password, and set the time, TA1610 will get the configuration file from the server automatically and regularly.

Note: if there is no user name and password for the server, leave these fields blank.

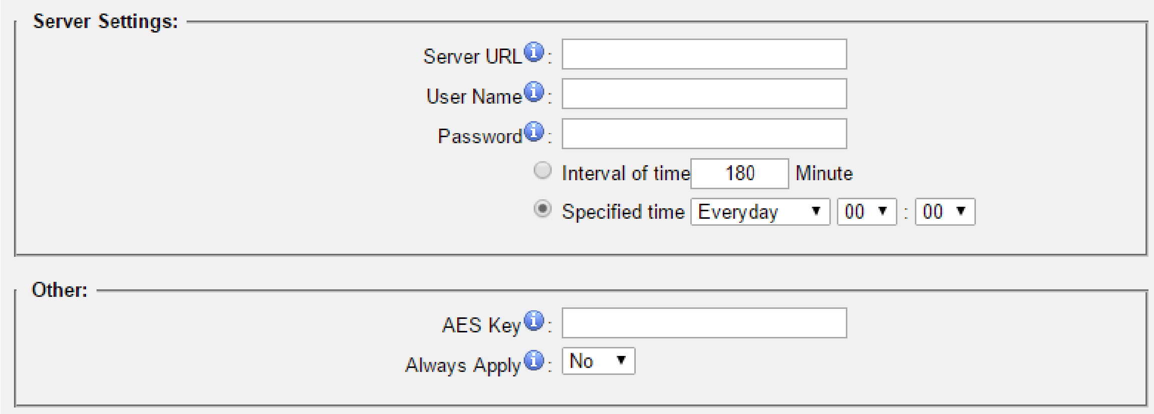


Figure 11-7 Server Address

- **AES Key**

  If the configuration file is encrypted by AES key, you need to fill the key in this field.

- **Always Apply**

  With No, it will compare the current configuration file with the last updated one, if the contents are the same no update will be applied. With Yes, it will always apply the updated configuration file.

Firmware Update

TA1610 can be upgraded to a new firmware version via network or locally. Users could upgrade firmware via HTTP or TFTP. Please go to **System→ System Preferences→ Firmware Update** to do upgrade.

Notes:

1. If "Reset configuration to Factory Defaults" is enabled, the system will be restored to factory default settings.
2. When updating the firmware, please don't turn off the power. Or the system will be damaged.
3. If you are trying to upgrade through HTTP, please make sure that your TA1610 is able to visit external network, or it cannot access Yeastar website to get the firmware file, causing the upgrade fail.

Upgrade through HTTP

On the Firmware Upgrade page, choose **HTTP URL**.

Step1. Enter the download link of the firmware file.
Note: the HTTP URL should be a **BIN** file download link.

Step2. Click “Start” to upgrade.

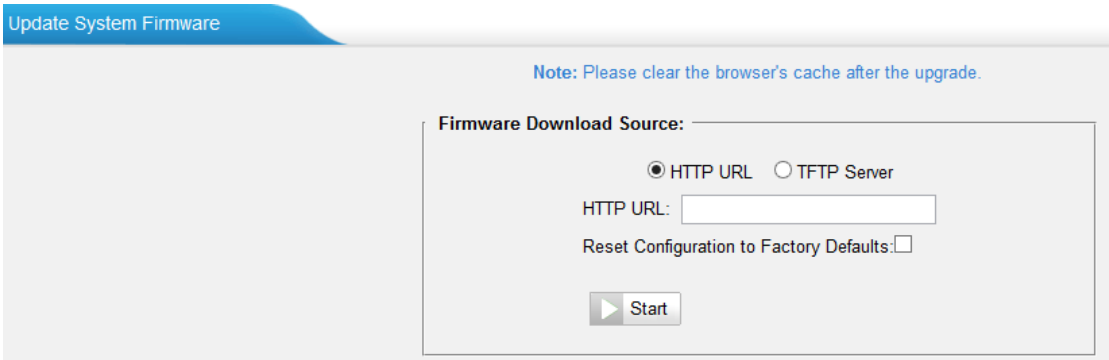


Figure 11-8 Upgrade through HTTP

Upgrade through TFTP

Step1. Download firmware file from Yeastar website.

Step2. Create a tftp Server (For example, tftpd on Windows).

1) Install tftpd32 software on computer.
   Download link: http://tftpd32.jounin.net/tftpd32_download.html

2) Configure tftpd32.

   On option “**Current Directory**”, click “**Browse**” button, choose the firmware file (BIN file) upgraded patch.

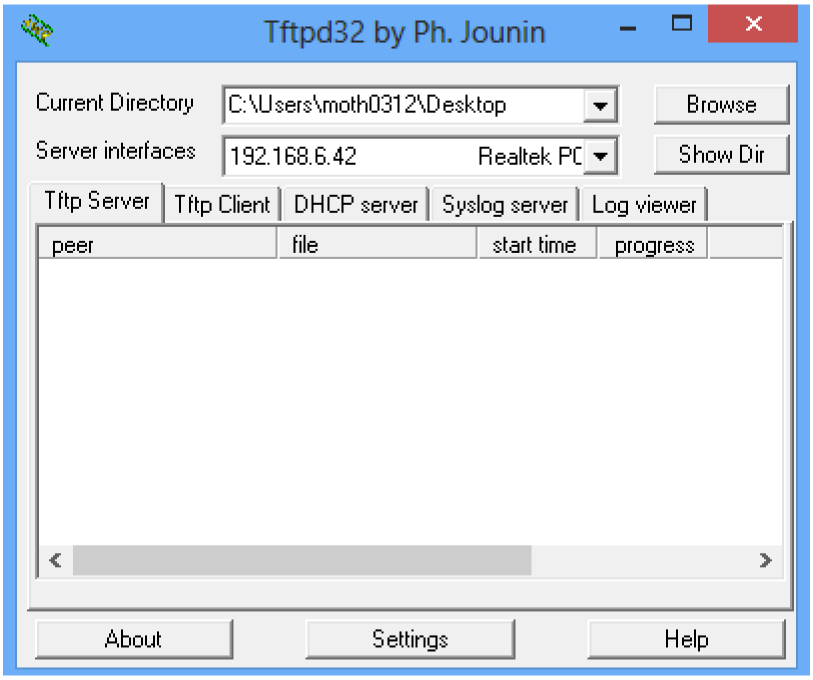


Figure 11-9 Configure Tftpd32

Step3. Logon the TA1610's Web page and go to **System→System Preferences→Firmware Update**, choose "**TFTP Server**".

1) TFTP Server: fill in IP address of tftpd32 server (your PC's IP address).

2) File Name: enter the name of firmware update. It should be a BIN file name.

3) Click "Start" to upgrade.

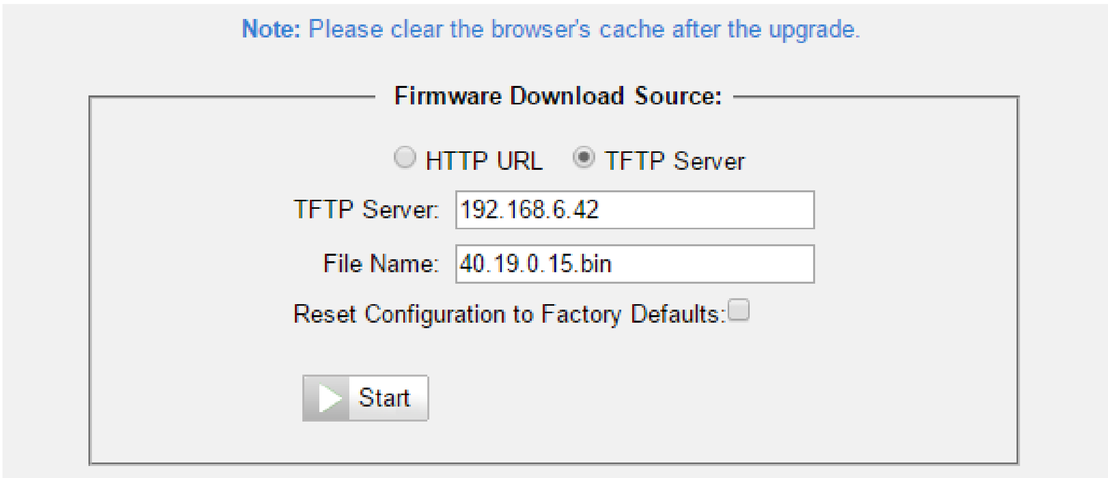


Figure 11-10 Upgrade through HTTP

Backup and Restore

TA1610 provides Backup and Restore feature, which allows you to create a complete backup of TA1610 configurations to a file.

Notes:

1. When you have updated the firmware version, it's not recommended to restore using an old package.
2. Backup from an earlier version cannot be restored on TA1610 of a later version.

- **Create a New Backup**

  Click Create a New Backup to create a new backup.

- **Upload a Backup**

  Click Upload a Backup to upload a backup.

- **Restore**

  To restore TA1610 configuration data, upload the backup file to TA1610 and click. Reboot the system to take effect.

  Please note the current configurations will be OVERWRITTEN with the backup data.

#	Name	Time	Options
1	backup_2015may9_174120.tar	Sat May 09 1:41:58 2015	

Figure 11-11 Restore Backup

Reset and Reboot

Users could reset and reboot the system under **System→ System Preferences→ Reset and Reboot**.

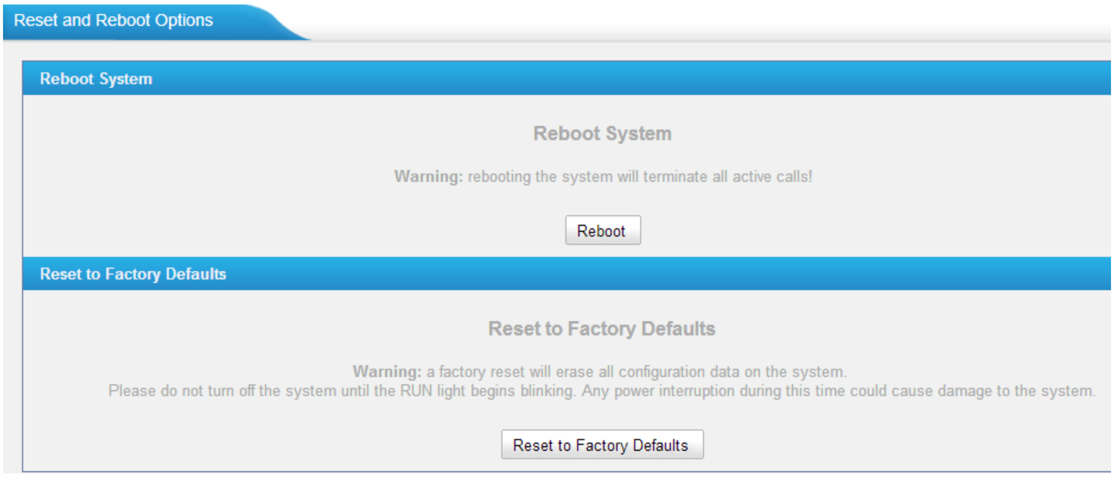


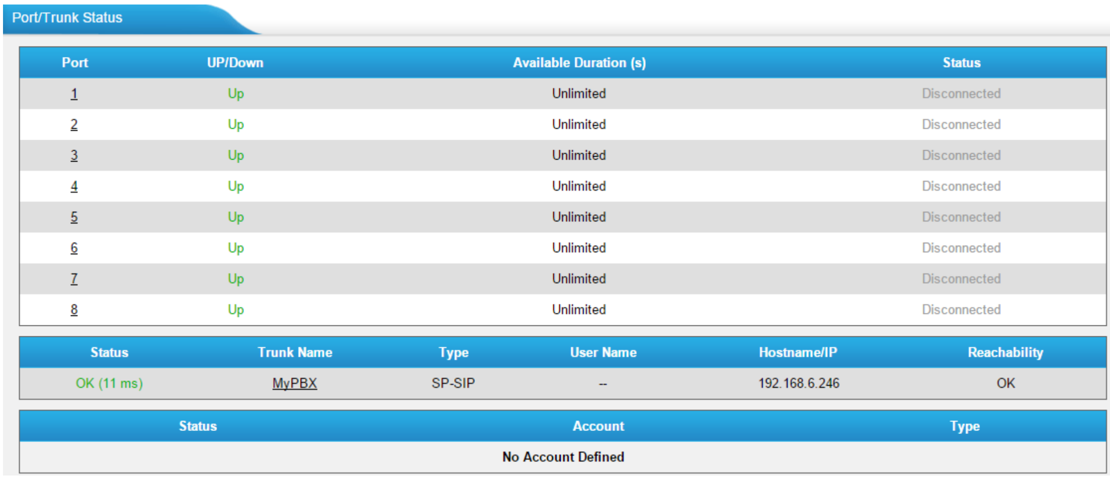
Figure 11-12 Reset and Reboot

Status

Users could check the system status on **Status→System Status**, where FXO Port and trunk Status, Network Status and System Info can be checked.

- Port/Trunk Status
- Network Status
- System Info

Port/Trunk Status



Port/Trunk Status

Port	UP/Down	Available Duration (s)	Status
1	Up	Unlimited	Disconnected
2	Up	Unlimited	Disconnected
3	Up	Unlimited	Disconnected
4	Up	Unlimited	Disconnected
5	Up	Unlimited	Disconnected
6	Up	Unlimited	Disconnected
7	Up	Unlimited	Disconnected
8	Up	Unlimited	Disconnected

Status	Trunk Name	Type	User Name	Hostname/IP	Reachability
OK (11 ms)	MyPBX	SP-SIP	--	192.168.6.246	OK

Status	Account	Type
	No Account Defined	

Figure 12-1 Port/Trunk Status

- **FXO Port Status**

Table 12-1 Description of FXO Port Status

Up/Down	
Up	The FXO module works well.
Down	The FXO module is broken.
Available Duration (s)	
The available duration of this PSTN trunk.	

Status	
Idle	The FXO port is idle.
Busy	The FXO port is busy.
Disconnect	There is no line connected to the FXO port.

- **VoIP Trunk Status**

1) SIP/IAX Type

Table 12-2 Description of SIP/IAX Trunk Status

Status	Description
Registered	Successful registration, trunk is ready for use.
Unregistered	Trunk registration failed.
Request Sent	Registering.
Waiting for Authentication	Wrong password.

2) SP-SIP/IAX Type

Table 12-3 Description of SP-SIP/IAX Trunk Status

Status	Description
OK	Successful registration, trunk is ready for use.
Unreachable	The trunk is unreachable.
Failed	Trunk registration failed.

3) VoIP Account

Table 12-4 Description of VoIP Account Status

Status	Description
Registered	The account is registered successfully on the SIP server.
Unregistered	Trunk registration failed.

Network status

In this page, the IP address of TA gateway will appear with their status.

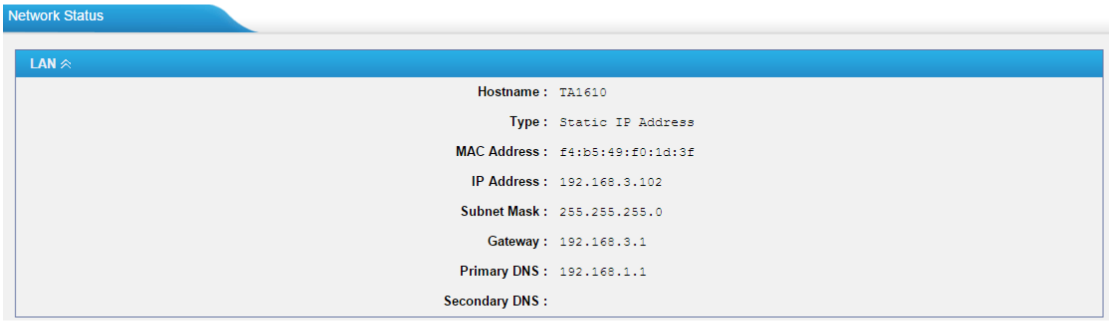


Figure 12-2 Network Status

If your VLAN or VPN are configured, you can check the status in this page also.

System Info

In this page, we can check the hardware/firmware version, or the disk usage of TA1610.

Figure 12-3 System Info

Reports

Users could check the call logs, system logs on **Status→Reports** page, and use the packet Tool and Port Monitor Tool to capture debug logs from TA1610.

- Call Logs
- System Logs
- Packet Tool
- Port Monitor Tool

Call Logs

The call log captures all call details, including call time, caller number, callee number, call type, call duration, etc. An administrator can search and filter call data by call date, caller/callee, trunk, duration, billing duration, status, or communication type.

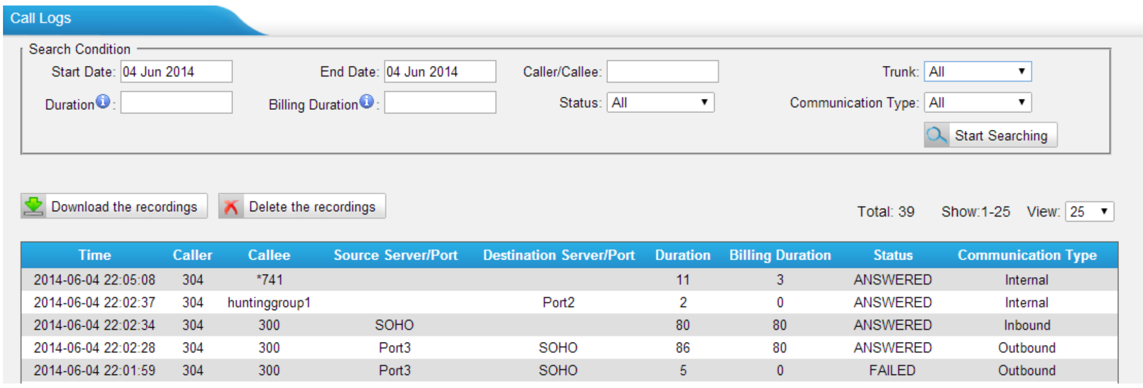


Figure 13-1 Call Logs

System Logs

You can download and delete the system logs of TA1610.

- **Enable Hardware Log**

  Save the information of hardware; (up to 4 log files)

- **Enable Normal Log**

Save the prompt information; (up to 16 log files)

- **Enable Web Log**

  Save the history of web operations (up to 2 log files)

- **Enable Debug Log**

  Save debug information (up to 2 log files)

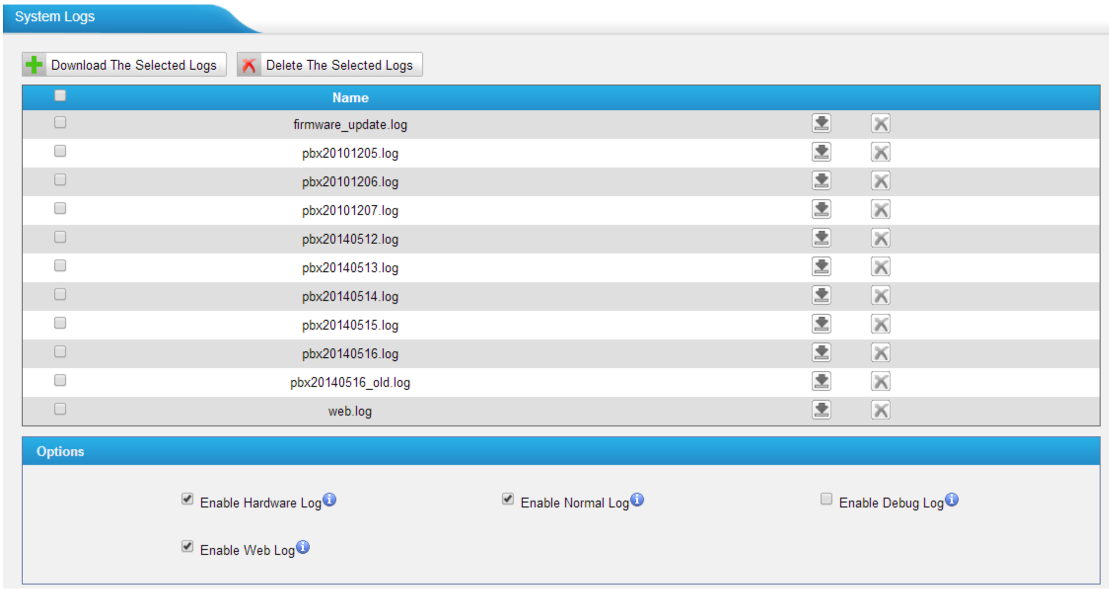


Figure 13-2 System Logs

Packet Tool

This feature is used to capture packets for technician. Integrate packet capture tool “Wireshark” in TA1610. Users also could specify the destination IP address and port to get the packets.

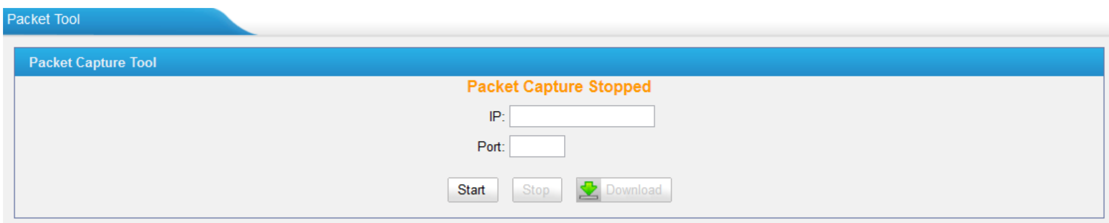


Figure 13-3 Packet Tool

- **IP**

  Specify the destination IP address to get the packets.

- **Port**

  Specify the destination Port to get the packets.

Port Monitor Tool

This tool is used to debug a FXO port. Select a FXO port and click “Start” to monitor the FXO port, stop monitoring by clicking “Stop” button.

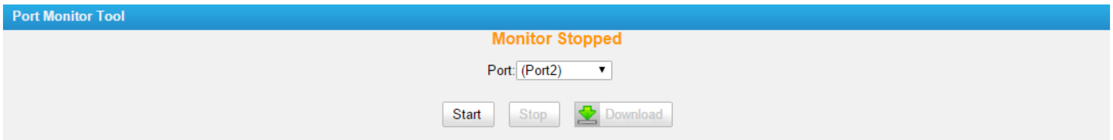


Figure 13-4 Port Monitor Tool