



Release Notes for TA1610

Version 40.19.0.X

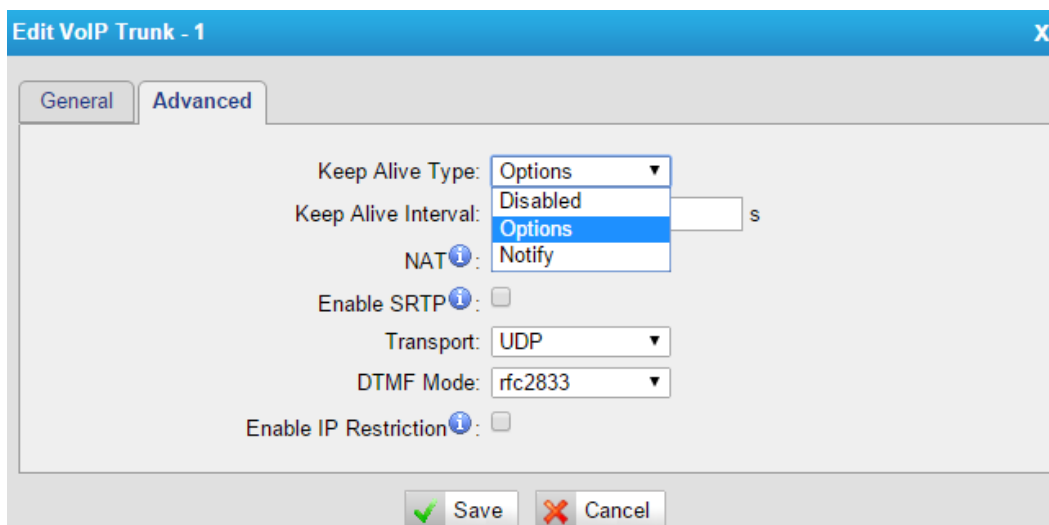
Yeastar Information Technology Co. Ltd

====Firmware Version: V40.19.0.19====**Applicable Model: TA1610****Release Date: October 11, 2015****Hardware Compatibility**

1. This firmware version is compatible with old hardware (version older than 1.2) and new hardware (version newer than 1.3).
2. You do not have to reset the system after upgrade.

New Features

1. If multiple routes (IP->Port or Port->IP) are created on the system, the system would route the incoming calls by the dial patterns set on the routes. If the routes have the same dial pattern, the system would route the incoming calls from the top route to the bottom route.
2. Added support for the time zones:
 - 8 Irkutsk
 - 9 Yakutsk
 - 11 Srednekolymsk
3. Rename “Qualify” setting on VoIP trunk as “Keep Alive Type”. You can select which SIP packet to send to the VoIP provider to keep the trunk alive. The system supports “OPTIONS” and “NOTIFY” packets.



====Firmware Version: V40.19.0.15====**Applicable Model: TA1610****Release Date: June 30, 2015****! Upgrade Notes**

In this version, we redesign the Web GUI to have new connection modes and routes settings to help you to connect your SIP server and TA1610 in an easier way. In the new version, we provide:

3 types of VoIP Trunk

- ✓ **Account**
- ✓ **SIP**
- ✓ **Service Provider**

i You can choose any one of these 3 types of VoIP trunk to connect your SIP server and TA1610. Please refer to *TA1610 UserManual* or relative solution documents for details.

2 types of Route

- ✓ **IP -> Port**
Control calls from your SIP server to TA1610 FXO ports
- ✓ **Port -> IP/Port**
Receiving incoming calls to PSTN trunks on TA1610 and route the calls to your SIP server or another PSTN trunk on TA1610.

IMPORTANT:

1. We strongly recommend that you back up the configurations and all the files before upgrade.
2. We suggest that you double check or reconfigure the route settings after upgrade.
3. Backup files from higher firmware version cannot be restored to the device with lower firmware version.
4. Please clear the browser's cache after the upgrade.

New Features

4. New Connection Mode for your SIP server and TA1610.
5. Added **SNMP** feature.
6. Added **Call Duration** Settings for FXO ports.
7. Added **Port Group** Settings.
8. Added **Trunk Group** Settings.
9. Added **Callback** Settings.
10. Added **Blacklist** settings to block incoming or outgoing calls.
11. Added **Alert settings** for IP attack or web-based attack.
12. Added **Port Monitor Tool** on web interface for PSTN trunk debugging.
13. Added **Auto Provision** feature.
14. Added **VAD** and **Echo Tail Length** settings.
15. Added support for **G723** and **G729AB** codec.

New Features (Instruction)

1. New Connection Mode for your SIP server and TA1610.

Instruction:

In the previous version, you should configure a VoIP Server, a Dial Pattern Template and apply them to a FXO port, then configure IP->Port or Port->IP route settings

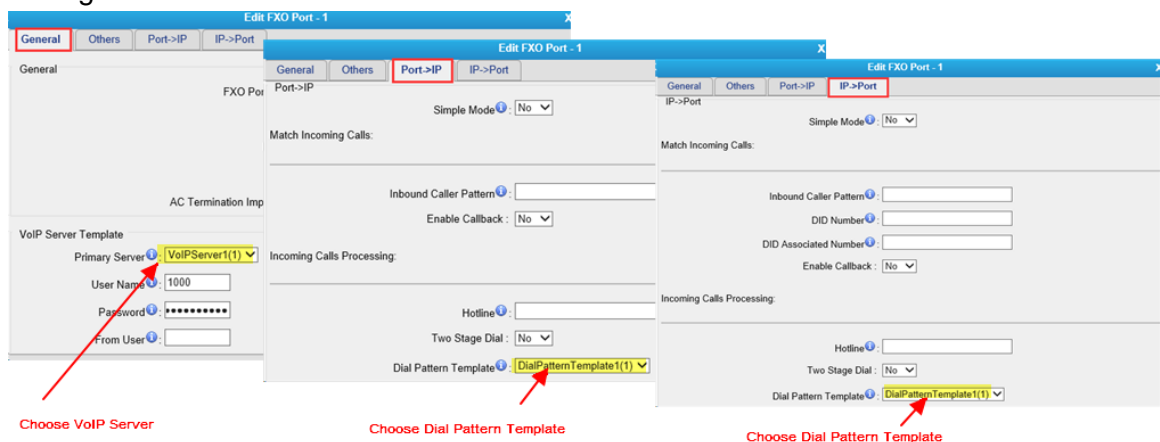


Figure 1 Configurations in old version

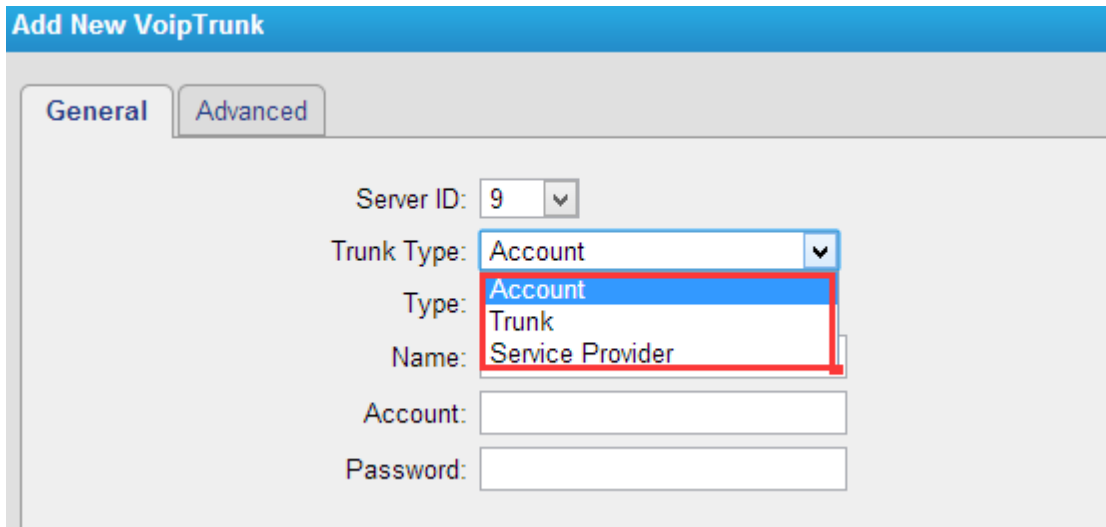
In the new version, we delete **VoIP Sever Settings** and **Dial Pattern Template Settings** on TA1610 and provide 3 types of VoIP Trunks and 2 types of Routes to make the configuration clearer and easier.

3 Types of VoIP Trunk (Account Mode, Trunk Mode, Service Provider Mode)

Path: Gateway→VoIPSettings→VoIP Trunk

Instruction:

Choose one type of VoIP trunk to connect your SIP server and TA1610.



Add New VoipTrunk

General | Advanced

Server ID: 9

Trunk Type: Account

Type: Account
Trunk

Name: Service Provider

Account:

Password:

Figure 2 VoIP Trunk

2 Types of Route (IP->Port, Port->IP/Port)

Path: Gateway → Routes Settings

Instruction:

- **IP -> Port**
Control calls from your SIP server to TA1610 FXO ports.
- **Port -> IP/Port**
Receiving incoming calls to PSTN trunks on TA1610 and route the calls to your SIP server or another PSTN trunk on TA1610.

The screenshot shows the 'IP->Port' configuration page. At the top, there is a blue header with the text 'IP->Port' and a close button 'X'. Below the header, the configuration is organized into sections:

- Route ID:** A dropdown menu set to '1'.
- Simple Mode:** A dropdown menu set to 'No'.
- Route Name:** A text input field containing 'RouteIP2Port1'.
- Match Incoming Calls:** A section header followed by a horizontal line.
- Call Source:** A dropdown menu set to 'SIP Trunk -- VoIPServer1_1'.
- Inbound Caller Pattern:** An empty text input field.
- DID Number:** An empty text input field.
- DID Associated Number:** An empty text input field.
- Enable Callback:** A dropdown menu set to 'No' with a link to 'Callback Settings'.
- Incoming Calls Processing:** A section header followed by a horizontal line.
- Call Destination:** A dropdown menu set to 'Port1 -- 4728'.
- Hotline:** An empty text input field.
- Two Stage Dial:** A dropdown menu set to 'No'.
- Outbound Dial Pattern:** An empty text input field.
- Strip:** An empty text input field followed by the text 'digits from left'.
- Prepend these digits:** An empty text input field followed by the text 'before dialing'.

Figure 3 Route Settings Page

2. Added SNMP feature.

Path: System→Network Preferences→ SNMP Settings

Instruction:

Simple Network Management Protocol (SNMP) is an Internet-standard protocol for managing devices on IP networks. NeoGate TA gateway supports three versions: V1, V2C and V3.

The screenshot shows the 'SNMP Settings' configuration page. At the top, there is a blue header with the text 'SNMP Settings'. Below the header, there are two notes:

- Note 1: If the managers want to access the device by SNMP v3 mode, 'SNMPv3 user' information must be configured.
- Note 2: If the managers want to access the device by SNMP v1/v2c mode, 'SNMP Community' information must be configured.

The main configuration area is titled 'SNMP Settings' and contains the following settings:

- SNMP is not running** (Status)
- Enable:** A dropdown menu set to 'No'.
- Local Port:** A text input field containing '161'.
- SNMPv3 User:** A section header.
- SNMPv3 User:** An empty text input field.
- Access Limit:** A dropdown menu set to 'NoAuth'.
- SNMP Community:** A section header.
- SNMP Mode:** A dropdown menu set to 'v2c'.
- Access:** Checkboxes for 'Read' (checked) and 'Write' (unchecked).
- Community:** An empty text input field.
- IP/SubnetMask:** An empty text input field.
- Trap Setting:** A section header.
- Trap Mode:** A dropdown menu set to 'v2c trap'.
- Trap Community:** A text input field containing 'public'.
- Trap IP:** A text input field containing '162'.

Figure 4SNMP Settings

3. Added Call Duration Settings for FXO ports.

Path: Gateway→PortList→FXO Port List

Instruction:

In this page you can configure the duration of the FXO port. A phonenotification or an email notification will be received if the balance reaches Alarm threshold you set for the port.

Figure 5 Call Duration Settings

4. Added Port Group Settings.

Path: Gateway→PortList→Port Group

Instruction:

You can select FXO ports and group them, then define the strategy for the group. When the Trunk Group is applied to a route for IP-to-Port calls or Port-to-Port calls, TA1610 will choose a PSTN trunk from the Trunk Group to call out according to the strategy.

- Round-robin: select the next available port in line
- Least used: select the port that is least used

Figure 6 Port Group Settings Page

5. Added Trunk Group Settings.

Path: Gateway→VoIPSettings→Trunk Group

Instruction:

Group the selected SIP trunks or SIP accounts.

Figure 7 Trunk Group

6. Added Callback Settings.

Path: Gateway→RoutesSettings→Callback Settings

Instruction:

- 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.

Callback Settings

Callback Number Settings

Note:
1. If you'd like to use callback feature, please make sure that it's enabled on the [IP->Port](#) / [Port->IP/Port](#) setting panel.
2. No callback rules need to be set if the trunk is able to call back with the caller ID directly.

Allow All Numbers ⓘ

<input type="checkbox"/>	ID	Callback Number
<input type="checkbox"/>	1	1589293883

Callback Rules Settings

No Callback Rules Defined

Figure 8 Callback Settings

7. Added Blacklist settings to block incoming or outgoing calls.

Path: Gateway→RouteSettings→Blacklist

Instruction:

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

Add New Blacklist X

ID: 1 ▼

Number : 456899876

Type : Inbound ▼

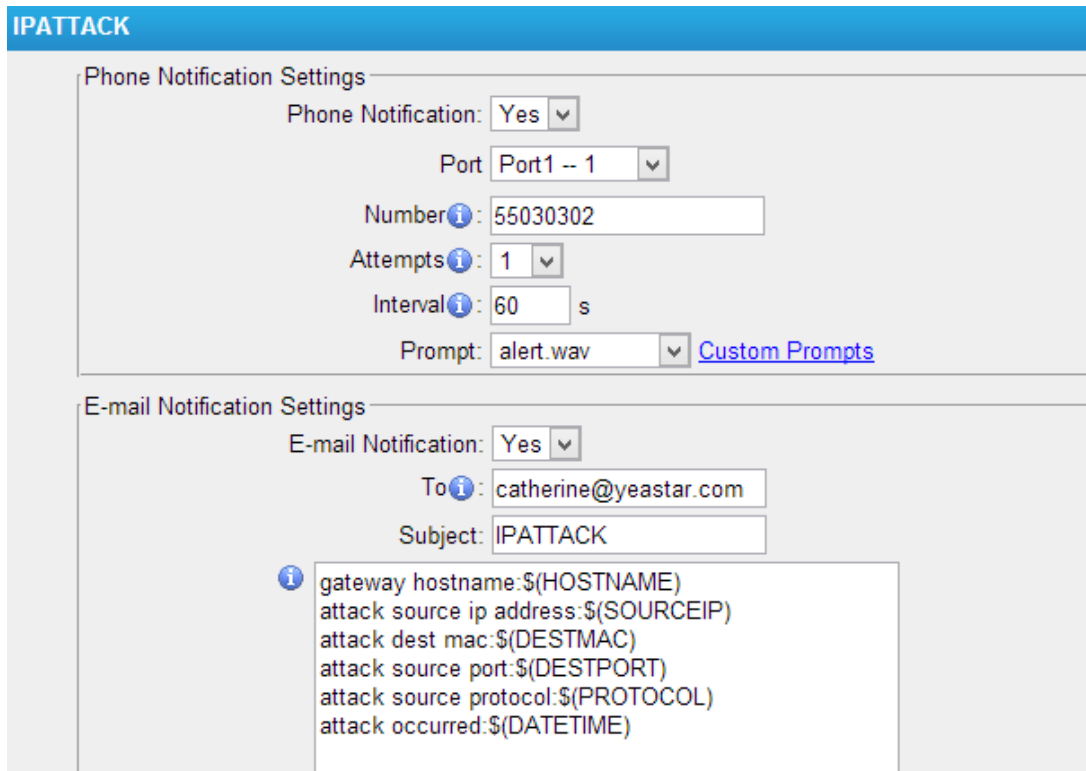
Figure 9 Blacklist Settings

8. Added Alert settings for IP attack or web-based attack.

Path: System→SecurityCenter→Alert Settings

Instruction:

After enabling this feature, phone notification or email notification will be sent to users if the system has been attacked.



IPATTACK

Phone Notification Settings

Phone Notification: Yes ▾

Port: Port1 -- 1 ▾

Number ⓘ: 55030302

Attempts ⓘ: 1 ▾

Interval ⓘ: 60 s

Prompt: alert.wav ▾ [Custom Prompts](#)

E-mail Notification Settings

E-mail Notification: Yes ▾

To ⓘ: catherine@yeastar.com

Subject: IPATTACK

gateway hostname:\$(HOSTNAME)
 attack source ip address:\$(SOURCEIP)
 attack dest mac:\$(DESTMAC)
 attack source port:\$(DESTPORT)
 attack source protocol:\$(PROTOCOL)
 attack occurred:\$(DATETIME)

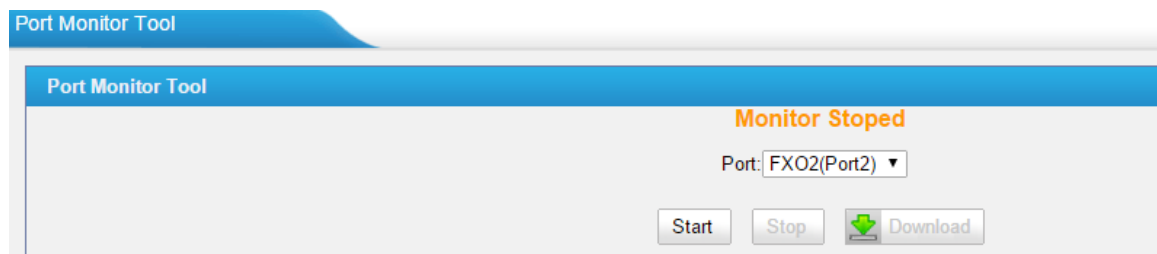
Figure 10 Alert Settings

9. Added Port Monitor Tool on web interface for FXO ports debugging.

Path: Status→Reports→Port Monitor Tool

Instruction:

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



Port Monitor Tool

Port Monitor Tool

Monitor Stopped

Port: FXO2(Port2) ▾

Start Stop Download

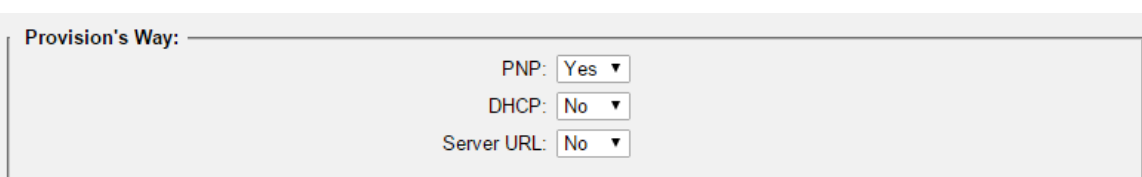
Figure 11 Port Monitor Tool

10. Added Auto Provision feature.

Path: System→SystemPreferences→Auto Provision Settings

Instruction:

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.



Provision's Way:

PNP: Yes ▾

DHCP: No ▾

Server URL: No ▾

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

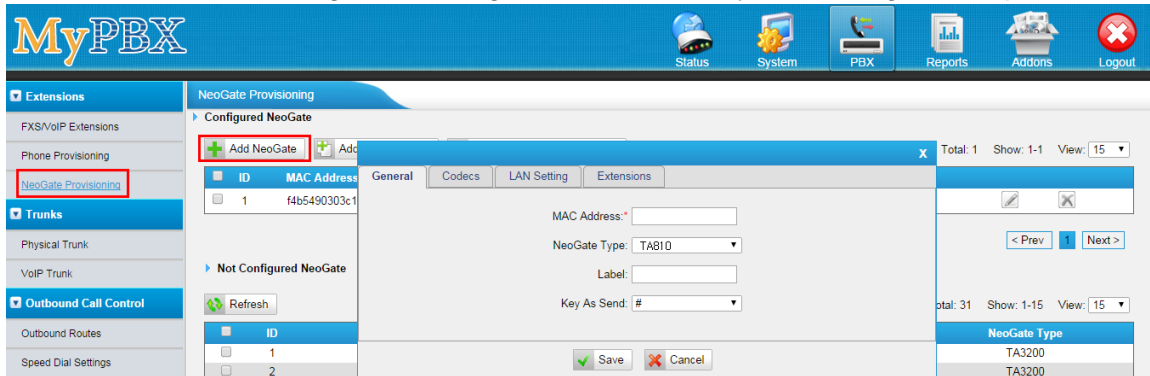


Figure 12 MyPBX NeoGate Provisioning

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

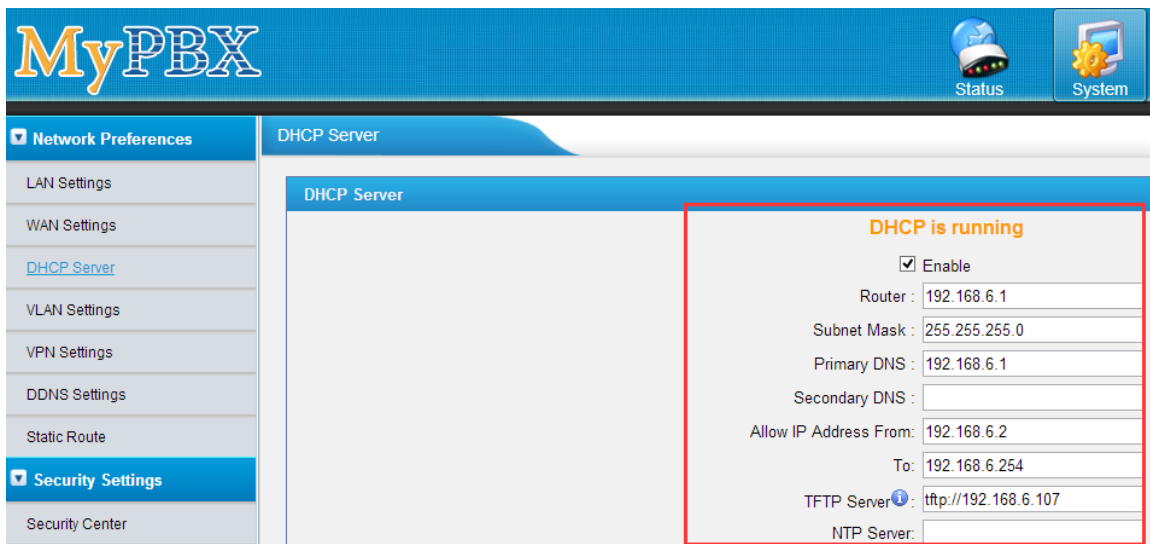


Figure 13 Set MyPBX as a DHCP Sever

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

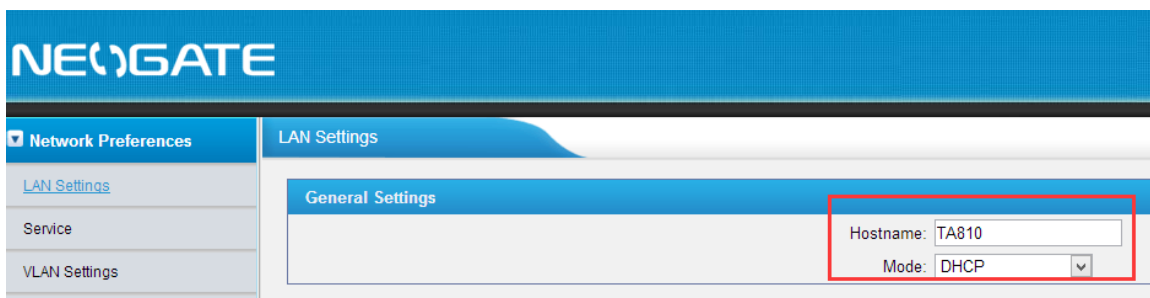


Figure 14 Set TA1610 as a DHCP Client

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.

The screenshot shows two configuration sections. The 'Server Settings' section includes fields for 'Server URL', 'User Name', and 'Password', each with an information icon. Below these are two radio buttons for scheduling: 'Interval of time' (set to 180 Minute) and 'Specified time' (set to Everyday 00:00). The 'Other' section includes an 'AES Key' field and an 'Always Apply' dropdown menu set to 'No'.

Figure 15 Server Address

Other Settings for Auto Provision

- **AES Key:** If the configuration file is encrypted by AES key, you need to fill the key in this field.
- **Always Apply:** whether to check the new configuration and apply to TA1610.

11. Added VAD and Echo Tail Length settings.

Path: Gateway→GatewaySettings→General Preferences

Instruction:

Adjust VAD and Echo Tail Length settings to get better voice quality.

The screenshot shows the 'General Settings' and 'Voice Settings' sections. In 'General Settings', 'MAX Call Duration(s)' is 6000, 'G723 Encoding Rate' is 6.3kbps, and 'FXO Mode' is FCC. In 'Voice Settings', 'Enable Jitterbuffer' is No, 'Jitter Buffer MaxSize' is 40, 'VAD' is Yes, and 'Echo Tail Length' is 128ms. The VAD and Echo Tail Length settings are highlighted with a red box.

Figure 16 VAD & Echo Tail Length

12. Added support for G723 and G729AB codec.

Path: Gateway→VoIPSettings→SIPSettings→Codec

Instruction:

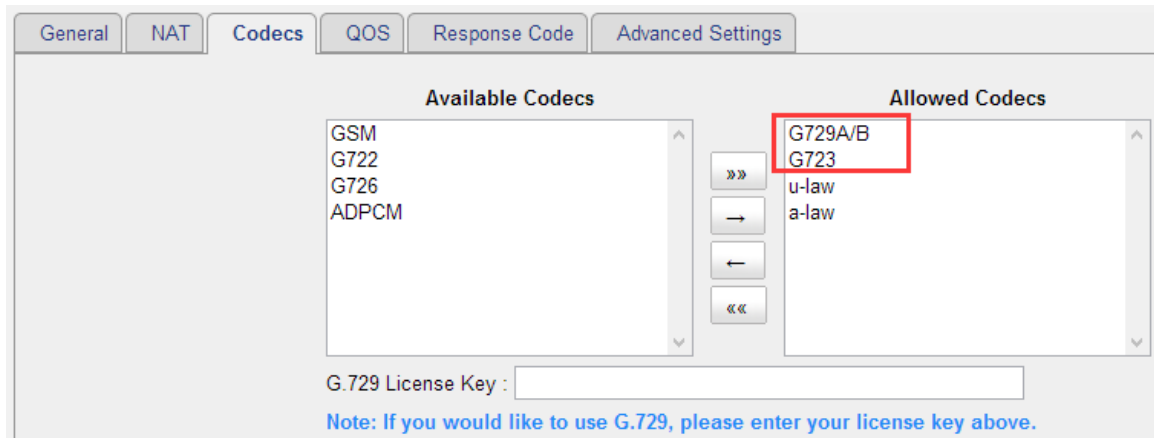


Figure 17 Codec

G729AB is compatible with G729, G729A and G729B.

G723 Encoding Rate can be adjusted on **Gateway**→**GatewaySettings**→**General Preferences** page.

[The End]