



Release Note for TA1600/2400/3200

Version 40.19.0.X

Yeastar Information Technology Co. Ltd.

===Firmware Version: V40.19.0.30===
Applicable Model: TA1600/2400/3200
Release Date: April 6, 2016

New Features

1. Added **Phone Status** for FXS ports.
2. Added **Random SIP Port** feature for VoIP server.
3. Added support for **exporting configuration file** into a .xml file.
4. Added **Interdigit Timeout** setting.
5. Added **Blacklist** feature.
6. Added **Hook Flash Event** feature.
7. Added enable/disable options for **three-way calling**.
8. Added support for the following **time zones**:
 - 3 Kuwait (Al Kuwait)
 - -4 Curaçao(Willemstad)
 - 0 Ghana (Accra)
 - 1 Nigeria (Abuja)
 - 2 Mozambique (Maputo)
 - 3 Bahrain (Manama)
 - 3 Qatar (Doha)
 - 3 Tanzania (Dar es Salaam)
 - 4 Oman (Muscat)
 - 5.5 Sri Lanka (Colombo)
 - 6 Bangladesh (Dhaka)
 - 7 Indonesia (Jakarta)
 - 8 Malaysia (Kuala Lumpur)
 - 10 Papua New Guinea (Port Moresby)
 - 0 Iceland (Reykjavik)
 - -3 Uruguay (Montevideo)
 - 1 Bosnia and Herzegovina (Sarajevo)
 - 2 Lithuania (Vilnius)
 - 2 Palestine (Gaza)
 - 3 Kenya (Nairobi)
 - 3 Saudi Arabia (Riyadh)
 - 3 Yemen (Aden)
 - 4 Mauritius (Port Louis)
 - 5.75 Nepal (Kathmandu)
 - 6.5 Yangon (Myanmar)
 - 7 Vietnam (Ho Chi Minh City)
 - 8 Philippines (Manila)
 - 12 Fiji (Suva)

Optimization

1. The backup file created on TA1600/2400/3200 is encrypted. The backup file will be generated into a .bak file.
2. Users could edit FXS port numbers in bulk by clicking **Modify the Selected Port** button.
3. Increased the time interval between sending each REGISTER SIP packet.
4. Reduced the range of default RTP ports from 10000-12000 to 10000-10200.
5. Increased the length for Hostname field to 63 bytes.
6. Added bubble tips for Volume settings Rxgain and Txgain.

Bug Fixes

1. Fixed the issue that if session timer was enabled, and the Session-minise was modified, when you called in TA gateway, the phone connected to TA could not ring.
2. Fixed the issue that when you made a call from TA, and the other party hang up

the call, the LED (which indicates new voicemail message) on the analog phone would turn off.

3. Fixed the issue that TA gateway could not accept incoming calls without caller ID.
4. Fixed the issue that TA gateway could not send or receive fax with Huawei device. Before sending fax, Huawei device will send an INVITE packet with sdp that describes fax information (X-mode/X-fax/mode/fax). When TA gateway receives the INVITE message, it should respond 200OK that containing the same fax mode information.

Instruction (New Features)

1. Added Phone Status for FXS ports.

Path: Status > System Status > FXS Port Status

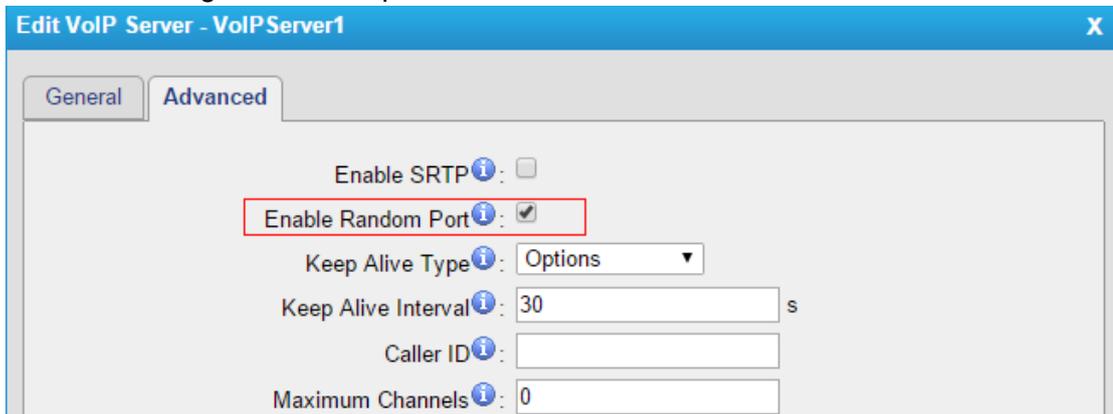
Instruction: this feature is supported on TA1600/2400/3200 with hardware version 1.3 or above. The phone status indicates whether the analog phone is connected to the FXS port or not.

Port	UP/Down/Break	Name	Status	Voice Mail(New/Old)	Off-hook/On-hook	Phone Status
1	Up	--	--	--	On Hook	Connected
2	Up	--	--	--	On Hook	Connected
3	Up	--	--	--	On Hook	Disconnected
4	Up	--	--	--	On Hook	Disconnected

2. Added Random SIP Port feature for VoIP Server.

Path: Gateway > VoIP Settings > VoIP Server Settings

Instruction: if the Random Port is enabled for a VoIP server, when registering using the VoIP server, TA will send a REGISTER packet with a random SIP port. The default range of random port is 5062-5162.



Edit VoIP Server - VoIP Server1

General | **Advanced**

Enable SRTP:

Enable Random Port:

Keep Alive Type: Options

Keep Alive Interval: 30 s

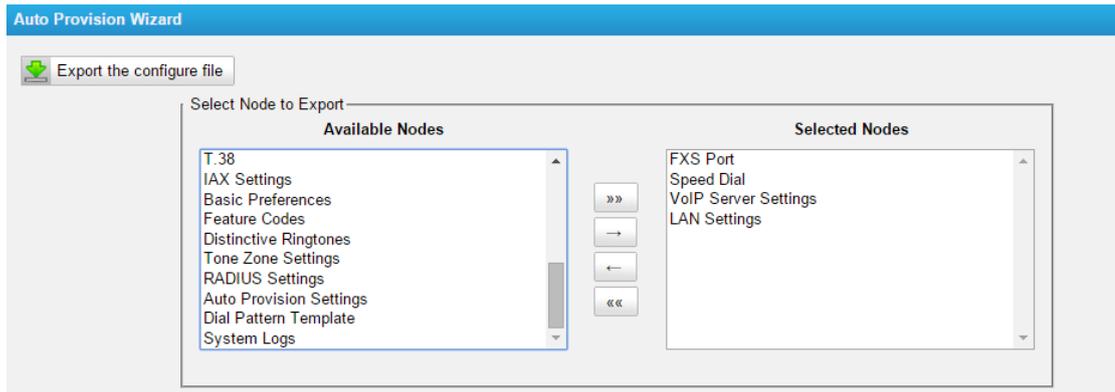
Caller ID:

Maximum Channels: 0

3. Added support for exporting configuration file into a .xml file.

Path: System > System Preferences > Auto Provision Settings

Instruction: select the nodes from the Available box to the Selected box, click "Export the Configuration File" button to export the file.



4. Added Interdigit Timeout setting.

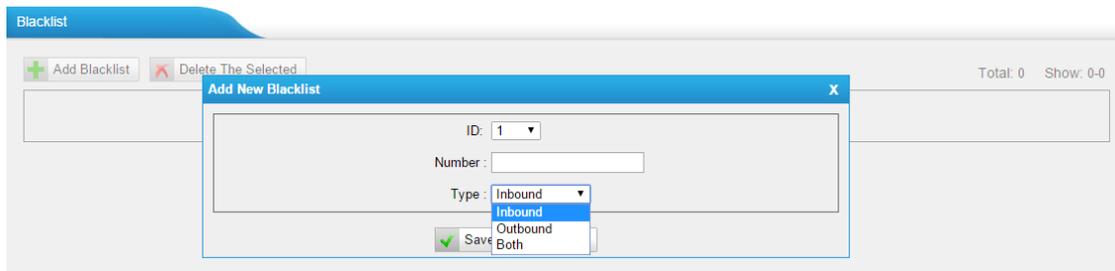
Path: Gateway > Gateway Settings > Basic Preferences

Instruction: the interdigit timeout is the time that you have between digits that you dial. The default value is 3 seconds.

5. Added Blacklist feature.

Path: Gateway > Gateway Settings > Blacklist

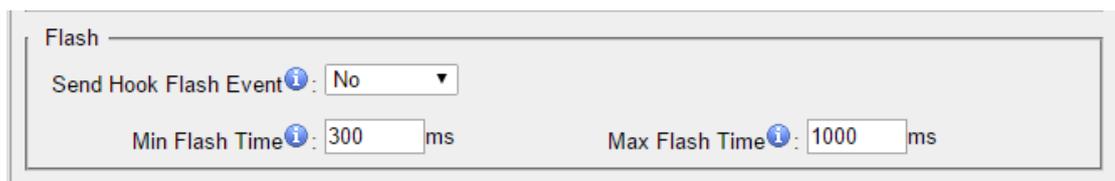
Instruction: this feature is used to limit inbound and outbound phone numbers.



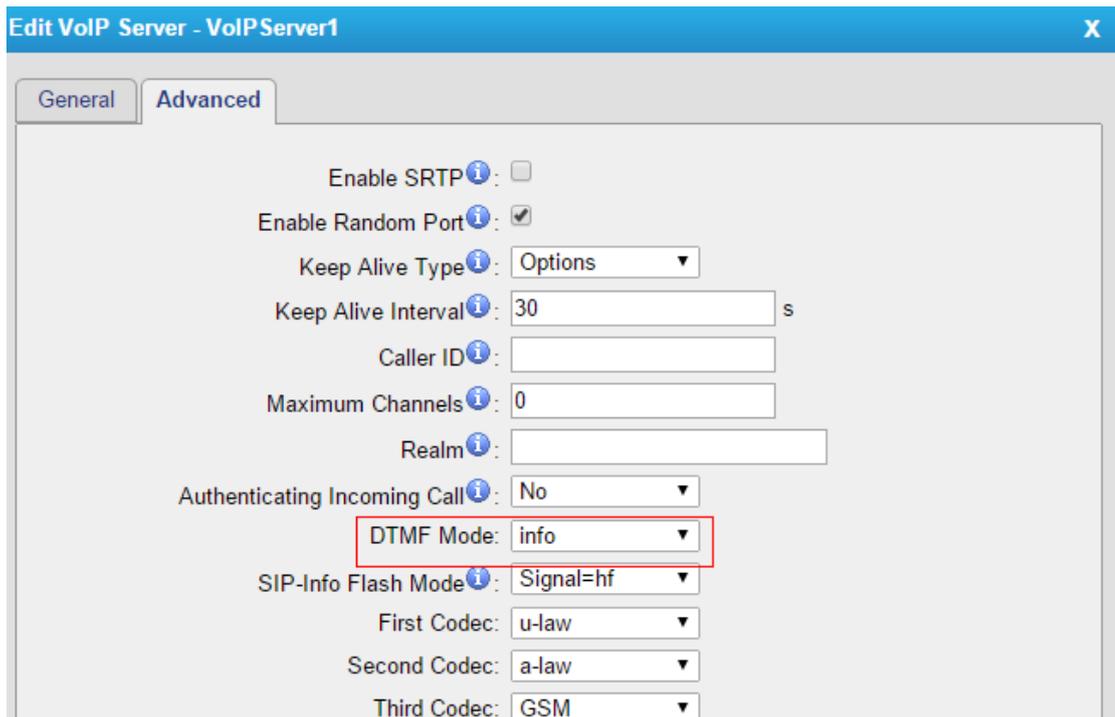
6. Added Hook Flash Event feature.

Path: Gateway > Port List > FXS Port List

Instruction: if Hook Flash Event is enabled, flash will send as DTMF event. Auto, rfc2833 or SIP info can be the signaling method of the Hook flash event.



The DTMF mode is set on VoIP server. For the info mode, you can choose Signal=hf or Signal=16.



7. Added enable/disable options for three-way calling.

Path: Gateway > Gateway Settings > Basic Preferences

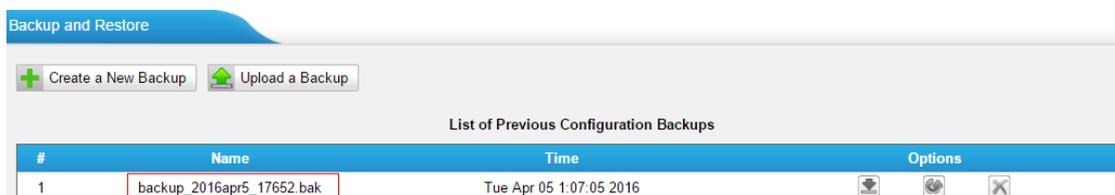
Instruction: in the former firmware version, when you pressed hook key during an active call, you would start to use three-way calling, but could not transfer a call using the hook key. In the new version, when you press the hook key during a call, the system will treat it as transferring by default, if you want to make a three-way call, you can enable the option “enable three-way calling”.

Instruction (Optimization)

1. The backup file created on TA1600/2400/3200 is encrypted. The backup file will be generated into a .bak file.

Path: System > System Preferences > Backup and Restore

Instruction:



2. Users could edit FXS port numbers in bulk by clicking Modify the Selected Port button.

Path: Gateway > Port List > FXS Port List

Instruction: when you edit selected FXS ports in bulk, you can edit the Name, number and other settings.

Modify the selected Port

General | Other Settings

Name
Prefix + Number starting from

Caller ID Number
Number starting from

VoIP Server Template
VoIP Server:

Password
 Fixed Password

3. Increased the time interval between each REGISTER SIP packet.

Instruction: the time interval is increased to 500 ms. You can edit the value in `/etc/asterisk/sip.conf`, modify the field "registerdelaytimer".

4. Reduce the range of default RTP ports from 10000-12000 to 10000-10200.

Path: Gateway > VoIP Settings > SIP Settings

Instruction: if you upgrade to the new version without reset, the old RTP port range setting will remain.

SIP Settings

General | NAT | QOS | Response Code | T.38 | **Advanced Settings**

UDP Port:

Enable TCP Port:

Enable TLS Port:

TLS Verify Server:

TLS Verify Client:

TLS Ignore Common Name:

TLS Client Method:

RTP Port Start:

RTP Port End:

Random SIP Port Start:

Random SIP Port End:

5. Increased the length for Hostname field to 63 bytes.

Path: System > Network Preferences > LAN Settings

Instruction:

LAN Settings

General Settings

Hostname: TA3200

Mode: Static IP Address

IP Address: 192.168.3.198

Subnet Mask: 255.255.255.0

Gateway: 192.168.3.1

Primary DNS: 192.168.1.1

Secondary DNS:

IP Address2:

Subnet Mask2:

6. Added bubble tips for Volume settings Rxgain and Txgain.

Path: Gateway > Port List > FXS Port List > Other Settings

Instruction:

Edit FXS Port - 1

General Other Settings

Other Options

Call Waiting DND Ring Out: 30

Follow me

Always Internal Port

Forward Type: No answer Destination: Hunt Group

When Busy Number

Prompt: No Music On Hold: None

Volume Settings

Rxgain: 20% Txgain: 20%

===Firmware Version: V40.19.0.14===
Applicable Model: TA1600/2400/3200
Release Date: September8, 2015

New Features

5. Added TR-069 feature.
6. Added Distinctive Ringtones feature.
7. Added Port Monitor Tool on web interface for FXS port debugging.
8. Added support for auto provision from a specified server.
9. Added Echo Cancellation enable and disable option for FXS ports.
10. Added Caller ID Sending Mode option for FXS ports.
11. Added support for G729A/B, G723 and iLBC codecs.
12. Added Ring Timeout setting for Hunt Group.
13. Added DSP Fax feature.
14. Added support for Broadsoft transfer mechanism.
15. Added Direct Caller ID Dialing feature for internal calls.
16. Added support for Pulse Dialing.
17. Added Virtual Ring Back Tone feature.
18. Added support for Serbia Tone Zone and Korea Tone Zone.

Optimization

1. Defined 3 register modes of VoIP Server.
2. Optimized Flash setting for FXS ports.
3. The characters [* , # -] are allowed in User Name filed and Authentication Name filed when registering FXS ports to a VoIP server.

Bug Fixes

1. Fixed the issue that if the FXS port's DID number contained any of the characters [n, x, z], no incoming calls would reach the FXS port.
2. Deleted the "Dial Timeout" setting on General Preferences page, which is a useless setting that limits the time to ring a device when calling out from TA gateway.

New Features (Instruction)

1. Added TR-069 feature.

Path: System→Network Preferences→TR-069 Settings

Instruction:

Service Providers, using TR-069, can have one common platform to manage all Yeastar TA devices.

TR069 Settings

General Settings

Enable TR069: Yes ▾

ACS Username:

ACS Password:

ACS URL:

Enable Periodic Inform: Yes ▾

Periodic Inform Interval(s):

Enable STUN:

STUN Server Address: Port:

STUN Server Username:

STUN Server Password:

Connection Request Address: Port:

Connection Request Username:

Connection Request Password:

2. Added Distinctive Ringtones feature.

Path: Gateway→Advanced Setting→Distinctive Ringtones

Instruction:

Users could configure different ringtones to match different incoming caller ID.

Edit Ringtones - 1

Ringtones ID:

Ring Cadence ⓘ:

Inbound Caller Pattern ⓘ:

Save Cancel

3. Added Port Monitor Tool on web interface for FXS port debugging.

Path: Status→Reports→Port Monitor Tool

Instruction:

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

Port Monitor Tool

Monitor Stopped

Port: (Port1) ▾

4. Added support for auto provision from a specified server.

Path: System→System Preferences→Auto Provision Settings

Instruction:

Besides provisioning Yeastar TA gateway with MyPBX, users could do auto provision from a specified server. Users could manually configure a provisioning server URL, TA gateway will get the configuration file from the server automatically and regularly.

Yeastar TA gateway supports HTTP, TFTP, FTP server.

AutoProvision Mode

Provision's Way:

PNP: No
 DHCP: No
 Server URL: Yes

Server Address:

Server URL
 User Name
 Password
 Interval of time 180 Minute
 Specified time Everyday : 00 : 00

Other:

AES Key
 Always Apply : No

5. Added Echo Cancellation enable and disable option for FXS ports.

Path: Gateway→Port List

Instruction:

If the port is connected to a POS machine, it is suggested that echo cancellation is disabled.

Edit FXS Port - 1

General Other Settings

General

Caller ID Name Caller ID Number

VoIP Server Template

VoIP Server : VoIPServer1(1)

Route Settings

Dial Pattern Template : DialPatternTemplate1(1) DID Number

Hotline

Enable Hotline : No
 Hotline Number Delay Dial : 2 s

Flash

Min Flash Time : 300 ms Max Flash Time : 1000 ms

Call Duration Setting

Max Call Duration : 6000 s

Echo Cancellation Setting

Enable Echo Cancellation : Yes

6. Added Caller ID Sending Mode option for FXS ports.

Path: Gateway→Port List

Instruction:

Choose the Caller ID Sending mode. TA gateway supports the following sending modes:

- Ring + Caller ID + Ring
- Caller ID + Ring
- Polarity + Caller ID + Ring

Note: If the FXS port is connected to a device's FXO port, Caller ID settings on FXS port and FXO port should be the same, or the call cannot be established.

The screenshot shows the 'Edit FXS Port - 1' configuration window with the 'Other Settings' tab selected. The 'Sending Mode' dropdown menu is highlighted with a red box and is set to 'Ring + Caller ID + Ring'. Other settings visible include 'Call Waiting' (unchecked), 'DND' (unchecked), 'Ring Out' (30), 'Follow me' options (Always, Internal Port, No answer, When Busy), 'Forward Type' (Destination: Hunt Group), 'Volume Settings' (Rxgain: 40%, Txgain: 40%), and 'Caller ID Settings' (Caller ID Signalling: FSK, Caller ID Type: Bell - USA).

7. Added support for G729A/B, G723 and iLBC codecs.

Path: Gateway→VoIP Settings→ SIP Settings

Instruction:

The screenshot shows the 'SIP Settings' configuration window with the 'Codecs' tab selected. The 'Allowed Codecs' list is highlighted with a red box and includes 'u-law', 'a-law', 'G723', 'G729A/B', and 'iLBC'. The 'Available Codecs' list includes 'ADPCM', 'G726', 'G722', and 'GSM'. The 'G.729 License Key' field is empty. A note at the bottom states: 'Note: If you would like to use G.729, please enter your license key above.'

8. Added Ring Timeout setting for Hunt Group.

Path: Gateway→ Port List → Hunt Group

Instruction:

Set the seconds to ring each port in the hunt group.

Edit Hunt Group

Group ID: 1

Group Name: HuntGroup1

Strategy: Simultaneous

Ring Timeout: 30 s

Group Members

Available FXS Port	Selected
	(Port1)
	(Port2)
	(Port3)
	(Port4)
	(Port5)
	(Port6)
	(Port7)
	(Port8)

Save Cancel

9. Added DSP Fax feature.

Path: Gateway → Gateway Settings → General Preferences

Instruction:

If the option is set to “Yes”, the system will process the Fax on DSP module, if set to “No”, Fax processing will be done on ARM module.

General Settings

MAX Call Duration: 6000 s

Music On Hold: calmriver

Enable Local Transfer: No

Key As Send: #

Enable DSP Fax: Yes

Internal Calling First: No

Virtual Ring Back Tone: No

Pulse Dialing: No

10. Added support for Broadsoft transfer mechanism.

Path: Gateway → Gateway Settings → General Preferences

Instruction:

By default, TA gateway use its own transfer mechanism. If this option is set to “No”, TA gateway will use the Broadsoft transfer mechanism.

Broadsoft transfer mechanism

- Blind transfer is implemented by a simple REFER method without Replaces in the REFER-TO header.
- Attended Transfer After Answer is implemented by a REFER method with *Replaces* in the REFER-TO header.

General Settings

MAX Call Duration: 6000 s

Music On Hold: calmriver

Enable Local Transfer: No

Key As Send: #

Enable DSP Fax: Yes

Internal Calling First: No

Virtual Ring Back Tone: No

Pulse Dialing: No

11. Added Direct Caller ID Dialing feature for internal calls.

Path: Gateway→Gateway Settings→General Preferences

Instruction:

If the option “Internal Calling First” is set to “Yes”, when making calls between internal ports (through Caller ID Number), the system will call the destination port directly, bypassing the VoIP Server.

If set to “No”, the system will call the destination port through the VoIP Server and only bypassing VoIP Server when the former fails.

General Settings

MAX Call Duration: 6000 s

Music On Hold: calmriver

Enable Local Transfer: No

Key As Send: #

Enable DSP Fax: Yes

Internal Calling First: No

Virtual Ring Back Tone: No

Pulse Dialing: No

12. Added support for Pulse Dialing.

Path: Gateway→Gateway Settings→General Preferences

Instruction:

Set “Pulse Dialing” to “Yes” to use pulse dialing.

General Settings

MAX Call Duration: 6000 s

Music On Hold: calmriver

Enable Local Transfer: No

Key As Send: #

Enable DSP Fax: Yes

Internal Calling First: No

Virtual Ring Back Tone: No

Pulse Dialing: No

13. Added Virtual Ring Back Tone feature.

Path: Gateway→Gateway Settings→General Preferences

Instruction:

Once the feature is enabled, when the caller dials out, the caller will only hear the virtual ring back tone generated by the system before the callee answers the call.

General Settings

MAX Call Duration: 6000 s

Music On Hold: calmriver

Enable Local Transfer: No

Key As Send: #

Enable DSP Fax: Yes

Internal Calling First: No

Virtual Ring Back Tone: No

Pulse Dialing: No

14. Added support for Serbia Tone Zone and Korea Tone Zone.

Path: Gateway → Advanced Settings → Tone Zone Settings

Instruction:

Users could choose the pre-configured Serbia Tone Zone and Korea Tone Zone.

Tone Zone Settings

Country/Region: United States / North America

Ring Cadence: Korea

Dial Tone: Lithuania

Ringback Tone: Malaysia

Busy Tone: Mexico

Call-Waiting Tone: Netherlands

Congestion Tone: New Zealand

2nd Dial Tone: Norway

Message Tone: Panama

VoiceMail Tone: Philippines

Poland

Portugal

Russian Federation

Serbia

Singapore

South Africa

Spain

+440/100,10/100,350+440

+440/100,10/100,1350+440/100

Optimization (Instruction)

1. Defined 3 register modes of VoIP Server.

Path: Gateway → VoIP Settings → VoIP Server

Instruction:

In the new version, we define 3 register modes for a VoIP server. Users could select one mode for the VoIP server and apply it to FXS ports.

- Service Provider** - IP Based VoIP Provider, do not generally require the TA gateway to register with the provider. Only IP address or domain is needed to configure on TA gateway.

Note: this is a peer-to-peer mode to connect VoIP server and TA gateway, you also need to create a service provider trunk connecting to TA gateway on your VoIP server.
- Port Register**—the VoIP server requires TA gateway to register with the provider using an authentication ID and password.

If you choose the VoIP server, you need to fill in User Name, Authentication Name and Password to register the FXS port.

- **Template Register**—register to your VoIP server and apply the template to FXS ports.

2. Optimized Flash setting for FXS ports.

Path: Gateway→Port List

Instruction:

The TA gateway will regard the flash duration between Min Flash and Max Flash as effective flash. Any flash lasting over the longest time will be considered by gateway as hang-up.

The screenshot shows a configuration window titled "Edit FXS Port - 1" with a blue header bar. There are two tabs: "General" (selected) and "Other Settings". The configuration is organized into several sections:

- General:** Contains "Caller ID Name" and "Caller ID Number" text input fields.
- VoIP Server Template:** Contains a "VoIP Server" dropdown menu set to "MyPBX_SPS(1)".
- Route Settings:** Contains a "Dial Pattern Template" dropdown menu set to "DialPatternTemplate1(1)" and a "DID Number" text input field.
- Hotline:** Contains an "Enable Hotline" dropdown menu set to "No", a "Hotline Number" text input field, and a "Delay Dial" spinner set to "2" seconds.
- Flash:** This section is highlighted with a red rectangular box. It contains two text input fields: "Min Flash Time" set to "300" ms and "Max Flash Time" set to "1000" ms.

[The End]