



## Release Note for TA400/800

Version 41.19.0.X

**Yeastar Information Technology Co. Ltd.**

**===Firmware Version: V41.19.0.32===****Applicable Model: TA400/800****Release Date: April 15, 2017****New Features**

1. Added **SNMP** feature (System > SNMP Settings).
2. Added **WAN Settings** for TA800 new hardware with two Ethernet ports. Added support for VLAN over WAN, static routing through WAN port, Auto Provisioning through WAN port.
3. Added support for **Auto Provisioning** through an **https** server (System > Auto Provision Settings > Server URL).
4. Added **Phone Status** for FXS ports to monitor if analog phones are connected to FXS ports or not.
5. Added **Random SIP Port** feature for VoIP server.
6. Added support for **exporting configuration file** into a .xml file (System > System Preferences > Auto Provision Settings).
7. Added **Interdigit Timeout** setting (Gateway > Gateway Settings > Basic Preferences).
8. Added **Blacklist** feature (Gateway > Gateway Settings > Blacklist).
9. Added **Hook Flash Event** feature for FXS ports.
10. Added enable/disable options for **three-way calling** (Gateway > Gateway Settings > Basic Preferences).
11. Added support for the following **time zones**:

• 3 Kuwait (Al Kuwait)	• 0 Iceland (Reykjavik)
• -4 Curaçao(Willemstad)	• -3 Uruguay (Montevideo)
• 0 Ghana (Accra)	• 1 Bosnia and Herzegovina (Sarajevo)
• 1 Nigeria (Abuja)	• 2 Lithuania (Vilnius)
• 2 Mozambique (Maputo)	• 2 Palestine (Gaza)
• 3 Bahrain (Manama)	• 3 Kenya (Nairobi)
• 3 Qatar (Doha)	• 3 Saudi Arabia (Riyadh)
• 3 Tanzania (Dar es Salaam)	• 3 Yemen (Aden)
• 4 Oman (Muscat)	• 4 Mauritius (Port Louis)
• 5.5 Sri Lanka (Colombo)	• 5.75 Nepal (Kathmandu)
• 6 Bangladesh (Dhaka)	• 6.5 Yangon (Myanmar)
• 7 Indonesia (Jakarta)	• 7 Vietnam (Ho Chi Minh City)
• 8 Malaysia (Kuala Lumpur)	• 8 Philippines (Manila)
• 10 Papua New Guinea (Port Moresby)	• 12 Fiji (Suva)

**Optimization**

1. Optimized Flash Transfer: supported REFER transfer when using flash key to transfer calls.
2. Optimized call flow: the INVITE packet will not contain callerid number in From field if the Caller ID Name is not set on the FXS port.
3. Optimized TR069 feature: stepped up TA gateway's respond to the ACS actions.

4. Added compatibility with Draytek PBX.
5. Added compatibility with SIP trunk with multiple DID numbers.
6. Optimized SIP Outgoing Registration Time setting: the default value is set to 1800s.
7. The backup file created on TA1600/2400/3200 is encrypted. The backup file will be generated into a .bak file.
8. Users could edit FXS port numbers in bulk by clicking **Modify the Selected Port** button.
9. Increased the time interval between sending each REGISTER SIP packet.
10. Reduced the range of default RTP ports from 10000-12000 to 10000-10200.
11. Increased the length for Hostname field to 63 bytes.
12. Added bubble tips for Volume settings Rxgain and Txgain.

### Bug Fixes

1. Fixed the issue that if session timer was enabled, and the Session-minse 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.
5. Fixed Attended Transfer issue: transferred a call to a 3<sup>rd</sup> party, then hang up the call if the 3<sup>rd</sup> party didn't answer the call, the call could not be successfully transferred.
6. Fixed the compatibility with China IMS: if the trunk user name contained character "@", this trunk could not receive inbound calls.
7. Fixed FlashTransfer issue: if the FXS port enabled T.38, using flash key to transfer an inbound call would fail.
8. Fixed the issue that "Allow Guest" setting could not take effect (SIP Settings > Advanced Settings).
9. Fixed the issue that "DNS SRV Look Up" settings could not take effect (SIP Settings > General).
10. Corrected Australia Tonezone settings.

**===Firmware Version: V41.19.0.16===**

**Applicable Model: TA400/800**

**Release Date: November 24, 2015**

## New Features

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

## Optimization

1. Defined 3 register modes of VoIP Server.
2. Optimized Flash setting for FXS ports.
3. The characters [\* , # - ] are allowed in User Name field and Authentication Name field 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.

The screenshot shows the 'Edit FXS Port - 1' configuration window with the following settings:

- General:** Caller ID Name: 460, Caller ID Number: 460
- VoIP Server Template:** VoIP Server: VoIPServer1(1), User Name: 460, Authentication Name: 460, Password: [redacted], From User: [empty]
- Route Settings:** Dial Pattern Template: DialPatternTemplate1(1), DID Number: 460
- Hotline:** Enable Hotline: No, Hotline Number: [empty], Delay Dial: 2 s
- Flash:** Send Hook Flash Event: No, 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 window is divided into several sections:

- Other Options:** Includes checkboxes for 'Call Waiting' and 'DND', and a 'Ring Out' field set to 30.
- Follow me:** Includes checkboxes for 'Always' and 'Internal Port', and radio buttons for 'Hunt Group' (selected) and 'Number'. The 'Hunt Group' dropdown is set to 'HuntGroup1(Group1)'. There are also checkboxes for 'No answer' and 'When Busy', and a 'Prompt' dropdown set to 'No'.
- Volume Settings:** Includes 'Rxgain' and 'Txgain' dropdowns, both set to 40%.
- Caller ID Settings:** Includes 'Caller ID Signalling' set to 'FSK', 'Caller ID Type' set to 'Bell - USA', and 'Sending Mode' set to 'Ring + Caller ID + Ring' (highlighted with a red box).

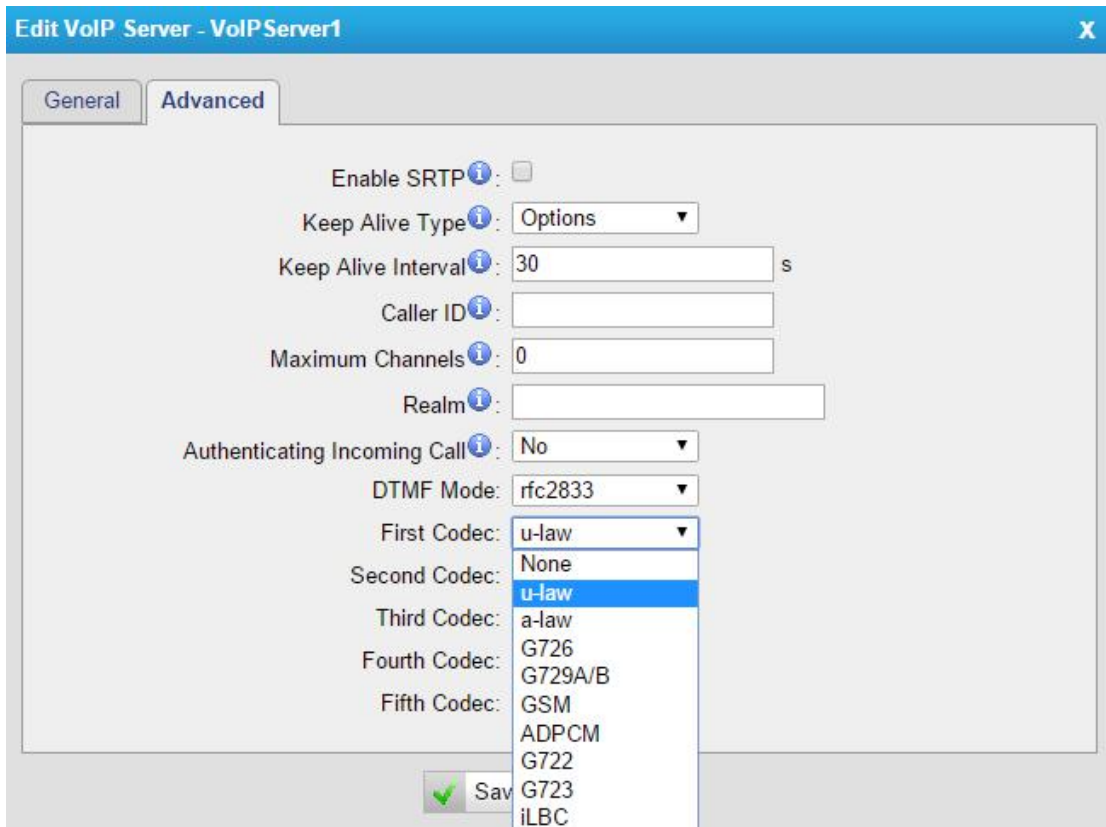
## 7. Added support for G729A/B, G723 and iLBC codecs.

**Path:** Gateway→ VoIP Settings→ VoIP Server Settings

### **Instruction:**

Codec tab is removed in SIP settings page. And it is added support for G729A/B, G723 and iLBC codecs.



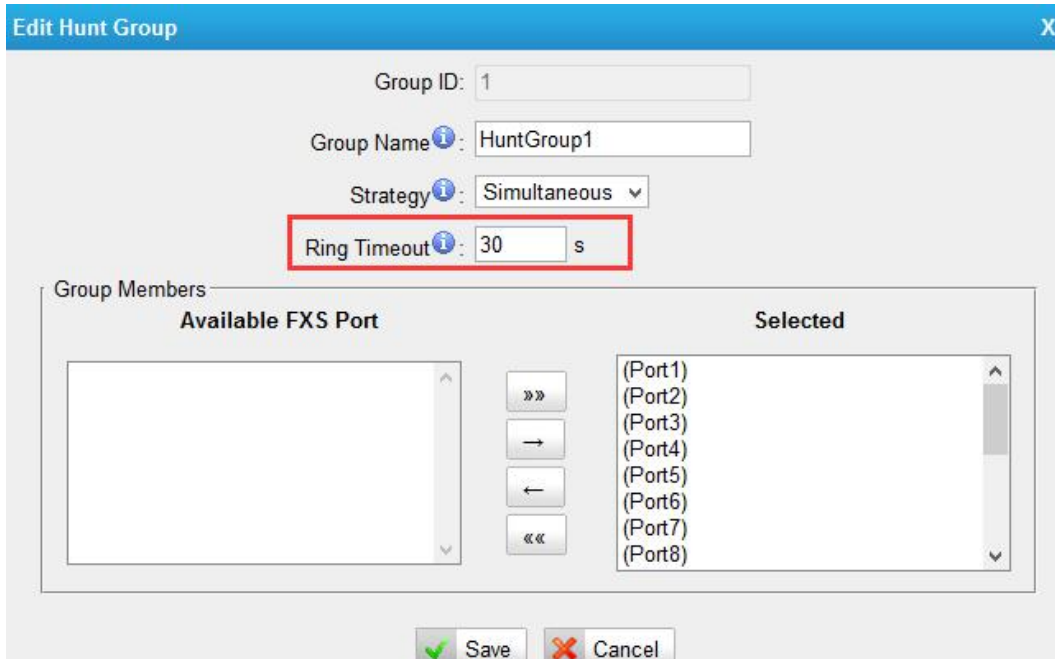


#### 8. Added Ring Timeout setting for Hunt Group.

**Path:** Gateway → Port List → Hunt Group

**Instruction:**

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



#### 9. Added DSP Fax feature.

**Path:** Gateway → Gateway Settings → Basic Preferences → General Settings

**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

Music On Hold: calmriver ▾

Enable Local Transfer ⓘ: Yes ▾

Key As Send: # ▾

Enable DSP Fax: Yes ▾

Internal Calling First ⓘ: No ▾

Virtual Ring Back Tone ⓘ: No ▾

Enable Call Logs ⓘ: Yes ▾

## 10. Added support for Broadsoft transfer mechanism.

**Path:** Gateway→ Gateway Settings→ Basic Preferences→ General Settings

### 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

Music On Hold: calmriver ▾

Enable Local Transfer ⓘ: Yes ▾

Key As Send: # ▾

Enable DSP Fax: Yes ▾

Internal Calling First ⓘ: No ▾

Virtual Ring Back Tone ⓘ: No ▾

Enable Call Logs ⓘ: Yes ▾

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

**Path:** Gateway→ Gateway Settings→ Basic Preferences→ General Settings

### 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

Music On Hold: calmriver ▾

Enable Local Transfer ⓘ: Yes ▾

Key As Send: # ▾

Enable DSP Fax: Yes ▾

Internal Calling First ⓘ: No ▾

Virtual Ring Back Tone ⓘ: No ▾

Enable Call Logs ⓘ: Yes ▾

## 12. Added Virtual Ring Back Tone feature.

**Path:** Gateway→ Gateway Settings→ Basic Preferences→ General Settings

**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.

The screenshot shows the 'General Settings' page with several configuration options. The 'Virtual Ring Back Tone' dropdown menu is highlighted with a red box and is currently set to 'No'. Other options include Music On Hold (calmriver), Enable Local Transfer (Yes), Key As Send (#), Enable DSP Fax (Yes), Internal Calling First (No), and Enable Call Logs (Yes).

## 13. 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.

The screenshot shows the 'Tone Zone Settings' page. The 'Country/Region' dropdown menu is open, showing a list of countries and regions. 'Korea' and 'Serbia' are highlighted with red boxes. Other countries listed include Japan, Lithuania, Malaysia, Mexico, Netherlands, New Zealand, Norway, Panama, Philippines, Poland, Portugal, Russian Federation, Singapore, and South Africa.

## 14. Added support for Hook Flash Event.

**Path:** Gateway→ Port List

**Instruction:**

If the option “Send Hook Flash Event” is set to “Yes”, when pressing the flash key on the analog phone during an active call. TA will send a SIP DTMF flash event to the VoIP server. It can be an interoperation with a FXO gateway which is used for informing FXO gateway to generate the flash signal to the CO line.

The screenshot shows the 'Edit FXS Port - 1' configuration window. The 'Flash' section is highlighted with a red box, showing the 'Send Hook Flash Event' dropdown menu set to 'No'. Other visible settings include: Caller ID Name: 460, Caller ID Number: 460, VoIP Server Template: VoIPServer1(1), VoIP Server: VoIPServer1(1), User Name: 460, Authentication Name: 460, Password: [redacted], From User: [empty], Dial Pattern Template: DialPatternTemplate1(1), DID Number: 460, Enable Hotline: No, Hotline Number: [empty], Delay Dial: 2 s, Min Flash Time: 300 ms, and Max Flash Time: 1000 ms.

### 15. Added Adaptive option for Jitter Buffer.

**Path:** Gateway→ Gateway Settings→ Basic Preferences→ Voice Settings

**Instruction:**

If you select the “Adaptive” option, it will calculate a best buffer size according to the current network conditions. If the buffer size is less than the one you filled in the Jitter Buffer Max Size, TA will use this buffer size instead of the one you filled in. Otherwise, TA will use the max size you filled in.

The screenshot shows the 'Voice Settings' configuration window. The 'Enable Jitter Buffer' dropdown menu is highlighted with a red box and set to 'Adaptive'. Other visible settings include: Jitter Buffer Max Size: 40 ms, Echo Tail Length: 128 ms, G723 Encoding Rate: 6.3 Kbps, and iLBC Frame Size: 30 ms.

### 16. Added Enable Call Logs option.

**Path:** Gateway→ Gateway Settings→ Basic Preferences→ General Settings

**Instruction:**

If this option is set to “Yes”, the system will store the call logs. If not, the system won’t keep call logs any more, but the logs stored previously will be still there.

**General Settings**

Music On Hold:  ▾

Enable Local Transfer ⓘ:  ▾

Key As Send:  ▾

Enable DSP Fax:  ▾

Internal Calling First ⓘ:  ▾

Virtual Ring Back Tone ⓘ:  ▾

**Enable Call Logs ⓘ:  ▾**

## 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.

The screenshot shows the 'Edit FXS Port - 1' configuration window. It has two tabs: 'General' and 'Other Settings'. The 'General' tab is selected. Under the 'General' section, there are two input fields: 'Caller ID Name' and 'Caller ID Number', both containing the value '301'. Below this is the 'VoIP Server Template' section, which is highlighted with a red border. It contains four fields: 'VoIP Server' (a dropdown menu showing 'pbx(2)'), 'User Name' (containing '301'), 'Authentication Name' (containing '301'), and 'Password' (masked with dots). There is also a 'From User' field which is empty.

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

Edit VoIP Server - VoIPServer3

General Advanced

Server ID: 3

Server Name: VoIPServer3

Type: SIP

Transport: UDP

Hostname/IP: 192.168.6.162 : 5060

Domain: 192.168.6.162

Enable Outbound Proxy Server

Failover Hostname/IP: : 5060

Register Mode: Template Register

User Name: 300

Authentication Name: 300

Password: ••••••••

From User:

## 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.

**Edit FXS Port - 1** [X]

**General** | Other Settings

**General**

Caller ID Name: 460      Caller ID Number: 460

**VoIP Server Template**

VoIP Server: VoIPServer2(2) ▼

User Name: 460      Authentication Name: 460

Password: .....      From User:

**Route Settings**

Dial Pattern Template: DialPatternTemplate1(1) ▼      DID Number: 460

**Hotline**

Enable Hotline: No ▼

Hotline Number:      Delay Dial: 2 s

**Flash**

Send Hook Flash Event: No ▼

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

**Call Duration Setting**

Max Call Duration: 6000 s

[The End]