



TB200/400
User Manual

Version 18.18.0.2

Yeastar Information Technology Co. Ltd.

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Introduction

Yeastar TB200/400 is a compact and reliable standalone VoIP BRI gateway (BRI-VoIP/VoIP-BRI) offering 2 or 4 BRI ports for companies using ISDN BRI lines an easy, cost-effective and flexible integration into any VoIP system or enabling any IP PBX to be connected to the public ISDN network.

Features

● 2 or 4 BRI ports
● Programmable NT/TE modes
● Type of connection: Point to Point, Point to Multipoint
● T.38 FAX
● Flexible number manipulation
● Least cost routing
● SIP Registrar for IP phones
● Simple Web-based management
● Trace and debug tools for diagnostics

For more information, please click:

<http://www.yeastar.com/Products/BRI-VoIP-Gateway-TB200-&-TB400>

Yeastar TB200/400 BRI Gateway features 2 or 4 BRI interfaces for connection of BRI providers one 10/100 Mbps LAN port.

For more information about the TB200/400 hardware specification and how to install the TB200/400, please refer to the document below:

http://www.yeastar.com/download/Yeastar_TB_BRI_VoIP_Gateways_Installation_Guide_en.pdf

Application Description

TB200/400 BRI VoIP Gateway supports up to 4 or 8 simultaneous phone calls from SIP to ISDN BRI or from BRI to SIP. TB200/400 is interoperable with most IP PBX and Unified Communication vendors such as MyPBX, Elastix, Asterisk, 3CX, Skype etc.

Three modes are available for you to connect your SIP server and TB200/400 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 TB200/400. SPS Mode is recommended.

Account Mode:

Create one SIP account on TB200/400, and take the SIP account to register one SIP trunk on your SIP server. Then TB400 and your SIP server are connected by the account.

➤ **Calls from SIP to BRI**

- 1) Create one outbound route on your SIP sever, and select the SIP trunk you have registered just now.
- 2) Configure a route on TB200/400, choose the SIP account in the field "Call Comes in From", and choose BRI trunk in the field "Send calls Through".
- 3) Make a call from your SIP Server and the call should match the outbound route dial rules.

➤ **Calls from BRI to SIP**

- 1) Create an inbound route on your SIP server, and select the SIP trunk you have registered just now.
- 2) Configure another route on TB200/400, choose BRI trunks in the field "Call Comes in From", and choose the SIP account in the filed "Send Calls Through".
- 3) When a call comes to BRI trunk on TB200/400, 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 BRI trunk on TB200/400 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 TB200/400 as a VoIP trunk. In this way, TB200/400 and your SIP server are connected.

- **Calls from SIP to BRI**
 - 1) Configure a route on TB200/400; choose the VoIP trunk in the field “Call Comes in From”, and choose BRI trunk in the field “Send calls Through”. **Enable Two-stage Dialing** on the route.
 - 2) Make a call from your SIP server, dial the SIP account number which is registered on TB200/400. You will hear a dial tone or two-stage dialing prompt, then dial the external number out through BRI trunk.
- **Calls from BRI to SIP**
 - 1) Configure another route on TB200/400, choose BRI trunks in the field “Call Comes in From”, and choose the SIP trunk in the field “Send Calls Through”. **Enable Two-stage Dialing** on the route.
 - 2) When an incoming call reaches BRI trunk on TB200/400, you will hear a dial tone or two-stage dialing prompt, then dial an extension number of the SIP server.

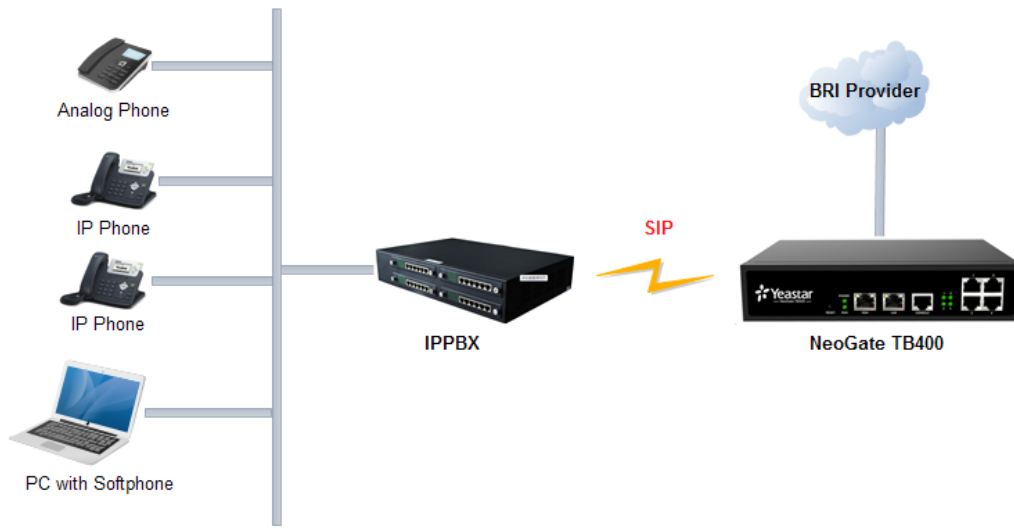
SPS Mode(Recommended)

Create a Service Provider SIP trunk on TB200/400 to connect to your SIP server. Add another Service Provider SIP trunk on your SIP server, connecting to TB200/400.

- **Calls from SIP to BRI**
 - 1) Create one outbound route on your SIP sever, and select the SIP trunk you have created just now.
 - 2) Configure a route on TB200/400, choose the SPS trunk in the field “Call Comes in From”, and choose BRI trunk in the field “Send calls Through”.
 - 3) Make a call from your SIP Server and the call should match the outbound route dial rules.
- **Calls from BRI to SIP**
 - 1) Configure another route on TB200/400, choose BRI trunks in the field “Call Comes in From”, and choose the SPS trunk in the field “Send Calls Through”.
 - 2) Create one inbound route on your SIP server and select the SIP trunk created just now.
 - 3) When an incoming call reaches BRI trunk on TB200/400, 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 TB200/400 route.

Typical Application



Typical Application

Configuration Guide

1. Login

TB200/400 provides web-based configuration interface for administrator. The user can manage the device by logging in the web interface. Check the factory defaults below:

IP address: <http://192.168.5.150>

User Name: **admin**

Default Password: **password**

In this guide, the IP address of TB200/400 is <http://192.168.6.125>.

1. Start the browser on PC. In the address bar, enter the IP address, click “Enter” button and then you can see the Web Configuration Panel login page (see Figure 1-1).
2. Enter the Admin User Name and Password to log in.

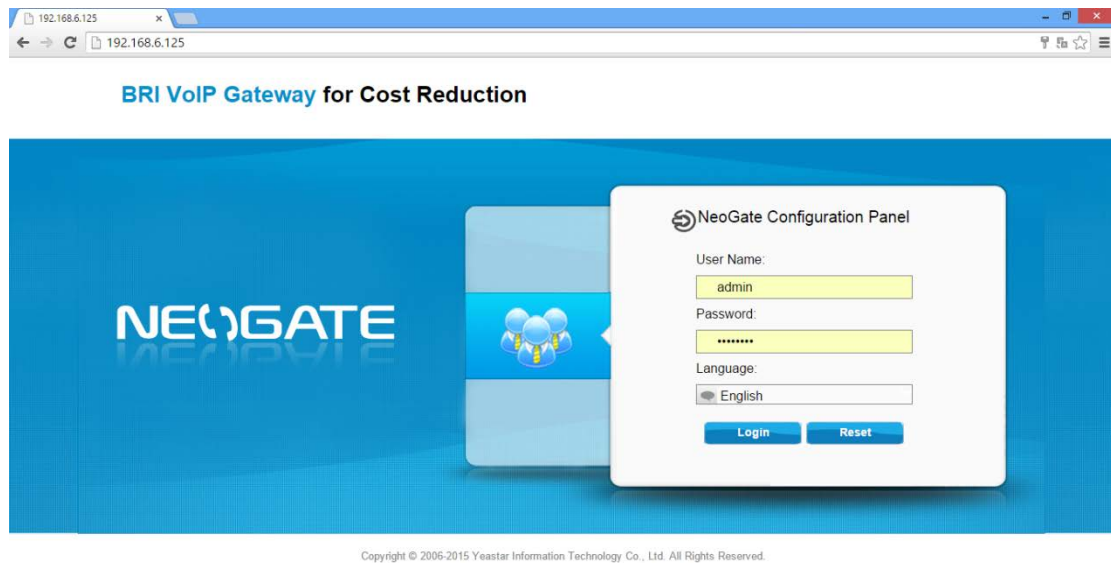



Figure 1-1 Web Configuration Panel Login Page

2. Status



Click  to check the status of TB200/400, including the system status and the detailed reports.

2.1 System Status

In this page, we can check the status of the system, including trunk status, network status and system information.

2.1.1 IP Trunk Status

Status of all the SIP trunks and SIP accounts are displayed on this page.

SIP Type

Table 2-1 Description of SIP Trunk Status

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

SP-SIP Type

Table 2-2 Description of SP-SIP Trunk Status

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

SIP Account

Table 2-3 Description of SIP Account Status

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

2.1.2 BRI Status

On this page, you can check the status of BRI trunks. If there is no BRI module inserted on TB400, you cannot see any BRI trunk here.

Table 2-4 Description of SIP Account Status

Status	Description
OK	The BRI trunk is connected and configured correctly, trunk is ready to use.
Disconnected	The BRI trunk is not connected or configured wrong.

2.1.3 Network Status

In this page, the IP address of LAN port will appear with their status. If your VLAN or VPN are configured, you can check the status in this page also.

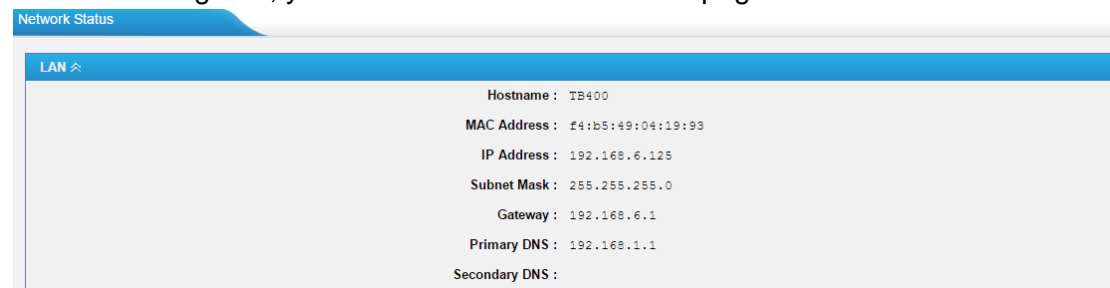


Figure 2-1 Network Status

2.1.4 System Info

In this page, you can check the hardware/firmware version, or the disk usage of TB200/400.

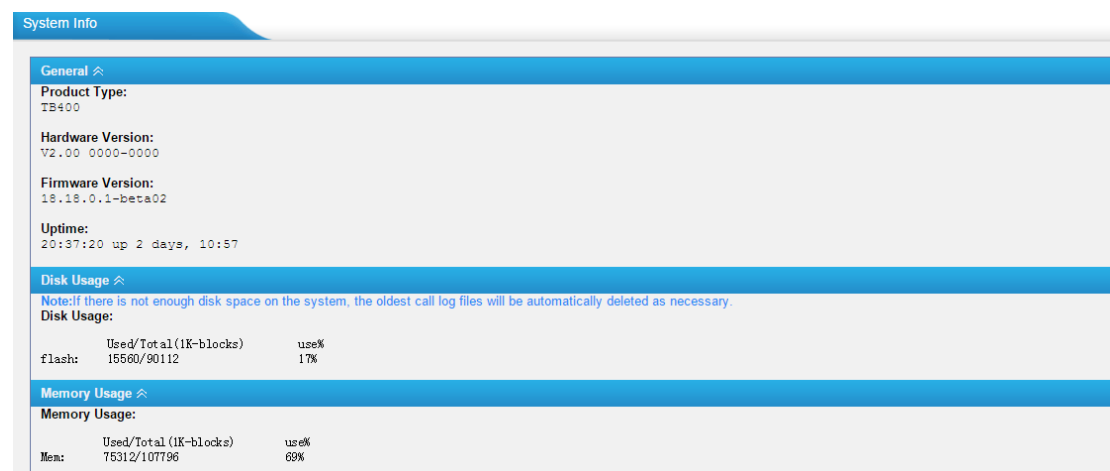


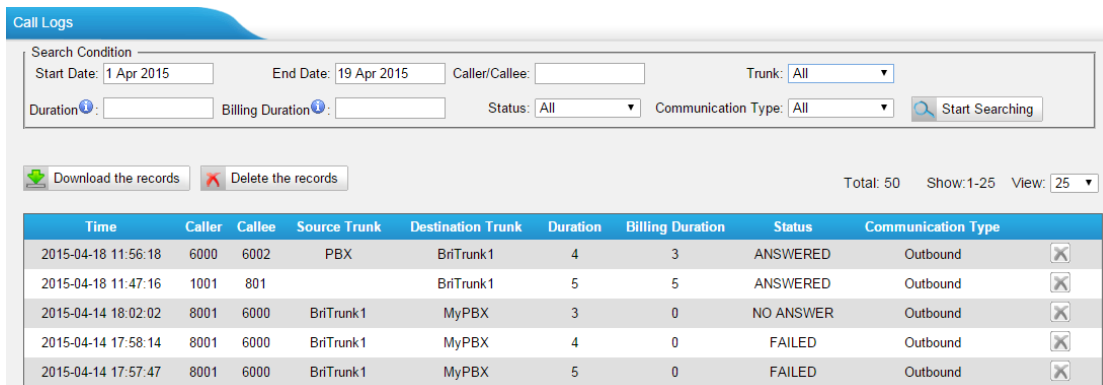
Figure 2-2 System Info

2.2 Reports

In this page, you can check the call detailed log, system log, and use the packet tool to debug the system when needed.

2.2.1 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.



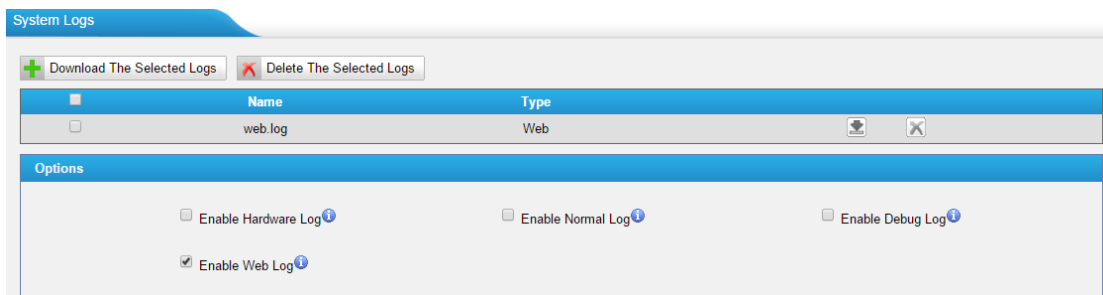
The screenshot shows the 'Call Logs' interface. At the top, there is a search bar with the following filters: Start Date (1 Apr 2015), End Date (19 Apr 2015), Caller/Callee (empty), Trunk (All), Duration (empty), Billing Duration (empty), Status (All), and Communication Type (All). There is a 'Start Searching' button. Below the search bar are two buttons: 'Download the records' and 'Delete the records'. On the right, it says 'Total: 50 Show: 1-25 View: 25'. The main part of the interface is a table with the following columns: Time, Caller, Callee, Source Trunk, Destination Trunk, Duration, Billing Duration, Status, and Communication Type. The table contains five rows of call records.

Time	Caller	Callee	Source Trunk	Destination Trunk	Duration	Billing Duration	Status	Communication Type
2015-04-18 11:56:18	6000	6002	PBX	BriTrunk1	4	3	ANSWERED	Outbound
2015-04-18 11:47:16	1001	801		BriTrunk1	5	5	ANSWERED	Outbound
2015-04-14 18:02:02	8001	6000	BriTrunk1	MyPBX	3	0	NO ANSWER	Outbound
2015-04-14 17:58:14	8001	6000	BriTrunk1	MyPBX	4	0	FAILED	Outbound
2015-04-14 17:57:47	8001	6000	BriTrunk1	MyPBX	5	0	FAILED	Outbound

Figure 2-3 Call Logs

2.2.2 System Logs

You can download and delete the system logs of TB200/400.



The screenshot shows the 'System Logs' interface. At the top, there are two buttons: 'Download The Selected Logs' and 'Delete The Selected Logs'. Below these is a table with two columns: 'Name' and 'Type'. The table contains one row with 'web.log' in the Name column and 'Web' in the Type column. Below the table is an 'Options' section with four checkboxes: 'Enable Hardware Log', 'Enable Normal Log', 'Enable Debug Log', and 'Enable Web Log'. The 'Enable Web Log' checkbox is checked.

Name	Type
web.log	Web

Options:

- Enable Hardware Log
- Enable Normal Log
- Enable Debug Log
- Enable Web Log

Figure 2-4 System Logs

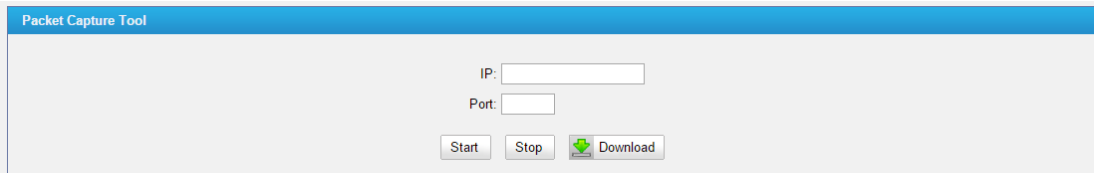
- **Enable Hardware Log**
Save the information of hardware; (up to 4 log files)
- **Enable Normal Log**
Save the prompt information; (up to 16 log files)
- **Enable Web Log**
Save the history of web operations (up to 2 log files)

- **Enable Debug Log**
Save debug information (up to 2 log files)

Packet Tool

TB200/400 provides a tool to capture packets for technician. Packet capture tool “Wireshark” is integrated in TB200/400.

Users also could specify the destination IP address and port to get the packets.




The screenshot shows a web-based interface for a packet capture tool. The title bar reads "Packet Capture Tool". The main area contains two text input fields, one labeled "IP:" and one labeled "Port:". Below the input fields are three buttons: "Start", "Stop", and "Download". The "Download" button includes a small green icon of a downward-pointing arrow.

Figure 2-5 Packet Tool

3. System



Click  to access. In this page, we can configure the network settings, security settings and some system preferences.

3.1 Network Preferences

3.1.1 LAN Settings

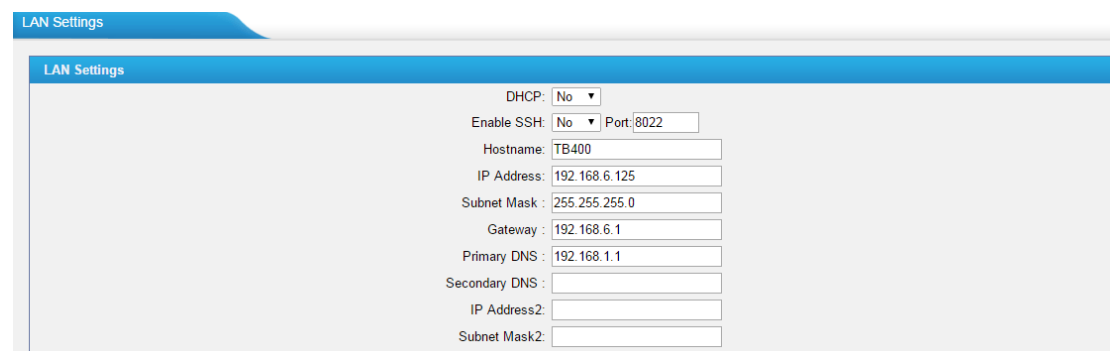


Figure 3-1LAN Settings

Table 3-1 Description of LAN Settings

Items	Description
DHCP	If this option is set as yes, TB200/400 will act as DHCP client to get an available IP address from your local network. We don't recommend enabling this, as without the right IP address you cannot access TB200/400.
Enable SSH	By using SSH, you can log in to TB200/400 and run commands. It's disabled by default. We don't recommend enabling it if not needed. The default port for SSH is 8022;
Hostname	Set the host name for TB200/400.
IP Address	Set the IP Address for TB200/400.
Subnet Mask	Set the subnet mask for TB200/400.
Gateway	Set the gateway for TB200/400.
Primary DNS	Set the primary DNS for TB200/400.
Secondary DNS	Set the secondary DNS for TB200/400.
IP Address2	Set the second IP Address for TB200/400.
Subnet Mask2	Set the second subnet mask for TB200/400.

3.1.2 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 3-2 DDNS Settings

Table 3-2 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.

3.1.3 Static Route

TB200/400 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 TB200/400 to force it to go out through different gateway when accessing to different internet.

The default gateway priority of TB200/400 from high to low is VPN/VLAN→LAN port.

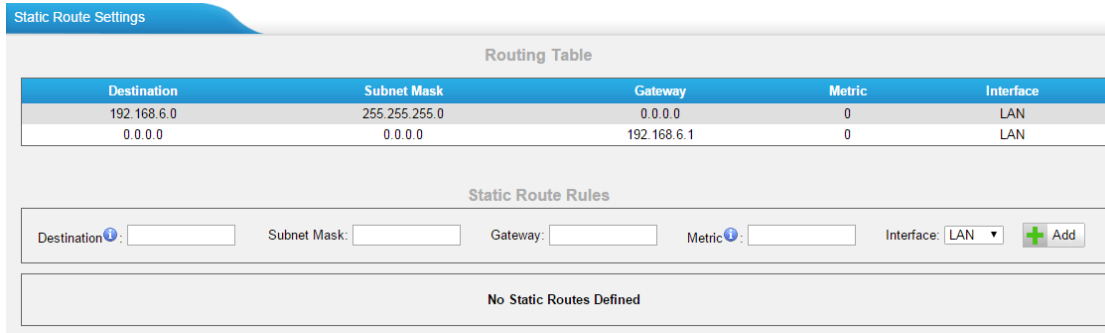


Figure 3-3 Static Route

1) Route Table

The current route rules of TB200/400.

2) Static Route Rules

You can add new static route rules here.

Table 3-3 Description of Static Route

Items	Description
Destination	The destination network to be accessed to by TB200/400.
Subnet Mask	Specify the destination network portion.
Gateway	Define which gateway TB200/400 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.

3.2 Security Center

3.2.1 Certificates

TB200/400 can support TLS trunk. Before you register TLS trunk to TB200/400, you should upload certificates first.

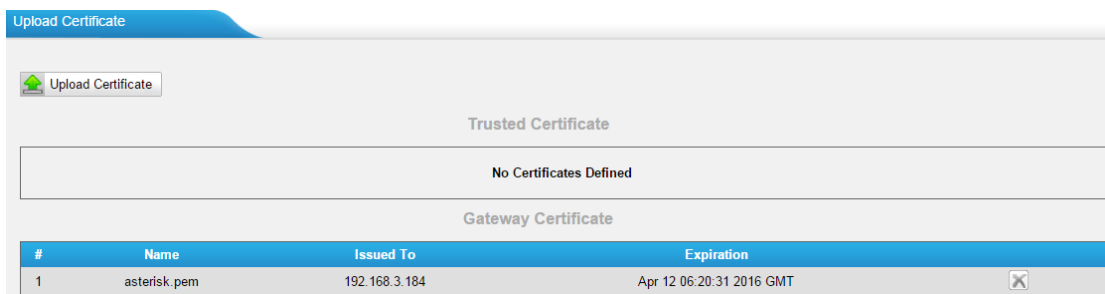


Figure 3-4 Certificates

- **Trusted Certificate**

This certificate is a CA certificate. When selecting “TLS Verify Client” as “Yes”, you should upload a CA. The relevant IPPBX 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 TB200/400. If IPPBX enables “TLS Verify server”, you should also upload this certificate on IPPBX.

3.2.2 Firewall Rules



Figure 3-5 Firewall Rules

1) General Settings

Table 3-4 Description of Firewall General Settings

Items	Description
Enable Firewall	Enable the firewall to protect the device. You should reboot the device to make the firewall run.
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 3-6 Common Rules

Table 3-5 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 192.168.5.0 to 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.

3.2.3 IP Blacklist

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.

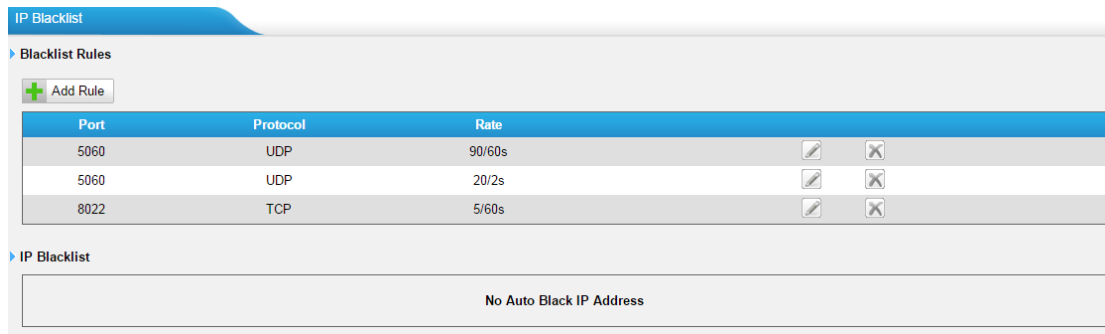


Figure 3-7 IP Blacklist

1) Blacklist rules

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

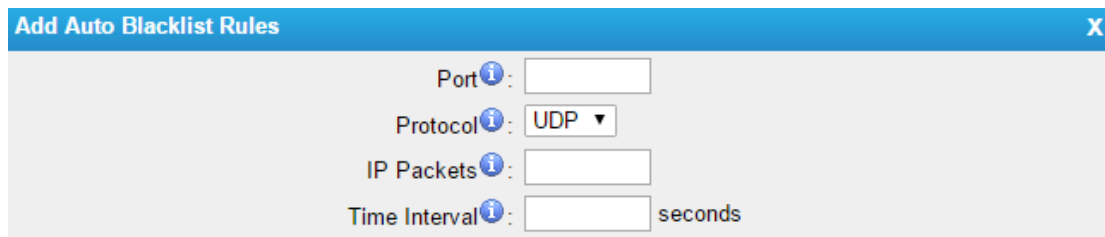


Figure 3-8 Auto Blacklist Rule

Table 3-6 Description of Auto Blacklist 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 blacklist

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

3.3 System Preferences

In this page, we can set other system preferences, like the password for admin account, system date and time, firmware update, backup and restore, reset and reboot.

3.3.1 Password Settings

The default password is “**password**”. To change the password, enter the new password and click “Save”. The system will then prompt you to re-login using your new password.

Figure 3-9 Password Settings

3.3.2 Date and Time

Set the date and time for TB200/400.

Figure 3-10 Date and Time

Table 3-7 Description of Date and Time Settings

Items	Description
Time Zone	You can choose your time zone here.
Daylight Saving Time	Set the mode to Automatic or disabled.
Automatically Synchronize With an Internet Time Server	Input the NTP server so that TB200/400 will update the time automatically.
Set Date & Time Manually	You can set the time to your local time manually here.

3.3.3 Backup and Restore

We can back up the configurations before resetting TB200/400 to factory defaults, and then restore it on this package.

Figure 3-11 Backup and Restore

Notes:

1. Only configurations, custom prompts will be backed up.

2. If you have updated the firmware, it's not recommended to restore using old package.

3.3.4 Reset and Reboot

We can reset or reboot TB200/400 directly on this page.

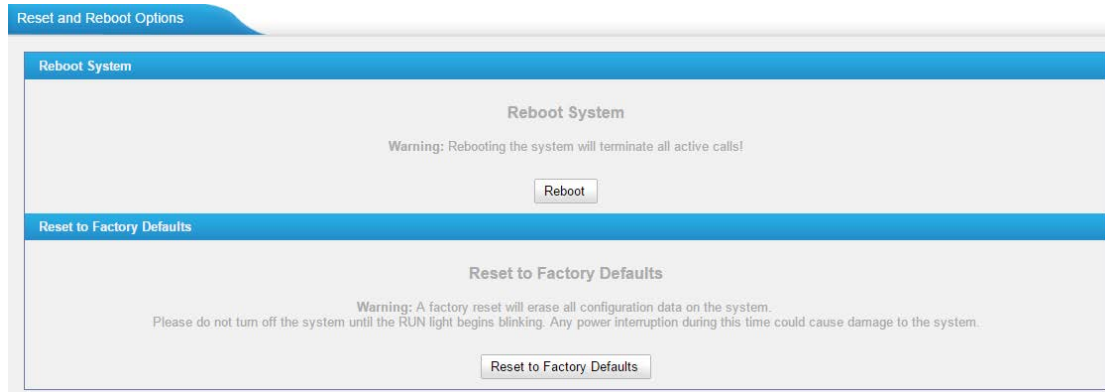


Figure 3-12 Restore and Reboot

- **Reboot System**
Warning: rebooting the system will terminate all active calls!
- **Reset to Factory Defaults**
Warning: a factory reset will erase all configuration data on the system. Please do not turn off the system until the RUN light begins blinking. Any power interruption during this time could cause damage to the system.

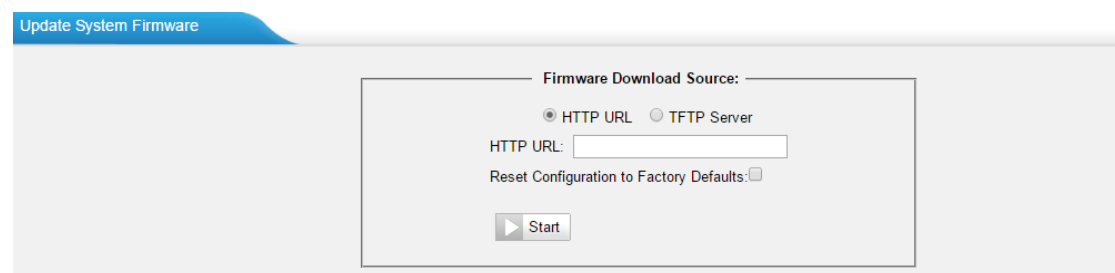
3.3.5 Firmware Update

Firmware upgrading is possible through the Administrator Web interface using a TFTP Server or an HTTP URL.

Enter your TFTP Server IP address and firmware file location, then click “Start” to update the firmware

Notes:

1. If “Reset configuration to Factory Defaults” is enabled, the system will restore to factory default settings.
2. When updating the firmware, please don't turn off the power. Or the system will get damaged.



Update System Firmware

Firmware Download Source:

HTTP URL TFTP Server

HTTP URL:

Reset Configuration to Factory Defaults:

Figure 3-13 Firmware Update

4. Gateway



Click [Gateway](#) to access the gateway configuration page. We can configure the details of BRI ports, VoIP settings, gateway settings and advanced settings.

4.1 Physical Trunk

4.1.1 Module List

All the BRI trunks are listed here. You can edit each BRI trunk by clicking “Edit” button. Before configure anything, please make sure the cable is OK, and you have got enough information from the ISDN provider.

Physical Trunk						
BRI Trunk						
Trunk Name	Port	Signaling	Max. Call Duration(min)	Call Duration(min)	Clear Stat.	
BriTrunk1	1	BRI-CPE	0	7	0	
BriTrunk2	2	BRI-NET-PTMP	0	0	0	

Figure 4-1 Module List

1) General Settings

Trunk Name	<input type="text" value="BriTrunk1"/>	Switch Type	<input type="text" value="euroisdn"/>
Signaling:	<input type="text" value="BRI-CPE"/>	Reset Interval	<input type="text" value="never"/>
Overlap Dial	<input type="text" value="no"/>	Enable Facility	<input type="text" value="Disabled"/>
PRI Indication	<input type="text" value="Inband"/>	Echo Cancellation	<input type="text" value="Off"/>
Nsf	<input type="text" value="none"/>	Codec:	<input type="text" value="alaw"/>
Hide Caller ID	<input type="text" value="No"/>		

Figure 4-2 General Settings of BRI Trunk

Table 4-1 Description of BRI General Settings

Items	Description
Trunk Name	Define a name for this trunk.
Signaling	Choose the signaling of BRI. TB400 supports the following signaling: <ul style="list-style-type: none"> ✓ BRI-NET ✓ BRI-NET-PTMP ✓ BRI CPE ✓ BRI-CPE-PTMP
Switch Type	Choose the switch type of BRI. <ul style="list-style-type: none"> ✓ national: National ISDN type2 (common in the US) ✓ ni1: National ISDN type 1

	<ul style="list-style-type: none"> ✓ dms100: Nortel DMS100 ✓ 4ess: AT&T 4ESS ✓ 5ess: Lucent 5ESS ✓ euroisdn: EuroISDN ✓ qsig: Minimalistic protocol to build a "network" with two or more PBX of different vendors
Overlap Dial	Define whether TB400 can dial this switch using overlap digits. If you need Direct Dial-in (DDI; in German "Durchwahl") you should change this to yes, then TB400 will wait after the last digit it receives.
Reset Interval	Set the time in seconds between restart of unused channels. Some PBXs don't like channel restarts. So set the interval to a very long interval e.g. 100000000 or "never" to disable entirely. If you are in Israel, the following is important. As Bezeq in Israel doesn't like the B-Channel resets happening on the lines, it is best to set the reset interval to 'never' when installing a box in Israel. Our past experience also shows that this parameter may also cause issues on local switches in the UK and China.
PRI Indication	<p>Tells how Device should indicate Busy () and Congestion() to the switch/user. Accepted values are:</p> <ul style="list-style-type: none"> ✓ inband: Device plays indication tones without answering; not available on all PRI/BRI subscription lines . ✓ Outof Band: Device disconnects with busy/congestion information code so the switch will play the indication tones to the caller. Busy() will now do same as setting PRI_CAUSE=17 and Hangup().
Enable Facility	To enable transmission of facility-based ISDN supplementary services (such as caller name from CPE over facility).
Nsf	Used with AT&T PRIs. If outbound calls are being rejected due to "Mandatory information element missing" and the missing IE is 0x20, then you need this setting.
Echo Cancellation	This disables or enables echo cancellation, it is recommended not to turn this off.
Hide Caller ID	Whether to hide Caller ID.
Codec	<p>Choose the codec:</p> <ul style="list-style-type: none"> ✓ alaw ✓ ulaw

2) Call Duration Settings

Call Duration Setting

Max. Call Duration(min): Clear Stat.

Figure 4-3 Call Duration Settings

Table 4-2 Description of Call Duration Settings

Items	Description
Max Call Duration (min)	Define the maximum call duration within a month through this BRI trunk. (0 means unlimited)
Clear Stat	Set the day in a month on which the statistics data on Max. Call Duration are deleted. This parameter is ignored if set to 0.

3) Caller ID Prefix

Figure 4-4 Caller ID Prefix

Table 4-3 Description of Caller ID Prefix

Items	Description
ISDN Dialplan	Whether the Dialplan Settings are set to make the caller ID prefix work according to information sent from the BRI provider.
International Prefix	When there are international calls coming in via this BRI trunk, the International Prefix you have set here will be added before the CID. So you can know this is an international call before you answer it.
National Prefix	When there are national calls coming in via this BRI trunk, the National Prefix you have set here will be added before the CID. So you can know this is a national call before you answer it.
Local Prefix	When there are Local calls coming in via this BRI trunk, the Local Prefix you have set here will be added before the CID. So you can know this is a local call before you answer it.
Private Prefix	When there are Private calls coming in via this BRI trunk, the Private Prefix you have set here will be added before the CID. So you can know this is a Private call before you answer it.
Unknown Prefix	When there are calls with unknown number coming via this BRI trunk, the Unknown Prefix you set here will be shown as the caller ID.

4) Dialplan

Figure 4-5 Dialplan

Table 4-4 Description of Dialplan

Items	Description
Remote Dialplan	Called number type
Remote Number Type	Called number identification
Location Dialplan	Calling number type
Location Number Type	Calling number identification

5) DOD Settings

DOD (Direct Outward Dialing) means the caller ID displayed when dialing out. Before configure this, please make sure the provider supports this feature.

You can set Global DOD for the BRI trunk.

Also, you can set different DOD numbers for different extensions.

Figure 4-6 DOD Settings

Note:

If you want to set continuous associated numbers to show continuous DOD numbers, you can choose the count of DOD number and associated number first, and then input starting number respectively.

The count of the DOD number must be only one or equal to the count of the associated number.

4.2 VoIP Settings

In this page, we can create VoIP trunk, trunk group for routing, and SIP settings.

4.2.1 VoIP Trunk

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

Name	Type	Transport	Hostname/IP	Max. Call Duration(min)	Call Duration(min)	Clear Stat.
1000	Account	udp	--	0	8	0
1001	Account	udp	--	0	0	0
PBX	VoIP Trunk	udp	192.168.6.31	0	0	0

Figure 4-7 VoIP Trunk

1) Account

It's an SIP account created in TB400 so that the other devices can register SIP trunk at their side using these information.

Figure 4-8 Account

Table 4-5 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 TB400 to register to the SIP provider, please make sure this trunk works properly in advance with provider before configuring TB400.

The screenshot shows a window titled "Add New Trunk" with a close button (X) in the top right corner. Below the title bar are two tabs: "General" (selected) and "Advanced". The main area contains the following fields:

- Trunk Type: A dropdown menu with "VoIP Trunk" selected.
- Provider Name: A text input field.
- Hostname/IP: A text input field followed by a port field containing "5060".
- Domain: A text input field.
- User Name: A text input field.
- Authorization Name: A text input field.
- Password: A text input field.

Figure 4-9 VoIP Trunk Settings

Table 4-6 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.

The screenshot shows a window titled "Add Service Provider" with a close button (X) in the top right corner. Below the title bar are two tabs: "General" (selected) and "Advanced". The main area contains the following fields:

- Trunk Type: A dropdown menu with "Service Provider" selected.
- Provider Name: A text input field.
- Hostname/IP: A text input field followed by a port field containing "5060".

Figure 4-10 Service Provider Trunk Settings

Table 4-7 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.

4.2.2 SIP Settings

This is the SIP settings in TB200/400, including General settings, NAT, Codecs, QoS, Response Code, T.38, and advanced settings.

1) General

The screenshot shows the 'SIP Settings' window with the 'General' tab selected. The settings are as follows:

- UDP Port: 5060
- Enable:
- TCP Port: 5060
- Enable:
- TLS Port: 5061
- TLS Verify Server: Yes
- TLS Verify Client: No
- TLS Ignore Common Name: Yes
- TLS Client Method: sslv2
- RTP Port Start: 10000
- RTP Port End: 12000
- DTMF Mode: rfc2833
- Max Registration/Subscription Time: 3600
- Min Registration/Subscription Time: 60
- Default Incoming/Outgoing Registration Time: 120
- Register Attempts: 0
- Register Timeout: 20
- Calling Channel Codec Priority: Yes
- DNS SRV Look Up: No
- User Agent: (empty)

Figure 4-11 SIP General Settings

Table 4-8 Description of SIP General Settings

Items	Description
UDP Port	Port used for SIP registrations. The default is 5060.
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 TB200/400 as a TLS client, whether or not to verify server's certificate. It is "No" by default.
TLS Verify Client	When using TB200/400 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 TB200/400 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. The default setting: rfc2833
MaxRegistration/ 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, TB200/400 will follow the priority order in your SIP/SPS trunks.
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 “TB200/400”; you can change it if needed.

2) NAT

SIP Settings

General NAT Codecs QOS T.38 Advanced Settings

Note: Configuration of this section is only required when you use remote extensions.

Enable STUN:

STUN Address:

STUN Port:

External IP Address:

External Host:

External Refresh Interval:

Local Network Identification:

NAT Mode:

Allow RTP Re-Invite:

Figure 4-12 NAT Settings

Table 4-9 Description of 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.
STUN Port	STUN Port
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 issued with this system. Please contact your ISP for more information.
External Refresh Interval	If an external host has been supplied, you may specify how often the system will perform a DNS query on this host. This value is specified in seconds.
Local Network Identification	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 streams 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) Codec

We can choose the allowed codec in TB200/400, 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

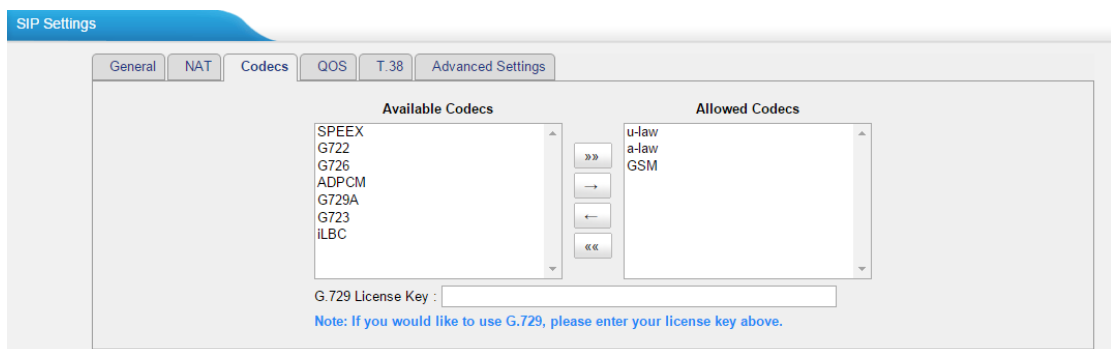


Figure 4-13 Codec

Note: 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.

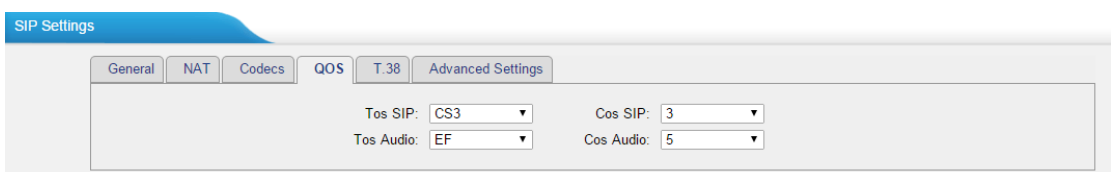
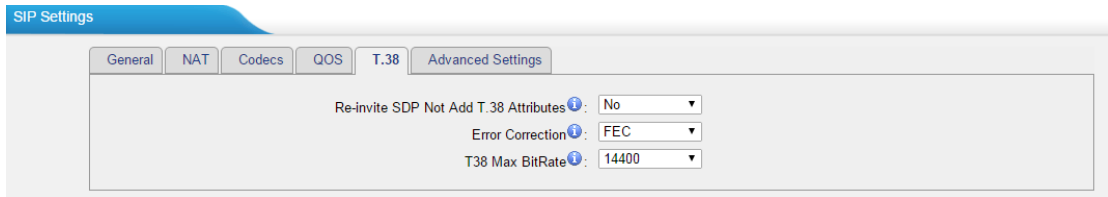


Figure 4-14 QoS

Note: it's recommended that you configure the QoS in your router or switch instead of TB200/400 side.

5) T.38

Settings on this page is for the purpose of improving receiving and sending T.38 FAX.



SIP Settings

General NAT Codecs QOS T.38 Advanced Settings

Re-invite SDP Not Add T.38 Attributes: No

Error Correction: FEC

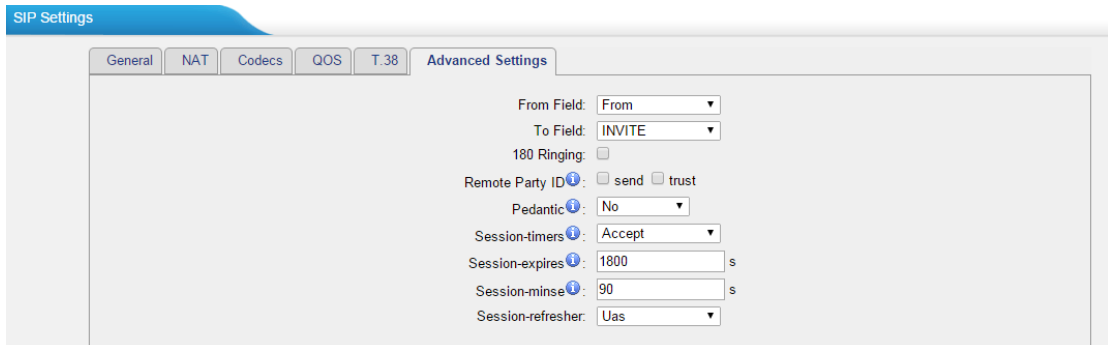
T38 Max BitRate: 14400

Figure 4-15 T.38

Table 4-10 Description of T.38 Settings

Items	Description
Re-invite SDP Not Add T.38 Attributes	If set to Yes, SDP in re-invite packet will not add T.38 attributes.
Error Correction	Re-invite SDP T38FaxUdpEc.
T.38 Max Bit Rate	Set T38 Max Bit Rate.

6) Advanced Settings



SIP Settings

General NAT Codecs QOS T.38 Advanced Settings

From Field: From

To Field: INVITE

180 Ringing:

Remote Party ID: send trust

Pedantic: No

Session-timers: Accept

Session-expires: 1800 s

Session-minse: 90 s

Session-refresher: Uas

Figure 4-16 Advanced Settings

Table 4-11 Description of Advanced Settings

Items	Description
From Field	Where to get the caller ID in SIP packet.
To 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 TB200/400 rejects "Register" or "Invite" packets, TB200/400 always respond the packets using "SIP404 NOT FOUND". It's recommended that it is enabled for security reason.

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-minise	The min refresh interval, which mustn't be shorter than 90s.
Session-refresher	Choose the session-refresher, the default is Uas.

4.2.3 Trunk Group

Trunk group is a feature that allows you to define specific SIP trunks or BRI 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.

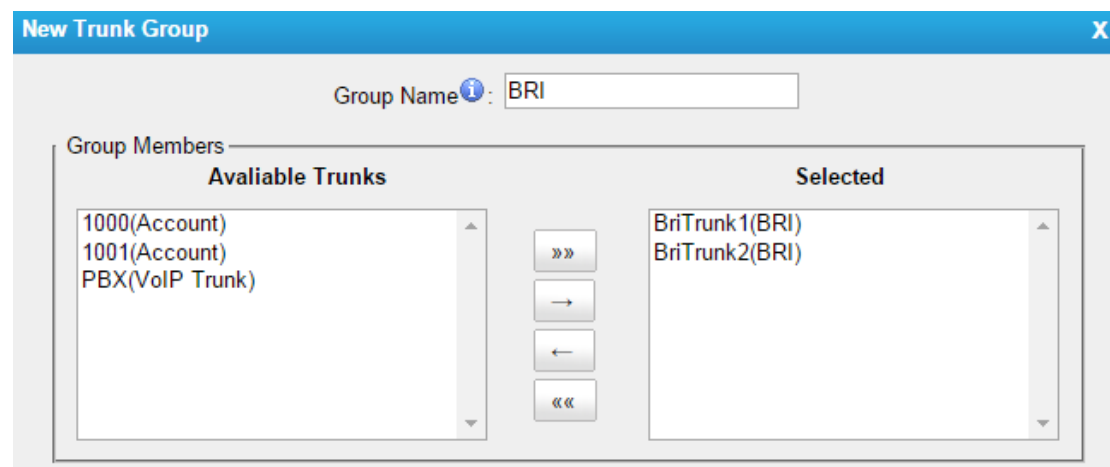


Figure 4-17 Trunk Group

Table 4-12 Description of Trunk Group

Items	Description
Group Name	Define the Group name.
Group Members	All the SIP trunks and BRI trunks are listed in the Available Trunks Box. Move the desired trunks to the Selected Box, they will be the group members.

4.2.4 General Preferences

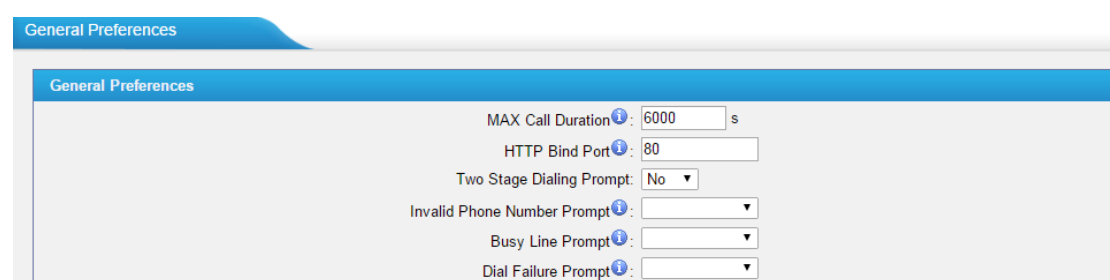


Figure 4-18 General Preferences

Table 4-13 Description of General Preferences

Items	Description
Max Call Duration	The absolute maximum amount of time permitted for a call. A setting of 0 disables the timeout. The default value is 6000s.
HTTP Bind Port	Port to use for HTTP sessions. Default: 80
Two-stage Dialing Prompt	Whether to play the prompt if Two-stage Dialing is enabled.
Invalid Phone Number Prompt	Choose the prompt for a call to an invalid phone number.
Busy Line Prompt	Choose the prompt for a busy call.
Dial Failure Prompt	Choose the prompt when calling failed.

4.3 Route Settings

4.3.1 Route List

There are two default routes on this page,

- **bri_2_sip**
Control incoming calls to TB400 BRI trunk and route calls to the SIP Server which is connected to TB400.
- **sip_2_bri**
Control calls from the SIP Server which is connected to TB400 and route calls to external numbers through BRI trunks on TB400.

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.

The screenshot shows the 'Edit Route' configuration window. At the top, there is a blue header with the title 'Edit Route' and a close button 'X'. Below the header, the configuration is organized into sections:

- Simple Mode:** A dropdown menu set to 'Yes'.
- Route name:** A text input field containing 'bri_2_sip'.
- Days of Week:** Two dropdown menus showing 'Monday' and 'Sunday'.
- Time:** Two pairs of dropdown menus showing '00' and '00' for the start time, and '23' and '59' for the end time.
- Match Incoming Calls:** A section header followed by a horizontal line and a dropdown menu for 'Call Comes in From' set to 'BRI -- BriTrunk1'.
- Handle Matched Incoming Calls:** A section header followed by a horizontal line and a dropdown menu for 'Send Call Through' set to 'Trunk -- PBX'.
- Hotline:** A text input field.

Figure 4-19 Simple Mode Route

Table 4-14 Description of Simple Mode Route

Items	Description
Route Name	Define the route name.
Days of Week	Limit the days that the route can be used.
Time	Limit the time when the route can be used.
Call Comes in From	Choose the trunk or trunk group for the incoming calls.
Send Call Through	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.

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.

Edit RouteX

Simple Mode ⓘ:

Route name ⓘ:

Days of Week: -

Time: : - :

Match Incoming Calls:

Call Comes in From

Inbound Caller Pattern ⓘ:

DID Number ⓘ:

DID Associated Number ⓘ:

Handle Matched Incoming Calls:

Send Call Through:

T.38 Support ⓘ:

Hotline ⓘ:

Two Stage Dial:

Outbound Dial Pattern ⓘ:

Strip ⓘ: digitals from front before dialing

Prepend these digitals ⓘ: before dialing

Figure 4-20 Detailed Mode Route

Table 4-15 Description of Match Incoming Calls Settings

Items	Description
Call Comes in From	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 4-16 Description of Handle Matched Incoming Calls Settings

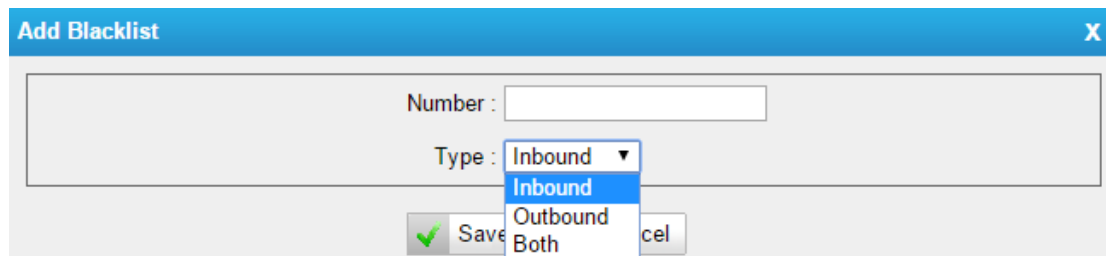
Items	Description
Send Call Through	Choose the trunk or trunk group to route the incoming calls to.
T.38 Support	Enable or disable T.38 FAX.

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.

4.3.2 Blacklist

Blacklist is used to block an incoming or outgoing call. If the number of incoming or outgoing call is listed in the number blacklist, 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.



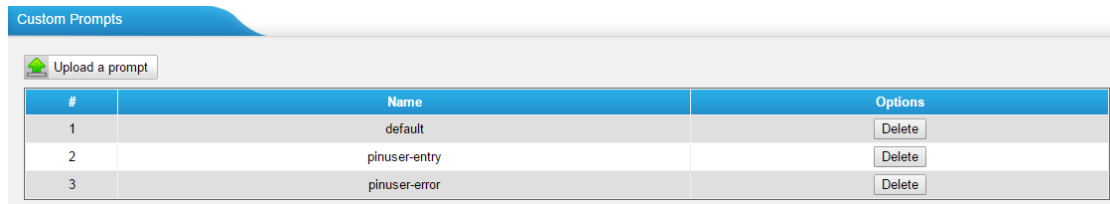
The screenshot shows a dialog box titled "Add Blacklist" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Number:". Below it is a dropdown menu labeled "Type:" with a downward arrow. The dropdown menu is open, showing three options: "Inbound" (highlighted in blue), "Outbound", and "Both". At the bottom of the dialog, there are two buttons: "Save" (with a green checkmark icon) and "Cancel".

Figure 4-21 Blacklist

4.4 Audio Settings

4.4.1 Custom Prompts

Upload custom prompts on this page. You can also download it and save it as a backup.



Custom Prompts

Upload a prompt

#	Name	Options
1	default	Delete
2	pinuser-entry	Delete
3	pinuser-error	Delete

Figure 4-22 Custom Prompts

The administrator can upload prompts following the steps:

- 1) Click "Upload Prompt".
- 2) Click "Browse" to choose the desired prompt.
- 3) Click "Upload" to upload the selected prompt.

Note:

The file 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.

5. Logout



Click  to log out TB200/400 configuration page.

[The End]