SBC300 Session Border Controller
User Manual V1.0

Shenzhen Dinstar Co., Ltd.

Address: 9th Floor, Guoxing Building, Changxing Road, Nanshan District, Shenzhen, China

Postal Code: 518052

Telephone: +86 755 61919966

Fax: +86 755 26456659

Emails: sales@dinstar.com, support@dinstar.com

Website: www.dinstar.com

Preface

Welcome

Thanks for choosing **SBC300 Session Border Controller**! We hope you will make full use of this rich-feature device. Contact us if you need any technical support: 86-755-26456110/112.

About This Manual

This manual gives introduction to the SBC300 device, and provides information about how to install, configure or use it. Please read the manual carefully before installing it.

Intended Audience

This manual is primarily aimed at the following people:

- Users
- Engineers who install, configure and maintain SBC300 device

Revision Record

Document Name	Document Version	Firmware Version
SBC300 Session Border Controller User Manual	V1.0 (2018/09/10)	1.91.1.5

Conventions

Device mentioned in this document refers to the SBC300 Session Border Controller. Those words specially noted in the document are the contents that users need to pay attention to.

Contents

1 Production Introduction

1.1 Overview

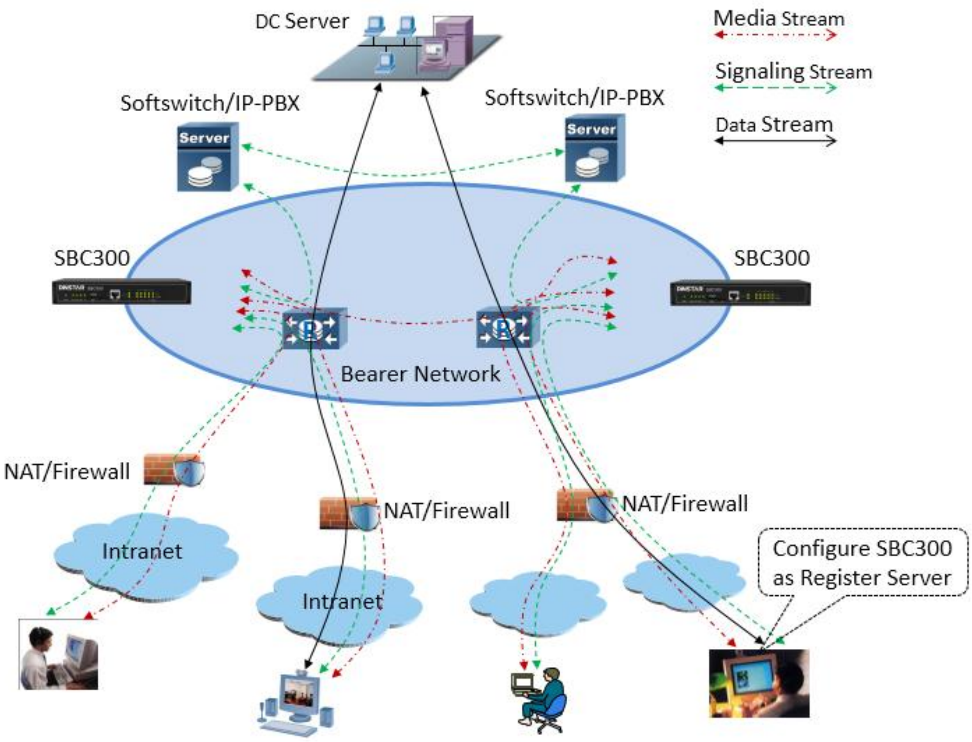
With the rapid development of unified communication and All-IP network, more and more enterprises begin to construct their own IP-based communication system by using IP-PBX and software to improve internal communication efficiency. However, they need to ensure the NAT traversal for IP multimedia services and the safe access of users. Dinstar SBC300 session border controller can help enterprises to solve the abovementioned problem.

Dinstar SBC300 provides rich SIP-based services such as safe network access, robust security, system interconnectivity, flexible session routing & policy management, QoS, media transcoding and media processing for enterprises. With distributed multi-core processor, hardware structure for non-blocking gigabit switch system as well as embedded Linux operating system, SBC300 delivers high capability while achieves low power dissipation. It is able to process up to 300 concurrent SIP sessions and transcode 100 concurrent calls. Meanwhile, it allows encrypted sessions via TLS and SRTP. Apart from traditional codecs like G.729, G.723, G.711 and G.726, SBC300 also supports the transcoding of iLBC, AMR and OPUS.

1.2 Application Scenario

The application scenario of SBC300 session border controller is shown as follows:

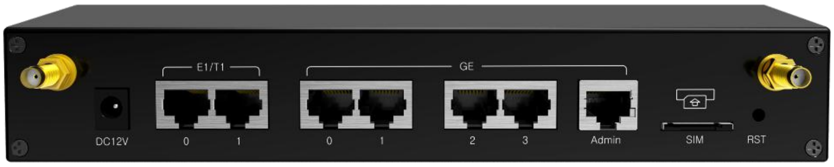
Figure 1-1 Application Scenario of SBC300



1.3 **Product Appearance**

Front View:

Back View:



1.4 **Desciption of LED Indicators**

Indicator	**Definition**	**Status**	**Description**
PWR	Power Indicator	Off	There is no power supply or power supply is abnormal
		On	The device is powered on
RUN	Running Indicator	Slow Flashing（1s）	The device is initialized successfully and is running normally
		Fast flash for two times, with interval of 1s	Image file is upgraded successfully
		Fast Flashing（200ms）	Image file fails to be upgraded
		Other Statuses	The device is in abnormal running
GE/Admin	Link indicator (Green)	Fast Flashing	The network port is connected normally
		Off	The network port is not connected, or is connected abnormally
	Speed Indicator (Yellow)	On	Network port works at 1000Mbps
		Off	Network port works 10/100Mbps
E1/T1	E1/T1 Status Indicator	Reserved	Reserved
SIM	SIM Card Indicator	Reserved	Reserved
TF	TF Card Indicator	Reserved	Reserved

1.5 Functions and Featurres

1.5.1 Key Features

- Support up to 3000 SIP registrations, with maximum RPS (registrations per second) of 20/s
- Forward 300 media calls, with maximum forwarding rate of 20/s
- Transcode 120 media calls or faxes
- Encrypted sessions through SRTP and ‘SIP over TLS’
- Support multiple softswitches, anti-blocking and topology hiding
- SIP trunks & flexible routing rules for accessing IMS
- Support regular expression and black/white list
- Embedded VoIP firewall, prevention of DoS and DDoS attacks
- Prevention of address spoofing, prevention of illegal SIP/RTP packages
- Bandwidth limitation and dynamic white list & black list
- Bandwidth limitation and dynamic white list & black list
- VLAN, QoS, static route, NAT traversal
- Double-device Hot Standby
- Hierarchical management of users, import & export of remote upgrade and configuration data
- User-friendly web interface, multiple management ways
- Support SIP protocols including UDP, TCP and TLS
- Support multiple codecs: : G.711A/U,G.723.1,G.729A/B, iLBC，AMR， OPUS
- Support multiple softswitches
- WebRTC gateway（to do）
- Video service（to do）

1.5.2 Physical Interfaces

- Ethernet Ports:
  4* 10/100/1000M Base-T Ethernet ports (GE0-GE3 for services)
  1* 10/100/1000M Base-T Admin port (for management)
- E1/T1 Ports:
  2* E1/T1, RJ48C
- 1* USB 2.0
- 1* TF Card Slot
- Serial Console

1* RS232, 115200bps, RJ45

- LTE Uplink (to do)

1.5.3 Capabilities

- Concurrent Calls

  Support 300 SIP sessions at maximum
- Transcoding

  Supports 100 transcoding calls
- CPS for call

  20 calls per second at maximum
- Registrations

  Maximum SIP registrations: 3000
- CPS for Registration

  20 registrations per second
- SIP Trunks

  128 SIP trunks at maximum

1.5.4 VoIP

- SIP 2.0 compliant, UDP, TCP, TLS,
- SIP trunk (Peer to peer)
- SIP trunk (Access)
- SIP registrations
- B2BUA (Back-to-Back User Agent)
- SIP Request rate limiting
- SIP registration rate limiting
- SIP registration scan attack detection
- SIP call scan attack detection
- SIP anti-attack
- SIP Header manipulation
- SIP malformed packet protection
- Multiple Soft-switches supported
- QoS (ToS, DSCP)

- NAT Traversal

1.5.5 Voice

- Codecs: G.711a/μ，G.723， G.729A/B，iLBC，G.726， AMR，OPUS
- RTP Transcoding
- Fax: T.38 and Pass-through
- No RTP detection
- One-way audio detection
- RTP/RTCP
- RTCP statistics reports
- DTMF: RFC2833, SIP Info, INBAND
- Silence Suppression
- Comfort Noise
- Voice Activity Detection (VAD)
- Echo Cancellation(G.168, 128ms)
- Adaptive Dynamic Buffer

1.5.6 Security

- Prevention of DoS and DDos attacks
- Control of access policies
- Policy-based anti-attacks
- Call Security with TLS/SRTP
- White List & Black List
- Access Rule List
- Embedded VoIP Firewall

1.5.7 Call Control

- Dynamic load balancing and call routing
- Flexible routing engine
- Call routing based on prefixes
- Call routing based on caller/called number
- Regular Expression
- Call routing based on time profile

- Call routing based on SIP URI
- Call routing based on SIP method
- Call routing based on endpoint
- Caller/called number manipulation

1.5.8 Maintenance

- Web-based GUI for Configurations
- Configurations Restore/Backup
- HTTP Firmware Upgrade
- CDR Report and CDR Export
- Ping and Tracert
- Network Capture
- System Logs
- Statistics and Reports
- Multiple Languages
- Centralized Management System
- Remote Web and Telnet

1.5.9 Environmental

- Power Supply: DC12V 2A
- Power Consumption: 10w
- Operating Temperature: 0 ℃ ~ 45 ℃
- Storage Temperature: -20 ℃ ~80 ℃
- Humidity: 10%-90% Non-Condensing
- Dimensions (W/D/H): 226×146×39mm
- Unit Weight: 0.85 kg
- Compliance: CE, FCC

2 Installation

2.1 Preparations before Installation

2.1.1 Attentions for Installation

Before you install the SBC300 device, please read the following safety guidelines:

- To guarantee SBC300 works normally and to lengthen the service life of the device, the humidity of the equipment room where SBC300 is installed should be maintained at 10%-90% (non-condensing), and temperature should be 0 ℃ ~ 45 ℃;
- Ensure the equipment room is well-ventilated and clean;
- It's suggested that personnel who has experience or who has received related training be responsible for installing and maintaining SBC300;
- Please wear ESD wrist strap when installing SBC300;
- Please do not hot plug cables;
- It's advised to adopt uninterruptible power supply (UPS).

2.1.2 Preparations about Installation Site

- Equipment Cabinet

  Ensure the cabinet is well-ventilated and strong enough to bear the weight of SBC300.
- Trunk

  Ensure telecom operator has approved to open a trunk.
- IP Network

  Ensure router under IP network has been prepared, since SBC300 is connected to the IP network through the standard 10/100/1000M Ethernet port.

2.1.3 Installation Tools

- Screwdriver
- ESD wrist strap
- Ethernet cables, power wires, telephone wires

- Hub, telephone set, fax, and small PBX
- Terminal (can be a PC which is equipped with hyperterminal simulation software)

2.1.4 Unpacking

Open the packing container to check whether the SBC300 device and all accessories have been in it:

- One SBC300 device
- One power adapter: 12V, 2A
- Two network cables
- One Serial console cable
- Screws

2.2 Installtion of SBC300

2.2.1 Put SBC300 into Shelf

1. Put the SBC300 device on the shelf or cabinet horizontally;

2.2.2 Connect SBC300 to Network

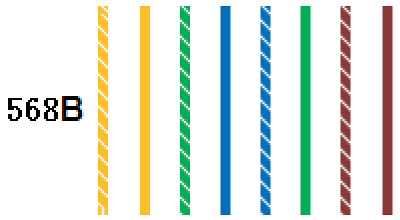
SBC300 has five network ports, namely the gigabit network port for services (from GE0 to GE3) and the gigabit network port for network management (Admin). It is advised to connect GE0, GE1, GE2 or GE3 to the IP network.

Both GE0/GE1/GE2/GE3 and Admin can be used to carry out management on SBC300, but generally GE0/GE1/GE2/GE3 are put in use. Admin is used when there is a need to separate management-related processing from service processing on SBC300.

2.2.3 How to make RJ45 Network Cable

Step1. Prepare a twisted-pair cable with a length of at least 0.6 meters, and then remove the shuck of the network cable;

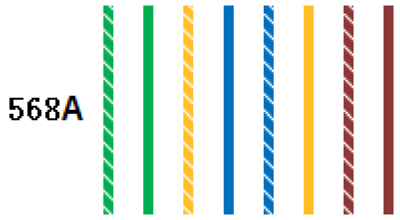
Step2. Sequence the wires of the cable according to EIA / TIA 568B Standard (as shown in the following figure);



Wire sequence of 568B: white & orange, orange, white & green, blue, white & blue, green, white & brown, brown.

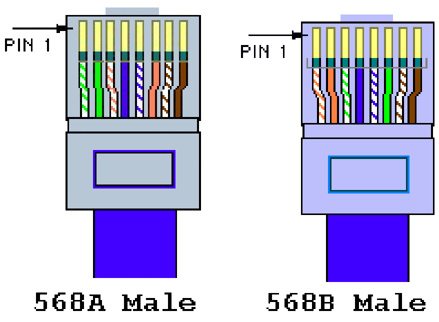
Step3. Put the wires into the PINs of a RJ45 joint according to the abovementioned wire sequence of EIA/TIA 568B, and then use a wire crimper to crimp the RJ45 joint.

Step4. On the other end of the network cable, sequence the wires of the cable according to EIA/TIA 568A Standard (as shown in the following figure);



Wire sequence of 568A: white & green, green, white & orange, blue, white & blue, orange, white & brown, brown.

Step5. Put the wires into the PINs of a RJ45 joint according to the abovementioned wire sequence of EIA/TIA 568A, and then use a wire crimper to crimp the RJ45 joint.



Step6. Test the usability of the network cable.

2.2.4 Troubleshooting about Network Connection

When the SBC300 device has been connected to gigabit Ethernet, but the SPEED and LINK indicators on the front panel of the device are still dull, it can be concluded that network connection fails.

You can try to find the reasons for network connection failure according to the following steps.

Step1: In case that the network cable is inserted into one of the service ports, please pull out the network cable and insert it into the ‘Admin’ port. If the indicator for the ‘Admin’ port is on, it can be concluded that the corresponding service port is faulty.

In case that the network cable is inserted into the ‘Admin’ port, please pull out the network cable and insert it into one of the service ports. If the indicator for the corresponding service port is on, it can be concluded that the ‘Admin’ port is faulty.

Step2: If the corresponding indicator is still dull after the network cable is inserted into other network port, please connect the network cable to a laptop or a PC, and then go to visit a website.

Step3: If the laptop or PC can visit a website normally, it can be concluded that the network cable is usable but the network port of SBC300 is faulty.

Step4: If the laptop or PC cannot visit a website, it can be concluded that the network cable is unavailable.

3 Configurations on Web Interface

3.1 How to Log in Web Interface

3.1.1 Preparations for Login

SBC300 has five network ports, namely the gigabit network ports for services (from GE0 to GE3) and the gigabit network port for management (Admin). It is advised to connect GE0/GE1/GE2/GE3 to the IP network.

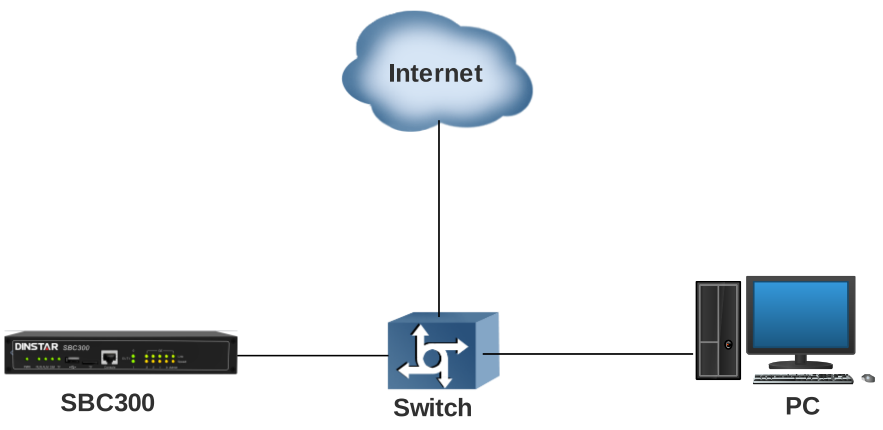
The default IP address of the 'Admin' port is 192.168.11.1, while those of GE0, GE1, GE2 and GE3 are 192.168.12.1, 192.168.13.1, 192.168.14.1 and 192.168.15.1 respectively.

First Use

At the first time that the SBC300 device is put in use, please connect the device's Admin port to a PC by using a network cable, and then modify the IP address of the PC to make it at the same network segment with of the default IP address of the Admin port. The format of PC IP address is 192.168.11.XXX, since the default IP of Admin port is 192.168.11.1

Daily Use

Connect the service port (GE0/GE1/GE2/GE3) of SBC300 to a 1000Mbps or 10/100mbps switch.



If SBC300 is connected to a 1000Mbps switch, the link indicators on the front panel turn green and flash, while the speed indicators turn yellow.

If SBC300 is connected to a 10/100Mbps switch, the link indicators on the front panel turn green and flash, while the speed indicators remain dull.

Note:

At the first time that the SBC300 device is used, only the Admin port is allowed to visit the Web interface (other network ports are disabled). If you want to connect the SBC300 device through other network ports, please connect the Admin port to a PC and log into the Web interface of the device, and then enable GE0, GE1, GE2 and GE3 ports on the **Security→Access** Control page.

3.1.2 Log in Web Interface

Open a web browser and enter the IP address of the Admin port of SBC300 (https:// 192.168.11.1). Then input username, password and verification code on the displayed login GUI. The default username is admin, while the default password is admin@123#.

Figure 3-1 Login GUI

For security consideration, it is suggested that you should modify the username and password on the **System → Users** page.

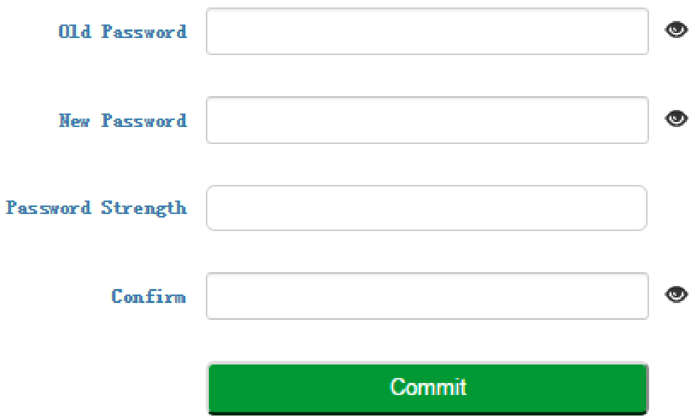


Figure 3-2 Modify Password

Note:

If you forget the IP address after modification and cannot log in the Web interface, please use a serial cable to connect the Console port of SBC300 with a PC. Enter the 'en' mode and input 'show interface' to query the IP address.

3.2 Introduction to Web Interface

The Web Interface of the SBC300 consists of the main menu bar, navigation tree and detailed configuration interfaces. Click a button of the main menu bar and select a node of the navigation tree on the left, you will see a detailed display interface or configuration interface:

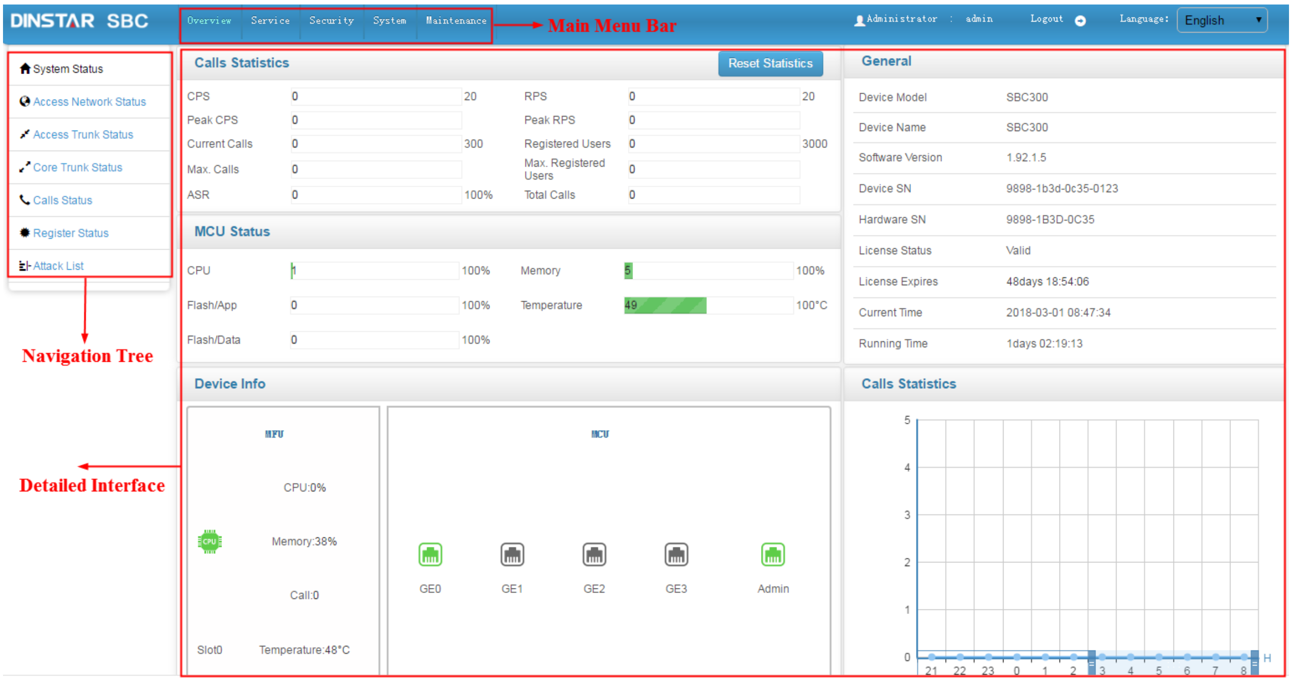


Figure 3-3 Structure of Web Interface

Table 3-1 Introduction to Web Interface

Index	Item	Description
1	Main Menu Bar	The main menu bar of SBC300, including buttons of Overview, Service, Security, System and Maintenance
2	Navigation Tree	The navigation tree of each button of the main menu bar
3	Detailed Interface	The detailed configuration interface or display interface of a node under navigation tree
4	Language	Choose Chinese or English
5	Logout	Click logout, and you will exit the Web interface

6	+ Add	To add configurations
7		To edit/modify configurations
8		To delete configurations

3.3 Configuration Flows

The following is the general configuration flows of SBC300:

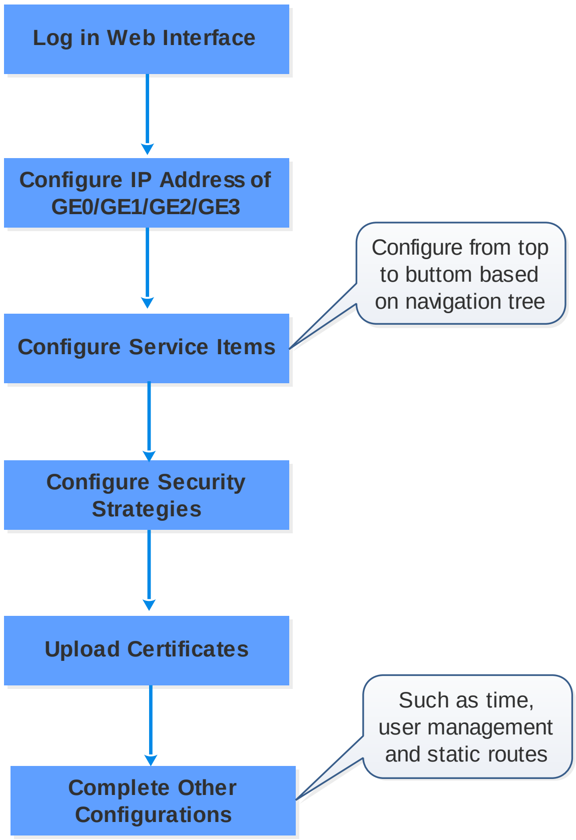


Figure 3-4 Configuration Flow

3.3.1 System Status

Log into the Web interface, and the 'System Status' page is displayed. On the page, call statistics and its graphic, device information, MCU (Main Control Unit) status as well as general information are shown.

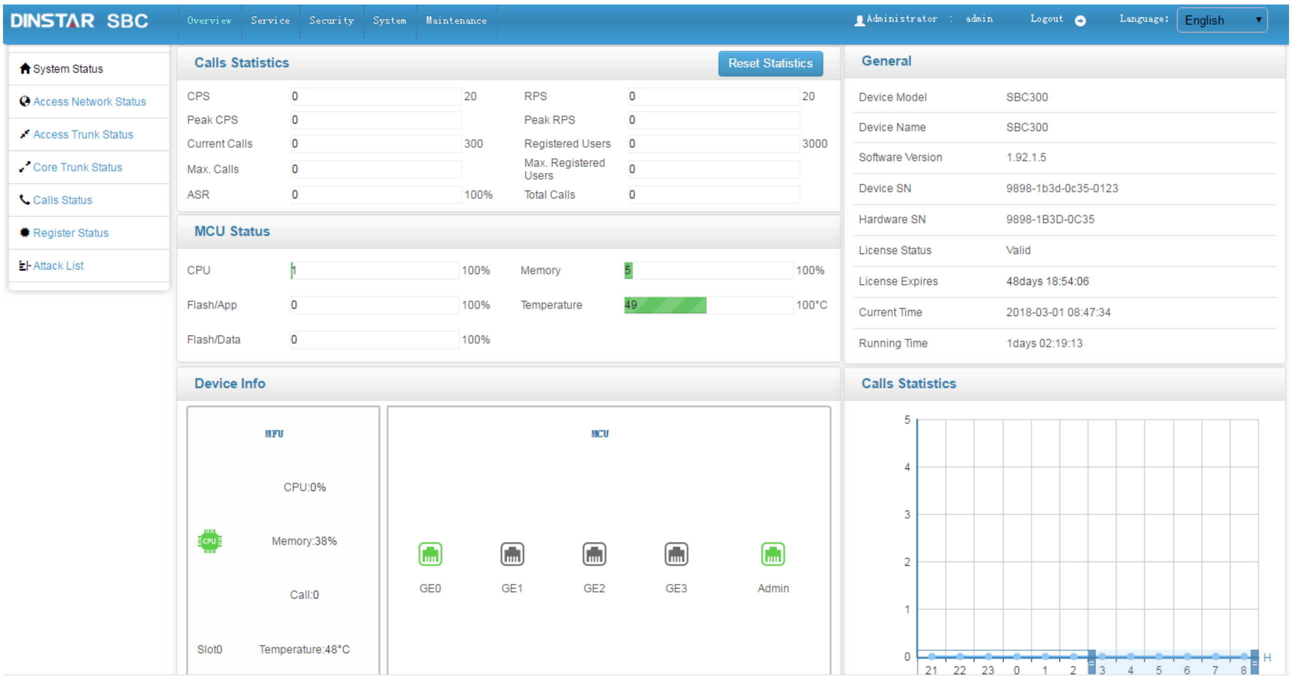


Figure 3-5 System Status

Table 3-2 Calls Statistics

CPS (Calls Per Second)	The number of new calls going through SBC300 every second at current time
Peak CPS	The peak CPS (calls per second) since SBC300 is booted up
Current Calls	The number of on-going calls at current time
Max. Calls	The maximum number of concurrent calls since SBC300 is booted up
ASR	ASR (Answer Success Rate) is a call success rate in telecommunication, which reflects the percentage of answered telephone calls with respect to the total call volume. ASR = answered call/total attempts of calls.
RPS (Registrations Per Second)	The number of new requests for registrations every second at current time
Peak RPS	The peak RPS (registrations per second) since SBC300 is booted up
Registered Users	The total number of registered users at current time
Max. Registered Users	The maximum number of registrations that are simultaneously processed since SBC300 is booted up
Total Calls	The total number of legal call requests since SBC300 is booted up

Table 3-3 MCU Status

CPU	The CPU occupancy rate at current time
Flash/App	The occupancy rate of application flash at current time
Flash/Data	The occupancy rate of data flash at current time

Memory	The occupancy rate of memory at current time
Temperature	The temperature of the CPU for MCU (Main Control Unit)

Table 3-4 Device Information

MFU (Main Function Unit)	CPU	The CPU occupancy rate of MFU at current time
	Memory	The memory occupancy rate of MFU at current time
	Call	The number of current calls that are being processed by MFU's CPU
	Temperature	The temperature of the CPU for MFU
MCU (Main Control Unit)	Network Ports （Admin/GE0/GE1/GE2/GE3）	All the network ports on the MCU, among which green ones refer to those network ports in use, while gray ones are idle.

Table 3-5 General Information

Device Model	SBC300
Device Name	The name of the device, which can be modified on the 'System →System Management' page
Software Version	The current software version No. running on SBC100
License Status	If the license is in its validity period, "Valid" will be displayed. If the license has expired, "Invalid" is shown
License Expires	The remaining time of license validity
Current Time	The current time of SBC300, which can be modified or synchronized on the 'System →Date & Time' page
Running time	The running time of the device since it is booted up

Note:

If the current time is still wrong after the system time has been synchronized or the device is restarted, it means the battery inside the device runs low and you need to replace the battery with a new one. Besides, only the Admin port can be used to synchronize time with NTP.

3.3.2 Access Network Status

Terminal users are registered to SBC300 through access network. The status of access network is always "true", which means the access network is normal and available.

On the **Overview→Access Network Status** page, detailed information about access network, including the status, name, CPS (Calls Per Second), number of registered users, ASR (Answered Success Ratio), number of calls that are being transcoded, number of current calls as well as number of total calls, are shown.

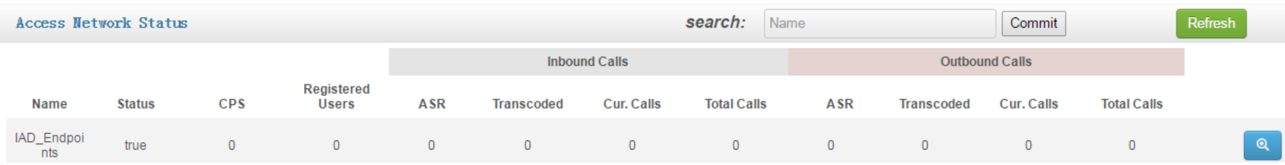


Figure 3-6 Access Network Status

Table 3-6 Access Network Status

Name	The name of the access network. It cannot be changed after the configuration is successfully applied
Status	The status of access network is always “true”, which means the access network is normal and available
CPS	The number of new calls going through the access network every second at current time
Registered	The total number of users that are successfully registered through the access network and are still in validity period
ASR	The ASR of the access network since the device is booted up; ASR = successful calls/total legal calling attempts
Transcoding	The number of calls that are being transcoded in the access network at current time
Current Calls	The number of current calls in the access network
Total Calls	The total number of legal calls since the device is booted up

Note:

Calls are grouped into inbound calls and outbound calls. Inbound calls go from terminal users to SBC300, while outbound calls are exactly the opposite.

Inbound calls and outbound calls have their own statistics of ASR, number of transcoded calls, number of current calls and number of total calls.

3.3.3 Access Trunk Status

Access SIP Trunk can realize the connection between terminal users and SBC300.

If both ‘Registration’ and ‘Keepalive’ are disabled for the SIP trunk on the **Service → Access SIP Trunk** page, the status of the SIP trunk will be ‘True’. If both ‘Registration’ and ‘Keepalive’ are enabled, the SIP trunk is successfully registered and meanwhile the option message for ‘Keepalive’ is successfully responded, the status of the SIP trunk will be ‘True’, otherwise, the status will be ‘False’.

If only ‘Registration’ is enabled and meanwhile the SIP trunk is successfully registered, the status of the SIP trunk will be ‘True’, otherwise, the status will be ‘False’. If only ‘Keepalive’ is enabled and meanwhile its option message is successfully responded, the status of the SIP trunk will be ‘True’, otherwise, the status will be ‘False’.

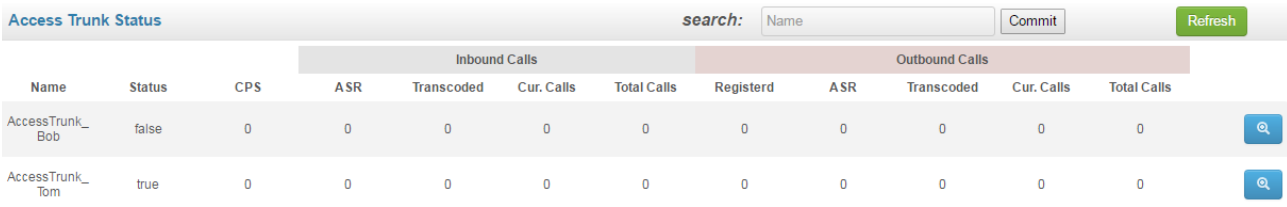


Figure 3-7 Access Trunk Status

Table 3-7 Access Trunk Status

Name	The name of the access SIP trunk. It cannot be changed after the configuration is successfully applied
Status	The status of the access SIP trunk. True: the access SIP trunk is connected normally and available; False: the access SIP trunk is disconnected and unavailable
CPS (Calls Per Second)	The number of new calls directed by the access SIP trunk every second at current time
ASR	The ASR of the access SIP trunk since the device is booted up; ASR = successful calls/total legal calling attempts
Transcoded	The number of calls that are being transcoded through the access SIP trunk at current time
Current Calls	The number of current calls routed by the access SIP trunk
Total Calls	The total number of legal calls routed by the access SIP trunk since the device is booted up
Registered	The total number of users that are successfully registered to SBC300 by the help of the access SIP trunk and are still in validity period

Note:
As for ASR, if the invite message of a call is successfully responded, we consider the call as a successful/answered call.

Calls are grouped into inbound calls and outbound calls. Inbound calls go from terminal users to SBC300, while outbound calls are exactly the opposite. Inbound calls and outbound calls have their own statistics of ASR, number of transcoded calls, number of current calls and number of total calls.

3.3.4 Core Trunk Status

Core network's SIP trunk can realize the connection between the core network and SBC300.

If both 'Registration' and 'Keepalive' are disabled for the SIP trunk, the status of the SIP trunk will be 'True'. If both 'Registration' and 'Keepalive' are enabled, the SIP trunk is successfully registered and meanwhile the option message for 'Keepalive' is successfully responded, the status of the SIP trunk will be 'True', otherwise, the status will be 'False'.

If only ‘Registration’ is enabled and meanwhile the SIP trunk is successfully registered, the status of the SIP trunk will be ‘True’, otherwise, the status will be ‘False’. If only ‘Keepalive’ is enabled and meanwhile its option message is successfully responded, the status of the SIP trunk will be ‘True’, otherwise, the status will be ‘False’.

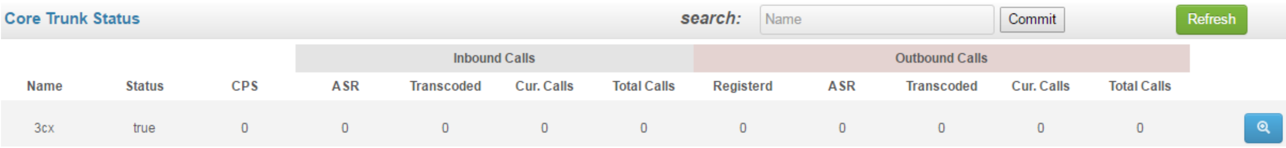


Figure 3-8 Core Trunk Status

Table 3-8 Core Trunk Status

Name	The name of the core SIP trunk. It cannot be changed after the configuration is successfully applied
Status	The status of the core SIP trunk. True: the core SIP trunk is connected normally and available; False: the core SIP trunk is disconnected and unavailable
CPS (Calls Per Second)	The number of new calls routed by the core SIP trunk every second at current time
Registered	The total number of users that are successfully registered to SBC300 by the help of the core SIP trunk and are still in validity period
ASR	The ASR of the core SIP trunk since the device is booted up; ASR = successful calls/total legal calling attempts
Transcoded	The number of calls that are being transcoded through the core SIP trunk at current time
Current Calls	The number of current calls routed by the core SIP trunk
Total Calls	The total number of legal calls routed by the core SIP trunk since the device is booted up

Note:
As for ASR, if the invite message of a call is successfully responded, we consider the call as a successful/answered call.

Calls are grouped into inbound calls and outbound calls. Inbound calls go from core network to SBC300, while outbound calls are exactly the opposite. Inbound calls and outbound calls have their own statistics of ASR, number of calls that are being transcoded, number of current calls and number of total calls.

3.3.5 Calls Status

On the **Overview→ Calls Status** page, the statuses, durations, caller number and callee number of current calls are displayed.

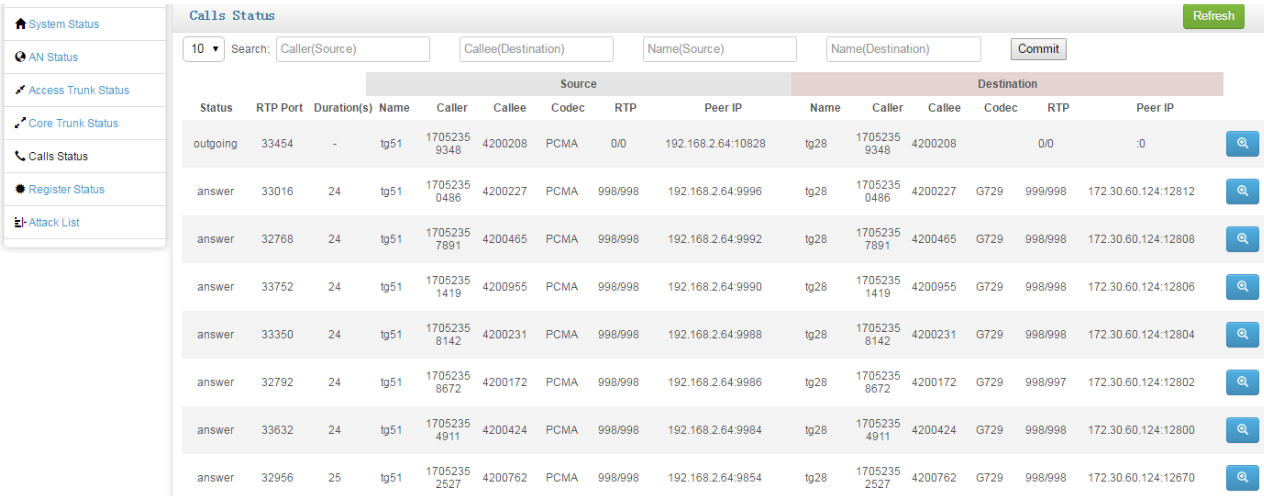


Figure 3-9 Calls Status

Table 3-9 Call Status

Status	**Init**: an invite request for calling is received and the call is initiated; **Outgoing**: the request for routing out the call is sent , and the system is waiting for response **Early**: the 18x response is received **Completed**: the 2xx response is received, and the system is waiting for the ack message **Answer**: the ack message is received, and the call is set up
RTP Port	The local RTP port of the call. If the RTP port is displayed as ‘0’, it means the RTP session has not been connected successfully
Duration(s)	The duration of the call
Name	The name of the call, which will be used when the call goes through access network’s SIP trunk, core network’s SIP trunk or access network
Caller	The caller number of the call
Callee	The callee number of the call
Codec	The codec adopted by the call. If it is a transcoded call, the source codec is different from the destination codec
RTP	The number of RTP messages that received or sent. The statistics is collected every five seconds
Peer IP	The peer IP address and peer RTP port

3.3.6 Register Status

On the **Overview→ Register Status** page, the registration statuses of terminal users on SBC300 are displayed.

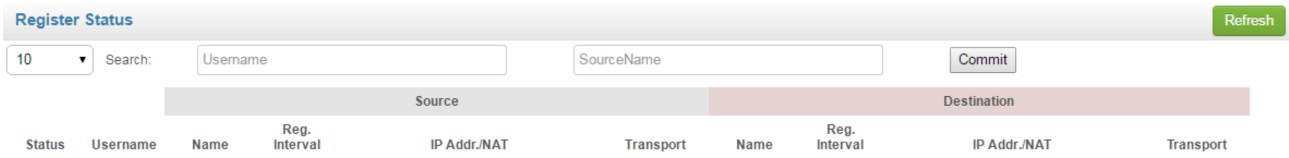


Figure 3-10 Register Status

Table 3-10 Register Status

Status	Registering: SBC300 has received the registration request send by terminal user, and is processing the request; Registered: The terminal user has been successfully registered and is in validity period
Username	The username of the terminal user, which will be used during registration
Name	Name (source): refers to the name of the access network where the registered terminal user is from; Name (destination): refers to the name of the core network's SIP trunk where the registration goes to
Reg. Interval	Register Interval (source): the interval of registering to SBC300 by terminal user Register Interval (destination): the interval of registering to core network's SIP trunk by SBC300
IP Addr./NAT	IP Addr./NAT (source): the IP address and NAT address of terminal user IP Addr./NAT (destination): the IP address and NAT address of core network's SIP trunk

3.3.7 **Attack List**

On the **Overview→ Attack List** page, the source, IP address and interface of attacks to SBC300 are shown.

Figure 3-11 Attack List

Table 3-11 Attack List

Source	The source of an attack inflicted on SBC300, for example, DDoS/DoS attacks
IP: Port	The IP address of the attack source, or the destination port that is attacked
Interface	The SBC300 device's network interface that is attacked, for example, GE1
Traffic	The traffic of the attack. When the traffic here mounts to the traffic threshold set on the **Security → Security Policy** page, the action such as 'Drop' or 'Flow Limited' will be executed.
Action	**Log Record**: when the security policy is triggered and takes effect, the attack event is recorded in a log **Flow Limited**: when the security policy is triggered and takes effect, the traffic of peer IP address or the set local port is limited, and those packets whose traffics exceed are dropped

	during the protection time. **Packet Rate Limited**: when the security policy is triggered and takes effect, the packet rate of peer IP address or the set local port is limited, and those packets with exceeding transmission rate are dropped during the protection time. **Drop**: when the security policy is triggered and takes effect, all the packets from peer IP address and those received by the set local port are dropped during the protection time.
Protection Time	The duration of the action conducted on attack source

3.4 Service

3.4.1 Media Detection

On the **Service → Media Detection** page, you can choose to enable/disable ‘Use called to match sessions’ and ‘RTP Detection’. If ‘RTP Detection’ is enabled, the SBC300 device will monitor the RTP packets of each call and will disconnect the call after it finds that no RTP packets are sent or received during the detection time.

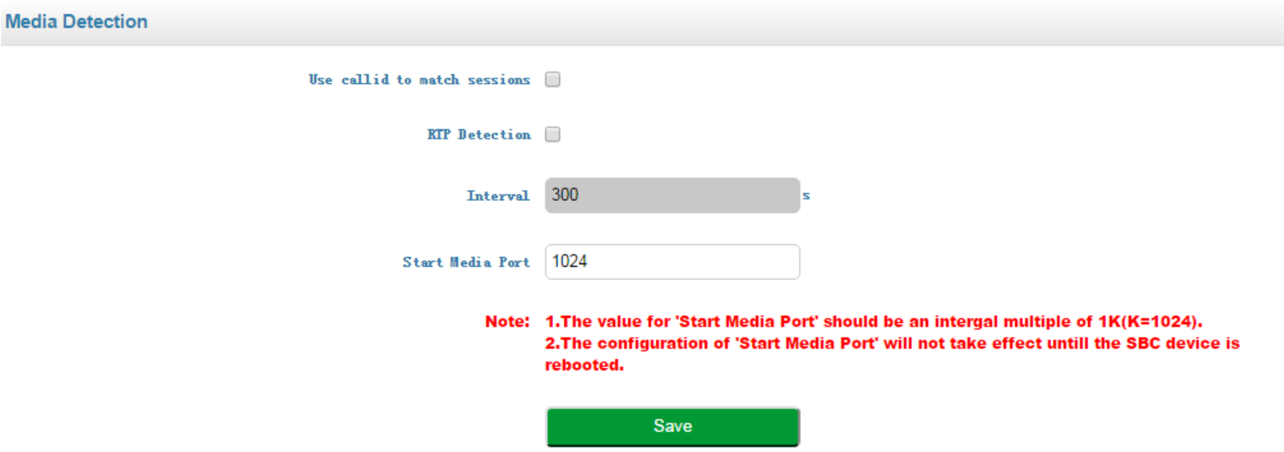


Figure 3-12 Media Detection

3.4.2 CDR

On the **Service →CDR** page, the CDR server defaults to ‘Disabled’, and you need to enable it to do corresponding configurations.

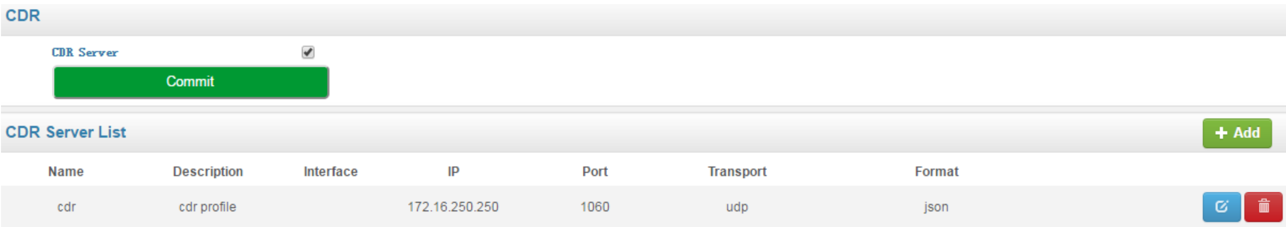


Figure 3-13 Configure CDR Server

Table 3-12 CDR

Name	The name of the CDR server. It cannot be modified after the CDR server has been successfully added
Description	The description of the CDR server
Interface	The interface through which the CDR server receives CDRs
IP	The IP address of the CDR server
Port	The SIP port through which the CDR server receives CDRs
Transport	The transport protocol adopted to transport CDRs, which can be UDP or TCP
Format	The coded format of CDRs, which only supports json currently

3.4.3 Number Profile

On the **Service →Number Profile** page, you can set a prefix for calling numbers or called numbers. When the prefix of a calling number or a called number matches the set prefix, the call will be passed to choose a route. Number profile does not support ‘Regular Expression’ currently.

Click 

, and you can add a number profile.

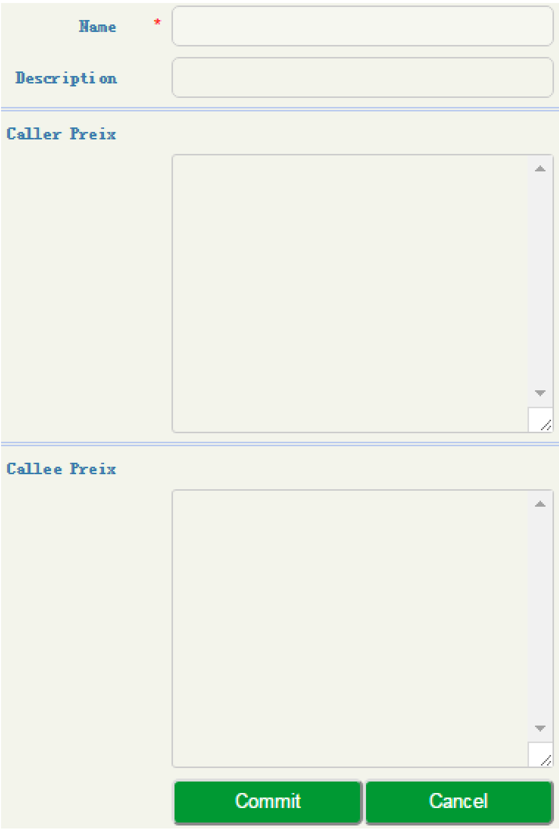


Figure 3-14 Add Number Profile

Table 3-13 Number Profile

Name	The name of the number profile. It cannot be modified after the number profile is added successfully
Description	The description of the number profile
Caller Prefix	The prefix set for caller numbers. It does not support regular expression. When the prefix of a caller number matches the set prefix, the call will be passed to choose a specific route.
Callee Prefix	The prefix set for callee numbers. It does not support regular expression. When the prefix of a callee number matches the set prefix, the call will be passed to choose a specific route.

3.4.4 Time Profile

On the **Service → Time Profile** page, you can set a time period for calls to choose routes. If the local time when a call is initiated falls into the set time period, the call will be passed to choose a corresponding route. If a call is initiated at other time, the call cannot be routed.

Click 

, and you can add a time profile.

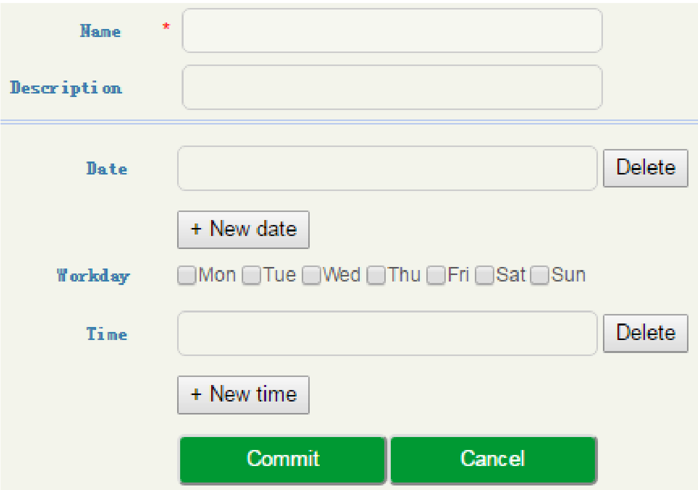


Figure 3-15 Add Time Profile

Table 3-14 Time Profile

Name	The name of the time profile. It cannot be modified after the time profile is added successfully
Description	The description of the time profile
Date	Configure the starting date and ending date of a period; You are allowed to configure multiple periods
Workday	Choose one or more working days (from Monday to Sunday)

Time	Choose the starting time and ending time of a day You are allowed to configure multiple time periods

3.4.5 Rate Limit

On the **Service → Rate Limit** page, you can configure the maximum registrations per second (RPS), maximum calls per second (CPS) and maximum concurrent calls for access network, access SIP trunk and core SIP trunk.

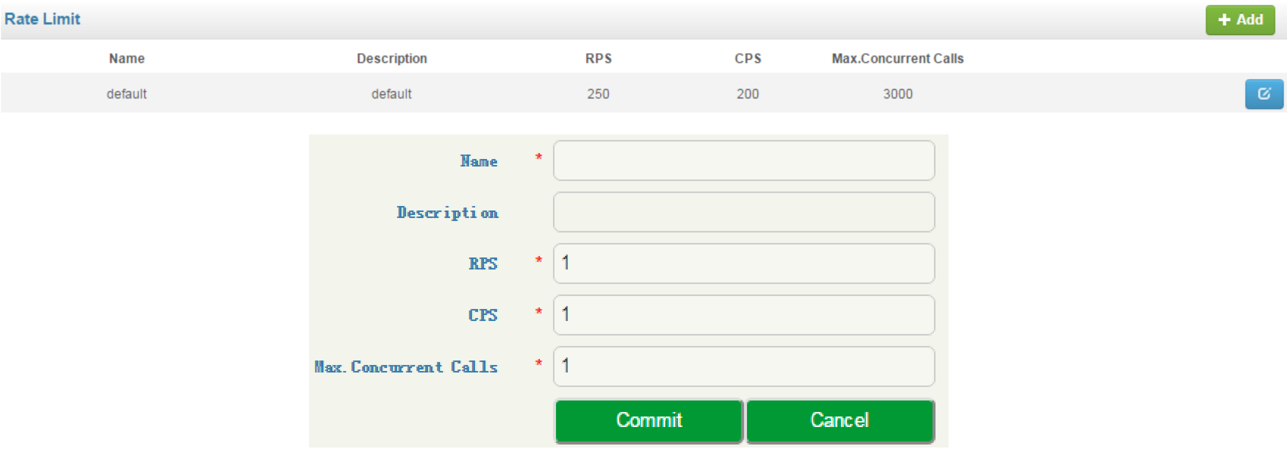


Figure 3-16 Add Time Limit

Table 3-15 Rate Limit

Name	The name of the rate limit rule. It cannot be modified after the rate limit rule is added successfully
Description	The description of the rate limit rule
RPS	The maximum number of registrations that is allowed per second
CPS	The maximum number of calls that is allowed per second
Max. Concurrent Calls	The maximum number of concurrent calls that is allowed

Note:

1. There is a default rate limit rule on the page. Its RPS, CPS and maximum number of concurrent calls are defined by License.

2. The RPS, CPS and maximum concurrent calls configured in other rate limit rules cannot be greater than those of default rule.

3.4.6 Black & White List

On the **Service → Black & White List** page, you can choose to put calling numbers on black list or white list. If a number is put on black list and the black list is linked to an access network, an access SIP trunk or a core SIP trunk, the SBC300 device will refuse the calls and registration requests from this number.

If a number is put on whitelist and the white list is adopted, the SBC300 device will accept the calls and registration requests from this number.

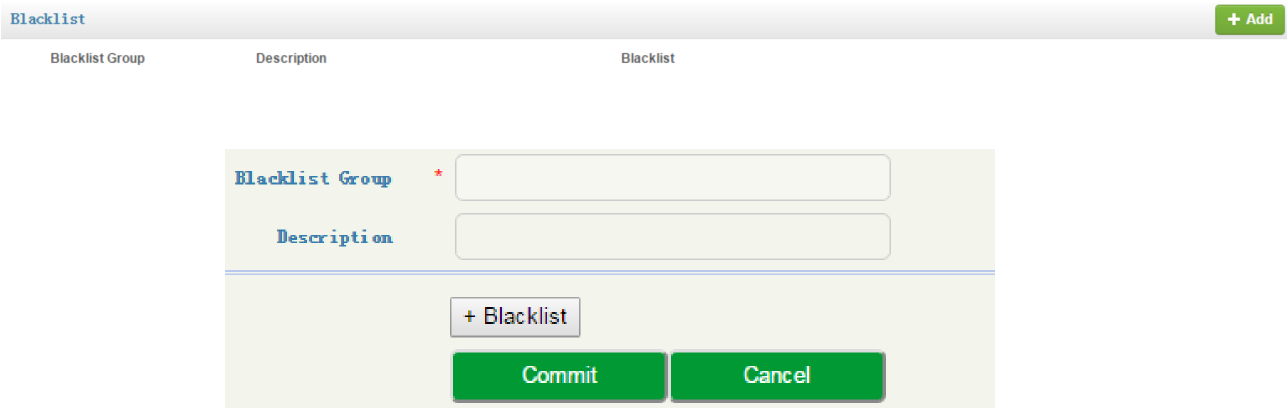


Figure 3-17 Blacklist

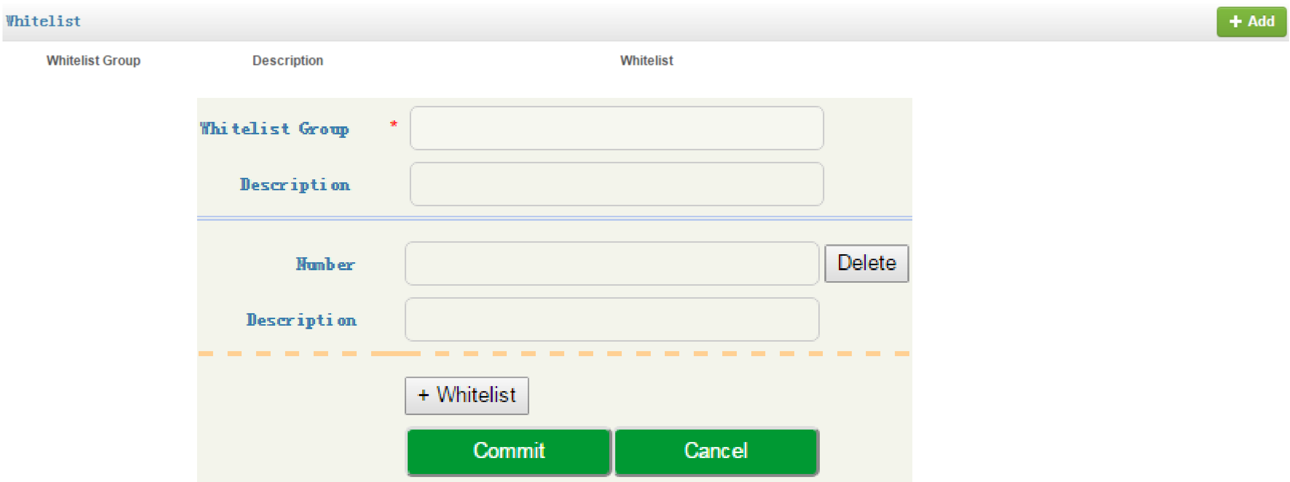


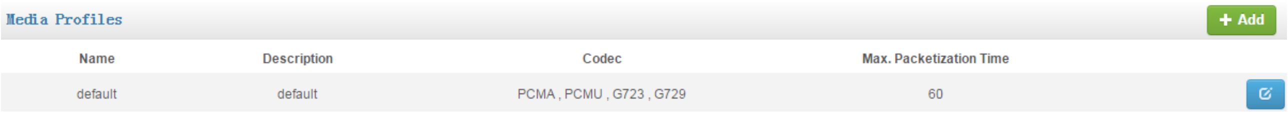
Figure 3-18 Whitelist

Table 3-16 Blacklist & Whitelist

Blacklist Group	The name of the blacklist. It cannot be modified after the blacklist group is added successfully
Whitelist Group	The name of the whitelist. It cannot be modified after the whitelist group is added successfully
Description	The description of the blacklist/ whitelist group
Number	The calling number(s) that is (are) put on blacklist/ whitelist. It does not support regular expression.
Description	The description of a specific blacklist/ whitelist

3.4.7 Codec Profile

SBC300 supports such codecs as G729, G723, PCMU, PCMA, ILBC_13K, ILBC_15K, OPUS and AMR. You can group these codecs and adjust their priority according to your needs.



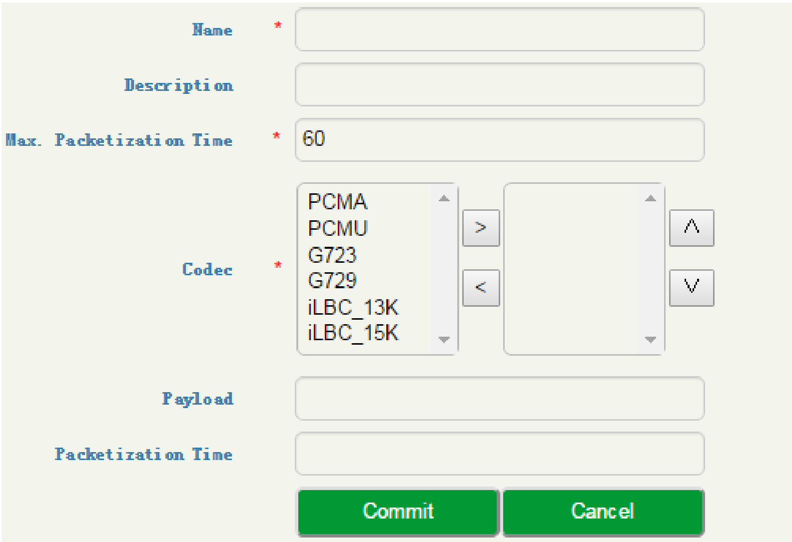


Figure 3-19 Edit Codec Profile

Table 3-17 Codec Group

Name	The name of the codec group. It cannot be modified after the codec group has been added successfully
Description	The description of the codec group
Max. Packetizing Time	The maximum packetizing time that the codec group supports
Codec	SBC300 supports codecs including PCMA, PCMU, G.729A/B, G.723, iLBC,_13K, iLBC_15K, AMR and OPUS
Payload	The codec value of each codec, which cannot be modified
Packetizing Time	The default packetizing time of each codec, which cannot be modified

Note:

There is a default codec group on the page. This codec group includes all the codecs by default. It can be modified but cannot be deleted.

3.4.8 Number Manipulation

Number manipulation refers to the change of a called number or a caller number during calling process when the called number or the caller number matches the preset rules.

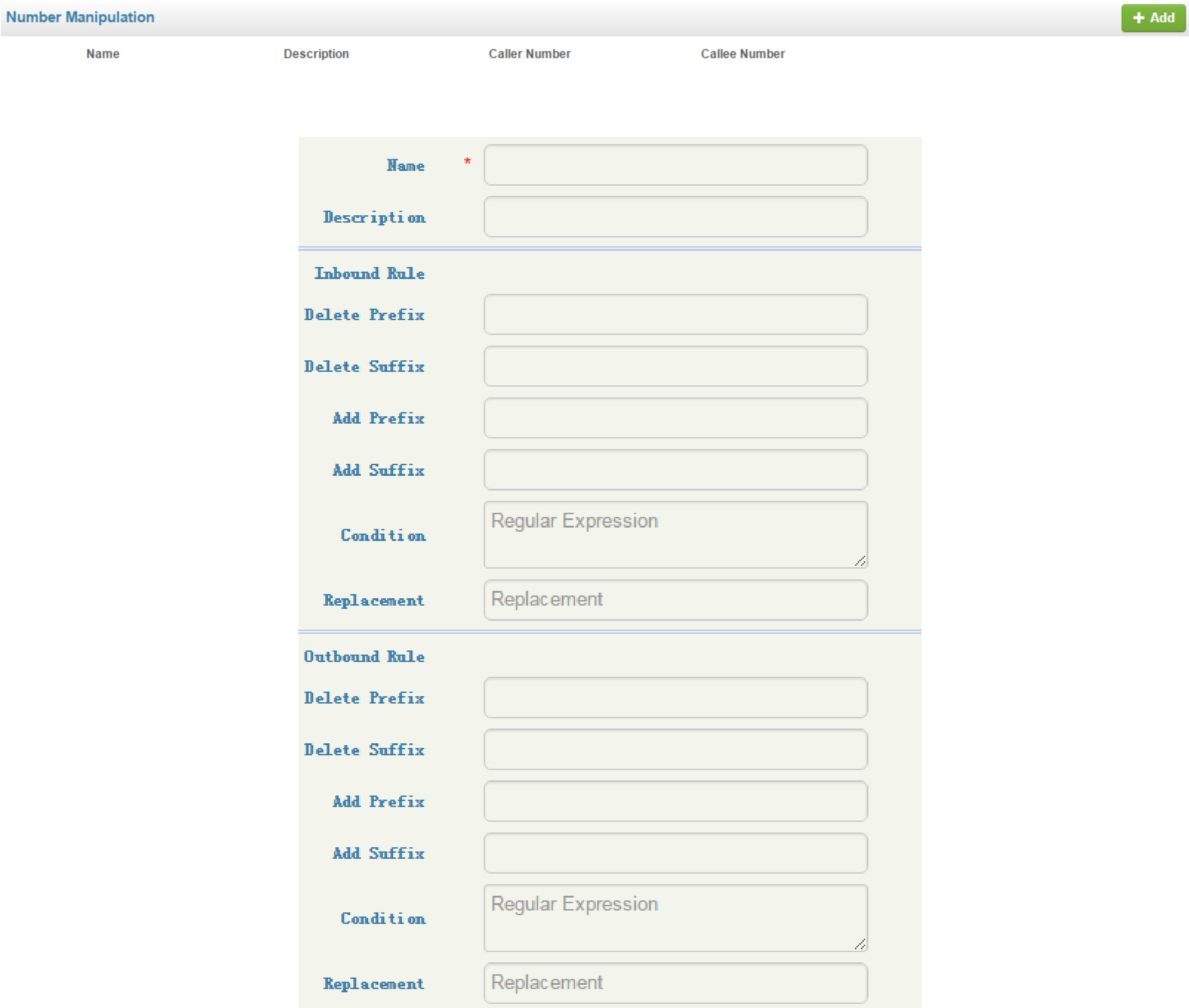


Figure 3-20 Configure Number Manipulation Rule

Table 3-18 Number Manipulation Rule

Name	The name of this manipulation rule. It cannot be modified after the manipulation rule has been added successfully
Description	The description of this manipulation rule
Delete Prefix	The prefix that will be deleted after it matches a caller/callee number. For example, if the prefix is set as 678 and the caller number is 67890000, then the caller number will be changed into 9000; The prefix supports regular expression;

	Multiple prefixes can be set for one manipulation rule.
Delete Suffix	The suffix that will be deleted after it matches a caller/callee number. For example, if the suffix is set as 123 and the caller number is 8000123, then the caller number will be changed into 8000; The suffix supports regular expression; Multiple suffixes can be set for one manipulation rule.
Add Prefix	The prefix added to the caller/callee number. For example, if the prefix is set as 678 and the caller number is 9000, then the caller number will be changed into 6789000 after the manipulation rule is matched; The prefix does not support regular expression;
Add Suffix	The suffix added to the caller/callee number For example, if the suffix is set as 678 and the caller number is 9000, then the caller number will be changed into 9000678 after the manipulation rule is matched; The suffix does not support regular expression;
Condition	The condition supports regular expression. If a caller/callee number can match one of the rules set in the 'Condition' parameter, the original number will be changed into the one set in the 'Replaced By' parameter.
Replaced By	If a caller/callee number can match one of the rules set in the 'Condition' parameter, the original number will be changed into the one set in the 'Replaced By' parameter. The value of the 'Replaced By' parameter does not support regular expression.

Note:

During number manipulation, 'Delete Prefix' and 'Delete Suffix' are carried out first, followed by 'Add Prefix' and 'Add Suffix'. If 'Condition' is also set, SBC300 will match the condition based on the result of the abovementioned rules.

If a number manipulation rule is used on the **Service →Access Network** page, the **Service → Access SIP Trunk** page or the **Service → Core SIP Trunk** page, it means the caller/callee number will be manipulated before the call chooses a route;

If a number manipulation rule is used on the **Service →Routing Profiles** page, it means the caller/callee number will be manipulated after the call has chosen a specific route.

3.4.9 Number Pool

On the **Service → Number Pool** page, you can set a number pool. If the number pool is used on the **Service → Routing Profiles** page, the caller/callee number will be randomly replaced by a number from the pool.

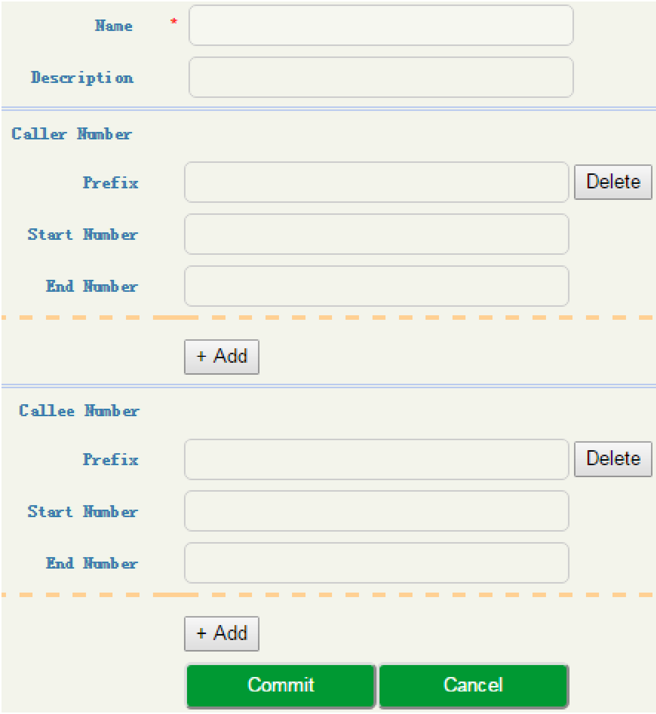


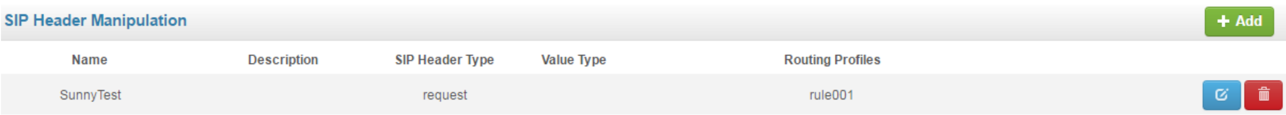
Figure 3-21 Configure Number Pool

Table 3-19 Number Pool

Name	The name of this number pool. It cannot be modified after the number pool has been added successfully
Description	The description of this manipulation rule
Caller/Callee Number	**Prefix**: If the prefix here is matched with a caller/callee number, the caller/callee number will be randomly replaced by a number from the pool; **Start Number**: The starting number of the number pool **End Number**: The ending number of the number pool

3.4.10 SIP Header Manipulation

When the SIP headers of the messages related to calls passing through access network, access SIP trunk and core SIP trunk are not consistent with those required, you need to set rules to manipulate original SIP headers.



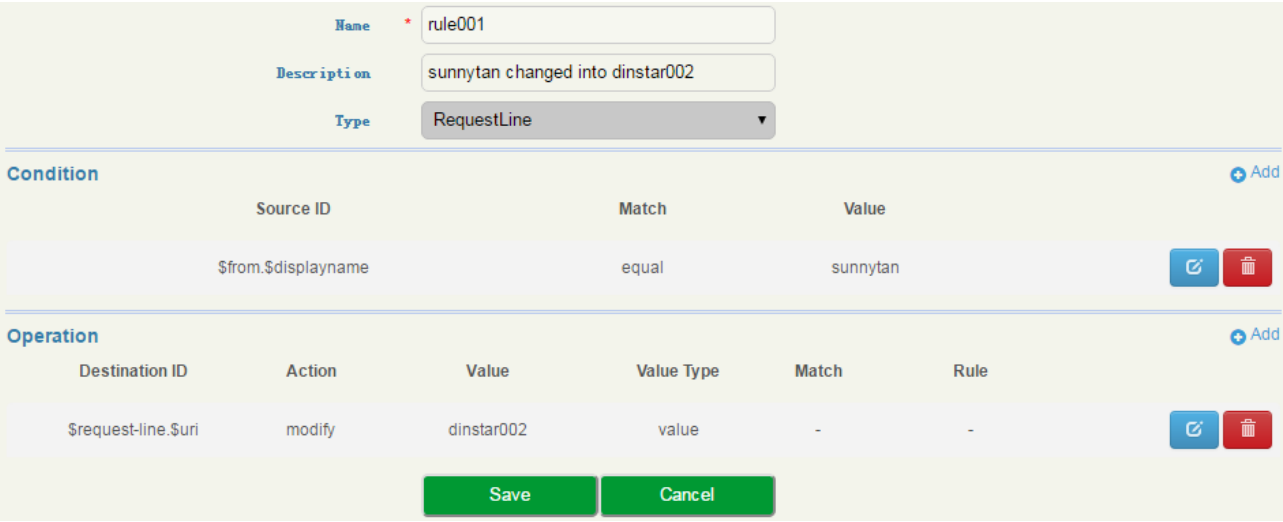


Figure 3-22 Configure SIP Header Manipulation Rule

Table 3-20 SIP Header Manipulation

Name	The name of the SIP header manipulation rule. It cannot be modified after the SIP header manipulation rule has been added successfully
Description	The description of the SIP header manipulation rule
Type	Request: The manipulation rule is only applied to SIP request messages; Response: The manipulation rule is only applied to SIP response messages; List: The manipulation rule is only applied to those SIP request and response messages that are selected
Operation	The operation rule will be applied when the set condition is met. For example, when the set value meets the source ID in Request Line, the actions(add, modify or remove) will be conducted on the destination ID. **Name**: the name of the operation rule. **Description**: the description of the operation rule. **Type**: the content type where the operation rule will be applied. Request-line: the content of the request line of SIP message. Status-line: the content of the status line of SIP message. Header: the content of the header of SIP message. **Condition**: the set condition for the operation rule. When the set value matches the source ID, the operation rule will be activated. **Source ID**: the original content of SIP message, it can be any parameter included in SIP message. **Match**: equal → when the source ID is equal to the set value, the operation rule is activate.

	Regex→ when the source ID matches the set regular expression, the operation rule will be activated. **Value**: the value set to match the source ID. **Destination ID**: the designated header to be modified. **Action**: The actions (add, modify or remove) to manipulate SIP header after the preset conditions is matched. Value Type: Token→ In the 'Value' field, the content with $ is the content which is from the designated header of original SIP message.

3.4.11 **SIP Header Passthrough**

On the **Service → SIP Header Passthrough** page, you can configure one or more 'SIP Header Passthrough' profiles. If the profiles are used on the **Service →Routing Profile** page, the designated extension fields of SIP messages of a specific route will be passed through.

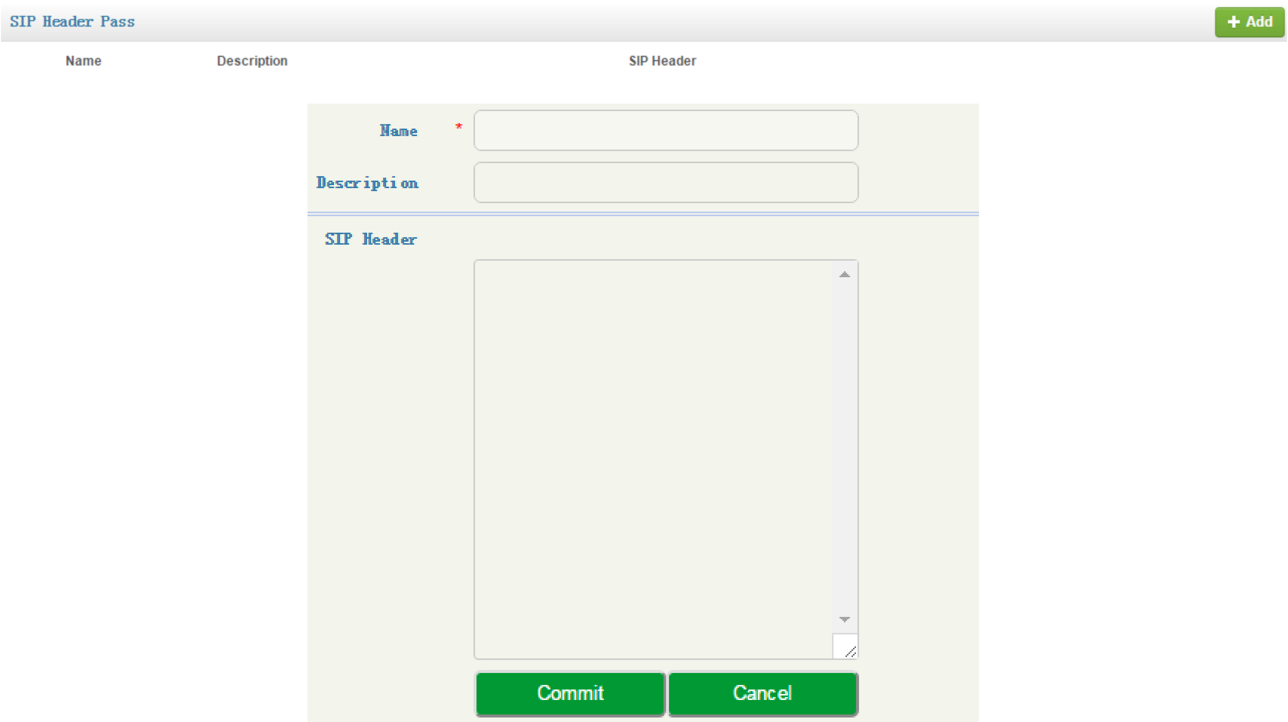


Figure 3-23 SIP Header Passthrough

Table 3-21 SIP Header Pass

Name	The name of the 'SIP header passthrough' profile. It cannot be modified after the 'SIP header pass' profile has been added successfully
Description	The description of the 'SIP header passthrough' profile

SIP	The SIP headers that are passed through. A SIP header in a row, case-sensitive, without any extra punctuation marks

Note:

1.The ‘Allow’ and ‘Supported’ SIP headers can only be passed through during registration. That is to say, they cannot be passed through during calling. Please think carefully before passing through these two SIP headers, as they might conflict with the configurations of SBC300.

2.The following SIP heads are not allowed to be passed through:

Network, To, From, Contact, Cseq, Max-Forwards, Content-Length, Content-Type, Via, Require, Proxy-Require, Unsupported, Authorization, Proxy-Authorization, Www-Authenticate, Proxy-Authenticate, Accept, Route, Record-Route, Refer-To, Referred-By, Auto-Defined。

3.4.12 **Access Network**

On the **Service →Access Network** page, you can configure the parameters of access network, which will be used when terminal users are registered to softswitch through the SBC300 device.

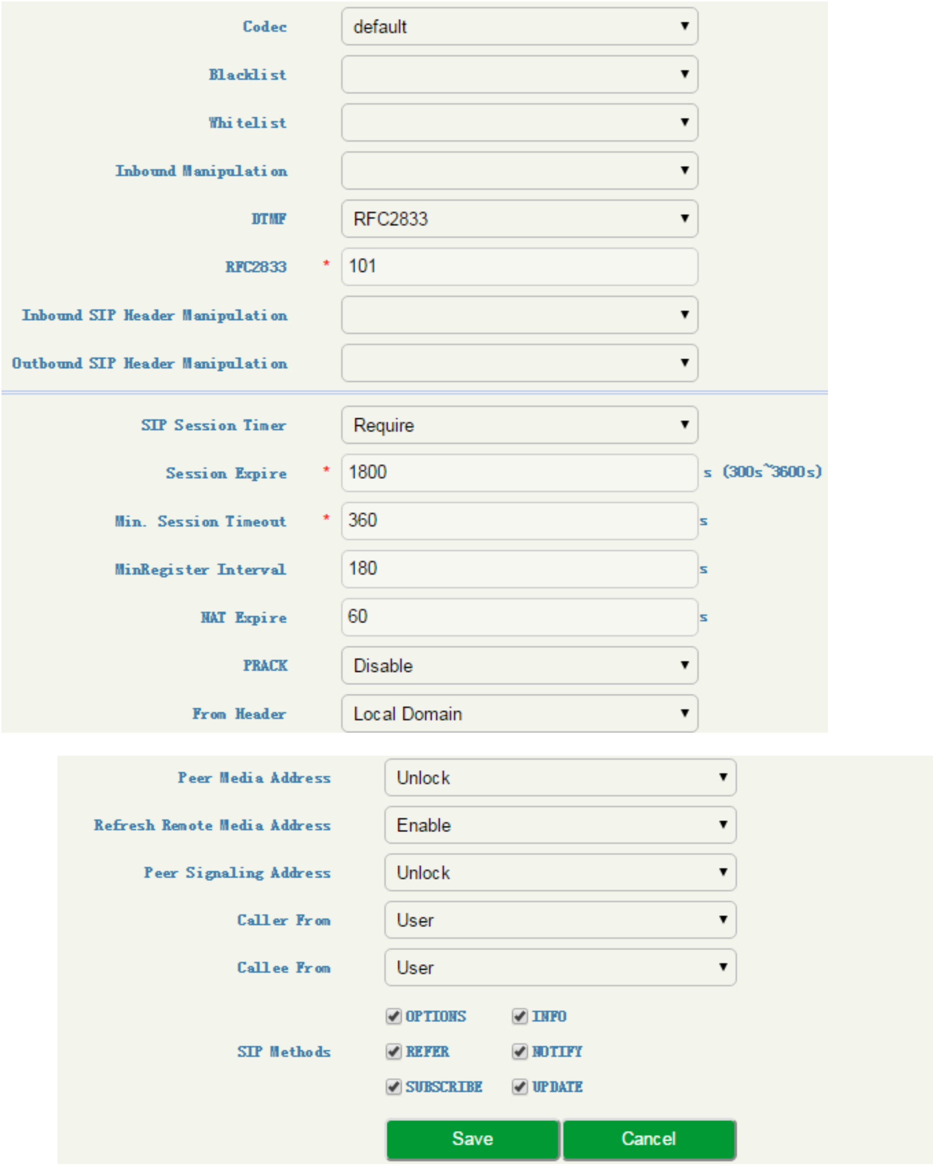


Figure 3-24 Configure Parameters of Access Network

Table 3-22 Access Network

Name	The name of the access network. It cannot be modified after the access network has been added successfully
Description	The description of the access network
Interface	The interface of the access network. It can be eth0, eth1, eth2 or eth3
Transport Protocol	Select a transport protocol for the access network. It can be UDP, TCP or TLS
SIP Port	The access network's SIP listening port on the Ethernet interface of SBC300
IPv4/IPv6	Select a network protocol for the access network. It can be IPv4 or IPv6. By default, the network protocol is IPv4

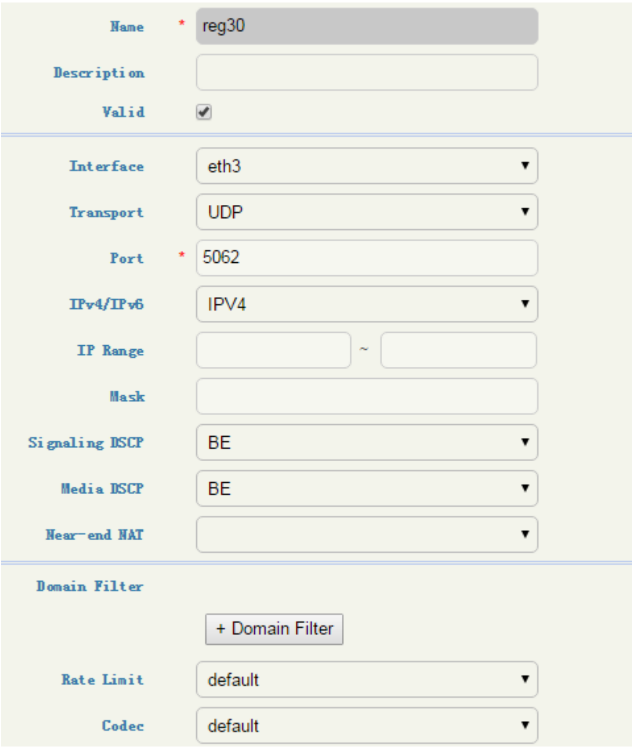
IP Range	Configure the range of legal IP addresses that send out SIP request can be received by the
Mask	The subnet mask of the IP range
Signaling DSCP	The QoS tag of SIP signaling messages
Media DSCP	The QoS tag of meida messages
Near-end NAT	Near-end NAT defaults to disabled. If it is enabled, the contact IP address contained in SIP messages sent out by SBC300 will be turned into the outbound IP address of public network. If NAT is enabled, you need to fill in the outbound IP address of public network.
Domain Filter	
Rate Limit	The maximum RPS(registrations per second), CPS(calls per second) and total call volume. Please refer to3.4.5
Codec	The codecs that the access network supports. Please refer to 3.4.7
Blacklist	Select a blacklist for the access network. Calls given by the caller numbers on the blacklist will be refused to go through the access network. Please refer to 3.4.6
Whitelist	Select a whitelist for the access network. Calls initiated by the caller numbers on the whitelist will be allowed to go through the access network. Please refer to 3.4.6 If no black list and white list are selected for the access network, all calls are allowed to go through the access network
Inbound Manipulation	Select a number manipulation rule or a number pool for the access network. When a call coming into the access network matches the manipulation rule, its number will be manipulated. Please refer to 3.4.8 and 3.4.9
DTMF	DTMF is short for Dual Tone Multi Frequency; There are three DTMF modes, including SIP Info, INBAND, RFC2833; If the DTMF mode of an access network differs from that of core network, SBC300 will convert it through DSP
Inbound SIP Header Manipulation	Select a SIP header manipulation rule for inbound calls of the access network. If a call matches the manipulation rule, the SIP header of the messages related to the call will be manipulated when it comes into the access network. Please refer to 3.4.10
Outbound SIP Header Manipulation	Select a SIP header manipulation rule for outbound calls of the access network. If a call matches the manipulation rule, the SIP header of the messages related to the call will be manipulated when it goes out the access network. Please refer to 3.4.10
SIP Session Timer	Session timer is a mechanism to keep activating sessions. If 'Supported' is selected, SBC300 will send 'reinvite' messages to keep activating

	sessions within the configured duration. If no messages are detected within the configured duration, sessions will be considered as ‘ended’, and then will be disconnected. If ‘Require’ is selected, the callee side of a call passing through the access network also needs to support session timer.
Session Expire	Configure the duration of the session. During the duration, SBC300 will send ‘reinvite’ messages to keep activating the session.
Min. Session Timeout	Minimum session duration is used to negotiate with the session timer on the callee side
MinRegister Interval	The minimum time allowed for terminal’s registration. That is to say, if the ‘expires’ value in the REGISTER message is smaller than this minimum time, SBC300 will refuse the register request.
NAT Expire	If a terminal is in private network and sends out messages through NAT, the registration time responded by SBC300 will automatically turned into the time configured here. The value of ‘NAT Expire’
PRACK	PRACK (Provisional Response ACKnowledgement): provide reliable provisional response messages. Disable: INVITE request and 1xx response sent out by SBC300 will not include *100rel* tag by default; Support: INVITE request and 1xx response sent out by SBC300 will include *100rel* tag in Supported header; Require: INVITE request and 1xx response sent out by SBC300 will include *100rel* tag in Require header; if the peer does not support 100rel, it will automatically reject INVITE request with 420; if the peer supports 100rel. it will send *PRACK* request to acknowledge the response.
From Header	It can be ‘Local Domain’ or ‘Peer Domain’. ‘Local Domain’ is the default value.
Peer Media Address	Lock: when the peer device works at public network, media address carried in SDP (Session Description Protocol) message is locked; when the peer device works at private network, the address that sends 30 messages continuously are locked. Unlock: remote address sending media messages is not locked.
Refresh Remote Media Address	If this parameter is enabled, the remote address receiving media messages will be refreshed.
Peer Signaling Address	Lock: when a calling account is successfully registered, the access network only receives those calls from the registered address of the caller.
Caller From	User: the USER field of FROM header of INVITE message is extracted as caller number Display: the DISPLAY field of FROM header of INVITE message is extracted as caller

	number
Callee From	User: the USER field of TO header of INVITE message is extracted as callee number； Display: the DISPLAY field of TO header of INVITE message is extracted as callee number； Request-uri: the USER NUMBER in REQUEST-URI of INVITE message is extracted as callee number；
SIP Methods	Configure the SIP request methods that can be accepted by the access network. If a SIP request method is not enabled, the system will reject the corresponding SIP request. By default, the INVITE request, REGISTER request and SESSION DISCONNECT request are accepted.

3.4.13 Access SIP Trunk

Access SIP trunk can realize the connection between access network and SBC300. On the **Service →Access SIP Trunk** page, you can configure the parameters of access SIP trunk.



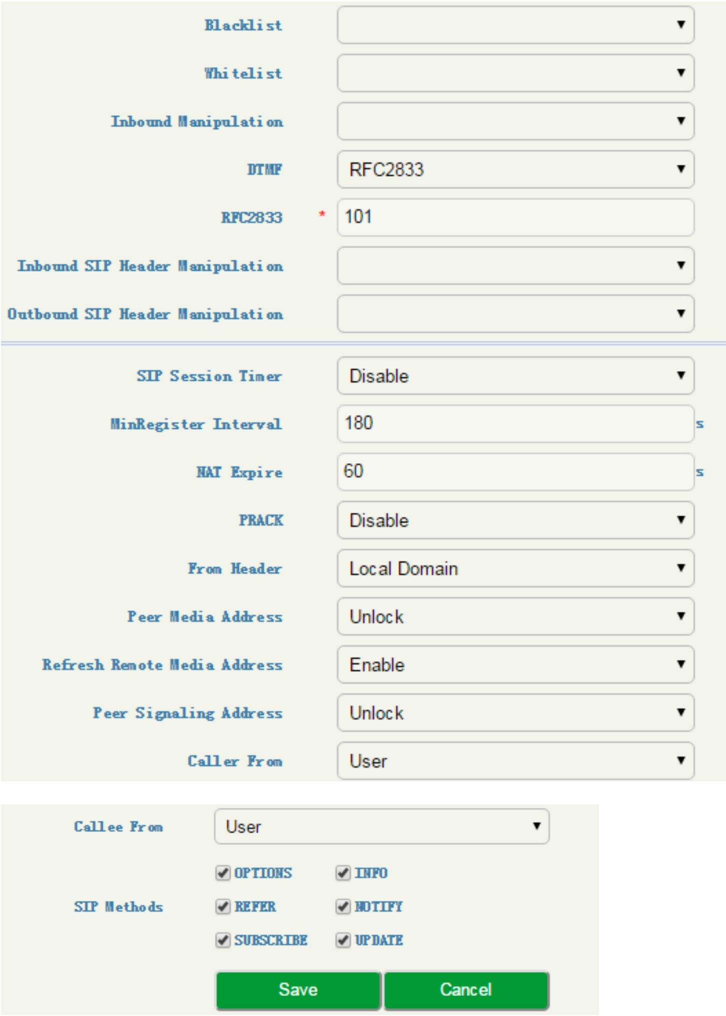


Figure 3-25 Configure Access SIP Trunk

Table 3-23 Access SIP Trunk

Name	The name of the access SIP trunk. It cannot be modified after the access SIP trunk has been added successfully
Description	The description of the access SIP trunk
Interface	The SBC300 device's Ethernet interface configured to connect the access SIP trunk. It can be eth0, eth1, eth2, eth3 or VLAN
Transport	Select a transport protocol for the access SIP trunk. It can be UDP, TCP or TLS
SIP Port	The access SIP trunk's SIP listening port on the Ethernet interface of SBC300
IPv4/IPv6	Select a network protocol for the access SIP trunk. It can be IPv4 or IPv6. By default, the network protocol is IPv4
Signaling DSCP	The QoS tag of SIP signaling messages
Media DSCP	The QoS tag of media messages

Near-end NAT	Near-end NAT defaults to disabled. If it is enabled, the contact IP address contained in SIP messages sent out by SBC300 will be turned into the outbound IP address of public network. If NAT is enabled, you need to fill in the outbound IP address of public network.
Rate Limit	The maximum RPS(registrations per second), CPS(calls per second) and total call volume of the access SIP trunk. Please refer to3.4.5
Codec	The codecs that the access SIP trunk supports. Please refer to 3.4.7
Blacklist	Select a blacklist for the access SIP trunk. Calls given by the caller numbers on the blacklist cannot be routed by the access SIP trunk. Please refer to 3.4.6
Whitelist	Select a whitelist for the access SIP trunk. Calls initiated by the caller numbers on the whitelist will be directed by the access SIP trunk. Please refer to 3.4.6 If no black list and white list are selected for the access SIP trunk, all calls can be routed by the access SIP trunk.
Inbound Manipulation	Select a number manipulation rule or a number pool for the access SIP trunk. When a call routed by the SIP trunk matches the manipulation rule, its number will be manipulated. Please refer to 3.4.8 and 3.4.9
DTMF	DTMF is short for Dual Tone Multi Frequency; There are three DTMF modes, including SIP Info, Inband, RFC2833; If the DTMF mode of an access SIP trunk differs from that of core network, SBC300 will convert it through DSP
Inbound SIP Header Manipulation	Select a SIP header manipulation rule for inbound calls of the access SIP trunk. If a call matches the manipulation rule, the SIP header of the messages related to the call will be manipulated when it comes into the access SIP trunk. Please refer to 3.4.10
Outbound SIP Header Manipulation	Select a SIP header manipulation rule for outbound calls of the access SIP trunk. If a call matches the manipulation rule, the SIP header of the messages related to the call will be manipulated when it goes out the access SIP trunk. Please refer to 3.4.10
Trunk Mode	**When SBC is connected to IMS,** **Static**: you need to manually configure the IP address and port of the peer device, for example, 192.168.2.159:5060 Remote domain name: the domain name of the peer **Dynamic**: the access SIP trunk works as a server, and you need to configure username, authentication ID and password for the SIP trunk, which will be used when a peer device tries to register to the SIP trunk. If the peer device registers to the SIP trunk successfully, the status of the SIP trunk will be 'True'. If the peer device fails to register or does not register to the SIP trunk, the status of the SIP trunk will be 'Flase'.

Registration	When 'Server IP Type' is configured as 'Static', registration will be displayed. If registration is enabled, the access IP trunk will be registered to the configured peer address and port, and the status of the access SIP trunk will become 'Ture'. Otherwise, the status is 'False'. For the status of access SIP trunk, please refer to 3.3.3 .
Keepalive	If 'Keepalive' is disabled, the system will not detect whether the access SIP trunk's peer device (generally it is the access network server) is reachable or not. If it is enabled, option message will be sent to detect the access network server is reachable. If response is received, it means the peer device is reachable, and the status of the access SIP trunk is 'True'. Otherwise, the status will be 'False'. For the status of access SIP trunk, please refer to 3.3.3 .
Times of No Response	The maximum number of timeouts for receiving response from the peer device after option messages are sent out.
Interval	The interval to send option message to the peer device
SIP Session Timer	Session timer is a mechanism to keep activating sessions. If 'Supported' is selected, SBC300 will send 'reinvite' messages to keep activating sessions within the configured duration. If no messages are detected within the configured duration, sessions will be considered as 'ended', and then will be disconnected. If 'Require' is selected, the callee side of a call passing through the access SIP trunk also needs to support session timer.
Session Expires	Configure the duration of the session. During the duration, SBC300 will send 'reinvite' messages to keep activating the session.
Min. Session Timeout	Minimum session duration is used to negotiate with the session timer on the callee side
PRACK	PRACK (Provisional Response ACKnowledgement): provide reliable provisional response messages. Disable: INVITE request and 1xx response sent out by SBC300 will not include *100rel* tag by default; Support: INVITE request and 1xx response sent out by SBC300 will include *100rel* tag in Supported header; Require: INVITE request and 1xx response sent out by SBC300 will include *100rel* tag in Require header; if the peer does not support 100rel, it will automatically reject INVITE request with 420; if the peer supports 100rel. it will send *PRACK* request to acknowledge the response.
From Header	It can be 'Local Domain' or 'Peer Domain'. 'Local Domain' is the default value.
Peer Media	Lock: when the peer device works at public network, media address carried in SDP

Address	(Session Description Protocol) message is locked; when the peer device works at private network, the address that sends 30 messages continuously are locked. Unlock: remote address sending media messages is not locked.
Refresh Remote Media Address	If this parameter is enabled, the remote address receiving media messages will be refreshed.
Peer Signaling Address	Lock: when a calling account is successfully registered, the access SIP trunk only receives those calls from the registered address of the caller.
Caller From	User: the USER field of FROM header of INVITE message is extracted as caller number Display: the DISPLAY field of FROM header of INVITE message is extracted as caller number
Callee From	User: the USER field of TO header of INVITE message is extracted as callee number； Display: the DISPLAY field of TO header of INVITE message is extracted as callee number； Request-uri: the USER NUMBER in REQUEST-URI of INVITE message is extracted as callee number；
SIP Methods	Configure the SIP request methods that can be accepted by the access SIP trunk. If a SIP request method is not enabled, the system will reject the corresponding SIP request. By default, the INVITE request, REGISTER request and SESSION DISCONNECT request are always accepted.

3.4.14 Core SIP Trunk

Core SIP trunk can realize the connection between SBC300 and the core network. On the **Service →Core SIP Trunk** page, you can configure the parameters of core SIP trunk.


Name *
Description
Valid
Interface eth0
Transport UDP
Port * 5060
IPv4/IPv6 IPV4
Signaling DSCP BE
Media DSCP BE
Near-end NAT
Codec default
Inbound Manipulation
DTMF RFC2833
RFC2833 * 101
Inbound SIP Header Manipulation
Outbound SIP Header Manipulation


Trunk Mode Static
Remote IP :Port *
Remote Server domain
Access Visit ACL table
+ Add
Registration
OutBound Proxy
Keepalive
SIP Session Timer Disable
PRACK Disable
From Header Local Domain
Peer Media Address Lock
Refresh Remote Media Address Enable
Peer Signaling Address Unlock
Caller From User

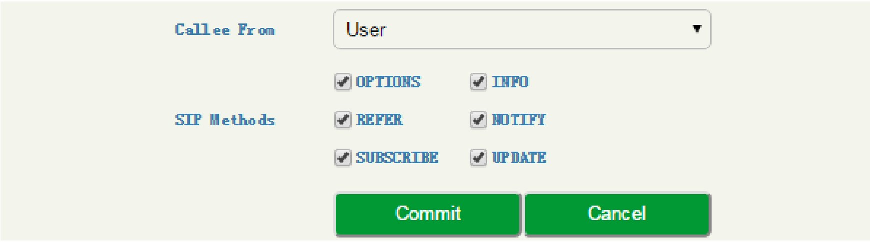


Figure 3-26 Core SIP Trunk

Table 3-24 Core SIP Trunk

Name	The name of the core SIP trunk. It cannot be modified after the access SIP trunk has been added successfully
Description	The description of the core SIP trunk
Interface	The SBC300 device's Ethernet interface configured to connect the core SIP trunk k. It can be eth0, eth1, eth2, eth3 or VLAN
Transport	Select a transport protocol for the core SIP trunk. It can be UDP, TCP or TLS
SIP Port	The core SIP trunk's SIP listening port on the Ethernet interface of SBC300
IPv4/IPv6	Select a network protocol for the core SIP trunk. It can be IPv4 or IPv6. By default, the network protocol is IPv4
Signaling DSCP	The QoS tag of SIP signaling messages
Media DSCP	The QoS tag of media messages
Near-end NAT	Near-end NAT defaults to disabled. If it is enabled, the contact IP address contained in SIP messages sent out by SBC300 will be turned into the outbound IP address of public network. If NAT is enabled, you need to fill in the outbound IP address of public network.
Rate Limit	The maximum RPS(registrations per second), CPS(calls per second) and total call volume of the core SIP trunk. Please refer to3.4.5
Codec	The codecs that the core SIP trunk supports. Please refer to 3.4.7
Blacklist	Select a blacklist for the core SIP trunk. Calls given by the caller numbers on the blacklist cannot be routed by the core SIP trunk. Please refer to 3.4.6
Whitelist	Select a whitelist for the core SIP trunk. Calls initiated by the caller numbers on the whitelist will be directed by the core SIP trunk. Please refer to 3.4.6 If no black list and white list are selected for the core SIP trunk, all calls can be routed by the core SIP trunk.
Inbound Manipulation	Select a number manipulation rule or a number pool for the core SIP trunk. When a call routed by the SIP trunk matches the manipulation rule, its number will be manipulated. Please refer to 3.4.8 and 3.4.9
DTMF	DTMF is short for Dual Tone Multi Frequency;

	There are three DTMF modes, including SIP Info, Inband, RFC2833; If the DTMF mode of an core SIP trunk differs from that of access network, SBC300 will convert it through DSP
Inbound SIP Manipulation	Select a SIP header manipulation rule for inbound calls of the core SIP trunk. If a call matches the manipulation rule, the SIP header of the messages related to the call will be manipulated when it comes into the core SIP trunk. Please refer to 3.4.10
Outbound SIP Manipulation	Select a SIP header manipulation rule for outbound calls of the core SIP trunk. If a call matches the manipulation rule, the SIP header of the messages related to the call will be manipulated when it goes out the core SIP trunk. Please refer to 3.4.10
Server IP Type	**When SBC is connected to IMS,** **Static**: you need to manually configure the IP address and port of the peer device, for example, 192.168.2.159:5060 Remote domain name: the domain name of the peer **Dynamic**: the access SIP trunk works as a server, and you need to configure username, authentication ID and password for the SIP trunk, which will be used when a peer device tries to register to the SIP trunk. If the peer device registers to the SIP trunk successfully, the status of the SIP trunk will be 'True'. If the peer device fails to register or does not register to the SIP trunk, the status of the SIP trunk will be 'Flase'.
Registration	When 'Server IP Type' is configured as 'Static', registration will be displayed. If registration is enabled, the core IP trunk will be registered to the configured peer address and port, and the status of the core SIP trunk will become 'Ture'. Otherwise, the status is 'False'. For the status of core SIP trunk, please refer to 3.3.4 .
Keepalive	If 'Keepalive' is disabled, the system will not detect whether the core SIP trunk's peer device (generally it is the core network server) is reachable or not. If it is enabled, option message will be sent to detect the core network server is reachable. If response is received, it means the core network server is reachable, and the status of the access SIP trunk is 'True'. Otherwise, the status will be 'False'. For the status of access SIP trunk, please refer to 3.3.3 .
Times of No response	The maximum number of timeouts for receiving response from the core network server after option messages are sent out.
Interval	The interval to send option message to the core network server
SIP Session Timer	Session timer is a mechanism to keep activating sessions. If 'Supported' is selected, SBC300 will send 'reinvite' messages to keep activating sessions within the configured duration. If no messages are detected within the configured duration, sessions will be considered

	as 'ended', and then will be disconnected. If 'Require' is selected, the callee side of a call passing through the core SIP trunk also needs to support session timer.
Session Expires	Configure the duration of the session. During the duration, SBC300 will send 'reinvite' messages to keep activating the session.
Mini Session Expires	The minimum session duration which is used to negotiate with the session timer on the callee side
PRACK	PRACK (Provisional Response ACKnowledgement): provide reliable provisional response messages. Disable: INVITE request and 1xx response sent out by SBC300 will not include *100rel* tag by default; Support: INVITE request and 1xx response sent out by SBC300 will include *100rel* tag in Supported header; Require: INVITE request and 1xx response sent out by SBC300 will include *100rel* tag in Require header; if the peer device does not support 100rel, it will automatically reject the INVITE request with 420; if the peer device supports 100rel, it will send the *PRACK* request to acknowledge the response.
From Header	It can be 'Local Domain' or 'Peer Domain'. 'Local Domain' is the default value.
Remote media send addresses	Lock: when the peer device works at public network, media address carried in SDP (Session Description Protocol) message is locked; when the peer device works at private network, the address that sends 30 messages continuously are locked. Unlock: remote address sending media messages is not locked.
Remote media receive address refresh	If this parameter is enabled, the remote address receiving media messages will be refreshed.
Peer Signaling IP	Lock: when a calling account is successfully registered, the core SIP trunk only receives those calls from the registered address of the caller.
Caller Number Field	User: the USER field of FROM header of INVITE message is extracted as caller number Display: the DISPLAY field of FROM header of INVITE message is extracted as caller number
Callee Number Field	User: the USER field of TO header of INVITE message is extracted as callee number； Display: the DISPLAY field of TO header of INVITE message is extracted as callee number； Request-uri: the USER NUMBER in REQUEST-URI of INVITE message is extracted

	as callee number;
SIP Methods	Configure the SIP request methods that can be accepted by the core SIP trunk. If a SIP request method is not enabled, the system will reject the corresponding SIP request. By default, the INVITE request, REGISTER request and SESSION DISCONNECT request are always accepted.

3.4.15 Routing Profile

1. SIP Trunk Group

On the **Routing Profiles → SIP Trunk Group** interface, you can group several access SIP trunks or core SIP trunks, and then set a strategy (backup or load balance) for choosing which truck will be used under a trunk group when a call comes in.

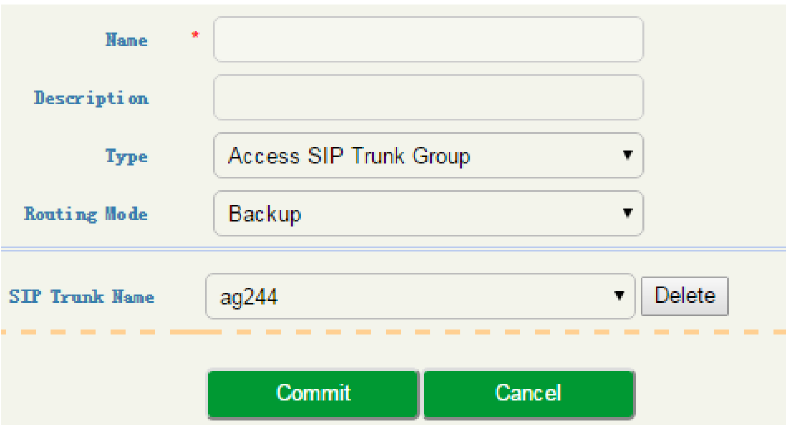


Figure 3-27 Configure SIP Trunk Group

Table 3-25 SIP Trunk Group

Name	The name of the SIP trunk group. It cannot be modified after the SIP trunk group has been added successfully
Description	The description of the SIP trunk group
Trunk Type	It can be access SIP trunk or core SIP trunk.
Routing Mode	The strategy for choosing which truck will be used under a trunk group when a call comes in. **Backup**: if the status of the first SIP trunk is 'True', the call will be always routed by the first SIP trunk. If the status of the first SIP trunk is 'False', the call will be routed by the next available SIP trunk.

	Load Balance: Trunk will be chosen according to the weight configured for it. For example, assuming the weight of a SIP trunk is 60% and that of the other SIP trunk in the same group is 40%, if there are 10 calls comes in, 6 calls will be routed by the first SIP trunk, and 4 calls will be routed by the second SIP trunk.
Trunk Name	The name of the access SIP trunk or core SIP trunk included in the trunk group

2. Call Routing

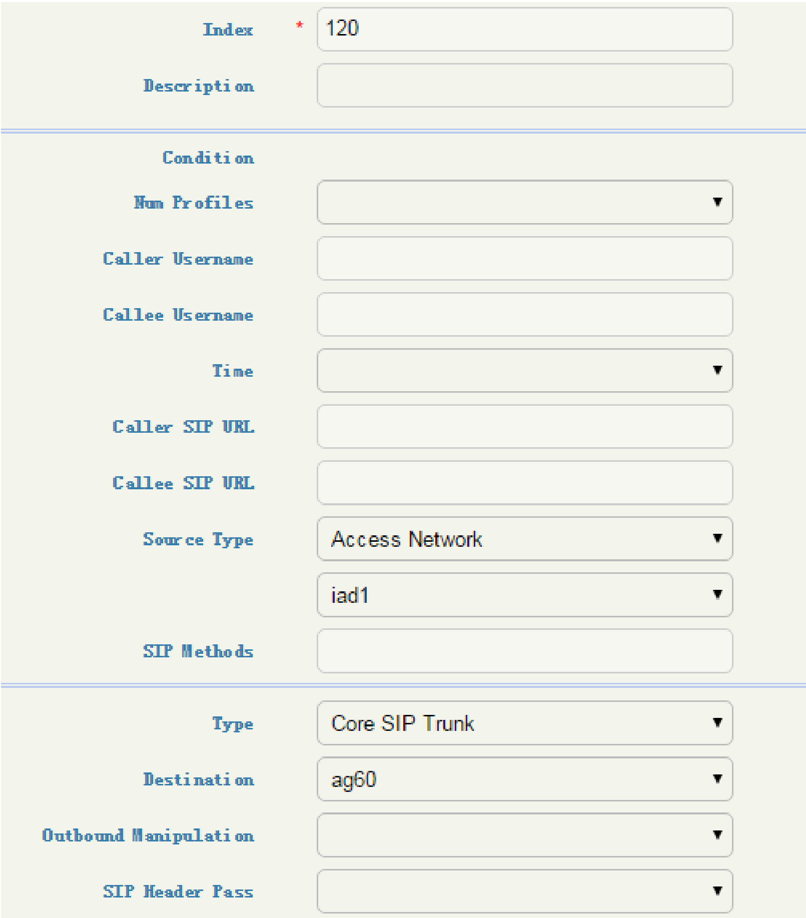


Figure 3-28 Call Routing

Table 3-26 Call Routing

Index	The index of the route, which determines the priority for a call to choose the route; the higher value, the lower priority.
Description	The description of the route, which is generally used to identify the route
Number Profile	The number profile set for matching the route. If the caller number or the called number of a call matches with a number in this profile, the call will be routed by the route. This parameter is optional to fill in. Make reference to 3.4.3 .

Caller Username	The caller number set for matching the route, which supports regular expression. If the caller number of a call matches with this number, the call will be routed by the route. If this parameter is null, it means caller number can be any number.
Callee Username	The callee number set for matching the route, which supports regular expression. If the callee number of a call matches with this number, the call will be routed by the route. If this parameter is null, it means callee number can be any number.
ime Profile	The profile of time during which the route can be used; If this parameter is null, it means the route can be used at anytime. Please make reference to 3.4.4
Caller SIP URL	If the 'SIP URL' field of the 'FROM' header of a request message sent by a caller number matches with the value configured here, the call will be routed by the route. If this parameter is null, it means the SIP URL from caller can be any.
SIP URL	If the 'SIP URL' field of the 'FROM' header of a request message sent by a callee number matches with the value configured here, the call will be routed by the route. If this parameter is null, it means the SIP URL from callee can be any.
Source Type	The source of the call routed by the route. If the source of a call is access network or access SIP trunk, the destination can only be core SIP trunk; If the source of a call is core SIP trunk, the destination can be access network or access SIP trunk.
SIP Methods	The SIP method(s) supported by the route. If this parameter is null, it means SIP methods can be any.
Destination Type	The destination of the call routed by the route. If the destination of a call is access network or access SIP trunk, the source can only be core SIP trunk; If the destination of a call is core SIP trunk, the source can be access network or access SIP trunk.
Destination	The specific SIP truck where a call will be routed
Number Manipulation	If it is on, the caller number or called number of a call routed by the route will be manipulated according to the configured manipulation rule; The parameter is off by default. For manipulation rule, please make reference to 3.4.8
SIP Header Passthrough	If it is on, the SIP header of a call routed by the route will be manipulated according to the configured manipulation rule; The parameter is off by default. For manipulation rule, please make reference to 3.4.10

Note:

Caller number or called number can also be manipulated when a call comes into an access network, access SIP trunk or core SIP trunk. In this section, number is manipulated after a call has finished choosing a route.

3.5 Security

In the **Security** section, you can configure the system security strategies, anti-attack strategies and access control strategies.

3.5.1 System

System security is mainly used to prevent SBC300 from being attacked by various DOS/DDOS floods, so as to ensure stable running of the device.

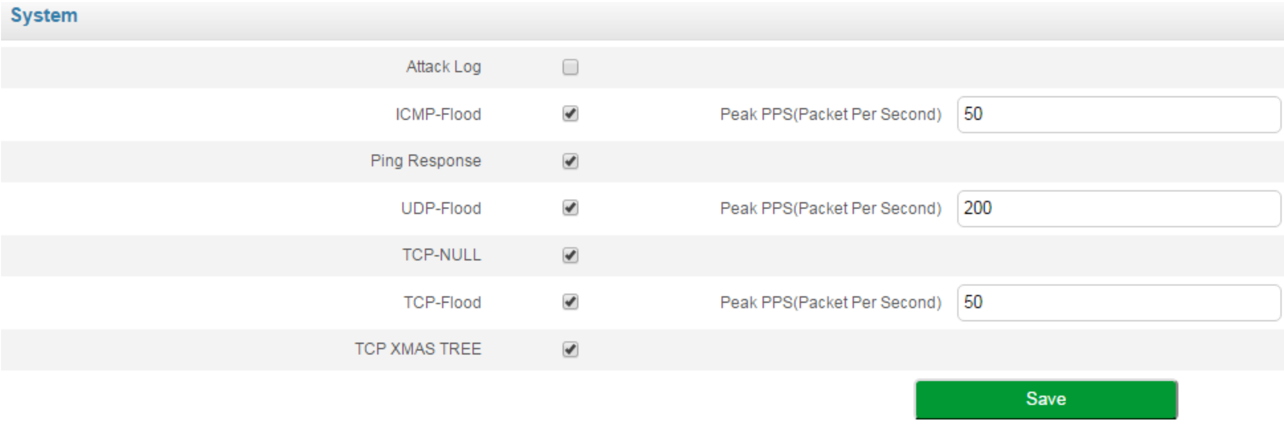


Figure 3-29 System Security

Table 3-27 System Security

Attack Log	If 'Attack Log' is enabled and SBC300 is attacked, the device will record the attack in logs which can be viewed on the **Maintenance →Log →Security Log** page.
ICMP-Flood	ICMP-Flood is a kind of DDOS attack. It can send a mass of ICMP packets to attack the SBC300 device. If this parameter is enabled, the device will drop those packets whose transmission rate exceeds the configured value of peak PPS(Packet Per Second); the range of the peak PPS is from 1 to 1000.
PING of Death	If this parameter is enabled, the SBC300 device will not give response to the PING request sent by devices in public network. It is disabled by default.
UDP-Flood	UDP-Flood is a kind of DDOS attack. It can send a mass of UDP packets to attack the SBC300 device. If this parameter is enabled, the device will drop those packets whose transmission rate exceeds the configured value of peak PPS (Packet Per Second); the range of the peak PPS is from 1 to 1000.
TCP-NULL	TCP NULL is a scan to determine if ports are closed on the target device.

	If this parameter is enabled, SBC300 will drop TCP packages, and the peer device cannot learn whether the ports of SBC300 are closed or not.
TCP-Flood	TCP-Flood is a kind of DDOS attack. It can send a mass of TCP requests to occupy the system resources of the target device and then to make the target device crash. If this parameter is enabled, the device will drop those packets whose transmission rate exceeds the configured value of peak PPS (Packet Per Second); the range of the peak PPS is from 1 to 1000.
TCP XMAS TREE	TCP XMAS TREE can send TCP packets with special tag to detect which ports are open on the target device. If this parameter is enabled, SBC300 will drop thoseTCP packages, and the peer device cannot learn which ports of SBC300 are open.

3.5.2 Access Control

On the **Security →Access Control** page, you can configure the access ports for Web and SSH as well as the access control of GE0, GE1, GE2 and GE3.

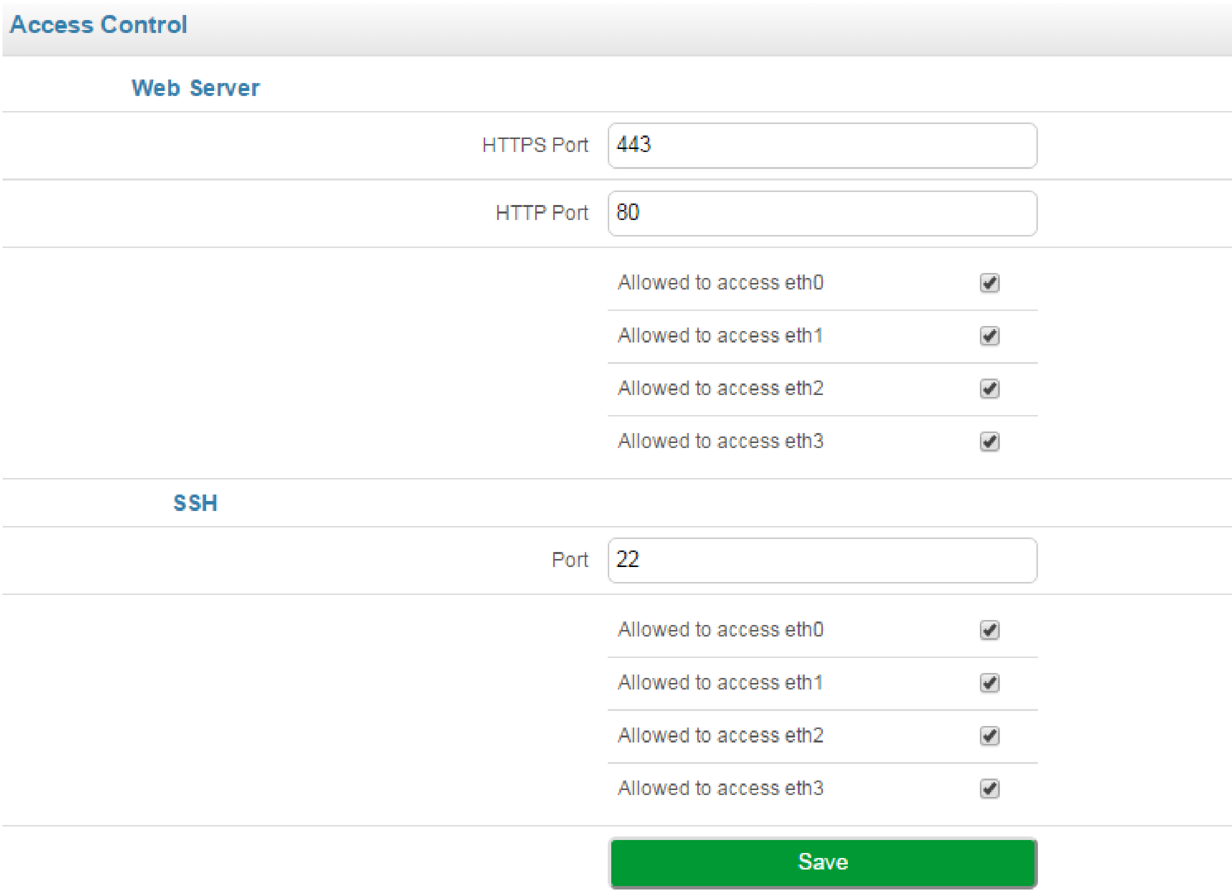


Figure 3-30 Access Control

Table 3-28 Access Control

Web Server	The Web interface of SBC300 only supports https, and the https port defaults to 443. You can modify the https port; If you select the checkbox on the right of GE0, GE1, GE2 or GE3, it means the selected port.is allowed to access the Web interface of SBC300. By default, GE0, GE1, GE2 and GE3 are not allowed to access the Web interface.
SSH	The SSH port of SBC300 defaults to 22. If you select the checkbox on the right of GE0, GE1, GE2 or GE3, it means the selected port.is allowed to access the SSH of SBC300. By default, GE0, GE1, GE2 and GE3 are not allowed to access the SSH.

3.5.3 **Security Policy**

1. IP Security Strategy

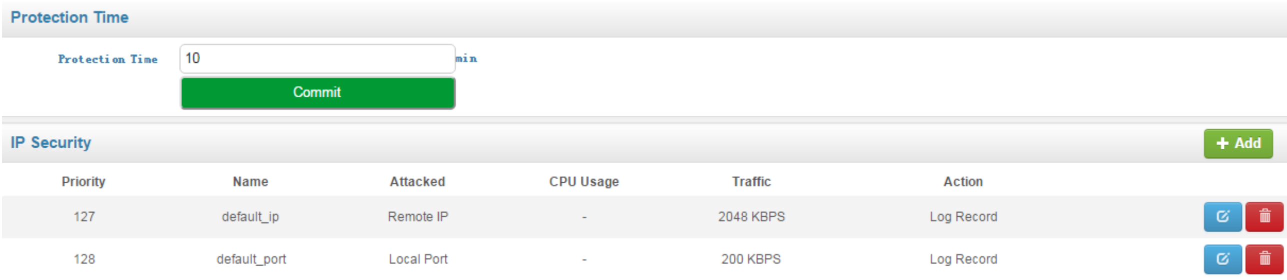


Figure 3-31 IP Security Strategy

Click + Add to add a strategy to prevent attacks from other IP addresses. Click to delete a strategy, while click to modify the strategy.

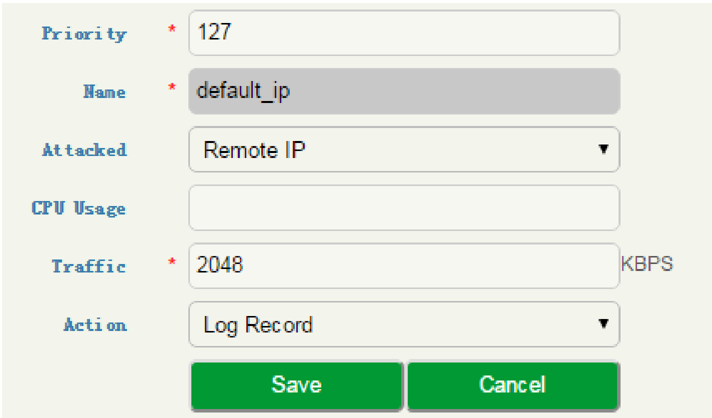


Figure 3-32 Add IP Security Strategy

Table 3-29 IP Security Strategy

Time Limiting	The validity time of the IP security strategy. When the validity time expires, the strategy needs to be retriggered, otherwise it will not takes effect.
Index	The greater digit, the lower priority
Description	The description of the IP security strategy. It cannot be modified after the strategy has been successfully added.
Detection	Remote IP: when the packet traffic sent by remote IP exceeds the configured traffic threshold (KBPS) or the CPU usage exceeds the configured threshold, SBC300 will execute the preset action. Local port: when the packet traffic received by local port exceeds the configured traffic threshold (KBPS) or the CPU usage exceeds the configured threshold, SBC300 will execute the preset action.
CPU Usage	The CPU usage rate If this parameter is null, it means CPU usage is not a condition for triggering security strategy.
Traffic（KBPS）	The maximum packet traffic sent by the peer IP or received by local port. If this threshold is surpassed, SBC300 will execute the configured action on the packets.

Action	Log Record: when the security strategy is triggered and takes effect, the attack event is recorded in a log Flow Limited: when the security strategy is triggered and takes effect, the traffic of peer IP address or the set local port is limited, and those packets whose traffics exceed are dropped during the limitation time. Packet Rate Limited: when the security strategy is triggered and takes effect, the packet rate of peer IP address or the set local port is limited, and those packets whose traffics exceed are dropped during the limitation time. Drop: when the security strategy is triggered and takes effect, all the packets from peer IP address and those received by the set local port are dropped during the limitation time.

2. SIP Security

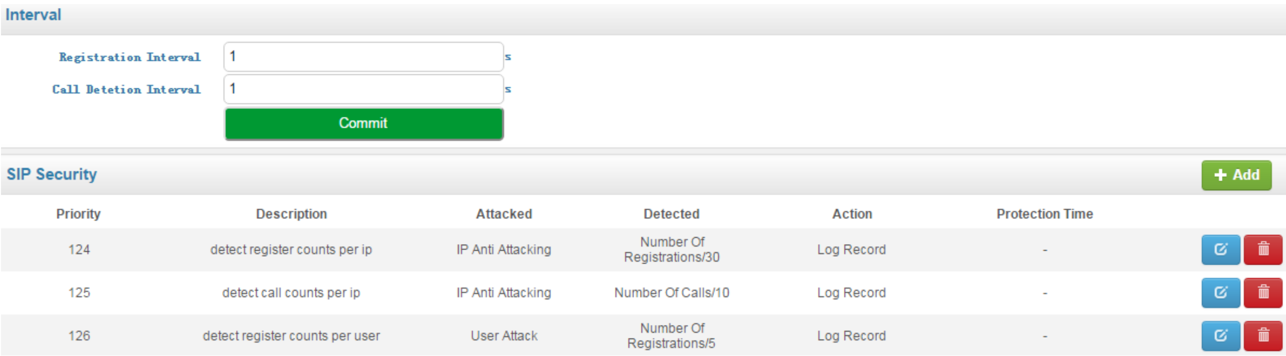


Figure 3-33 SIP Security Strategy

Click + Add to add a strategy to prevent attacks from SIP-based devices. Click to delete a strategy, while click to modify the strategy.

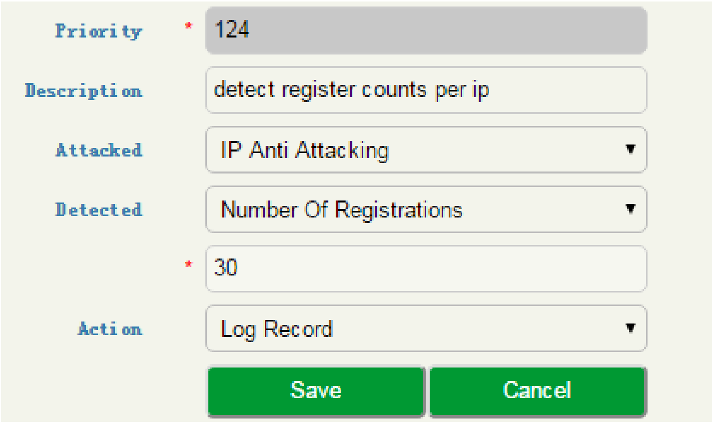


Figure 3-34 Add SIP Security Strategy

3.6 System

On the System pages, you can configure the device name, certification, network, port mapping, static routes, username & password as well as time zone & current time. You can also upgrade software versions, backup or restore configuration data, and update license and certificate.

3.6.1 Device Name

On the **System → System Management** page, you can configure the name of the SBC300 device.

Figure 3-35 Device Name

3.6.2 Web Configuration

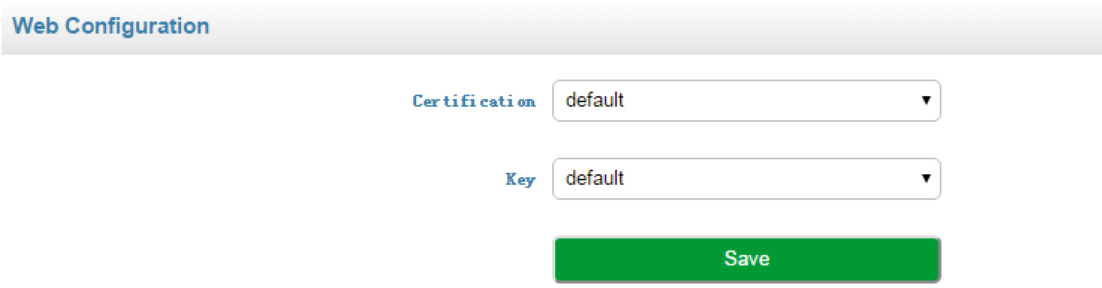
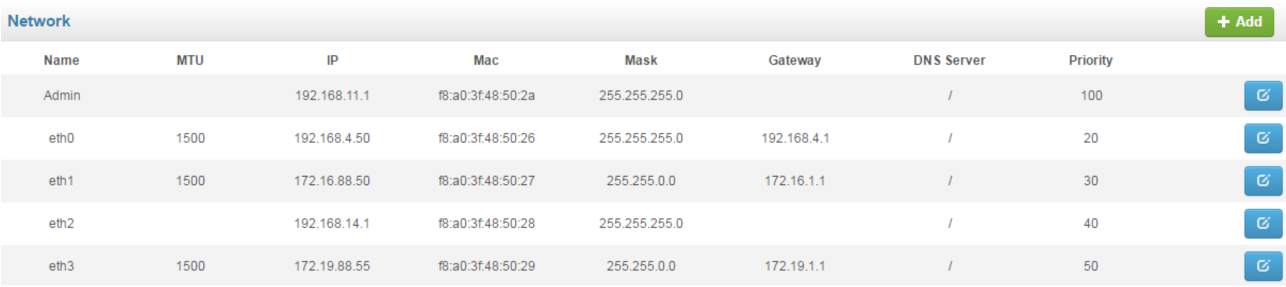


Figure 3-36 Web Configuration

3.6.3 Network

On the **System → Network** page, you can configure the IP address, Subnet mask, gateway and DNS server. You can also add VLAN on the page.



Network + Add

Name	MTU	IP	Mac	Mask	Gateway	DNS Server	Priority
Admin		192.168.11.1	f8:a0:3f:48:50:2a	255.255.255.0		/	100
eth0	1500	192.168.4.50	f8:a0:3f:48:50:26	255.255.255.0	192.168.4.1	/	20
eth1	1500	172.16.88.50	f8:a0:3f:48:50:27	255.255.0.0	172.16.1.1	/	30
eth2		192.168.14.1	f8:a0:3f:48:50:28	255.255.255.0		/	40
eth3	1500	172.19.88.55	f8:a0:3f:48:50:29	255.255.0.0	172.19.1.1	/	50

Figure 3-37 Network Port

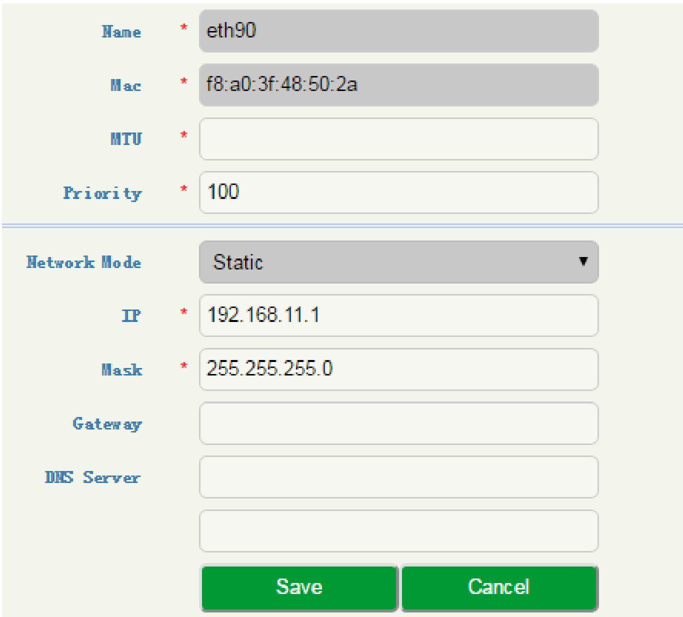


Figure 3-38 Modify Port Infomation

Click to add a VLAN and click to modify the information of each network port or VLAN, while click to delete a VLAN.

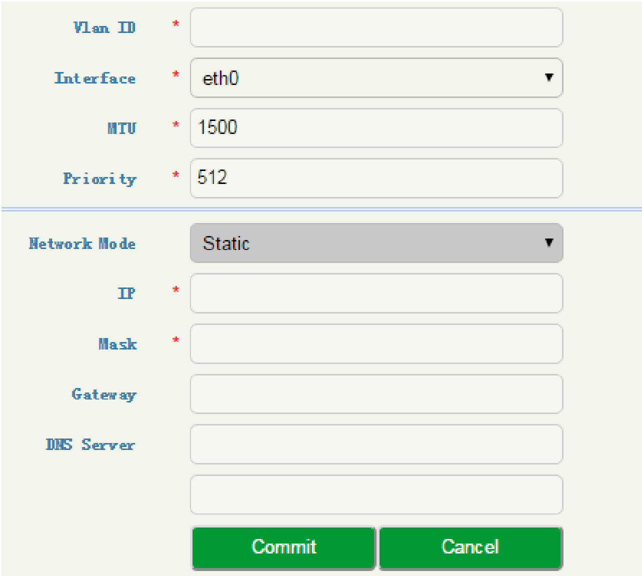


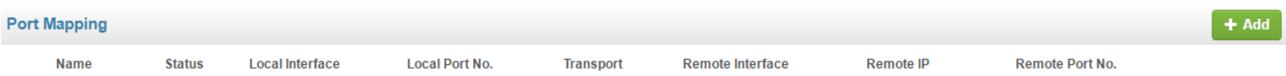
Figure 3-39 Add VLAN

Table 3-30 Network Configuration

VLAN ID	The ID of the added VLAN
Interface	Network port: Admin, GE0, GE1, GE2 and GE3
MTU	The MTU (Maximum Transmission Unit) of the network port
Priority	When SBC300 visits an IP address of other network segment and this peer IP address is not directed by static route, SBC300 will go out from the network port or VLAN with the highest priority. The smaller digit, the higher priority.
Network Mode	The way for network port (Admin, GE0, GE1, GE2 and GE3) to get its IP address. Currently, SBC300 only supports static IP address.
IP address	The IP address of network port or VLAN
Mask	The subnet mask of network port or VLAN
Gateway	The gateway of network port or VLAN
DNS Server	The address of DNS server of network port or VLAN

3.6.4 Port Mapping

To ensure the security of the LAN (local-area network), SBC300 will reject the connection request from the wide-area network (WAN). Port mapping allows a client in the wide-area network to visit the SBC300 device in the local-area network.



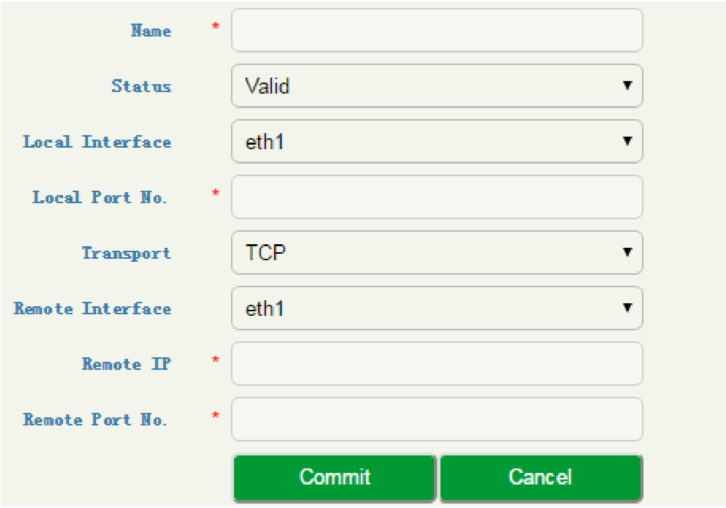


Figure 3-40 Configure Port Mapping

Table 3-31Port Mapping

Name	The name of this port mapping
Status	To enable or disable
Local Interface	The mapped interface of the SBC300 device in local-area network
Local Port Number	The mapped port of the SBC300 device in local-area network (this port cannot conflict with the in-use port of the SBC300 device)
Transport Protocol	Choose TCP, UDP or TCP\UDP
Remote Interface	The interface of the client in the wide-area network, which is to visit the SBC300 device in local-area network1
Remote Port Number	The port of the client in the wide-area network, which is to visit the SBC300 device in local-area network
Remote IP Address	The IP address of the client in the wide-area network, which is to visit the SBC300 device in the local-area network.

3.6.5 Static Route

On the **System → Static Route** interface, you can configure static routes for the network. After a static route is successfully set, related packets will be sent to the designated destination according to the static route. Click

to enter into the setting page of static route.

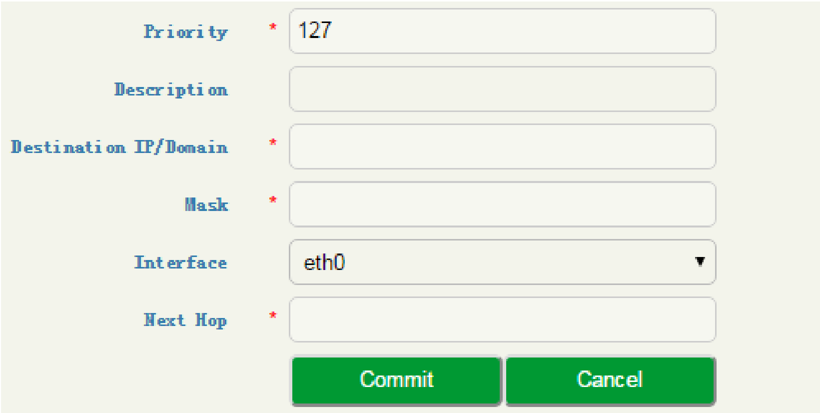


Figure 3-41 Add Static Route

Table 3-32 Static Route

Priority	The priority of the static route. The smaller digit, the higher priority
Description	The description of the static route
IP Destination IP	The destination IP address of the static route
Mask	The netmask of the static route, such as 255.255.255.0
Interface	The source interface of the static route, such as GE0, GE1,GE2 and GE3
Nexthop	The next hop address, namely the router address passed by the packets before they reach the destination address

3.6.6 User Manager

On the **System → User Manager → Password** page, you can modify administrator's password for logging in the SBC300 device. Factory defaults for administrator's username and password are 'admin' and 'admin@123#' which are also used to log in SSH.

Password

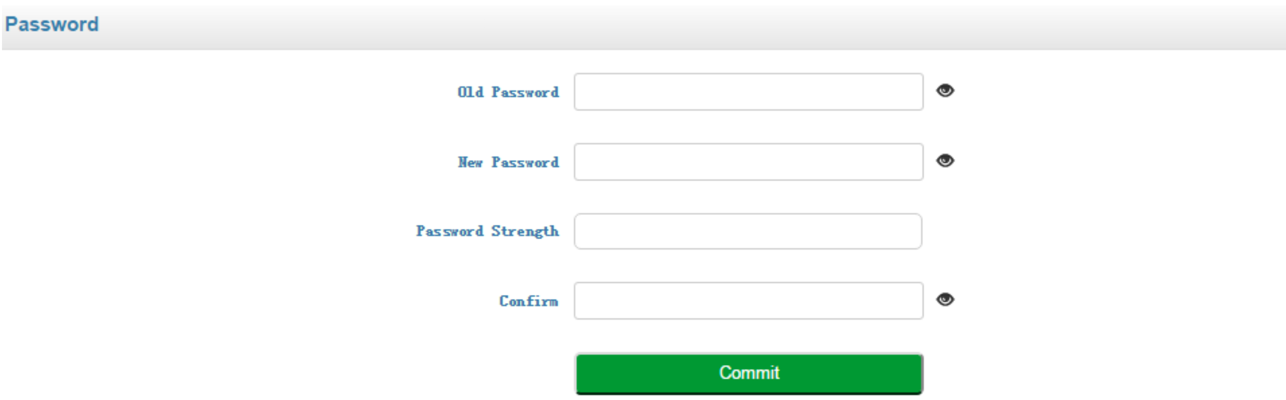


Figure 3-42 Modify Password

User List

On the **System → User Manager →User List** page, the administrator can add the users that are allowed to log in the Web interface, specify their roles and allocate permissions to them.

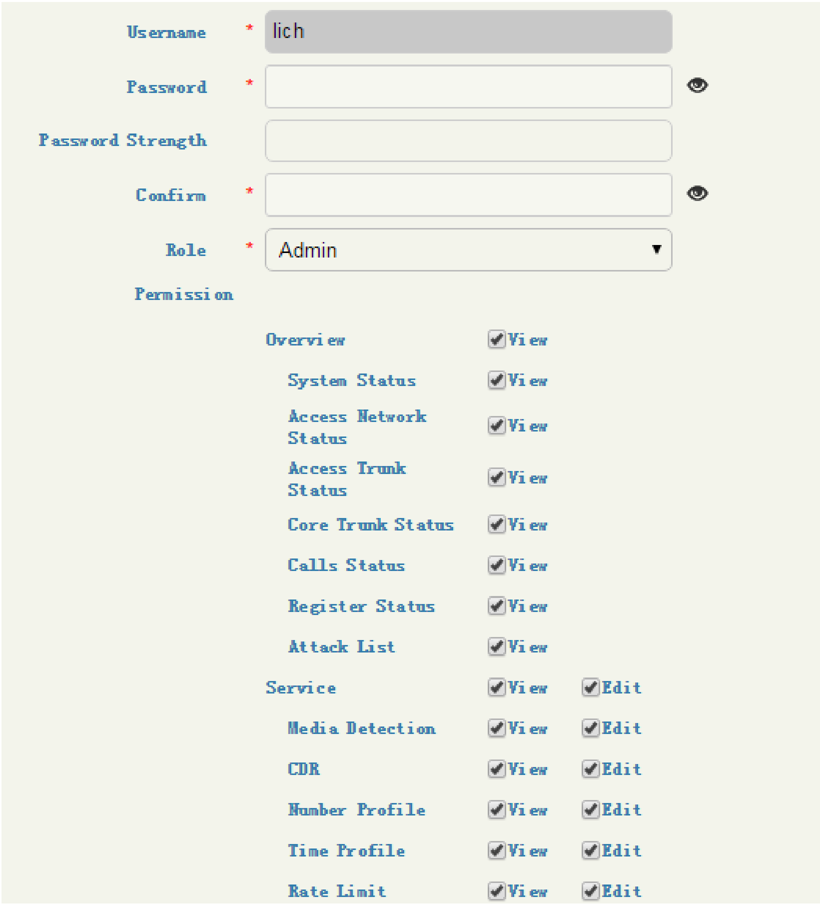


Figure 3-43 Add User and Assign Permissions

Table 3-33 User List

Username	The name of the user, which is used to log in the SBC300 device
Password	The password for the user to log in the SBC300 device
Confirm	Confirm the password
Password Strength	The security strength of the password
Role	Admin: has the permission to add users whose role is operator or observer, to modify the passwords of users, to add/delete/modify configurations. Only one administrator is allowed for one SBC300 device. Operator: has the permission to view configurations, or modify part of the configurations. Observer: has the permission to view existing configurations, but cannot delete or modify them.

3.6.7 Date & Time

On the **System → Date & Time** page, you can set a new time zone, synchronize local time and add NTP server.

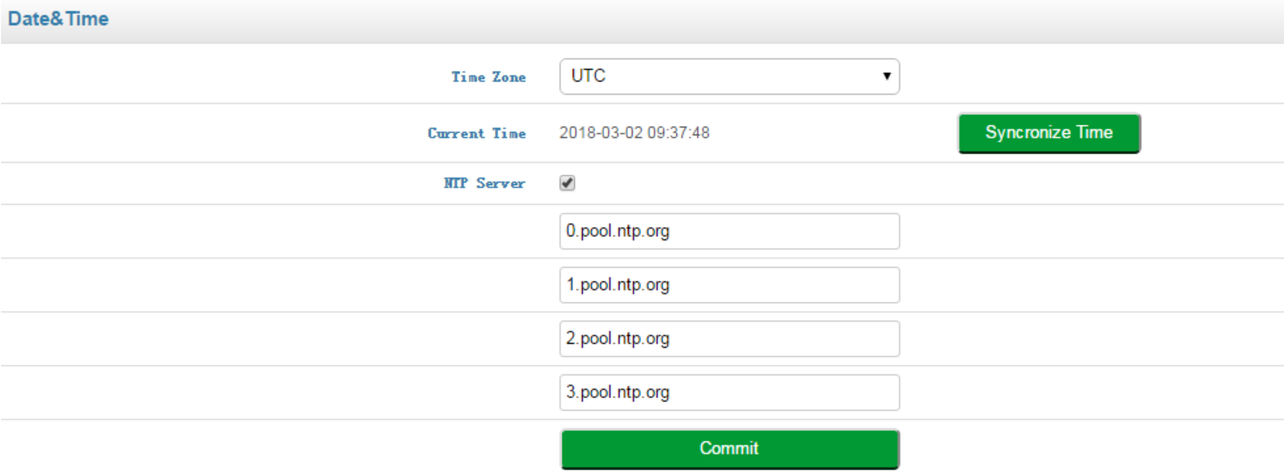


Figure 3-44 Configure Date & Time

Table 3-34 Date & Time

Time Zone	Choose a time zone for the SBC300 device according to the location where the device is placed.
Synchronize Time	If the current time of SBC300 is wrong and the device fails to synchronize with a NTP server, you can synchronize the current time to that of the PC which is used to log in the SBC300.
NTP Server	If NTP server is enabled, the time of SBC300 will be synchronize to that of NTP server.

3.6.8 Upgrade

On the **System → Upgrade** interface, you can upgrade the SBC300 to a new version. But you need to restart the device for the change to take effect after executing upgrade.

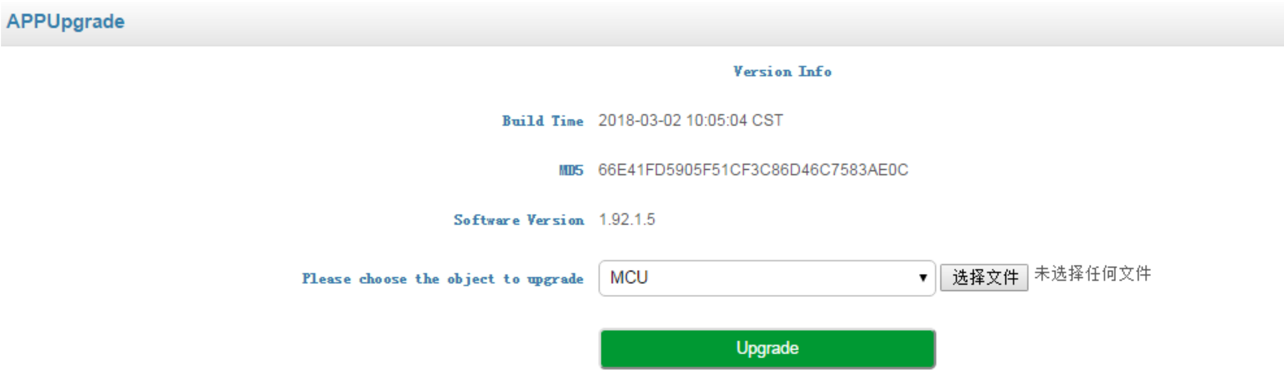


Figure 3-45 Software Upgrade

The version file used for upgrade is generally named as ‘1.91.x.x.ldf’. Please do not use other products’ version files to upgrade the SBC300 device.

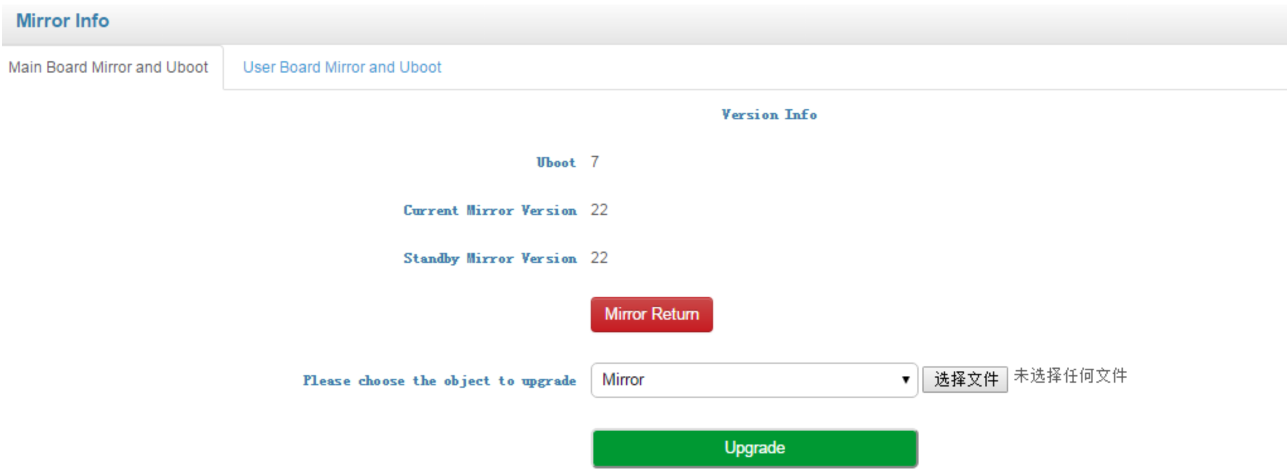


Figure 3-46 Mirror Upgrade

3.6.9 Backup & Restore

On the **System → Backup & Restore** interface, you can back up or restore all the configuration data, including service configurations, network configurations and license & certificate. After the configuration data is restored, the SBC300 device will automatically restart.

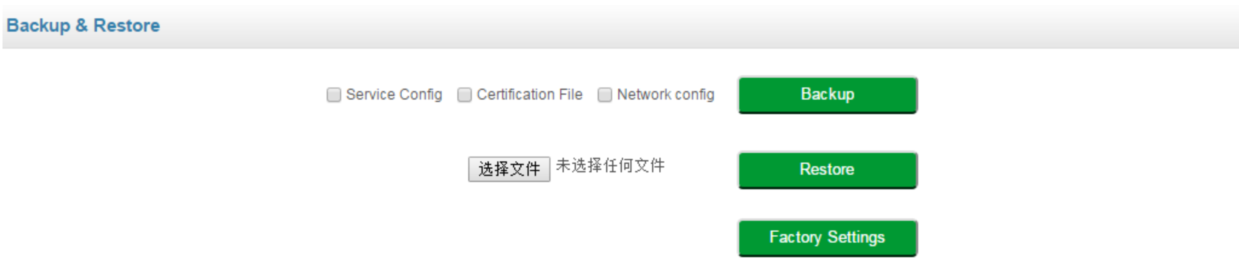


Figure 3-47 Backup & Restore

Table 3-35 Backup & Restore

Backup	You can download the configuration data to be taken as backup. Select any of the checkboxes on the right of Service Config, Certification File and Network Config, and then click **Backup**
Restore	Choose a backup file, and then click **Restore**.
Factory Settings	Click **Factory Settings**, and the configurations of the SBC300 device will become factory settings.

3.6.10 Double-device Hot Standby

Two SBC300 devices can be connected with each other through the 'Admin' port for the sake of hot standby. That is to say, the two SBC300 devices work in the active/standby mode. When the active device fails, it changes to the standby state while the standby device changes to the active state and take over the functionality of the failed device. In this way, services such as calling and transcoding, provided by SBC300, will not be interrupted in case that one of the SBC300 devices malfunctions.

3.6.11 License

On the **System → License** page, the license information, including license beginning time, license expiry time, maximum concurrent calls, maximum transcoded sessions, maximum registered users, RPS (registrations per second) and CPS(calls per second), is displayed. The SBC300 device will not accept registrations and calls after the license expires.

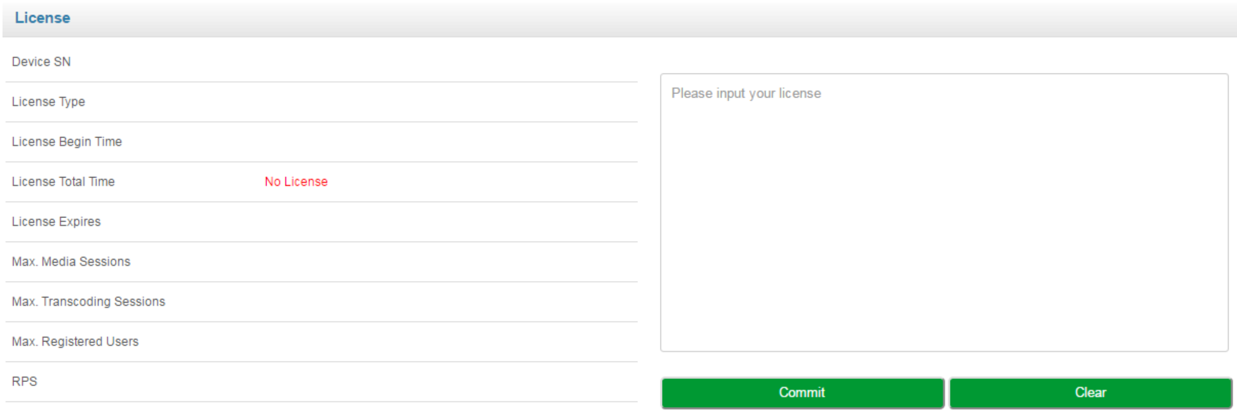


Figure 3-48 License Information

3.6.12 Certificate

On the **System → Certificate** page, you need to upload a certificate to ensure the secure login to the Web interface of the SBC300 device. You cannot log in the device until you has uploaded a certificate.

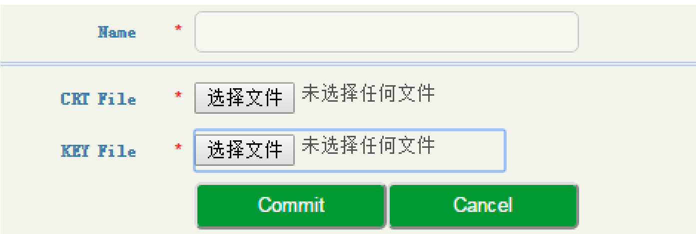


Figure 3-49 Upload Certificate

3.7 Maintenance

3.7.1 Login Log

The logs tracing the logins of the SBC300 device can be viewed on the **Maintenance → Login Log** page. You are allowed to set query criteria to view the logs that you want.

Login Log

10 ▾ Search: Name | Type | Begin Time | End Time | Source | Commit

Index	Username	Role	Time	Login IP	Source	Description
1	admin	admin	2018-01-26 06:34:05	172.19.120.143:53289	web	Login success
2	admin	admin	2018-01-24 12:16:13	172.19.165.114:56018	web	Login success
3	admin	admin	2018-01-24 12:15:54	172.19.165.114:56018	web	CAPTCHA FAILED
4	admin	admin	2018-01-22 06:50:35	172.19.17.71:54873	web	Login success
5	admin	admin	2018-01-22 06:49:55	172.19.17.71:54873	web	Login failed
6	admin	admin	2018-01-22 06:49:38	172.19.17.71:54873	web	CAPTCHA FAILED
7	admin	admin	2018-01-22 06:48:07	172.19.17.71:54873	web	CAPTCHA FAILED
8	admin	admin	2018-01-22 06:36:57	172.19.17.71:54873	web	Login failed
9	admin	admin	2018-01-17 09:49:33	172.19.120.143:55372	web	Login success
10	admin	admin	2018-01-17 08:37:09	172.19.120.143:54181	web	Login success

Figure 3-50 Login Log

3.7.2 Operation Log

The logs tracing the operations carried out on the Web interface can be queried on the **Maintenance → Operation Log** page. You are allowed to set query criteria to view the logs that you want.

Operation Log

10 ▾ Search: Name | Type | Begin Time | End Time | Source | Commit

Index	Username	Role	Time	Login IP	Source	Operation	Content
1	admin	admin	2018-01-26 06:37:05	172.19.120.143:53404	web	Reboot	System
2	admin	admin	2018-01-26 06:36:55	172.19.120.143:53404	web	Reboot	UserBoard
3	admin	admin	2017-10-26 12:35:01	172.19.120.143:49578	Web	撤销	IP Security
4	admin	admin	2017-10-23 12:33:53	172.19.120.143:57868	Web	Mod.	Time Limiting/10

Figure 3-51 Operation Log

3.7.1 Security Log

The logs related to security can be viewed on the **Maintenance → Security Log** page. You are allowed to set query criteria to view the logs that you want.

Figure 3-52 System Log

3.7.2 Log Management

On the **Maintenance → Log Management** page, you can set the log level to filter logs, and can export the logs of different level.

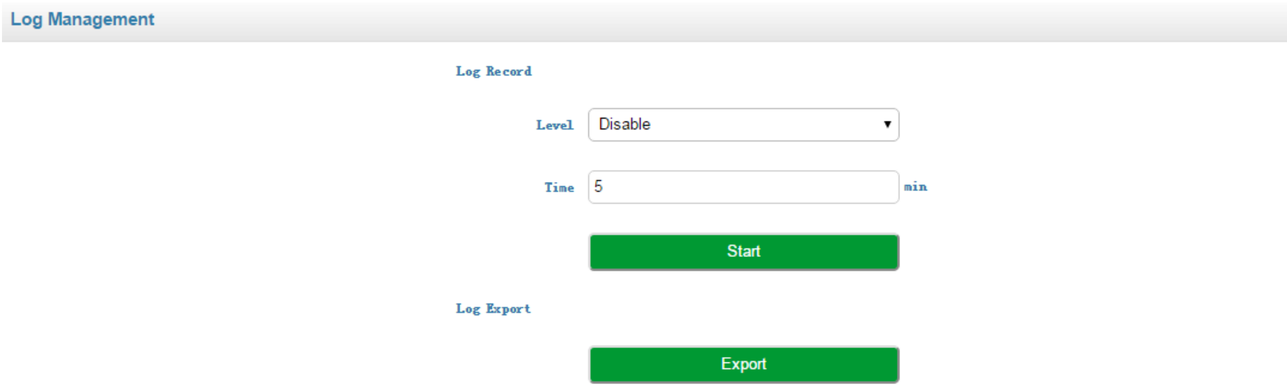


Figure 3-53 Log Management

3.7.3 Tools

On the **Maintenance → Tools** page, you can use three network utilities including Ping, Traceroute and Nslookup to diagnose the network, and can capture data packages of the available network ports.

[PING]

Ping is used to examine whether a network works normally through sending test packets and calculating response time.

Instructions for using Ping:

1. Enter the IP address or domain name of a network, a website or a device in the input box of Ping, and then click **Ping**.
2. If related messages are received, it means the network works normally; otherwise, the network is not connected or is connected faultily.

[**Traceroute**]

Traceroute is used to determine a route from one IP address to another.

Instruction for using Traceroute:

Step1. Enter the IP address or domain name of a destination device in the input box of Traceroute, and then click **Traceroute**.

Step2. View the route information from the returned message.

[Network Capture]

On the following interface, you can capture data packages of the available network ports. You can also set source IP, source port, destination IP or destination port to capture the packages that you want.

4 Abbreviation

SBC: （Session Border Controller）

SIP: （Session Initiation Protocol）

DTMF: （Dual Tone Multi Frequency）

NAT：（Network Address Translation）

VLAN：（Virtual Local Area Network）