



# **SBC1000 Session Border Controller**

## **User Manual V1.0**



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# Preface

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## Welcome

Thanks for choosing **SBC1000 Session Border Controller**! We hope you will make full use of this rich-feature device. Contact us if you need any technical support: 86-0755-26456110/112.

## About This Manual

This manual gives introduction to the SBC1000 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 SBC1000 device

## Revision Record

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## Conventions

Device mentioned in this document refers to the SBC1000 Session Border Controller. Those words specially noted in the document are the contents that users need to pay attention to.



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# 1 Production Introduction

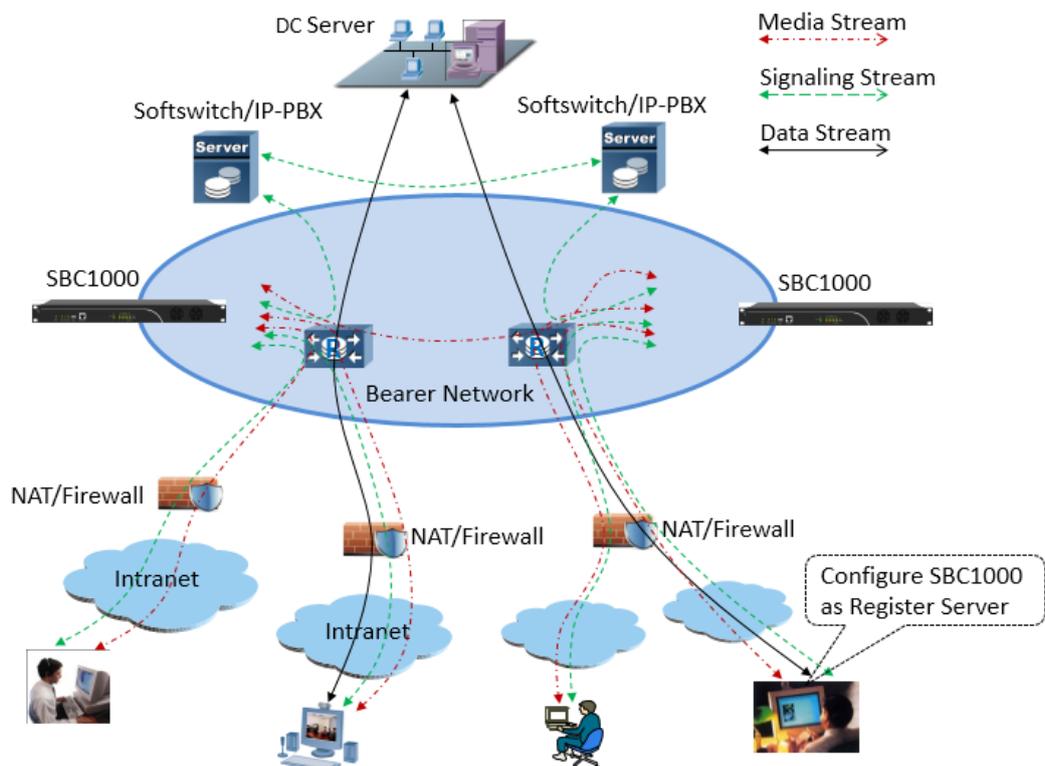
## 1.1 Overview

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 SBC1000 session border controller can help enterprises to solve the abovementioned problem.

**Dinstar SBC1000** 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, SBC1000 delivers high capability while achieves low power dissipation. It is able to process up to 500 concurrent SIP sessions and transcode 200 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, SBC1000 also supports the transcoding of iLBC, AMR and OPUS.

## 1.2 Application Scenario

Figure 1-1 Application Scenario of SBC1000



## 1.3 Product Appearance

Front View:



Back View:



## 1.4 Description 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 ( 0-3 ) /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

## 1.5 Functions and Features

### 1.5.1 Key Features

- Support up to 5000 SIP registrations
- Support 500 concurrent SIP sessions and transcode 200 concurrent sessions
- SIP trunks & flexible routing rules for accessing IMS
- Media transcoding and **fax transcoding (to do)**
- Embedded VoIP firewall, prevention of DoS and DDos attacks
- Bandwidth limitation and dynamic white list & black list
- QoS, static route, NAT traversal
- Import & export of remote upgrade and configuration data
- Encrypted sessions through SRTP and ‘SIP over TLS’
- 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\* SIM Card Slot
- Serial Console
  - 1\* RS232, 115200bps, RJ45
- **LTE Uplink (to do)**

### 1.5.3 Capabilities

- Concurrent Calls  
Support 500 SIP sessions at maximum
- Transcoding  
Supports 200 transcoding calls
- CPS for call  
25 calls per second at maximum
- Registrations  
Maximum SIP registrations: 5000
- CPS for Registration  
25 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

- Redundant Power Supply: 100-240VAC, 50-60 Hz
- Power Consumption: 15w
- Operating Temperature: 0 °C ~ 45 °C
- Storage Temperature: -20 °C ~80 °C
- Humidity: 10%-90% Non-Condensing
- Dimensions (W/D/H): 437×300×44mm (1U)
- Unit Weight: 4.5 kg
- Compliance: CE, FCC

# 2 Installation

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## 2.1 Preparations before Installation

### 2.1.1 Attentions for Installation

Before you install the SBC1000 device, please read the following safety guidelines:

- To guarantee SBC1000 works normally and to lengthen the service life of the device, the humidity of the equipment room where SBC1000 is installed should be maintained at 10%-90% (non-condensing), and temperature should be 0 °C ~ 45 °C;
- Ensure the equipment room is well-ventilated and clean;
- Power supply of SBC1000 should be 100 ~ 240V AC, and its socket is a three-pin socket which should be grounded well;
- It's suggested that personnel who has experience or who has received related training be responsible for installing and maintaining SBC1000;
- Please wear ESD wrist strap when installing SBC1000;
- 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 SBC1000.
- **Trunk**  
Ensure telecom operator has approved to open a trunk.
- **IP Network**  
Ensure router under IP network has been prepared, since SBC1000 is connected to the IP network through the standard 10/100/1000M Ethernet port.
- **Power Supply**  
Ensure the socket of SBC1000 is a three-pin socket and power supply is grounded well.

### 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 SBC1000 device and all accessories have been in it:

- One SBC1000 device
- One 1.8-meter-long of power wire (AC 250V/4A)
- Two network cables
- One grounding cable
- One serial console cable
- Flanks and screws

## 2.2 Installation of SBC1000

### 2.2.1 Put SBC1000 into Shelf

Put the SBC1000 device on the shelf or cabinet horizontally;

### 2.2.2 Connect SBC1000 to Network

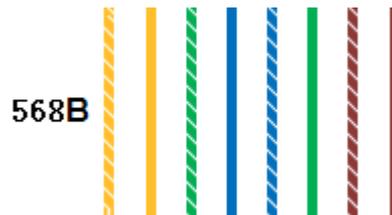
SBC1000 has five network ports, namely the gigabit network port for services (from GE0 to GE3) and the gigabit network port for network management (Admin). It is advised to connect GE0, GE1, GE2 or GE3 to the IP network.

Both GE0/GE1/GE2/GE3 and Admin can be used to carry out management on SBC1000, 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 SBC1000.

## 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.

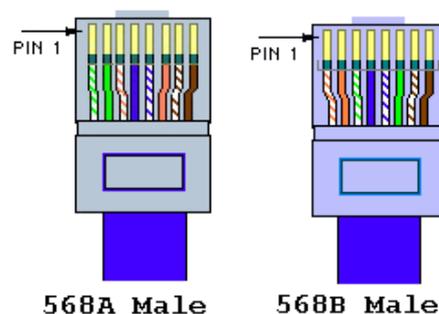
**Step3.** Put the wires into the PINs of a RJ45 joint according to the abovementioned wire sequence of EIA/TIA 568B, and then use a wire crimper to crimp the RJ45 joint.

**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 Connect SBC1000 to Network

SBC1000 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.

Both GE0/GE1/GE2/GE3 and Admin can be used to carry out management on SBC1000, but only GE0/GE1/GE2/GE3 is put in use generally. Admin is used when there is a need to separate management processing from service processing on SBC1000.

## 2.2.5 Troubleshooting about Network Connection

When the SBC1000 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 SBC1000 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

SBC1000 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.

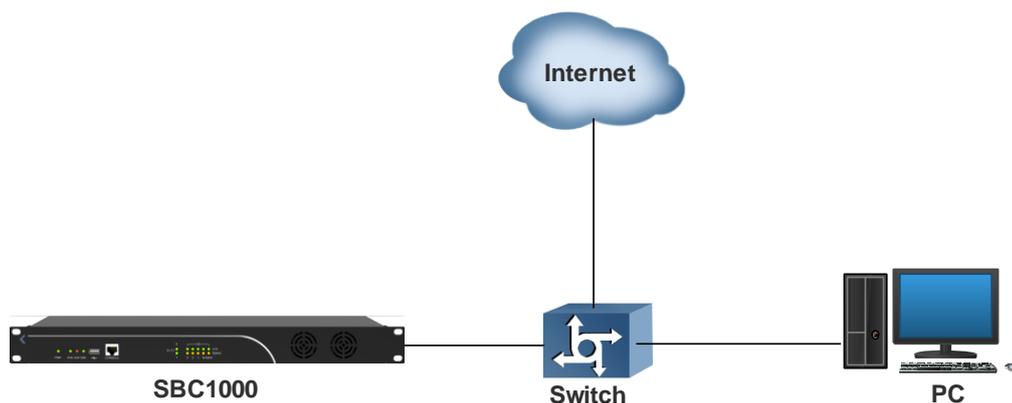
The default IP address of the 'Admin' port is 192.168.11.1, while those of GE0, GE1, GE2 and GE3 are 192.168.12.1, 192.168.13.1, 192.168.14.1 and 192.168.15.1 respectively.

#### First Use

At the first time that the SBC1000 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 Admin1 port is 192.168.11.1

#### Daily Use

Connect the service port (GE0/GE1/GE2/GE3) of SBC1000 to a 1000Mbps or 10/100Mbps switch.



If SBC1000 is connected to a 1000Mbps switch, the link indicators on the front panel turn green and flash, while the speed indicators turn yellow.

If SBC1000 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 SBC1000 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 SBC1000 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 SBC1000 (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.

Old Password

New Password

Password Strength

Confirm

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 SBC1000 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 SBC1000 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:

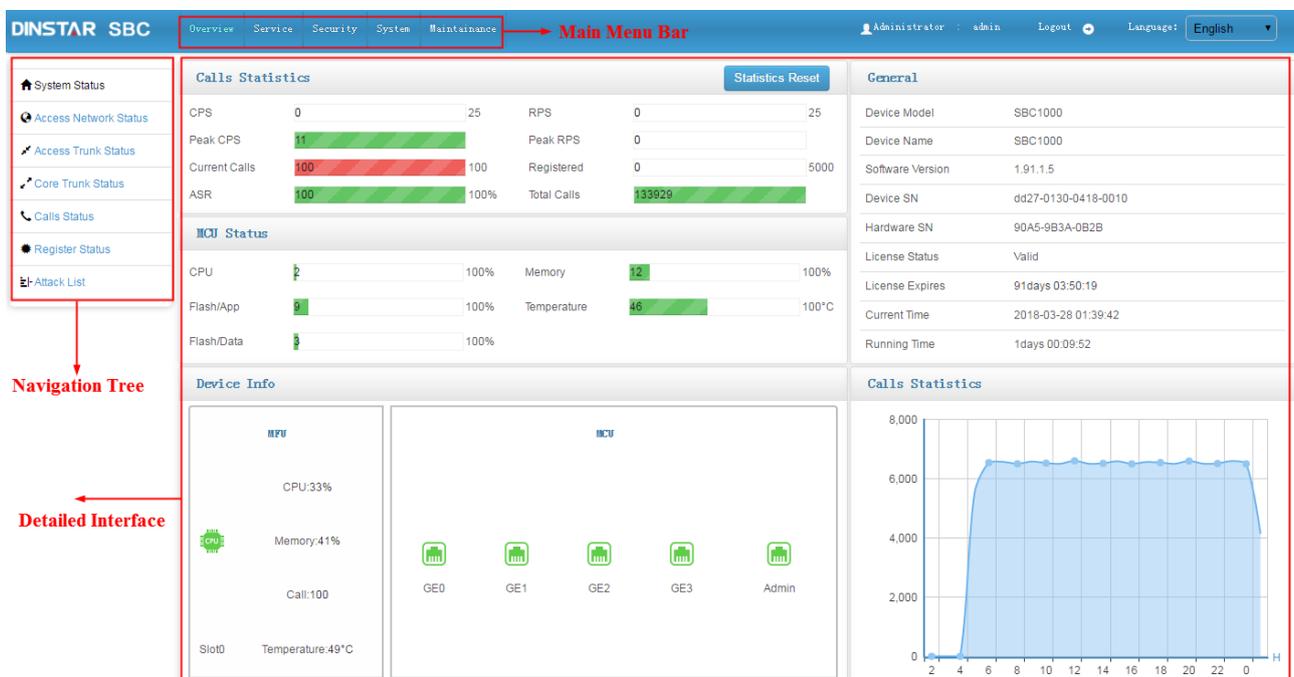


Figure 3-3 Structure of Web Interface

Table 3-1 Introduction to Web Interface

Index	Item	Description
1	Main Menu Bar	The main menu bar of SBC1000, 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		To add configurations
7		To edit/modify configurations
8		To delete configurations

### 3.3 Configuration Flows

The following is the general configuration flows of SBC1000:

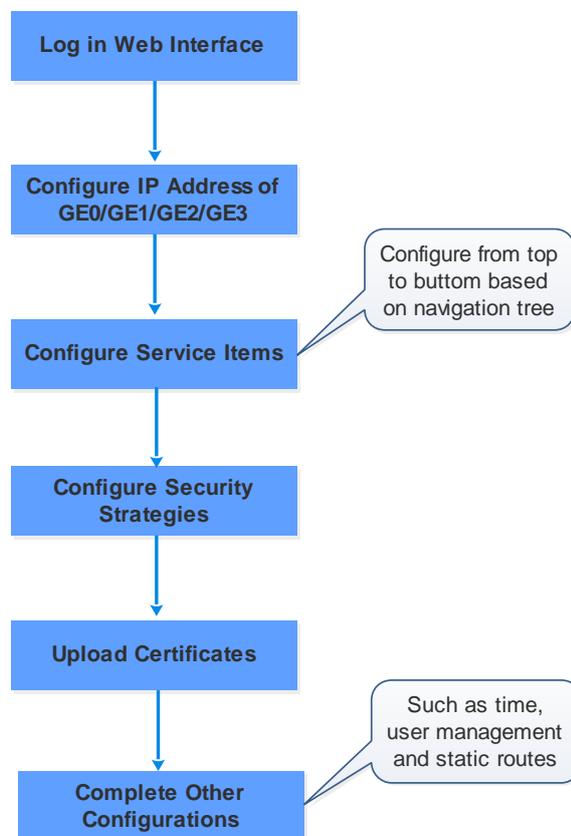


Figure 3-4 Configuration Flow

#### 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.

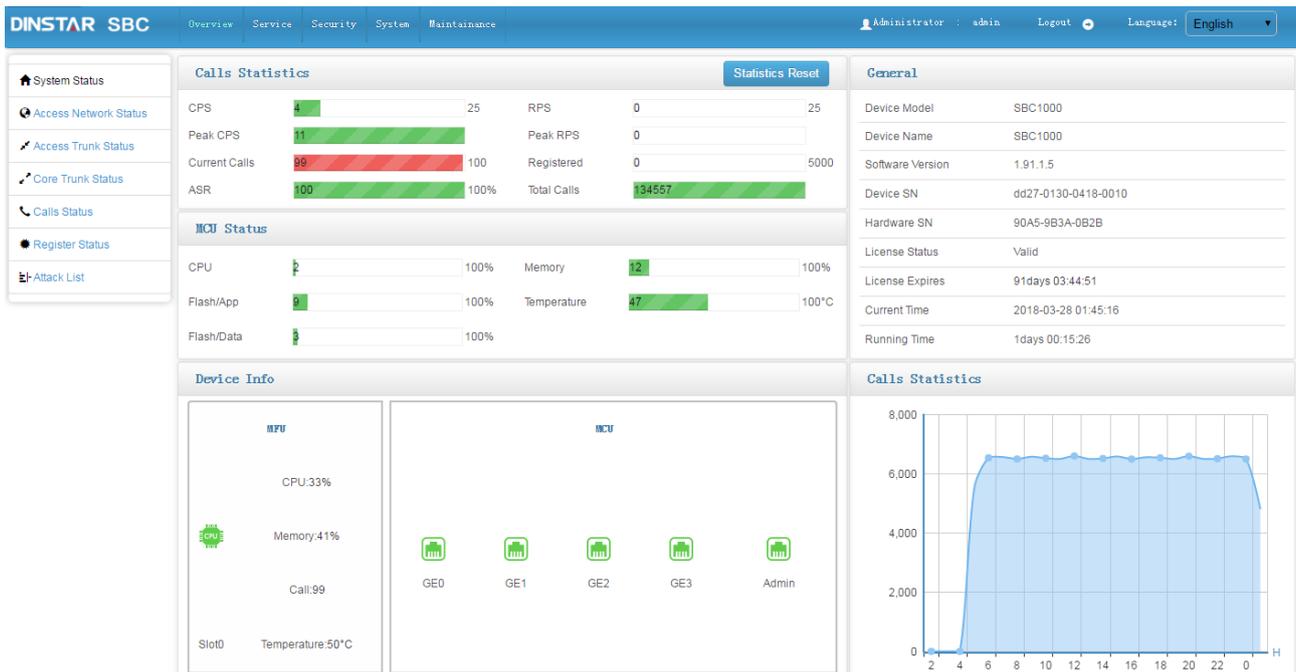


Figure 3-5 System Status

Table 3-2 Calls Statistics

CPS (Calls Per Second)	The number of new calls going through SBC1000 every second at current time
Peak CPS	The peak CPS (calls per second) since SBC1000 is booted up
Current Calls	The number of on-going calls at current time
Max. Calls	The maximum number of concurrent calls since SBC1000 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 SBC1000 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 SBC1000 is booted up
Total Calls	The total number of legal call requests since SBC1000 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	SBC1000
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 SBC1000, 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 SBC1000 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.

Access Network										+ Add		
Name	Description	Interface	Port	IPv4/IPv6	Transport	Codec	DTMF	MinRegister Interval	NAT Expire			
AccessNetwork1	AccessNetwork1	eth0	5060	ip4	udp	default	RFC2833	180	60			
AccessNetwork2	AccessNetwork2	eth1	5060	ip4	udp	default	RFC2833	180	60			

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 SBC1000, 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 SBC1000.

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’.

Access Trunk Status			search: <input type="text" value="Name"/> <input type="button" value="Commit"/> <input type="button" value="Refresh"/>										
Name	Status	CPS	Inbound Calls				Outbound Calls						
			ASR	Transcoded	Cur. Calls	Total Calls	Registerd	ASR	Transcoded	Cur. Calls		Total Calls	
AccessTrunk_Bob	false	0	0	0	0	0	0	0	0	0	0	0	
AccessTrunk_Tom	true	0	0	0	0	0	0	0	0	0	0	0	

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 SBC1000 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 SBC1000, 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 SBC1000.

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'.

Name	Status	CPS	Inbound Calls				Outbound Calls				
			ASR	Transcoded	Cur. Calls	Total Calls	Registerd	ASR	Transcoded	Cur. Calls	Total Calls
3cx	true	0	0	0	0	0	0	0	0	0	0

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 SBC1000 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 SBC1000, 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.

Calls Status																	Refresh
10	Search:	Caller(Source)	Callee(Destination)	Name(Source)	Name(Destination)	Commit											
Source								Destination									
Status	RTP Port	Duration(s)	Name	Caller	Callee	Codec	RTP	Peer IP	Name	Caller	Callee	Codec	RTP	Peer IP			
outgoing	33454	-	tg51	1705235 9348	4200208	PCMA	0/0	192.168.2.64:10828	tg28	1705235 9348	4200208		0/0	.0			
answer	33016	24	tg51	1705235 0486	4200227	PCMA	998/998	192.168.2.64:9996	tg28	1705235 0486	4200227	G729	998/998	172.30.60.124:12812			
answer	32768	24	tg51	1705235 7891	4200465	PCMA	998/998	192.168.2.64:9992	tg28	1705235 7891	4200465	G729	998/998	172.30.60.124:12808			
answer	33752	24	tg51	1705235 1419	4200955	PCMA	998/998	192.168.2.64:9990	tg28	1705235 1419	4200955	G729	998/998	172.30.60.124:12806			
answer	33350	24	tg51	1705235 8142	4200231	PCMA	998/998	192.168.2.64:9988	tg28	1705235 8142	4200231	G729	998/998	172.30.60.124:12804			
answer	32792	24	tg51	1705235 8672	4200172	PCMA	998/998	192.168.2.64:9986	tg28	1705235 8672	4200172	G729	998/997	172.30.60.124:12802			
answer	33632	24	tg51	1705235 4911	4200424	PCMA	998/998	192.168.2.64:9984	tg28	1705235 4911	4200424	G729	998/998	172.30.60.124:12800			
answer	32956	25	tg51	1705235 2527	4200762	PCMA	998/998	192.168.2.64:9854	tg28	1705235 2527	4200762	G729	998/998	172.30.60.124:12670			

Figure 3-9 Calls Status

Table 3-9 Call Status

Status	<p><b>Init:</b> an invite request for calling is received and the call is initiated;</p> <p><b>Outgoing:</b> the request for routing out the call is sent , and the system is waiting for response</p> <p><b>Early:</b> the 18x response is received</p> <p><b>Completed:</b> the 2xx response is received, and the system is waiting for the ack message</p> <p><b>Answer:</b> the ack message is received, and the call is set up</p>
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 SBC1000 are displayed.

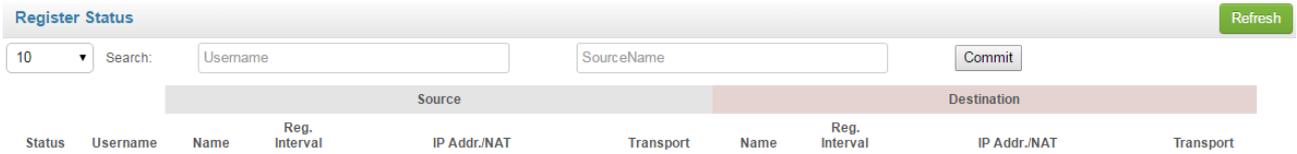


Figure 3-10 Register Status

Table 3-10 Register Status

Status	Registering: SBC1000 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 SBC1000 by terminal user Register Interval (destination): the interval of registering to core network’s SIP trunk by SBC1000
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 SBC1000 are shown.

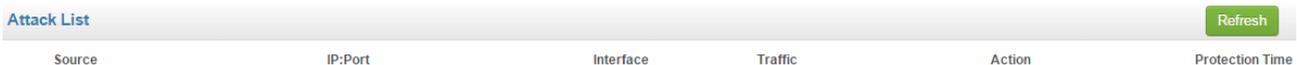


Figure 3-11 Attack List

Table 3-11 Attack List

Source	The source of an attack inflicted on SBC1000, for example, DDoS/DoS attacks
IP: Port	The IP address of the attack source, or the destination port that is attacked
Interface	The SBC1000 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 <b>Security</b> → <b>Security Policy</b> page, the action such as ‘Drop’ or ‘Flow Limited’ will be executed.
Action	<b>Log Record:</b> when the security policy is triggered and takes effect, the attack event is recorded in a log <b>Flow Limited:</b> 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

	<p>during the protection time.</p> <p><b>Packet Rate Limited:</b> 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.</p> <p><b>Drop:</b> 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.</p>
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 SBC1000 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.

Media Detection

Use callid to match sessions

RTP Detection

Interval  s

Start Media Port

**Note:** 1.The value for 'Start Media Port' should be an intergal multiple of 1K(K=1024).  
2.The configuration of 'Start Media Port' will not take effect until the SBC device is rebooted.

Save

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.

CDR

CDR Server

Commit

CDR Server List + Add

Name	Description	Interface	IP	Port	Transport	Format
cdr	cdr profile	172.16.250.250	1060	udp	json	

Figure 3-13 Configure CDR Server

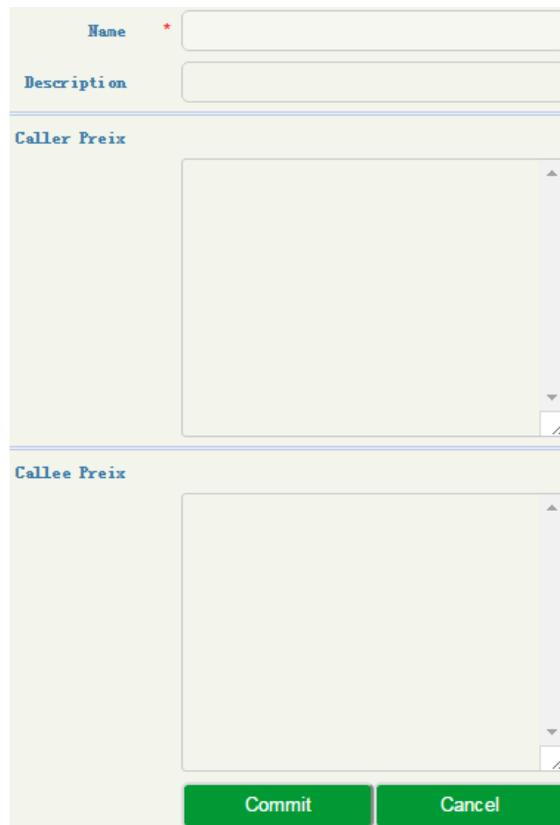
Table 3-12 CDR

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.



The screenshot shows a web form for adding a number profile. It has the following fields and sections:

- Name**: A text input field with a red asterisk indicating it is required.
- Description**: A text input field.
- Caller Preix**: A large text area for entering the caller prefix.
- Callee Preix**: A large text area for entering the callee prefix.
- Buttons**: 'Commit' and 'Cancel' buttons at the bottom.

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.

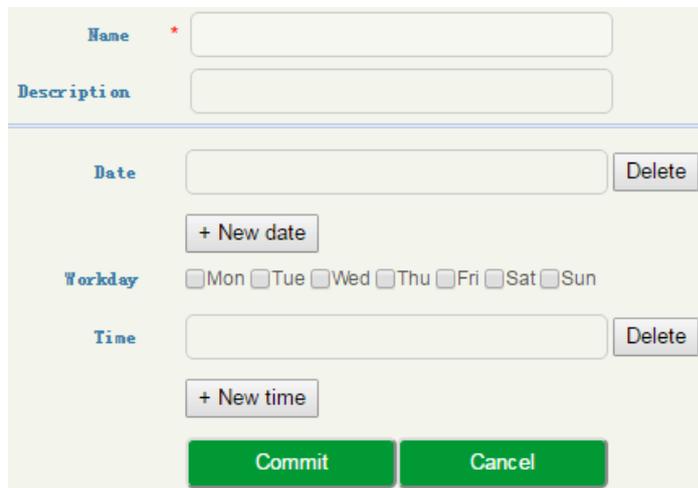


Figure 3-15 Add Time Profile

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.

The screenshot shows the 'Rate Limit' configuration page. At the top right, there is a '+ Add' button. Below it is a table with the following data:

Name	Description	RPS	CPS	Max. Concurrent Calls
default	default	250	200	3000

Below the table is a form to add a new rate limit rule. The form fields are:

- Name \*
- Description
- RPS \* 1
- CPS \* 1
- Max. Concurrent Calls \* 1

At the bottom of the form are two buttons: 'Commit' and 'Cancel'.

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 SBC1000 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 SBC1000 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

SBC1000 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.

Media Profiles <span style="float: right;">+ Add</span>			
Name	Description	Codec	Max. Packetization Time
default	default	PCMA , PCMU , G723 , G729	60

The screenshot shows the 'Edit Codec Profile' form with the following fields and values:

- Name:** (empty field)
- Description:** (empty field)
- Max. Packetization Time:** 60
- Codec:** A list containing PCMA, PCMU, G723, G729, iLBC\_13K, and iLBC\_15K. There are navigation arrows (>, <, ^, v) to the right of the list.
- Payload:** (empty field)
- Packetization Time:** (empty field)

At the bottom of the form are two green buttons: 'Commit' and 'Cancel'.

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	SBC1000 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.

Number Manipulation
+ Add

Name	Description	Caller Number	Callee Number
<div style="display: flex; justify-content: space-between;"> <div style="width: 45%;"> <p><b>Name</b> *</p> <input type="text"/> <p><b>Description</b></p> <input type="text"/> </div> <div style="width: 50%; border-left: 1px solid #ccc; padding-left: 10px;"> <p><b>Inbound Rule</b></p> <p><b>Delete Prefix</b></p> <input type="text"/> <p><b>Delete Suffix</b></p> <input type="text"/> <p><b>Add Prefix</b></p> <input type="text"/> <p><b>Add Suffix</b></p> <input type="text"/> <p><b>Condition</b></p> <input type="text" value="Regular Expression"/> <p><b>Replacement</b></p> <input type="text" value="Replacement"/> <hr/> <p><b>Outbound Rule</b></p> <p><b>Delete Prefix</b></p> <input type="text"/> <p><b>Delete Suffix</b></p> <input type="text"/> <p><b>Add Prefix</b></p> <input type="text"/> <p><b>Add Suffix</b></p> <input type="text"/> <p><b>Condition</b></p> <input type="text" value="Regular Expression"/> <p><b>Replacement</b></p> <input type="text" value="Replacement"/> </div> </div>			

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, SBC1000 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.

The screenshot shows a web interface for configuring a Number Pool. It consists of several sections:

- Name:** A text input field with a red asterisk indicating it is required.
- Description:** A text input field.
- Caller Number:** A section containing:
  - Prefix:** A text input field with a 'Delete' button to its right.
  - Start Number:** A text input field.
  - End Number:** A text input field.
  - A '+ Add' button below the fields.
- Callee Number:** A section containing:
  - Prefix:** A text input field with a 'Delete' button to its right.
  - Start Number:** A text input field.
  - End Number:** A text input field.
  - A '+ Add' button below the fields.
- Buttons:** At the bottom, there are two green buttons: 'Commit' and 'Cancel'.

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	<p><b>Prefix:</b> 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;</p> <p><b>Start Number:</b> The starting number of the number pool</p> <p><b>End Number:</b> The ending number of the number pool</p>

### 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.

The screenshot shows the 'SIP Header Manipulation' configuration table. It has the following structure:

Name	Description	SIP Header Type	Value Type	Routing Profiles
SunnyTest		request		rule001

At the top right of the table, there is a '+ Add' button. At the bottom right of the table, there are edit and delete icons.

The screenshot shows a web interface for configuring a SIP Header Manipulation Rule. At the top, there are three input fields: 'Name' with the value 'rule001', 'Description' with the value 'sunnytan changed into dinstar002', and 'Type' with a dropdown menu set to 'RequestLine'. Below these fields are two tables. The first table, titled 'Condition', has columns for 'Source ID', 'Match', and 'Value'. It contains one row with the source ID '\$from.\$displayname', the match type 'equal', and the value 'sunnytan'. The second table, titled 'Operation', has columns for 'Destination ID', 'Action', 'Value', 'Value Type', 'Match', and 'Rule'. It contains one row with the destination ID '\$request-line.\$uri', the action 'modify', the value 'dinstar002', the value type 'value', a match of '-', and a rule of '-'. At the bottom of the form are two green buttons labeled 'Save' and 'Cancel'.

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	<p>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.</p> <p><b>Name:</b> the name of the operation rule.</p> <p><b>Description:</b> the description of the operation rule.</p> <p><b>Type:</b> the content type where the operation rule will be applied.</p> <p>Request-line: the content of the request line of SIP message.</p> <p>Status-line: the content of the status line of SIP message.</p> <p>Header: the content of the header of SIP message.</p> <p><b>Condition:</b> the set condition for the operation rule. When the set value matches the source ID, the operation rule will be activated.</p> <p><b>Source ID:</b> the original content of SIP message, it can be any parameter included in SIP message.</p> <p><b>Match:</b> equal → when the source ID is equal to the set value, the operation rule is activate.</p>

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

### 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.

The screenshot shows a web interface for configuring SIP Header Passthrough profiles. At the top, there is a header bar with the text 'SIP Header Pass' on the left and a green '+ Add' button on the right. Below this, there is a table with three columns: 'Name', 'Description', and 'SIP Header'. The 'Name' column contains a text input field with a red asterisk next to it. The 'Description' column contains a text input field. The 'SIP Header' column contains a large text area. At the bottom of the form, there are two green buttons: 'Commit' and 'Cancel'.

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 SBC1000.

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 SBC1000 device.

<b>Name</b>	* nanshan
<b>Description</b>	
<b>Valid</b>	<input checked="" type="checkbox"/>
<b>Interface</b>	eth1
<b>Transport</b>	UDP
<b>Port</b>	* 5070
<b>IPv4/IPv6</b>	IPV4
<b>IP Range</b>	<input type="text"/> ~ <input type="text"/>
<b>Mask</b>	<input type="text"/>
<b>Signaling DSCP</b>	BE
<b>Media DSCP</b>	BE
<b>Header-end NAT</b>	
<b>Domain Filter</b>	
	<input type="button" value="+ Domain Filter"/>
<b>Rate Limit</b>	default

The image shows two panels of configuration options for an access network. The top panel includes dropdown menus for Codec (default), Blacklist, Whitelist, Inbound Manipulation, DTMF (RFC2833), RFC2833 (101), Inbound SIP Header Manipulation, and Outbound SIP Header Manipulation. The middle panel features input fields for SIP Session Timer (Require), Session Expire (1800s), Min. Session Timeout (360s), MinRegister Interval (180s), NAT Expire (60s), PRACK (Disable), and From Header (Local Domain). The bottom panel contains dropdowns for Peer Media Address (Unlock), Refresh Remote Media Address (Enable), Peer Signaling Address (Unlock), Caller From (User), and Callee From (User). It also has checkboxes for SIP Methods: OPTIONS, INFO, REFER, NOTIFY, SUBSCRIBE, and UPDATE. At the bottom are Save and Cancel buttons.

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 SBC1000
IPv4/IPv6	Select a network protocol for the access network. It can be IPv4 or IPv6. By default, the network protocol is IPv4

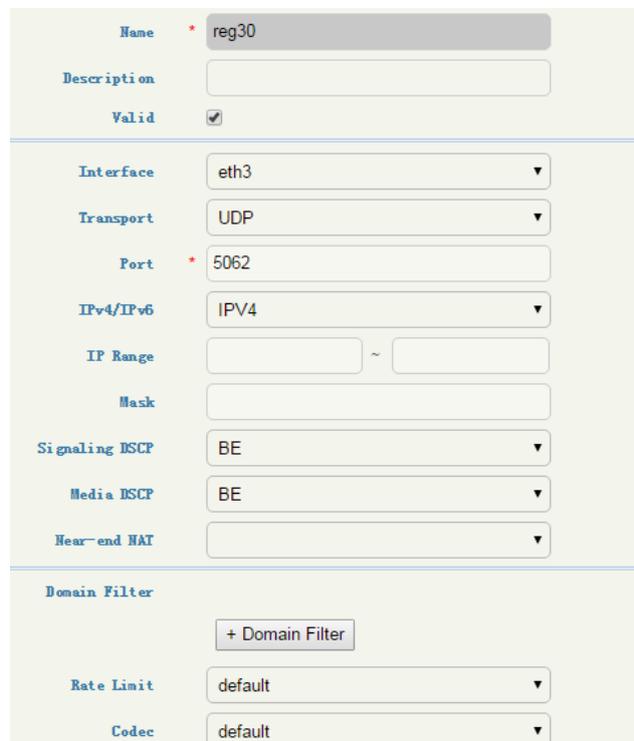
IP Range	Configure the range of legal IP addresses that send out SIP request can be received by the
Mask	The subnet mask of the IP range
Signaling DSCP	The QoS tag of SIP signaling messages
Media DSCP	The QoS tag of media messages
Near-end NAT	Near-end NAT defaults to disabled. If it is enabled, the contact IP address contained in SIP messages sent out by SBC1000 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 to 3.4.5
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; <a href="#">If the DTMF mode of an access network differs from that of core network, SBC1000 will convert it through DSP</a>
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, SBC1000 will send 'reinvite' messages to keep activating

	<p>sessions within the configured duration.</p> <p>If no messages are detected within the configured duration, sessions will be considered as 'ended', and then will be disconnected.</p> <p>If 'Require' is selected, the callee side of a call passing through the access network also needs to support session timer.</p>
Session Expire	Configure the duration of the session. During the duration, SBC1000 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, SBC1000 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 SBC1000 will automatically turned into the time configured here. The value of 'NAT Expire'
PRACK	<p>PRACK (Provisional Response ACKnowledgement): provide reliable provisional response messages.</p> <p>Disable: INVITE request and 1xx response sent out by SBC1000 will not include <i>100rel</i> tag by default;</p> <p>Support: INVITE request and 1xx response sent out by SBC1000 will include <i>100rel</i> tag in Supported header;</p> <p>Require: INVITE request and 1xx response sent out by SBC1000 will include <i>100rel</i> 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 <i>PRACK</i> request to acknowledge the response.</p>
From Header	<p>It can be 'Local Domain' or 'Peer Domain'.</p> <p>'Local Domain' is the default value.</p>
Peer Media Address	<p>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.</p> <p>Unlock: remote address sending media messages is not locked.</p>
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	<p>User: the USER field of FROM header of INVITE message is extracted as caller number</p> <p>Display: the DISPLAY field of FROM header of INVITE message is extracted as caller</p>

	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 SBC1000. On the **Service → Access SIP Trunk** page, you can configure the parameters of access SIP trunk.



The screenshot shows the configuration page for an Access SIP Trunk. The fields are as follows:

- Name:** reg30
- Description:** (empty)
- Valid:**
- Interface:** eth3
- Transport:** UDP
- Port:** 5062
- IPv4/IPv6:** IPV4
- IP Range:** (empty) ~ (empty)
- Mask:** (empty)
- Signaling DSCP:** BE
- Media DSCP:** BE
- Near-end NAT:** (empty)
- Domain Filter:** + Domain Filter
- Rate Limit:** default
- Codec:** default

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 SBC1000 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 SBC1000
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	<p>Near-end NAT defaults to disabled. If it is enabled, the contact IP address contained in SIP messages sent out by SBC1000 will be turned into the outbound IP address of public network.</p> <p>If NAT is enabled, you need to fill in the outbound IP address of public network.</p>
Rate Limit	The maximum RPS(registrations per second), CPS(calls per second) and total call volume of the access SIP trunk. Please refer to 3.4.5
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	<p>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</p> <p>If no black list and white list are selected for the access SIP trunk, all calls can be routed by the access SIP trunk.</p>
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	<p>DTMF is short for Dual Tone Multi Frequency;</p> <p>There are three DTMF modes, including SIP Info, Inband, RFC2833;</p> <p><a href="#">If the DTMF mode of an access SIP trunk differs from that of core network, SBC1000 will convert it through DSP</a></p>
Inbound SIP Header Manipulation	<p>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.</p> <p>Please refer to 3.4.10</p>
Outbound SIP Header Manipulation	<p>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.</p> <p>Please refer to 3.4.10</p>
Trunk Mode	<p><a href="#">When SBC is connected to IMS,</a></p> <p><b>Static:</b> you need to manually configure the IP address and port of the peer device, for example, 192.168.2.159:5060</p> <p>Remote domain name: the domain name of the peer</p> <p><b>Dynamic:</b> 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 'False'.</p>

Registration	<p>When 'Server IP Type' is configured as 'Static', registration will be displayed.</p> <p>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 'True'. Otherwise, the status is 'False'. For the status of access SIP trunk, please refer to 3.3.3 .</p>
Keepalive	<p>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.</p> <p>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 .</p>
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	<p>Session timer is a mechanism to keep activating sessions.</p> <p>If 'Supported' is selected, SBC1000 will send 'reinvite' messages to keep activating sessions within the configured duration.</p> <p>If no messages are detected within the configured duration, sessions will be considered as 'ended', and then will be disconnected.</p> <p>If 'Require' is selected, the callee side of a call passing through the access SIP trunk also needs to support session timer.</p>
Session Expires	Configure the duration of the session. During the duration, SBC1000 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	<p>PRACK (Provisional Response ACKnowledgement): provide reliable provisional response messages.</p> <p>Disable: INVITE request and 1xx response sent out by SBC1000 will not include <i>100rel</i> tag by default;</p> <p>Support: INVITE request and 1xx response sent out by SBC1000 will include <i>100rel</i> tag in Supported header;</p> <p>Require: INVITE request and 1xx response sent out by SBC1000 will include <i>100rel</i> 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 <i>PRACK</i> request to acknowledge the response.</p>
From Header	<p>It can be 'Local Domain' or 'Peer Domain'.</p> <p>'Local Domain' is the default value.</p>
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 SBC1000 and the core network. On the **Service → Core SIP Trunk** page, you can configure the parameters of core SIP trunk.

Name *	<input type="text"/>
Description	<input type="text"/>
Valid	<input checked="" type="checkbox"/>
Interface	<input type="text" value="eth0"/>
Transport	<input type="text" value="UDP"/>
Port *	<input type="text" value="5060"/>
IPv4/IPv6	<input type="text" value="IPV4"/>
Signaling DSCP	<input type="text" value="BE"/>
Media DSCP	<input type="text" value="BE"/>
Near-end NAT	<input type="text"/>
Codec	<input type="text" value="default"/>
Inbound Manipulation	<input type="text"/>
DTMF	<input type="text" value="RFC2833"/>
RFC2833 *	<input type="text" value="101"/>
Inbound SIP Header Manipulation	<input type="text"/>
Outbound SIP Header Manipulation	<input type="text"/>
Trunk Mode	<input type="text" value="Static"/>
Remote IP :Port *	<input type="text"/>
Remote Server domain	<input type="text"/>
Access Visit ACL table	<input type="text"/>
	<input type="button" value="+ Add"/>
Registration	<input type="checkbox"/>
OutBound Proxy	<input type="checkbox"/>
Keepalive	<input type="checkbox"/>
SIP Session Timer	<input type="text" value="Disable"/>
PRACK	<input type="text" value="Disable"/>
From Header	<input type="text" value="Local Domain"/>
Peer Media Address	<input type="text" value="Lock"/>
Refresh Remote Media Address	<input type="text" value="Enable"/>
Peer Signaling Address	<input type="text" value="Unlock"/>
Caller From	<input type="text" value="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 SBC1000 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 SBC1000
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 SBC1000 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 to 3.4.5
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;

	<p>There are three DTMF modes, including SIP Info, Inband, RFC2833;</p> <p>If the DTMF mode of an core SIP trunk differs from that of access network, SBC1000 will convert it through DSP</p>
Inbound SIP Manipulation	<p>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.</p> <p>Please refer to 3.4.10</p>
Outbound SIP Manipulation	<p>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.</p> <p>Please refer to 3.4.10</p>
Server IP Type	<p><b>When SBC is connected to IMS,</b></p> <p><b>Static:</b> you need to manually configure the IP address and port of the peer device, for example, 192.168.2.159:5060</p> <p>Remote domain name: the domain name of the peer</p> <p><b>Dynamic:</b> 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 'False'.</p>
Registration	<p>When 'Server IP Type' is configured as 'Static', registration will be displayed.</p> <p>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 'True'. Otherwise, the status is 'False'. For the status of core SIP trunk, please refer to 3.3.4 .</p>
Keepalive	<p>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.</p> <p>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 .</p>
Times of No response	<p>The maximum number of timeouts for receiving response from the core network server after option messages are sent out.</p>
Interval	<p>The interval to send option message to the core network server</p>
SIP Session Timer	<p>Session timer is a mechanism to keep activating sessions.</p> <p>If 'Supported' is selected, SBC1000 will send 'reinvite' messages to keep activating sessions within the configured duration.</p> <p>If no messages are detected within the configured duration, sessions will be considered</p>

	<p>as 'ended', and then will be disconnected.</p> <p>If 'Require' is selected, the callee side of a call passing through the core SIP trunk also needs to support session timer.</p>
Session Expires	Configure the duration of the session. During the duration, SBC1000 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	<p>PRACK (Provisional Response ACKnowledgement): provide reliable provisional response messages.</p> <p>Disable: INVITE request and 1xx response sent out by SBC1000 will not include <i>100rel</i> tag by default;</p> <p>Support: INVITE request and 1xx response sent out by SBC1000 will include <i>100rel</i> tag in Supported header;</p> <p>Require: INVITE request and 1xx response sent out by SBC1000 will include <i>100rel</i> 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 <i>PRACK</i> request to acknowledge the response.</p>
From Header	<p>It can be 'Local Domain' or 'Peer Domain'.</p> <p>'Local Domain' is the default value.</p>
Remote media send addresses	<p>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.</p> <p>Unlock: remote address sending media messages is not locked.</p>
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	<p>User: the USER field of FROM header of INVITE message is extracted as caller number</p> <p>Display: the DISPLAY field of FROM header of INVITE message is extracted as caller number</p>
Callee Number Field	<p>User: the USER field of TO header of INVITE message is extracted as callee number;</p> <p>Display: the DISPLAY field of TO header of INVITE message is extracted as callee number;</p> <p>Request-uri: the USER NUMBER in REQUEST-URI of INVITE message is extracted</p>

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

### 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.

The screenshot shows a configuration form for a SIP Trunk Group. The fields are:
 

- Name**: A text input field with a red asterisk indicating it is required.
- Description**: A text input field.
- Type**: A dropdown menu currently set to "Access SIP Trunk Group".
- Routing Mode**: A dropdown menu currently set to "Backup".
- SIP Trunk Name**: A dropdown menu currently set to "ag244", with a "Delete" button next to it.

 At the bottom of the form are two green buttons: "Commit" and "Cancel".

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	<p>The strategy for choosing which truck will be used under a trunk group when a call comes in.</p> <p><b>Backup</b>: 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.</p>

	<b>Load Balance:</b> 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

The screenshot shows a configuration form for Call Routing. The fields are as follows:

- Index:** 120
- Description:** (empty text box)
- Condition:**
  - Num Profiles:** (dropdown menu)
  - Caller Username:** (text box)
  - Callee Username:** (text box)
  - Time:** (dropdown menu)
  - Caller SIP URL:** (text box)
  - Callee SIP URL:** (text box)
  - Source Type:** Access Network (dropdown menu)
  - SIP Methods:** (text box)
- Type:** Core SIP Trunk (dropdown menu)
- Destination:** ag60 (dropdown menu)
- Outbound Manipulation:** (dropdown menu)
- SIP Header Pass:** (dropdown menu)

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.
Time 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 SBC1000 from being attacked by various DOS/DDOS floods, so as to ensure stable running of the device.

System		
Attack Log	<input type="checkbox"/>	
ICMP-Flood	<input checked="" type="checkbox"/>	Peak PPS(Packet Per Second) 50
Ping Response	<input checked="" type="checkbox"/>	
UDP-Flood	<input checked="" type="checkbox"/>	Peak PPS(Packet Per Second) 200
TCP-NULL	<input checked="" type="checkbox"/>	
TCP-Flood	<input checked="" type="checkbox"/>	Peak PPS(Packet Per Second) 50
TCP XMAS TREE	<input checked="" type="checkbox"/>	

[Save](#)

Figure 3-29 System Security

Table 3-27 System Security

Attack Log	If 'Attack Log' is enabled and SBC1000 is attacked, the device will record the attack in logs which can be viewed on the <b>Maintenance →Log →Security Log</b> page.
ICMP-Flood	ICMP-Flood is a kind of DDOS attack. It can send a mass of ICMP packets to attack the SBC1000 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 SBC1000 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 SBC1000 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, SBC1000 will drop TCP packages, and the peer device cannot learn whether the ports of SBC1000 are closed or not.
TCP-Flood	<p>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.</p> <p>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.</p>
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, SBC1000 will drop those TCP packages, and the peer device cannot learn which ports of SBC1000 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.

**Access Control**

**Web Server**

---

HTTPS Port

---

HTTP Port

---

Allowed to access eth0

---

Allowed to access eth1

---

Allowed to access eth2

---

Allowed to access eth3

---

**SSH**

---

Port

---

Allowed to access eth0

---

Allowed to access eth1

---

Allowed to access eth2

---

Allowed to access eth3

---

Save

Figure 3-30 Access Control

Table 3-28 Access Control

Web Server	<p>The Web interface of SBC1000 only supports https, and the https port defaults to 443. You can modify the https port;</p> <p>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 SBC1000.</p> <p>By default, GE0, GE1, GE2 and GE3 are not allowed to access the Web interface.</p>
SSH	<p>The SSH port of SBC1000 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 SBC1000.</p> <p>By default, GE0, GE1, GE2 and GE3 are not allowed to access the SSH.</p>

### 3.5.3 Security Policy

#### 1. IP Security Strategy

**Protection Time**

Protection Time:  min

**IP Security**

Priority	Name	Attacked	CPU Usage	Traffic	Action
127	default_ip	Remote IP	-	2048 KBPS	Log Record
128	default_port	Local Port	-	200 KBPS	Log Record

Figure 3-31 IP Security Strategy

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

**Priority** \*

**Name** \*

**Attacked**

**CPU Usage**

**Traffic** \*  KBPS

**Action**

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, SBC1000 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, SBC1000 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, SBC1000 will execute the configured action on the packets.

Action	<p>Log Record: when the security strategy is triggered and takes effect, the attack event is recorded in a log</p> <p>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.</p> <p>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.</p> <p>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.</p>
--------	--

## 2. SIP Security

**Interval**

Registration Interval  s

Call Detetion Interval  s

**SIP Security**

Priority	Description	Attacked	Detected	Action	Protection Time	
124	detect register counts per ip	IP Anti Attacking	Number Of Registrations/30	Log Record	-	<input type="button" value="edit"/> <input type="button" value="delete"/>
125	detect call counts per ip	IP Anti Attacking	Number Of Calls/10	Log Record	-	<input type="button" value="edit"/> <input type="button" value="delete"/>
126	detect register counts per user	User Attack	Number Of Registrations/5	Log Record	-	<input type="button" value="edit"/> <input type="button" value="delete"/>

Figure 3-33 SIP Security Strategy

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

Priority *	124
Description	detect register counts per ip
Attacked	IP Anti Attacking ▼
Detected	Number Of Registrations ▼
*	30
Action	Log Record ▼
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

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 SBC1000 device.

<b>System Management</b>	
Device Name	SBC1000
<input type="button" value="Save"/>	

Figure 3-35 Device Name

### 3.6.2 Web Configuration

<b>Web Configuration</b>	
Certification	default ▼
Key	default ▼
<input type="button" value="Save"/>	

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 <span style="float: right;">+ Add</span>							
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

**Name** \* eth90

**Mac** \* f8:a0:3f:48:50:2a

**MTU** \*

**Priority** \* 100

---

**Network Mode** Static ▼

**IP** \* 192.168.11.1

**Mask** \* 255.255.255.0

**Gateway**

**DNS Server**

Figure 3-38 Modify Port Information

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

The screenshot shows a web form for adding a VLAN. The fields are as follows:

- Vlan ID**: \* [ ]
- Interface**: \* eth0 (dropdown)
- MTU**: \* 1500
- Priority**: \* 512
- Network Mode**: Static (dropdown)
- IP**: \*
- Mask**: \*
- Gateway**: [ ]
- DNS Server**: [ ]

Buttons: **Commit** (green), **Cancel** (green)

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 SBC1000 visits an IP address of other network segment and this peer IP address is not directed by static route, SBC1000 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, SBC1000 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), SBC1000 will reject the connection request from the wide-area network (WAN). Port mapping allows a client in the wide-area network to visit the SBC1000 device in the local-area network.

Port Mapping							+ Add
Name	Status	Local Interface	Local Port No.	Transport	Remote Interface	Remote IP	Remote Port No.

The screenshot shows a configuration form for port mapping. The fields are as follows:

- Name**: \* (empty text input)
- Status**: Valid (dropdown menu)
- Local Interface**: eth1 (dropdown menu)
- Local Port No.**: \* (empty text input)
- Transport**: TCP (dropdown menu)
- Remote Interface**: eth1 (dropdown menu)
- Remote IP**: \* (empty text input)
- Remote Port No.**: \* (empty text input)

At the bottom of the form are two green buttons: **Commit** and **Cancel**.

Figure 3-40 Configure Port Mapping

Table 3-31 Port Mapping

Name	The name of this port mapping
Status	To enable or disable
External host interface	The interface of the client in the wide-area network, which is to visit the SBC1000 device in local-area network, such as GE0, GE1 , GE2 and GE3
External host port number	The port of the client in the wide-area network, which is to visit the SBC1000 device in local-area network, such as 5060
Transport Protocol	Choose TCP, UDP or TCP\UDP
Internal host interface	The mapped interface of the SBC1000 device in local-area network
Internal host IP address	The mapped IP address of the SBC1000 device in local-area network
Internal host port number	The mapped port of the SBC1000 device in local-area network (this port cannot conflict with the in-use port of the SBC1000 device )

### 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

**+ Add**

to enter into the setting page of static route.

The screenshot shows a configuration form for adding a static route. The fields are as follows:

- Priority**: 127
- Description**: (empty)
- Destination IP/Domain**: (empty)
- Mask**: (empty)
- Interface**: eth0
- Next Hop**: (empty)

At the bottom of the form are two buttons: **Commit** and **Cancel**.

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 SBC1000 device. Factory defaults for administrator's username and password are 'admin' and 'admin@123#' which are also used to log in SSH.

#### Password

The screenshot shows the 'Modify Password' configuration form. The fields are as follows:

- Old Password**: (empty)
- New Password**: (empty)
- Password Strength**: (empty)
- Confirm**: (empty)

At the bottom of the form is a **Commit** button.

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.

The screenshot shows a web form for adding a user. The fields are as follows:

- Username**: \* lich
- Password**: \* (empty field with an eye icon for visibility toggle)
- Password Strength**: (empty field)
- Confirm**: \* (empty field with an eye icon for visibility toggle)
- Role**: \* Admin (dropdown menu)
- Permission**: A list of system components with checkboxes for View and Edit permissions.
  - Overview:  View
  - System Status:  View
  - Access Network Status:  View
  - Access Trunk Status:  View
  - Core Trunk Status:  View
  - Calls Status:  View
  - Register Status:  View
  - Attack List:  View
  - Service:  View  Edit
  - Media Detection:  View  Edit
  - CDR:  View  Edit
  - Number Profile:  View  Edit
  - Time Profile:  View  Edit
  - Rate Limit:  View  Edit

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 SBC1000 device
Password	The password for the user to log in the SBC1000 device
Confirm	Confirm the password
Password Strength	The security strength of the password
Role	<p>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 SBC1000 device.</p> <p>Operator: has the permission to view configurations, or modify part of the configurations.</p> <p>Observer: has the permission to view existing configurations, but cannot delete or modify them.</p>

### 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 SBC1000 device according to the location where the device is placed.
Synchronize Time	If the current time of SBC1000 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 SBC1000.
NTP Server	If NTP server is enabled, the time of SBC1000 will be synchronize to that of NTP server.

### 3.6.8 Upgrade

On the **System** → **Upgrade** interface, you can upgrade the SBC1000 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 SBC1000 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 SBC1000 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 <b>Backup</b>
Restore	Choose a backup file, and then click <b>Restore</b> .
Factory Settings	Click <b>Factory Settings</b> , and the configurations of the SBC1000 device will become factory settings.

### 3.6.10 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 SBC1000 device will not accept registrations and calls after the license expires.

License	
Device SN	
License Type	
License Begin Time	
License Total Time	No License
License Expires	
Max. Media Sessions	
Max. Transcoding Sessions	
Max. Registered Users	
RPS	

Please input your license

Commit
Clear

Figure 3-48 License Information

### 3.6.11 Certificate

On the **System** → **Certificate** page, you need to upload a certificate to ensure the secure login to the Web interface of the SBC1000 device. You cannot log in the device until you has uploaded a certificate.

**Name** \*

---

**CRT File** \*  Browse...

**KEY File** \*  Browse...

Commit
Cancel

Figure 3-49 Upload Certificate

## 3.7 Maintenance

### 3.7.1 Login Log

The logs tracing the logins of the SBC1000 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.

Security Log								
10	Search: Begin Time	End Time	Type	Source	IP	Interface	Port	Commit
Index	Time	Attacked	Source	IP	Interface	Port	Condition	Action

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)