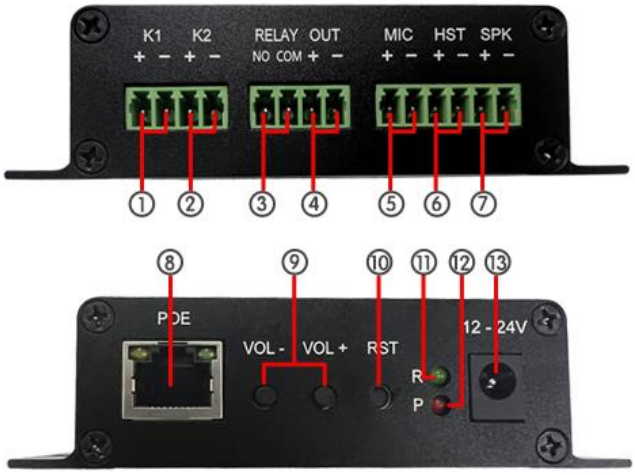
SIP Paging Adapter
SIP-T20
User Manual

Content

1. Overview

SIP-T20 is an IP based paging adapter that can covert analog to SIP. It's small and portable design with black outlook. The various interfaces (MIC, headset and speaker) make it possible for quickly configure intercom and paging solution. It's compatible with SIP & ONVIF protocol that can be used in VoIP and security field. It supports two-way intercom communication. Flexible Alarm in and out solutions (GPIO, HTTP URL, and relay out) are widely applied in daily life. The 48K OPUS Audio Codec enables excellent sound quality to make announcement, play background music, security alarm in school, factory and hospital, etc.

2. Interface Description



① K1 DSS Key	Connect to: 1. external keys 2. infrared probe and emergency switch 3. door sensor and other switch components
② K2 DSS Key	Connect to: 1. external keys 2. infrared probe and emergency switch 3. door sensor and other switch components
③ Relay NO COM Interface	Control the alarm system on/off
④ IO Output Interface	Responding to: 1. control the external amplifier power switch 2.the short-circuit input interface 3. login device security page settings 4. control the alarm light, electric locks and other equipment 5. with the adjacent power port connection for external equipment
⑤Microphone Interface	2.2K Ohm impedance electric condenser microphone is recommended.
⑥ Headset Interface	Speaker audio line signal output impedance 32 Ohm, single ended output voltage 1.2V, used for external headphones or amplifier.
⑦ Speaker Interface	Maximum support 15W speaker.
⑧ Ethernet Interface	WAN port, standard RJ45 interface, 10/ 100M

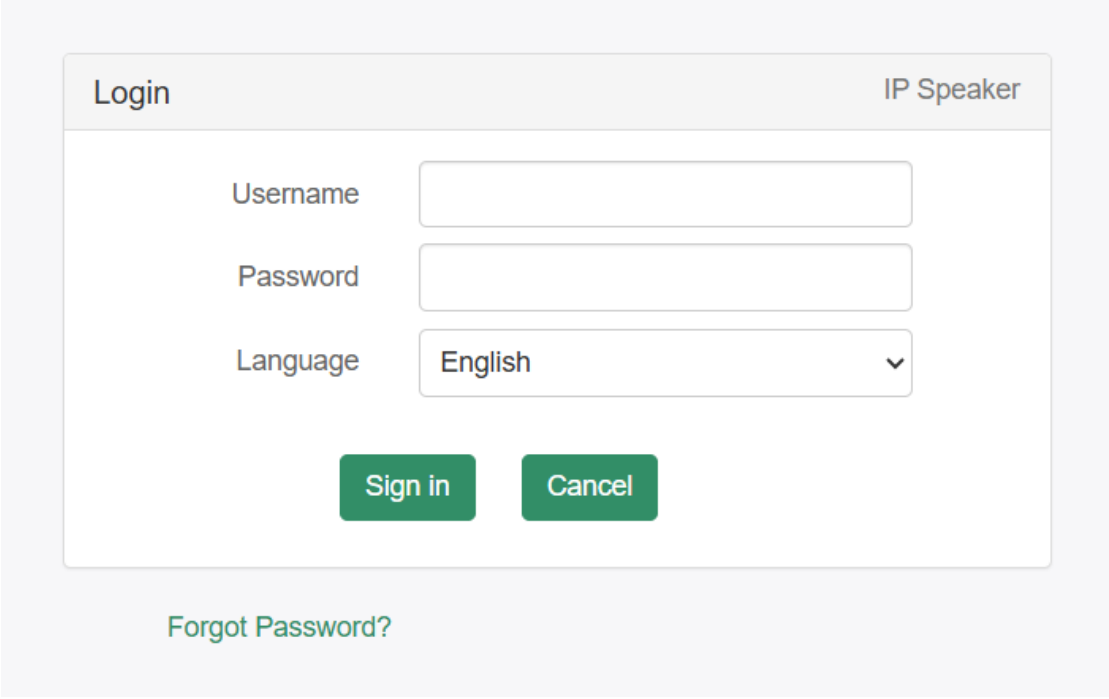
	adaptive, support POE input.
⑨ Volume Control Key	These two keys are to adjust the volume of the device's, bell, phone call and broadcasting, etc.
⑩ System Reset Key	Press Rest key and hold for 3 seconds, the devices will restart to factory setting.
⑪ Run Indicator	The light is on shows that the device is working well.
⑫ Power Indicator	The light on shows that the power is connected.
⑬ Power Input Interface	12V ~ 24V 2A input, according to the input voltage to determine the maximum output power amplifier.

3. Web Configuration

Web configuration includes complete function setting. When the device and your computer are connected to a same network, please open a browser and type in http://192.168.5.200, then log in with defaulted username and password as below.

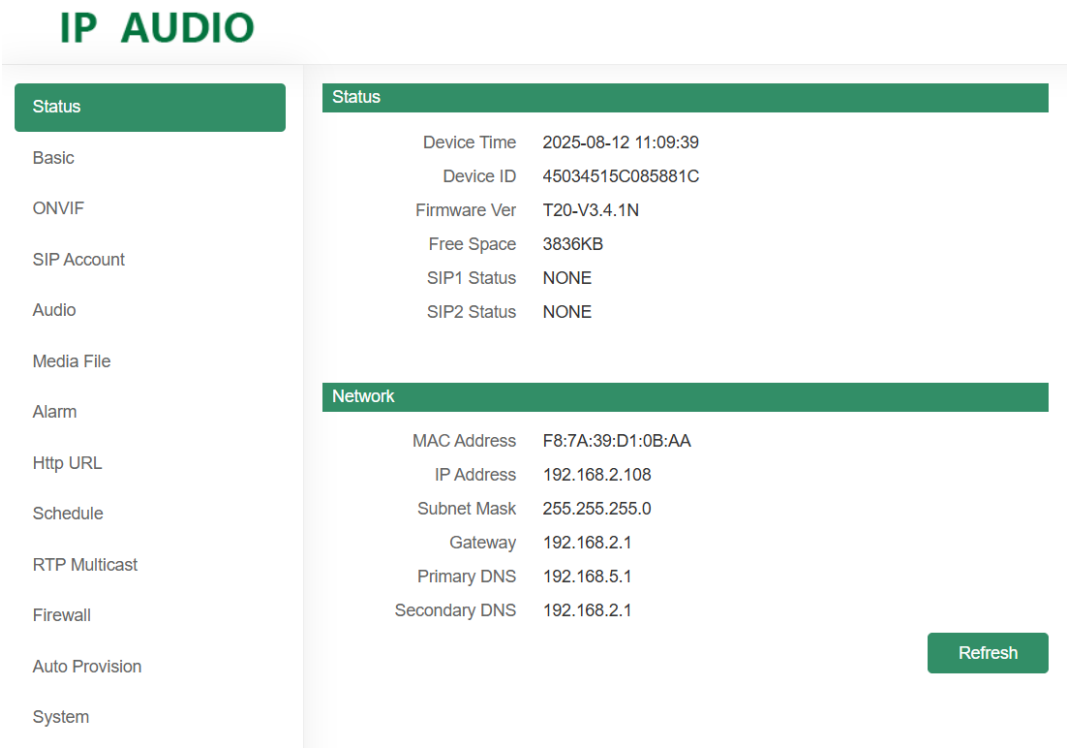
Username: admin

Password: tm1234

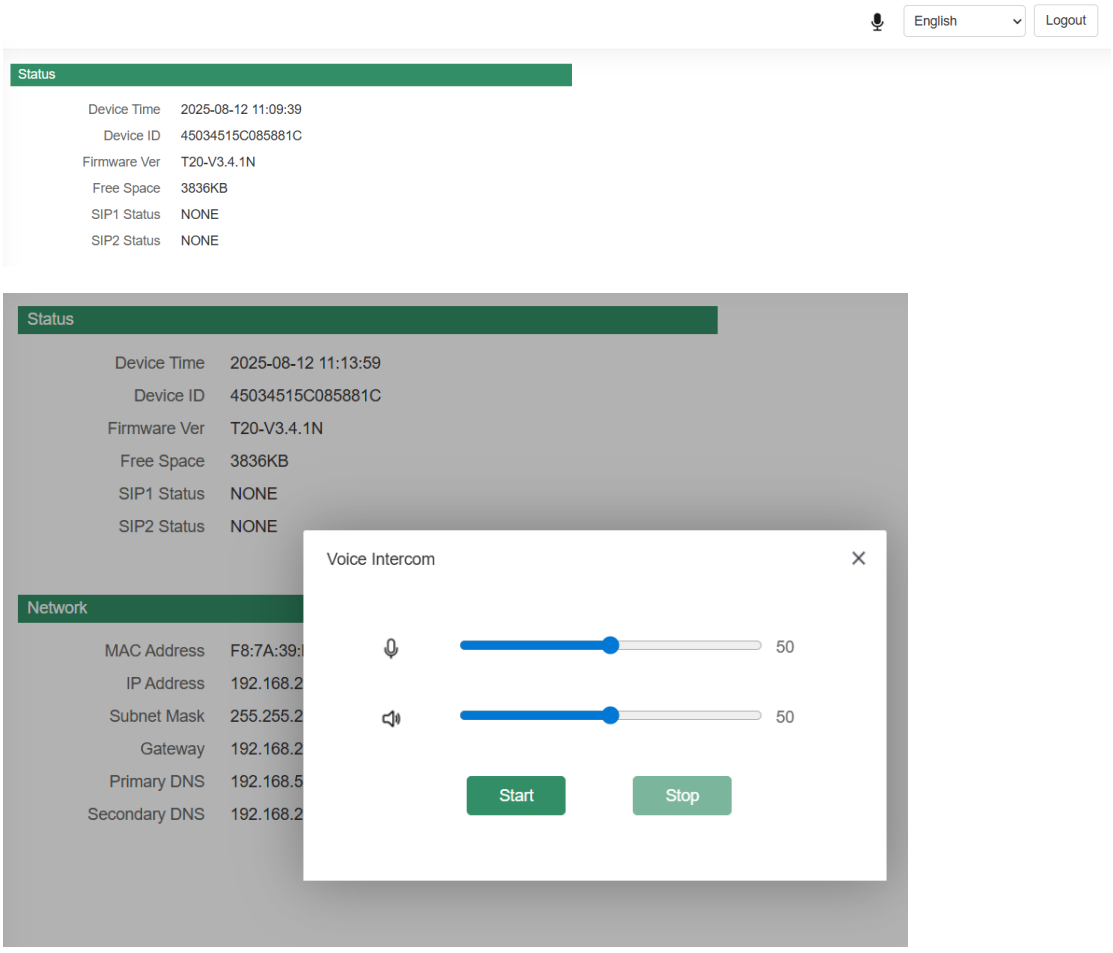


3.1 Status

You can check out firmware version, free space and two SIP accounts status of SIP-T20, also can locate the current network information here, like MAC, IP address and gateway etc.



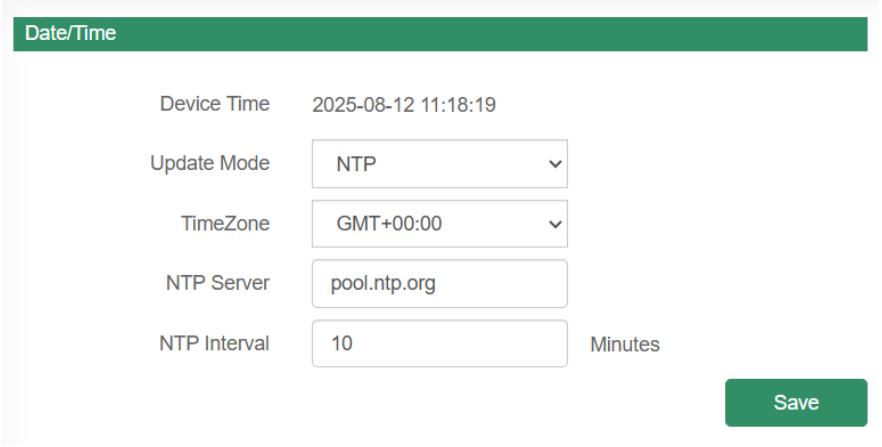
Web Calling Support: Click the microphone logo in the top right corner. You will be redirected to an HTTPS page. Click again on the new page to initiate the call.

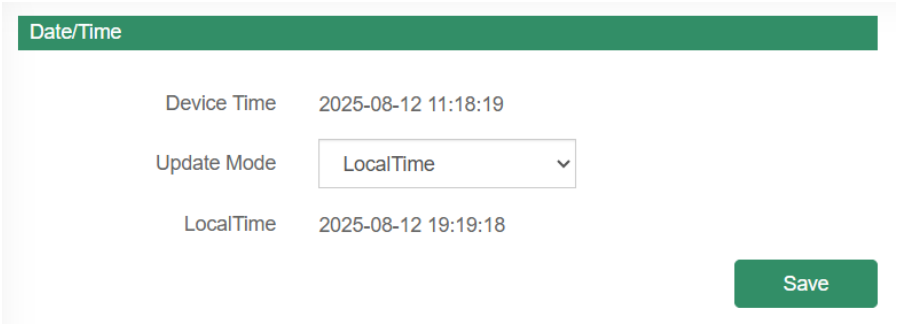


3.2 Basic

3.2.1 Date/ Time

There are two update modes for time： NTP/ local time, choose one and set the time zones, NTP sever and interval can choose default setting, then save the configuration.

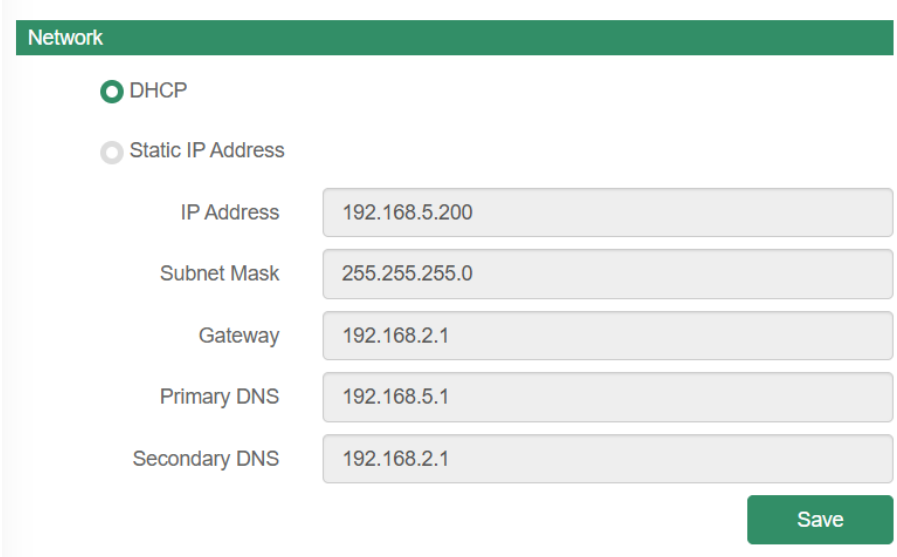




3.2.2 Network

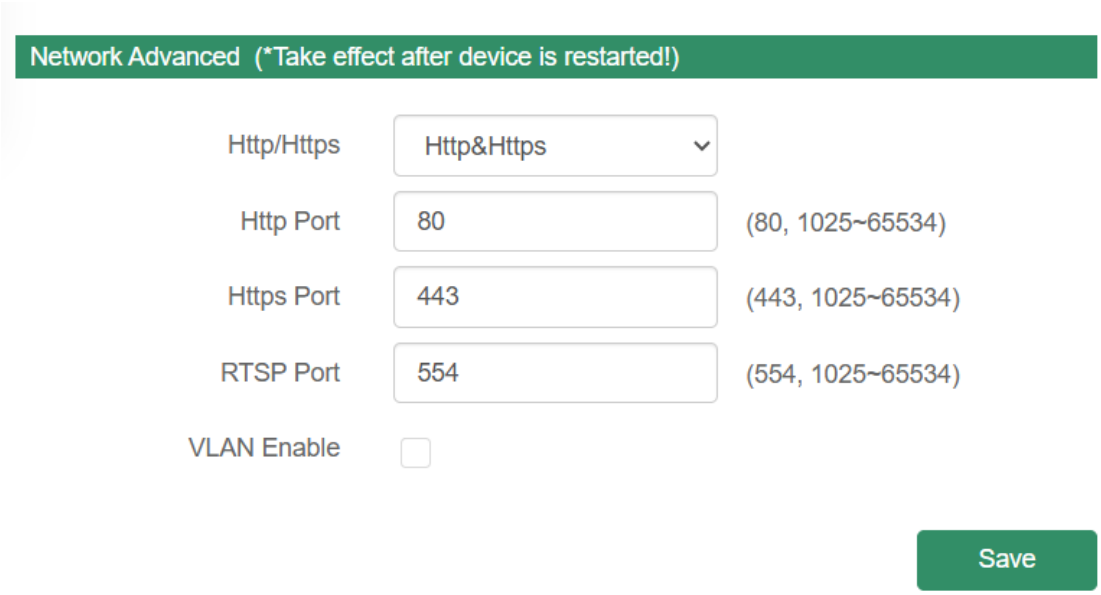
When you choose DHCP and save it, IP address will be created automatically by a DHCP server, then you need to login again with the new IP address on browser: 192.168.X.XXX.

Status IP address: It is a default IP and will not be changed as following.



3.2.3 Network Advanced

Allows manual configuration of HTTP and HTTPS port settings.

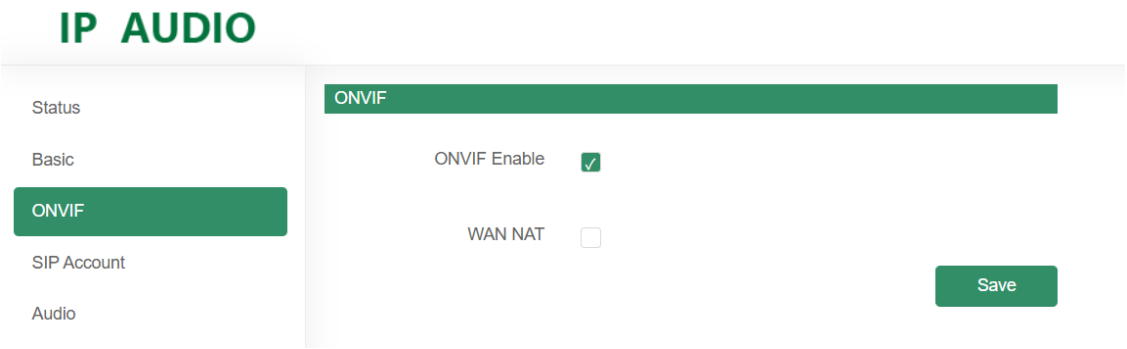


3.3 ONVIF

Select Enable ONVIF, then the device be searched by ONVIF VMS.

ONVIF credentials match the device credentials; changing the device credentials will also update the ONVIF credentials.

Default user name: admin, password:tm1234.

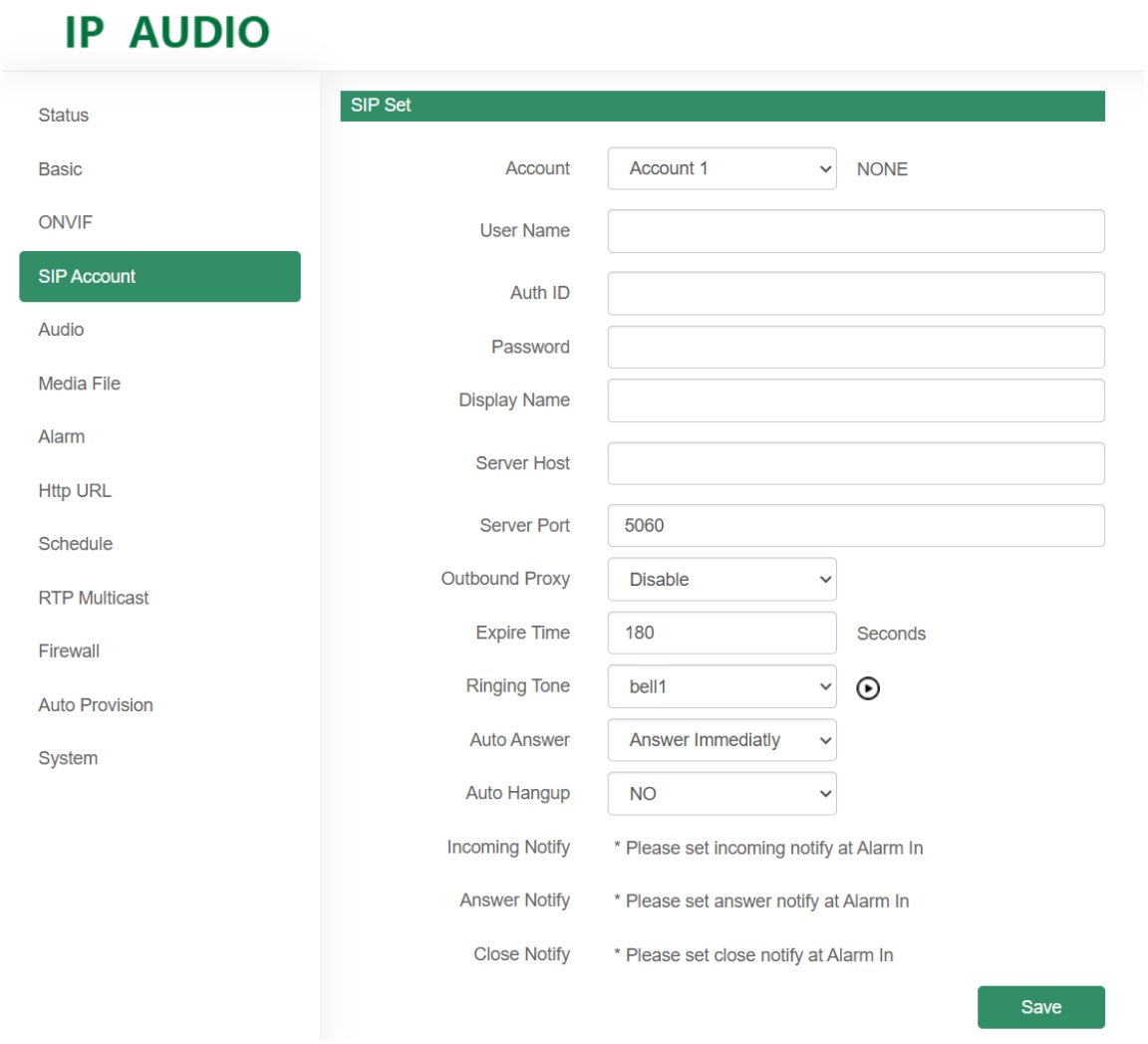


3.4 SIP Account

3.4.1 SIP Set

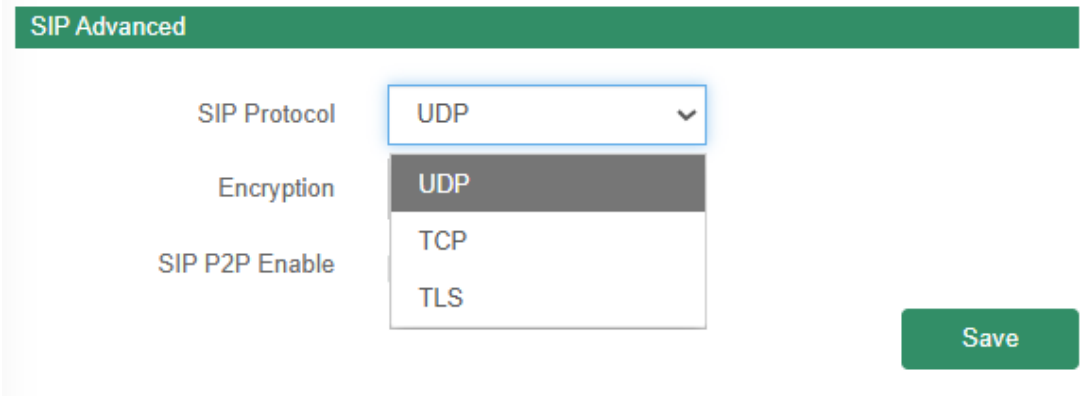
Each device has two SIP accounts, put SIP extension messages into the blanks and save the configuration, then you can check if it registers successfully or not on status.

Expire time	Set the expire time of registered account information
Ringing tone	5 system ringtones and 10 users upload media files
Auto Answer	answer immediately and answer delay when a calling income

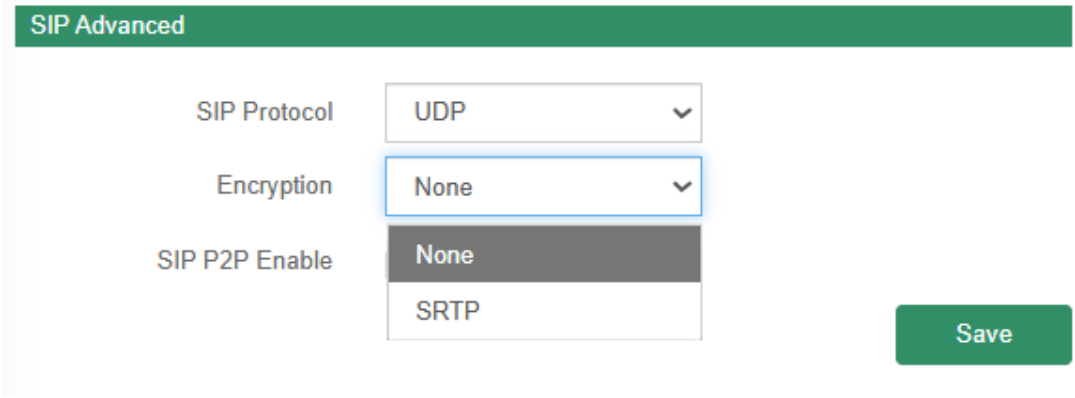


3.4.2 SIP Advanced

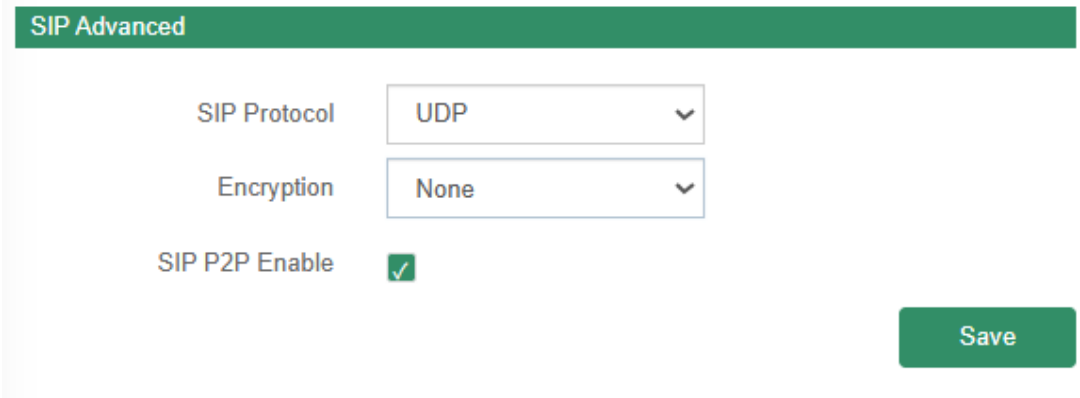
SIP Protocol: Select from UDP, TCP, or TLS.



Encryption: Choose either None or SRTP.

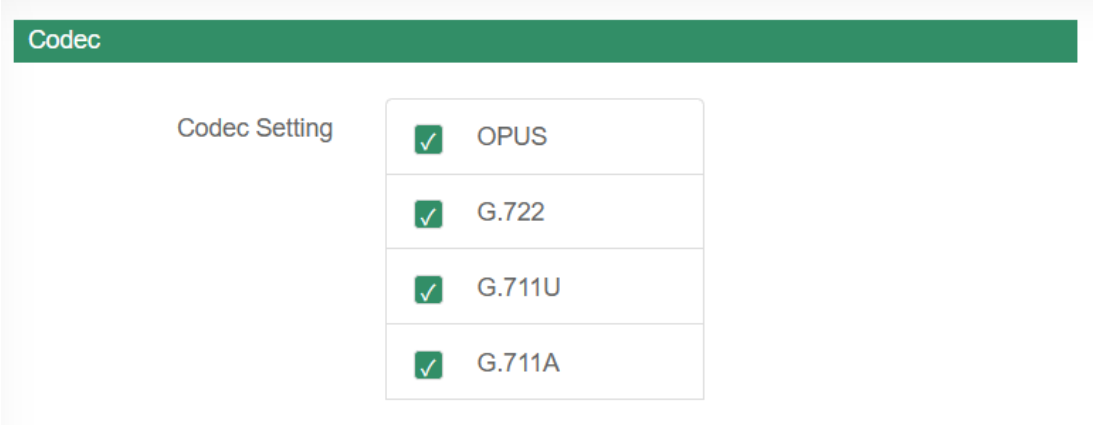


SIP P2P Enable: Check to activate SIP peer-to-peer functionality



3.5 Audio

Codec: Four audio codes to compatible with major audio sources.



Speaker:

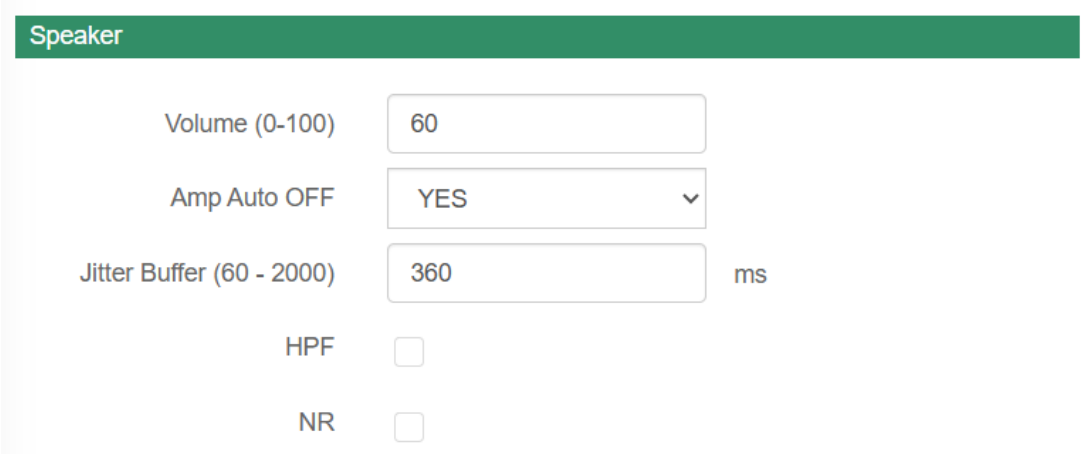
Volume: Adjust output volume at 0-100.

Amp auto off: It's set defaulted as ON, then there is no noise when not broadcasting.

Jitter buffer: To make the audio more stable.

HPF: High Pass Filter.

NR: Noise Reduction.



MIC:

Gain: Options: None, Low, Middle, High.

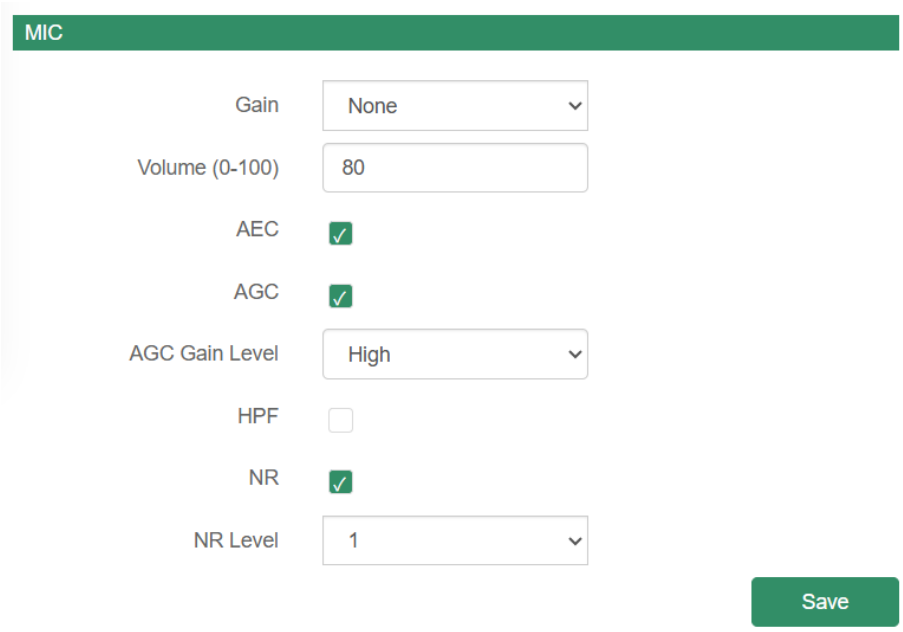
Volume: Adjust MIC volume at 0-100.

AEC: Acoustic Echo Cancellation.

AGC: Automatic Gain Control.

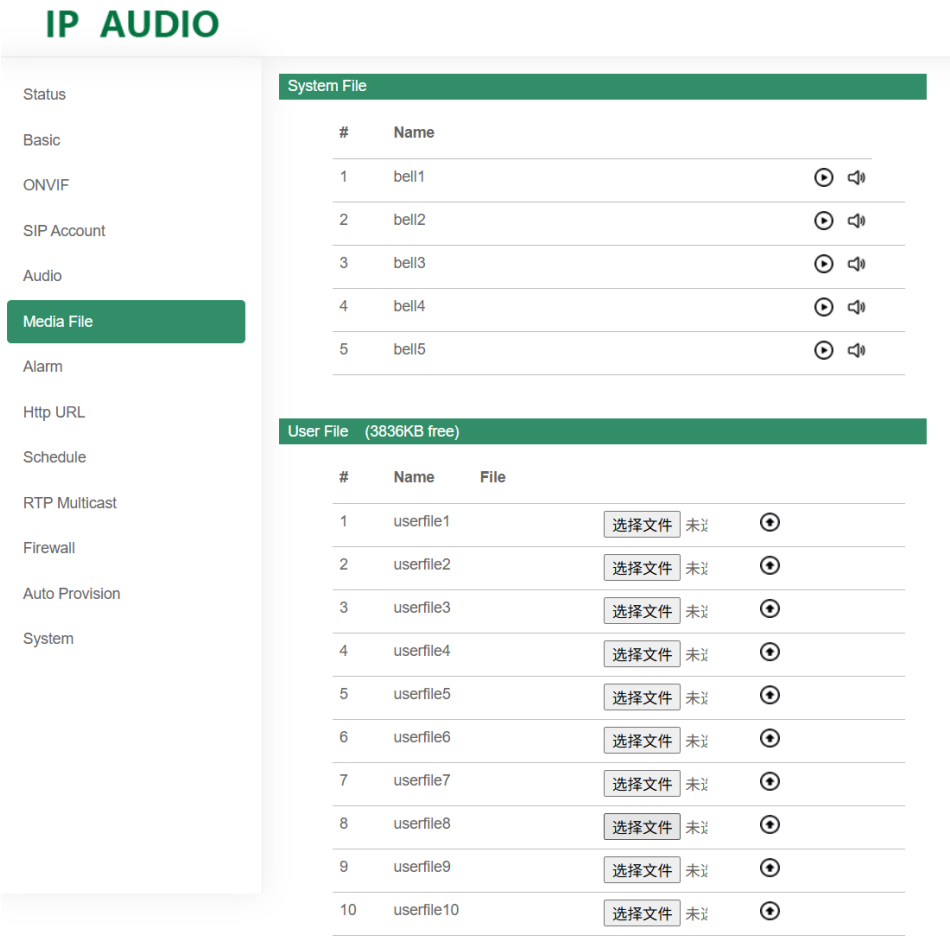
HPF: High Pass Filter

NR: Noise Reduction



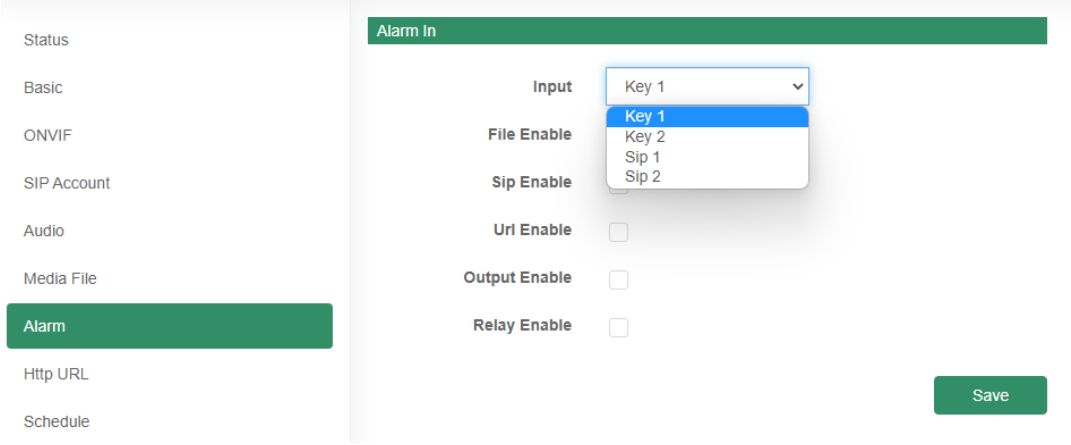
3.6 Media File

There are five system ringtones, and you can upload 10 media files as customers' demands: music, announcement, bells, etc.



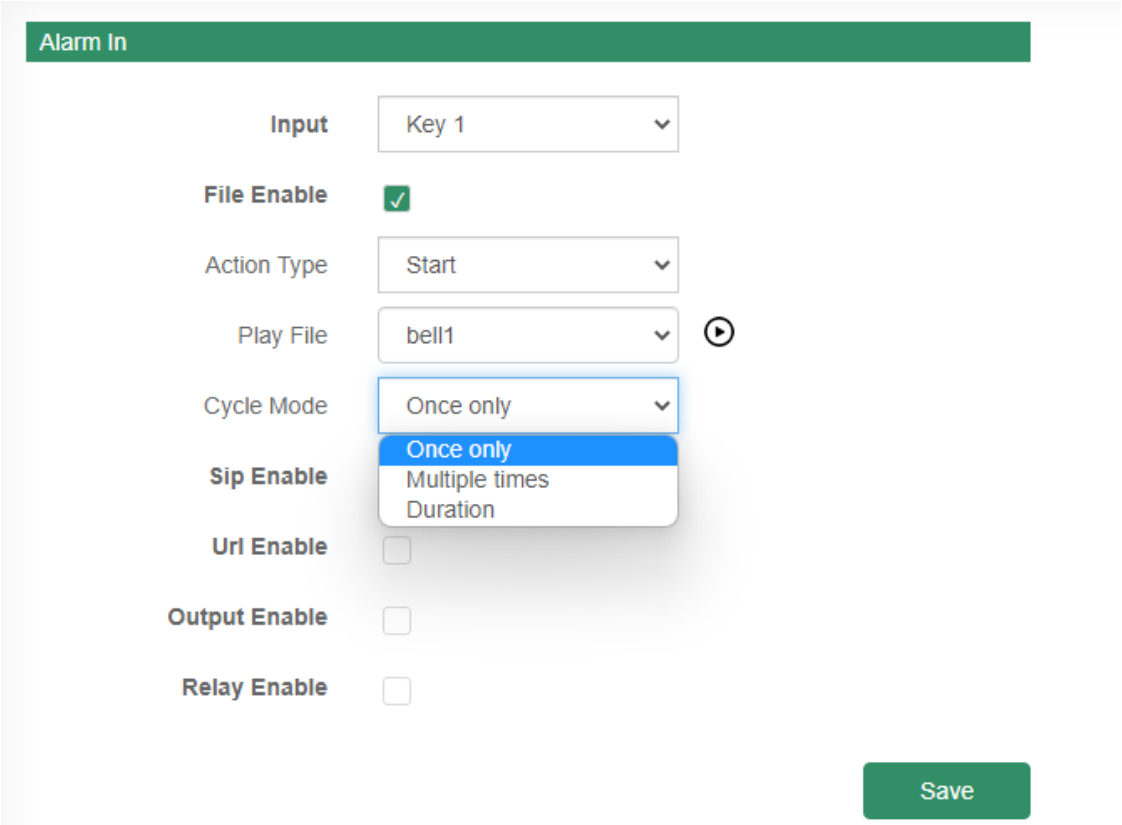
3.7 Alarm

We can set 2 DSS keys and 2 SIP accounts to realize alarm function, to ready the combination with alarm system.

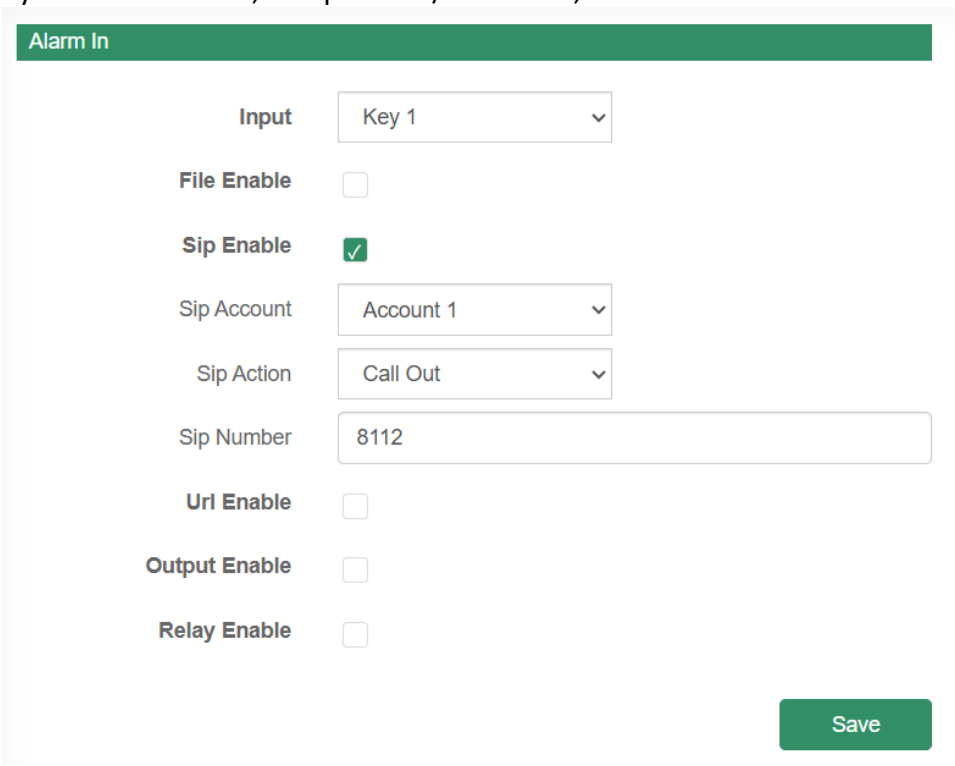


3.7.1 DSS Key Setting

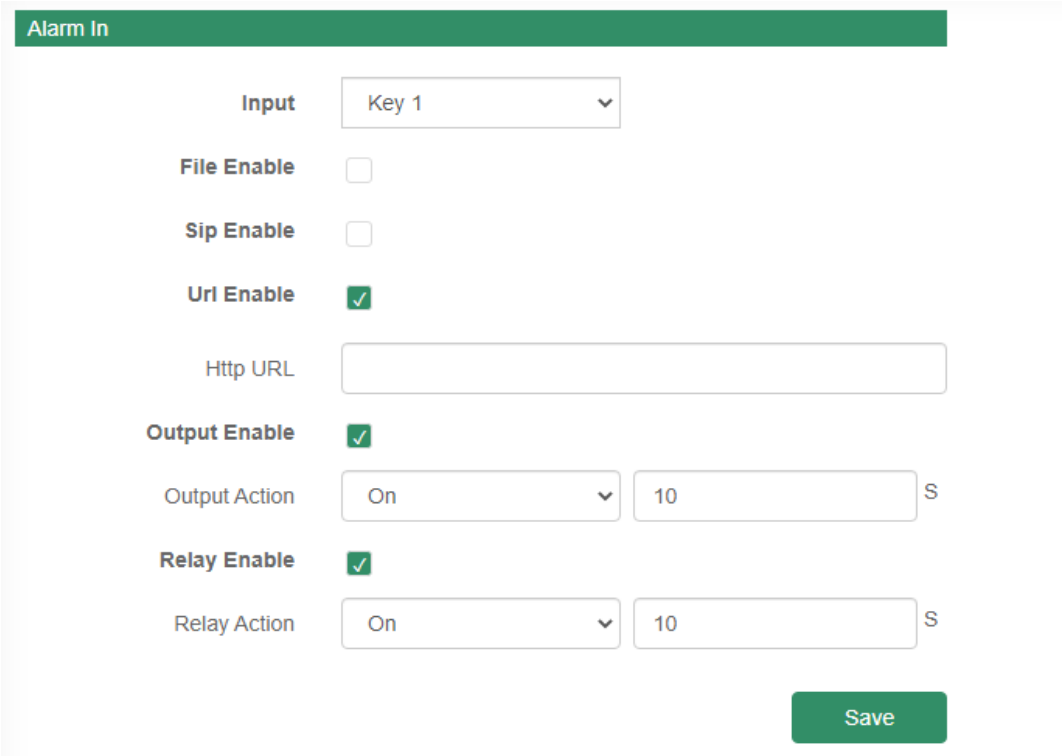
• Enable the file, you select an action type (start/ stop), play file and cycle mode, save the configuration, then press buttonK1 & K2, the bell will ring/close.



• SIP enable: Choose a SIP account you register, SIP action: call out/hang up, you can put the SIP number, e.g.: 8112, make sure it's the extensions which connected to the same IP sever with SIP account 1&2.
If you select call out, and press K1/K2 button, then extension 8112 will receive a call.



• Enable URL: Put the HTTP URL, after pressed K1/K2, the URL will be working.
• Output Enable: Turn on/off the output, press K1/K2 , the output succeeds.
• Relay Enable: Turn on/off the output, press K1/K2 , the relay succeed.



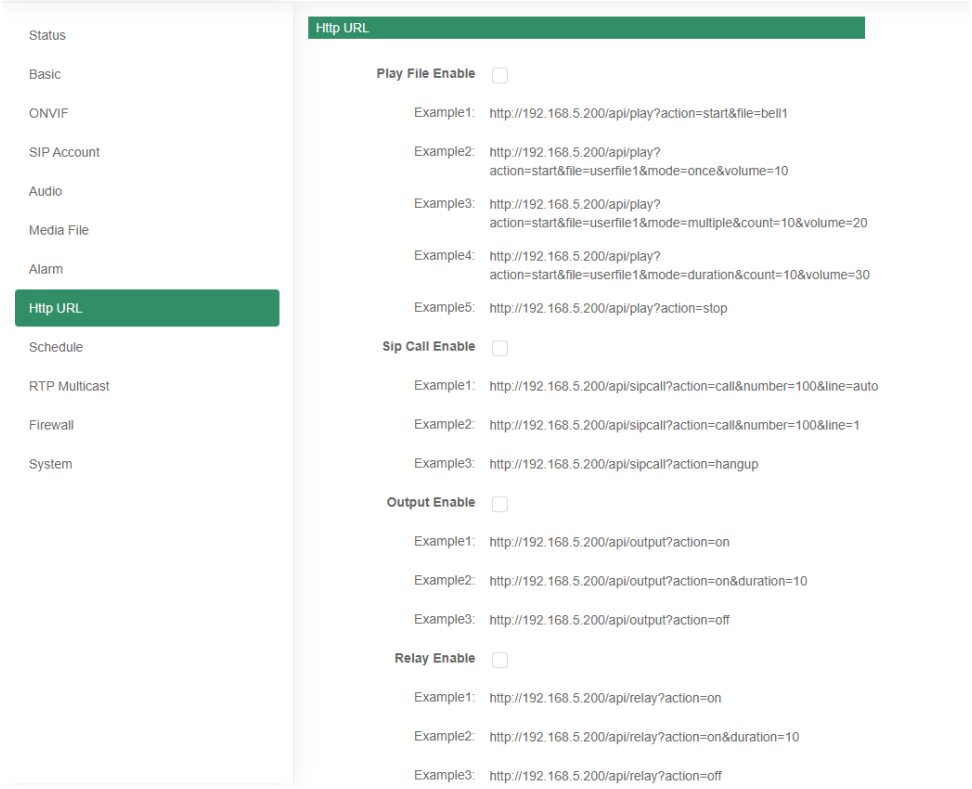
3.8 HTTP URL

User can control the alarm by HTTP URL:

(1) Enable the selection;

(2) Open any browser you have in computer;

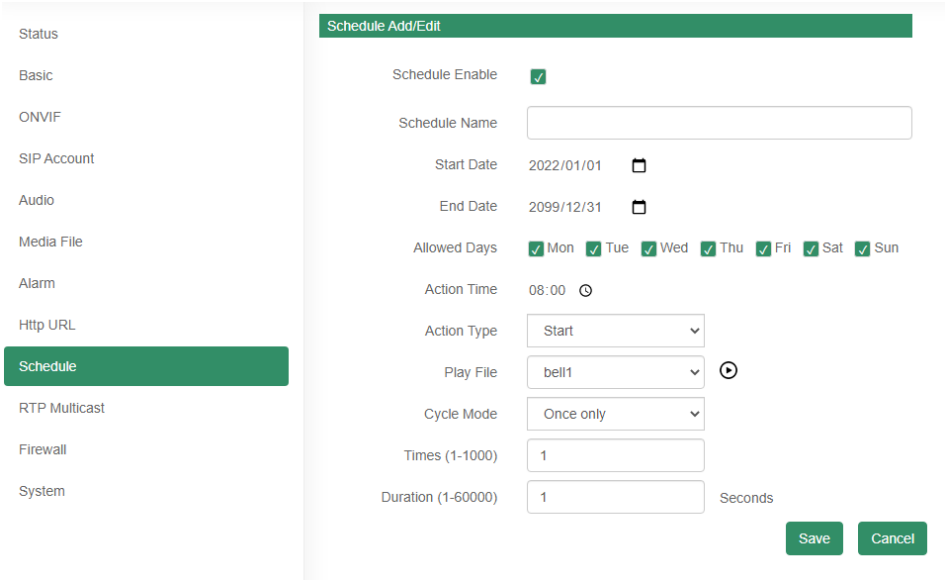
(3) Put the URL as the following examples, enter it.



3.9 Schedule

This function is widely use in school, factory and office projects. Making a regular bell, announcement and alarm.

Enable the schedule, you can name the schedule. then setting it step by step.



3.10 RTP Multicast IP

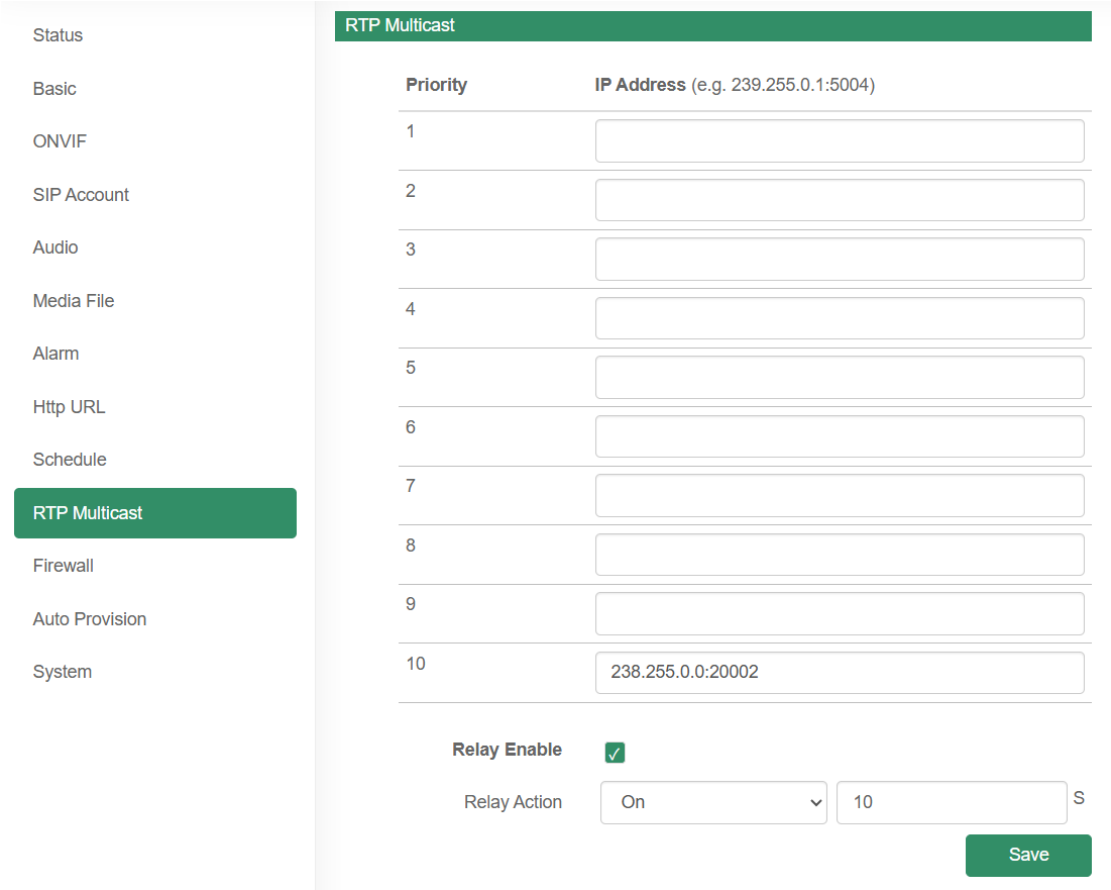
There are 10 RTP addresses can be received for each device, please note that: port numbers do not use continuous numbers when setting the same RTP addresses. Use discontinuous numbers. e.g.:

239.255.1.2:8000, 239.255.0.1:8001, 239.255.0.1:8002 (×)

239.255.0.1:8000, 239.255.0.1:8002, 239.255.0.1:8004 (√)

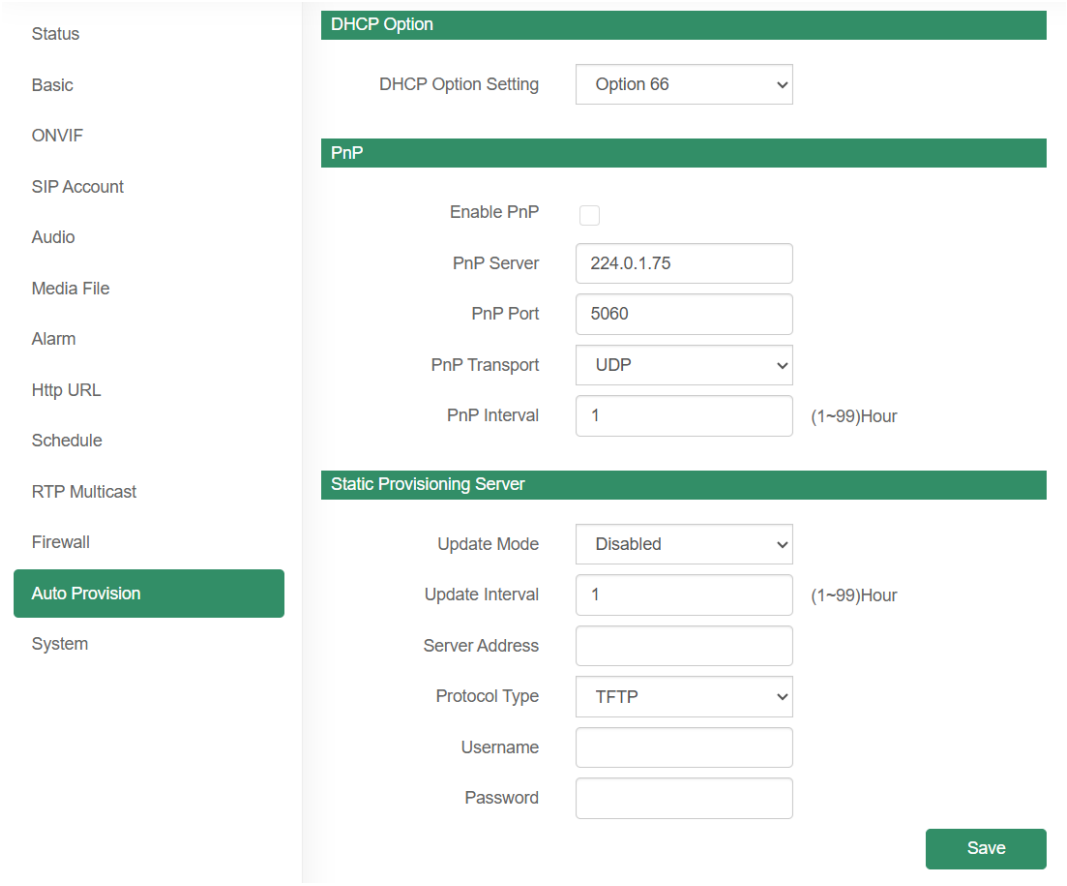
- Multicast address range: 224.0.0.0-239.255.255.
- Ports range: 1024-65536

Relay Enable: When Relay Enable is checked, the relay action will be activated during RTP playback, and its duration can be set.



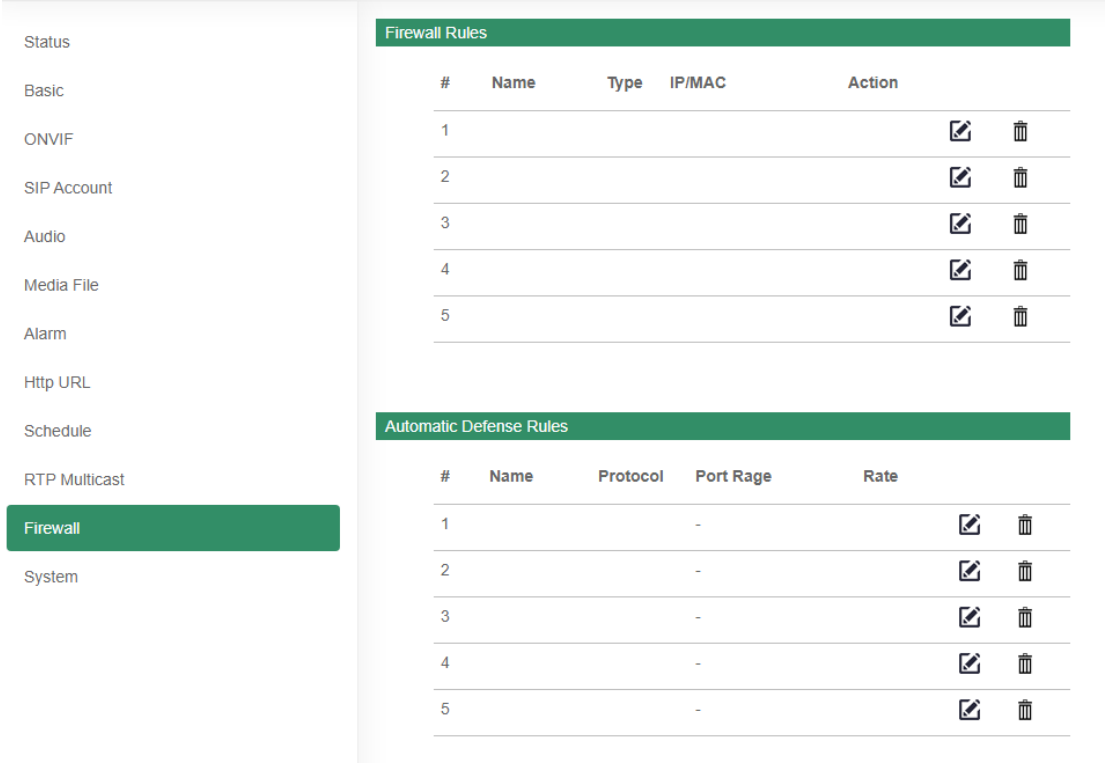
3.11 Auto Provision

Configures DHCP Option, PnP, and Static Provisioning Server parameters for automatic device configuration.



3.12 Firewall

This function is use to protect your network safety. You can edit the firewall automatic defence rules as you need as follows.

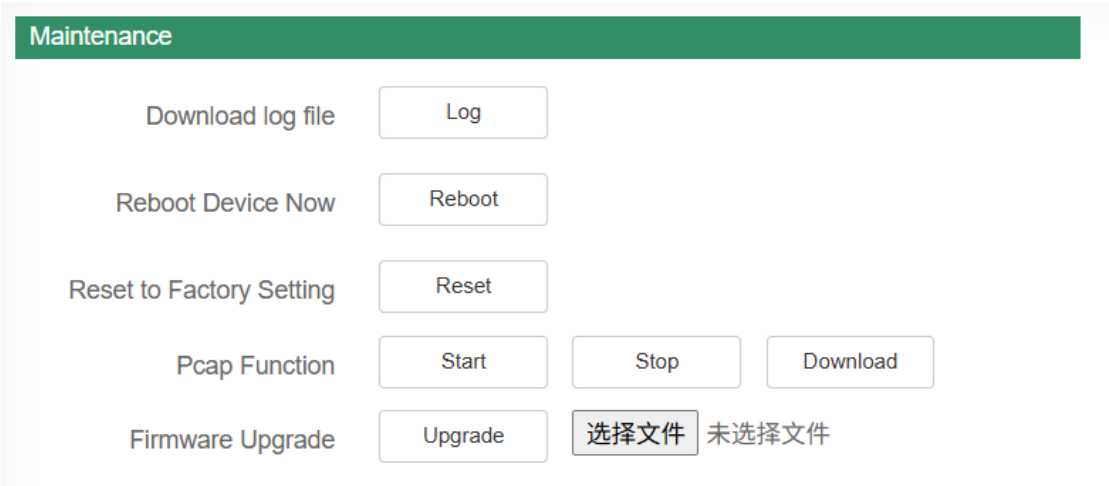


3.12 System

3.12.1 Maintenance

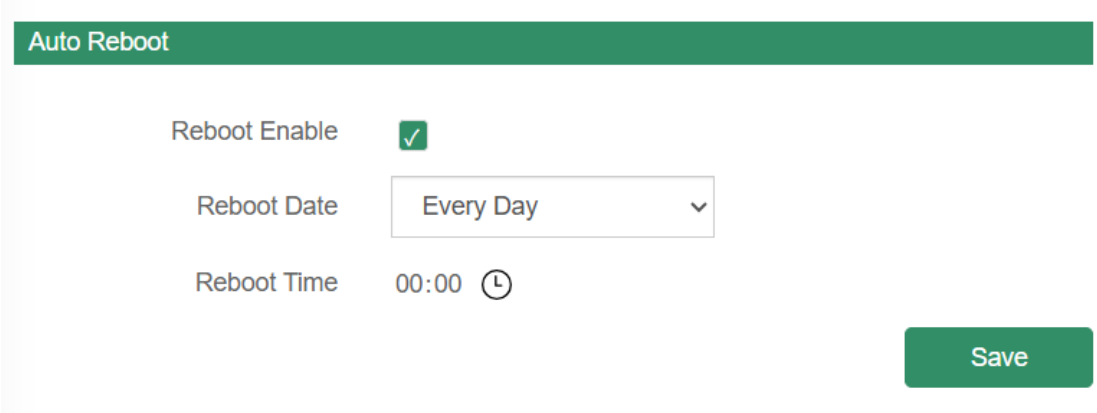
Log: Click to view or download system logs.
Reboot: Click to restart the device. Automatically redirects to login page after reboot.
Reset: Click to restore factory settings. Automatically redirects to login page after reset.
Pcap Function: Start: Begin packet capture; Stop: Halt packet capture; Download: Retrieve captured data

How to upgrade SIP-T20 firmware version in web interface?
(1) Select the latest version firmware T20-xxx-bin.
(2) Click upgrade to refresh, it would require about 20s.
(3) Re-login the web interface, latest version has upgraded.



3.12.2 Auto Reboot

Checked Reboot Enable option and set Reboot Date/Time to trigger automatic device restart at the scheduled time.



3.12.3 Security

Set a new user name and password as you need, save the configuration and restart login.