Tonmind IP Paging Adapter

SIP-T20

User Manual



Xiamen Tonmind Technologies Co., Ltd.

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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.

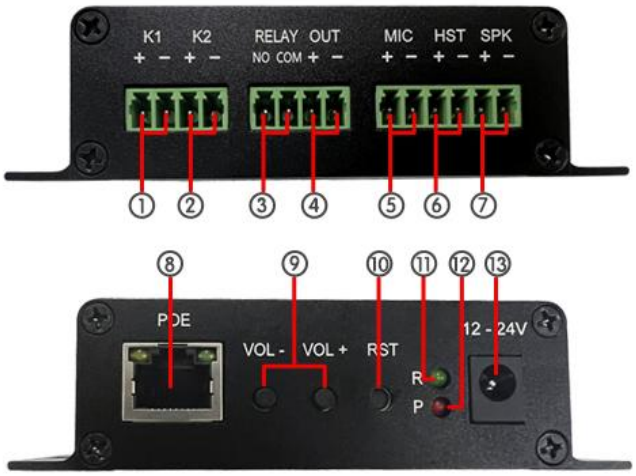








2. Interface Description

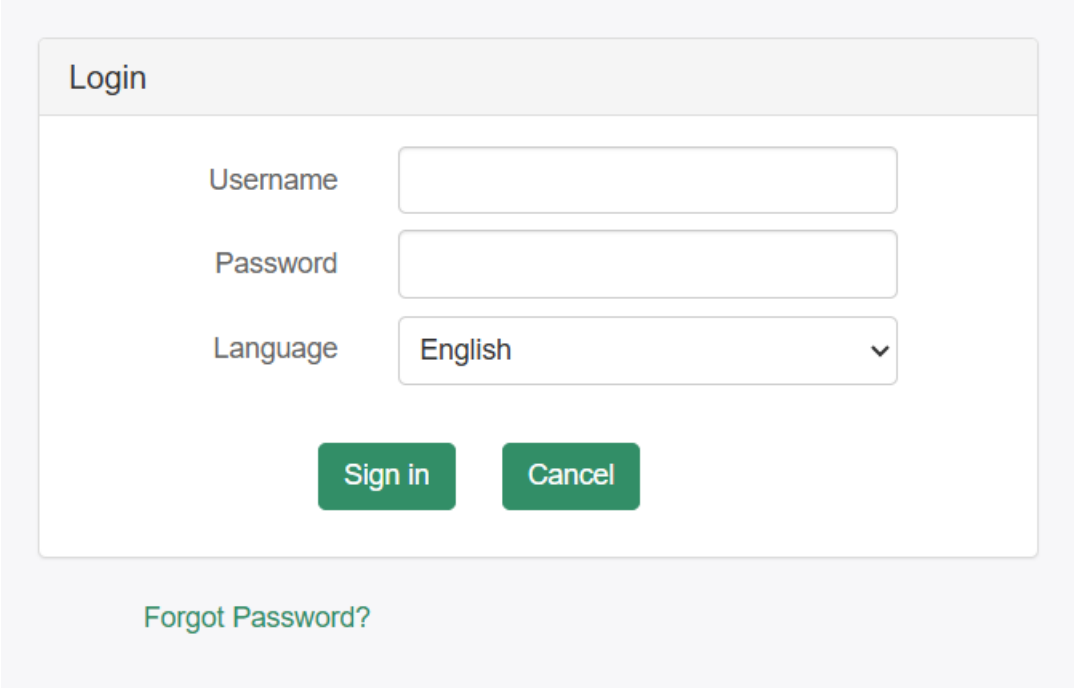


① K1 DSS Key	Connect to: 1. external keys 2. infrared probe and emergency switch 3. door sensor and other switch components
② K2 DSS Key	Connect to: 1. external keys 2. infrared probe and emergency switch 3. door sensor and other switch components
③ Relay NO COM Interface	Control the alarm system on/off
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
⑤ Microphone 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.

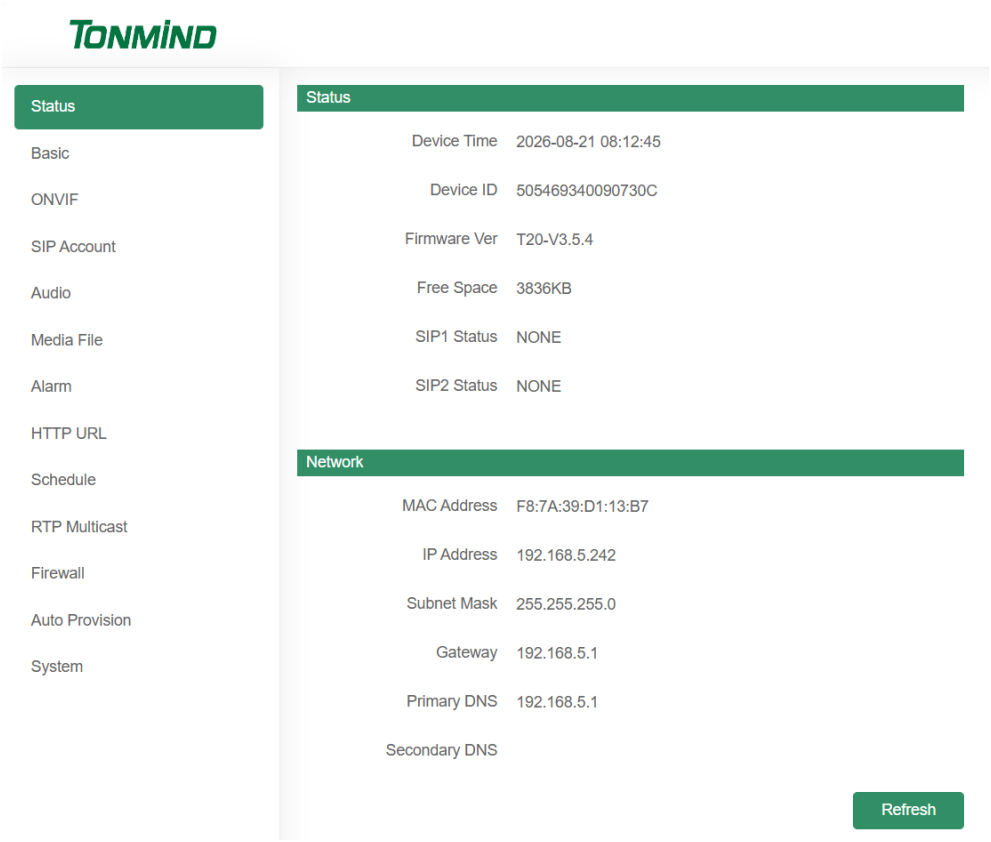
3. Web Configuration

Web configuration includes complete function setting, after the device and PC are connected to a same network, type the device's IP address in a web browser, the default IP address is 192.168.5.200, then log in with defaulted username and password as below, and there are different language options.
Username: admin
Password: tm1234

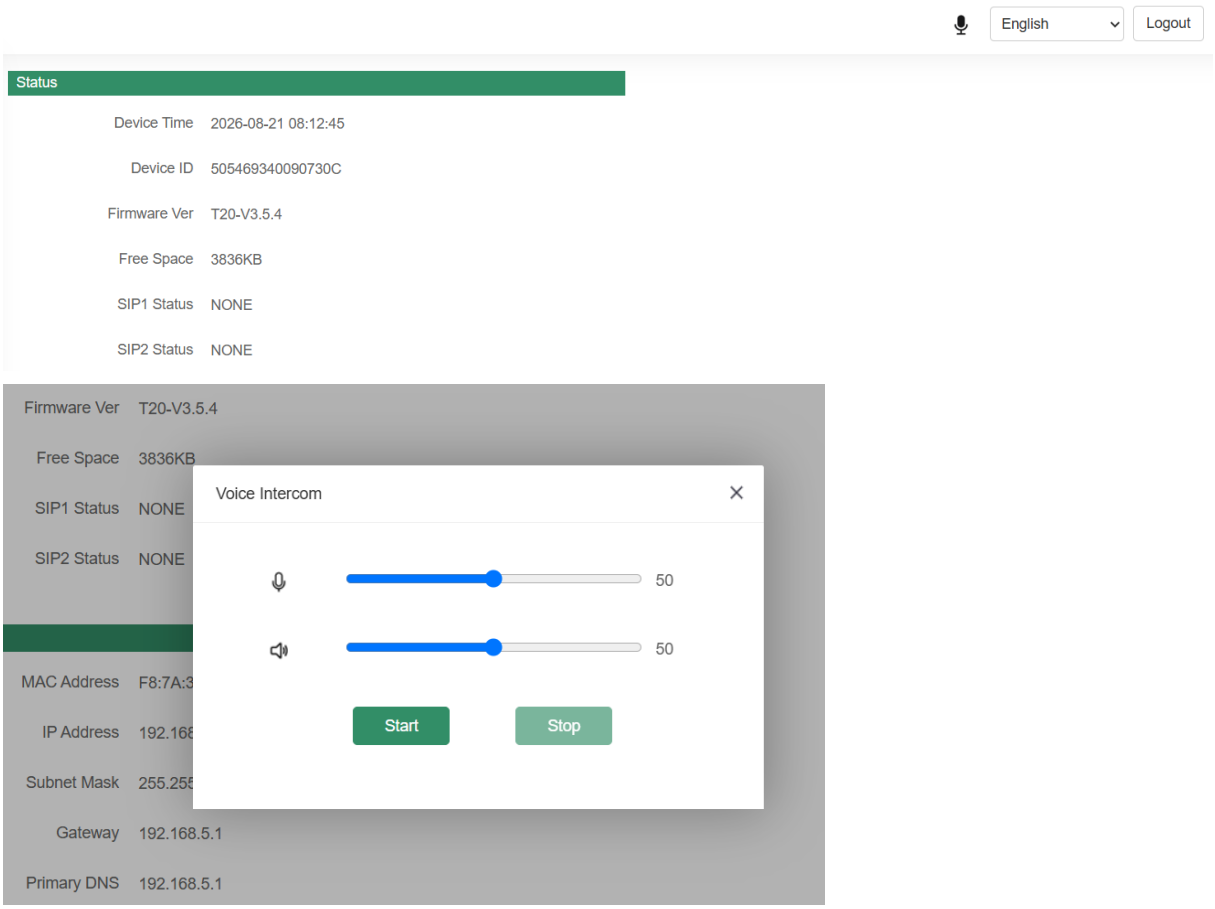


3.1 Status

The device status page displays the device time, firmware version, available storage space, SIP account status, and network information including MAC address, IP address, subnet mask, and gateway.



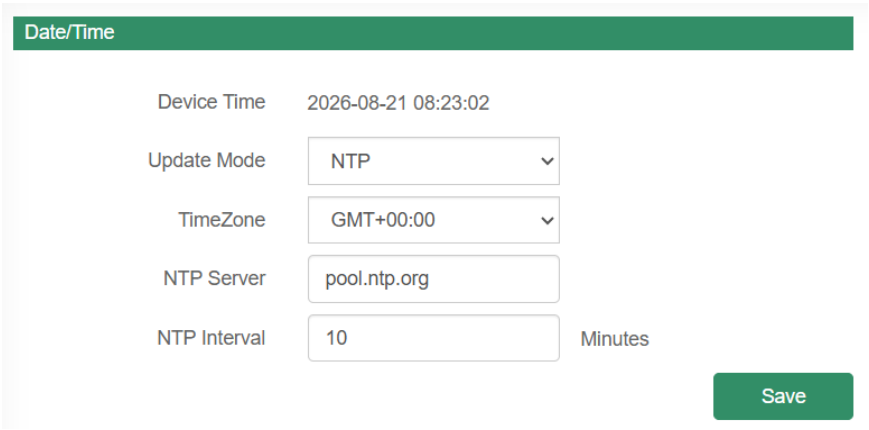
Web Calling Support: Click the microphone logo in the top right corner. You will be redirected to an HTTPS page. Click again on the new page to initiate the call.



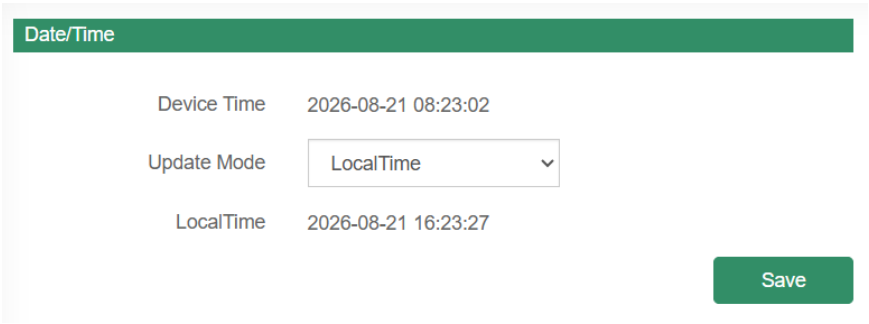
3.2 Basic

3.2.1 Date/Time

There are two update modes for time: NTP and Local Time.
NTP: Select a time zone and save. The device time will sync automatically.

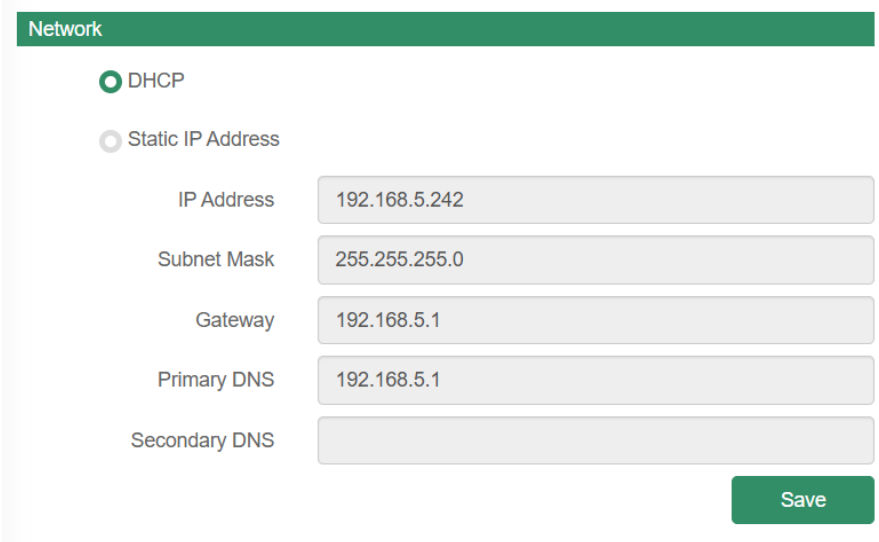


Local Time: Save to sync the device time with the current computer time.



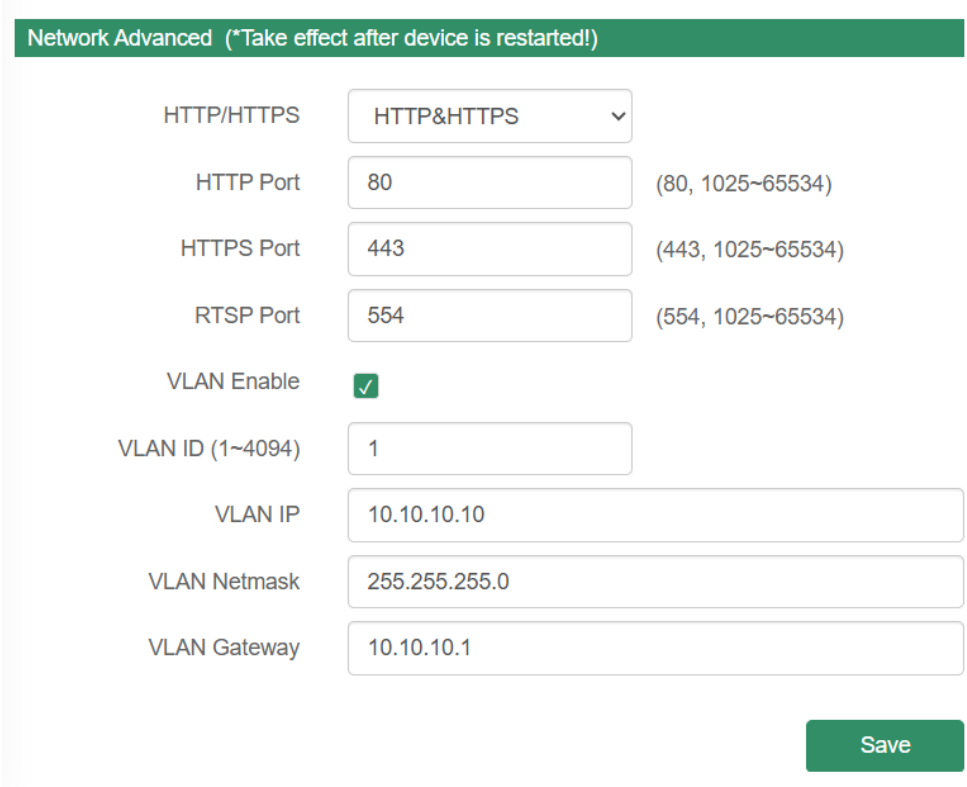
3.2.2 Network

- DHCP: The IP address is automatically assigned by a DHCP server.
- Static IP Address: Manually configure the IP Address, Subnet Mask, Gateway, Primary DNS, and Secondary DNS as required.



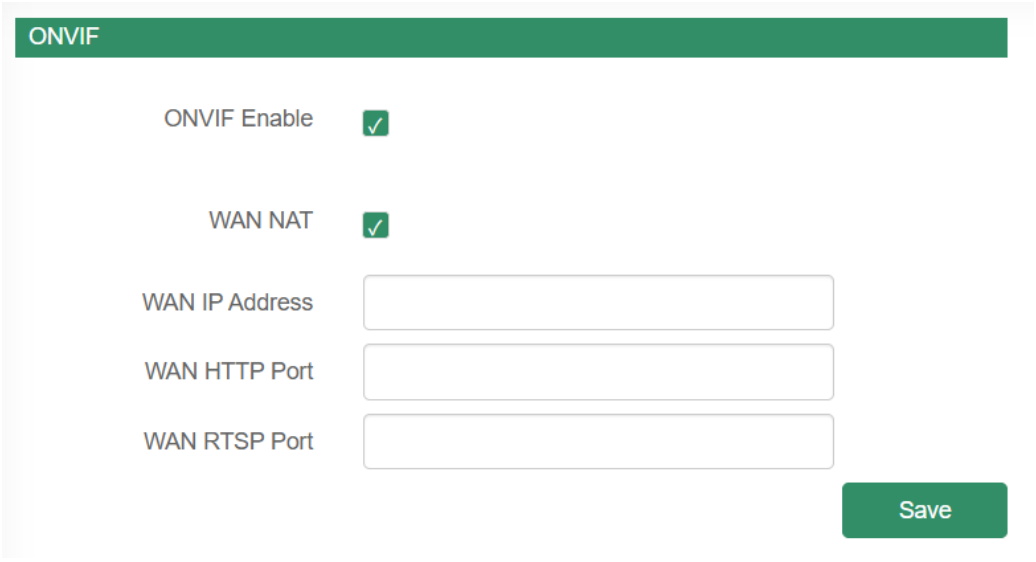
3.2.3 Network Advanced

Supports HTTP, HTTPS, or HTTP&HTTPS modes.
Users can set the HTTP Port, HTTPS Port, and RTSP Port.
Enable “VLAN Enable” to configure the VLAN ID, VLAN IP, VLAN Netmask, and VLAN Gateway.



3.3 ONVIF

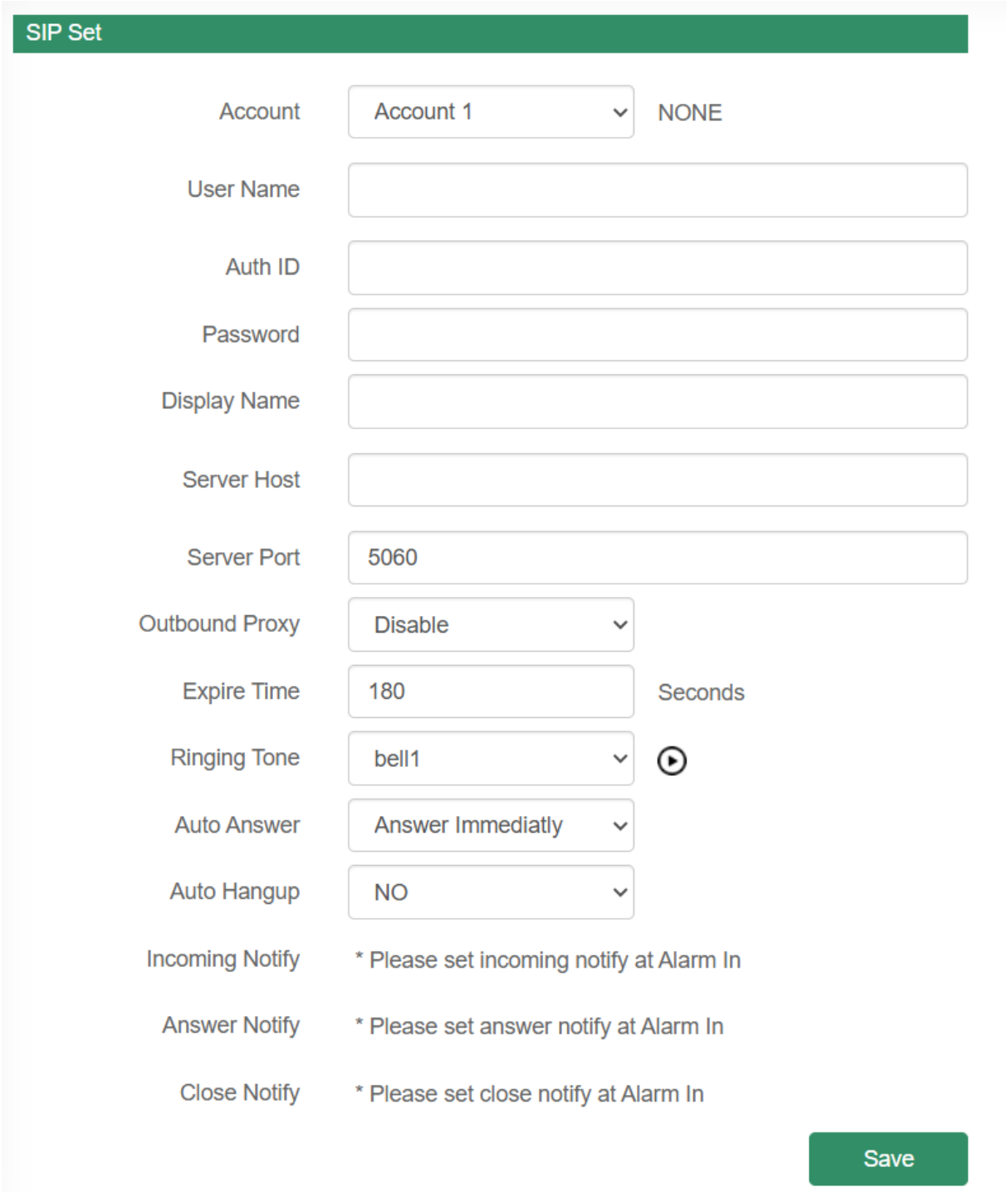
● Enable ONVIF to allow the device to be discovered by ONVIF VMS systems. The ONVIF username and password are the same as the T20 login credentials.
● For WAN NAT, configure port forwarding on the router to ensure proper communication between the server and the public internet. Then enable WAN NAT on the web page and enter the IP address, HTTP Port, and RTSP Port.



3.4 SIP Account

3.4.1 SIP Set

Each device supports two SIP accounts. Enter the SIP server and extension information, save the configuration, and check the registration status to confirm successful registration.



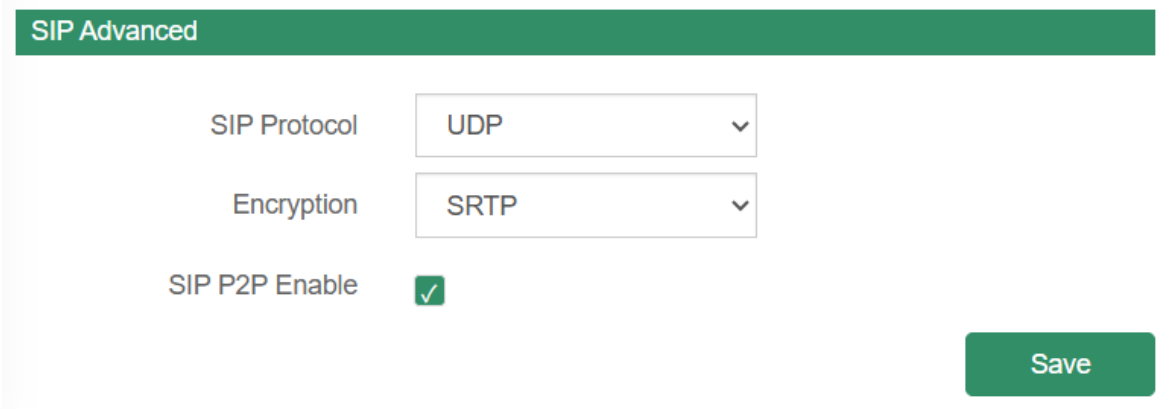
User Name	User account, provided by SIP server
Auth ID	SIP service subscriber's ID used for authentication.
Password	Account password provided by SIP server
Display Name	SIP service subscriber's name

Server Host	SIP server address
Server Port	SIP port, default to be 5060
Outbound Proxy	It is used to process signals and help data streams to go through firewall or NAT if there have.
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
Incoming Notify	On the Alarm page, select SIP1/SIP2 to configure this function. Enter a URL, and when an incoming SIP call starts ringing, the URL will be triggered automatically.
Answer Notify	On the Alarm page, select SIP1/SIP2 to configure this function. Enter a URL. When an incoming SIP call is answered, the URL will be triggered automatically.
Close Notify	On the Alarm page, select SIP1/SIP2 to configure this function. Enter a URL. When the SIP call is ended or disconnected, the URL will be triggered automatically.

3.4.2 SIP Advanced

● The devices support UDP, TCP, and TLS transport protocols, with optional SRTP encryption for secure audio communication.

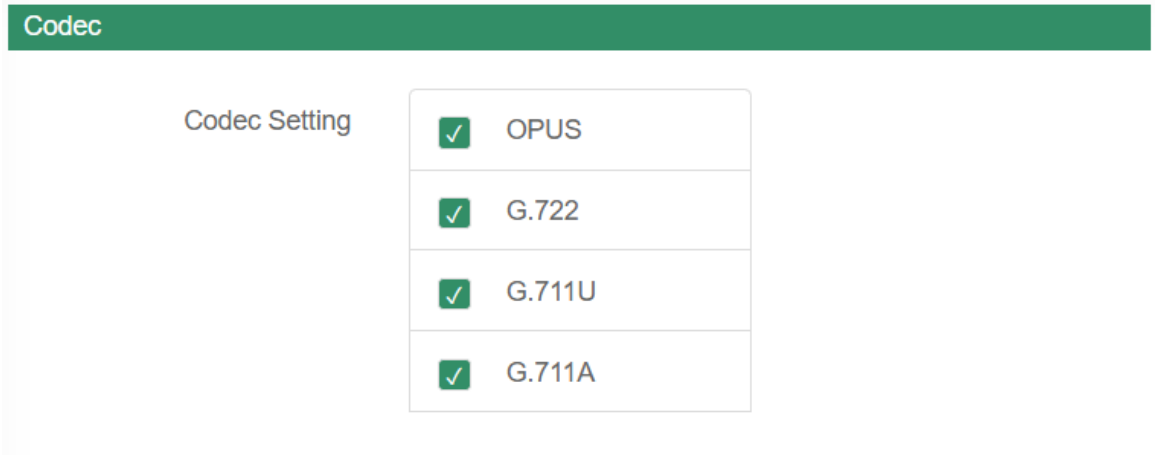
● To enable SIP P2P (Peer-to-Peer SIP), check “SIP P2P Enable”. In a local network, devices can communicate directly without a SIP server by using the SIP address of the target device. A typical SIP address format is: sip:192.168.5.200.



3.5 Audio

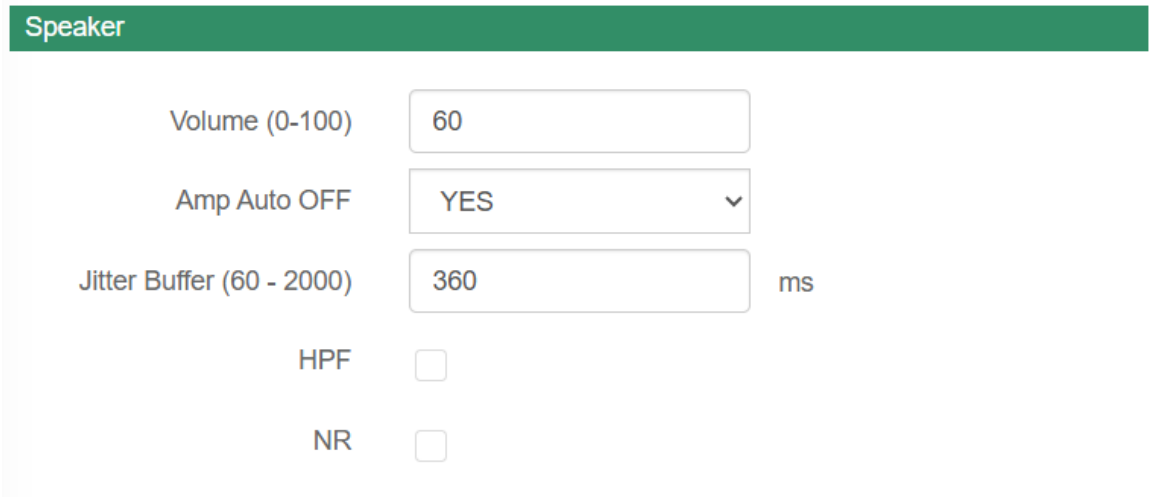
3.5.1 Codec

Under Audio settings, select at least one audio codec according to the required audio quality and network environment. The device supports four codec options: OPUS, G.722, G.711U, and G.711A.



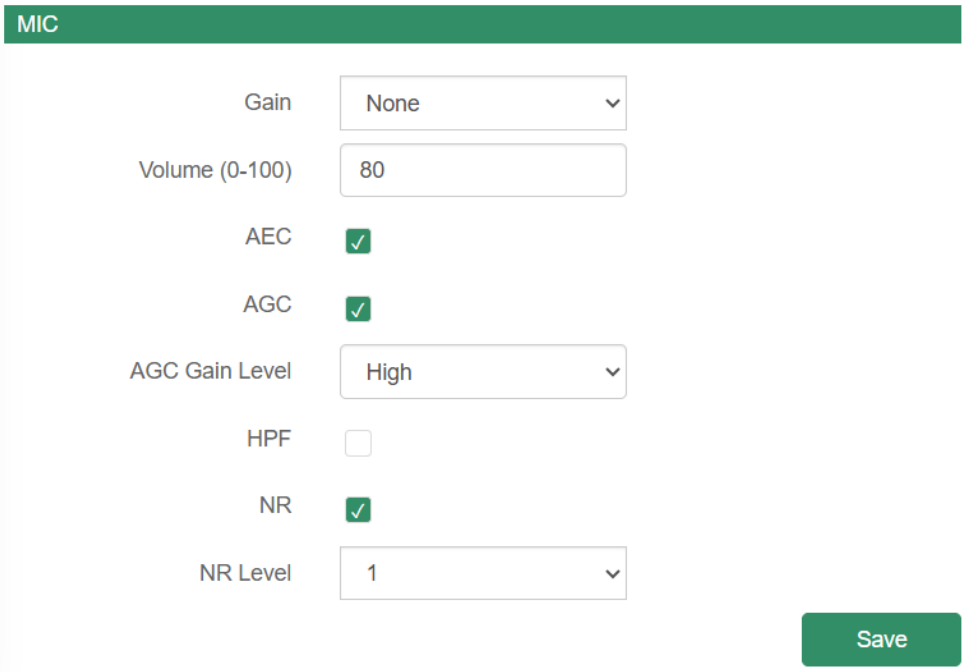
3.5.2 Speaker

- Volume: Adjust the speaker output volume.
- Amp Auto OFF: Default setting is “YES”. When the speaker is not playing audio, the built-in amplifier will automatically turn off to reduce power consumption.
- Jitter Buffer: Improves audio playback stability under network transmission.
- HPF: High Pass Filter. Filters frequencies below 150Hz to improve audio clarity.
- NR: Noise Reduction. Reduces background noise from the audio input to improve sound quality when enabled. The function is processed by the built-in audio chip.



3.5.3 MIC

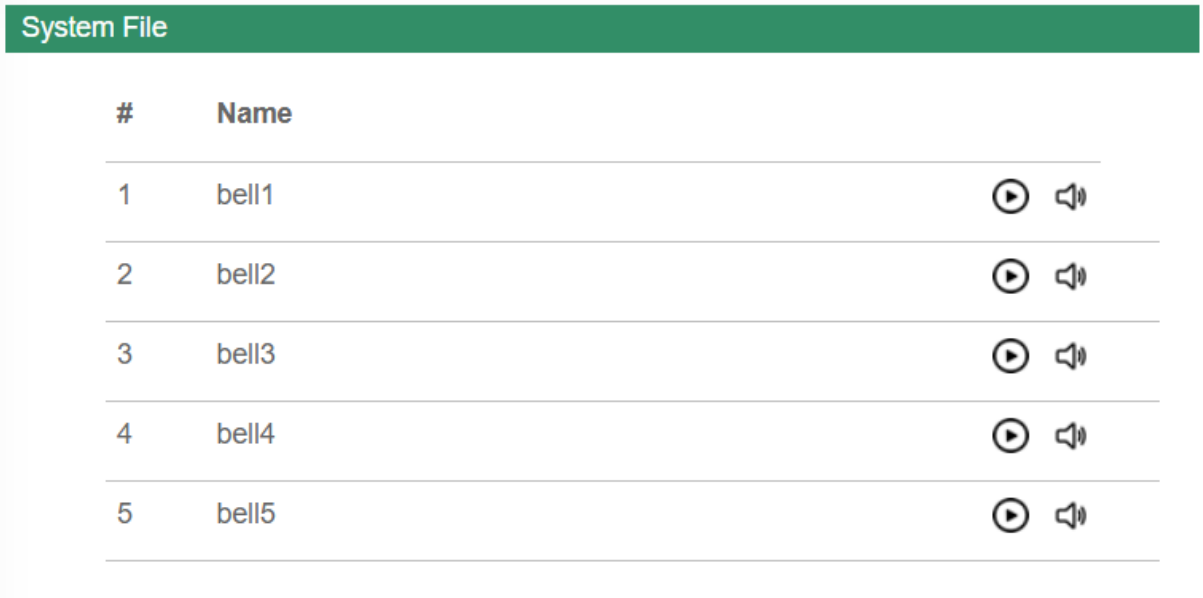
- Gain: Four levels (None, Low, Middle, High) for different input sensitivity.
- Volume: Adjusts microphone output volume.
- AEC (Acoustic Echo Cancellation): Reduces echo during communication.
- AGC (Automatic Gain Control): Automatically adjusts mic gain for stable audio levels.
- AGC Gain Level: Sets AGC strength (High, Middle, Low).
- HPF (High Pass Filter): Filters low-frequency noise below 150 Hz.
- NR (Noise Reduction): Reduces ambient noise from mic input. Three levels (0, 1, 2).



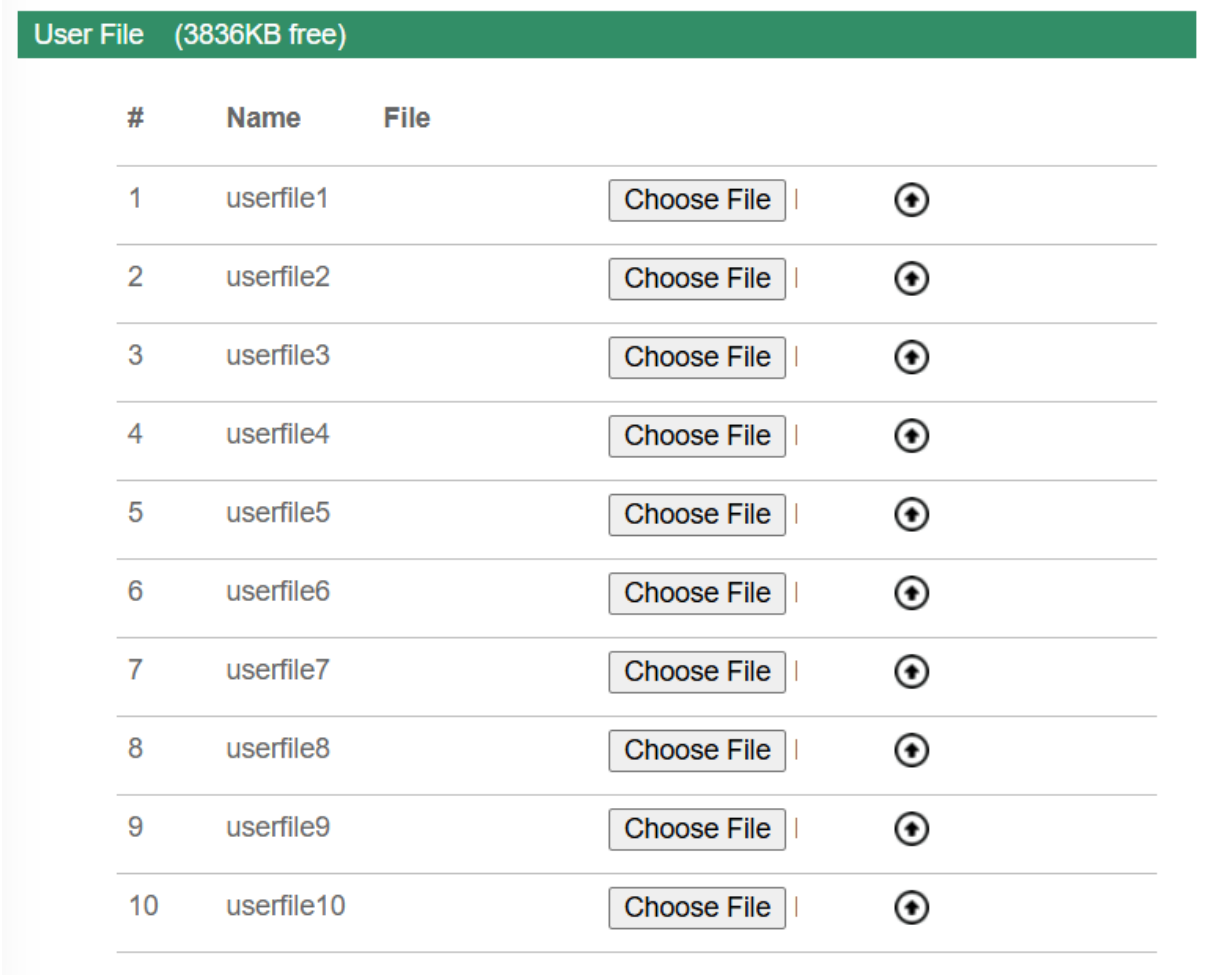
3.6 Media File

Click ▶ to listen on the PC, click 🔊 to play audio on the device side.

● System File: Includes 5 built-in system audio clips.



● User File: Supports uploading up to 10 custom audio clips. Available storage space is approximately 3800 KB.



3.7 Alarm

Alarm In supports three modes: Key 1/Key 2, SIP 1/SIP 2, and DTMF 1/ DTMF 2.

3.7.1 DSS Key

When the T20 Alarm In interfaces(K1/K2) is triggered, the selected actions will be activated. Users can enable: File Enable, SIP Enable, URL Enable, Output Enable and Relay Enable.

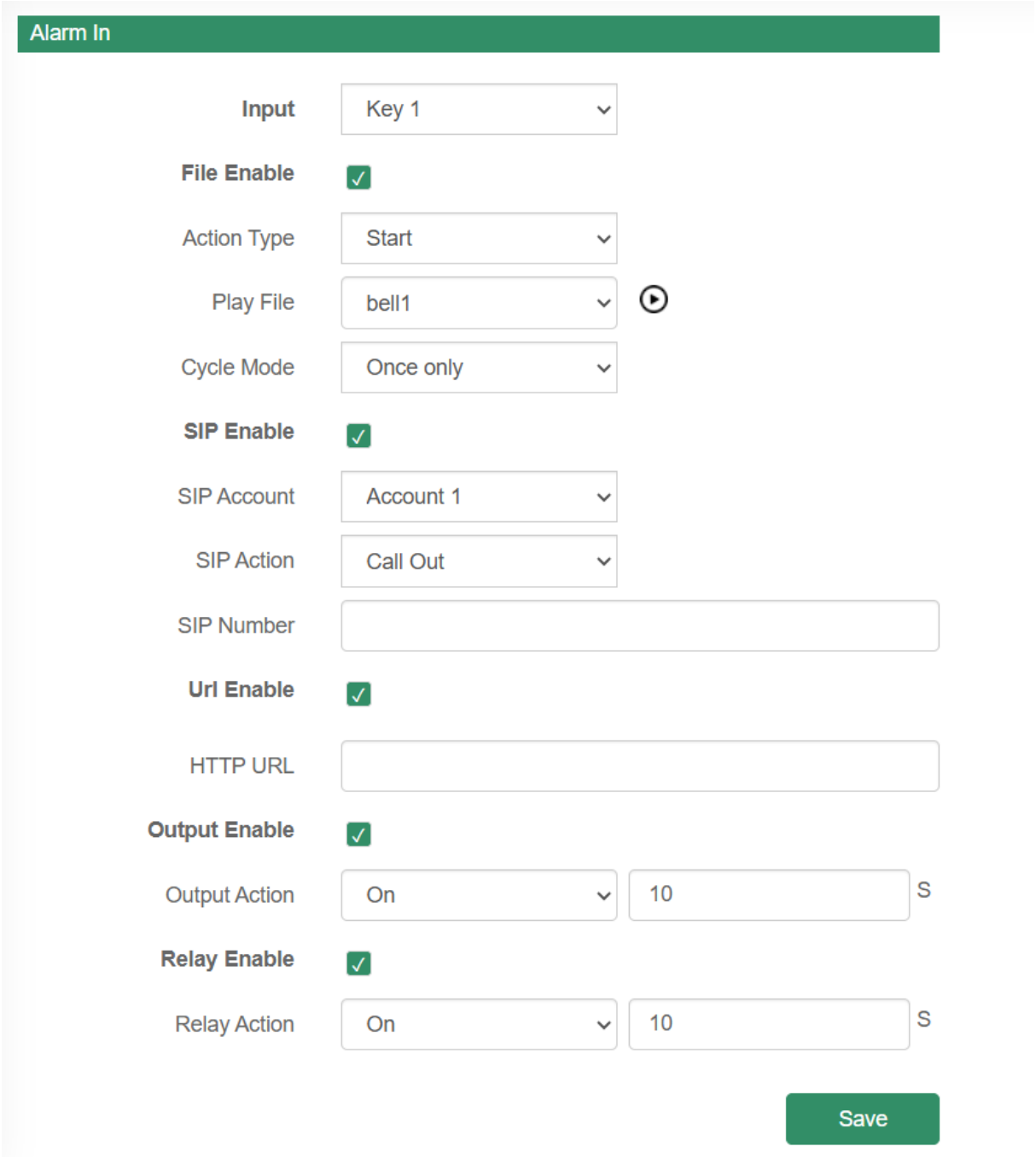
● File Enable: Play a stored media file from the device. Cycle modes include: Once Only, Multiple Times, Duration, and Infinite.

● SIP Enable: Select SIP Account 1 or 2 and enter the target SIP Number. When triggered, the speaker will call the specified SIP number.

● URL Enable: Enter an HTTP URL command, such as http://192.168.5.226/api/play?action=start&file=bell1. The command will be executed after Alarm In is triggered.

● Output Enable: When enabled, select On and set the activation duration in seconds, or select Off to disable the Output action.

● Relay Enable: When enabled, select On and set the relay activation duration in seconds, or select Off to disable the Relay action.



3.7.2 SIP

The Alarm In page allows users to configure actions for different stages of an incoming SIP call. When the selected SIP account receives a call, separate actions can be configured for the Incoming Notify, Answer Notify, and Close Notify stages.

● **Incoming Notify**

When an incoming SIP call starts ringing, the configured actions can be triggered automatically.

*URL Enable: Enable this option and enter an HTTP URL command, for example:

http://192.168.5.226/api/play?action=start&file=bell1

When the SIP call starts ringing, the URL will be triggered automatically.

*Output Enable: Enable the Output action. Select On and set the activation duration in seconds, or select Off to turn off the Output.

*Relay Enable: Enable the Relay action. Select On and set the activation duration in seconds, or select Off to turn off the Relay.

● **Answer Notify**

When the incoming SIP call is answered, the configured actions can be triggered automatically.

*URL Enable: Triggers the configured HTTP URL when the call is answered.

*Output Enable: Activates or turns off the Output according to the configured action.

*Relay Enable: Activates or turns off the Relay according to the configured action.

● **Close Notify**

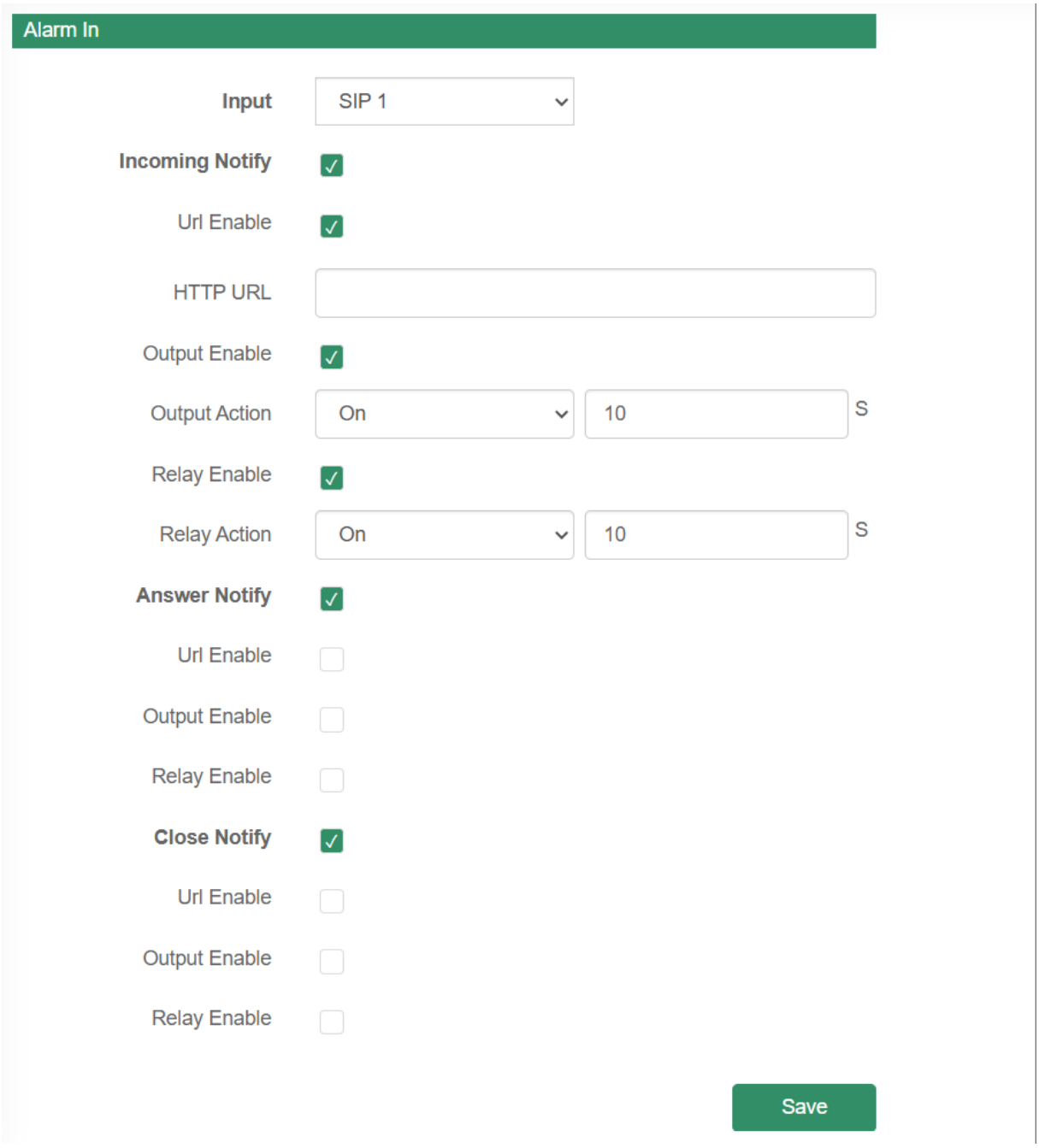
When the SIP call ends or is disconnected, the configured actions can be triggered automatically.

*URL Enable: Triggers the configured HTTP URL when the call is closed.

*Output Enable: Activates or turns off the Output according to the configured action.

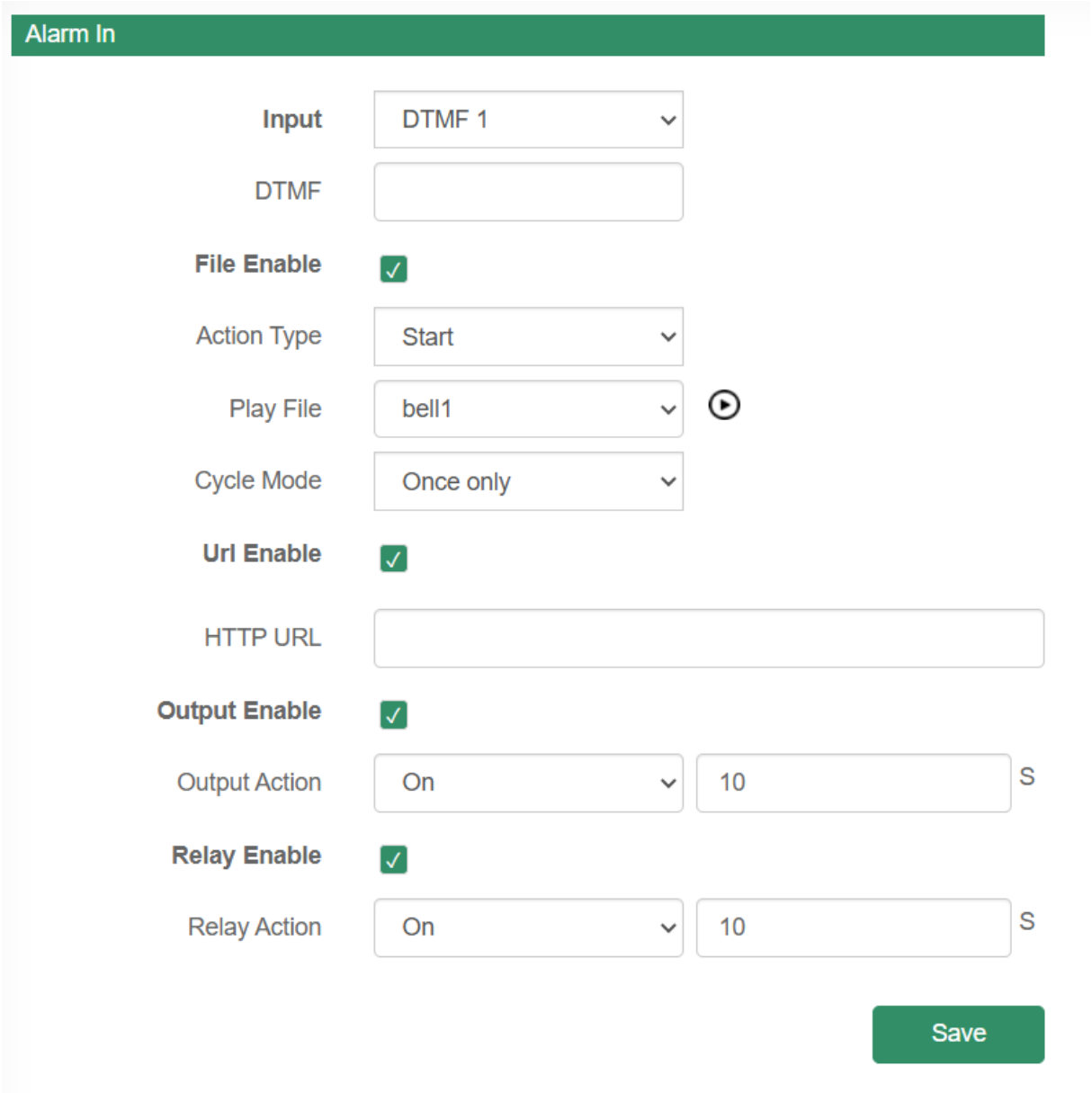
*Relay Enable: Activates or turns off the Relay according to the configured action.

Click Save to apply the settings.



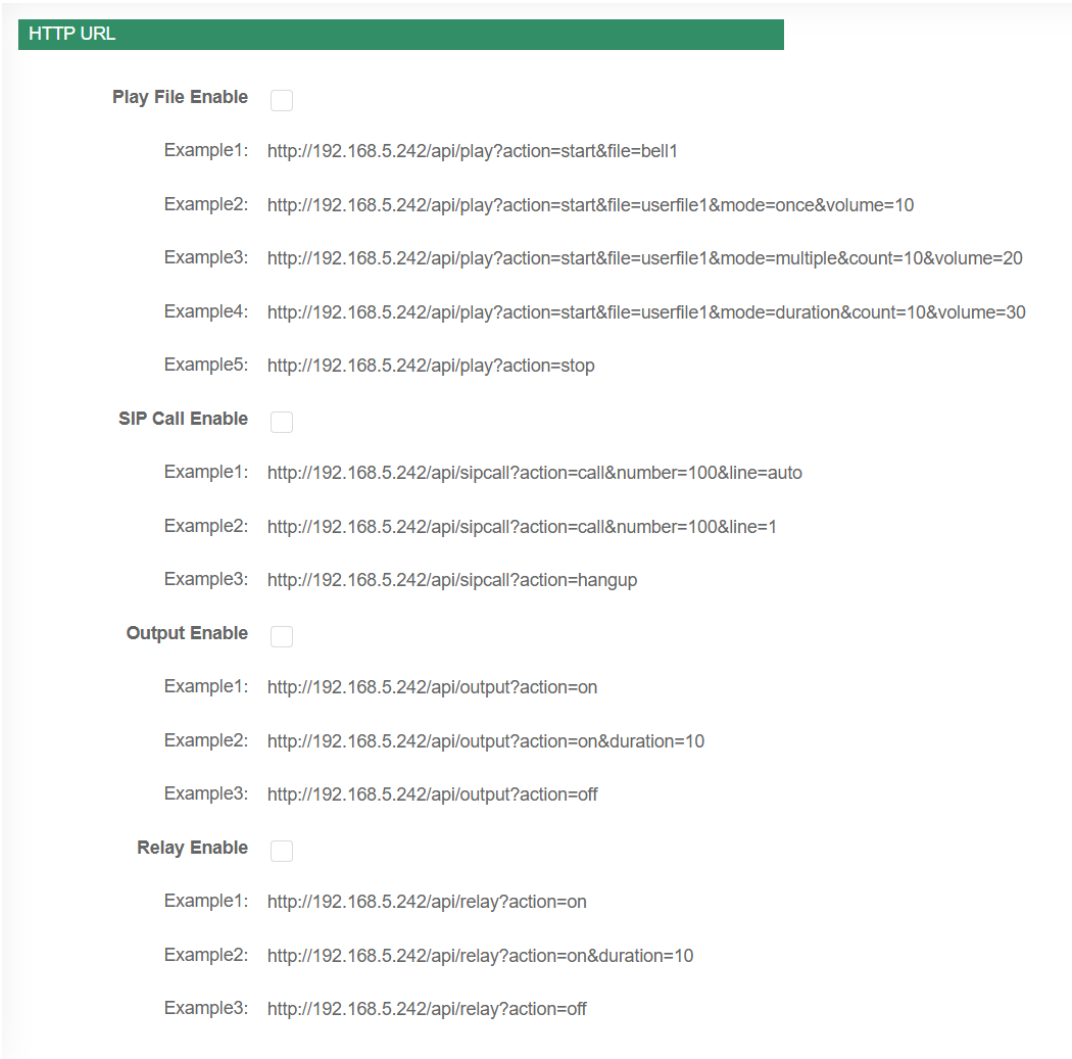
3.7.3 DTMF

In DTMF1 or DTMF2 mode, the T20 can be triggered by receiving the configured DTMF code during a call (such as from an IP Phone). After the correct DTMF code is received, the device can trigger File Enable, URL Enable, Output Enable or Relay Enable.



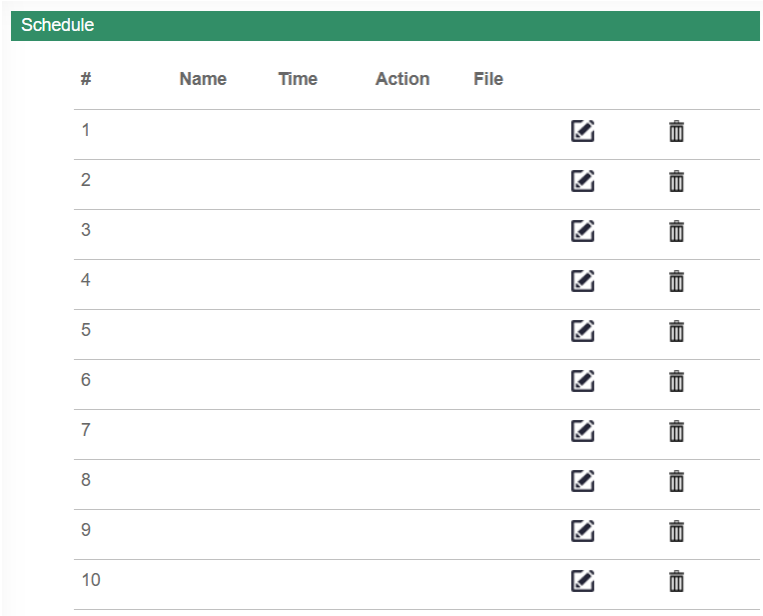
3.8 HTTP URL

Users can enable Play File Enable, SIP Call Enable, Output Enable and Relay Enable.
After enabling, the T20 can play audio through HTTP URL commands and receive URL control commands from other devices.
Under the options, several HTTP URL examples are provided for reference.

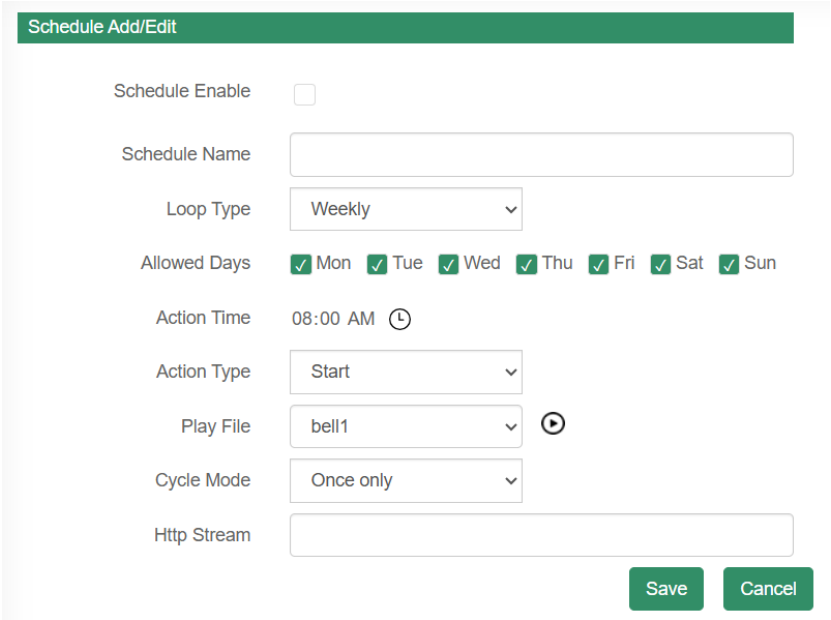


3.9 Schedule

Each T20 supports up to 10 schedules. Select a schedule and click to enter the Schedule Add/Edit page.



- Schedule Enable: Enable or disable the schedule.
- Schedule Name: Enter a name for the schedule.
- Loop Type: Three modes are available:
  - Once Only: Set Start Date and End Date.
  - Daily: Repeat every day.
  - Weekly: Select the days of the week for the schedule to take effect.
- Action Time: Click to set the schedule start time.
- Action Type: Select Start or Stop.
- Play File: Select a media file stored in the IP Speaker.
- Cycle Mode: Available modes: Once Only, Multiple Times, Duration, and Infinite.
- HTTP Stream: Enter an HTTP stream URL to play during the schedule, such as http://listen.livestreamingservice.com/181-greatoldies_128k.mp3



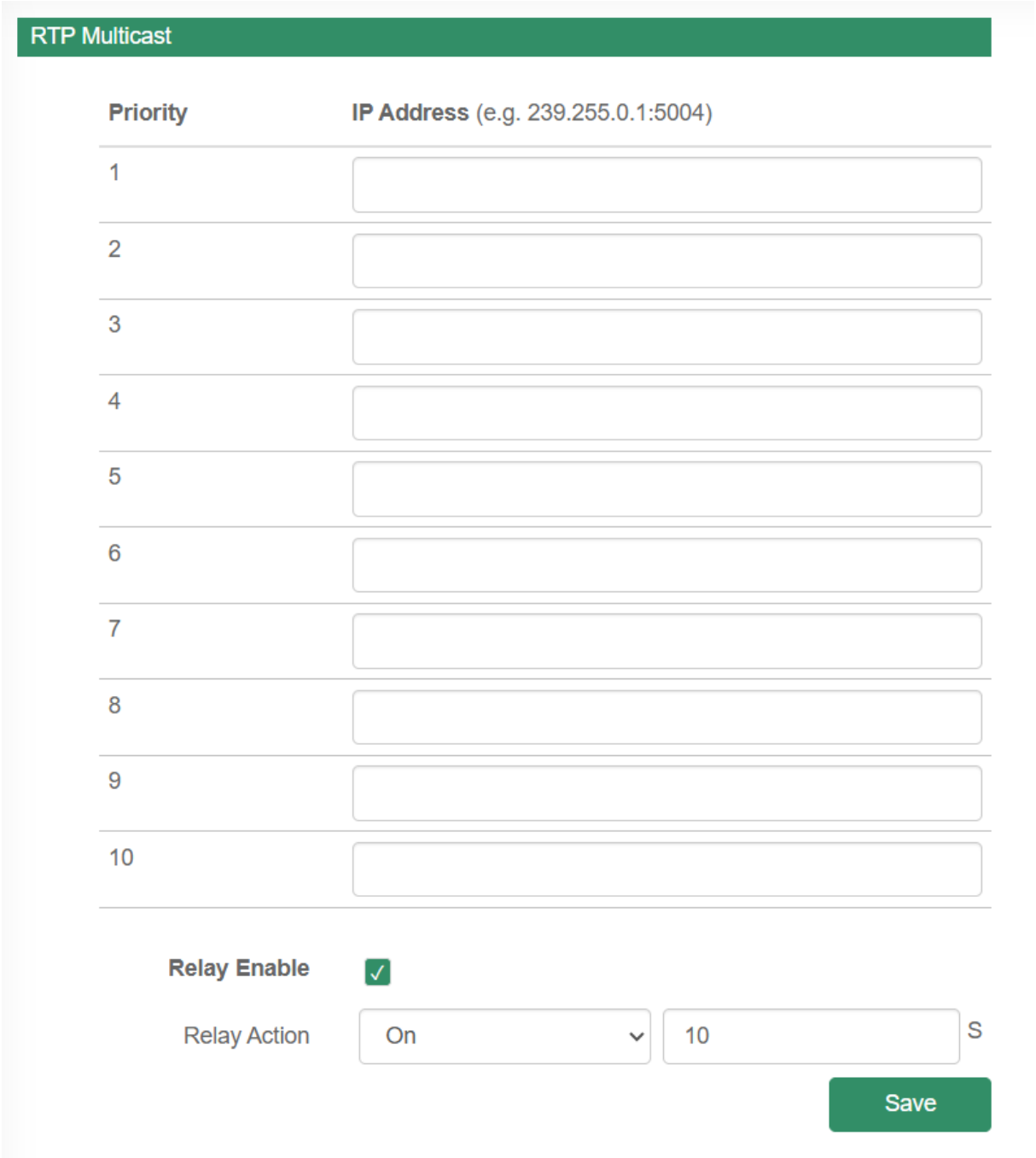
3.10 RTP Multicast

- Support 10 RTP addresses, 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 (√)

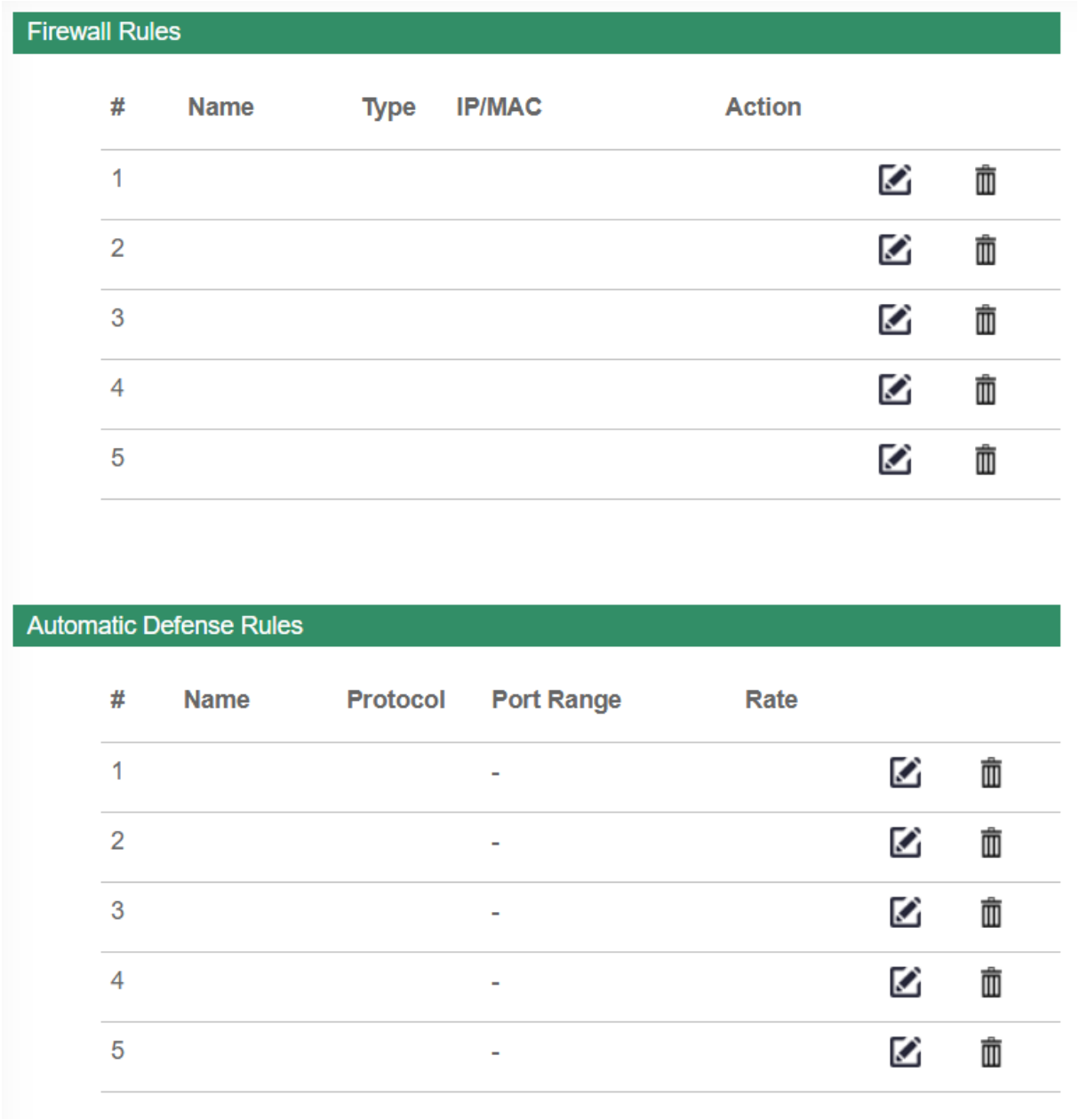
- Multicast address range: 224.0.0.0-239.255.255.
- Ports range: 1024-65536.

- Relay Enable: Select Relay Enable to control the Relay Output during RTP Multicast playback. The Relay Output can be set to On or Off, and the On duration can be configured in seconds.



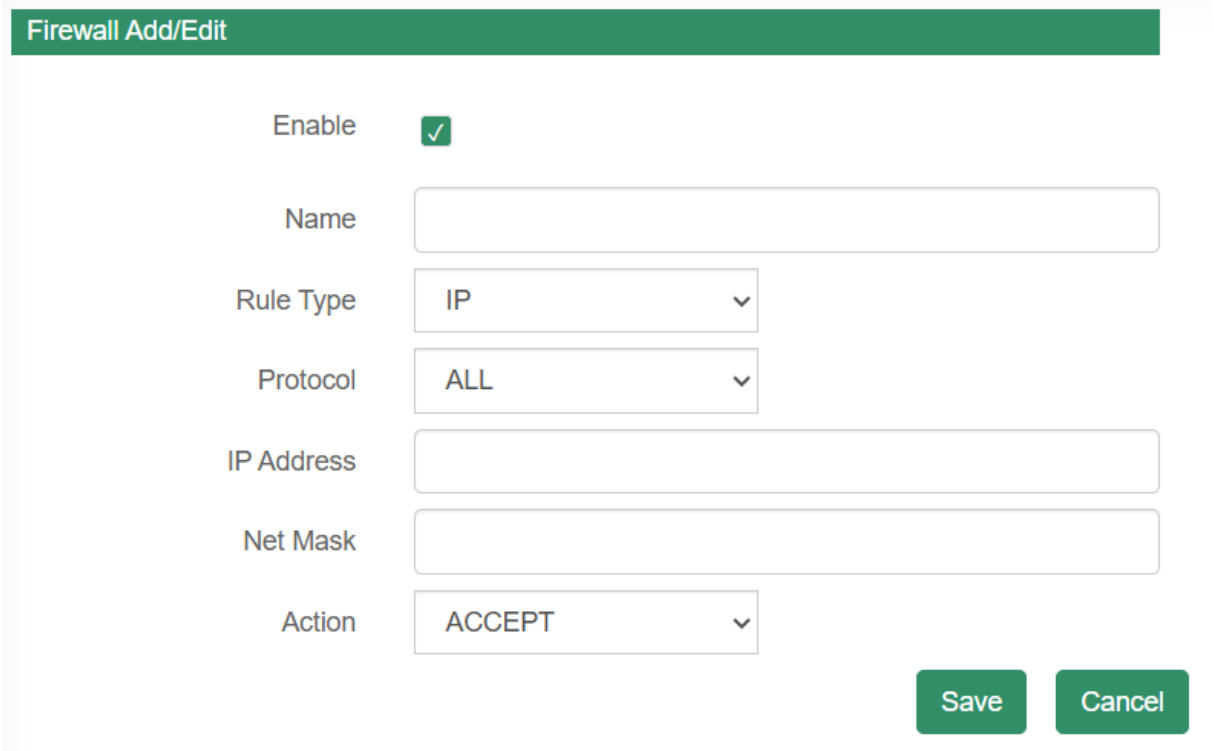
3.11 Firewall

The Firewall provides two types of protection: Firewall Rules and Automatic Defense Rules. Select one and click to enter the Add/Edit page.



3.11.1 Firewall Rules

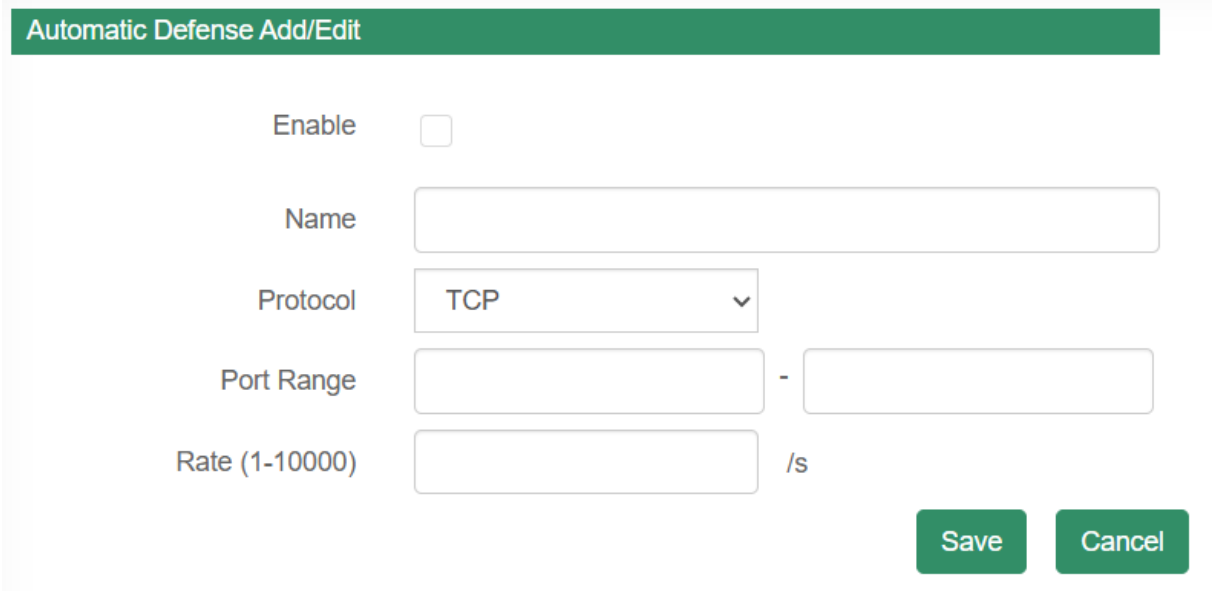
- Enable: Click to activate the firewall rule.
- Name: Enter a name for the rule.
- Rule Type: Choose IP Address or MAC Address.
- IP Address: Select protocol (ALL, TCP, or UDP), then enter the IP Address and Netmask.
- MAC Address: Enter the MAC address directly.
- Action: Choose ACCEPT or DROP.



3.11.2 Automatic Defense Rules

Used to automatically protect against abnormal traffic:

- Name: Enter a name for the rule.
- Protocol: Select TCP, UDP, or ICMP.
- Rate: Define the allowed request rate (per second).
- Port Range: Set the applicable port range.



3.12 Auto Provision

The Auto Provision section allows automatic device configuration based on MAC address. It supports third party servers using DHCP, PnP, TFTP, FTP, and HTTP protocols.

There are three provisioning methods available:

● DHCP Option:

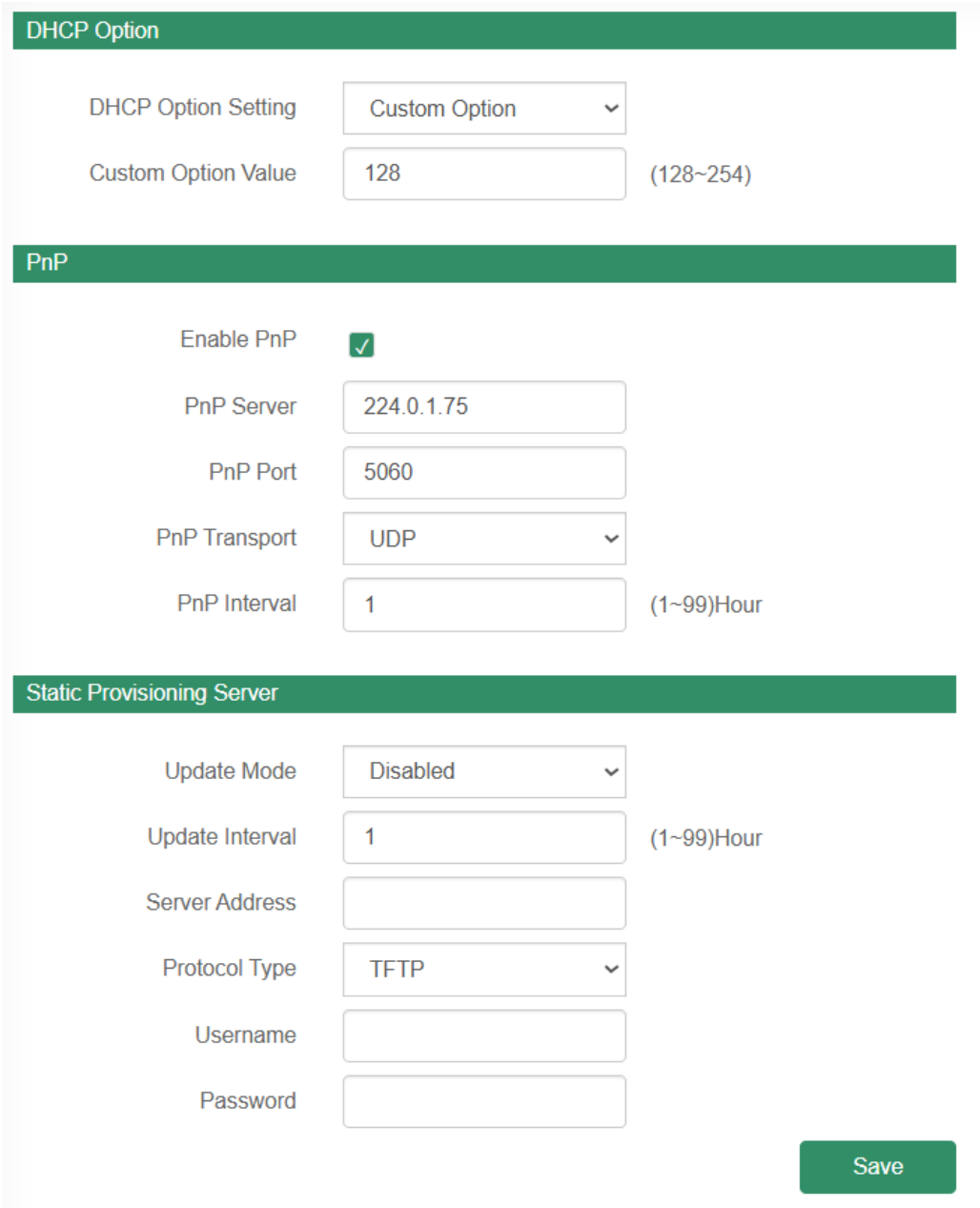
Set up a DHCP server first, then select the option code (Option 66, Option 43, or Custom Option). Configure the path to the provisioning file and run the DHCP server.

● PnP (Plug and Play):

Set up a PnP server, configure the provisioning file path, and run the PnP service for automatic device discovery and configuration.

● Static Provisioning Server:

Supports TFTP, FTP, and HTTP protocols. Prepare the corresponding server, then configure the server address and file path for the provisioning file. If authentication is required, enter the username and password.

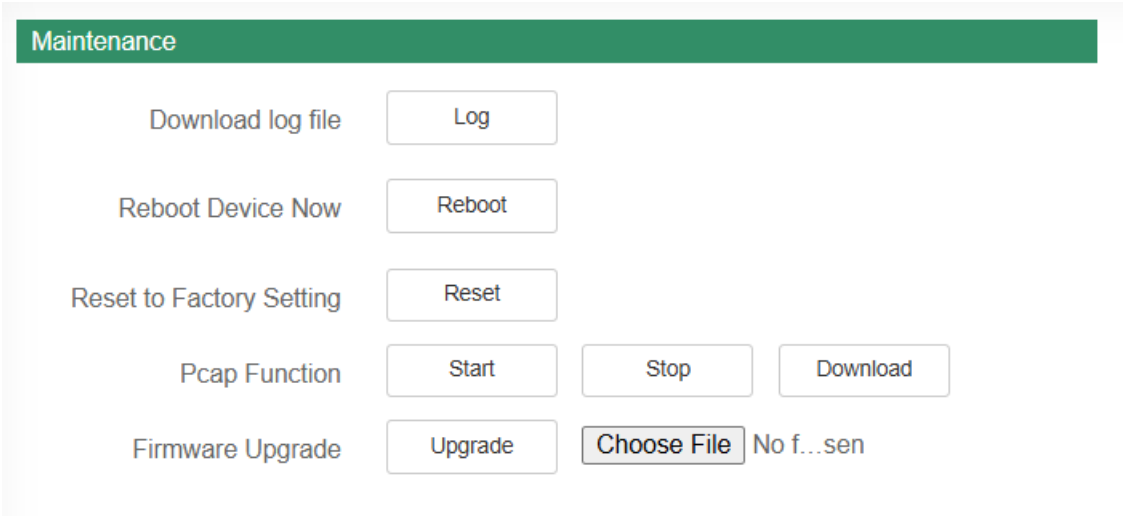


3.13 System

The Maintain section includes Maintenance, Password, and Date/Time settings for device management and system configuration.

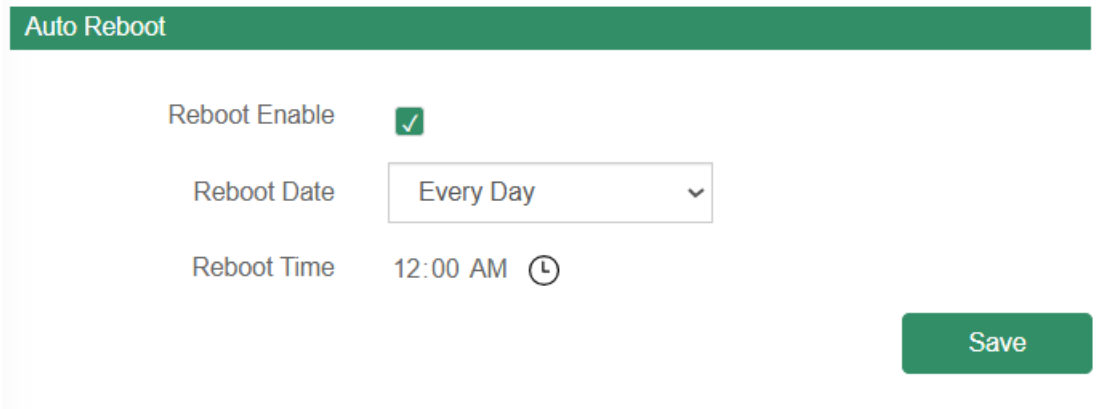
3.13.1 Maintenance

- Download Log File: Click Log to view and download device logs.
- Reboot Device Now: Click Reboot to restart the device.
- Reset to Factory Setting: Click Reset to restore factory default settings.
- Capture: Used for packet capture. Click Start to begin capturing, Stop to end, and Download to export the capture file.
- Firmware Upgrade:
  - Select the latest firmware file (xxx.bin).
  - Click Upgrade. The process takes about 20 seconds.
  - After the upgrade is complete, return to the speaker login web interface.



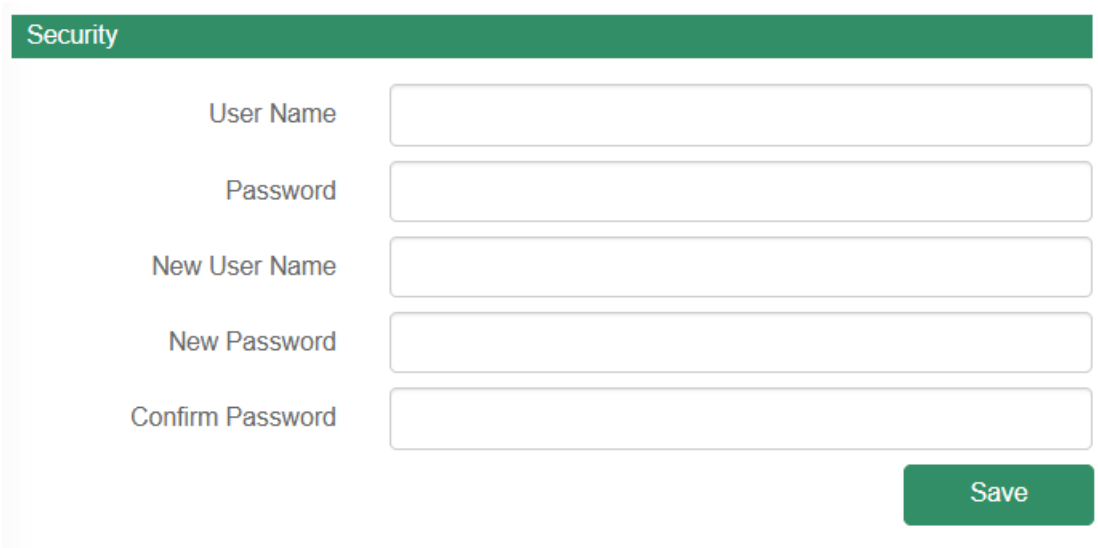
3.13.2 Auto Reboot

Check Reboot Enable to activate Auto Reboot.
Select the reboot date, including Every Day or specific days from Monday to Sunday.
Set the reboot time, then click Save. The IP Speaker will automatically reboot at the scheduled time.



3.13.3 Security

Set a new username and password. Save the configuration and log in again after changes take effect.



3. IPTool Configuration

Besides Web configuration, IPTool provides a quick way to configure basic settings such as SIP account, volume, RTP Multicast, and firmware upgrade.

Please follow the steps below:

Download IPTool from: https://www.tonmind.com/category/downloads/5

Run IPTool, select the correct network from Network, then click Search to discover devices and configure settings.

