



# **M100 Dispatch Microphone Console User Guide**



[www.zycoo.com](http://www.zycoo.com)

[zycoo@zycoo.com](mailto:zycoo@zycoo.com)

© 2023 Zycoo Communications LLC All rights reserved

---

# Contents

<b>1. Preface .....</b>	<b>1</b>
1.1 Audience .....	1
1.2 Revision History .....	1
<b>2. Overview .....</b>	<b>2</b>
2.1 Product Overview .....	2
2.2 Product Specifications .....	2
<b>3. Login the Device .....</b>	<b>3</b>
3.1 Accessing the Web GUI .....	3
3.2 Device Info .....	4
<b>4. SIP Settings .....</b>	<b>7</b>
4.1 SIP Account Settings .....	7
4.2 P2P Account Settings .....	10
4.3 Audio Codecs .....	11
4.4 Advance SIP Settings .....	11
<b>5. Basic Settings .....</b>	<b>14</b>
5.1 Audio Management .....	14
5.2 Buttons Settings .....	16
5.3 Event Scheduler .....	25
5.4 Volume Settings .....	28
5.5 I/O Settings .....	29
5.6 API Settings .....	31
5.7 System Language .....	33
5.8 Ring File .....	33
<b>6. System Settings .....</b>	<b>35</b>
6.1 Network .....	35
6.2 Time .....	36
6.3 Password Settings .....	37
6.4 Upgrade .....	38
6.5 Reboot & Reset .....	39
<b>7. Maintenance .....</b>	<b>40</b>
7.1 Diagnostic .....	40
7.2 Ethernet Capture .....	40

7.3 Import/Export .....	40
7.4 Auto Provisioning .....	41
7.5 Test .....	42
<b>8. Reports .....</b>	<b>44</b>
8.1 Call Logs .....	44
8.2 Operation Logs .....	44

# 1. Preface

## 1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the M100 Dispatch Microphone Console. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the M100 Dispatch Microphone Console.

## 1.2 Revision History


Document Version	Applicable Firmware Version	Update Content	Update Date
2.0.3	2.0.3	Added operating instructions for Firmware version v2.0.3.	2025.3
2.0.1	2.0.1	Added operating instructions for Firmware version v2.0.1.	2023.9.21
1.0.0	1.0.0	Added operating instructions for Firmware version v1.0.0.	2023.7.5

## 2. Overview

### 2.1 Product Overview

The M100 Dispatch Microphone Console is a versatile and high-performance SIP-enabled device for seamless communication. It features 30 programmable fast keys that can be used for paging, intercom, music playback, outbound phone calls, and emergency alarm activation. With multicast and peer-to-peer technology, it serves as a standalone serverless console for individual and group paging, internal calls, and more. The console also allows users to save music and prerecorded messages in the local storage TF card, ensuring reliable and efficient communication capabilities. The M100 Dispatch Microphone Console is ideal for a wide range of industries, including emergency services, schools, transportation, and hospitality, thanks to its advanced features and reliable performance.

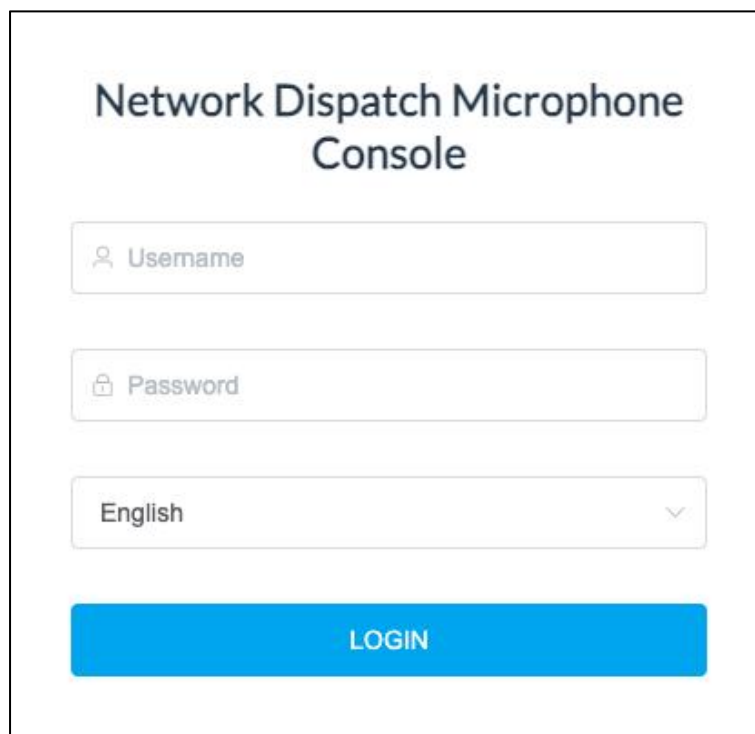
### 2.2 Product Specifications

M100 Dispatch Microphone Console Specifications		
Speaker Components	φ45mm full frequency	
Sensitivity	95±3dB / 1W / 1m	
Max Sound Pressure Level	100dB	
Rated Power:	8Ω 3W	
Microphone Sensitivity	36±2dB	
Microphone Max Sound Pressure Level	110dB	
Microphone Impedance	680Ω	
LCD Display Size	4.3 inches	
LCD Display Resolution	480p*272p	

## 3. Login the Device

### 3.1 Accessing the Web GUI

M100 obtains the IP address through DHCP by default, please ensure that there is an available DHCP server in your LAN (if DHCP fails to obtain an address, it will use a static IP address: 192.168.1.101), press the "RST" button on the back of the device and the IP address will be displayed on the screen, enter the IP address in the browser to access the Web Graphical User Interface.

The image shows a web-based login interface for the 'Network Dispatch Microphone Console'. At the top, the title 'Network Dispatch Microphone Console' is centered. Below the title are three input fields: a 'Username' field with a person icon, a 'Password' field with a lock icon, and a language selection dropdown menu currently set to 'English'. At the bottom of the form is a large blue button labeled 'LOGIN' in white capital letters.

#### Login Interface

After entering the correct username and password, you can log in to the device's web management interface.

**Default username: admin**

**Default password: admin**

For the safety purpose, it is recommended to change the default password on the first login, please go to **System --> Password Settings** page to change the password.

## 3.2 Device Info

After successful login, you will see the information interface of the device, and you can view the basic information of the device.

The screenshot displays the ZYCOO M100 Dispatch Microphone Console web interface. The sidebar on the left contains navigation links: Device Info, SIP Settings, Basic Settings, System, Maintenance, and Reports. The main content area is titled 'Device Info' and is divided into three sections: SIP STATUS, DEVICE INFORMATION, and NETWORK INFORMATION.

**SIP STATUS**

Primary SIP Account	Secondary SIP Account-1	Secondary SIP Account-2	Registered	Idle
1001@192.168.16.109:5060			Registered	Idle
			Unconfigured	
			Unconfigured	

**DEVICE INFORMATION**

Device Model	Hardware Version	Software Version	Uptime	Speaker Volume	Mic Volume	Device Description
M100	Ver2.0	v2.0.3	17 days, 20:54	3 (0-9)	8 (0-9)	M100

**NETWORK INFORMATION**

Mac Address	Connection Mode	IP Address	Subnet Mask	Gateway	Primary DNS	Alternative DNS
68:69:2E:28:00:0E	DHCP	192.168.16.211	255.255.255.0	192.168.16.1	223.5.5.5	223.6.6.6

SIP STATUS				
Primary SIP Account	9000@192.168.11.109:5060		Registered	Idle
Secondary SIP Account-1	103@192.168.11.43:5060		Unactivated	
Secondary SIP Account-2			Unconfigured	

### SIP Status

- SIP Account:** Display the SIP number configured on this device.
- SIP Server:** Display the SIP server (ZYCOO IP Audio Center or IP PBX) address.
- Register Status:** Display the SIP number registration status.

DEVICE INFORMATION	
Device Model	M100
Hardware Version	Ver2.0
Software Version	s2.0.1-dev8
Start Time	2023-09-20 17:33:19
Speaker Volume	8 (0-9) <a href="#">?</a>
Mic Volume	5 (0-9) <a href="#">?</a>
Device Description	M100 <a href="#">?</a>

## Device Information

- **Device Model:** Displays the model of the device.
- **Hardware Version:** Displays the hardware version number of the device.
- **Software Version:** Display the system version number of the device.
- **Start Time:** Displays the last time the device was started up.
- **Speaker Volume:** Displays the current volume of the device.
- **Mic Volume:** Displays the current device microphone input volume.
- **Device Description:** Remark the device information. The description will be displayed in a browser tab. After the Device Description is set, the description will be displayed in the browser tab, which is convenient for distinguishing different terminals when there are many terminal configuration pages.

NETWORK INFORMATION	
Mac Address	68:69:2E:28:00:09
Connection Mode	DHCP
IP Address	192.168.11.232
Subnet Mask	255.255.255.0
Gateway	192.168.11.1
Primary DNS	114.114.114.114
Alternative DNS	None

## Network Information

- **Mac Address:** Display the MAC address of the current device.
- **Connection Mode:** Display the network acquisition method of the device, DHCP (dynamic acquisition) or STATIC (static configuration).
- **IP Address:** The current IP address of the device.



- **Subnet Mask:** The current subnet mask of the device.
- **Gateway:** The gateway address currently used by the device.
- **Primary DNS:** The primary domain name server address used by the device.
- **Alternative DNS:** The secondary domain name server address used by the device.

## 4. SIP Settings

### 4.1 SIP Account Settings

There are three (3) SIP accounts under the SIP Settings, one (1) primary and two (2) secondary for the use of different SIP accounts to proceed with various tasks. If the current device needs to cooperate with the ZYCOO IP Audio Center, please turn on the 'Enable Integration with ZYCOO IP Audio Center' option. After any one of the SIP accounts has successfully registered, the UP/Down page buttons on the M100 will turn on with green light. If all SIP accounts registration failed, the UP/Down page buttons will be shown as red lights. Please go to **SIP Settings --> Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2**

Primary SIP Account

\* SIP Server: 192.168.17.110

\* SIP Port: 

—

 5060 

+

\* User ID: 5001

Auth User: 5001

Domain: 192.168.17.110

Password: \*\*\*\*\* 

👁

\* Register Expiration(Sec): 

—

 180 

+

\* Transport: UDP 

▼

Auto Answer: Yes 

▼

NAT Mode: Disabled 

▼

Enable Integration with  
ZYCOO IP Audio Center:

Activate:

Submit

## Primary SIP Account

Secondary SIP Account-1

\* SIP Server:

192.168.17.83

\* SIP Port:

—

5060

+

\* User ID:

1019

Auth User:

1019

Domain:

192.168.17.83

Password:

\*\*\*\*\*

👁

\* Register Expiration(Sec):

—

180

+

\* Transport:

UDP

▼

Auto Answer:

Yes

▼

NAT Mode:

Disabled

▼

Activate:

☒

Submit

## Secondary SIP 1 Account

Secondary SIP Account-2

\* SIP Server: 192.168.17.23

\* SIP Port:  5060

\* User ID: 100

Auth User: 100

Domain: 192.168.17.23

Password: \*\*

\* Register Expiration(Sec):  180

\* Transport: UDP

Auto Answer: Yes

NAT Mode: Disabled

Activate: ☐

### Secondary SIP 2 Account

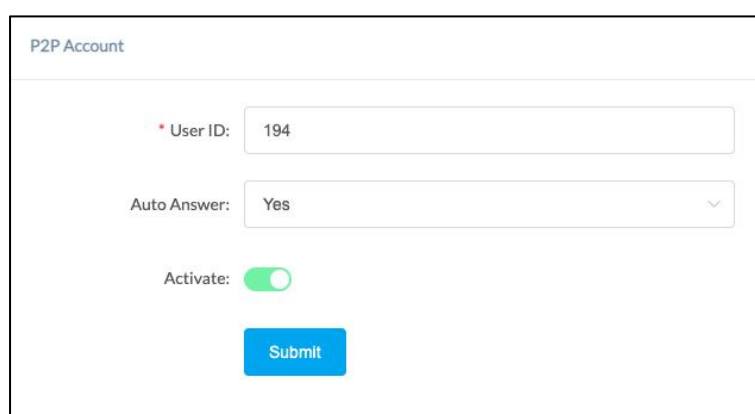
- **SIP Server:** Enter the IP address or domain name of the SIP server.
- **SIP Port:** Default SIP port is 5060. If the SIP server uses another port number as the SIP port, please modify this setting.
- **User ID:** The SIP account number provided by the SIP server.
- **Auth User:** Enter the authorized SIP account's username.
- **Domain:** Enter the SIP Domain.
- **Password:** Authorized SIP account password.
- **Register Expiration (sec):** SIP register expiration time, the default expiration time is 180 seconds.
- **Transport:** Set up the transport protocol, there are UDP, TCP, TLS options to choose.

- **Auto Answer:** Yes/No/Answer Delay, default in the Yes option.
- **NAT Mode:** Select the NAT mode and fill out the corresponding data.  
STUN, TURN, and ICE modes are supported.
- **Enable Integration with ZYCOO IP Audio Center:** This option is disabled by default.  
If you need to connect and use it with ZYCOO IP Audio Center, please enable this option.  
Only the main SIP account has this option.
- **Activate:** Once enabled, the account will be activated and registered to the SIP server.

## 4.2 P2P Account Settings

P2P stands for Peer to Peer. In a P2P network, the peers are connected to each other via the Internet, files can share, or peers can call each other directly between systems on the network without the need for a central server.

After configuring the P2P account, it can be used with the Outgoing Call feature in **Basic Settings --> I/O Settings**, or use the Outgoing API in **Basic Settings ---> API Settings** to make a P2P call.



The screenshot shows a web form titled "P2P Account". It contains three fields: "User ID" with a red asterisk and the value "194", "Auto Answer" with a dropdown menu showing "Yes", and "Activate" with a green toggle switch. A blue "Submit" button is at the bottom.

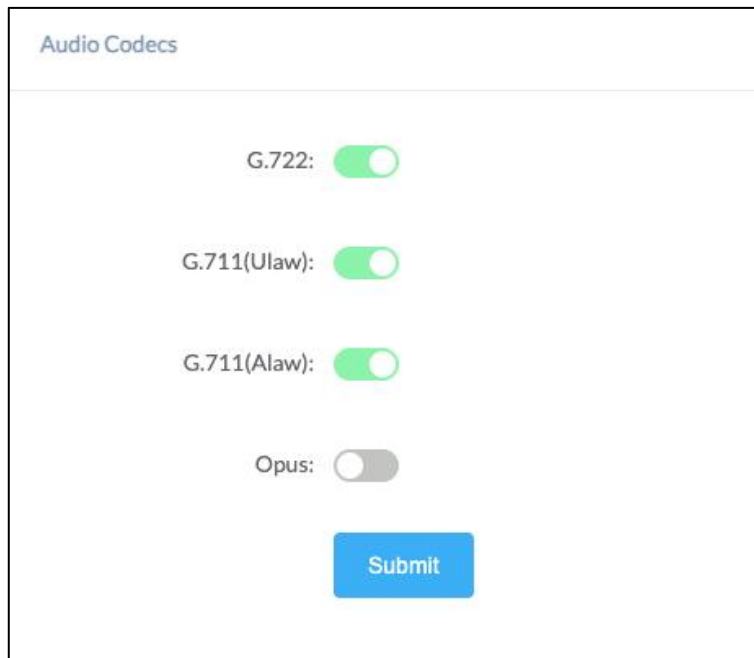
### P2P Account

- **User ID:** The User ID will be displayed as the outgoing number when calling out, or the number that another device needs to dial.
- **Auto Answer:** Yes/No/Answer Delay, default in the Yes option.
- **Activate:** Enable/Disable the P2P feature.

## 4.3 Audio Codecs

M100 supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw), and Opus.

To enable or disable an audio codec/codecs, please go to **SIP Settings --> Audio Codecs** page.



Audio Codecs

G.722: ☒

G.711(Ulaw): ☒

G.711(Alaw): ☒

Opus: ☐

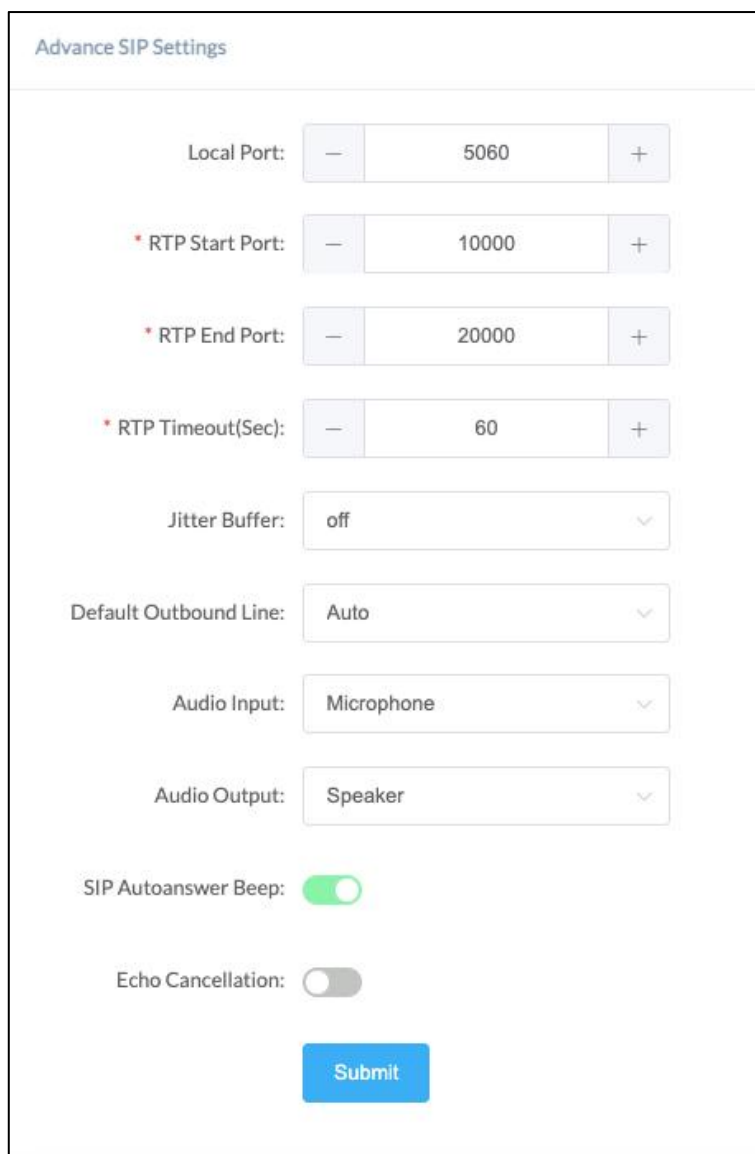
### Audio Codecs

Please keep at least one codec enabled and supported by the SIP server, otherwise, SIP paging will not work.

## 4.4 Advance SIP Settings

Configure some advanced parameters of the SIP protocol.

Please go to **SIP Settings --> Advance SIP Settings**.



Advance SIP Settings

Local Port:

\* RTP Start Port:

\* RTP End Port:

\* RTP Timeout(Sec):

Jitter Buffer:

Default Outbound Line:

Audio Input:

Audio Output:

SIP Autoanswer Beep: ☒

Echo Cancellation: ☐

### Advance SIP Settings

- **Local Port:** This setting represents the port used to receive SIP packets.
- **RTP Start Port:** This setting represents the starting RTP port that will use for media sessions.
- **RTP End Port:** This setting represents the end RTP port that the system will use for media sessions.
- **RTP Timeout (sec):** This setting means that within a specific time range, if the system does not receive the RTP stream, the call will end.
- **Jitter Buffer:** This setting represents the Jitter buffer where voice packets can be collected, stored, and sent to the voice processor in even intervals. Three options are

provided, off/adaptive/fixed. A fixed jitter buffer adds a fixed delay to voice packets. An adaptive jitter buffer can adjust based on the delays in the network.

- **Default Outbound Line:** The default line to use for dialing calls from the dial pad.
- **Incoming Audio Input:** The audio input interface which used for answering incoming calls by default.
- **Audio Output:** The audio output interface which used for incoming and outgoing calls by default.
- **SIP Autoanswer Beep:** Enable/Disable. This setting represents the ringtone beep when a call comes and only applies when the SIP Autoanswer feature is enabled.
- **Echo Cancellation:** Set whether to enable echo cancellation or not.



## 5. Basic Settings

### 5.1 Audio Management

The audio management page is used to manage the music files and playlist of the external TF card, and the audio can be used in the **Audio Source** ---> **Audio Input** in the button settings.

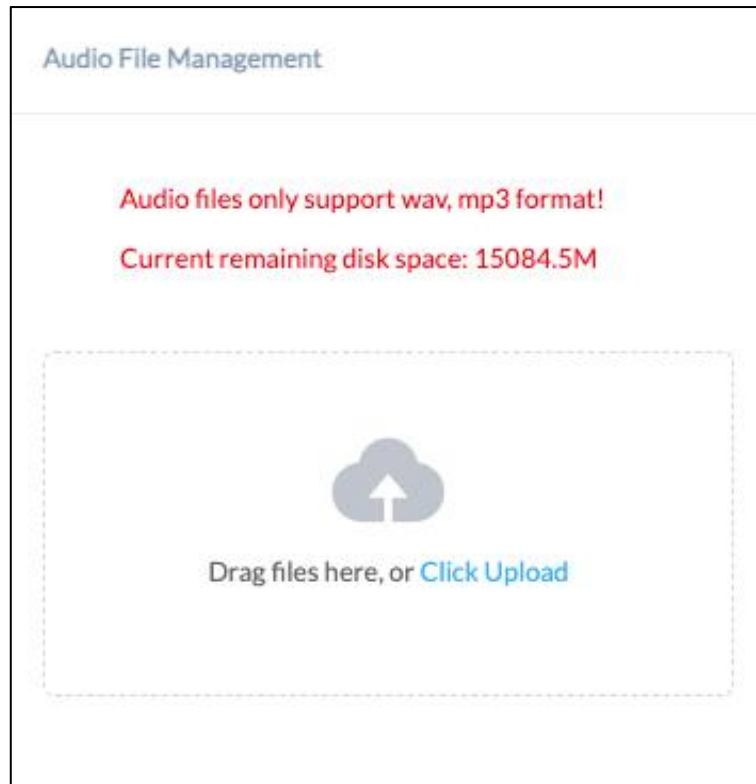
The audio file format only supports wav and mp3 formats.

Click **Basic Settings** ---> **Audio Management** to enter the audio file management page.



#### Audio File Management

Click the "Click Upload" button to select the local audio file to be uploaded. Click the "Delete" button to delete the audio file.



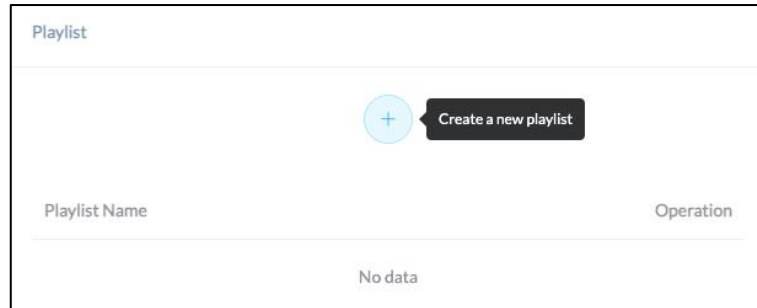
### Upload Audio File

Click **Basic Settings** ---> **Audio Management** and click Playlist to create, check and edit the playlists.



### Playlist

In the playlist interface, click the "+" button to select the required audio files to add, and click "Edit" and "Delete" to edit and delete the created playlist.

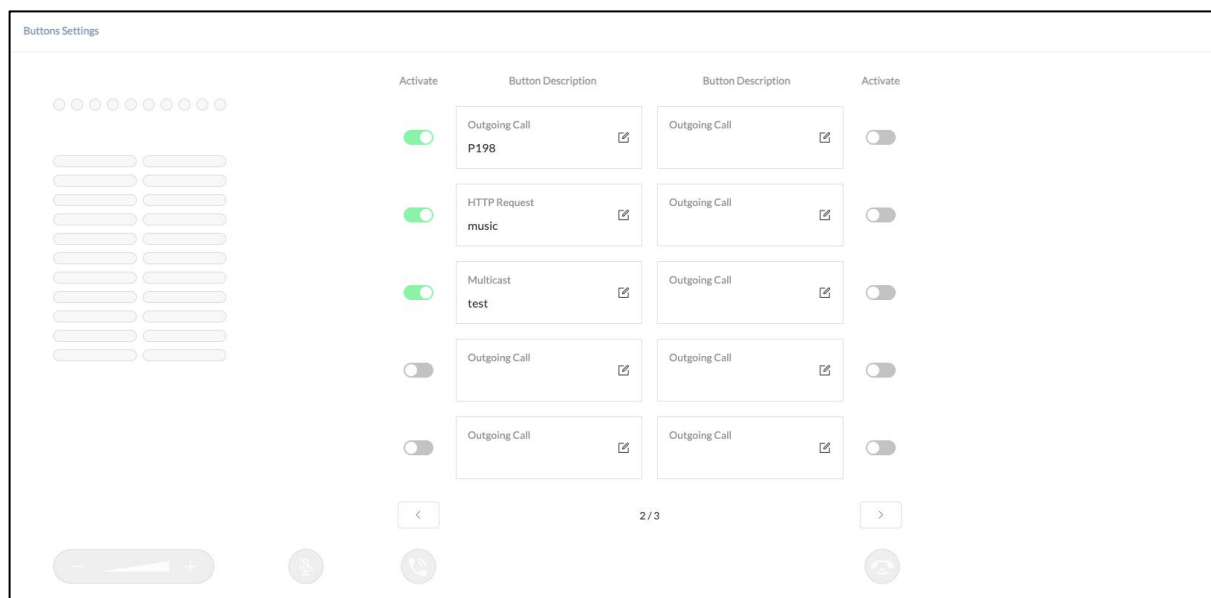


## Create a New Playlist

## 5.2 Buttons Setting

Configure some actions for the buttons on the M100 device, including Outgoing Call, HTTP request, DTMF request, Relay control, and Multicast. The configured buttons will display green lights, and when the buttons are executed, the green lights will turn into red lights. If the button is not configured as the feature key, then it will not light up.

Click **Basic Settings**--->**Buttons Setting** to enter the buttons settings page.



Button Setting

Activate: ☐

\* Name:  ?

Actions:  ^

Audio Source:

\* Destination:

Line:  v

Cancel Submit

### Button Settings

- **Activate:** Activate/deactivate the fast key.
- **Name:** Set the fast key name, which will be displayed on the device screen.
- **Actions:** Set the fast key action. (Including Outgoing Call, HTTP request, DTMF request, Relay control, and Multicast)

Button Setting

Activate: ☐

\* Name:  ⓘ

Actions:

Action Settings

Audio Source:

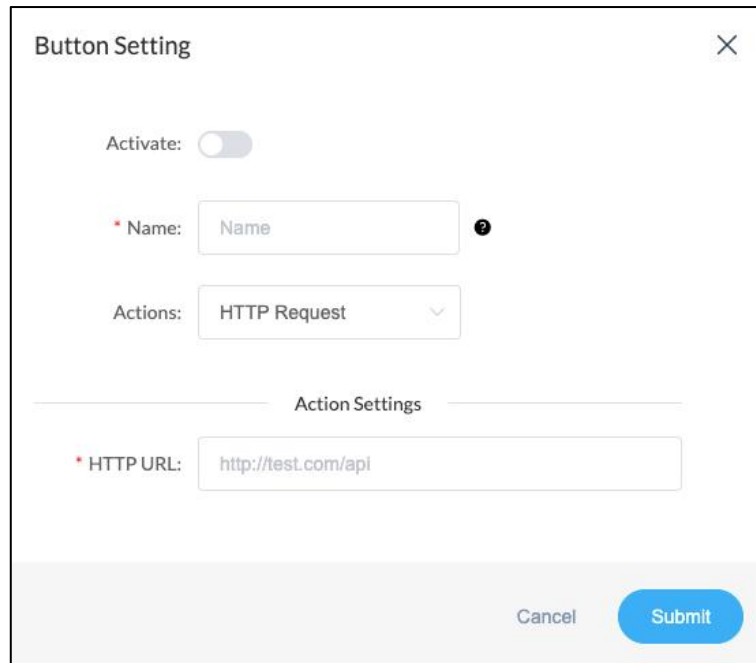
\* Destination:

Line:

Cancel Submit

## Outgoing Call

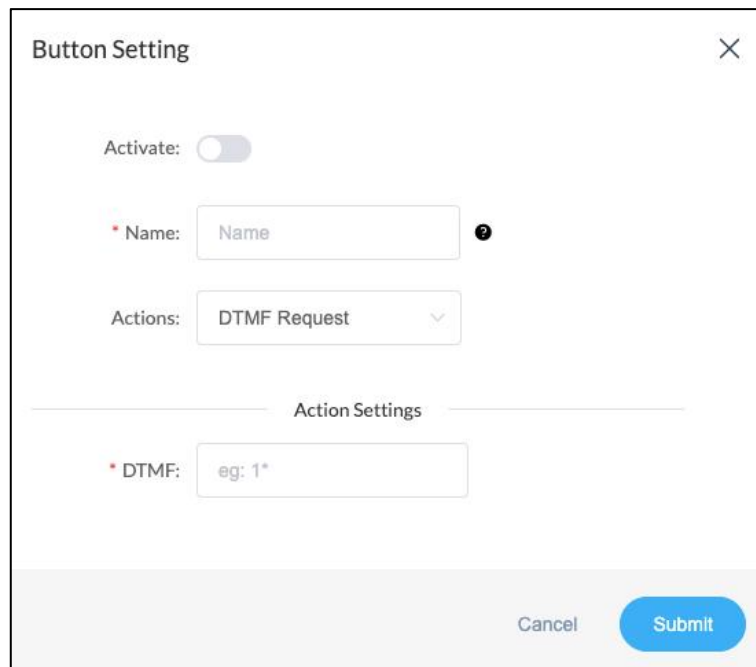
- **Audio Source:** Input settings for the audio source, including Microphone and Audio Input.
- **Destination:** Set the destination number for outgoing calling.
- **Line:** Set the specific line for outgoing calling.



The 'Button Setting' dialog box features a close button (X) in the top right corner. It includes an 'Activate' toggle switch, which is currently turned off. Below this is a required 'Name' field with a placeholder 'Name' and a help icon. The 'Actions' dropdown menu is set to 'HTTP Request'. A section titled 'Action Settings' contains a required 'HTTP URL' field with the placeholder 'http://test.com/api'. At the bottom right, there are 'Cancel' and 'Submit' buttons.

### HTTP Request

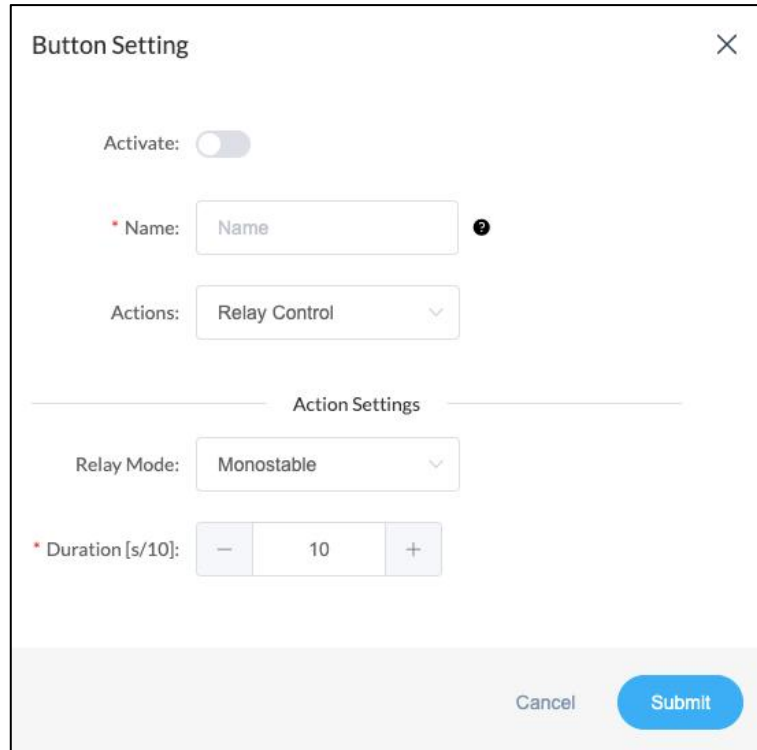
- **HTTP URL:** Set the HTTP request response action. For details, please refer to the "API Settings" instructions.



The 'Button Setting' dialog box features a close button (X) in the top right corner. It includes an 'Activate' toggle switch, which is currently turned off. Below this is a required 'Name' field with a placeholder 'Name' and a help icon. The 'Actions' dropdown menu is set to 'DTMF Request'. A section titled 'Action Settings' contains a required 'DTMF' field with the placeholder 'eg: 1\*'. At the bottom right, there are 'Cancel' and 'Submit' buttons.

### DTMF Request

- **DTMF:** Set the key number to be used when triggering DTMF.



The screenshot shows a 'Button Setting' dialog box with a close button (X) in the top right corner. Inside the dialog, there is an 'Activate' toggle switch which is currently turned off. Below this is a 'Name' field with a red asterisk and a placeholder text 'Name', followed by a help icon (question mark). Underneath is an 'Actions' dropdown menu currently set to 'Relay Control'. A horizontal line separates this from the 'Action Settings' section. In the 'Action Settings' section, there is a 'Relay Mode' dropdown menu set to 'Monostable'. Below that is a 'Duration [s/10]' field with a red asterisk, featuring minus, plus, and reset buttons, with the value '10' displayed. At the bottom right of the dialog are 'Cancel' and 'Submit' buttons.

### Relay Control

- **Relay Mode:** Monostable mode and bistable mode can be selected.
- **Duration:** In monostable mode, the activation duration can be set. (If bistable mode is selected, you need to manually press the button to turn the relay on or off.)

Button Setting

Activate:

\* Name:

Name

Actions:

Multicast

Action Settings

Audio Source:

Microphone

\* Address:

239.168.4.1

\* Port:

-

2

+

Codec:

G.722

Prompt sound:

Cancel

Submit

## Multicast-Microphone



Button Setting

Activate: ☐

\* Name:  ?

Actions:

Action Settings

Audio Source:

\* Address:

\* Port:

Codec:

Prompt sound: ☐

Cancel Submit

### Multicast-Audio Input

- **Audio Source:** Select the audio source as microphone input or external audio input.
- **Address:** Set the multicast address.
- **Port:** Set the multicast port.
- **Codec:** When selecting microphone or audio input as the audio source, the codec that can be selected are MP3, G.722, G.711(Ulaw), and G.711(Alaw).
- **Prompt Sound:** Set whether to enable the prompt sound.

Button Setting

Activate:

\* Name:

Name

?

Actions:

Multicast

Action Settings

Audio Source:

Audio File

Audio File:

alarm\_tone0.wav

Play Times:

-

0

+

?

\* Address:

239.168.4.1

\* Port:

-

2

+

Codec:

g722

Prompt sound:

Cancel

Submit

## Multicast-Audio File

Button Setting

Activate: ☐

\* Name:

Actions:

Action Settings

Audio Source:

Playlist:

Play Times:

\* Address:

\* Port:

Codec:

Prompt sound: ☐

Cancel Submit

### Multicast-Playlist

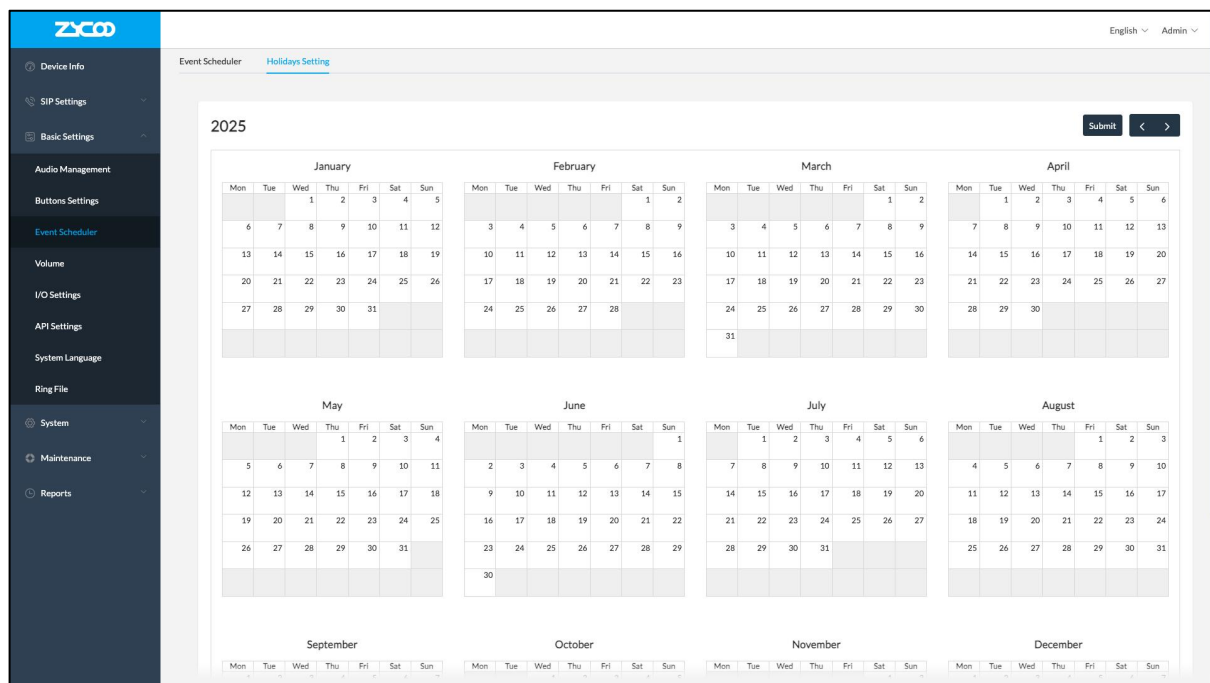
- **Audio Source:** Select audio file input or playlist.
- **Audio File:** Select an audio file to play.
- **Playlist:** Select a playlist created in the audio management interface to play.
- **Play Times:** Set the number of playbacks. When set to "0", it is loop playback.
- **Address:** Set the multicast address.
- **Port:** Set the multicast port.

- **Codec:** When selecting microphone or audio input as the audio source, the codec that can be selected are MP3, G.722, G.711(Ulaw), and G.711(Alaw).
- **Prompt Sound:** Set whether to enable the prompt sound.

## 5.3 Event Scheduler

The event scheduler can edit up to 30 time plans, you can click the corresponding option to edit or delete them. Before you edit/create an event scheduler, please go to the **Basic Settings--->Buttons Setting** page and activate an empty button to use as the corresponding button for the event.

Holiday Setting can set holidays according to needs, and you can choose to enable the holiday exception feature or disable it in the event scheduler.



### Holiday Setting

Event Scheduler				
Activate	ID	Name	Description	
<input type="checkbox"/>	1			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	2			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	3			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	4			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	5			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	6			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	7			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	8			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	9			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	10			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	11			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	12			<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	13			<a href="#">Edit</a> <a href="#">Delete</a>

### Event Scheduler List

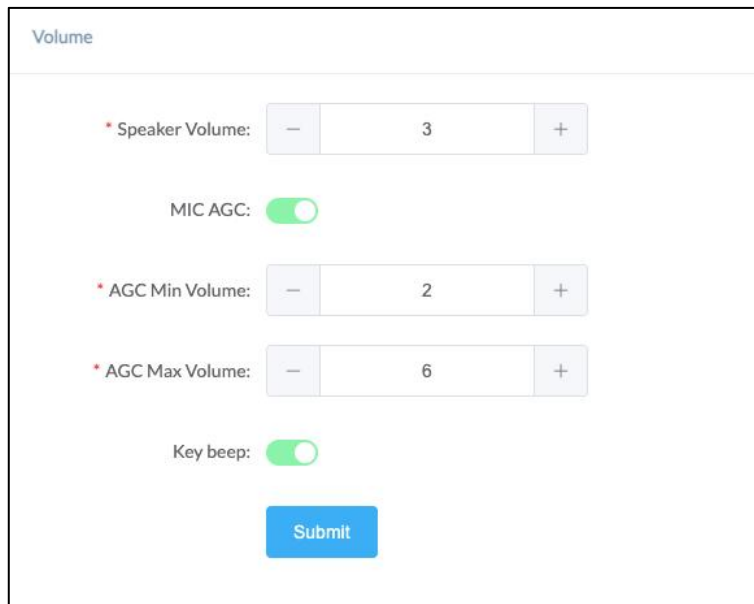
## Time Settings

- **Activate:** Activate/Deactivate the schedule.
- **Name:** Set the name of the schedule.
- **Description:** Comment information for the time schedule.
- **Date Selection:** Set the date range for the time schedule.
- **Weekday:** Set the execution week day in the date range.
- **Holiday Exceptions:** Enable the holiday feature or not.
- **Time Selection:** Set the specific time period for executing the action.
- **Interval(min):** Set the interval time for performing actions.
- **Task Type:** You can choose set Button or Custom tasks.
- **Action (Button):** Select the programmed button that you activated to execute the

scheduled action.

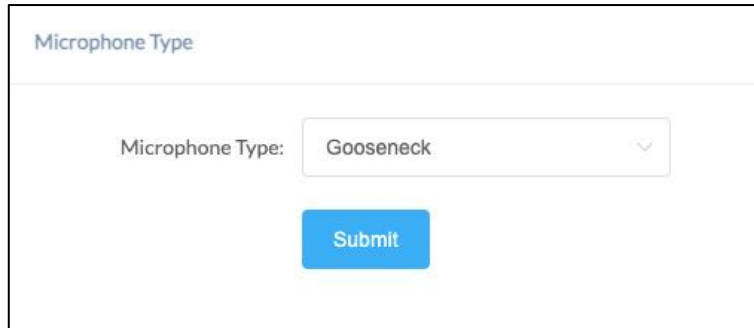
## 5.4 Volume Settings

M100's volume level can be adjusted from its web management interface, on the **Basic Settings --> Volume** page.

The screenshot shows a web interface titled "Volume". It contains five settings: "Speaker Volume" with a value of 3, "MIC AGC" which is a toggle switch turned on, "AGC Min Volume" with a value of 2, "AGC Max Volume" with a value of 6, and "Key beep" which is a toggle switch turned on. At the bottom of the settings is a blue "Submit" button.

### Volume Settings

- **Speaker Volume:** The default speaker volume is 7, adjustable range is 0 ~ 9.
- **MIC AGC:** When this setting is enabled, the system will automatically adjust the microphone volume according to the environment. Users are able to adjust the microphone volume manually when this setting is disabled.
- **AGC Min Volume:** This setting represents the minimum value of the automatic gain control.
- **AGC Max Volume:** This setting represents the maximum value of the automatic gain control.
- **Key Beep:** Enable/Disable the beep sound from the key button.



Microphone Type

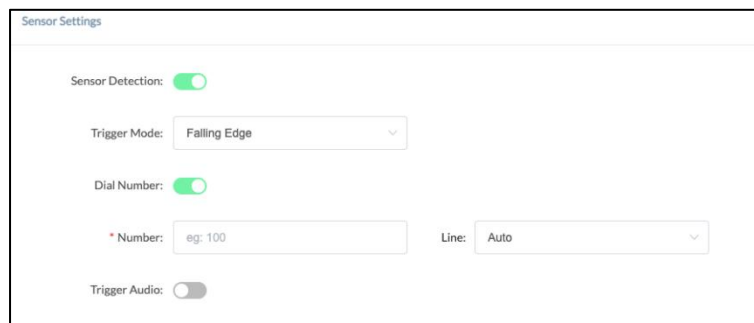
Microphone Type: Gooseneck

Submit

- **Microphone Type:** You can choose the microphone type as gooseneck or handheld.

## 5.5 I/O Settings

This page is used to configure configuration parameters related to security linkage, such as: sensor settings, trigger settings, relay settings and other related configurations. Please go to the **Basic Settings --> I/O Settings** page.



Sensor Settings

Sensor Detection: ☒

Trigger Mode: Falling Edge

Dial Number: ☒

\* Number: eg: 100 Line: Auto

Trigger Audio: ☐

### Sensor Settings

- **Sensor Detection:** Enable/Disable the input detection switch.
- **Trigger Mode:** Select the trigger mode as rising edge trigger or falling edge trigger.
- **Dial Number:** Enable/Disable the trigger dial number switch.
- **Number:** Set the number to dial automatically when triggered and its the corresponding line selection.
- **Trigger Audio:** Enable/Disable the trigger audio switch. (only one of the Trigger Audio and Dial Number options can be enabled)



- **Audio File:** Set the audio file to be played automatically and the number of times to repeat when triggered.

Trigger Setting

Trigger by Input Signal: ☒

Trigger by DTMF Signal: ☒

\* DTMF:

Trigger by Call Status: ☒

Event:

### Trigger Settings

- **Trigger by Input Signal:** Enable/Disable the input trigger switch.
- **Trigger by DTMF Signal:** Enable/Disable the DTMF signal to trigger the change of the output port status (when enabled, it will be triggered after pressing the configured DTMF trigger number during a call).
- **DTMF:** Set the button number to be used when DTMF is triggered.
- **Trigger by Call Status:** Enable/Disable to trigger the output port state change through the call state change.
- **Event:** Set the corresponding call state, you can choose **【Outgoing】** , **【Incoming】** , **【Incoming/Outgoing】** , **【Answered】** and **【Hangup/end】** .

Relay Control

Trigger Type:

Mode:

\* Duration(Sec):

### Relay Control

- **Trigger Type:** This setting represents the responses by the triggers, there are 'On', 'Fast Flashing', and 'Slow Flashing' options to choose from.
- **Mode:** This setting represents the reset mode after the trigger is responded, there are 'Delay Reset' and 'Hang-up Reset' options to choose from.
- **Duration (sec):** This setting is only available if the reply control mode is on Delay Reset, it represents the time duration when the configure interface status changed.

## 5.6 API Settings

This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

Please go to **Basic Settings --> API Settings**.

The screenshot displays the 'API Settings' configuration page. It is divided into two main sections: 'Call Event URL Callback' and 'Relay Event URL Callback'.  
 In the 'Call Event URL Callback' section:  
 - 'Incoming Enable' is a green toggle switch, currently turned on.  
 - 'Incoming Callback URL' is a text input field containing the URL: `http://192.168.11.109/incoming.cgi?ip=${ip}`.  
 - 'Outgoing Enable' is a green toggle switch, currently turned on.  
 - 'Outgoing Callback URL' is a text input field containing the URL: `http://192.168.11.109/outgoing.cgi?ip=${ip}`.  
 - 'Answered Enable' is a grey toggle switch, currently turned off.  
 - 'Hangup Enable' is a grey toggle switch, currently turned off.  
 The 'Relay Event URL Callback' section is partially visible at the bottom:  
 - 'On Enable' is a grey toggle switch, currently turned off.  
 - 'Off Enable' is a grey toggle switch, currently turned off.

### Call Event URL Callback & Relay Event URL Callback

When the call status changes, it will trigger an HTTP GET request to call a URL address.

Within the URL address, you may use variables to identify some current information.

For example:

<code>\${ip}</code> :	The current IP address of the device
<code>\${mac}</code> :	The current MAC address of the device
<code>\${ua}</code> :	The account of the current call
<code>\${number}</code> :	The number of the current call

When the relay status changes, it will trigger an HTTP GET request to call a URL address.

Within the URL address, you may use variables to identify some current information.

For example:

<code>\${ip}</code> :	The current IP address of the device
<code>\${mac}</code> :	The current MAC address of the device

Call API Enable: ☒

Outgoing API: <http://192.168.17.54/api/sipphone?action=call&number=101&line=auto>

Answer API: <http://192.168.17.54/api/sipphone?action=answer>

Hangup API: <http://192.168.17.54/api/sipphone?action=hangup>

Relay API Enable: ☒

On API: <http://192.168.17.54/api/relay?action=on>

Off API: <http://192.168.17.54/api/relay?action=off>

Delay API: <http://192.168.17.54/api/relay?action=on&duration=5>

Play API Enable: ☒

Start Play API: <http://192.168.17.54/api/player?action=start&id=1&repeat=0&volume=7>

Stop Play API: <http://192.168.17.54/api/player?action=stop>

## API Settings

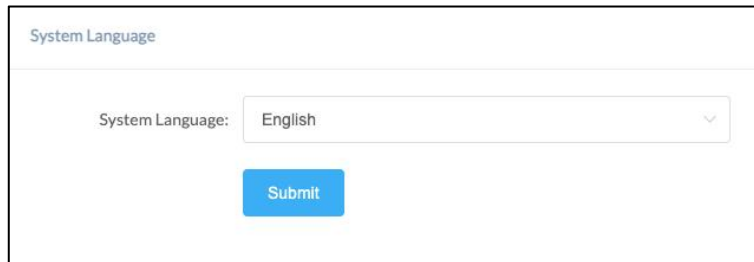
Using the API interface to realize features such as device linkage, call control, relay control, and play sound by the systems.

*Note: Authentication and encryption are not used in the API interface, so please pay attention to the security of the network environment when opening and using these API interfaces.*

## 5.7 System Language

The language of local voice prompts, like IP address announcements, can be set on **Basic Settings --> System Language** page.

Currently, only Chinese and English are provided.

A screenshot of a web interface titled "System Language". It features a label "System Language:" followed by a dropdown menu currently showing "English". Below the dropdown is a blue "Submit" button.

**System Language**

## 5.8 Ring File

The Ring File section allows users to upload 4.6M audio files to the endpoint and use them as ringtones or play API audio files. Click **Basic Settings --> Ring File** to enter the custom ringtone management page.






Click the "Select Audio File" button to select the local audio file to be uploaded, and then click the "Upload" button to upload.

Click "Play" to play the audio file and test the audio. Click "Delete" to delete the audio file.

Ring File Upload

Ring files only accept wav format!

Current disk space remaining: 4.6M

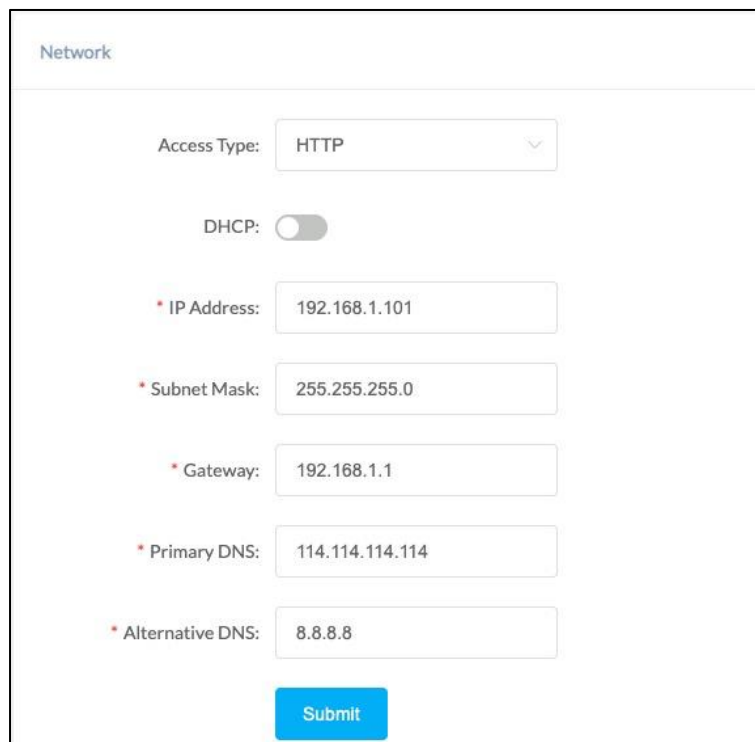
Custom audio file 1	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 2	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 3	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 4	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 5	Currently set to default	Select audio file 	Upload	Play	Delete

### Ring File

## 6. System Settings

### 6.1 Network

M100 uses DHCP to dynamically obtain IP addresses by default. To change the IP assignment from DHCP to Static IP, please go to **Settings --> Network** page. Turn the DHCP switch button off to show the network parameter settings.



Network

Access Type: HTTP

DHCP: ☐

\* IP Address: 192.168.1.101

\* Subnet Mask: 255.255.255.0

\* Gateway: 192.168.1.1

\* Primary DNS: 114.114.114.114

\* Alternative DNS: 8.8.8.8

Submit

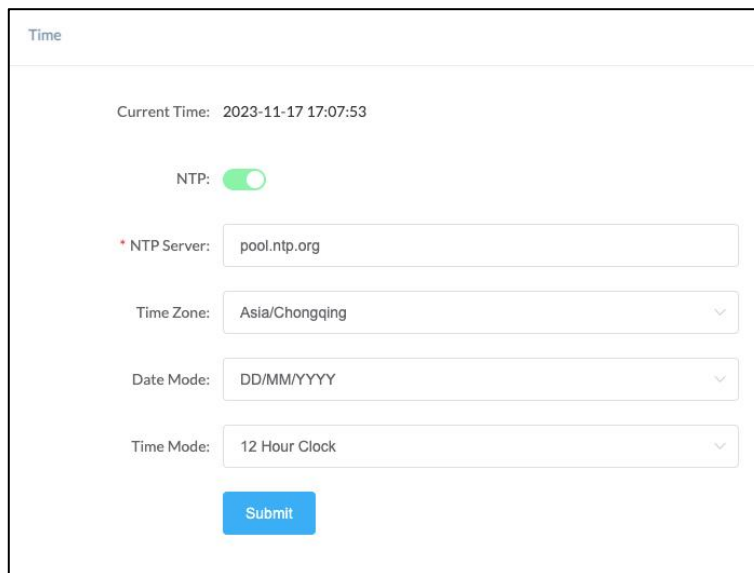
#### Network Configuration

- **Access Type:** Specify the access method of the website, which currently supports HTTP and HTTPS.
- **IP Address:** Enter a vacant IP address within your LAN.
- **Subnet Mask:** Enter the subnet mask of your LAN.
- **Gateway:** Enter the default gateway of your LAN, this is essential for the device when the IP Audio Center or other SIP server is installed outside the LAN.
- **Primary DNS:** Enter an effective primary DNS server address.

- **Alternative DNS:** Enter an alternative DNS server address, when the primary DNS fails, alternative DNS will be used.

## 6.2 Time

M100 obtains the time from the network time servers using NTP, to change the NTP settings please go to **Settings --> Time** page.



Time

Current Time: 2023-11-17 17:07:53

NTP: ☒

\* NTP Server:

Time Zone:

Date Mode:

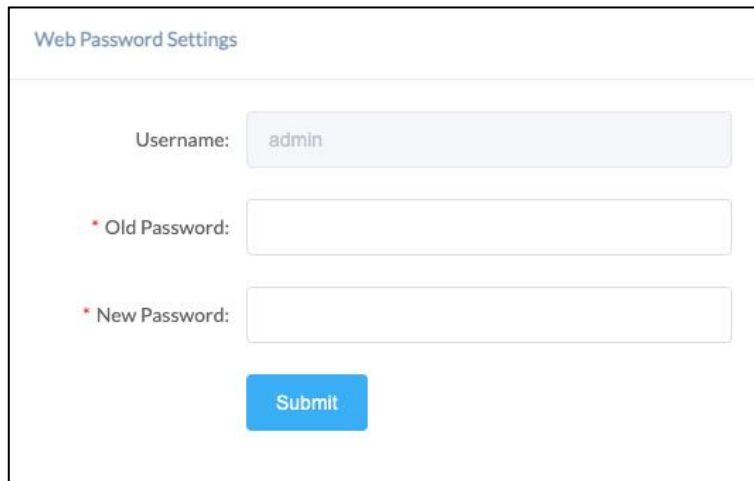
Time Mode:

### Time Settings

- **Activate:** Activate/deactivate the fast key.
- **Name:** Set the fast key name, which will be displayed on the device screen.
- **Current Time:** Display the current system time of the device.
- **NTP:** Enable/Disable using NTP to obtain the time.
- **NTP Server:** The network time server used to obtain the time.
- **Time Zone:** Set the time zone used by the device.

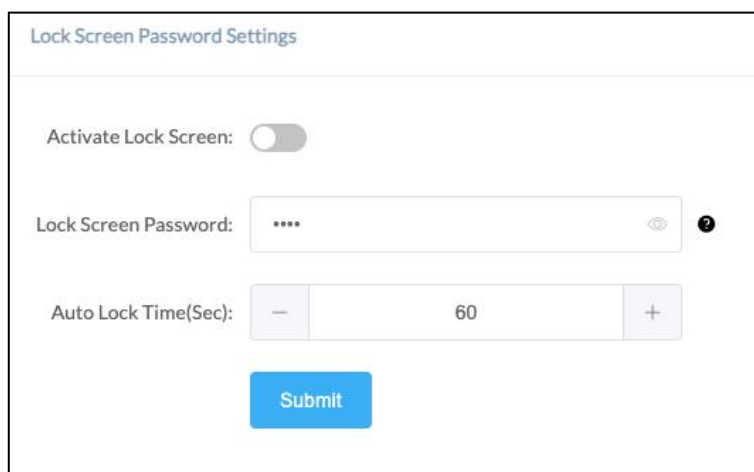
## 6.3 Password Settings

Change the login password of the web management interface and the lock screen password of the device. Please go to **Settings --> Password Settings**.

A screenshot of the 'Web Password Settings' form. It has a title bar 'Web Password Settings'. Below it, there are three input fields: 'Username:' with the value 'admin', '\* Old Password:', and '\* New Password:'. Each field has a red asterisk icon to its left. At the bottom of the form is a blue 'Submit' button.

### Web Password Settings

- **Old Password:** This setting represents the current user password.
- **New Password:** This setting represents the new password user would like to set up.

A screenshot of the 'Lock Screen Password Settings' form. It has a title bar 'Lock Screen Password Settings'. Below it, there are three settings: 'Activate Lock Screen:' with a toggle switch, 'Lock Screen Password:' with a masked input field (four dots) and an eye icon, and 'Auto Lock Time(Sec):' with a numeric input field showing '60' and minus/plus buttons. At the bottom is a blue 'Submit' button.

### Lock Screen Password Settings

- **Activate Lock Screen:** Turn the lock screen function on or off.



- **Lock Screen Password:** Set the lock screen password, the lock screen password is 4 digits, which can be numbers, asterisks and pound signs.
- **Auto Lock Time:** Set the auto-lock time when there is no operation in standby.

## 6.4 Upgrade

To upgrade the device's firmware, please go to **System --> Upgrade** page.

Current Firmware Info

Current Firmware Version: **s2.0.1**


Last Update: 2023-10-18

Upgrade

Warning: It will take around 250 seconds to complete the upgrade process, during the upgrade process please DO NOT power-off the device!

Reset Factory Defaults: ☐

Firmware:



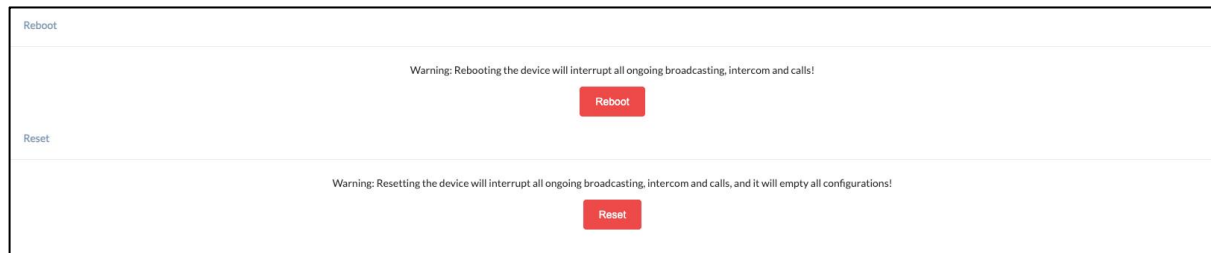
Drag files here, or [Click Upload](#)

### Upgrade Settings

- **Current Firmware Version:** Displays the version currently used by the system.
- **Last Update:** Displays the last system updating time.
- **Reset Factory Defaults:** Specify whether to restore factory settings when upgrading.
- **Firmware:** Click to select the firmware that needs to be used to upgrade the current device.

## 6.5 Reboot & Reset

M100 can be rebooted and reset from the web management interface on the **System --> Reboot & Reset** page.



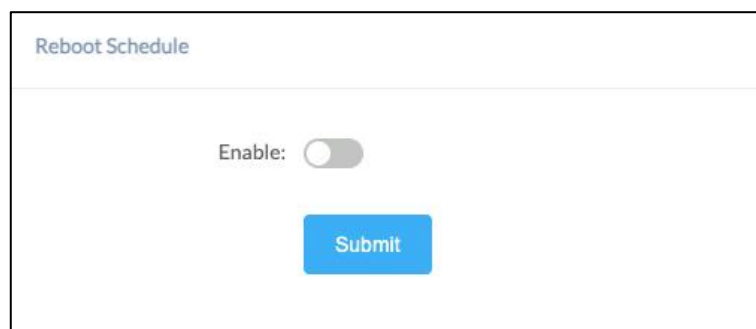
The screenshot shows a web interface with two sections. The top section is titled 'Reboot' and contains a warning message: 'Warning: Rebooting the device will interrupt all ongoing broadcasting, intercom and calls!'. Below the warning is a red button labeled 'Reboot'. The bottom section is titled 'Reset' and contains a warning message: 'Warning: Resetting the device will interrupt all ongoing broadcasting, intercom and calls, and it will empty all configurations!'. Below the warning is a red button labeled 'Reset'.

### Reboot & Reset Settings

Users can restart the device without power failure on this page. The restart process takes about 10 seconds. After the restart is complete, refresh the page to log in again.

If you need to restore the factory settings of the M100, you can reset it through this page or you can press and hold the RST button for more than 10 seconds and release it. After hearing the broadcast voice, the device will enter the state of restoration. The key will flash once. After restarting, the pop-up window disappears, and the device is restored successfully.

*Note: Restoring factory settings will erase all user settings, please operate with caution!*



The screenshot shows a web interface titled 'Reboot Schedule'. It features a toggle switch labeled 'Enable:' which is currently turned off. Below the toggle switch is a blue button labeled 'Submit'.

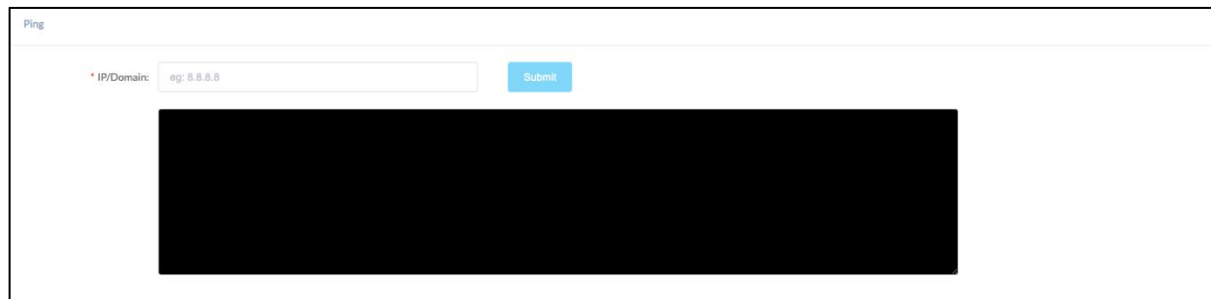
### Reboot Schedule

When the Reboot Schedule feature is Enabled, you can set up the automatic reboot daily, weekly, or monthly at a specified time.

## 7. Maintenance

### 7.1 Diagnostic

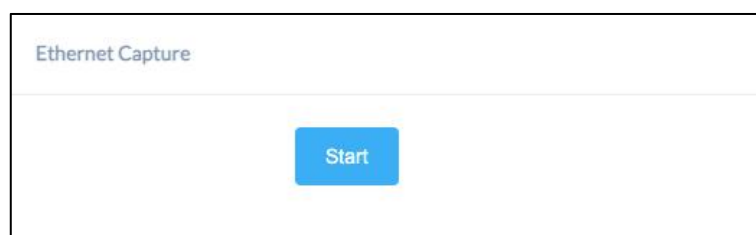
Ping is a network administration utility or tool used to test connectivity on an IP network. Input other devices' IP addresses and click on the submit button to trace the network route.



The screenshot shows a web interface titled "Ping". It features a text input field labeled "IP/Domain:" with the placeholder text "eg: 8.8.8.8". To the right of the input field is a blue button labeled "Submit". Below the input field is a large black rectangular area, likely intended for displaying the results of the ping test.

### 7.2 Ethernet Capture

The purpose of the Ethernet capture tool is to capture Ethernet network packets and store them in a standard Wireshark-compatible packet capture '.pcap' file for immediate viewing and data analysis.



The screenshot shows a web interface titled "Ethernet Capture". It features a large blue button labeled "Start" centered on the page.

### 7.3 Import/Export

This page is used to import and export the current configuration of the device, and you may use this configuration file to backup and/or recover. Both YAML and JSON formats are supported.

The screenshot displays a web interface for configuring the M100 Dispatch Microphone Console. It is divided into two main sections: 'Export' and 'Import'.

**Export Section:** Contains two blue buttons labeled 'Export YAML' and 'Export JSON'.

**Import Section:** Features a red warning message: 'Warning: Only yaml/json configuration file format is permitted. When importing a new configuration file, all ongoing broadcasts and/or calls will be interrupted. The device will restart with the new configuration after the process is done.' Below the warning is a dashed rectangular box containing a cloud upload icon and the text 'Drag files here, or [Click Upload](#)'.

## 7.4 Auto Provisioning

The system is supporting DHCP Option 066 and static TFTP/HTTP two auto provisioning methods.

When the system starts by default and the network mode is in DHCP, it will try to grab option 066 from the DHCP data as the TFTP server address. If the system couldn't get the option information, it will use the below Static Provisioning Server data to obtain the configuration file. When the system starts, and the network mode is in Static, it will use the below Static Provisioning Server data to directly obtain the configuration file.

The configuration file name's format rules:

- 1) all letters in the server MAC address need to be uppercase.
- 2) all colons ":" need to be removed. For example, 68692E290012.

DHCP Provisioning Server

When the system start by default and the network mode is in DHCP, it will try to grab option 066 from the DHCP data as the TFTP server address. If the system couldn't get the option information, it will use the below Static Provisioning Server data to obtain the configuration file. When the system starts, and the network mode is in Static, it will use the below Static Provisioning Server data to directly obtain the configuration file.

The configuration file name's format rules:

- 1) all letters in the server MAC address need to be uppercase
- 2) all colons ":" need to be removed. For example, 68692E290012

Static Provisioning Server

Access Mode: TFTP

TFTP Server Address: 10.10.1.5

Configuration Format: JSON

Configuration Filename: \$mac.json

Update Mode: Update after reboot

Submit

## 7.5 Test

The detection feature provides an option for the user to check whether the speaker, microphone and relay will work functionally before registering it to the server.

The screenshot displays a 'Test Settings' interface with three distinct test sections, each separated by a horizontal grey bar. The first section, 'Speaker Test', contains a blue 'Start Test' button. The second section, 'Microphone Loop Test', also contains a blue 'Start Test' button. The third section, 'Relay Test', contains a blue 'Test' button. Each button is highlighted with a yellow glow effect.

### Test Settings

- **Speaker Test:** Click on the Start Test button, and the speaker will play a ringtone to test whether the speaker is working. If the speaker is working functionally, you should hear the voice back.
- **Microphone Loop Test:** Click on the Start Test button, then start speaking to the device.
- **Relay Test:** Click on the Test button and the device will output signals to the relay for testing.

## 8. Reports

### 8.1 Call Logs

This page shows the call log records of incoming and outgoing calls of the M100 device.

Date	Time	Account	Telephone Number	Duration	Type	Status
2023-11-13	18:43:22	1001	1006	00:00:10	Outbound	Answered
2023-11-13	18:41:55	1001	1003	00:00:03	Outbound	Answered
2023-11-13	18:41:29	1001	6005	00:00:08	Outbound	Answered
2023-11-13	18:37:42	1001	9003	00:00:50	Inbound	Answered
2023-11-13	18:37:38	1001	9003	00:00:00	Inbound	Missed Call
2023-11-13	18:37:04	1001	9003	00:00:28	Inbound	Answered
2023-11-10	17:53:29	1001	1003	00:00:13	Inbound	Answered
2023-11-08	14:51:29	1001	9003	00:00:00	Inbound	Missed Call
2023-11-08	14:50:43	1001	9003	00:00:00	Inbound	Missed Call
2023-11-08	14:50:34	1001	9003	00:00:00	Inbound	Missed Call
2023-11-08	14:50:33	1001	9003	00:00:00	Inbound	Missed Call
2023-11-08	14:50:30	1001	9003	00:00:00	Inbound	Missed Call
2023-11-08	14:50:28	1001	9003	00:00:00	Inbound	Missed Call
2023-11-08	14:50:23	1001	9003	00:00:00	Inbound	Missed Call
2023-11-07	17:04:09	1001	1003	00:00:45	Outbound	Answered
2023-11-07	17:03:27	1001	1006	00:00:34	Outbound	Answered

#### Call Log

- **Date:** Shows the date the call occurred.
- **Time:** Shows the specific time when the call occurred.
- **Account:** Shows the line account used for calling/answering.
- **Telephone Number:** Shows the phone number or address of the caller.
- **Duration:** Shows the call duration.
- **Type:** Shows the call type, incoming or outgoing call.
- **Status:** Shows call status, answered or missed.

### 8.2 Operation Logs

This page shows various operations and execution logs of the M100 device.

Time	Type	Event	Action
2023-11-13 18:46:05	FUNCTION	http	HTTP request: http://192.168.11.99/api/player?action=stop
2023-11-13 18:46:04	BUTTON	9	Click
2023-11-13 18:45:58	BUTTON	end	Click
2023-11-13 18:45:43	FUNCTION	http	HTTP request: http://192.168.11.99/api/player?action=start&id=4&repeat=66&volume=5
2023-11-13 18:45:43	BUTTON	8	Click
2023-11-13 18:45:21	FUNCTION	multicast	Stop
2023-11-13 18:45:20	BUTTON	4	Click
2023-11-13 18:45:17	FUNCTION	multicast	Multicast: mic <239.168.12.102:2000>
2023-11-13 18:45:12	BUTTON	4	Click
2023-11-13 18:44:57	FUNCTION	multicast	Stop
2023-11-13 18:44:56	BUTTON	4	Click
2023-11-13 18:44:53	FUNCTION	multicast	Multicast: mic <239.168.12.102:2000>
2023-11-13 18:44:49	BUTTON	4	Click
2023-11-13 18:44:38	FUNCTION	multicast	Stop
2023-11-13 18:44:38	BUTTON	4	Click

### Operation Log

- **Time:** Shows the specific operation time.
- **Type:** Shows operation type, BUTTON, FUNCTION, SCHEDULE.
- **Event:** Shows events type, such as the key value, function name, etc.
- **Action:** Shows action content, such as clicking buttons, dialing numbers, playing music, etc.





[www.zycoo.com](http://www.zycoo.com)

[zycoo@zycoo.com](mailto:zycoo@zycoo.com)

© 2023 Zycoo Communications LLC All rights reserved

---