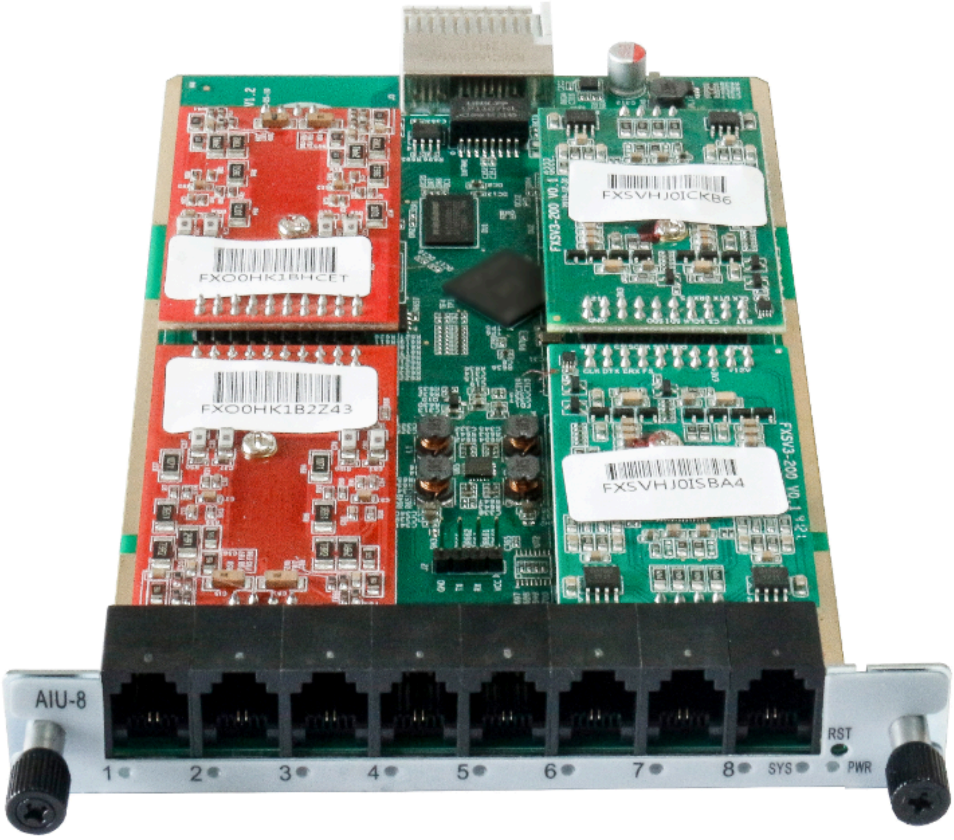
AIU-8 Analog Gateway User Manual



Version 2.0

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Revision History

Version	Release Date	Description
2.0	25/5/2026	Full update

Document Information

- Product: AIU-8
- Document Type: User Manual
- Interface Type: Three options: 8 FXS, 8 FXS, 4 FXS and 4 FXO

Product Overview

The AIU-8 is an analog VoIP gateway designed for SMB and SOHO scenarios. It provides 4 FXS ports and 4 FXO ports, enabling stable interconnection between IPPBX systems, fax machines, analog phones, and carrier networks, and supports concurrent voice and fax processing.

1. Overview

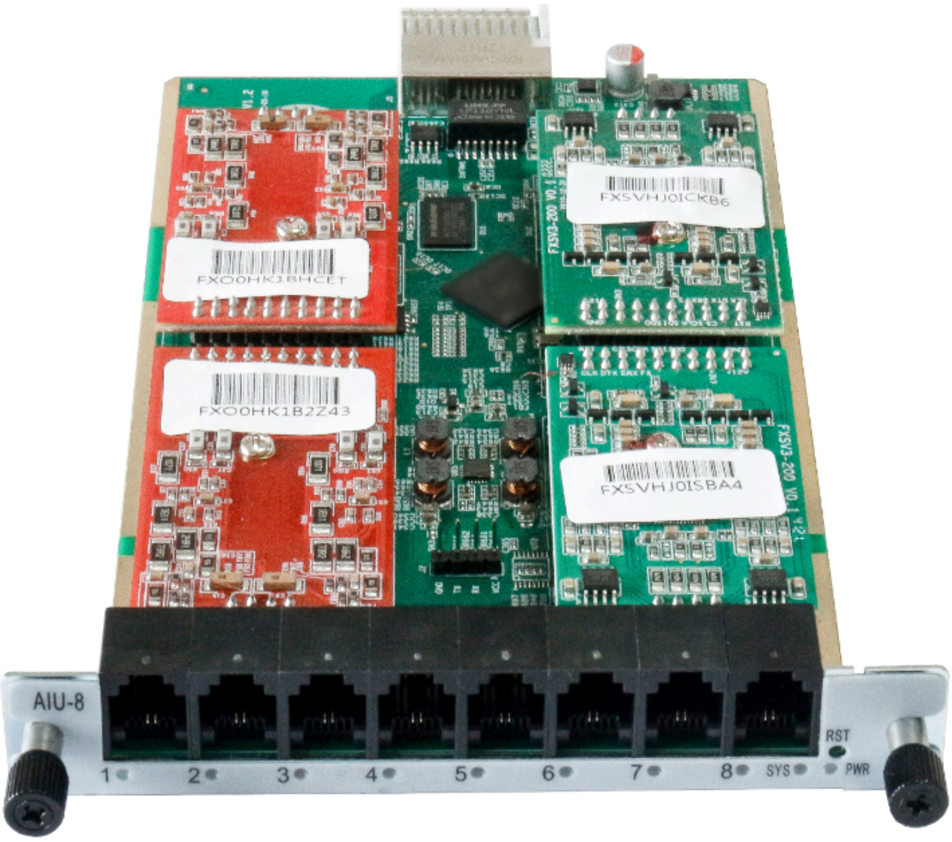
1.1 Product Introduction

AIU-8 is a multi-functional analog gateway that provides three options: 8 FXS, 8 FXS, 4 FXS and 4 FXO, enabling seamless connection between IP PBX systems, fax machines, analog phones, and carriers. It also provides excellent concurrent voice/fax processing capability, strong performance, and high stability, delivering high-quality call services for carriers, enterprises, call centers, and residential users in communities.

The AIU-8(4S4O) analog gateway is a cost-effective product in the iAG series and is an ideal analog VoIP gateway solution for SMB and SOHO environments. It provides a user-friendly interface and a distinctive design, allowing users to customize and configure the gateway easily. A complete API document is also available.

The AIU-8(4S4O) analog gateway is designed for interconnection with multiple codecs, including G.711A/U, G.723.1, G.729A, G.722, iLBC, OPUS, AMR, and AMR-WB. It uses the standard SIP protocol and is compatible with leading VoIP platforms, IPPBX systems, and SIP servers such as Asterisk, Issabel, 3CX, FreeSWITCH, BroadSoft, and VOS VoIP operating platforms.

1.2 Product Appearance



1.3 Software Features

	AIU-8
Phone Ports	Three optional configurations: 8 FXS ports, 8 FXO ports, 4 FXS ports, 4 FXO ports
Accounts and Templates	4 templates, each S port provides a SIP account
Voice Codecs	G.711A/U, G.723.1, G.729A, G.722, iLBC, OPUS, AMR, and AMR-WB
Fax	T.38 compliant Group 3 fax relay up to 14.4 kbps, with automatic fallback to G.711 fax transmission. T.38 fax relay uses V.17, V.21, V.27ter, and V.29 fax data pumps.
QoS	DiffServ, ToS, and 802.1P/Q VLAN tagging
Telephony Features	Caller ID display or blocking, call waiting, blind transfer and attended transfer, call forwarding, DND, callback, paging, message waiting indicator and interval prompt tone, auto dialing, and flexible dialing rules
DTMF Mode	Flexible DTMF transmission modes, user audio interface, RFC2833 and/or SIP Info
SIP Signaling	SIP (RFC 3261) over UDP/TCP/TLS
Security	SRTP/TLS/SIPS, HTTPS, 802.1x
Upgrade and Provisioning	TFTP, HTTP, HTTPS

	AIU-8
Network Protocols	TCP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS, DHCP, NTP, TFTP, PPPoE, STUN

1.4 Hardware Features

Parameter	AIU-8
Port Type	RJ11
Weight	189 g
Dimensions	160*100 mm
Maximum Power	12 W
Operating Temperature	0°C to 50°C
Storage Humidity	10% to 90%, non-condensing
Storage Temperature	-20°C to 70°C
Certification	CE

1.5 Software Access

Default IP: 192.168.6.65

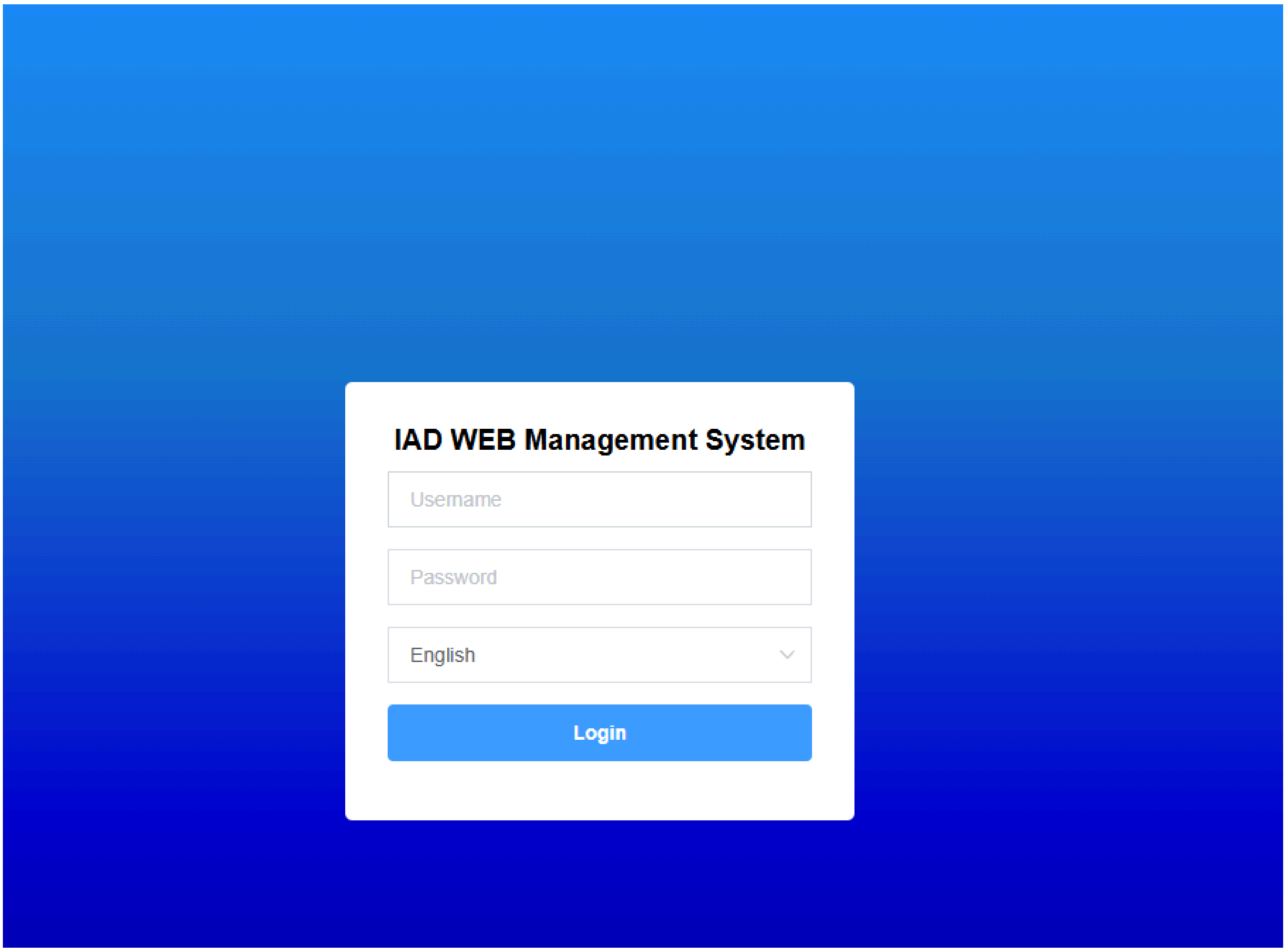
Username: admin

Password: admin

Connect the network cable to LAN1/LAN2, enter the default IP address in a browser, and log in to the gateway for configuration.

Note: The default network mode of this product is bridge mode. Whether the network cable is connected to the WAN port or the LAN port, the IP address is the same.

Figure 1-6-1 Login Interface

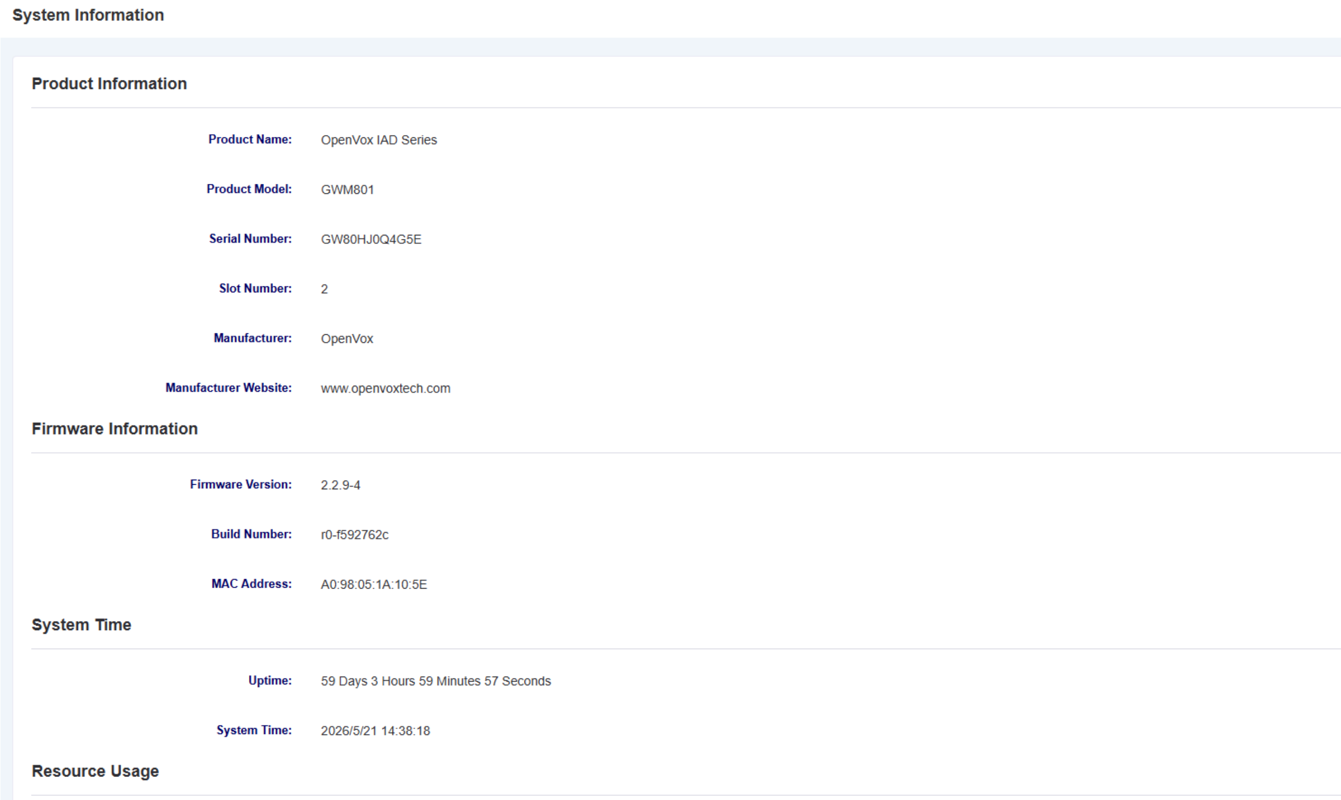


2. Status

2.1 System Status

The System Status page displays product information, firmware information, system time, and resource usage.

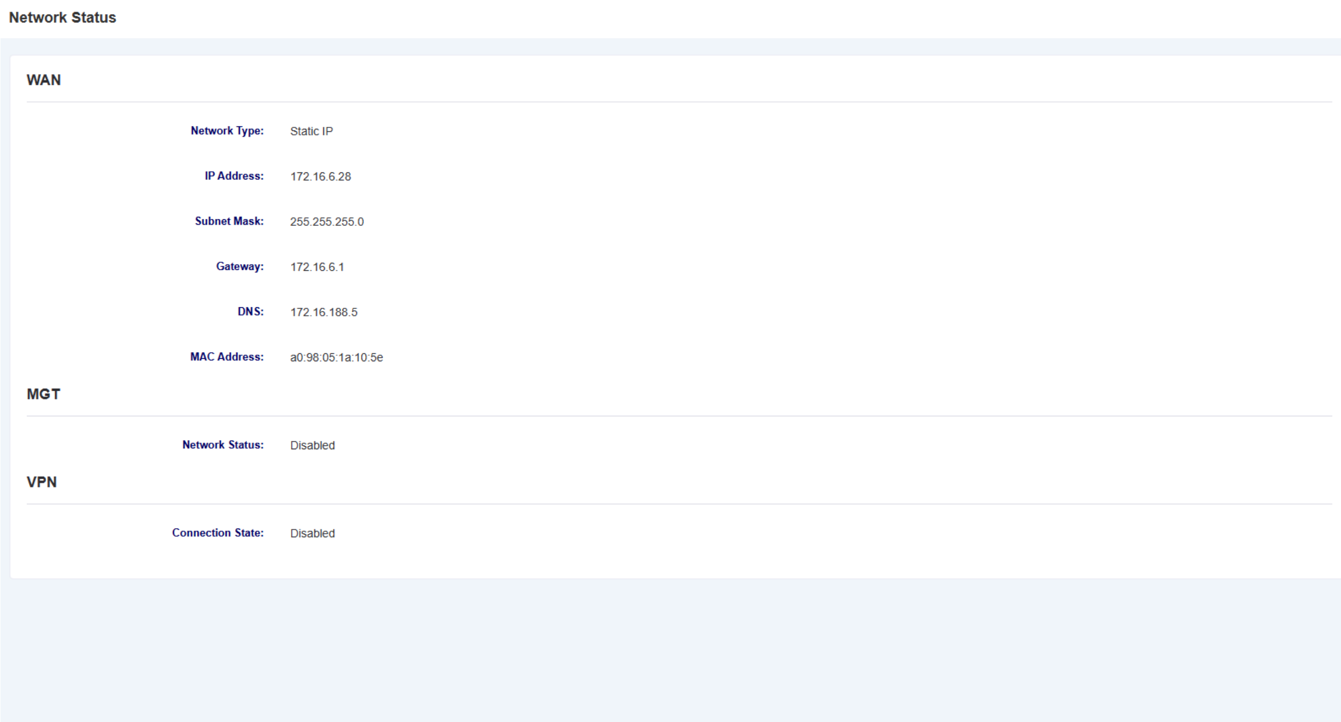
Figure 2-1-1 System Status Display



2.2 Network Status

The Network Status page displays network status and VPN connection status.

Figure 2-2-1 Network Status



2.3 Port Status

The Port Status page displays port type, enable status, registration status, and hook status. Click the slot number drop-down list to switch between interface boards. You can check the status of FXO or FXS according to the optional port.

Figure 2-3-1 FXS Port Status

Port Status

Port Number	Port Type	SIP Account	Enabled	Model	Group Number	Voltage	Register	Status
1	FXS	4001	Yes	S2	4	47	Unknown	on hook
2	FXS	4002	Yes	S2	4	47	Unknown	on hook
3	FXS		Yes	S2		47	Unknown	on hook
4	FXS		Yes	S2		47	Unknown	on hook

Figure 2-3-2 FXO Port Status

Port Status

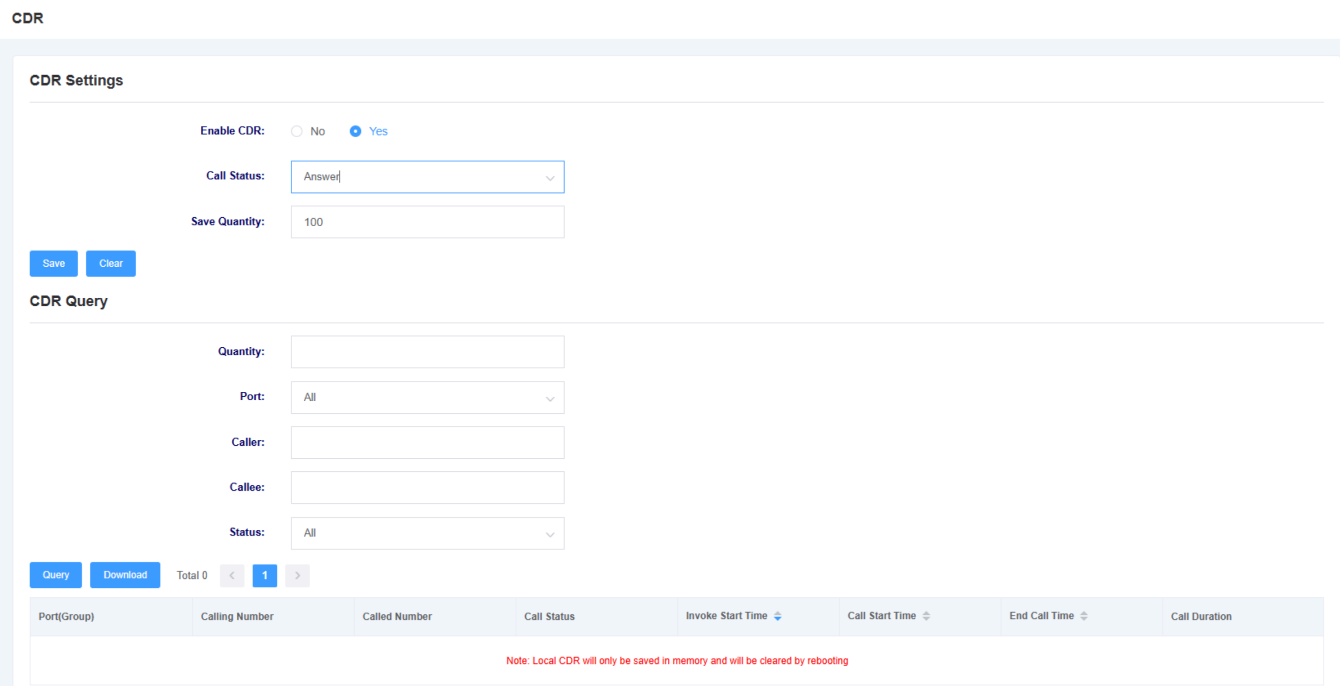
Port Number	Port Type	SIP Account	Enabled	Model	Group Number	Voltage	Register	Status
1	FXO		No	O0		50	Unknown	Connected
2	FXO		No	O0		1	Unknown	Disconnected
3	FXO		No	O0		2	Unknown	Disconnected
4	FXO		No	O0		1	Unknown	Disconnected
5	FXO		No	O0		1	Unknown	Disconnected
6	FXO		No	O0		1	Unknown	Disconnected
7	FXO		No	O0		1	Unknown	Disconnected
8	FXO		No	O0		1	Unknown	Disconnected

Total 8 < 1 >

2.4 CDR

On the CDR page, users can configure CDR settings and query CDR records.

Figure 2-4-1 CDR



Note: CDR records are stored only in memory and will be cleared after reboot.

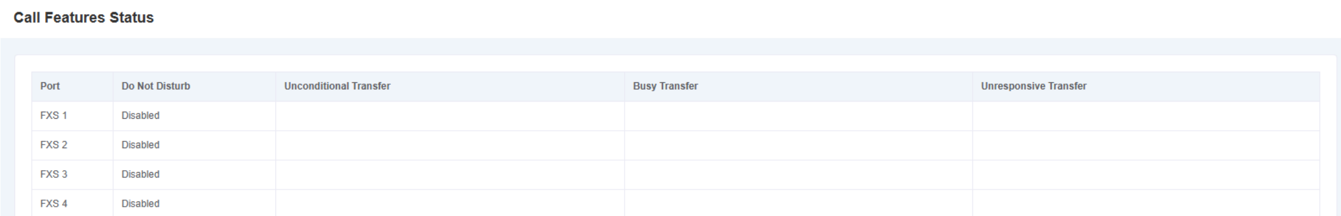
Table 2-4-1 CDR Options

Option	Description
Enable CDR	Select whether to enable CDR.
Call Status	Select the call status saved in CDR.
Save Quantity	Set the number of CDR entries to save.
Slot Number	Select the slot number for CDR queries.
Quantity	Select the number of CDR records to query.
Port	Select the port for CDR queries.
Caller	Filter CDR query items by caller number.
Callee	Filter CDR query items by callee number.

2.5 Call Feature Status

The Call Feature Status page displays the DND enable status, unconditional forwarding status, and busy forwarding status. Click the slot number drop-down list to switch between interface boards.

Figure 2-5-1 Call Feature Status Interface



3. Network Settings

3.1 Local Network

Figure 3-1-1 Local Network Interface

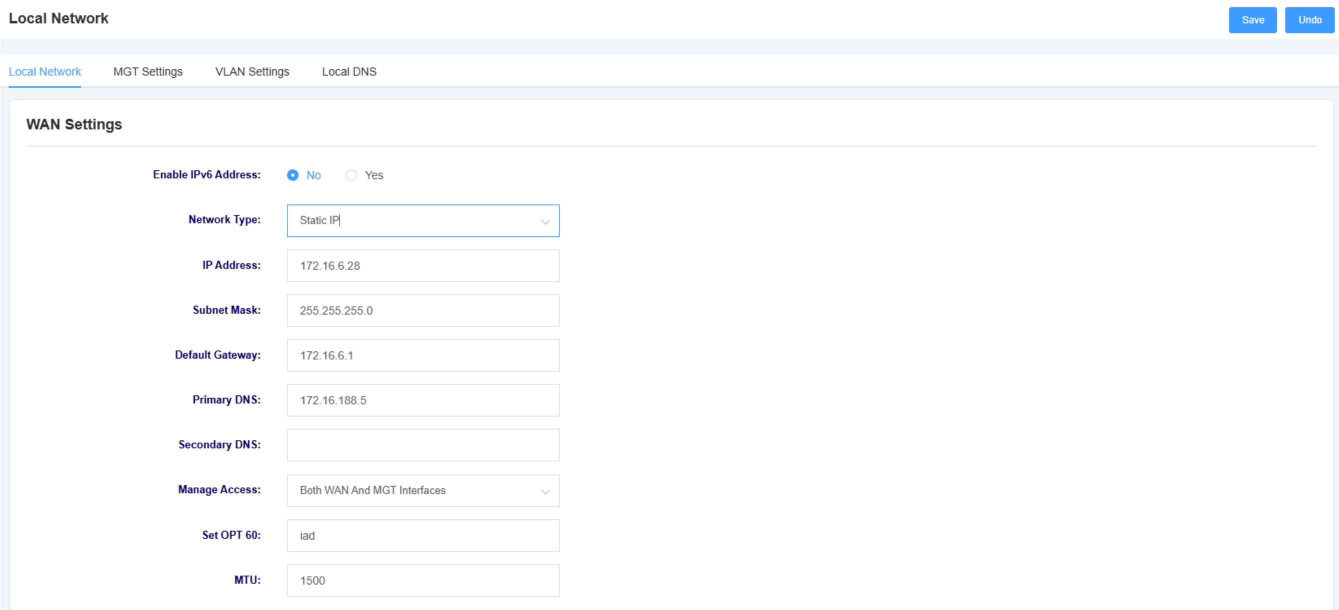


Table 3-1-1 WAN Settings Parameter Description

Option	**Description**
Network Mode	Select the device network mode.
Network Type	Select the network type: DHCP, Static IP, or PPPoE.
IP Address	Set the IP address of the device.
Subnet Mask	Set the subnet mask of the device.
Default Gateway	Set the default gateway of the device.
Primary DNS	Set the primary DNS of the device.
Secondary DNS	Set the secondary DNS of the device.
Management Access Options	Set web login restrictions.
Set OPT 60	Set OPT 60.
Option	Description
IP Address	Set the IP address of the LAN port.
Subnet Mask	Set the subnet mask of the LAN port.

Figure 3-1-2 Management Network Port Settings Interface

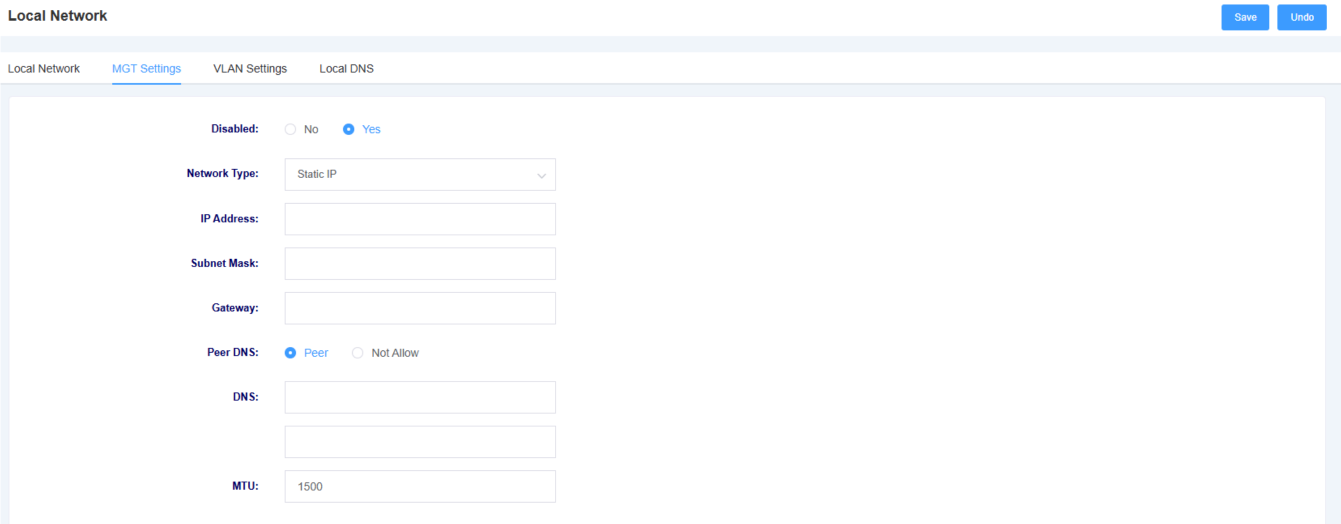


Table 3-1-2 Management Network Port Settings Parameter Description

Option	Description
Disable	Select whether to enable the management network port.
Network Type	Select the network type: DHCP, Static IP, or PPPoE.
IP Address	Set the IP address of the device.
Subnet Mask	Set the subnet mask of the device.
Gateway	Set the gateway of the device.
Remote DNS	Select whether to allow remote DNS.
DNS	Set the DNS of the device.

Figure 3-1-3 VLAN Settings Interface

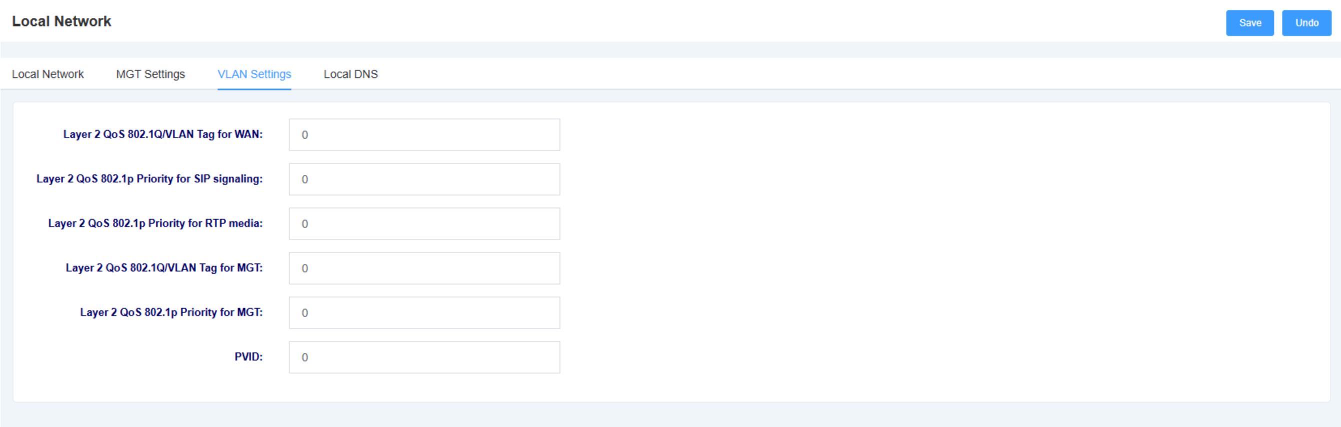


Table 3-1-3 VLAN Settings Parameter Description

Option	Description
Layer 2 QoS 802.1Q/VLAN Tag on WAN Port	Set the WAN port tag.
Layer 2 SIP Signaling QoS 802.1p Priority	Set the SIP signaling priority.

Option	Description
Layer 2 Voice QoS 802.1p Priority	Set the voice priority.
Layer 2 QoS 802.1Q/VLAN Tag on Management Port	Set the management port tag.
Layer 2 QoS 802.1p Priority on Management Port	Set the management port priority.

Figure 3-1-4 Local DNS Settings Interface

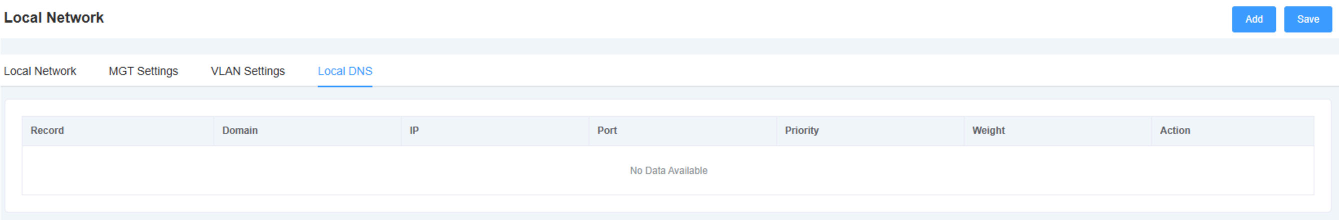


Figure 3-1-5 Add Local DNS Settings Interface

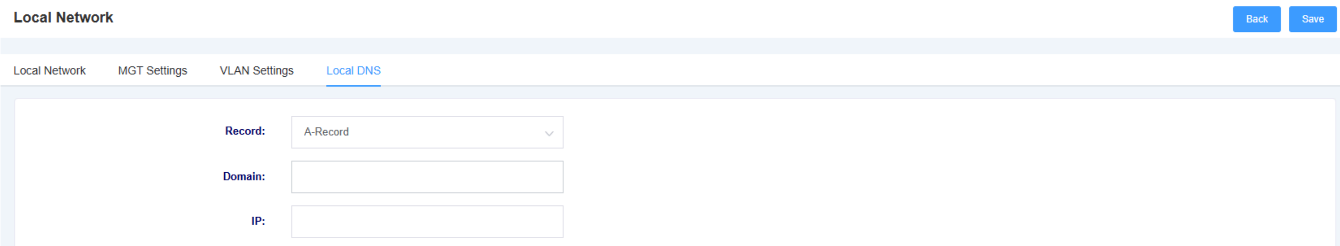


Table 3-1-4 Local DNS Parameter Description

Option	Description
Domain Name	Set the device domain name.
Resolved IP	Set the IP address to be resolved.

3.3 Static Route

On the "Static Route" page, static route network interface, target IP address, subnet mask, gateway, hop count, and operations are displayed. Static routes can be added here. Click the Add button to add a static route.

Figure 3-3-1 Static Route Interface

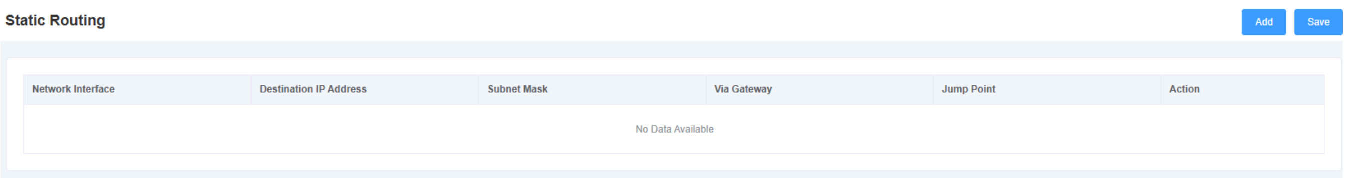
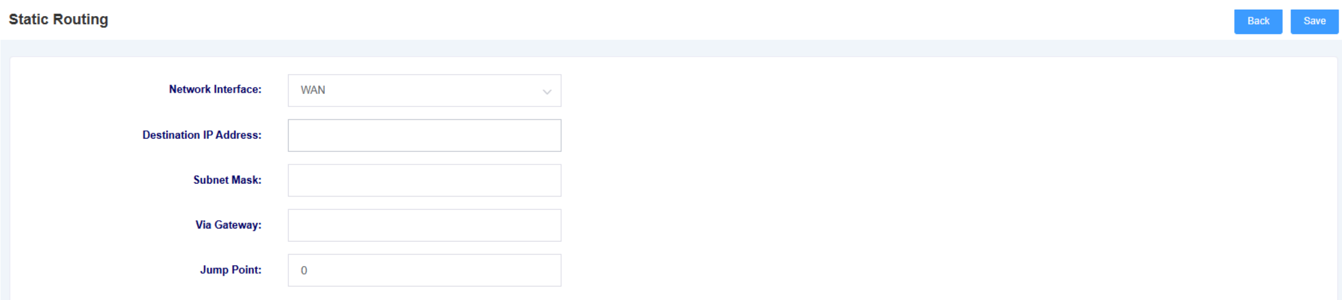


Figure 3-3-2 Add Static Route Interface



3.4 Firewall

On the "Firewall" page, firewall rule names, protocols, source network domain, source IP, source port, target network domain, target IP, target port, and rule actions are displayed. Firewall rules can be added here to ensure device security. Click the Delete button to delete firewall rules or the Add button to add firewall rules.

Figure 3-4-1 Firewall Interface

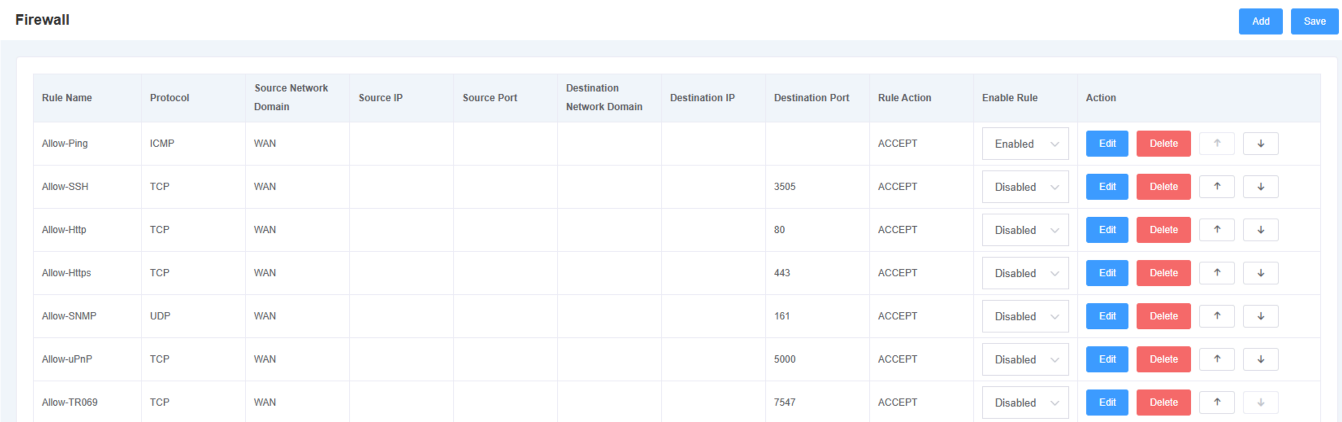


Figure 3-4-2 Add Firewall Rule Interface

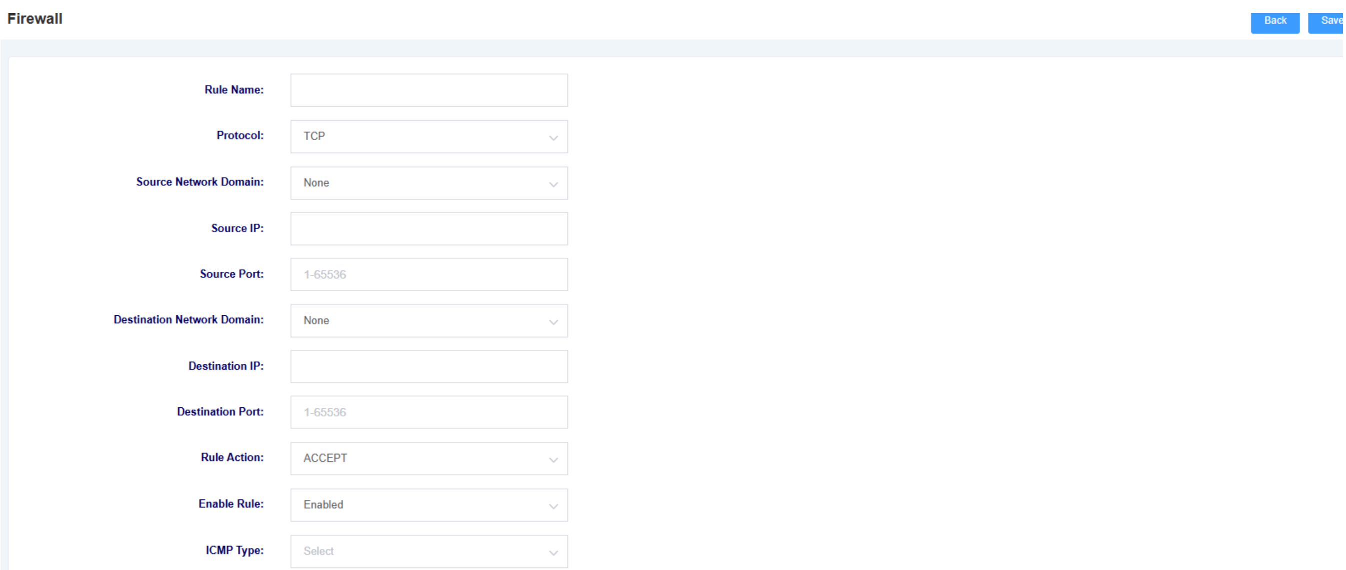


Table 3-4-1 Add Firewall Rule Parameters

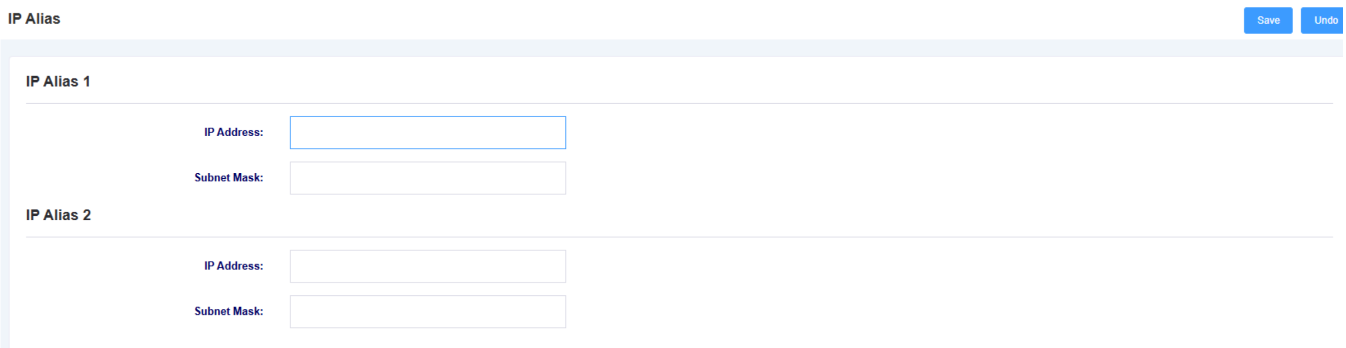
Option	Description
Name	Firewall rule name

Protocol	Protocol specified by firewall rule
Source Network Domain	Source network domain of firewall rule
Option	Description
Source IP	Source IP defined by firewall rule; if left empty, applies to all IPs
Source Port	Define source port, range 1-65535
Target Network Domain	Target network domain of firewall rule
Target IP	Target IP defined by firewall rule; if left empty, applies to all IPs
Target Port	Define target port, range 1-65535
Rule Action	Define rule action, options: ACCEPT, REJECT, DROP

3.5 IP Alias

The AIU-8 supports setting multiple IP addresses and can be configured in the IP Alias interface.

Figure 3-5-1 IP Alias Interface



3.6 VPN Settings

On this interface, VPN can be enabled and configured. The AIU-8 currently only supports OpenVPN.

Figure 3-6-1 VPN Settings Interface

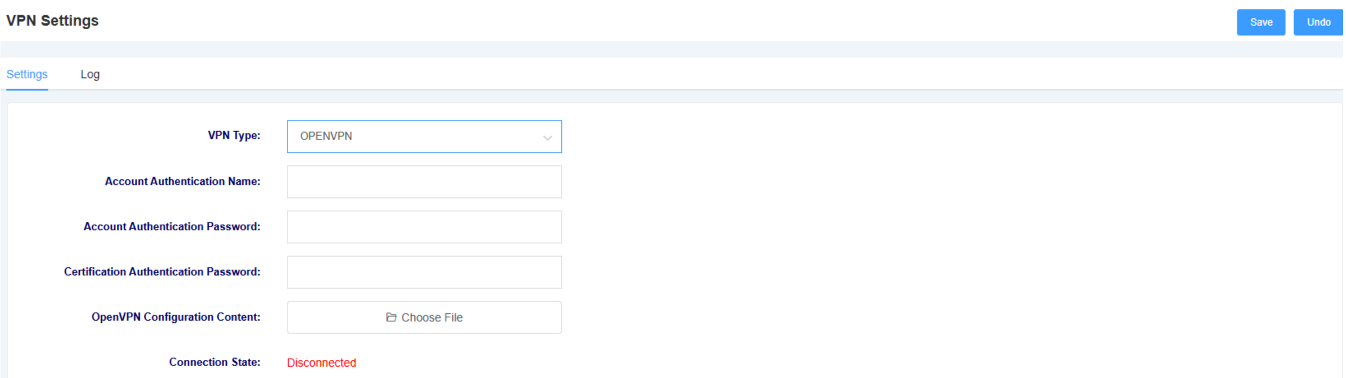
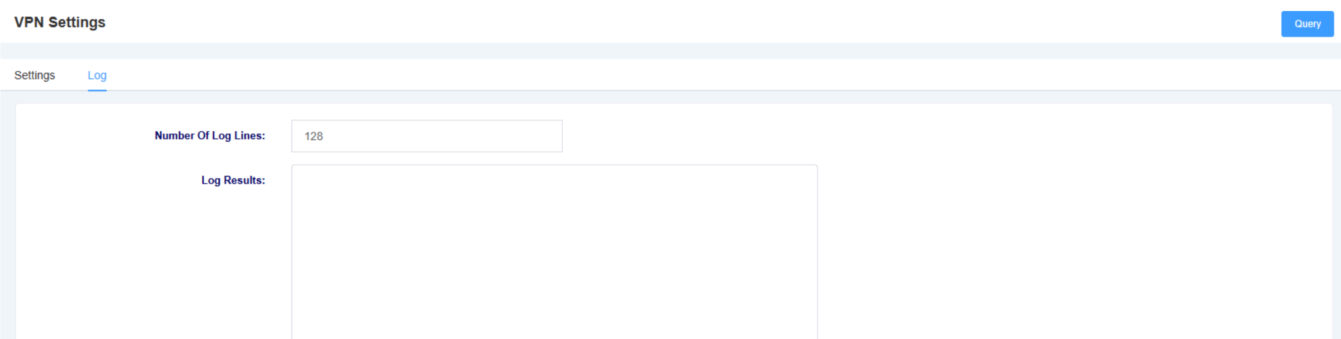


Table 3-6-1 VPN Settings Description

Option	Description
VPN Type	Can choose to disable VPN or use OpenVPN
Account Authentication Name	Authentication name used by OpenVPN
Account Authentication Password	Authentication password used by OpenVPN
Cert Authentication Password	Cert authentication password
OVPN Configuration Content	Upload OpenVPN configuration file
Connection Status	Show VPN connection status

On the Log page, you can select the number of log lines to display and then click the Query button. The logs will be displayed in the "Log Results" box.

Figure 3-6-2 VPN Log Interface



4. Templates

The iAG200 provides a convenient SIP registration method. Users can easily apply pre-configured templates to FXS ports. A total of 2/4 templates are available for configuration.

4.1 SIP Settings

Figure 4-1-1 SIP Settings

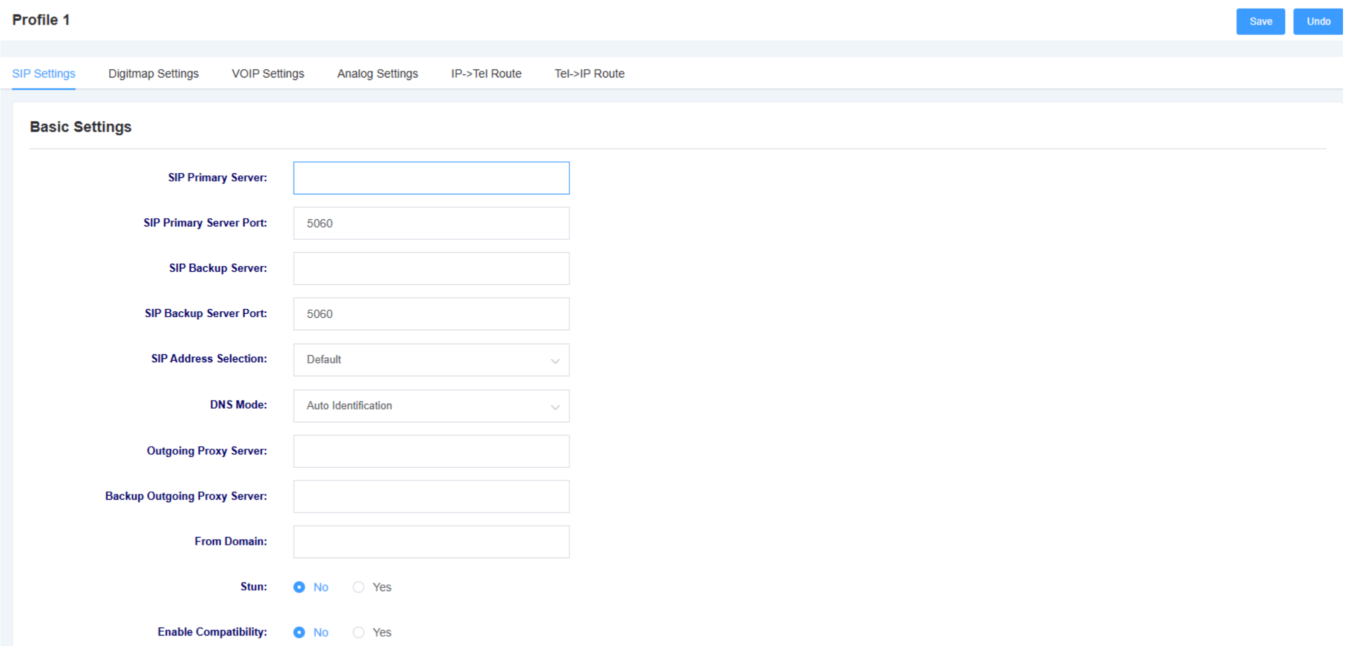


Table 4-1-1 SIP Settings Parameters

Option	Description
SIP Primary Server	Set SIP primary server
SIP Primary Server Port	Set SIP primary server port
SIP Backup Server	Set SIP backup server
Option	Description
SIP Backup Server Port	Set SIP backup server port
SIP Address Selection	Select which network port SIP service registers on
DNS Mode	Set DNS mode, optional Auto or DNSSRV
Outbound Proxy Server	Set outbound proxy server. The gateway will send signaling to this external proxy instead of directly to the peer.

From Domain	Set domain name for peer verification
STUN	Select whether to enable STUN service
Compatibility Mode	Select whether to enable compatibility mode

Figure 4-1-2 SIP Settings

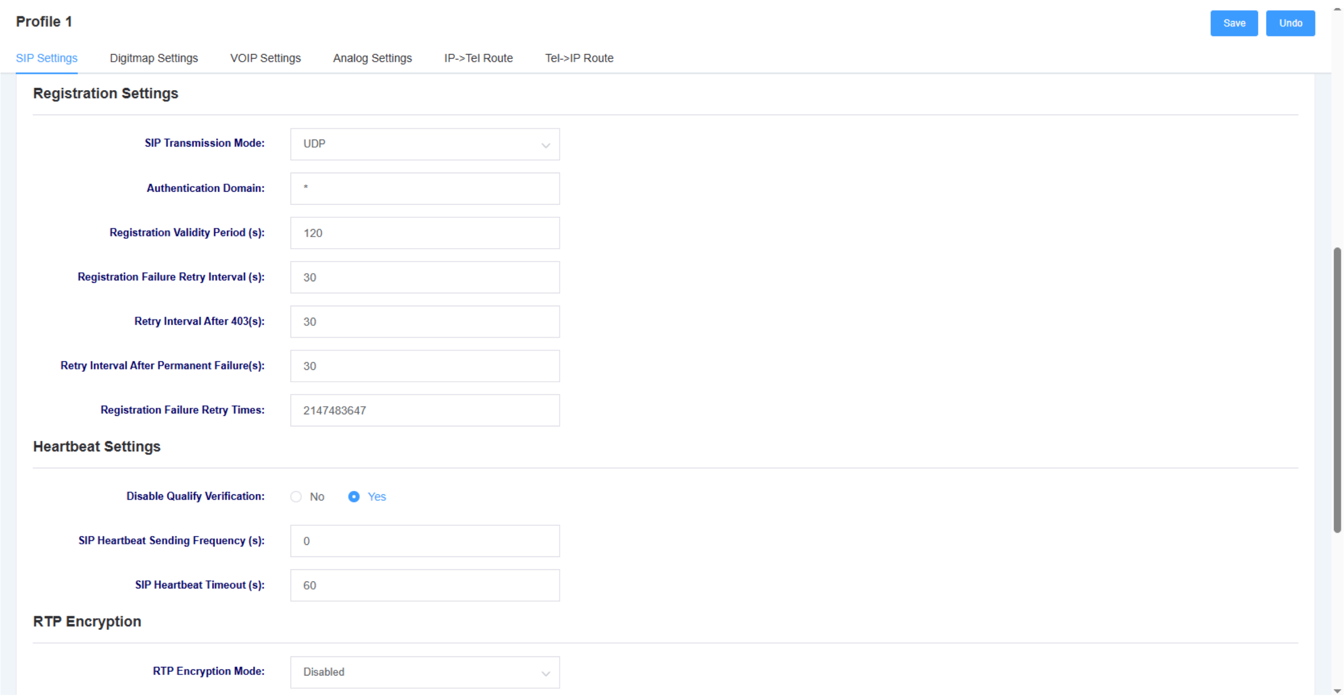
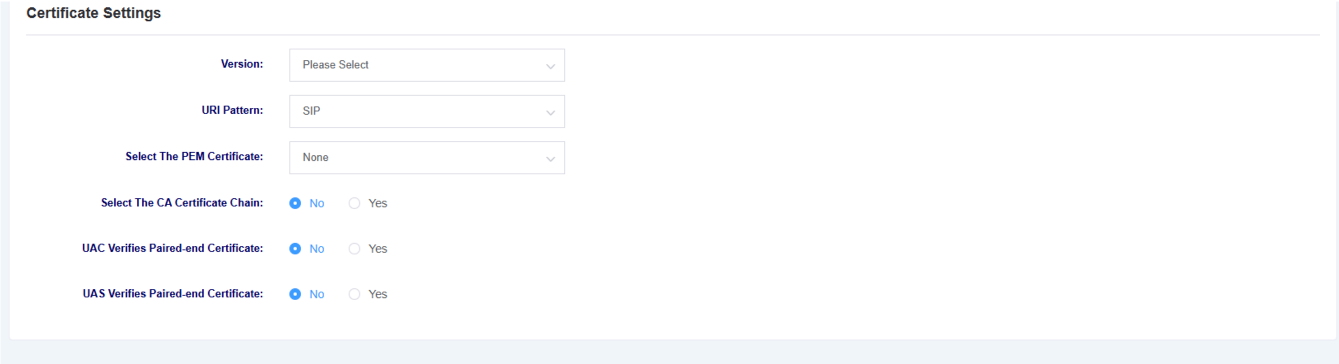


Table 4-1-2 SIP Settings Parameters

Option	Description
SIP Transport Method	Set SIP transport method, optional UDP, TCP, and TLS
Authentication Domain	Set SIP registration authentication domain
Option	Description
Registration Validity Period	Set registration validity period, default 3600 seconds
Registration Failure Retry Interval	Set registration failure retry interval, default 30 seconds
Registration Failure Retry Count	Set registration failure retry count, default 10 times
Qualify Verification	Whether to enable qualify verification
SIP Heartbeat Send Frequency	Set SIP heartbeat packet send frequency
SIP Heartbeat Timeout Time	Set SIP heartbeat packet timeout time
RTP Encryption Mode	Whether to enable RTP encryption

Figure 4-1-3 SIP Settings



Option	**Description**
Version	Select certificate version. Device supports different versions of TLS, SSL, SS certificates
URI Mode	Select URI mode, supporting SIP and SIPS
Select Device PEM Certificate	Select device PEM certificate
Select CA Certificate Chain	Select whether to enable CA certificate chain
Option	Description
UAC Verify Peer Certificate	As the calling party, select UAC to use telephone as refresher. Or select UAS to use callee or proxy server as refresher.
UAS Verify Peer Certificate	As the called party, select UAC to use callee or proxy server as refresher, or select UAS to use telephone as refresher.

4.2 Dial Plan Settings

On this page, dial rules and function key settings can be configured.

Figure 4-2-1 Dial Plan Settings

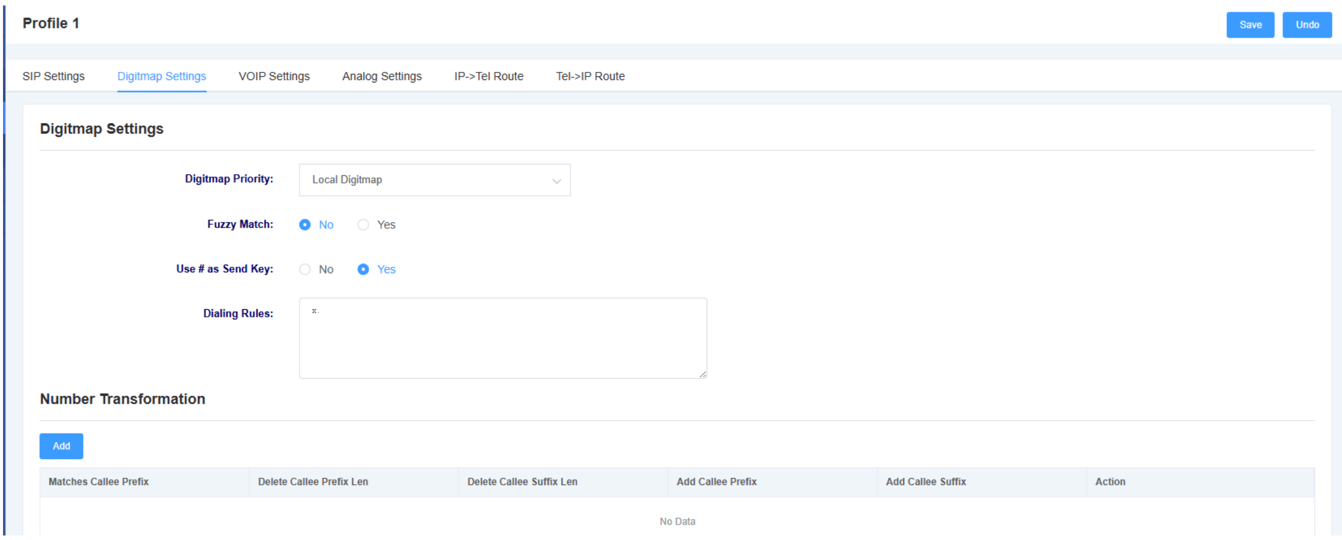


Table 4-2-1 Dial Plan Settings Parameters

Option	Description
Dial Plan Model	Select whether dial plan is local-first or remote dial plan. If using Openvox IPPBX, you can use remote dial plan to prioritize IPPBX dial rules
Fuzzy Matching	Select whether to enable fuzzy matching
Use # as Send Key	After enabling, pressing # after dialing will send the dialed number
Dial Rules	1. If no dial plan is configured, the soft switch server's dial plan will be used. 2. Valid characters include: 0-9, x, . 3. X represents matching any single digit 0-9. 4. . represents matching any number of the preceding digit (total not exceeding 32 bits). 5. . can only appear once and only at the end. 6. Even with variable-length dial plan configured, you can press # to dial quickly. 7. Configure multiple dial rules separated by commas

Figure 4-2-2 Function Key Settings

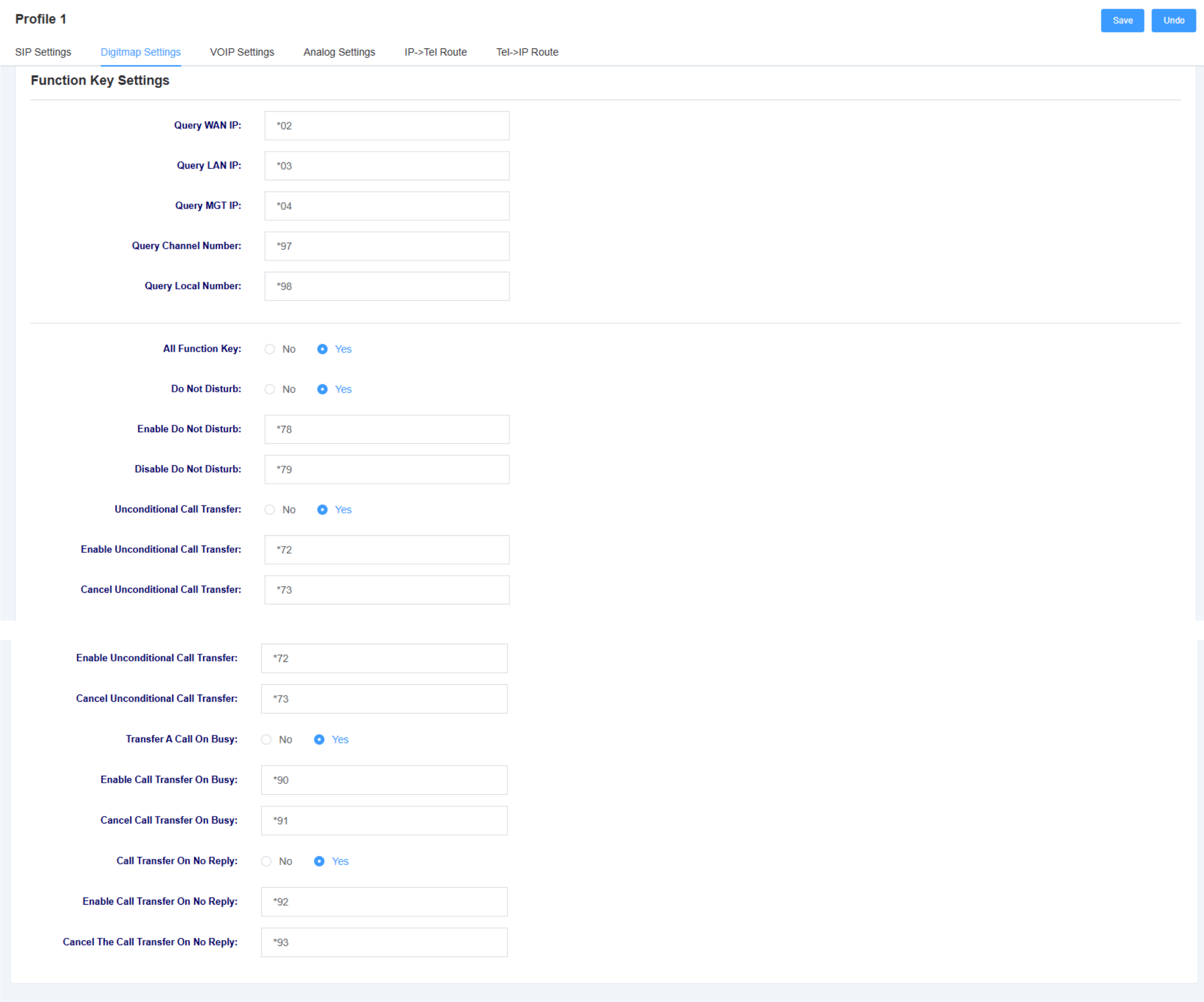


Table 4-2-2 Function Key Settings Parameters

Option	Description
Query IP	Set function key to query IP. After dialing, the device IP will be announced
Query Channel Number	Set function key to query channel number. After dialing, the channel number will be announced
Query Local Number	Set function key to query local number. After dialing, the local number will be announced
All Function Keys	Select to enable or disable function keys
Do Not Disturb	Select to enable or disable do not disturb function
Enable Do Not Disturb	Set function key to enable do not disturb. After dialing, do not disturb will be enabled on this extension

Disable Do Not Disturb	Set function key to disable do not disturb. After dialing, do not disturb will be disabled on this extension
Unconditional Call Forwarding	Select to enable or disable unconditional call forwarding function
Enable Unconditional Call Forwarding	Set function key to enable unconditional call forwarding. After dialing this key plus the forwarding extension, unconditional call forwarding will be enabled on this extension
Disable Unconditional Call Forwarding	Set function key to disable unconditional call forwarding. After dialing, unconditional call forwarding will be disabled on this extension
Call Forwarding on Busy	Select to enable or disable call forwarding on busy function
Enable Call Forwarding on Busy	Set function key to enable call forwarding on busy. After dialing this key plus the forwarding extension, call forwarding on busy will be enabled on this extension
Disable Call Forwarding on Busy	Set function key to disable call forwarding on busy. After dialing, call forwarding on busy will be disabled on this extension
Call Forwarding on No Answer	Select to enable or disable call forwarding on no answer function
Enable Call Forwarding on No Answer	Set function key to enable call forwarding on no answer. After dialing this key plus the forwarding extension, call forwarding on no answer will be enabled on this extension
Disable Call Forwarding on No Answer	Set function key to disable call forwarding on no answer. After dialing, call forwarding on no answer will be disabled on this extension

4.3 VOIP Settings

On this interface, users can configure VOIP-related parameters.

Figure 4-3-1 VOIP Settings

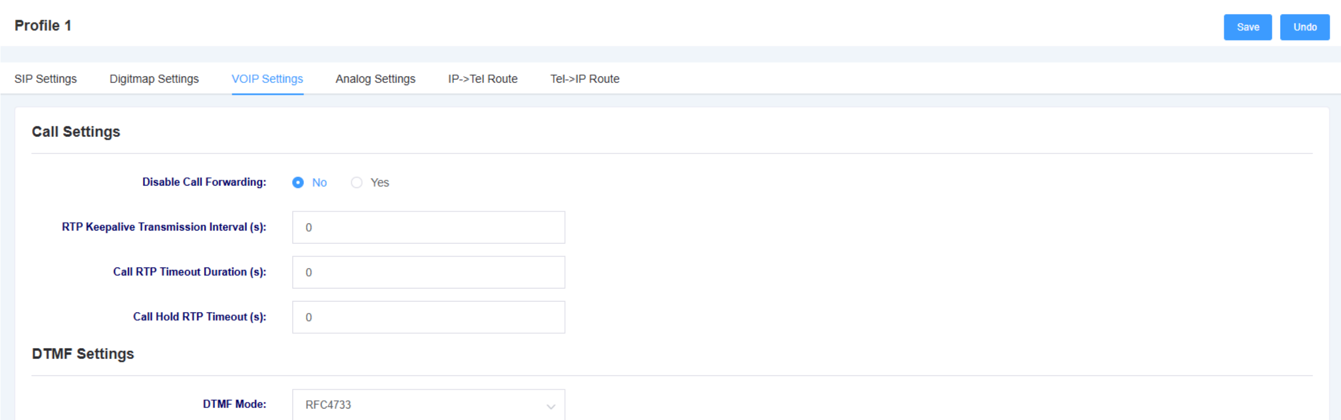


Table 4-3-1 VOIP Settings Parameters

Option	Description
Allow Call Transfer	Select whether to enable call transfer
RTP Keep-Alive Send Interval	Set RTP keep-alive send interval
Call RTP Timeout Time	Set call RTP timeout time
Call Hold RTP Timeout Time	Set call hold RTP timeout time
DTMF Mode	Set DTMF mode, options: RFC4733, inband, info, auto, and auto_info

Figure 4-3-2 VOIP Settings

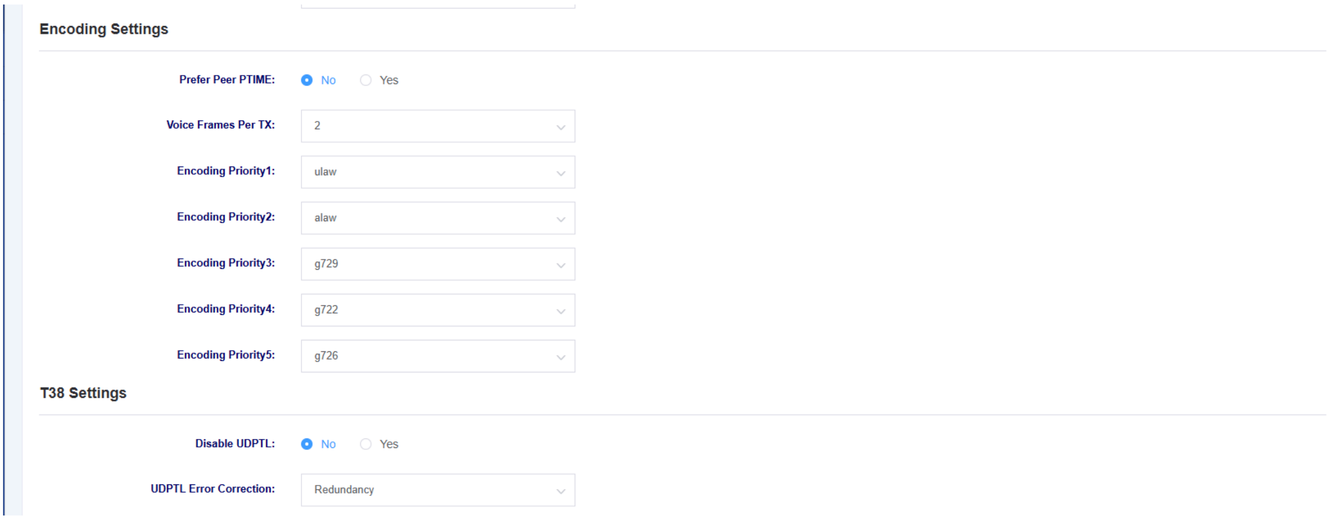


Table 4-3-2 VOIP Settings Parameters

Option	Description
Use Codec Packing Duration	Select whether to use codec packing duration for more efficient bandwidth and resource utilization in transmission, storage, and processing
Codec Priority	Set codec priority
Enable UDPTL	Select whether to enable UDPTL
UDPTL Error Correction	Select UDPTL error correction method

4.4 Analog Settings

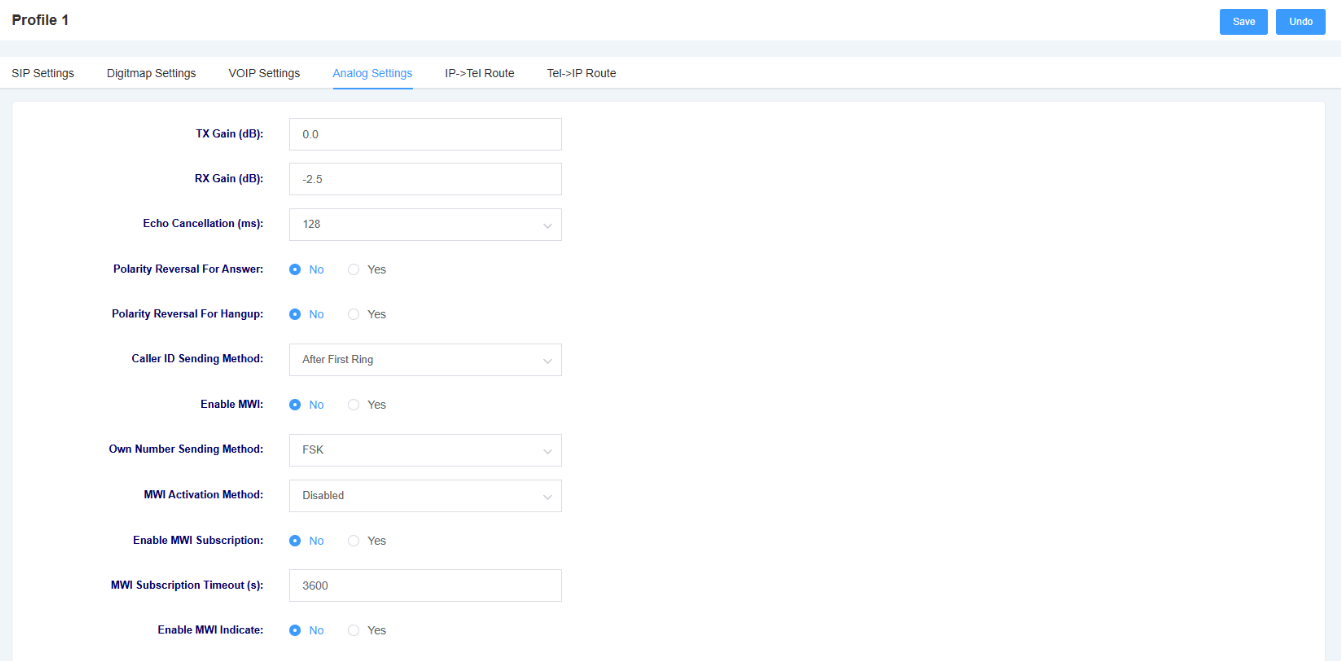


Table 4-4-1 Analog Settings Parameters

Option	Description
TX Gain	Set sending voice gain
RX Gain	Set receiving voice gain
Echo Cancellation	Select whether to enable echo cancellation and the milliseconds to enable
Polarity Reversal Indicates Answer	Select whether to enable polarity reversal to indicate answer
Polarity Reversal Indicates Hangup	Select whether to enable polarity reversal to indicate hangup
Caller ID Sending Method	Select caller ID sending method
Enable MWI Subscription and Local Number Display	Set whether to enable MWI subscription and local number display. When enabled, the local number will be displayed on the phone screen when on-hook
Local Number Display Method	Select local number display method
Message Waiting Indicator Light Method	Select message waiting indicator light method

5. FXS Port Settings

On this page, You can set the FXS port and FXO port.

5.1 FXS port settings

Figure 5-1-1 Basic Settings

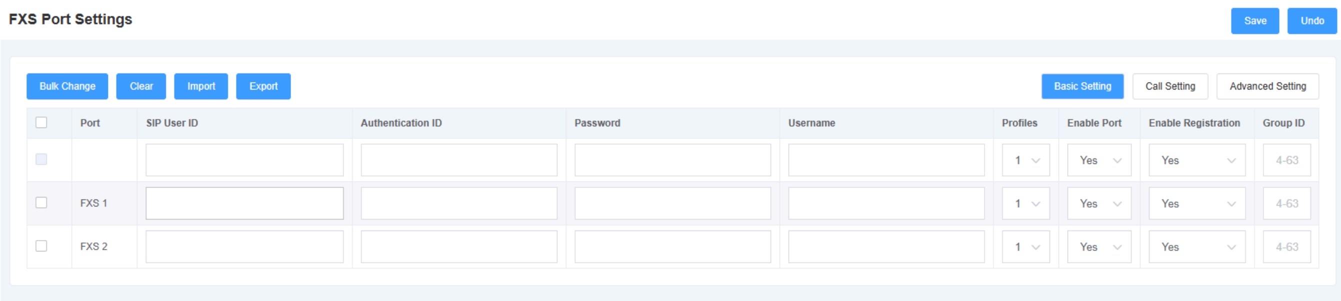


Table 5-1-1 Basic Settings Parameters

Option	Description
SIP User ID	Set SIP user ID corresponding to this FXS port
Authentication ID	Set authentication ID corresponding to this SIP user ID
Password	Set password corresponding to authentication ID
Username	Set caller display name
Template	Select template to use
Enable Port	Select whether to enable this port
Enable Registration	Select whether to enable registration

Figure 5-1-2 Call Settings

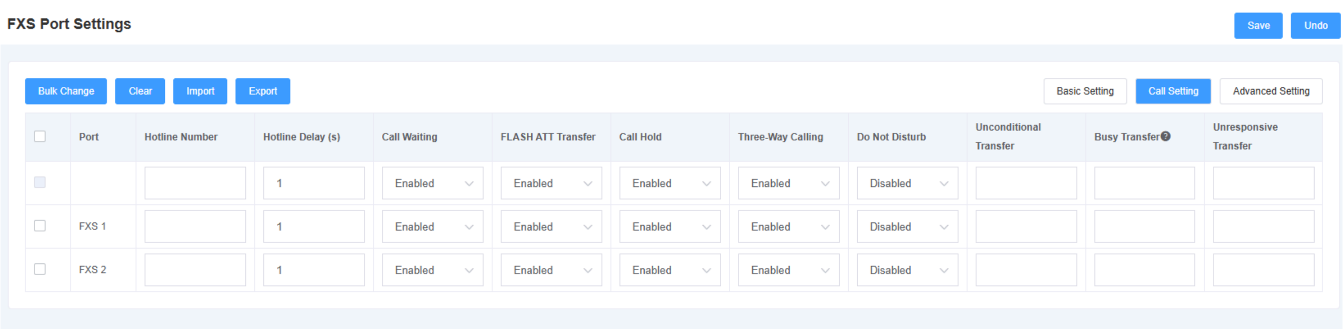


Table 5-1-2 Call Settings Parameters

Option	Description

Hotline Number	Set hotline number for the port. If no number is dialed within the hotline delay time after picking up, the hotline number will be dialed automatically
Hotline Delay	Set hotline delay
Call Waiting	Select whether to enable call waiting
Call Transfer	Select whether to enable call transfer
Call Hold	Select whether to enable call hold
Three-Way Call	Select whether to enable three-way call
Do Not Disturb	Select whether to enable do not disturb
Unconditional Transfer	Set unconditional transfer number
Transfer on Busy	Set transfer on busy number
Transfer on No Answer	Set transfer on no answer number

Figure 5-1-3 Advanced Settings

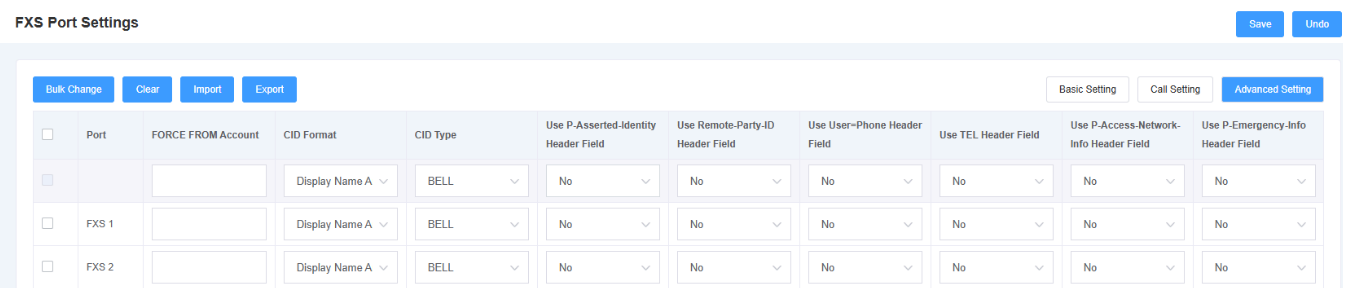
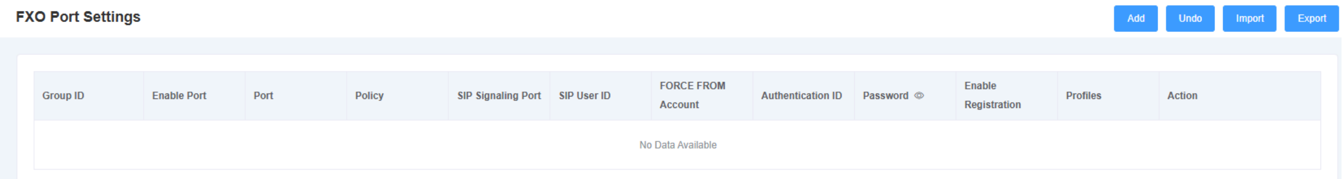


Table 5-1-3 Call Settings Parameters

Option	Description
FROM Force User	Set FROM force user
CID Message Signal	Set CID message signal
Use P-Asserted-Identity Header	Carry "P-Preferred-Identify" in INVITE. In anonymous calls, user identity can be indicated through P-Preferred-Identify header
Use Remote Party ID Header	Use Remote Party ID header to obtain CID
Use User=Phone Header	Carry "user=phone" in URI. When calling to PSTN network, extract called number from username

Use P-Accessed-Network-Info Header	Use P-Accessed-Network-Info header to obtain CID
Use P-Emergency-Info Header	Use P-Emergency-Info header to obtain CID

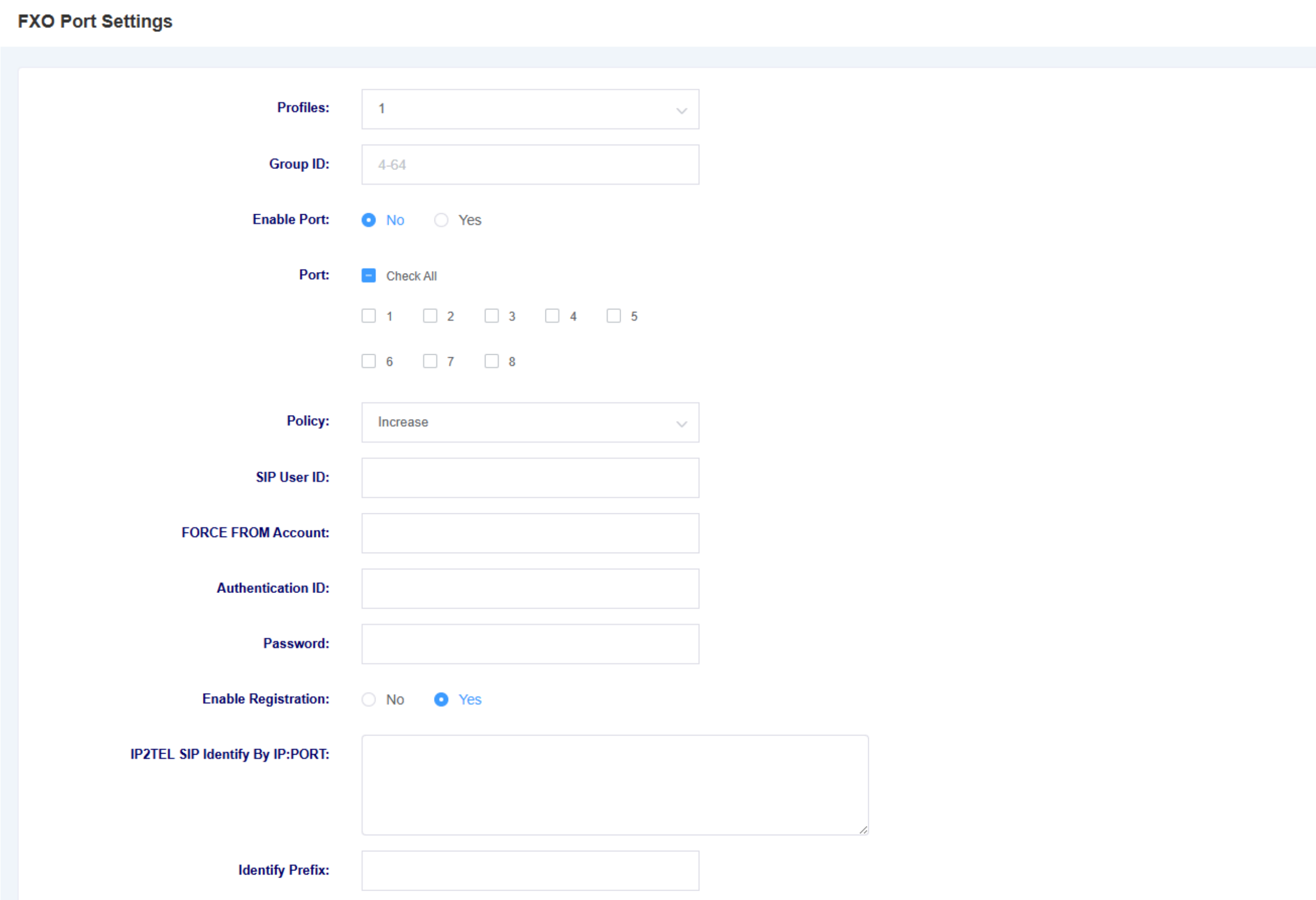
5.2 FXO port settings



Field	Description
Group ID	Used to identify and manage multiple SIP entries. Entries in the same group usually have similar attributes or business purposes, making batch management and filtering easier.
Enable Port	Used to control whether the SIP configuration is active. Set to “Yes” to enable, or “No” to disable.
Port	The local or remote port number used for SIP communication, usually 5060. Make sure it does not conflict with other services.
Strategy	Defines how SIP messages are handled, such as registration on demand, initial registration, or backup routing.
SIP Signaling Port	The port number used to send and receive SIP signaling, such as INVITE, REGISTER, BYE, etc. If not specified separately, it usually matches the “Port” value.
SIP User ID	The user identifier assigned by the carrier or PBX, used for authentication and routing, usually a numeric string.
FROM Signaling Address	The address filled in the From header of SIP messages, usually in the format `sip:userID@domain` or `sip:userID@IP`.
Authentication ID	The username used for identity verification during registration or calling, which may be the same as the “SIP User ID” or may be provided separately by the carrier.
Password	The authentication password used together with the “Authentication ID” for verification during registration or call initiation.
Enable Registration	Indicates whether to actively send registration requests to the SIP server. When enabled, the device periodically sends REGISTER messages; when disabled, it only supports direct IP-based calls.

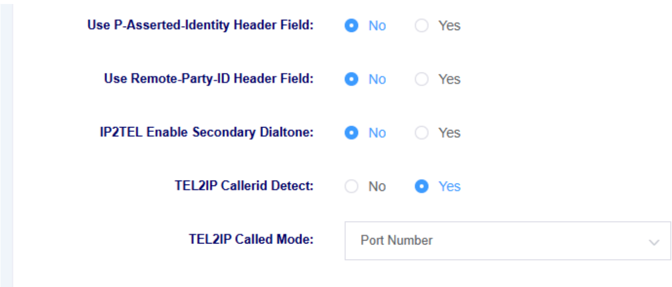
Field	Description
Template	Used to quickly apply predefined parameter settings. You can choose a built-in system template, or save the current configuration as a custom template for later reuse.
Actions	Operations performed on this record, usually including edit, delete, copy, and test.

The click to add button interface is shown below



Field	Description
Template	Used to quickly load a predefined configuration. After entering the template ID, the system automatically fills in the parameter values for that template, which is usually used for quickly deploying similar FXO line configurations.
Group ID	Used to classify multiple FXO ports into the same logical group for unified management. The valid range is usually 4–64, and ports within the same group can share the same application strategy and routing rules.
Enable Port	Used to control whether the FXO port configuration is active. Select “Yes” to enable, or “No” to disable.
Port	Used to select the physical FXO port to which the current configuration applies. You may select a single port, or choose “All” to apply it to all ports at once.

Field	Description
Strategy	Defines the line selection or number replacement rules used when an FXO port receives a call. “Seize in order” means ports in the group are selected sequentially by availability, and is commonly used for load balancing or failover.
SIP User ID	The user identifier used when the FXO port needs to register with a SIP server or initiate a call. It is usually provided by the carrier or PBX, and may be a numeric value or a URI format.
FROM Forced Account	Used to force replacement of the account portion in the From header of SIP messages. If this value is filled in, the system sends the specified content in the From header; if left blank, the default identifier is used.
Authentication ID	The username used for identity verification during registration or calling. It can be the same as the SIP User ID or different, and is usually provided by the carrier.
Password	The authentication password used together with the Authentication ID to complete SIP registration or call authentication.
Enable Registration	Used to control whether to actively register with the SIP server. When set to “Yes”, the device periodically sends REGISTER requests to stay online; when set to “No”, registration is not performed.
IP Direct Dial List	Used to configure a peer list for calls made directly based on IP address without SIP registration. Usually the peer IP and port are entered, and multiple fixed peers for direct dialing are supported.
Authentication Prefix	A number prefix used to trigger inbound authentication checks. Numbers matching this prefix must pass password or IP verification before routing, and it can also be used to distinguish dialing rules for different trunks.



Field	Description
Use P-Asserted-Identity Header	Whether to carry the P-Asserted-Identity header for transmitting caller identity.
Use Remote-Party-ID Header	Whether to carry the Remote-Party-ID header for transmitting caller information.
IP2TEL Called Party Secondary Dialing	Whether to allow continued digit input after an IP call reaches the telephone side.

Field	Description
Enable Incoming Caller ID Detection	Whether to detect the caller ID of incoming calls on FXO.
TEL2IP Called Party Mode	Defines the mapping method of the called number when a call comes in from the telephone side.

6. Advanced Configuration

6.1 Fax Parameters

On this page, fax-related parameters can be configured.

Figure 6-1-1 Fax Parameters

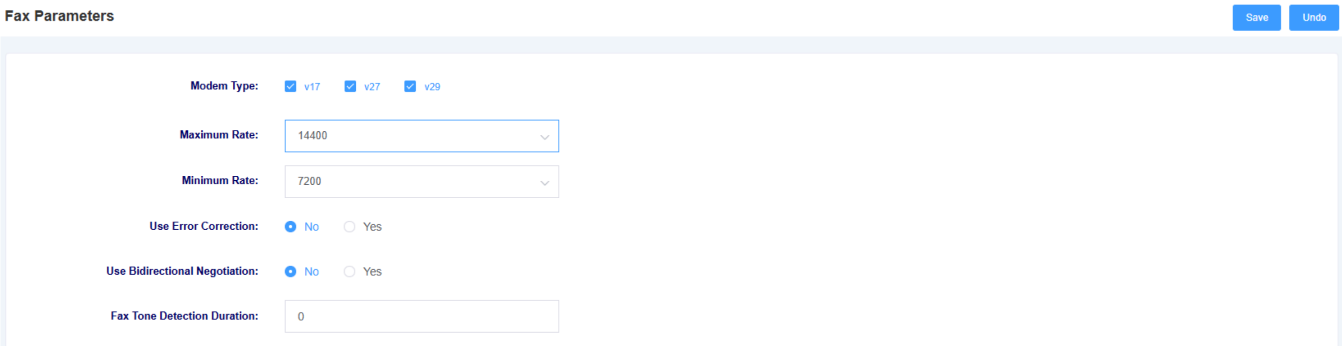


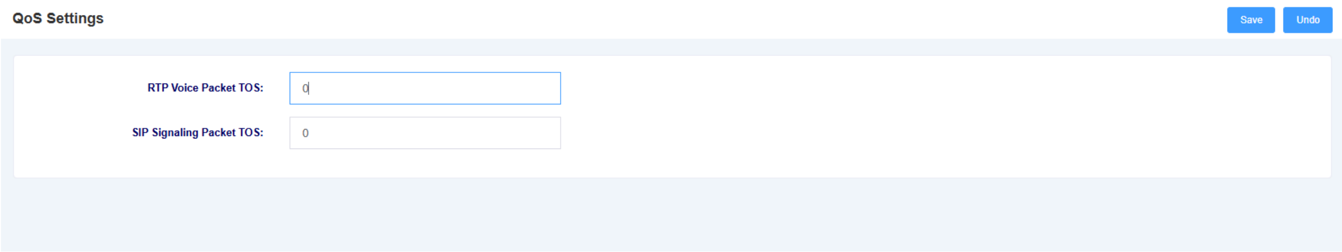
Table 6-1-1 Fax Parameters

Option	Description
Modem Type	Set supported modem type
Maximum Speed	Select maximum fax speed supported
Minimum Speed	Select minimum fax speed supported
Error Verification	Select whether to enable error verification
Bidirectional Negotiation	Select whether to enable bidirectional negotiation
Fax Tone Detection Duration	Set fax tone detection duration

6.2 QoS Settings

On this interface, RTP voice packet ToS and SIP signaling packet ToS can be set.

Figure 6-2-1 QoS Settings Interface



6.3 Analog Settings

On this interface, analog line-related parameters can be set, such as echo cancellation and jitter buffer.

Figure 6-3-1 Analog Settings Interface

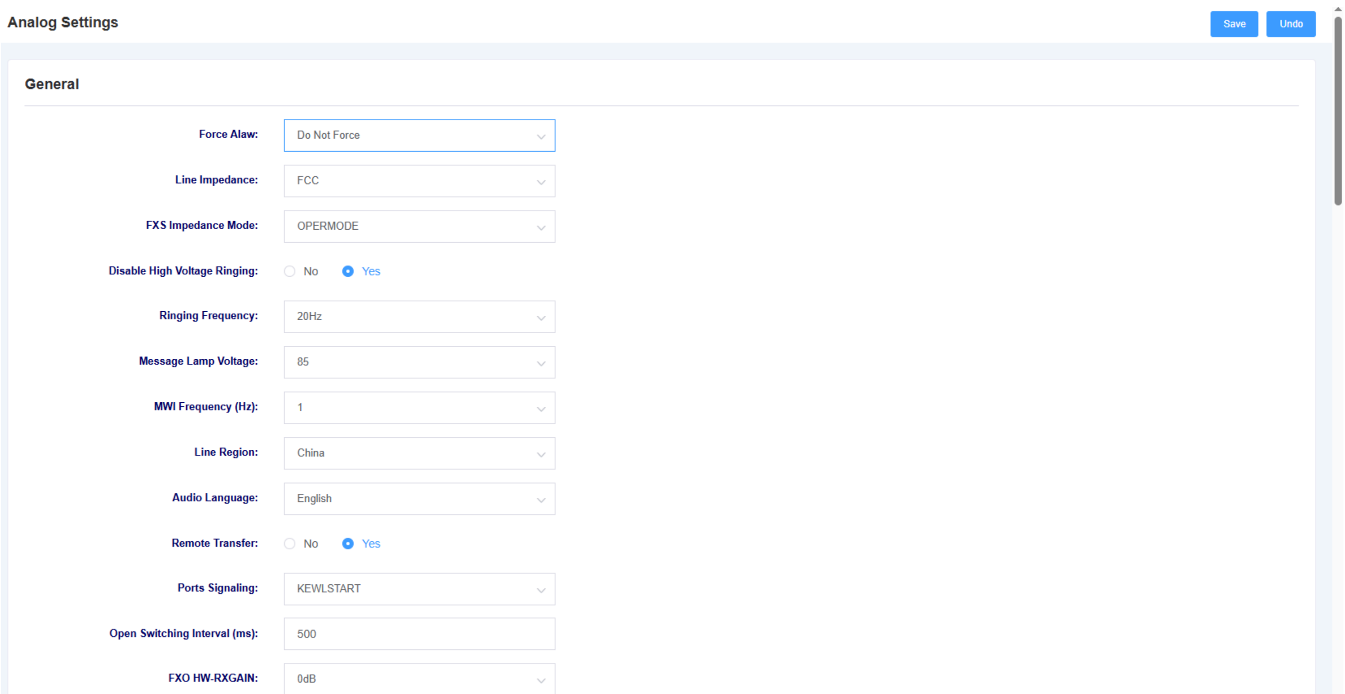


Table 6-3-1 Analog Settings Parameters

Option	Description
Force alaw	Select whether to enable this option. When enabled, alaw will be enforced
Line Impedance	Select line impedance
FXS Impedance Mode	Select FXS impedance mode
High Voltage Ringing	Select whether to enable high voltage ringing
Ringing Frequency	Select ringing frequency
Line Area	Select area where the line is located
Audio Language	Select language for voice prompts
Remote Transfer	Select whether to enable remote transfer function

Signaling	Select KEWLSTART / LOOPSTART protocol
Feeding Disconnect Duration (milliseconds)	Select disconnect duration
FXO Hardware RX	Select the level/gain and direction of voice signals received by this port from the remote line entering this device (receive path)
FXO Hardware TX	Select the level/gain and direction of voice signals sent by this device to the remote line (transmit path)

Figure 6-3-2 Analog Settings Interface

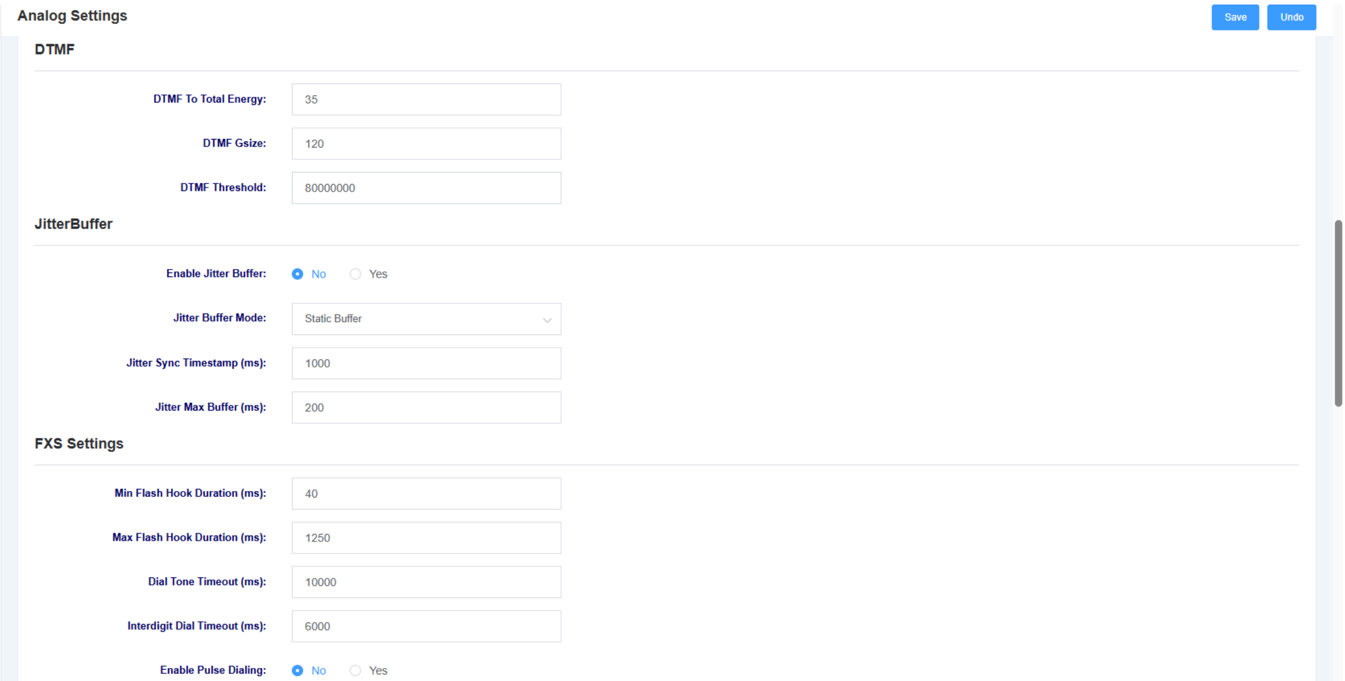
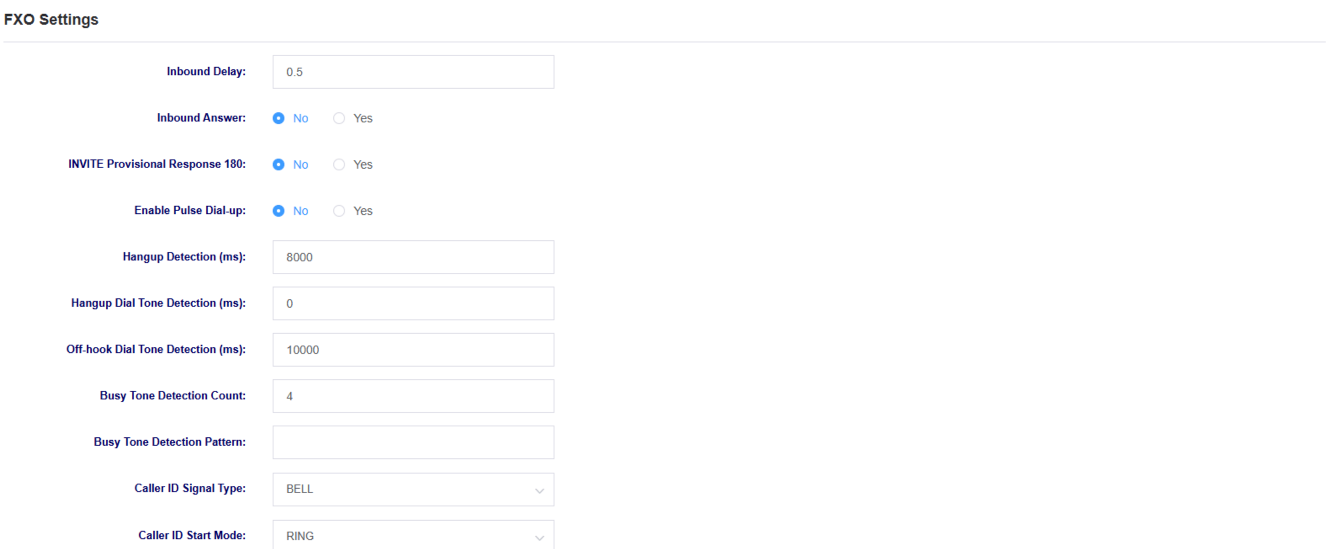


Table 6-3-2 Analog Settings Parameters

Option	Description
Total Energy Ratio Coefficient	Adjust energy ratio coefficient
Samples Per Detection Segment	Set number of samples
Energy Threshold	Set energy threshold
Jitter Buffer	Select whether to enable jitter buffer
Jitter Buffer Method	Select jitter buffer method
Jitter Synchronization Timestamp	Set jitter synchronization timestamp
Jitter Maximum Buffer	Set jitter maximum buffer
Option	Description

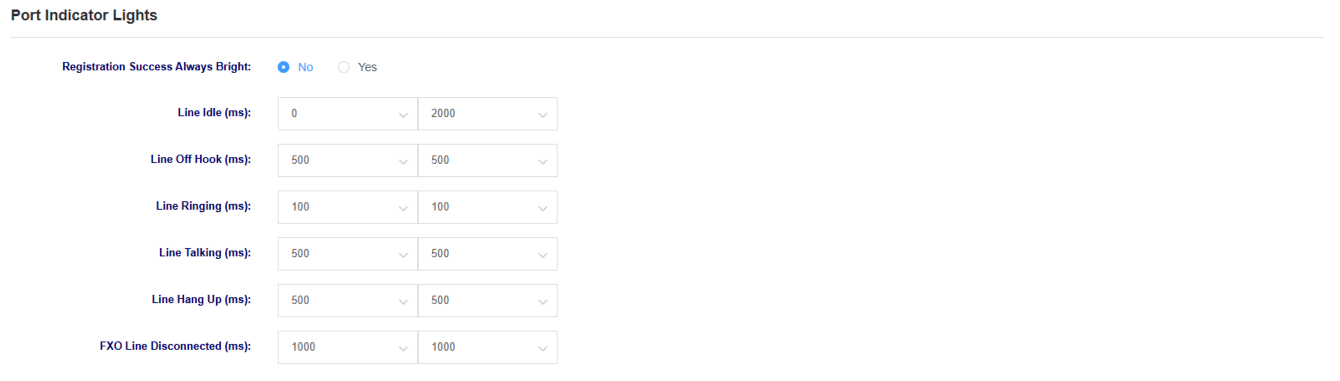
Minimum Hook Flash Duration	Set minimum hook flash duration
Maximum Hook Flash Duration	Set maximum hook flash duration
First Digit Timeout	Set first digit timeout time
Inter-Digit Timeout	Set inter-digit timeout time
Dial Matching Timeout	Set dial matching timeout time
Pulse Dialing	Select whether to enable pulse dialing
Maximum Pulse Duration	Set maximum pulse duration
Hangup Detection Duration	Set hangup detection duration

Figure 6-3-3 Analog Settings Interface



Option	**Description**
Call In Delay	Set FXO incoming call answer delay before connecting
Call In Auto Answer	Whether to auto answer on incoming call
INVITE Provisional Answer 180	Whether to send 180 Ringing
Enable Pulse Dialing	Whether to allow pulse dialing
Hangup Detection (milliseconds)	Hangup voltage/current drop detection duration
Hangup Dial Tone Detection (milliseconds)	Duration for determining hangup through dial tone
Off-Hook Dial Tone Detection (milliseconds)	Timeout for detecting dial tone after off-hook
Busy Tone Detection Times	Number of busy tone cycles needed for matching
Busy Tone Detection Rhythm	Busy tone rhythm/slot configuration

Option	Description
Caller ID Signal Type	FSK caller ID format (e.g., Bell/ETSI)
Caller ID Start Method	Caller ID trigger method (ringing/polarity reversal, etc.)



6.4 VOIP Settings

On this page, VoIP-related settings can be configured, such as call settings and session settings.

Figure 6-4-1 VoIP Settings

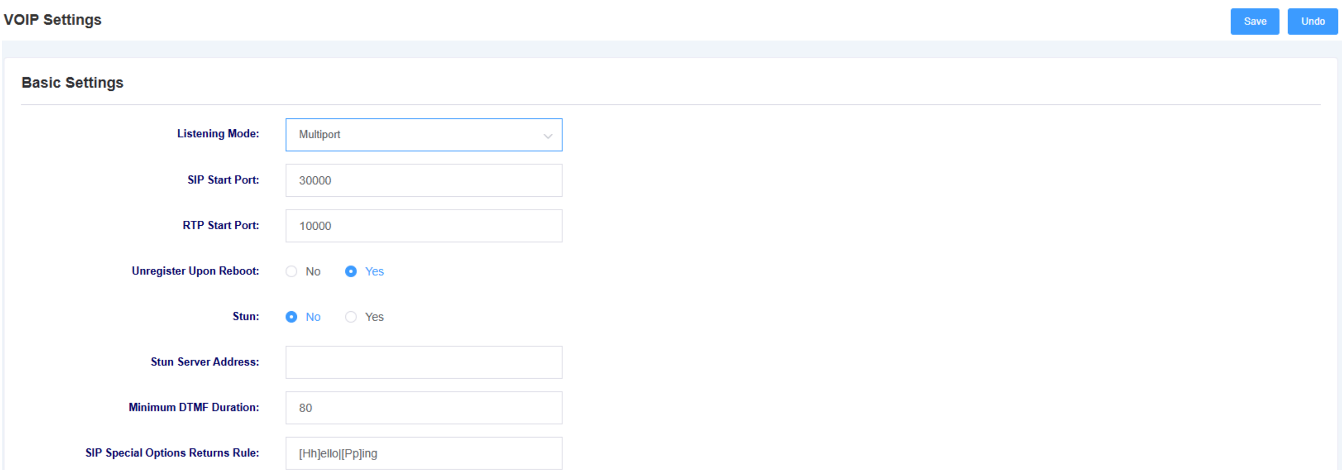


Table 6-4-1 VoIP Settings Parameters

Option	Description
Monitor Mode	Select monitor mode, optional multi-port or single-port
SIP Start Port	Set SIP start port
RTP Start Port	Set RTP start port
Unregister on Restart	Select whether to unregister on restart
STUN	Select whether to enable STUN
STUN Server Address	Set STUN server address

Figure 6-4-2 VoIP Settings

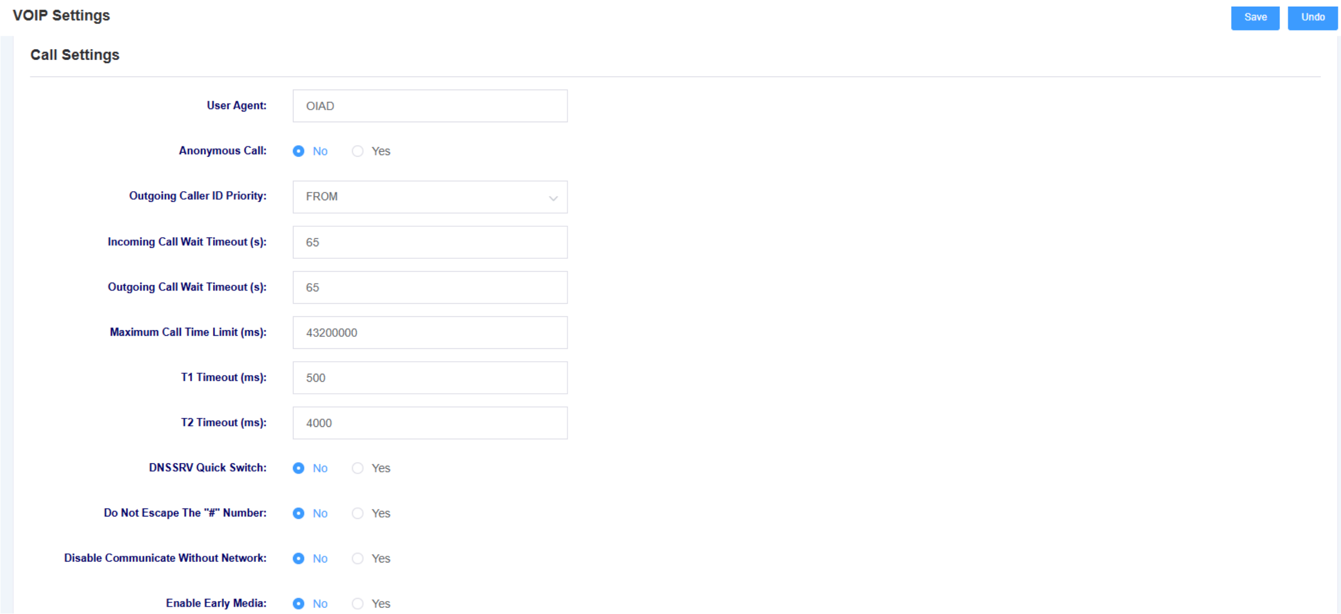


Table 6-4-2 VoIP Settings Parameters

Option	Description
User Agent	Set User Agent
Anonymous Call In	Select whether to allow anonymous call in
Caller ID Display Priority	Select caller ID priority to display from FROM field or P-Asserted-Identity field
Call In Waiting Timeout	Set call in waiting timeout time
Call Out Waiting Timeout	Set call out waiting timeout time
Call Maximum Duration	Set call maximum duration. Calls will be disconnected after exceeding this time
T1 Timeout	Set T1 timeout time
Network Disconnection Escape	Select whether to enable network disconnection escape. When enabled, internal extensions can call each other even when external network is disconnected
Early Media	Select whether to enable Early Media

Figure 6-4-3 VoIP Settings

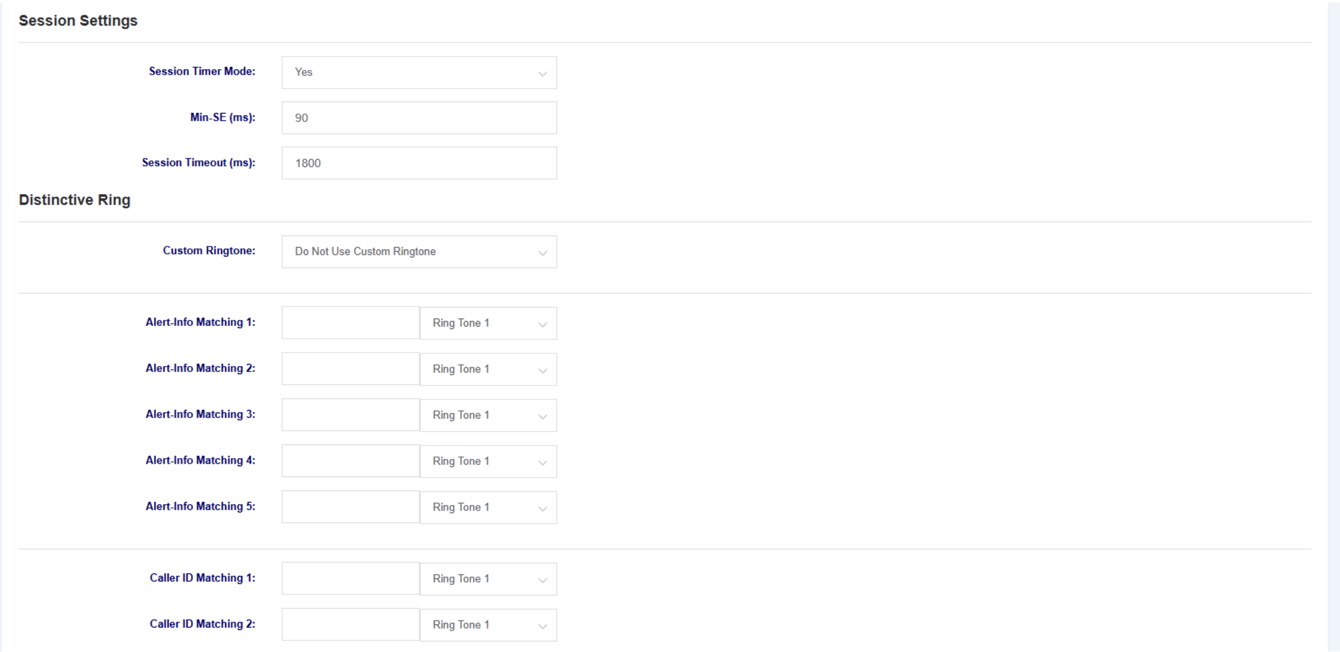
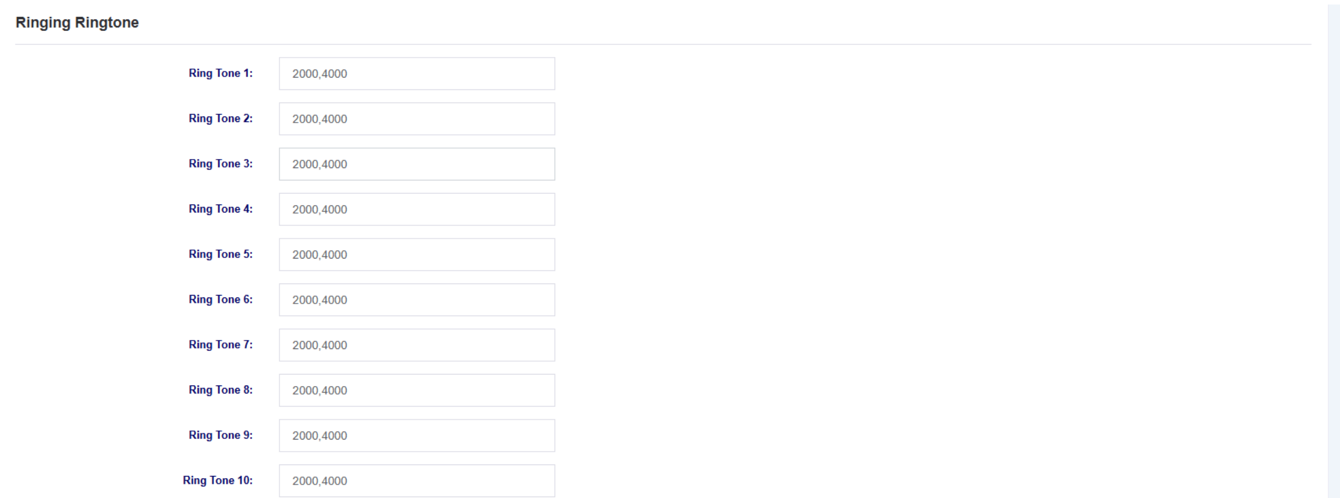


Table 6-4-3 VoIP Settings Parameters

Option	Description
Session Timer Mode	Select session timer mode
Min-SE	Set minimum session timeout duration
Session Timeout Time	Set session timeout time
Distinctive Ringing	Set different ringing tones

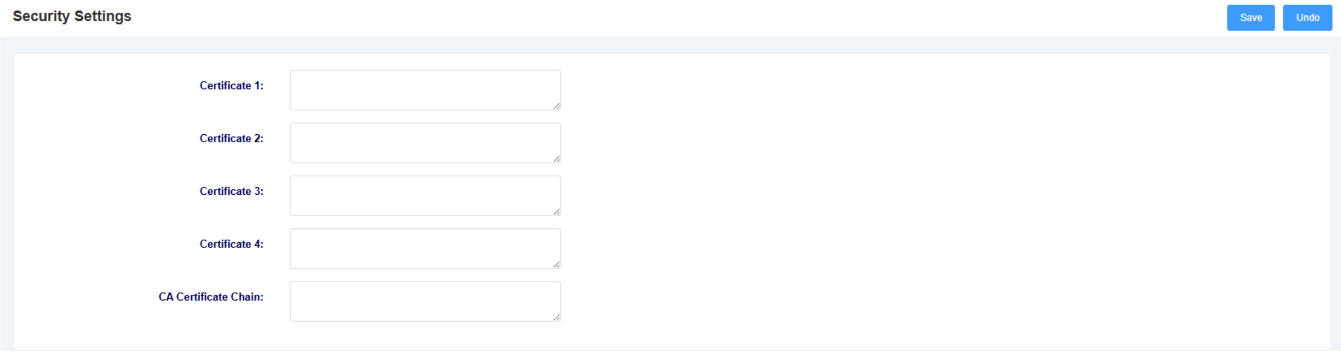
Figure 6-4-4 VoIP Settings



6.5 Security Settings

On this page, certificates can be uploaded.

Figure 6-5-1 Security Settings Interface



6.6 VEX

On this page, auto switching configuration can be set.

Figure 6-6-1 VEX Configuration Settings Interface

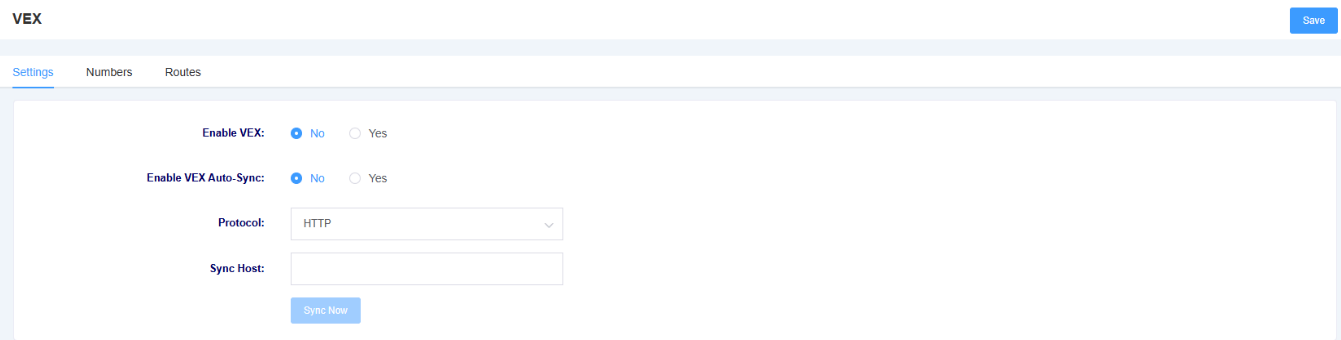


Table 6-6-1 VEX Parameters

Option	Description
Enable Auto Switching	Whether to enable auto switching function
Enable Auto Switching Auto Sync	Whether to enable auto switching auto sync function
Protocol	Freely select HTTP/HTTPS protocol
Sync Address	Enter the address to sync to

Figure 6-6-2 VEX Number Table Settings Interface

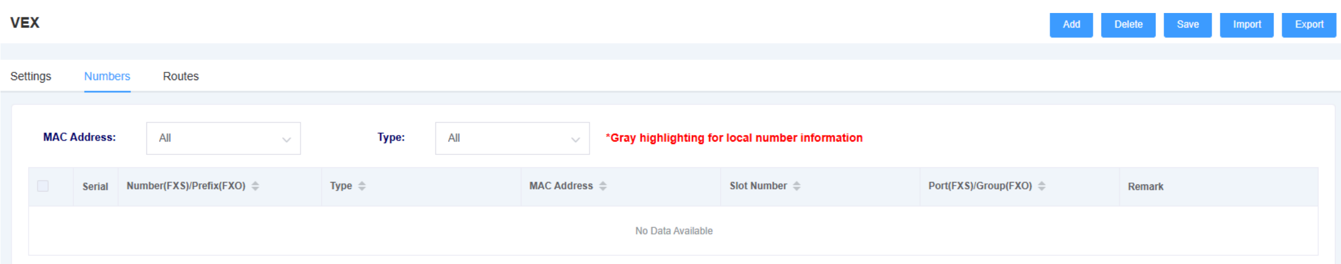
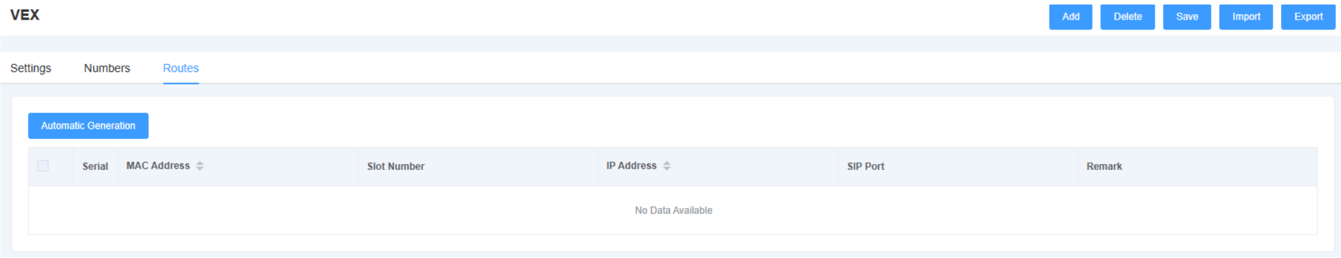


Figure 6-6-3 Routing Table Display Interface



7. Maintenance

7.1 Restart

On this page, restart functions can be configured, including system restart, network restart, and VOIP restart.

Figure 7-1-1 Restart

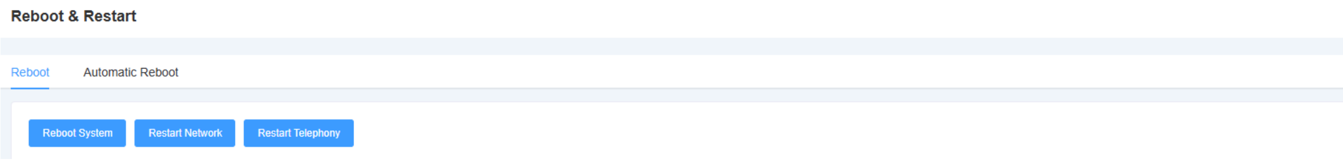
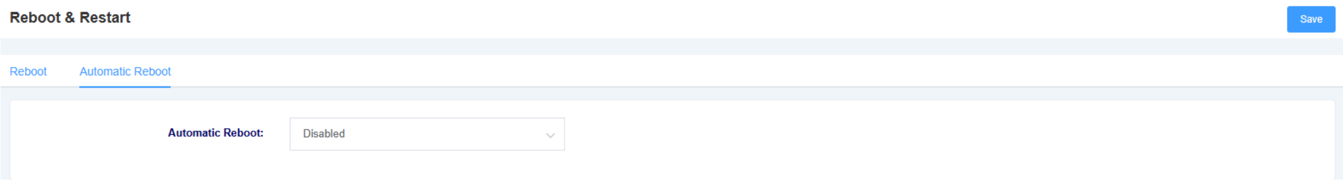


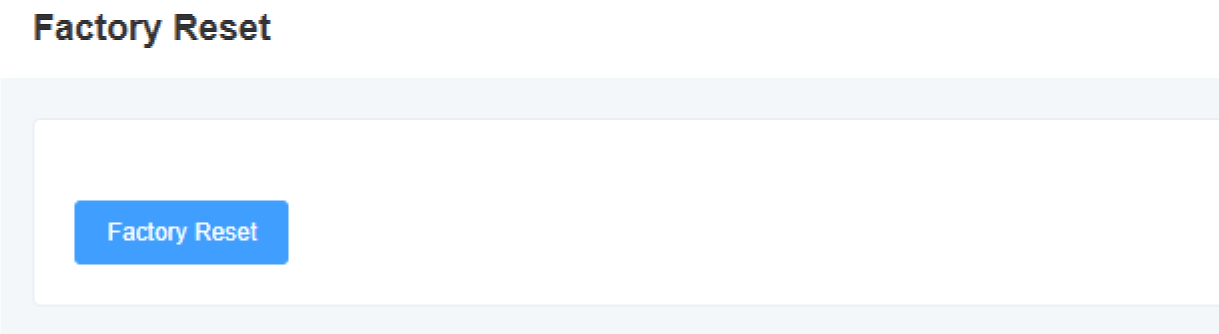
Figure 7-1-2 Auto Restart. You can adjust to disable auto restart, enable weekly auto restart, or enable daily auto restart.



7.2 Factory Reset

After clicking the factory reset button, the device will automatically restart and restore to factory settings.

Figure 7-2-1 Factory Reset Interface



7.3 Auto Deployment

The iAG200 has configuration file and firmware upgrade functions. These can be configured on this page.

Figure 7-3-1 Auto Deployment Interface

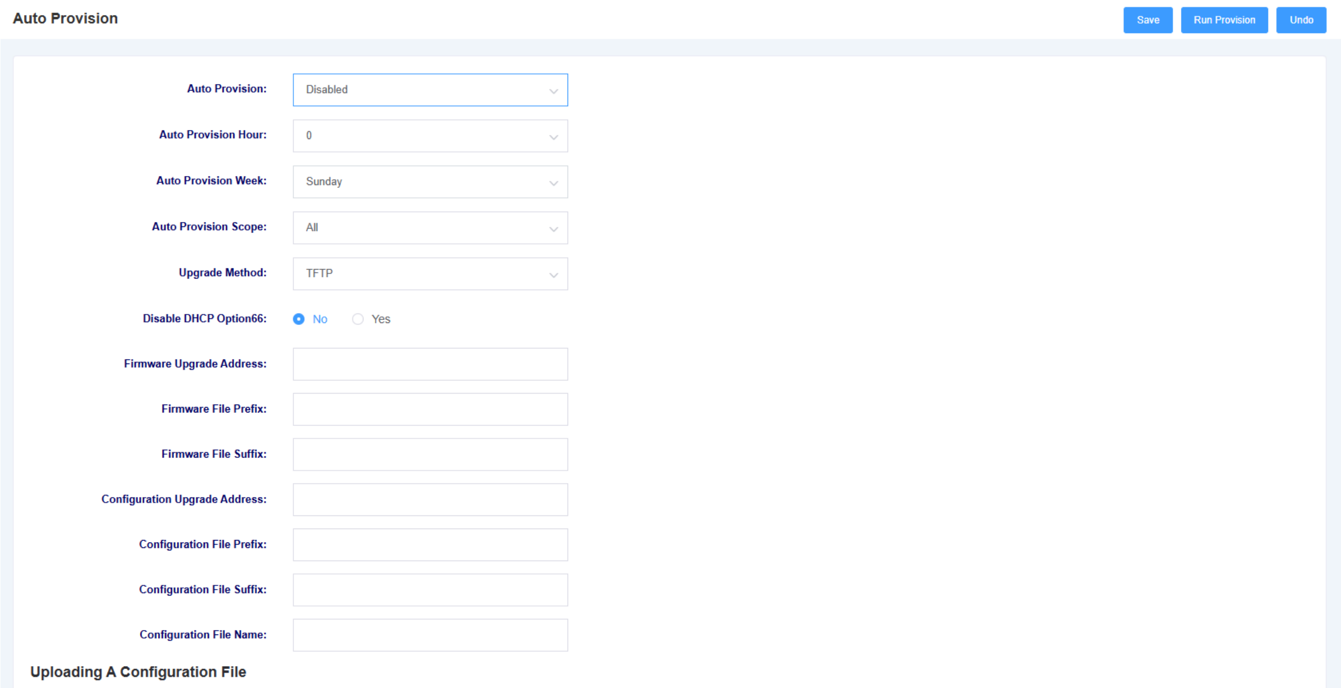


Table 7-3-1 Auto Deployment Parameters

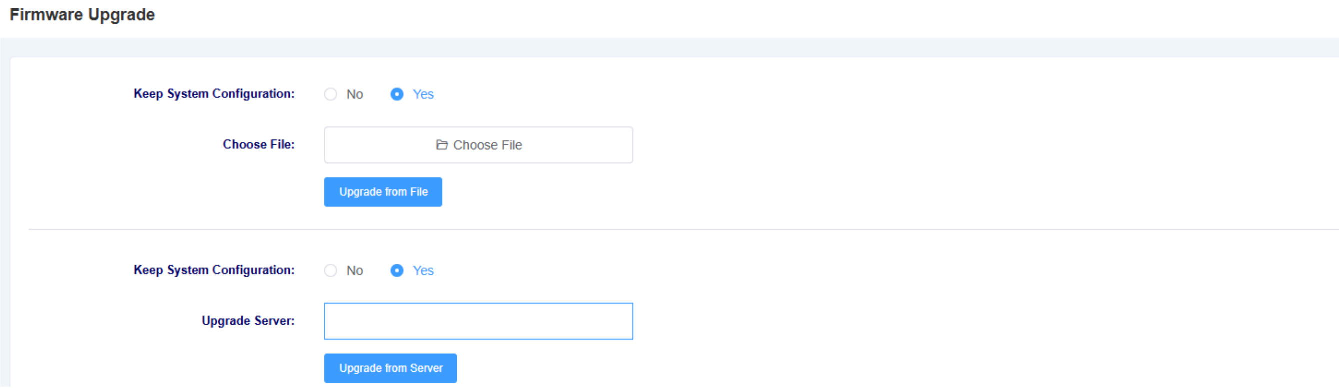
Option	Description
Auto Deployment	Set auto deployment mechanism. Can choose to deploy automatically after each boot or deploy at set time intervals
Auto Deployment Range	Select auto deployment range. Options: configuration files and firmware upgrades
Upgrade Method	Select auto deployment upgrade method. Supports tftp, http, https
Enable DHCP Option 66	Select whether to enable DHCP option 66 to get files
Option	Description
Firmware Upgrade Address	Set firmware upgrade path
Firmware File Prefix	Set firmware file prefix
Firmware File Suffix	Set firmware file suffix
Configuration Upgrade Address	Set configuration upgrade path
Configuration File Prefix	Set configuration file prefix
Configuration File Suffix	Set configuration file suffix
Upload Configuration	Upload configuration file
Download Configuration	Download device current configuration file

File names must be modified according to rules. Main control firmware file name rule: (pre)(firmware model).img(post). Interface board firmware file name rule: (pre)ixu(mac).img(post). Configuration file name rule: (pre)cfg(mac)(post). Pre is prefix, post is suffix. Prefix and suffix can be left empty.

7.4 Firmware Upgrade

On this page, firmware upgrade can be performed. Select the appropriate firmware type and upload the file to perform the upgrade. You can choose whether to keep system configuration. If not keeping system configuration, the device will clear system configuration after upgrade.

Figure 7-4-1 Firmware Upgrade



7.5 Time Settings

On this page, device time settings can be configured. Users can set the timezone and set the NTP server address to automatically synchronize time.

Figure 7-5-1 Time Settings

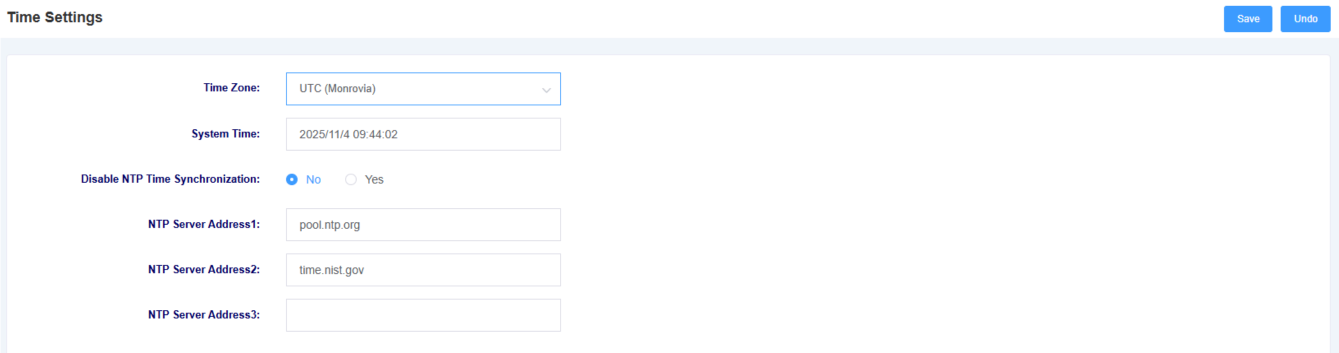


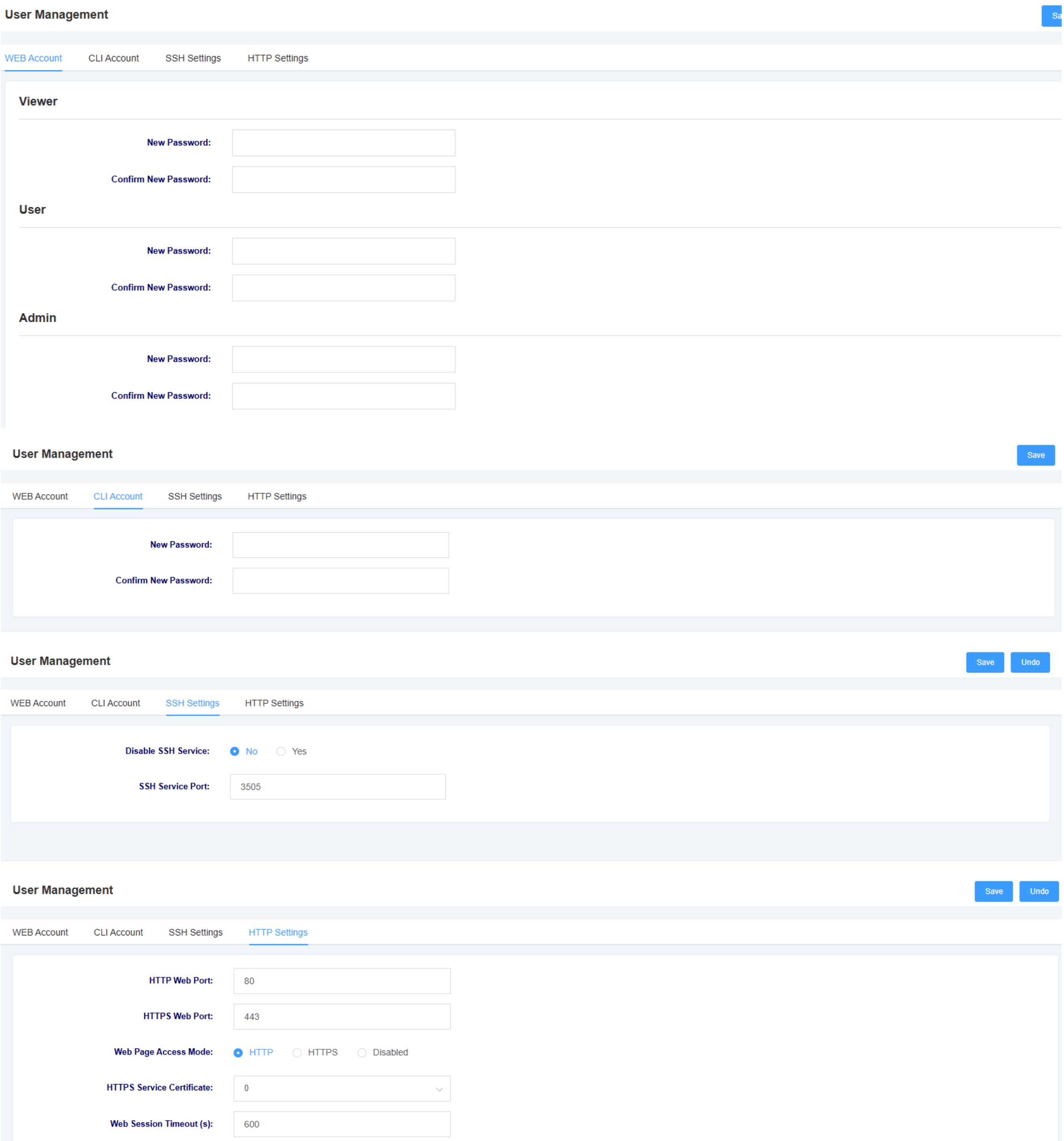
Table 7-5-1 Time Settings Parameters

Option	Description
Timezone	Set device timezone
System Time	Display system time
Enable NTP Time Sync	Select whether to enable NTP time sync
NTP Server Address	Set NTP server address

7.6 User Management

The AIU-8 supports different user roles to log in with different permissions. On the user management page, you can modify passwords, enable/disable SSH function, and configure HTTP settings for different roles.

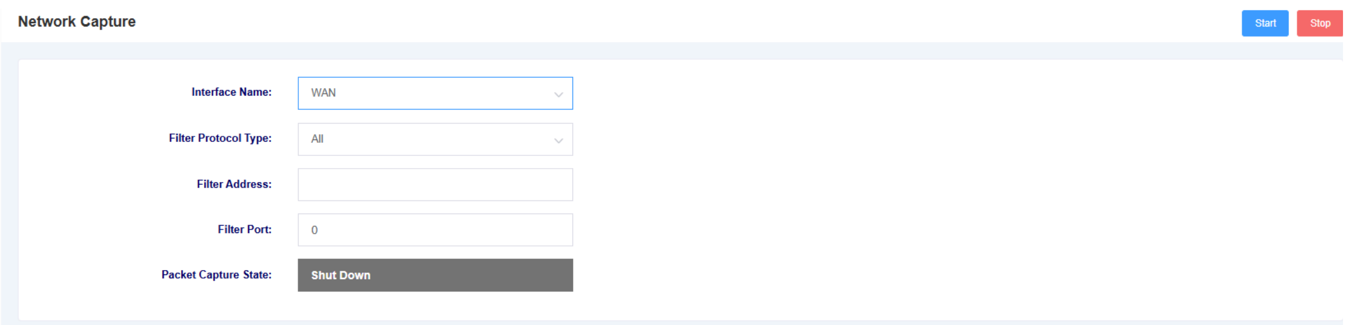
Figure 7-6-1 User Management



7.7 Network Packet Capture

The AIU-8 can conveniently locate network problems. Users can define the capture interface and select protocol type, address, and port on this interface.

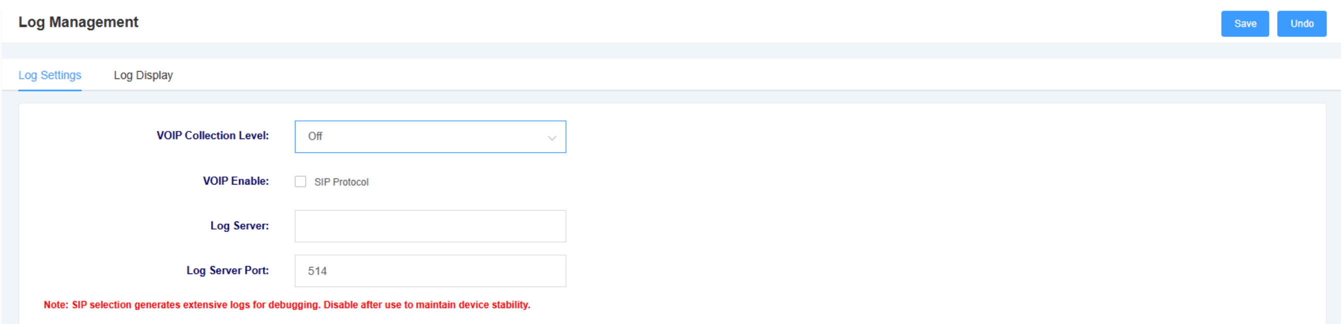
Figure 7-7-1 Network Packet Capture



7.8 Log Management

On the log management interface, you can set the log server address and port, and select kernel log level to facilitate viewing device logs for technical analysis.

Figure 7-8-1 Log Management



Syslog is commonly called system log or system record. It is a standard for transmitting record messages in Internet Protocol (TCP/IP) networks. This term commonly refers to the actual syslog protocol or applications/databases that send syslog messages. The syslog protocol is a client-server protocol: syslog senders transmit small text information (less than 1024 bytes) to syslog receivers. The receiving end is usually called "syslogd", "syslog daemon", or syslog server. System log messages can be transmitted via UDP protocol and/or TCP protocol.

Syslog Level Introduction:

• EMERG Failure

• ALERT Warning

• CRIT Needs immediate solution

• ERROR Error condition that prevents tool or some subsystem partial functionality

• WARNING Warning message

• NOTICE Normal condition with importance

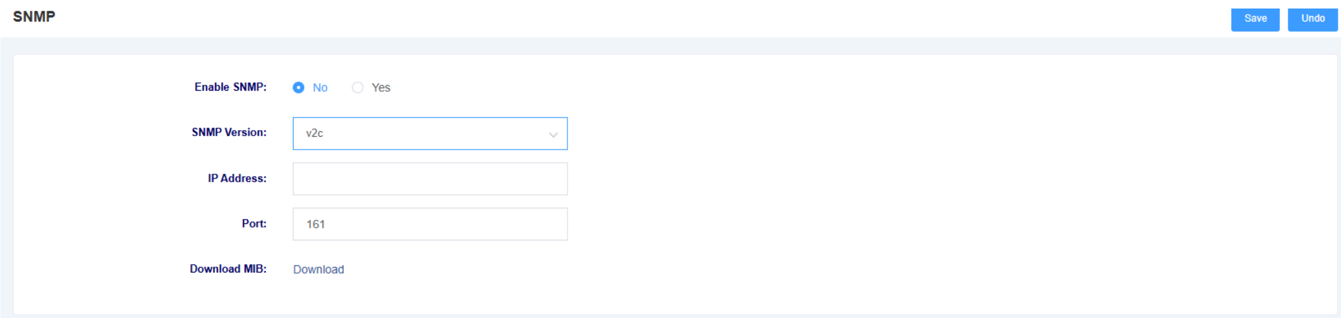
• INFO Information

• DEBUG Other information without function conditions or problems

7.9 SNMP

On this page, SNMP service-related information can be set. The AIU-8 supports SNMPv1, v2c.

Figure 7-9-1 SNMP



7.10 Cloud Management

On this page, cloud management-related information can be set. The AIU-8 supports Openvox cloud management function. After entering the server address, port, and binding code, you can manage the device on the cloud management platform.

Figure 7-10-1 Cloud Management Settings

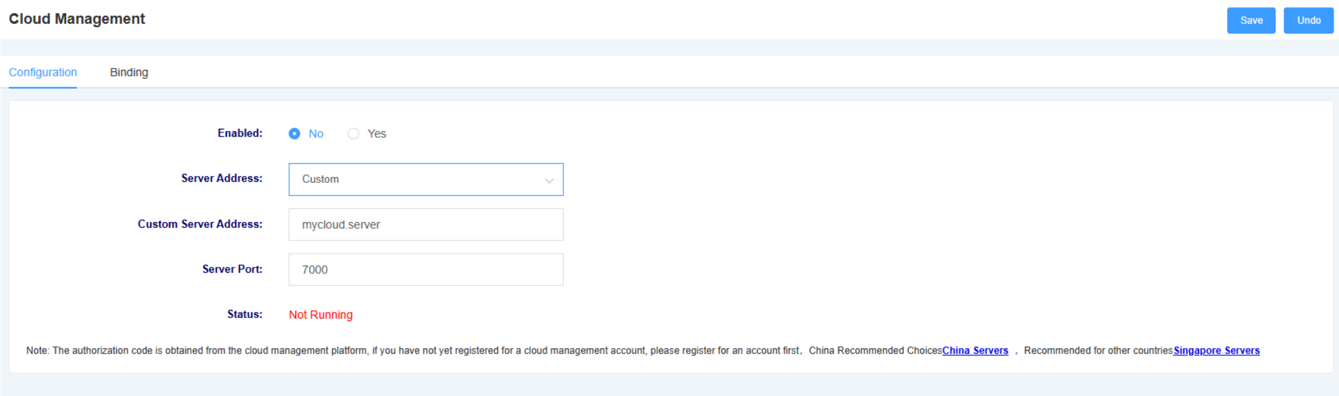
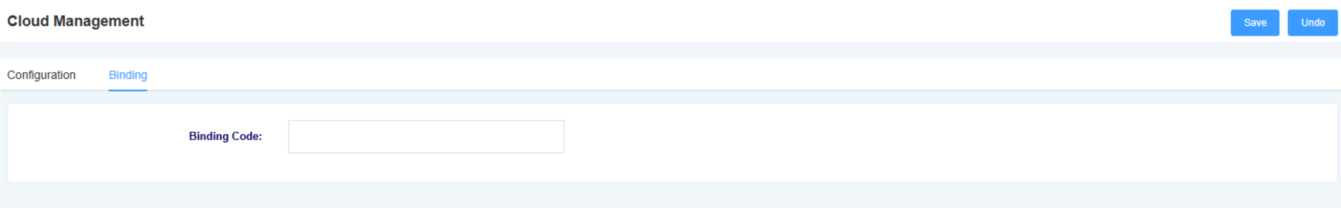
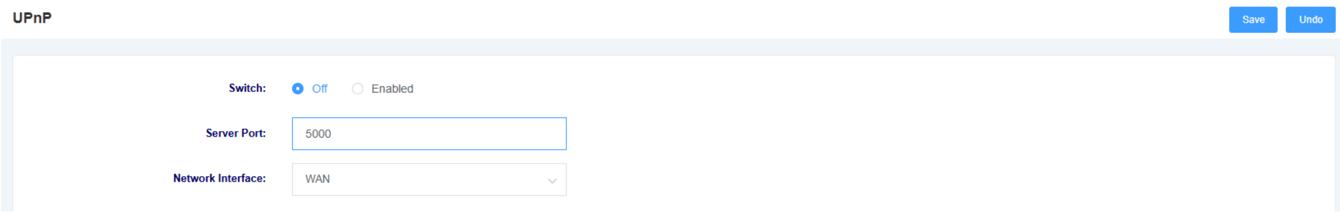


Figure 7-10-2 Cloud Management Binding



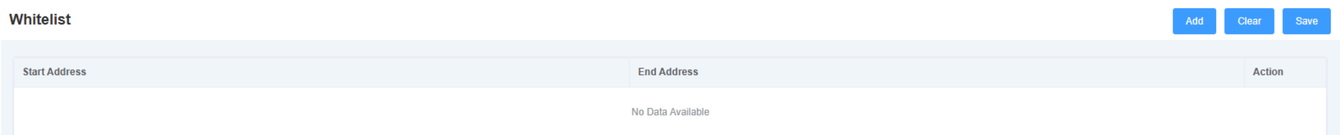
7.11 UPnP

On this page, UPnP-related information can be set. When enabled, applications in the LAN can automatically request port mapping from the router and perform device discovery and control, thereby achieving zero-configuration internal network service exposure and interconnection.



7.12 Whitelist

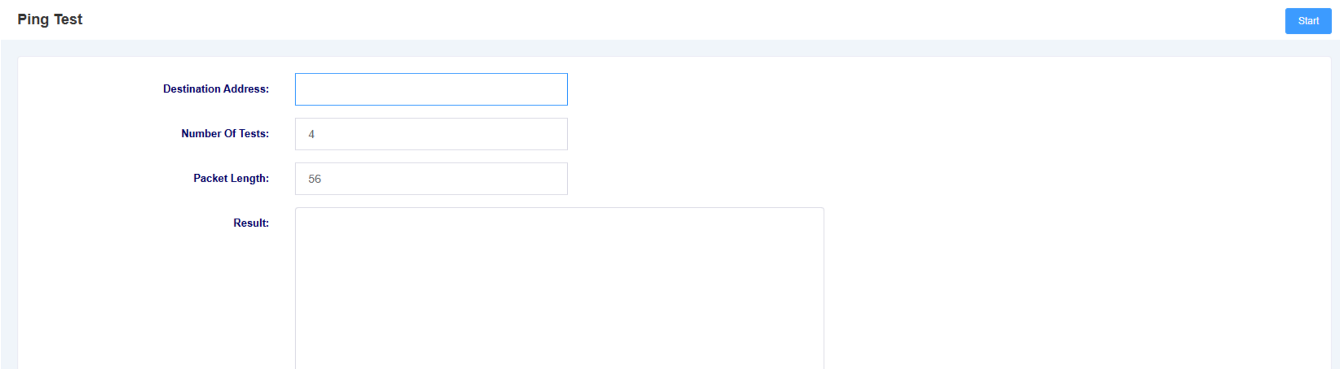
On this page, whitelist-related information can be set. After configuration, only IPs in the whitelist can access the device.



7.13 Ping Test

On this page, the ping command can be used to test network connectivity.

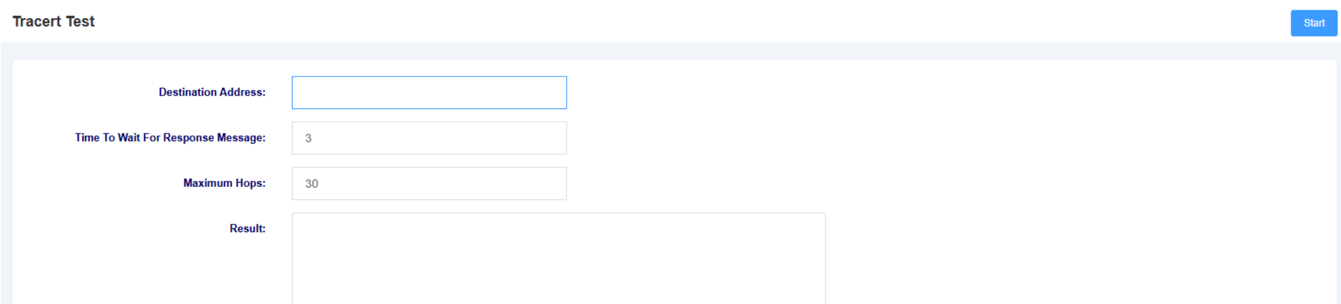
Figure 7-13-1 Ping Test



7.14 Tracert Test

On this page, the tracert command can be used to test network connectivity.

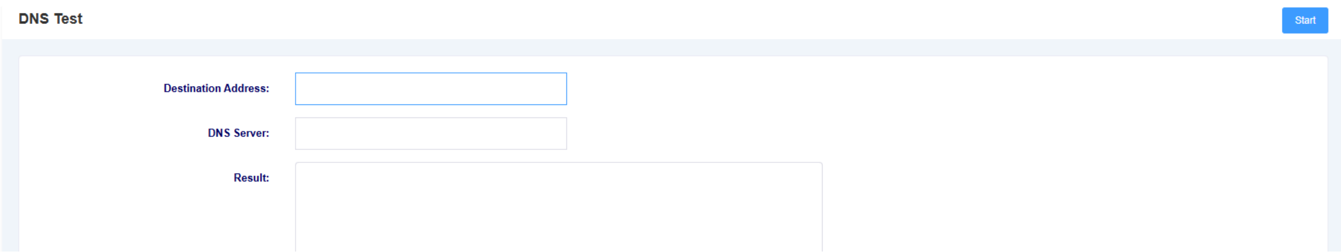
Figure 7-14-1 Tracert Test



7.15 DNS Test

On this page, DNS testing can be performed on a specified DNS.

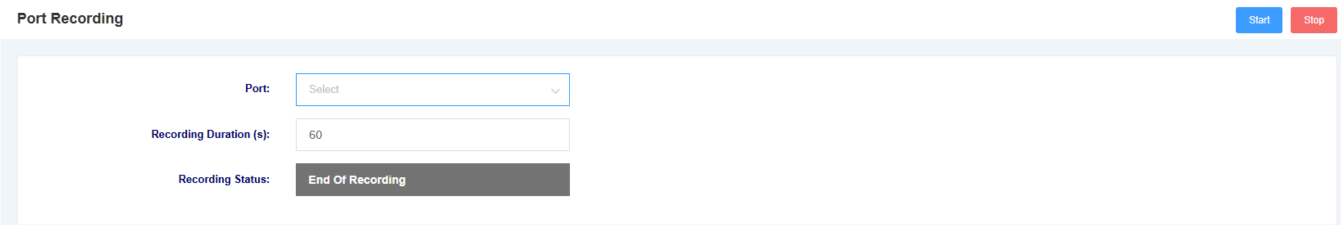
Figure 7-15-1 DNS Test



7.16 Port Recording

On this page, a specified port can be selected for recording to troubleshoot problems.

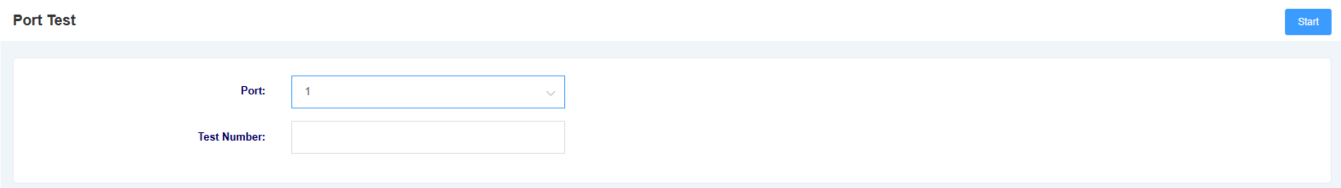
Figure 7-16-1 Port Recording



7.17 Port Test

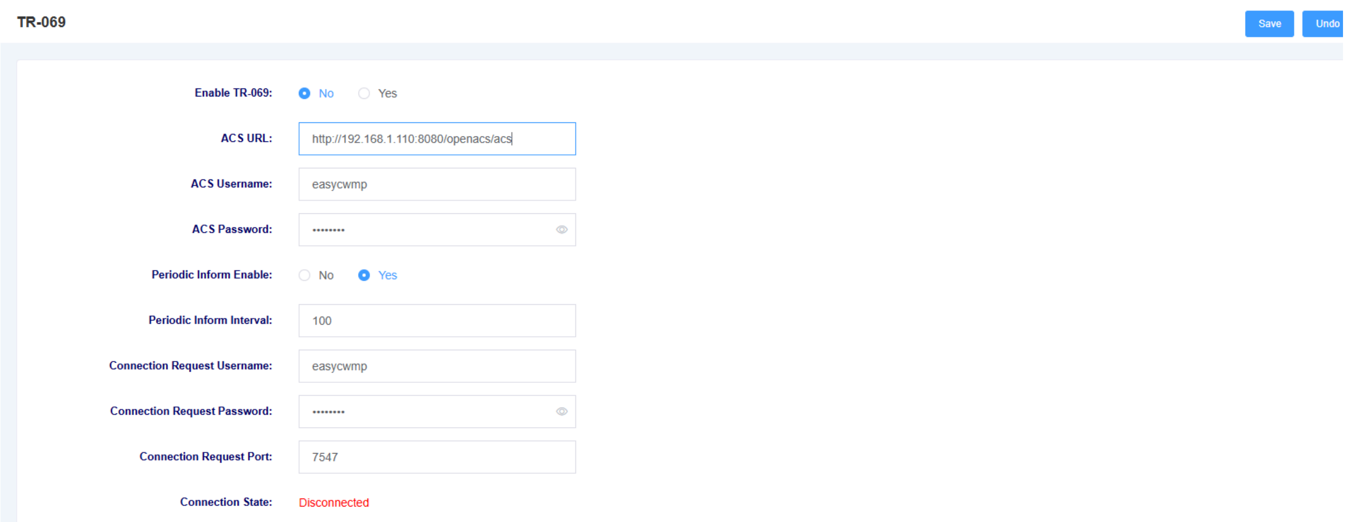
On this page, a specified port can be selected for port testing to quickly detect if the port is functioning normally.

Figure 7-17-1 Port Test



7.18 TR-069

On this page, TR-069-related information can be set. When enabled, the device will establish a secure connection with the ACS to realize automatic configuration, remote upgrade, parameter delivery, and status reporting centralized operation and maintenance functions.



- Enable TR-069: Whether to enable CWMP-based remote management, which establishes a session with the ACS for provisioning, monitoring, and upgrades.

- ACS Address: The ACS URL the device connects to (including scheme/host/port/path); used to discover and initiate the management session with the ACS.
- ACS Username: Account name used by the CPE for HTTP/HTTPS authentication when connecting to the ACS.
- ACS Password: Authentication credential paired with the ACS username, used to establish a secure session.
- Enable Periodic Connection: Whether to enable periodic Inform reports to upload device status and poll for new commands.
- Periodic Connection Interval (seconds): PeriodicInformInterval, the time interval for periodic Inform messages, in seconds.
- ACS Connection Request Username: Username validated on the device when the ACS initiates a Connection Request to the device.
- ACS Connection Request Password: Credential paired with the above username to verify the origin of the ACS-initiated request.
- ACS Connection Request Port: Local port on which the device listens for ACS Connection Requests; must be allowed/mapped on firewall/NAT.
- Connection Status: Indicates whether the current session/registration with the ACS is successful, helping assess the health of the management link.

Glossary

• DNS: Domain Name System

• SIP: Session Initiation Protocol

• TCP: Transmission Control Protocol

• UDP: User Datagram Protocol

• RTP: Real Time Protocol

• PPPOE: Point-to-Point Protocol Over Ethernet

• VLAN: Virtual Local Area Network

• ARP: Address Resolution Protocol

• CID: Caller Identity

• DND: Do NOT Disturb

• DTMF: Dual Tone Multi Frequency

• NTP: Network Time Protocol

• STUN: Simple Traversal of UDP over NAT

• PSTN: Public Switched Telephone Network