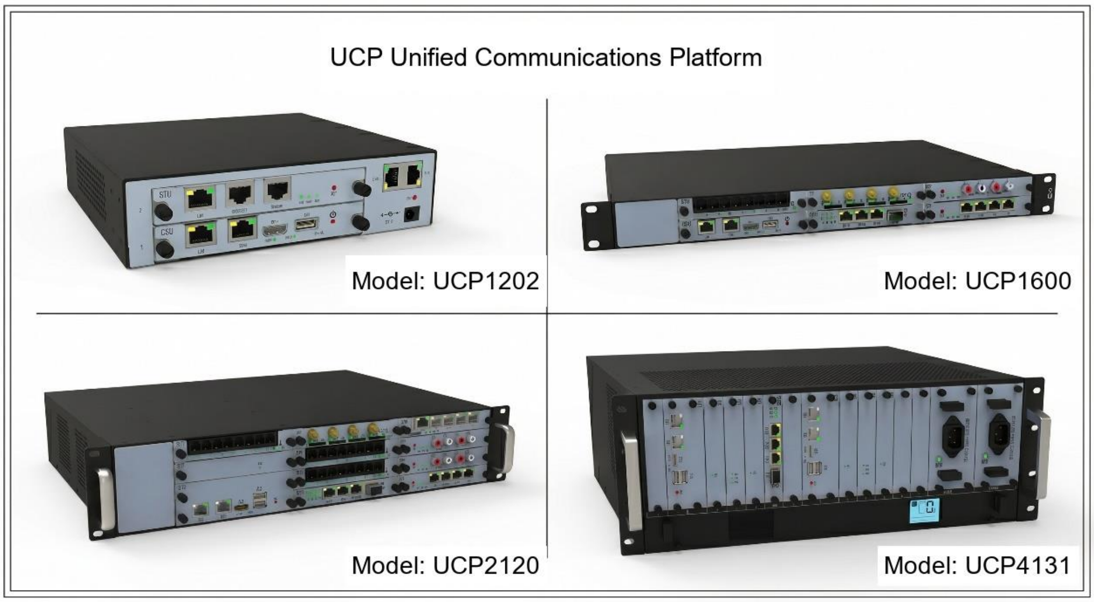
UCP Unified Communications Platform



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Welcome

Thank you for choosing the UCP Unified Communications Platform. We hope you make full use of this feature-rich communications platform. If you need technical support, please contact us at +86-755-66630978.

About This Manual

This manual introduces the UCP Unified Communications Platform. Please read this manual carefully before configuring its features.

Declaration

OpenVox Communication Co., Ltd. hereby declares that this series complies with the essential requirements and other relevant provisions of CE and FCC. The information in this document is subject to change without notice. OpenVox Communication Co., Ltd. makes no warranties with respect to this guide, including but not limited to the implied warranties of merchantability and fitness for a particular purpose. OpenVox Communication Co., Ltd. assumes no liability for any errors contained in this guide, nor for any incidental or consequential damages arising from the furnishing, performance, or use of this guide.

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- Documentation Guide

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- General convention

Describes common conventions in documents.

- Revision history

Introduce the revision version of the document, revision date, etc.

1.2 Documentation Guidelines

Introduces the architecture, content and acquisition methods of supporting materials for the unified communications platform.

Content introduction

The content of the unified communications platform supporting materials is briefly described in Table 1.

Table 1 Introduction to unified communications platform materials

Classification	Document name	Content and positioning	Technical support engineer	Enterprise users
product description	Product positioning and highlights	If you need to understand the highlights, architecture, features, functions, operation and maintenance methods, and networking applications of the unified communications platform, you can refer to the content described in this document.	√	√

Classification	Document name	Content and positioning	Technical support engineer	Enterprise users
UCP1600/2120/4131 product structure	You can use this document to learn about the product structure, hardware structure, parameter attributes, etc. of the unified communications platform.	√	√	
Technical indicators	You can use this document to learn about the physical parameters, performance and capacity, interfaces and protocols, and standards followed by the unified communications platform.	√	√	
Application scenarios	You can learn about unified communications platform application scenario examples through this document.	√	√	
Installation and configuration	Quick installation guide	During deployment, you can complete the hardware wiring, configuration, and power-on operations by referring to this document.	√	√
Business board interconnection configuration	You can use this document to learn how to connect the unified communications platform to the system.	√	-	
User Manual	You can use this document to find the user manual for each accessory.	√	-	

1.3 General Conventions

Describes common conventions in documents.

Common formatting conventions

Format	Description
Song Dynasty	The main text is expressed in Song font, with size 11 font.
black body	First-level titles are in boldface, and the first-level font is size 21. Second-level and third-level titles are in Song font size, with the second-level font size being 15 and the third-level font being 13 size.
regular script	Warnings, tips and other content should all be in italics, and lines should be added before and after the content to separate it from the main text.
"Terminal Display" format	The "Terminal Display" format represents screen output information. In addition, the information input by the user from the terminal included in the screen output information is shown in bold font.

Mouse operation convention

Format	meaning
click	Quickly press and release a mouse button.
Double click	Press and release a mouse button twice in succession.
Drag	While holding down a mouse button, move the mouse.

1.4 Revision history

Introduce the revision version of the document, revision date, etc.

version number	Release date	Description
1.0	15/07/2022	First release of Chinese version
2.0	10/5/2023	New modules CCU-I-TGL and DTU-301
3.0	30/6/2023	Added single board module user manual
4.0	31/8/2023	Added MIU module product description and quick installation manual
5.0	15/8/2024	Updated business version user manual and CCU-I-TGL master control information
6.0	3/4/2026	Added UCP1202 and deleted discontinued main control information

2. Product Overview

2.1 Product description

Introduce the characteristics, structure, features and functions of the unified communications platform.

- Product positioning and highlights Describe the product positioning and product highlights of the UCP series unified gateway.
- UCP1202/UCP1600/UCP2120/UCP4131 product structure Mainly introduces the hardware structure of UCP1202/UCP1600/UCP2120/UCP4131 such as chassis, single boards, cables and so on.
- Technical indicators Introduce the key technical indicators of the unified communications platform system.

2.2 Product positioning and highlights

Describe the product positioning and product highlights of the UCP series unified communications platform.

- Product positioning The UCP series unified communications platform is a switching device for IP voice solutions, meeting the communication needs of enterprises of different sizes and types, and providing professional IP voice solutions.
- Product Highlights The unified communications platform has rich business and interface capabilities, and is easy to deploy and maintain.

2.3 Product positioning

The UCP series unified communications platform is a switching device for IP voice solutions, meeting the communication needs of enterprises of different sizes and types, and providing professional IP voice solutions.

The product models and application scenarios of UCP series unified communications (hereinafter referred to as "UCP") are shown in Table 1.

Table 1 UCP series product models and application scenarios

Product model	Application scenarios
ucp1202 	The UCP1202 supports one main control board slot and one service board slot, which can meet the service requirements of less than 100 users.
UCP1600 	The voice switching equipment of small and medium-sized enterprises can also be used as the local access gateway of small and medium-sized branches to meet the business needs of less than 100 users. UCP1600 takes up less space (only 1U in height).

Product model	Application scenarios
UCP2120 	The voice switching equipment of medium-sized enterprises can also be used as the local access gateway of medium-sized branches to meet the business needs of 300 to 1000 users (height is 2U).
UCP4131 	The voice switching equipment of large enterprises can also be used as the local access gateway of large branches to meet the business needs of 1,000 to 10,000 users. UCP4131 takes up less space (4U height).

The product adopts SIP soft switching core, with high integration and integrated broadband and narrowband design, which can effectively improve communication efficiency and reduce operating costs.

The product realizes hybrid networking of analog phones and IP phones:

- Direct access to local analog phones.
- Use the IP bearer network to access the analog phone through the access gateway IAD (hereinafter referred to as IAD).
- Use the IP bearer network to access IP phones.
- The product connects to PSTN (Public Switched Telephone Network) or private network voice switching equipment through digital or analog trunks and broadband SIP trunks.

2.4 Product Highlights

UCP has rich business and interface capabilities, and is easy to deploy and maintain.

Rich interface capabilities

- Can be expanded to multiple types of telecommunications interfaces, analog/digital E1&T1/GSM/LTE/wireless intercom/audio broadcast/magnet.
- The main control and various business boards adopt modular design.

High reliability, low cost

- It adopts carrier-grade software and hardware platform and a customized operating system based on Linux.
- UCP2120 and UCP4131 support 1+1 power backup.
- UCP4131 supports active and standby main control boards. Business interruptions caused by main board network or hardware failures can be restored on the standby board (main and standby board software support is required).
- As an IP voice switching device, it can reduce enterprise operation and maintenance costs.

Flexible deployment and easy maintenance

- It supports multiple networking modes such as single-node networking and distributed networking, and is flexible in deployment.
- All service boards can be hot-swapped during system operation.
- All business boards have a built-in Web management system, providing quick deployment and daily configuration functions.

3. Hardware structure

3.1 UCP1202/UCP1600/2120/4131 product structure

Mainly introduces the hardware structure of UCP1202/UCP1600/2120/4131 such as chassis, single boards, cables, etc.

- UCP1202 chassis The UCP1202 supports one main control board slot and one service board slot, which can meet the service requirements of less than 100 users.
- UCP1600 chassis The UCP1600 chassis provides a centralized and interconnected space for internal components, while preventing component contamination and protecting components from damage caused by external factors.
- UCP2120 chassis The UCP2120 chassis provides a centralized and interconnected space for internal components, while preventing component contamination and protecting components from damage caused by external factors.
- UCP4131 chassis The UCP4131 chassis provides a centralized and interconnected space for internal components, while preventing component contamination and protecting components from damage caused by external factors.
- Veneer The single boards of UCP1600/UCP2120/UCP4131 include main control board (CCU), wireless user service unit (WTU), analog user service unit (AIU), audio broadcast service unit (ACU), wireless trunking service unit (RIU) and digital trunking service unit (DTU).
- power supply UCP2120/UCP4131 can support dual power supply. For UCP2120/UCP4131, when configured with dual power modules, current sharing and backup are supported.
- Cable The cables of UCP1600/UCP2120/UCP4131 include digital trunk cables, high-density user cables and DC power cables.

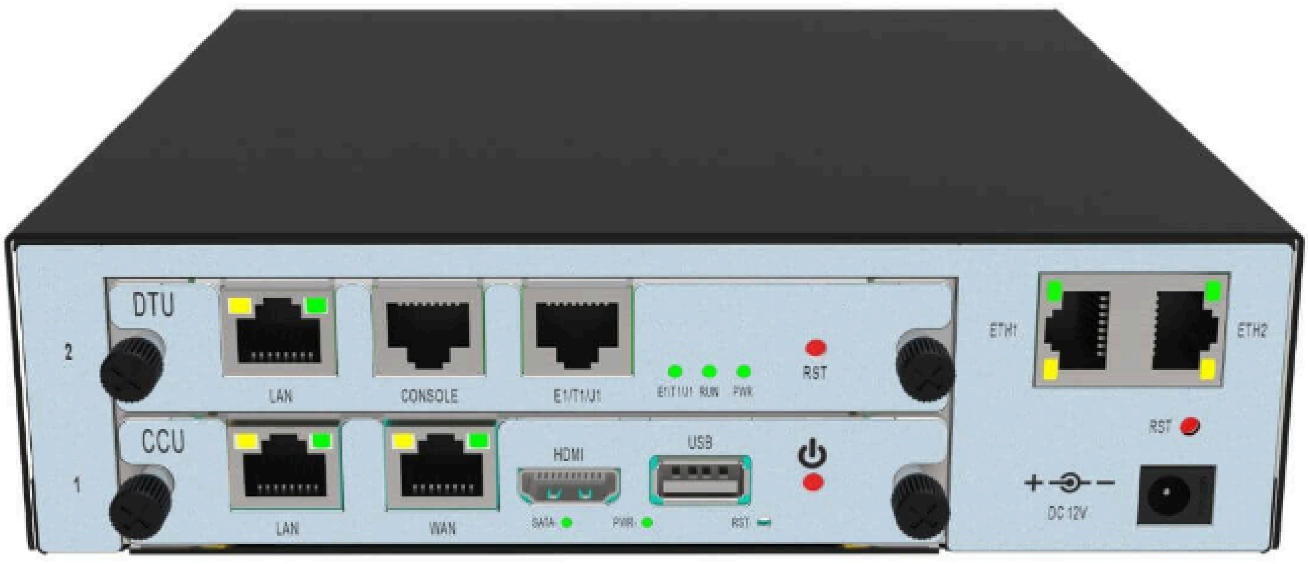
3.2 Chassis structure

3.2.1 UCP1202 chassis

The UCP1202 chassis provides a centralized and interconnected space for internal components, while preventing component contamination and protecting components from damage caused by external factors.

Appearance

UCP1202 is 188mm wide, 202mm deep, and 50mm high. Its appearance is shown in Figure 1.



slot

The slots are located on the front of the chassis. UCP1202 provides 1 main control board slot and 1 service board slot.

- Slot 2 is the service board slot, used to install various service boards.
- Slot 1 is the main control board slot, used to install the CCU-N-ALDER main control board.
- The general service board (except ACU/RIU) automatically identifies the initial IP based on the slot number. The corresponding IP for slot 2 is `172.16.80.2`.

The slot distribution map is shown in Figure 1

1
2

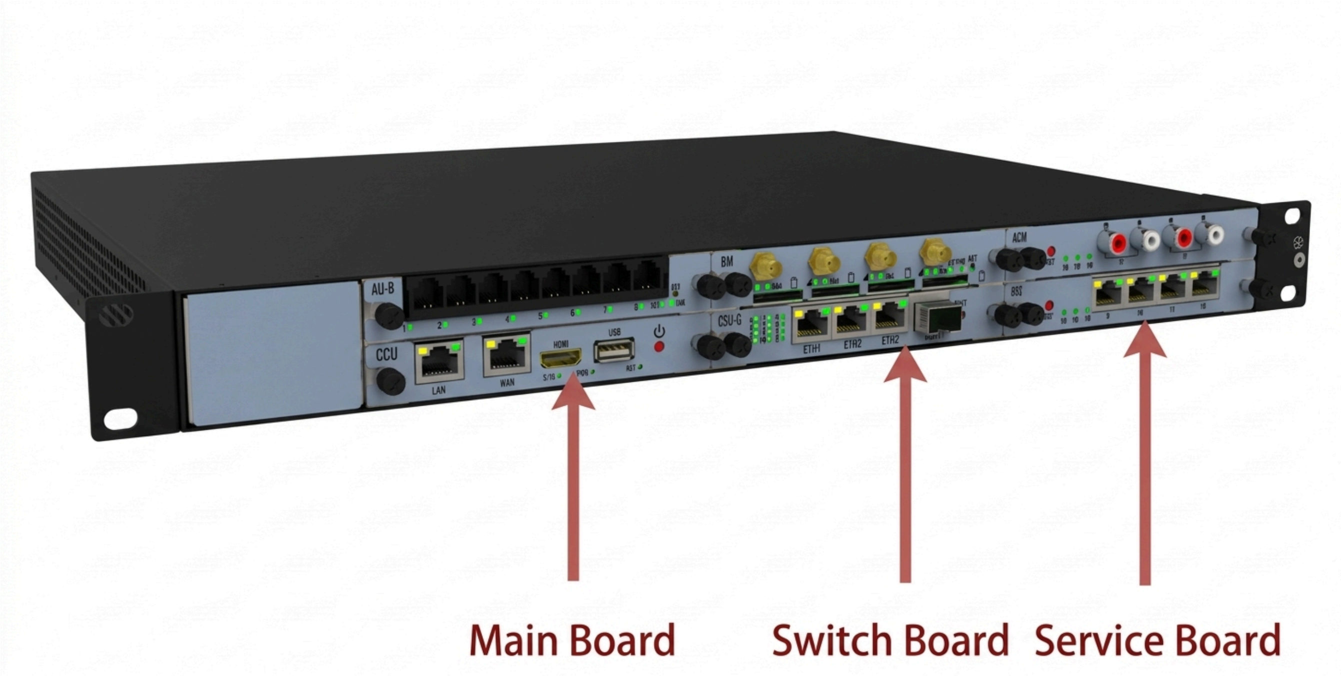
3.2.2 UCP1600 chassis

The UCP1600 chassis provides a centralized and interconnected space for internal components, while preventing component contamination and protecting components from damage caused by external factors.

Appearance

UCP1600 adopts a 1U standard chassis with a width of 434mm, a depth of 330mm, and a height of 44mm. It can be installed in a 19-inch cabinet that complies with IEC (International Electrotechnical Commission) standards. Its appearance is shown in Figure 1.

Figure 1 Appearance of the chassis



slot

The slots are located on the front of the chassis. UCP1600 provides 1 main control board slot and 4 service board slots.

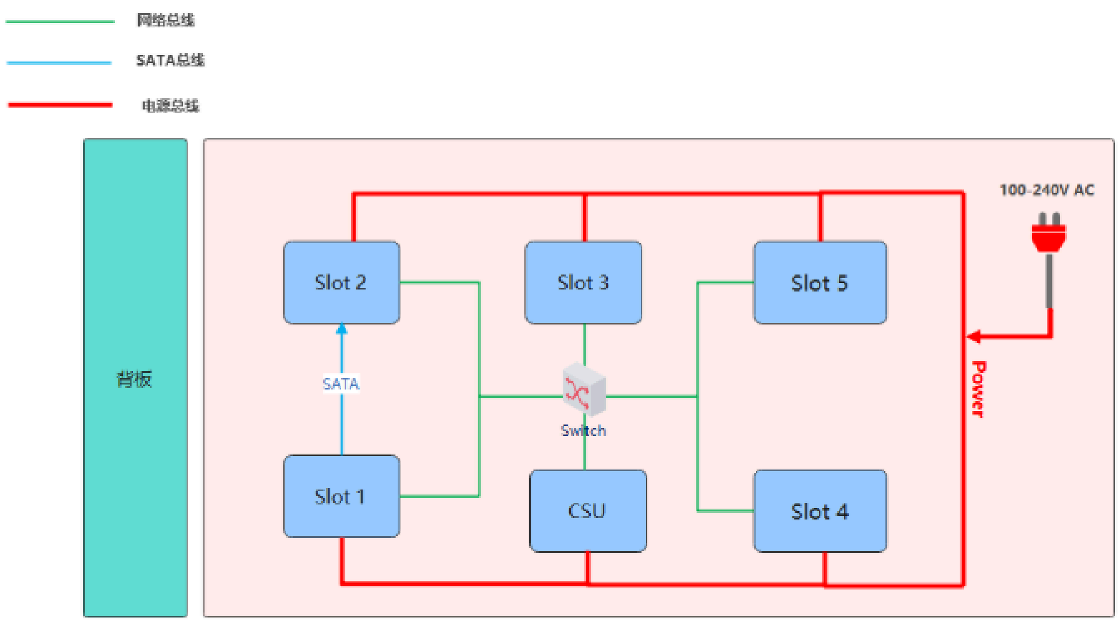
- Slots 2 to 5 are service board slots, which are used to install various service boards and support mixed insertion.
- Slot 1 is the main control board slot, used to install the **CCU-N-ALDER** main control board.
- If you want to expand the hard disk board on the main control board, you can insert the hard disk board into slot 2 and use it together.

The slot distribution and internal logic diagram are shown in Figure 2 and Figure 3.

Figure 2 Chassis slot distribution

2	3	5
1	CSU-F/G	4

Figure 3 Internal logic diagram



Explanation

- The UCP1600 service board can be selected according to the system capacity. If an empty slot is not configured with an interface board, you need to install an empty panel.
- To ensure the normal use of basic functions, UCP1600 must be equipped with only one main control board.
- If you use mounting ear screws to install the device on the cabinet, you do not need to use M4 grounding screws; if the chassis is installed on a tray and you do not use mounting ear screws, you need to use M4 grounding screws for grounding.
- The service board (except ACU and RIU) will identify the initial IP address based on the slot number, for example, slot 2: 172.16.80.2

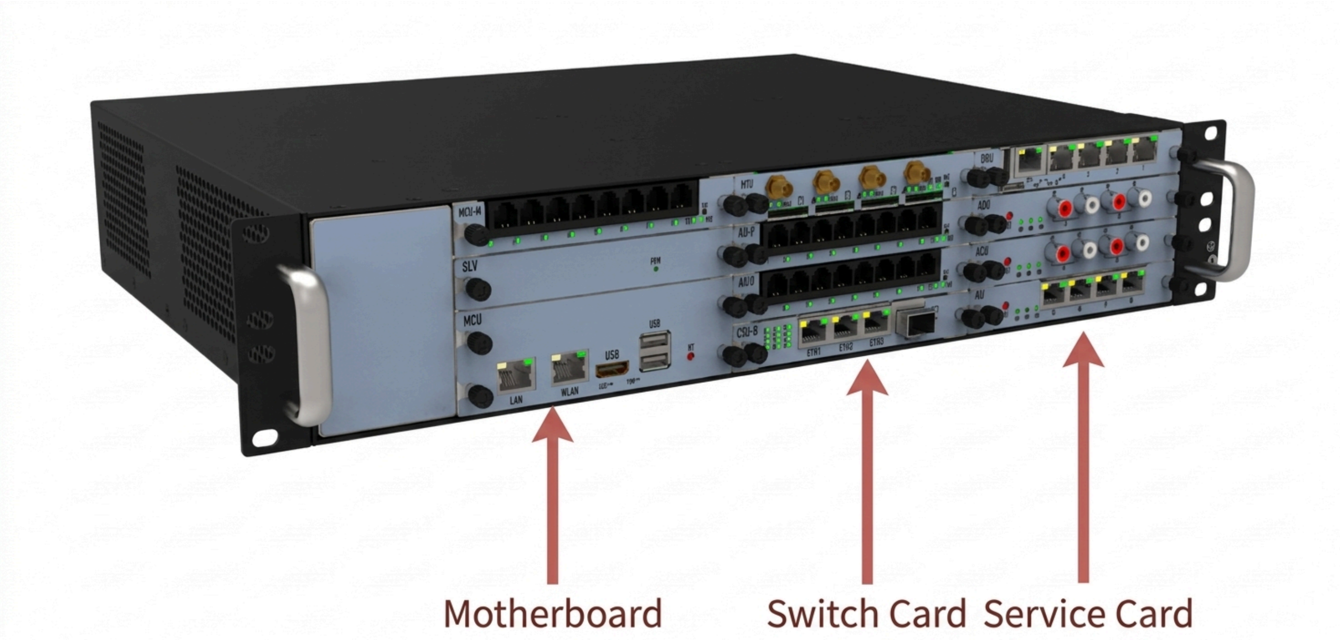
3.2.3 UCP2120 chassis

The UCP2120 chassis provides a centralized and interconnected space for internal components, while preventing component contamination and protecting components from damage caused by external factors.

Appearance

UCP2120 adopts a 2U standard chassis with a width of 434mm, a depth of 330mm, and a height of 88mm. The chassis can be installed in a 19-inch cabinet that complies with IEC (International Electrotechnical Commission) standards. Its appearance is shown in Figure 1.

Figure 1 Appearance of the chassis



slot

The slots are located on the front of the chassis. UCP2120 provides 2 main control board slots and 9 service board slots.

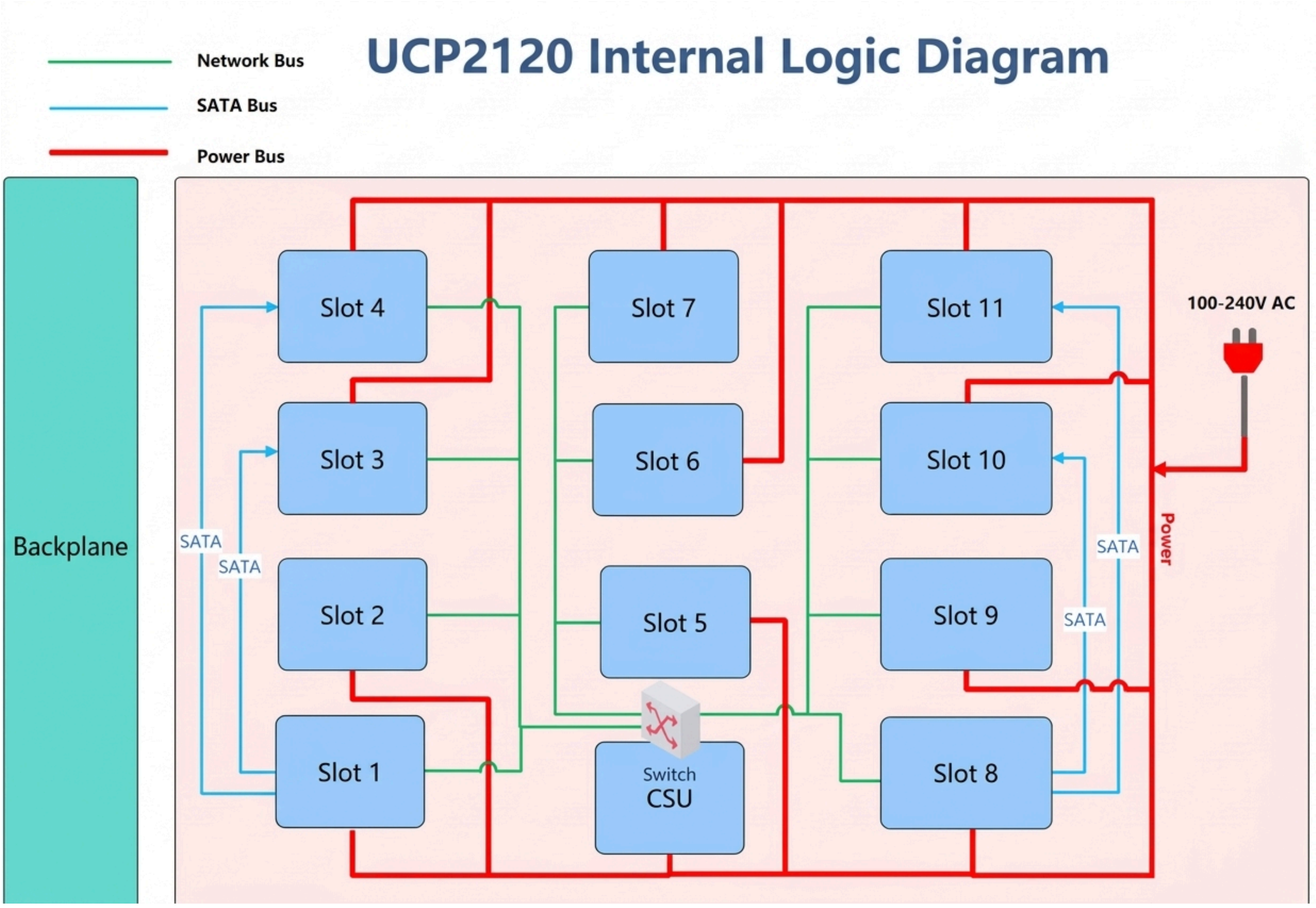
- Slots 2-7 and 9-11 are service board slots, which are used to install various service boards and support mixed insertion.
- Slots 1 and 8 are main control board slots, which are used to install CCU series main control boards and can support dual main control boards.
- If the main control board wants to expand the hard disk board, the hard disk board can be inserted into slots 3 and 10. For example, the main control board is in slot 1 and the hard disk board can be placed in slot 3.
- If you want to expand the hard disk board for the CCU-N-GML and CCU-I-KABYLR main control boards, you can insert the hard disk board into slots 3-4 and 10-11. For example, the main control board is in slot 1, and the hard disk board can be placed in slots 3 and 4.

The slot distribution and internal logic diagram are shown in Figure 2 and Figure 3.

Figure 2 Chassis slot distribution

4	7	11
3	6	10
2	5	9
1	CSU-F/G	8

Figure 3 Internal logic diagram



Explanation

- The UCP2120 interface board can be selected according to the system capacity. If an empty slot is not configured with an interface board, you need to install an empty panel.
- To ensure normal use of basic functions, UCP2120 needs to be equipped with a main control board.
- If you use mounting ear screws to install the device on the cabinet, you do not need to use M4 grounding screws; if the chassis is installed on a tray and you do not use mounting ear screws, you need to use M4 grounding screws for grounding.
- The service board (except ACU and RIU) will identify the initial IP address based on the slot number, for example, slot 2: 172.16.80.2

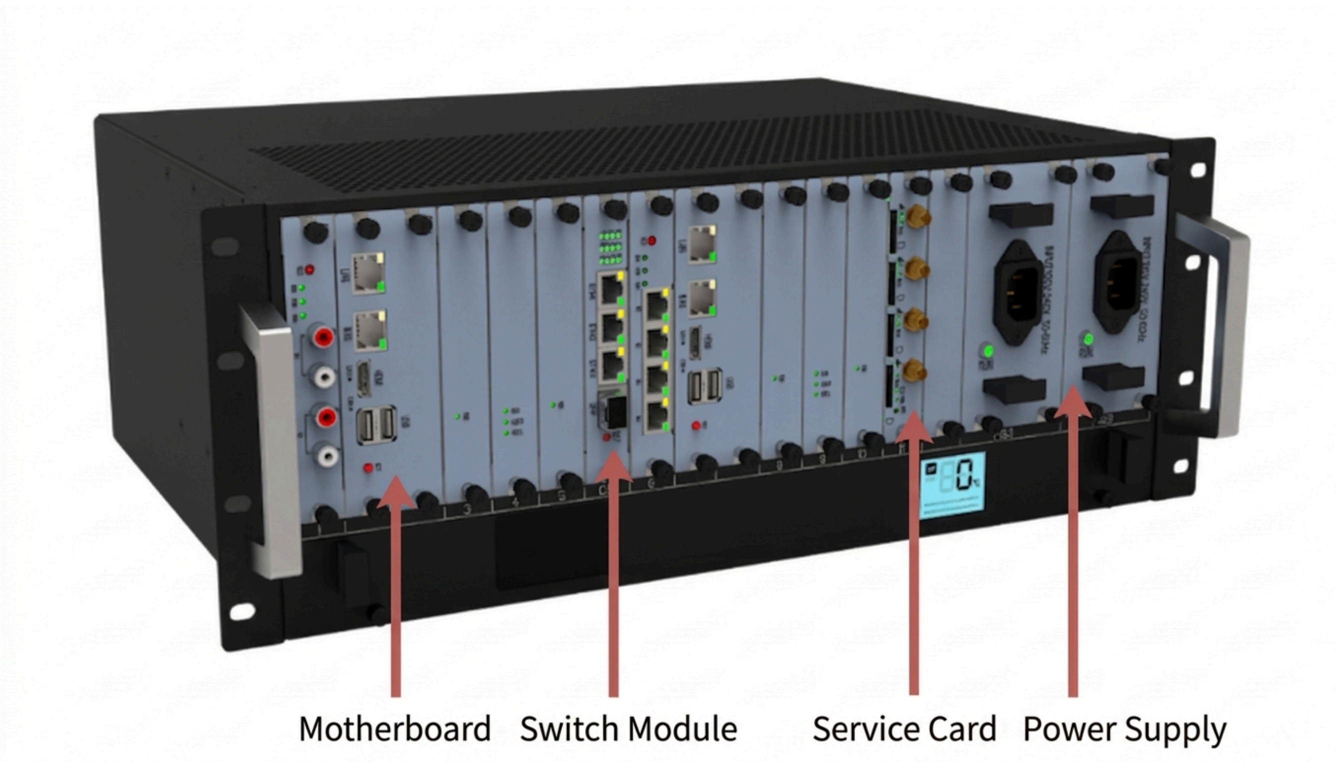
3.2.4 UCP4131 chassis

The UCP4131 chassis provides a centralized and interconnected space for internal components, while preventing component contamination and protecting components from damage caused by external factors.

Appearance

UCP4131 uses a 4U standard chassis with a width of 435.8mm, a depth of 330mm, and a height of 176.8mm. The chassis can be installed in a 19-inch cabinet that complies with IEC (International Electrotechnical Commission) standards. Its appearance is shown in Figure 1.

Figure 1 Appearance of the chassis



slot

The slots are located on the front of the chassis. UCP4131 provides 2 main control board slots, 9 interface board slots, and 2 power supply slots.

- Slots 1, 3 to 6, and 8 to 11 are service board slots, which are used to install various service boards and support mixed insertion.
- Slots 2 and 7 are main control board slots, used to install CCU boards and support dual main control boards.
- For the case without RAID module: If the main control board wants to expand the hard disk board, it can support up to 2 hard disk boards. If the main control board is in slot 2, you can insert the hard disk boards into slots 1 and 4; if the main control board is in slot 7, you can insert the hard disk boards into slots 6 and 9.
- For the case of accessing the RAID module: ① Insert the CCU main control board into slot 2. If you want to expand RAID, insert the RAID card into slot 4 and the RAID hard disk into slots 3 and 5; ② The CCU main control board is inserted into slot 7. If you want to expand RAID, insert the RAID card into slot 9 and the RAID hard disk into slots 8 and 10.

The slot distribution and internal logic diagram are shown in Figure 2 and Figure 3.

Figure 2 Chassis slot distribution

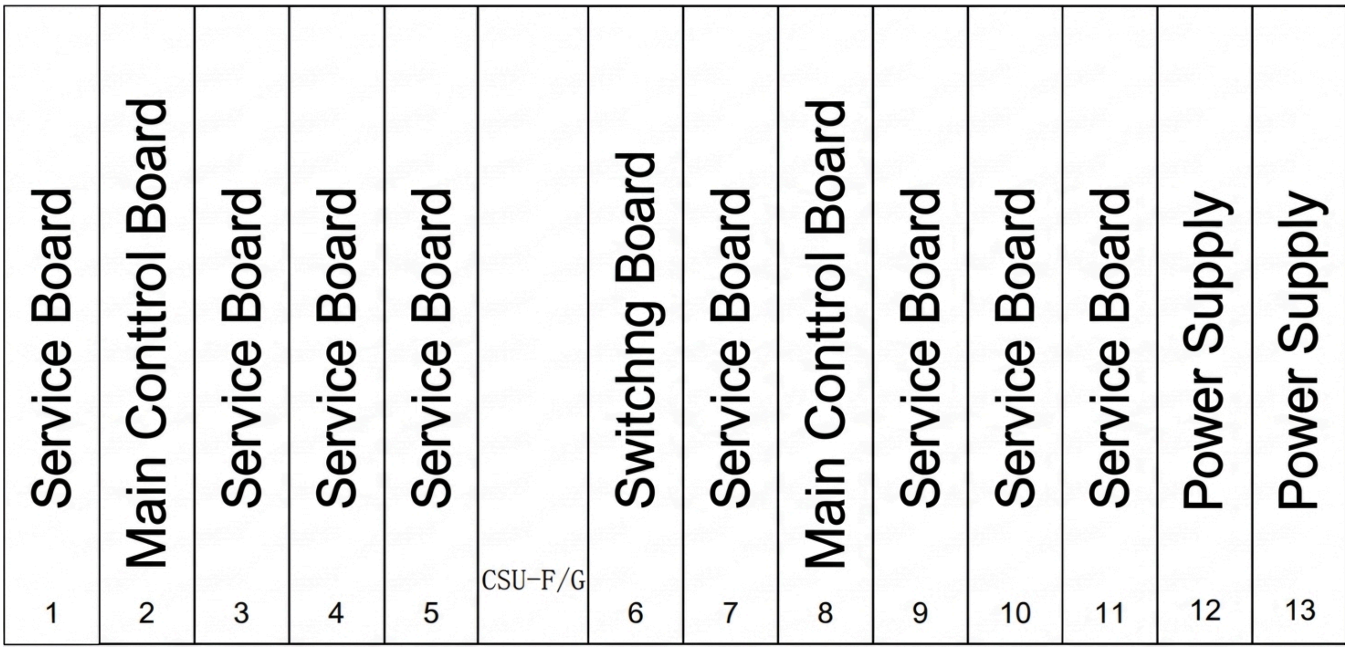
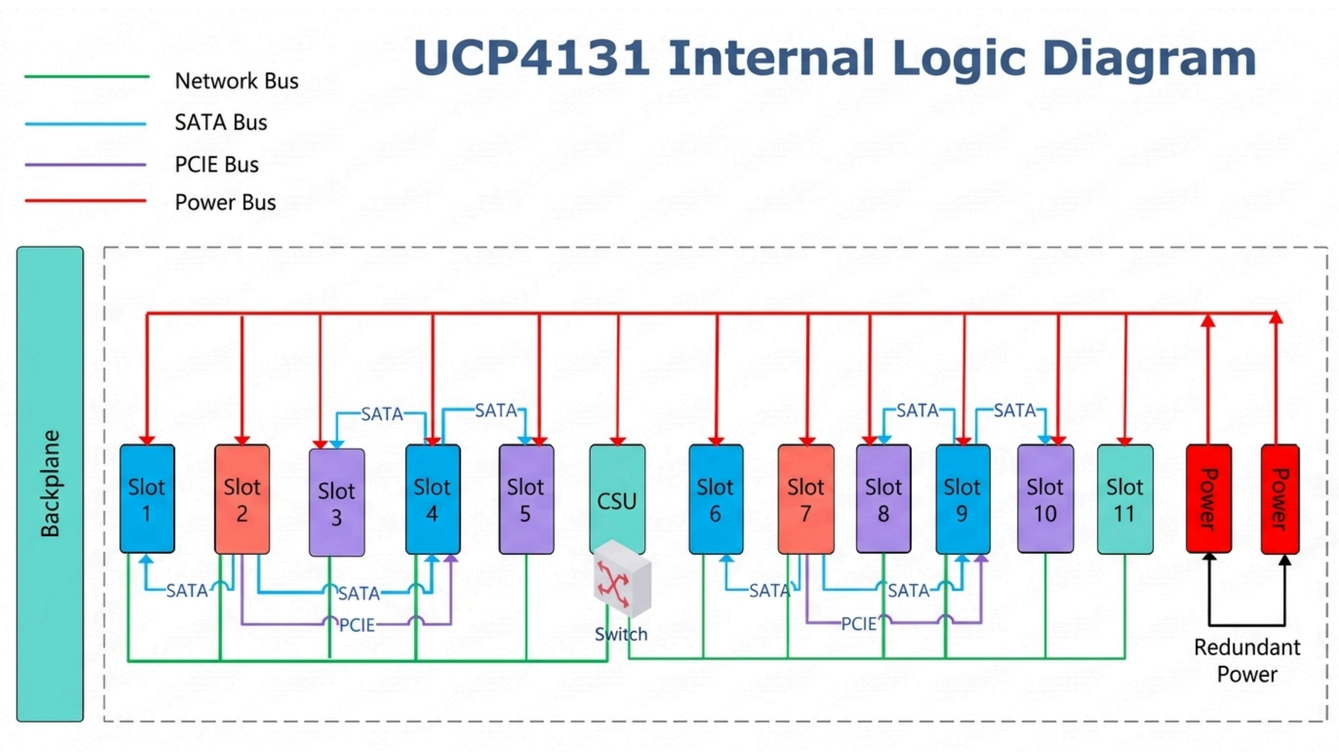


Figure 3 Internal logic diagram



- When only one main control board is installed in the two main control board slots, the system runs in the single main control mode; when the two main control board slots are fully configured, the system runs in the active and backup main control mode, which has higher reliability.
- The UCP4131 interface board can be selected according to the system capacity. Empty slots that are not configured with interface boards or power supplies need to be installed with empty panels.
- To ensure the normal use of basic functions, UCP4131 is equipped with at least one main control board.

3.3 Board introduction

The single boards of UCP1202/UCP1600/UCP2120/UCP4131 include switching unit (CSU), main control unit (CCU), wireless service unit (WTU), analog service unit (AIU), audio broadcast service unit (ACU), wireless trunking service unit (RIU) and digital service unit (DTU).

- Switch board CSU CSU is a switching board developed for VoIP applications. It provides switching interconnection for the main control board and service boards.
- Main control board CCU CCU is a main control board developed for VoIP applications. It uses Intel X86 high-performance processor to bring you clear, high-fidelity, and high-definition audio and video calls.
- Business board The service board includes wireless service board (WTU), analog service board (AIU), digital service board (DTU), audio broadcast service board (ACU), wireless cluster service board (RIU), RAID module board (RSU) and hard disk board (SEU). It supports access to GSM/WCDMA/LTE lines, analog lines, digital lines, audio lines, wireless intercom clusters and RAID.

3.3.1 Switch board CSU

CSU (Core Switch Unit) core switching board is divided into two types: CSU-F and CSU-G. CSU-F is a 100M switching board, and CSU-G is a 1000M switching board. In addition, CSU-F provides 2 ETH network ports, 1 CONSOLE port, and 1 RST key; CSU-G provides 3 ETH network ports, 1 SFP port, and 1 RST key. The CSU functions as a switch and connects the remaining main control boards and service boards in the chassis through the switch boards.

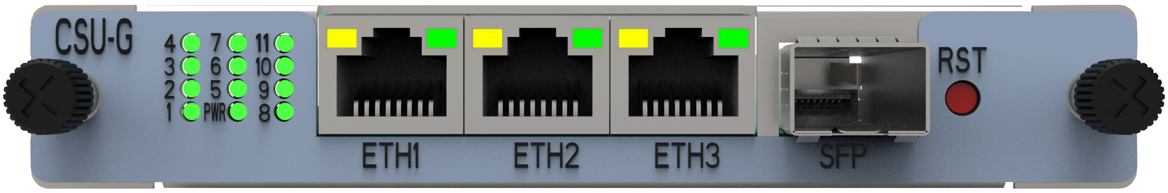
Panel appearance

The panel appearance of the CSU service board is shown in Figure 1.

Figure 1 CSU-F panel appearance

Note: The ETH1/ETH2 network port of CSU-F is a 100M network port

Figure 2 CSU-G panel appearance



Note: The ETH1/ETH2/ETH3 network ports of CSU-G are Gigabit network ports

3.3.2 Main control board CCU

CCU (Core Control Unit) is a main control board developed for VoIP applications. It uses Intel quad-core processor to bring you clear, high-fidelity, high-definition audio and video calls. In order to bring you clear, high-fidelity audio and video calls, CCU integrates a rich set of high-definition voice and video codecs and supports comprehensive protocol processing. The IP side supports SIP, IAX2 and other protocols, and the CPE side supports BRI, PRI, SS7, R2, GSM, WCDMA and other protocols. CCU can support Asterisk, Issabel, Elastix, FreePBX, VitalPBX, BrikerPBX and IPPBX/IVR and other open source software applications and private switch, firewall, IVR and voice gateway applications.

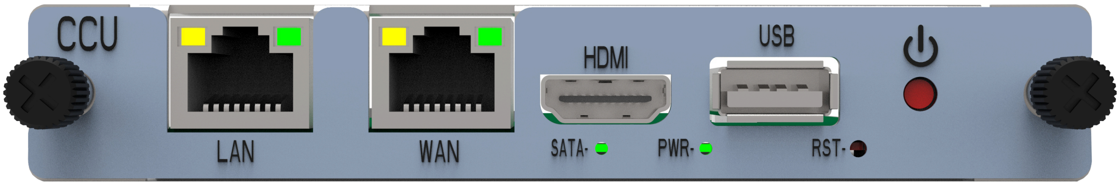
- CCU-N-ALDER CCU-I-TGL is a standard main control module developed for VoIP applications, using Intel quad-core processor
- CCU-I-TGL CCU-I-TGL is a high-performance main control module developed for VoIP applications, using Intel quad-core processor

CCU-N-ALDER

CCU-N-ALDER is a standard performance main control module developed for VoIP applications, using Intel quad-core processors. Suitable for small and medium-sized enterprise scenarios. Within the recommended number of users of 800, CCU-N-ALDER can handle up to 2,000 concurrent calls with G.711 codec or 620 concurrent calls with G.729 codec. In order to bring you clear, high-fidelity, high-definition audio and video calls, CCU-I-TGL integrates rich high-definition voice and video codecs.

CCU-I-TGL supports comprehensive protocol processing. The IP side supports SIP, IAX2 and other protocols, and the CPE side supports BRI, PRI, SS7, R2, GSM, WCDMA and other protocols. CCU-I-TGL can support Asterisk, Issabel, Elastix, FreePBX, VitalPBX, 3CX and IPPBX/IVR and other open source software applications and private switch, firewall, IVR and voice gateway applications. CCU-I-TGL will be pre-installed with OpenVox UC PBX system, Issabel, FreePBX and other software to take full advantage of the open source platform.

Panel Appearance



CCU-I-TGL

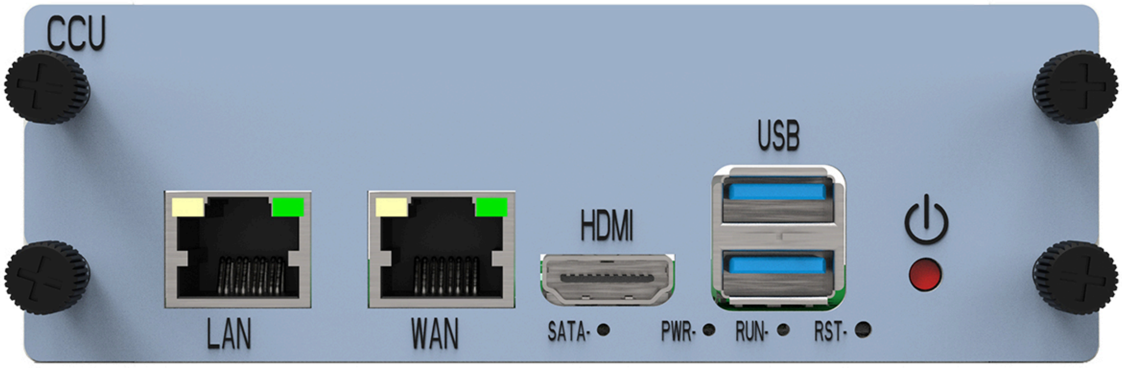
CCU-I-TGL is a main control module developed for VoIP applications. Using Intel quad-core processor, CCU-I-TGL can handle up to 3200 concurrent calls with G.711 codec or 1250 concurrent calls with G.729 codec (measured under laboratory conditions). In order to bring you clear, high-fidelity, high-definition audio and video calls, CCU-I-TGL integrates rich high-definition voice and video codecs.

CCU-I-TGL supports comprehensive protocol processing. The IP side supports SIP, IAX2 and other protocols, and the CPE side supports BRI, PRI, SS7, R2, GSM, WCDMA and other protocols. CCU-I-TGL can support Asterisk, Issabel, Elastix, FreePBX, VitalPBX, 3CX and IPPBX/IVR and other open source software applications and private switch, firewall, IVR and voice gateway applications. CCU-I-TGL will be pre-installed with OpenVox UC PBX system, Issabel, FreePBX and other software to take full advantage of the open source platform.

Panel appearance

The panel appearance of the CCU-I-TGL board is shown in Figure 1.

Figure 1 CCU-I-TGL panel appearance



Note: The network port of the main control board CCU is a Gigabit network port

Performance testing

1. test platform

CPU	Main frequency	memory	storage	operating system	Asterisk version
i5-1135 four cores and eight threads	2.40GHz	32G	MSATA 32G	Centos7.9 64-bit	16.23.0

Related instructions:

```
1、Equipment matching: i5-1135 motherboard connected to test machine through sip relay
2、Test data: Test different encodings, and the maximum number of concurrent SIP calls when the voice is normal
3、Call rate: 1 minute per round of calls, internal call delay 10s, maximum number of initiated and stopped calls per second set to 30
4、Asteirsk is compiled from source code, and the g723 and g729 encoding modules use core2-sse4 64-bit so files.
5、The performance screenshot is the top command output at the moment when WinSIP calls are full.
```

Call flow: consistent with the test call flow of **CCU-N-BAYL**.

2. Test method

   Prepare a test server B with Asterisk installed. WinSIP initiates a call to the A (i5-1135) motherboard and forwards it to Server B via SIP relay for playback. Before each test, make the CPU work at full speed.

   Set /sys/module/pcie_aspm/parameters/policy to performance, and use cpupower -c all frequency-set -g performance to put the CPU in full-speed working mode.

3. Test results

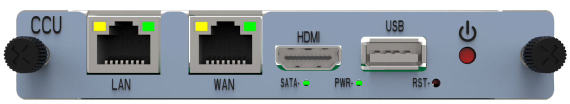
encoding	Number of concurrencies	sound quality	Performance (Cpu Idle)	Memory usage (Mb)
ulaw	3200	The voice is clear and there is no obvious lag	7.2%	7549
g723	1300	The voice is clear and there is no obvious lag	0.5%	3533
g729	1250	The voice is clear and there is no obvious lag	0.2%	3552

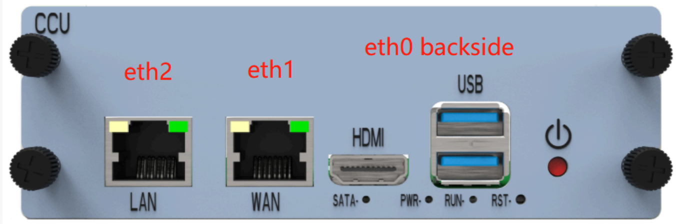
4. Things to note
   1. During the test, the critical number of concurrencies was determined by gradually increasing the number of concurrencies until the voice lags after full calls;
   2. In the test, the sound quality is judged by human ear monitoring. There are subjective differences. After the call is full, use the soft phone to call all the way to listen to the voice. The passing standard is clear voice without lag;
   3. During the test, the power limiting option in the BIOS was removed, and the CPU was working at full speed during the test.

CCU Notes

It should be noted that the eth numbers of CCU-N-ALDER and CCU-I-TGL are different on the management page, as shown in Table 1.

Table 1 Comparison of eth numbers

Motherboard model	LAN	WAN	Switch board	panel plot
CCU-N-ALDER	eth2	eth1		

Motherboard model	LAN	WAN	Switch board	panel plot
CCU-I-TGL	eth2	eth1	eth0	

3.3.3 Business board

The service board includes wireless service board (WTU), analog service board (AIU), digital service board (DTU), audio broadcast service board (ACU) and wireless trunking service board (RIU)), which supports access to wireless lines, analog lines, digital lines, audio lines and trunking lines.

- WTU The WTU (Wireless Trunk Unit) series of wireless business modules allow the UCP unified communications platform to support GSM/WCDMA/LTE to connect VoIP devices. Each module provides 4 2G/3G/4G channels.
- AIU-8 The AIU-8 (Analog Interface Unit 8 Port) series of analog business modules, including AIU-8O, AIU-8S and AIU-8O/S, are used with the UCP unified homogeneous platform to connect VoIP and PSTN. Each module provides 8 FXO/8 FXS/4 FXO and 4 FXS channels.
- AIU-16 The AIU-16 (Analog Interface Unit 16 Port) series of analog business modules, AIU-16S, is used with the UCP unified homogeneous platform to connect VoIP and PSTN. Each module provides 16 FXS channels.
- DTU The DTU (Digital Trunk Unit) business board is a VoIP trunk voice module specially designed for operators and call centers. It can support 1/2/4 E1/T1 interfaces.
- DTU-301 DTU-301 (Digital Trunk Unit) service board is a cost-effective single-port trunk voice module designed for operators and call centers. It supports 1 E1/T1 interface.
- ACU The ACU (Audio Communication Unit) series audio broadcast business board is a fully functional voice access device that facilitates the connection of microphones, telephones and conference audio mixer systems, and realizes the reception and amplification of telephones and microphones for transmission through telephone lines.
- RIU The RIU (Radio Interface Unit) series wireless trunking service board is a fully functional voice access device that facilitates the integration of the intercom trunking system and the telephone system.
- MIU MIU-4/8 magnet relay gateway is a powerful voice access device, including MIU-4 and MIU-8, which provides 4-way or 8-way magnet phone access capabilities to the IP network.
- RSU RSU (RAID Storage Unit) is a RAID module board that uses virtualized storage technology to combine multiple hard drives into one or more hard drive array groups to improve performance or data redundancy, or both at the same time.
- SEU SEU (Storage Expand Unit) is a hard drive board, you can choose 2.5-inch mechanical hard drive or 2.5-inch solid-state drive.

WTU

The WTU (Wireless Trunk Unit) series of wireless business modules allow the UCP unified communications platform to support GSM/WCDMA/LTE to connect VoIP devices. Each module provides 4 4G channels. They can bring you excellent voice services, support voice codecs including G.711U, G.711A, GSM, G.722, G.726, G.729, provide flexible SMS services and HTTP-based API interfaces. WTU series wireless business modules are fully compatible with VoIP system platforms such as Asterisk, 3CX, FreePBX, FreeSWITCH and VOS, providing users with more diverse telecommunications access methods and reducing communication costs.

Main functions

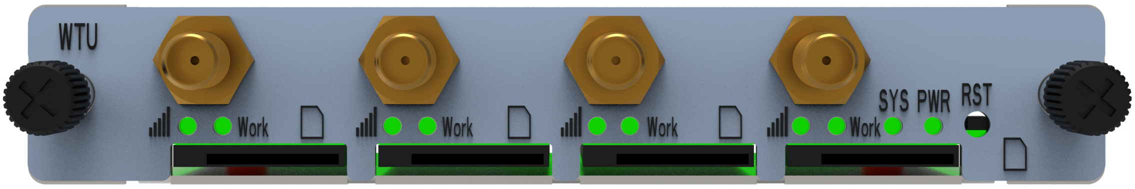
- SIP/IAX2/Wireless Group Settings
- Call duration limit, call number limit, text message limit
- call waiting
- call transfer
- echo cancellation
- gain control
- Caller/callee number filtering and modification
- Support two-step dialing
- Flexible routing
- Codecs:G.711-ulaw,G.711-alaw,gsm,g722,g726,g729
- DTMF
- CDR
- call statistics
- Number blacklist
- Support PIN code verification
- USSD and USSD API
- SMS API
- SMS
- Bulk SMS
- SMS to email
- SMS forwarding
- Support SMS remote control
- Automatic restart
- Supports custom scripts and dial plans
- Support OpenVox cloud platform
- Number inquiry, balance inquiry
- Backup configuration files
- Online updates
- Syslog remote log

- SIP/RTP packet capture

Panel appearance

The panel appearance of the WTU service board is shown in Figure 1.

Figure 1 Appearance of WTU panel



AIU-8

The AIU-8 (Analog Interface Unit 8 Port) series of analog business modules, including AIU-8O, AIU-8S and AIU-8O/S, are used with the UCP unified homogeneous platform to connect VoIP and PSTN. Each module provides 8 FXO/8 FXS/4 FXO and 4 FXS channels. The AIU-8 series analog business modules can bring you high-quality high-definition voice services, including G.711U, G.711A, GSM, G.722, G.726, G.726, G.729 and other codecs. The AIU-8 series analog business modules will be 100% compatible with Asterisk, 3CX, FreePBX, FreeSWITCH and VOS VoIP operating platforms to help users reduce telecommunications and communication costs.

Product features

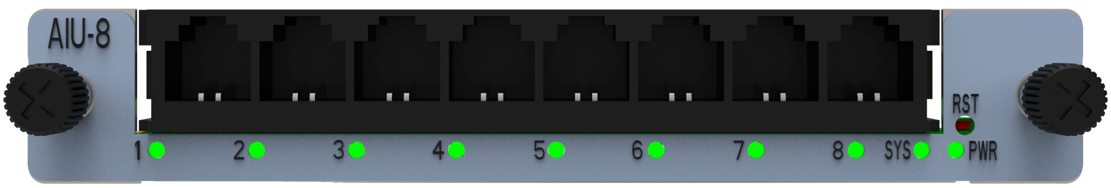
- Simple and convenient configuration via Web GUI
- Rich logging functions
- Open API interface
- Supports T.38 fax relay, T.30 fax transparent transmission, and continuous faxing of multiple pages
- Echo cancellation and static jitter buffering
- Supports DTMF mode settings
- Support ringing rhythm and frequency settings
- Support volume control
- Support MWI (Message Waiting Indicator)
- Support DHCP, DNS/DDNS, NAT networking
- Support multiple SIP protocols
- Diverse scalable applications, such as IVR, DISA, etc.

- Support NTP and client time synchronization
- Supports Voice Dynamic Detection (VAD) and Comfort Noise Generation (CNG)
- Supports modifying username and password for web login
- Support SSH remote operation
- Support SRTP
- Support SIP server mode
- Support firmware online upgrade
- Support configuration file backup and upload
- Supports custom scripts and dial plans
- Supports restoring factory settings
- Support volume adjustment, gain adjustment, call hold, call waiting, call forward, caller ID

Panel appearance

The panel appearance of the AIU-8 service board is shown in Figure 1.

Figure 1 AIU-8 panel appearance



AIU-16

The AIU-16 (Analog Interface Unit 16 Port) series of analog business modules, AIU-16S, is used with the UCP unified homogeneous platform to connect VoIP and PSTN. Each module provides 16 FXS channels. The AIU-16 series analog business modules can bring you high-quality high-definition voice services, including G.711U, G.711A, GSM, G.722, G.726, G.726, G.729 and other codecs. The AIU-16 series analog business modules will be 100% compatible with Asterisk, 3CX, FreePBX, FreeSWITCH and VOS VoIP operating platforms to help users reduce telecommunications and communication costs.

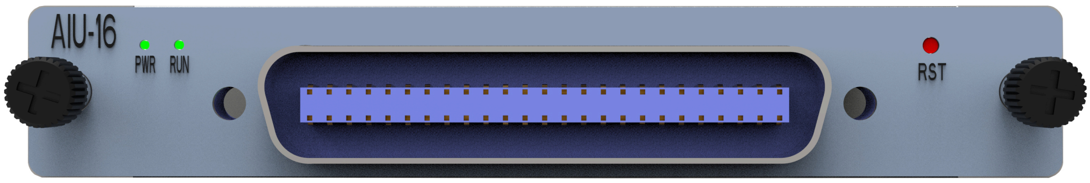
Product features

- Simple and convenient configuration via Web GUI
- Rich logging functions
- Open API interface
- Supports T.38 fax relay, T.30 fax transparent transmission, and continuous faxing of multiple pages
- Echo cancellation and static jitter buffering
- Supports DTMF mode settings
- Support ringing rhythm and frequency settings
- Support volume control
- Support MWI (Message Waiting Indicator)
- Support DHCP, DNS/DDNS, NAT networking
- Support multiple SIP protocols
- Diverse scalable applications, such as IVR, DISA, etc.
- Support NTP and client time synchronization
- Supports Voice Dynamic Detection (VAD) and Comfort Noise Generation (CNG)
- Supports modifying username and password for web login
- Support SSH remote operation
- Support SRTP
- Support SIP server mode
- Support firmware online upgrade
- Support configuration file backup and upload
- Supports custom scripts and dial plans
- Supports restoring factory settings
- Support volume adjustment, gain adjustment, call hold, call waiting, call forward, caller ID

Panel appearance

The panel appearance of the AIU-16 service board is shown in Figure 1.

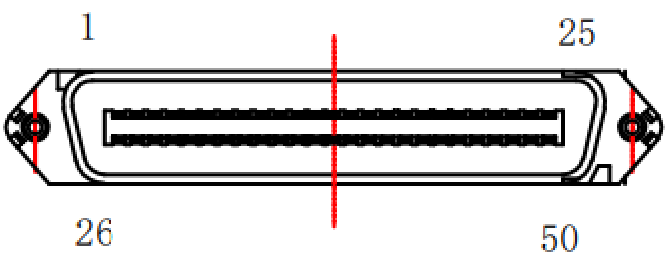
Figure 1 AIU-16 panel appearance



Interface description

The interface description of the AIU-16 service board is shown in Figure 2.

Figure 2 AIU-16 interface



The interface description of the AIU-16 service board is shown in the following table.

Tip	Ring	Phone number
26	1	Port 1
27	2	Port 2
28	3	Port 3
29	4	Port 4
30	5	Port 5
31	6	Port 6
32	7	Port 7
33	8	Port 8
34	9	Port 9

Tip	Ring	Phone number
35	10	Port 10
36	11	Port 11
37	12	Port 12
38	13	Port 13
39	14	Port 14
40	15	Port 15
41	16	Port 16
42	17	-
43	18	-
44	19	-
45	20	-
46	21	-
47	22	-
48	23	-
49	24	-
50	25	-

DTU

DTU digital gateway module is a VoIP trunk voice gateway module specially designed for operators and call centers. This is a converged media gateway module that can be used with OpenVox UCP series products to connect traditional phone systems with IP networks and achieve seamless connection between VoIP PBX and PRI/SS7/R2 networks. The user-friendly interface and simple operation methods allow users to easily set up personalized gateways. Users can realize secondary development of gateway functions through API. Among them, there are three models of DTU digital gateway modules, namely 301, 302, and 304

- The DTU-301 digital trunk voice gateway module provides 1 T1/E1 interface and can support up to 30 concurrent calls.
- The DTU-302 digital trunk voice gateway module provides 2 T1/E1 interfaces and can support up to 60 concurrent calls.
- The DTU-304 digital trunk voice gateway module provides 4 T1/E1 interfaces and can support up to 120 concurrent calls.

Main functions

- Provide 1/2/4 optional T1/E1/PRI interfaces;
- Supports up to 120 concurrent calls;
- Supports most VoIP system platforms, such as Asterisk, Elastix, 3CX, FreeSWITCH, Broadsoft, etc.;

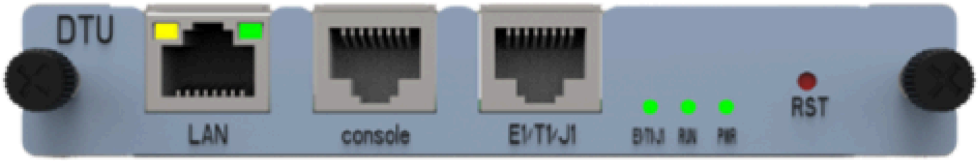
target application

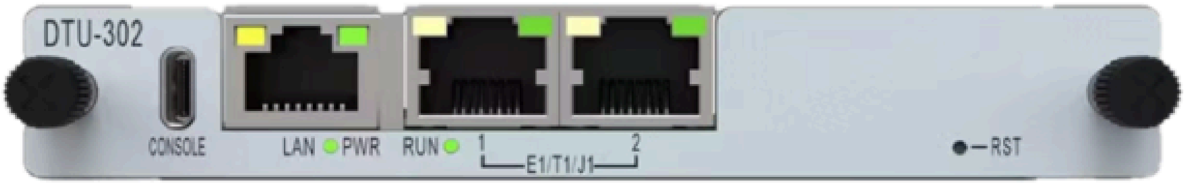
- Upgrade traditional PBX to VoIP service
- Traditional PBX connection to remote site for private Internet phone contact
- TLS/SRTP/HTTPS security technology ensures call and account security
- Configurable E1/T1 port, supports PRI, SS7, MFC/R2
- IP PBX connects to traditional TDM services
- Phased transition from traditional PBX to IP PBX
- Virtualized system connected to TDM service
- Support multiple voice coding: G722, G729, G723, GSM, G711
- Adopting advanced echo cancellation technology to ensure high-definition sound quality
- Support T.30, T.38 fax
- Support multi-language voice prompts
- SNMP v1/v2/v3

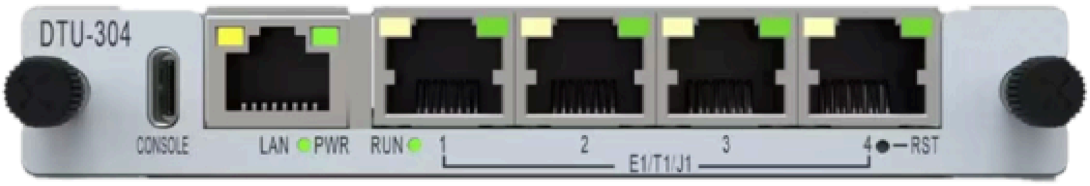
Panel appearance

The panel appearance of the DTU service board is shown in Figure 1.

Figure 1 Appearance of DTU panel







ACU

The ACU (Audio Communication Unit) audio broadcast business board is a powerful voice access device that facilitates the docking of microphones, telephones and conference audio mixer systems. It realizes the reception and amplification of telephone calls and microphones and transmits them through telephone lines, meeting the needs of video and telephone conferencing applications, ensuring clear voice quality and powerful telephone access control capabilities. The device provides 2 channels of audio input and output interfaces. Users can achieve different audio playback control effects by connecting different external devices.

Main functions

- Strong compatibility
- Echo suppression, noise suppression
- Unparalleled Noise Cancellation Technology
- Patented voice algorithm to ensure clear voice quality
- Multi-channel telephone (SIP, PSTN, GSM) access processing capabilities
- Can output two audio channels at the same time
- Moderate cost performance, simple integrated application
- 1U standardized design saves deployment space
- Simple and convenient configuration via Web GUI

Strong compatibility

The equipment supports the use of different telephone lines and can be connected to PBX user lines, telecommunications office user lines, simulated user gateway IAD and other equipment. In addition, it can also support SIP applications of IP-PBX, softswitch, and SIP servers of major companies, and can be seamlessly connected to a wide range of dispatching and command systems, conference videos, and central control systems.

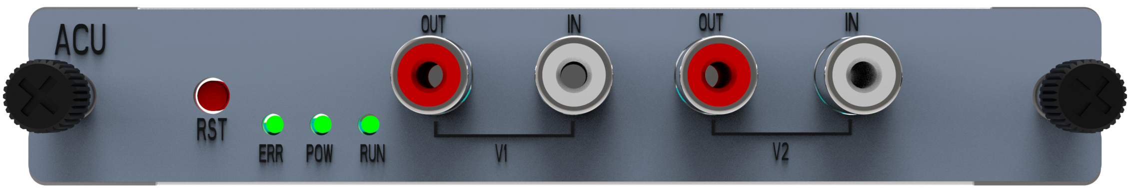
Application scenarios

- Large conference room
- court meeting room
- Dispatch command center
- Army, police
- Government department conference room
- Field command conference room
- Building broadcast
- training classroom
- Command vehicle conference room

Panel appearance

The panel appearance of the ACU service board is shown in Figure 1.

Figure 1 ACU panel appearance



RIU

The RIU (Radio Interface Unit) wireless trunking service board is a powerful voice access device that facilitates the integration of the intercom trunking system and the telephone system. Users can conveniently call the intercom through the phone, or use the intercom to make calls. The system supports traditional PSTN phone lines and SIP-based VoIP phone lines. It is very convenient to deploy and use, and can be plug-and-play.

The wireless trunking business module adopts carrier-grade product design and has strong networking capabilities and sound processing capabilities. It adopts microcomputer chip technology and electronic switch technology. Each channel of control is independent of each other. The operation of cutting in and out of audio signals is sensitive, and it can realize simultaneous access of 2-way intercom. Wireless cluster gateway can be widely used in special command and dispatch systems in various industries.

The device provides 4-way intercom interfaces, using RJ45 plugs, equipped with professional intercom control cables, and is compatible with mainstream intercom handsets and car radios such as Motorola and Kenwood. In addition, the wireless trunking module combines experience in the multi-party voice field and develops a voice algorithm that makes its call effect higher than similar products currently on the market. The wireless cluster intercom gateway is flexible and simple to deploy and can be used with a simple connection. Supports PSTN and SIP, compatible with multiple models of mobile phones and radio stations.

Main functions

- Simulate trunking single call and group call
- Patented voice algorithm to ensure clear voice quality
- Unparalleled Noise Cancellation Technology
- Various dialing and number collection rule configurations
- Multi-channel telephone (SIP, PSTN) access processing capabilities
- Standard SIP protocol, can be connected to telephone system and dispatch system
- Adaptive VOX (voice activation), adjustable sensitivity
- Input and output volume adjustable
- Adjustable line level control
- Voice delay cache setting to avoid lost words and characters
- Noise cancellation, tone detection and trigger settings
- COR and PTT valid signals can be set by the user
- Support WEB-based management method
- Strong compatibility supports multiple brands of walkie-talkies such as Motorola and Kenwood
- Support SIP INFO power grabbing function
- Support DTMF dialing
- Support PPT hotline call

Strong compatibility

The equipment supports the use of different telephone lines and can be connected to PBX user lines, telecommunications office user lines, simulated user gateway IAD and other equipment. It can also support SIP applications of IP-PBX, softswitch and SIP servers of major companies. And can be seamlessly connected to the vast dispatching and command system.

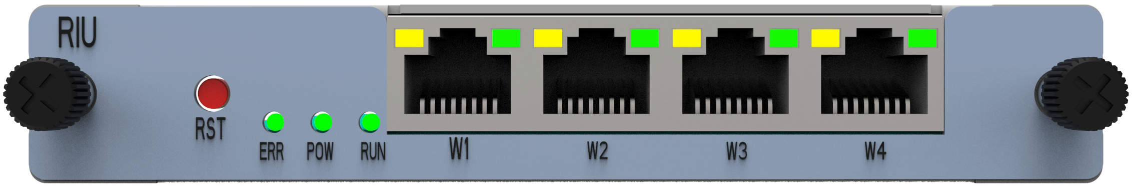
Cluster intercom gateway application scenarios

RIU wireless cluster business board can be widely used in command and dispatch systems of public security, armed police, firefighting, military, railways, civil air defense, industrial and mining enterprises, forestry, petroleum, electric power, and government. Achieve rapid response to emergency response and integrated communication of multiple communication methods.

Panel appearance

The panel appearance of the RIU service board is shown in Figure 1.

Figure 1 Appearance of RIU panel



MIU

MIU-4/8 (Magnet Interface Unit) magnet relay gateway is a powerful voice access device, including MIU-4 and MIU-8, which provides the ability to access 4-way or 8-way magnet phones to the IP network. Using the standard SIP protocol, it can be connected to the standard SIP soft switching system to provide VoIP/MoIP solutions for magnet phone users. The device is easy to deploy quickly, has a high degree of security and strong anti-interference capabilities, and is widely used in military or other special command and dispatch systems and other fields.

The MIU-4/8 (Magnet Interface Unit) panel supports 1/2 magnet channel interface, Console interface, running status indicator light, power status indicator light, channel status indicator light and RST restart button. In terms of software docking, MIU uses standard SIP protocol, is compatible with mainstream IPPBX and SIP servers, and supports most VoIP operating system platforms, such as Asterisk, Issabel, 3CX, FreeSwitch, BroadSoft, VOS, etc.

Main functions

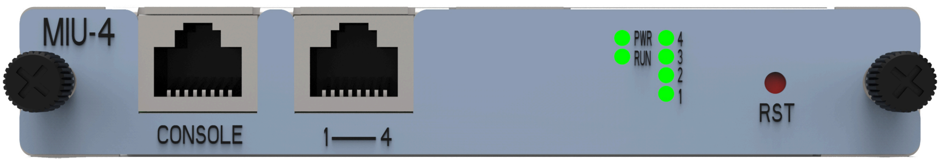
- Support 4 or 8-way magnet phone interface
- Support WEB network management configuration function
- Supports multiple network protocols and can integrate multiple interface types with the unified communications platform
- Can be connected to enterprise IP phone systems and various unified communications systems to improve communication efficiency
- Supports automatic off-hook when calling the opposite end and automatic hanging up when the opposite end hangs up;
- Support DHCP, DNS/DDNS, NAT networking
- Supports Voice Dynamic Detection (VAD) and Comfort Noise Generation (CNG)

- Can adapt to different network environments and ensure stable transmission
- Adopt codec adaptive technology to ensure call quality
- Open API interface (AMI)
- Support volume adjustment, gain adjustment, call hold, call waiting, call forward, caller ID
- Support SSH remote operation

Panel appearance

The panel appearance of the MIU service board is shown in Figure 1.

Figure 1 MIU panel appearance



RSU

RSU (RAID Storage Unit) is a RAID module board that uses virtualized storage technology to combine multiple hard drives into one or more hard drive array groups to improve performance or data redundancy, or both at the same time.

The RAID module unit supports RAID1. Only the 4u chassis supports the RAID function and requires two SEUs (hard disk boards). On the basis of improving the reading and writing speed, RAID 1 also has the function of data backup; its principle is to store data on the main hard disk and write the same data on the mirror hard disk at the same time. When the main hard disk (physical) is damaged, the mirror hard disk takes over the work of the main hard disk. Therefore, as long as one disk is normal, it can maintain operation and the reliability is high.

Panel appearance

The panel appearance of the RSU service board is shown in Figure 1.

Figure 1 RSU panel appearance

SEU

SEU (Storage Expand Unit) is a hard drive board, you can choose 2.5-inch mechanical hard drive or 2.5-inch solid-state drive. The front panel is equipped with a PWR status indicator light, making the device operating status clear at a glance. The 1U chassis supports expansion of **1** hard drive; the 2U chassis supports expansion of **4** hard drives; the 4U chassis supports expansion of **6** hard drives.

Panel appearance

The panel appearance of the SEU service board is shown in Figure 1.

Figure 1 SEU panel appearance

3.3.4 Power module

UCP2120/UCP4131 can support dual power supply. For UCP2120/UCP4131, when configured with dual power modules, current sharing and backup are supported.

UCP2120 power supply

UCP2120 can be configured with single power supply or dual power supply modules. The dual power supply module can choose between DC and AC power supplies, dual DC power supplies or dual AC power supplies.

When configured with dual power modules, it has the following functions:

- Supports current sharing and backup
- During normal operation, multiple power modules can each output current to share the load;
- When a certain power module stops working, other power modules serve as backup and assume its normal power supply tasks.

UCP4131 power supply

UCP4131 can be configured with single power supply or dual power supply modules. The dual power supply module currently only supports dual AC power supplies.

When configured with dual power modules, it has the following functions:

- Supports current sharing and backup
- During normal operation, multiple power modules can each output current to share the load;
- When a certain power module stops working, other power modules serve as backup and assume its normal power supply tasks.
- Support hot swap

Without turning off the power of the entire machine, you can directly add power modules to empty slots in the power distribution frame; in the case of power redundancy backup, you can directly pull out a power module without affecting the normal operation of the equipment.

Power specifications

- AC power

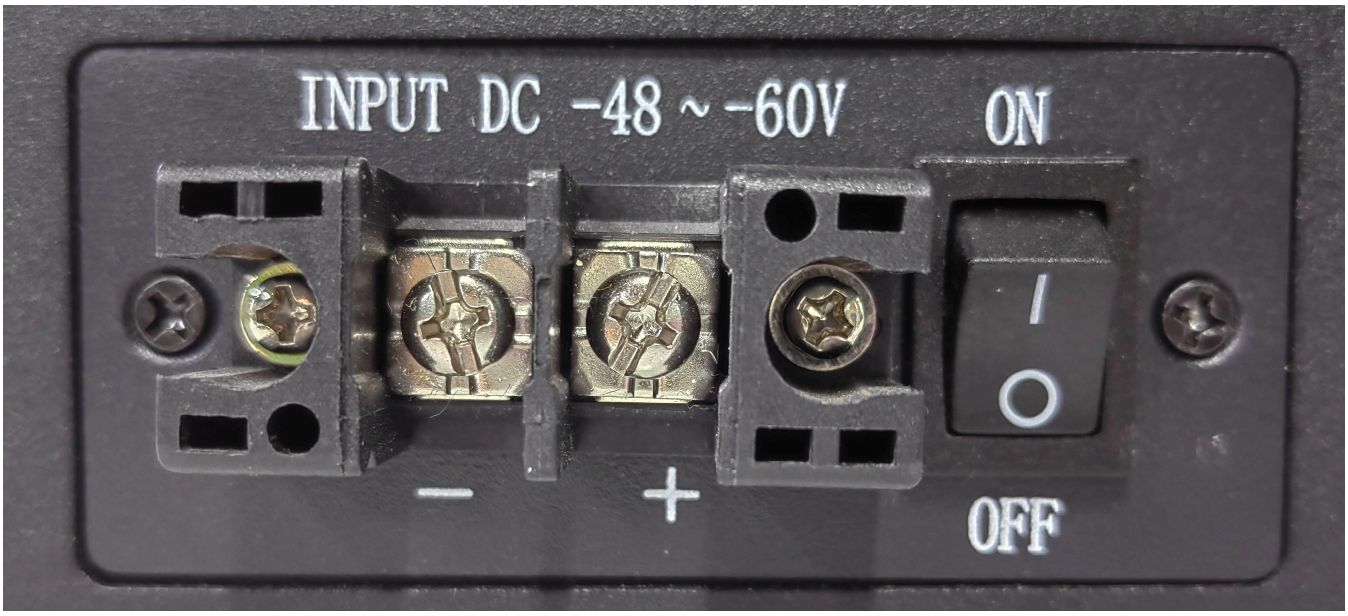
To use AC power, you need to use an AC power cord, as shown in Figure 1.

The AC power parameters are shown in the table below.

AC power parameters	
Rated power	200W
input voltage	100-240V/50-60Hz

- DC power supply

To use a DC power supply, a DC power cord is required. The panel interface is shown in Figure 2. (The product does not come with a DC power cord, you need to prepare it yourself)



The DC power supply parameters are shown in the table below.

DC power parameters	
Rated power	200W
input voltage	-48-60V

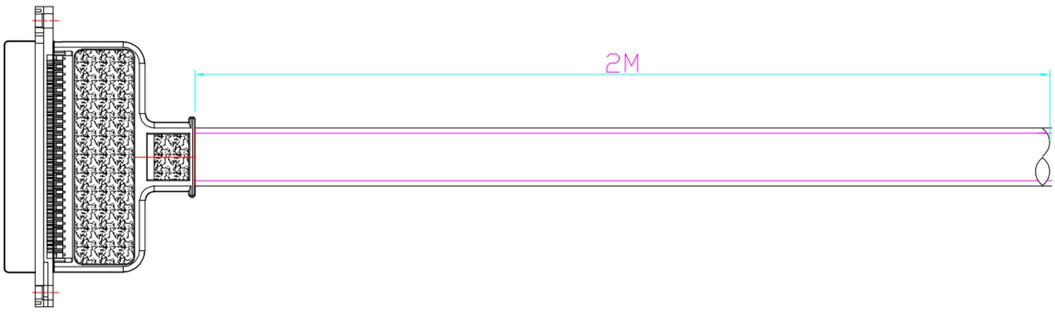
3.3.5 Cables

The cables of UCP1600/UCP2120/UCP4131 include analog trunk cables, digital trunk cables, and AC power cords.

Analog trunk cable

AIU-16 analog trunk cable is a cable that connects UCP1600/UCP2120/UCP4131 and various inter-office trunk equipment, as shown in Table 1 and Table 2.

Table 1 Analog trunk cable

Name	pictures
RJ21 cable (AIU-16)	Figure 1 RJ21 

The cable color description of the AIU-16 analog service board is as shown in the following table.

Table 2 Analog trunk cable

color	Tip	Ring	color	Phone number
blue	26	1	white	Port 1
Orange	27	2	white	Port 2
green	28	3	white	Port 3
Brown	29	4	white	Port 4
gray	30	5	white	Port 5
blue	31	6	red	Port 6
Orange	32	7	red	Port 7
green	33	8	red	Port 8
Brown	34	9	red	Port 9
gray	35	10	red	Port 10
blue	36	11	black	Port 11
Orange	37	12	black	Port 12

color	Tip	Ring	color	Phone number
green	38	13	black	Port 13
Brown	39	14	black	Port 14
gray	40	15	black	Port 15
blue	41	16	Yellow	Port 16
-	42	17	-	-
-	43	18	-	-
-	44	19	-	-
-	45	20	-	-
-	46	21	-	-
-	47	22	-	-
-	48	23	-	-
-	49	24	-	-
-	50	25	-	-

Digital trunk cable

Digital trunk cables are cables that connect UCP1600/UCP2120/UCP4131 to various inter-office relay equipment, as shown in Table 3.

Table 3 Digital trunk cable

Name	pictures
BNC adapter	Figure 2 1E1BNC adapter 

power cord

To use AC power, you need to use an AC power cord, as shown in Figure 3.

Figure 3 AC power cord

4. Technical indicators

Introduce the key technical indicators of the unified communications platform system.

- Physical parameters

  Mainly introduces the performance and capacity of UCP1202/UCP1600/UCP2120/UCP4131.
- Interfaces and protocols Mainly introduces the interfaces of UCP1202/UCP1600/UCP2120/UCP4131.
- standards to follow Mainly introduces the standards followed by UCP1202/UCP1600/UCP2120/UCP4131.

4.1 Physical parameters

The physical parameters of the unified communications platform are shown in the following table.

Table 1 Physical parameters

Parameter	UCP1202	UCP1600	UCP2120	UCP4131
Dimensions	50 mm (H) × 188 mm (W) × 202 mm (D)	44 mm (H) × 434 mm (W) × 330 mm (D)	88 mm (H) × 434 mm (W) × 330 mm (D)	176.8 mm (H) × 435.8 mm (W) × 330 mm (D)
Weight	1.5 kg	3.9 kg	5.6 kg	9.6 kg
Maximum Fully Loaded Power Consumption	< 36 W	< 64 W	< 170 W	< 340 W
Input Voltage (AC Power Supply)	\\	100 V ~ 240 V AC	100 V ~ 240 V AC	100 V ~ 240 V AC
Input Current (AC Power Supply)	Input current (DC power supply) 3 A	1 A	2 A	4 A
Power Frequency (AC Power Supply)	\\	50 Hz / 60 Hz	50 Hz / 60 Hz	50 Hz / 60 Hz
Maximum Output Power (AC Power Supply)	36 W	75 W	200 W	400 W
Input Voltage (DC Power Supply)	12 V	Not supported	-48 V ~ -60 V	-48 V ~ -60 V
User Loop Distance	3.0 km (under the condition that the telephone line has no bridge tap)			
Storage Temperature	-40°C ~ 70°C	-20°C ~ 70°C	-20°C ~ 70°C	-20°C ~ 70°C
Long-term Operating Temperature	0°C ~ 50°C			
Short-term Operating Temperature	Note: Short-term operation means continuous working time not exceeding 12 hours, and the annual cumulative time not exceeding 8 days.			
Ambient Humidity	5% ~ 95% RH, non-condensing			
Airborne Particle Concentration	Less than 180 mg/m³			

4.2 Performance and capacity

Unified communications platform performance and capacity are shown in the table below.

Table 1 Performance and capacity

parameters	**UCP1202**	**UCP1600**	**UCP2120**	**UCP4131**
Number of recommended users	800	800	4000	4000
Recommended number of concurrencies	200	200	800	800
Maximum number of built-in simulated users	16	64	160	160
Maximum number of FXO interfaces	8	32	80	80
Maximum number of E1/T1 interfaces	4	16	40	40
Maximum number of wireless interfaces	4	16	40	40
Maximum number of wireless trunking interfaces	4	16	40	40

4.3 Interfaces and protocols

The number and purpose of each interface of the unified communications platform are shown in the following table.

Table 1 Interfaces and protocols

Interface type	Business board that provides interfaces	Number of service board interfaces	Interface purpose			
UCP1600	UCP2120	UCP4131				
network interface	CCU-N-GML	2 (network card speed Gigabit)	It is used to connect the device to the LAN and is the external IP service interface of the device.	Yes	Yes	Yes
CCU-N-BAYL	2 (network card speed Gigabit)	It is used to connect the device to the LAN and is the external IP service interface of the device.	Yes	Yes	Yes	
CCU-I-KABYLR	2 (network card speed Gigabit)	It is used to connect the device to the LAN and is the external IP service interface of the device.	No	Yes	Yes	
DTU	1 (network card speed 100M)	It is used to connect the device to the LAN and is the external IP service interface of the device.	Yes	Yes	Yes	
USB interface	CCU-N-GML	1 (USB speed 3.0)	Business extension interface	Yes	Yes	Yes
CCU-N-BAYL	1 (USB speed 2.0)	Business extension interface	Yes	Yes	Yes	
CCU-I-KABYLR	2 (USB speed 3.0)	Business extension interface	No	Yes	Yes	
DTU	1 (USB speed 2.0)	Business extension interface	Yes	Yes	Yes	

Interface type	Business board that provides interfaces	Number of service board interfaces	Interface purpose			
E1/T1 interface	DTU	Maximum 4 channels	Provide E1/T1 access	Yes	Yes	Yes
FXS interface	AIU-16	Maximum 16 channels	Provides access to 16 analog phones.	Yes	Yes	Yes
FXO interface	AIU-8	Maximum 8 channels	Provides access to 8 analog trunks.	Yes	Yes	Yes
HDMI interface	CCU-N-GML	1	HDMI display access	Yes	Yes	Yes
CCU-N-BAYL	1	HDMI display access	Yes	Yes	Yes	
CCU-I-KABYLR	1	HDMI display access	No	Yes	Yes	
MiniHDMI	DTU	1	MiniHDMI display access	Yes	Yes	Yes

4.4 Standards followed

The standards followed by the unified communications platform are shown in Table 1.

Table 1 Standards followed

Protocol/Technology	Compatible with standards
PRA	YDN 034-1997 ISDN User - Network Interface Technical Specification, ITU-T G.962, ITU-T I.431, ITU-T Q.921, ITU-T Q.931
QSIG	ITU-T G.962，ITU-T I.431，ITU-T Q.921，ECMA-142，ECMA-143，ECMA-148，ECMA-157，ECMA-163，ECMA-164，ECMA-165，ECMA-173，ECMA-174，ECMA-185，ECMA-186，ECMA-241，ECMA-242
SIP	RFC 3261-3263，RFC 3265，RFC 2976，RFC 3311，RFC 3420，RFC 3515，RFC 3842
SDP	RFC 2327-1998，RFC 3264
T.30	ITU-T T.30
T.38	ITU-T T.38
echo cancellation	Oslec

Protocol/Technology	Compatible with standards
DTMF	•DTMF (Dual Tone Multi-Frequency) detection and reporting meets standards ITU-T Q.23 and ITU-T Q.24. The reported information includes: DTMF corresponding number, signal energy, and signal duration. •Supports transmission of DTMF via RTP. •Supports RFC 2833.
Voice Qos	•Complies with IEEE 802.1P/Q. •Supports TOS, priority 0-7, service type 0-4. •Supports DSCP, priority 0 - 63.
RTP Media Stream Encryption (SRTP)	Supports AES (128-bit) encryption algorithm.
TCP/IP protocol	•Supports IPv4. •UCP1600/UCP2120/UCP4130 currently only supports intra-office phone calls in IPv6 (excluding IPSec) scenarios, and does not support the coexistence of IPv4 and IPv6.

5. Installation and configuration

5.1 Installation and configuration

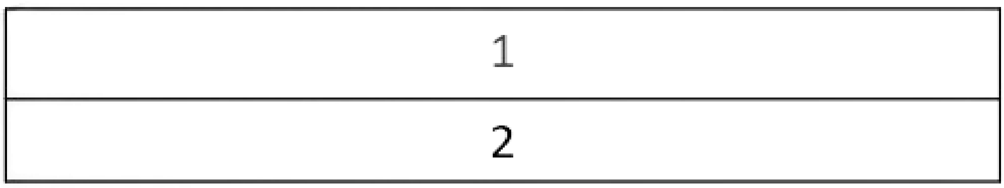
The unified communications platform is suitable for the following application scenarios.

- Quick installation guide This article mainly introduces how to install the service board into UCP1202/UCP1600/UCP2120/UCP4131.
- Business Board Interconnection Guide It mainly introduces how the single board interfaces with 3CX, Asterisk, FreePBX, FreeSwitch and Elastix.
- User Manual It mainly collects the user manuals of single boards.

5.2 Quick Installation Guide

Slot number corresponding IP relationship

UCP1202



- The slots are located on the front of the chassis. UCP1202 provides 1 main control board slot and 1 service board slot.
- The general service board (except ACU/RIU) automatically identifies the initial IP based on the slot number. The corresponding IP for slot 2 is `172.16.80.2`.
- **Slot 1 (Main Control)**: Only one **CCU-N-ALDER** main control board must be installed.
- **Empty Slot Treatment**: Unused slots need to be installed with empty panels to ensure dust prevention and heat dissipation.

UCP1600

2	3	5
1	CSU-F/G	4

- The slots are located on the front of the chassis. UCP1600 provides 1 main control board slot and 4 service board slots.
- Slots 2 to 5 are service board slots, which are used to install various service boards and support mixed insertion; the corresponding IP is 172.16.80.1---172.16.80.5
- Slot 1 is the main control board slot, and only one **CCU-N-ALDER** main control board must be installed.
- If the main control board wants to expand the SATA slot, it can insert the SATA slot into slot 2 and configure it for use.

UCP2120

4	7	11
3	6	10
2	5	9
1	CSU-F/G	8

- The slots are located on the front of the chassis. UCP2120 provides 2 main control board slots and 9 service board slots.
- Slots 2 to 7 and 9 to 11 are service board slots, which are used to install various service boards and support mixed insertion; the corresponding IP is: 172.16.80.1---172.16.80.11
- Slots 1 and 8 are main control board slots, used to install CCU series main control boards.

UCP4131

Service Board	Main Control Board	Service Board	Service Board	Service Board	CSU-F/G	Switchng Board	Service Board	Main Control Board	Service Board	Service Board	Service Board	Power Supply	Power Supply
1	2	3	4	5		6	7	8	9	10	11	12	13

- The slots are located on the front of the chassis. UCP4131 provides 2 main control board slots, 9 interface board slots, and 2 power supply slots.

- Slots 1, 3 to 6, and 8 to 11 are interface board slots, which are used to install WTU boards, AIU boards, ACU boards, RIU boards, and DTU boards. They support mixed insertion; the corresponding IPs are: 172.16.80.1---172.16.80.11
- Slots 2 and 7 are main control board slots, used to install CCU boards and support dual main control boards.

5.3 Single board installation guide

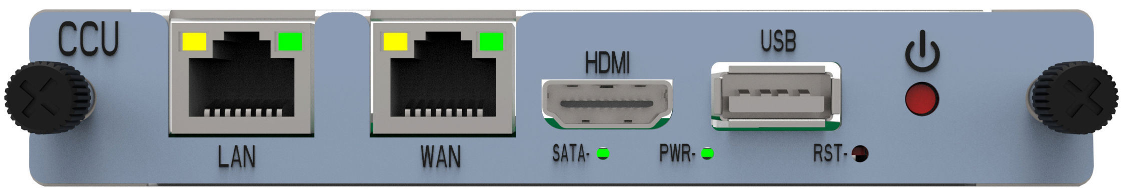
5.3.1 CCU main control board

CCU (Core Control Unit) is a main control board specially developed for VoIP applications. It uses Intel's high-performance processor to bring you high-fidelity, high-definition audio and video calls. In order to bring you clear, high-fidelity audio and video calls, CCU integrates a rich set of high-definition voice and video codecs and supports comprehensive protocol processing. The IP side supports SIP, IAX2 and other protocols, and the CPE side supports BRI, PRI, SS7, R2, GSM, WCDMA and other protocols. CCU can support Asterisk, Issabel, Elastix, FreePBX, VitalPBX, BrikerPBX and IPPBX/IVR and other open source software applications and private switch, firewall, IVR and voice gateway applications.

Product Details

Model	CPU	CPU core	Main frequency	memory	network card	Default storage	HDMI	USB
CCU-N-ALDER	3.40 G Intel N100	Quad core four threads	0.8 GHz	8GB DDR5	3	128G MSATA SSD	1	1
CCU-I-TGL	i5-1135	Four cores and eight threads	2.40GHz	Up to 32GB DDR4-3200	3	32G MSATA SSD	1	2

CCU main control board



Configure CCU main control board

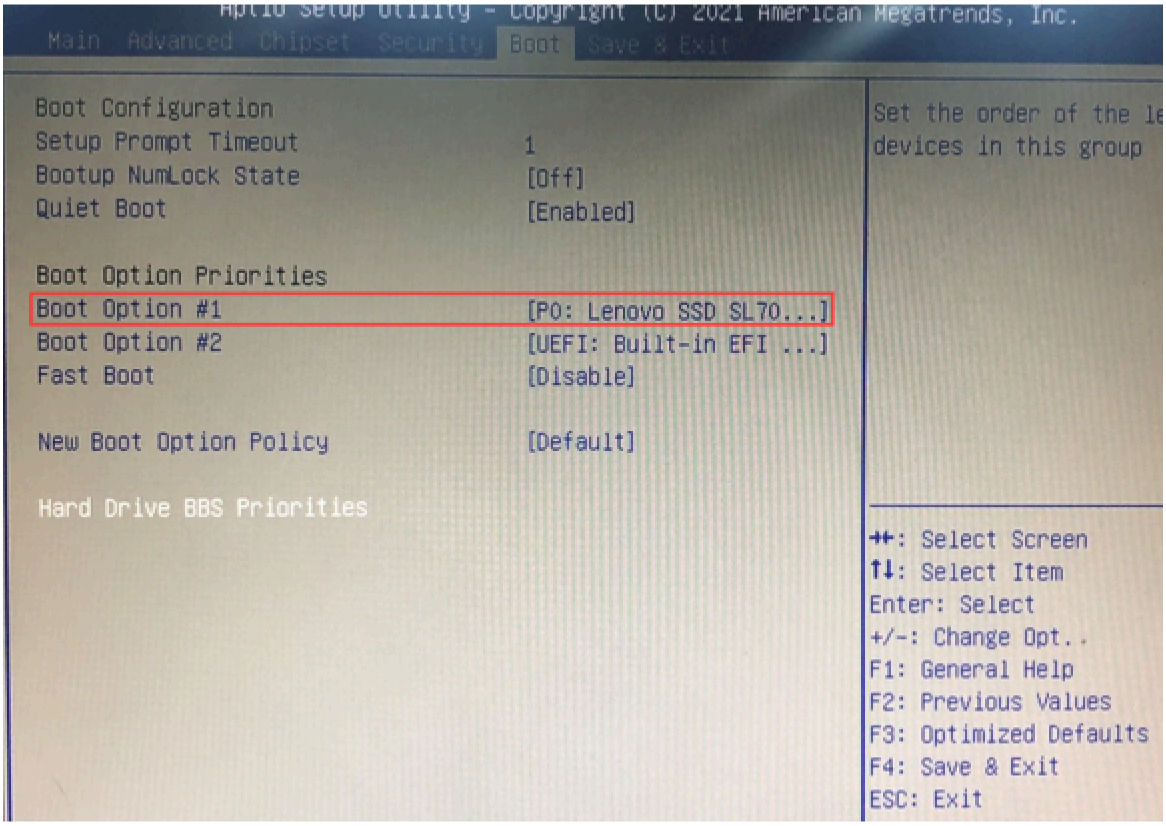
1. If the UC system is pre-installed, the default factory IP address for accessing the mainboard through the switch board CSU is: 172.16.80.200

The CCU series main control board can be configured through the PC's web browser. Please refer to the following steps:

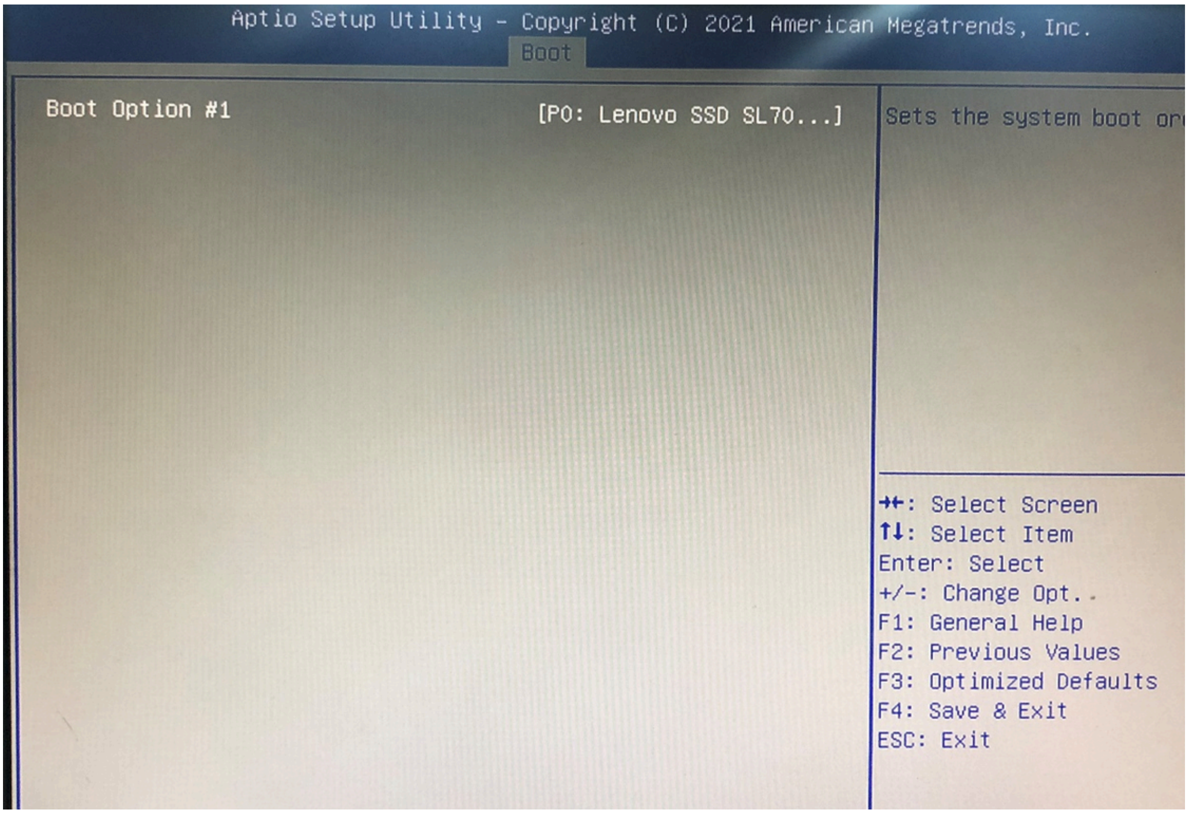
```
1. Please use AC/DC with correct specifications to power your UCP chassis.
2. Open a web browser on your computer
3. Enter the default IP address of the CCU main control board in the browser address bar.
4. Enter the administrator password to access the web configuration menu (by default, the
administrator username and password are admin)
```

Note: Please make sure your computer has an IP address in the 172.16.80.XXX/255.255.0.0 range so that you can access CCU's Web GUI

2. If Issabel, Elastix, FreePBX and other open source systems are pre-installed, you need to connect the network to the WAN port of the CCU motherboard. The default factory IP address for accessing the motherboard through the WAN port is: 172.16.99.98
3. If you need to manually install other Linux/windows systems, you can refer to the following steps:
    1. Connect the monitor through the HDMI interface on the CCU motherboard
    2. Insert the USB optical drive or U disk into the USB interface.
    3. Turn on the circuit.
    4. Steps to select boot from hard disk in BIOS: Power on, press the "Delete" button on the keyboard continuously to enter the BIOS setting interface, and select the SSD or HDD hard disk option as the first boot item in the Boot directory of the BIOS.

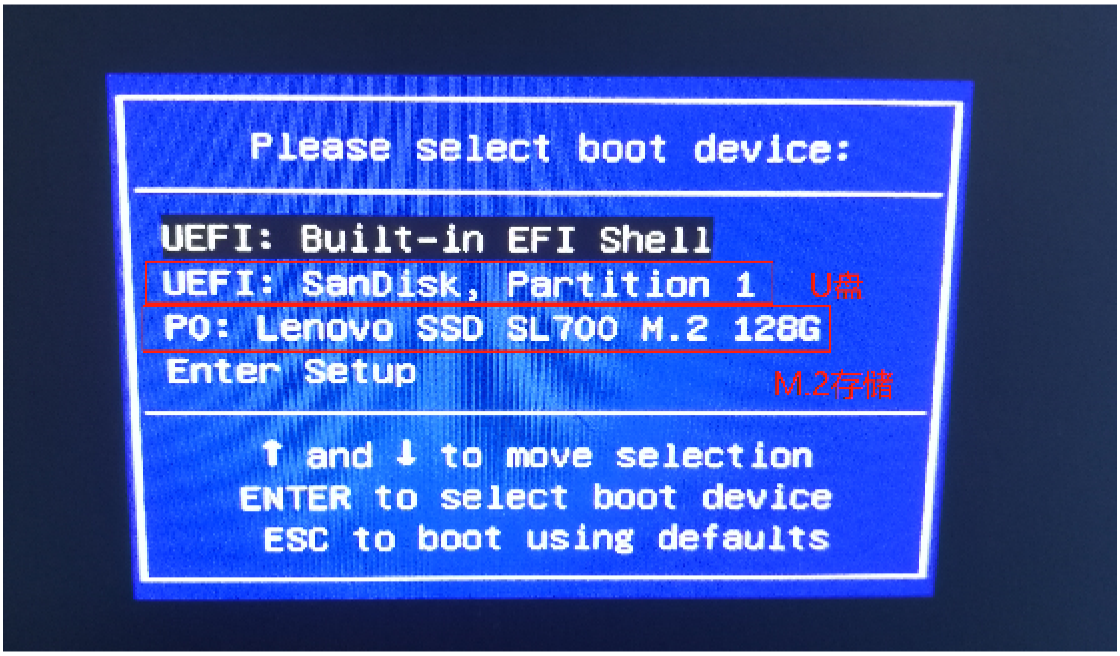


Click the "Hard Drive BBS Priorities" option in the Boot directory, select the SSD or HDD hard disk option as shown in the figure below, press F10 to save and exit the BIOS, and then restart the system.



5. Temporarily select the boot method from the USB optical drive/U disk from the startup item:

Power on, press the F7 button on the keyboard quickly and continuously, and the following interface will pop up. Select the U disk/USB optical drive or SSD/HDD hard disk option and press the Enter key on the keyboard. The user can install the system according to needs.



6. Next, you can directly boot and install the operating system through a U disk or USB optical drive.

5.3.2 WTU

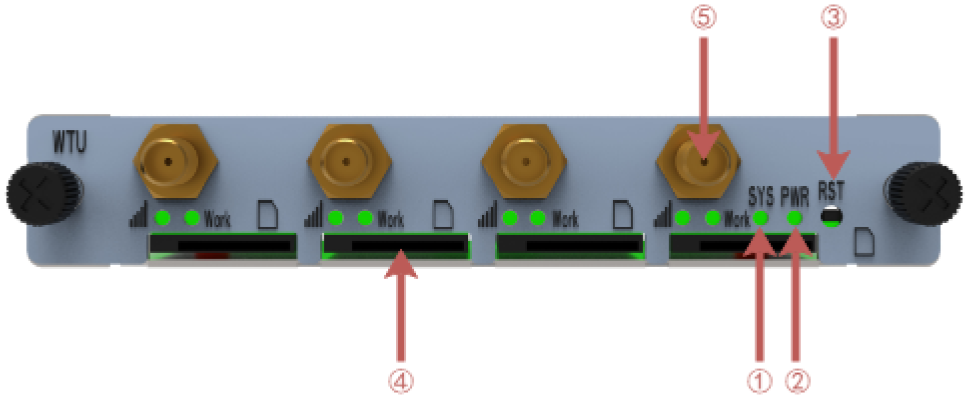
The WTU series wireless business boards are composed of two models: WTU-G that supports the 2G frequency band, and WTU-L that supports the 3G/4G frequency band. Each module provides 4 2G/3G/4G channels. Each WTU series wireless service board comes with its own main control and is an independent gateway system. It can be easily combined with other service boards to form a hybrid gateway in the UCP chassis.

The WTU series can provide excellent voice services, supporting voice codecs including G.711U, G.711A, GSM, G.722, G.726, and G.729, as well as flexible SMS services and HTTP-based APIs. WTU series wireless business boards are fully compatible with VoIP system platforms such as Asterisk, 3CX, FreePBX, FreeSWITCH and VOS, providing users with more diverse telecommunications access methods and reducing communication costs.

Product Details

Product Model	WTU-G	WTU-L		
Module Type	GSM	LTE		
Number of Channels	4			
Supported Bands	GSM: 850 / 900 / 1800 / 1900 MHz	**WTU-L-CE** China / India / Bay of Brazil LTE FDD: B1 / B3 / B5 / B8 / B1 / B2 / B3 / B4 / B5 / B7 / B8 / B28 LTE TDD: B40 TD-SCDMA: B34 / B39 CDMA: BC0 GSM: 900 / 1800 MHz	**WTU-L-E** Europe / Middle East / Africa / Korea / Thailand LTE FDD: B1 / B3 / B5 / B7 / B8 / B20 LTE TDD: B38 / B40 / B41 GSM: B3 / B8	**WTU-L-AU** Australia / New Zealand / Taiwan LTE FDD: LTE TDD: B38 / B40 / B41 GSM: B2 / B3 / B5 / B8
Power	28W			
Weight (excluding antenna)	164 g			
Dimensions (excluding antenna and mounting ear)	124 mm × 185 mm × 21 mm			
SIM Card	Hot-Swap			
Operating Temperature	0°C ~ 40°C			
Operating Humidity	10% ~ 90%			
Storage Temperature	-20°C ~ 70°C			

WTU Wireless Business Board



1. Running status indicator light
2. Power status indicator
3. reset button
4. SIM card slot
5. antenna

Connect to WTU wireless service board

The WTU series wireless service boards are designed for easy configuration and installation. To connect the WTU series wireless service boards, please follow the steps below:

```
1、Please ensure that a switch board is inserted into the UCP chassis, and plug the Ethernet cable into the LAN port of the switch board.
2、Insert the WTU series wireless service board into the chassis slot.
3、Plug the power adapter into the power port of the chassis and connect it to the wall socket to power on.
```

Configure WTU wireless service board

The default factory IP address is: 172.16.80.X (where X represents the slot number where the WTU service board is inserted into the chassis slot)

WTU series wireless service boards can be configured through the PC's web browser. Please refer to the following steps:

```
1、Please use AC/DC with correct specifications to power your UCP chassis.
2、Open a web browser on your computer
3、Enter the IP address of the WTU series wireless service board in the browser address bar
4、Enter the administrator password to access the web configuration menu (by default, the administrator username and password are admin)
```

Note: Please make sure your computer has an IP address in the 172.16.80.XXX range so that you can access the Web GUI of the WTU series wireless business board.

How to judge whether the WTU wireless service board is working normally

After the WTU series wireless service board is inserted into the chassis, you can determine whether it is working properly through the following methods:

```
1、After the UCP chassis is powered on, the PWR power indicator light is always on.
2、When the wireless service board system is running normally, the running status indicator SYS is in a slow flashing green state (the green light is on for 2 seconds and flashes for 0.1 seconds).
3、Before the SIM card is inserted, the SIM card working status indicator light is always off; the SIM card signal strength indicator light is red slowly flashing (on for 0.5s, off for 0.5s).
4、After inserting the SIM card, the SIM card working status indicator light and the SIM card signal strength indicator light are both steady green.
5、The Web GUI of the WTU wireless service board can be accessed normally using a computer web browser.
```

5.3.3 AIU-8

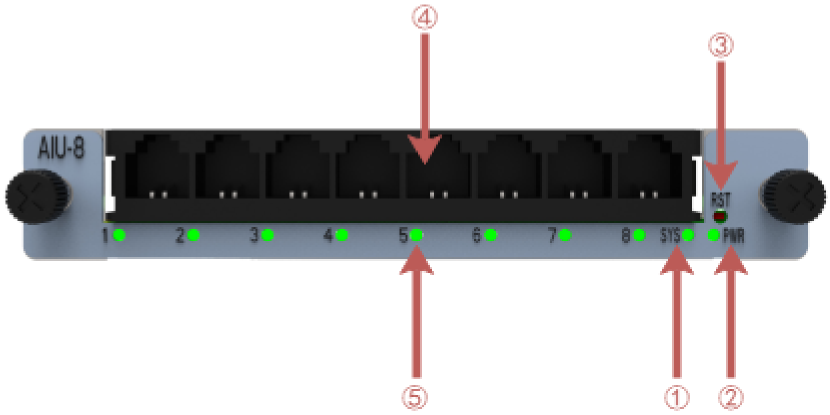
OpenVox AIU-8 series analog gateway modules are used with OpenVox VS-GW1200/1600/2120 series analog gateways to connect VoIP and PSTN. Each module provides 8 FXO/8 FXS/4 FXO and 4 FXS channels.

AIU-8 series analog gateway modules can bring you high-quality high-definition voice services, including G.711U, G.711A, GSM, G.722, G.729, ILBC and other codecs. The AIU-8 series analog gateway modules will be 100% compatible with Asterisk, 3CX, FreePBX, FreeSWITCH and VOS VoIP operating platforms to help users reduce telecommunications and communication costs.

Product Details

OpenVox AIU-8 Series Analog Gateway Module	
Product model	AIU-8
Number of channels	8
Power	12W
weight	162g
Size	130mm*210mm*210mm
Working environment temperature	0℃ ~ 50℃
Working environment humidity	10% ~ 90%
Storage environment temperature	-40℃ ~ 125℃

AIU-8 module port



1. Running status indicator light
2. Power status indicator
3. reset button
4. Analog interface
5. Interface working indicator light

Connect AIU-8 module

The AIU-8 series analog gateway module board is designed for easy configuration and installation. To connect the AIU-8 analog gateway module board is designed for easy configuration and installation, please follow the steps below.

1. Please ensure that a switch board is inserted into the UCP chassis, and plug the Ethernet cable into the LAN port of the switch board.
2. Insert the AIU-8 series analog gateway module board into the chassis slot.
3. Plug the power adapter into the power port of the chassis and connect it to the wall socket to power on.

Configure AIU-8 module

The default factory IP address is: 172.16.80.x

The AIU-8 series analog gateway module board can be configured through the PC's web browser. Please refer to the following steps:

1. Please use AC/DC with correct specifications to power your UCP chassis.
2. Open a web browser on your computer;
3. Enter the IP address of the AIU-8 module in the browser address bar;
4. Enter the administrator password to access the web configuration menu (by default, the administrator username and password are admin).

Note: Please ensure that your computer has an IP address in the 172.16.80.XXX range so that you can access the AIU-8's Web GUI

How to confirm whether the AIU-8 module is working properly

1. After the UCP chassis is powered on, the PWR power indicator light is always on.
2. According to different statuses, AIU-8 displays different lighting effects. Use the following figure to determine the status of the AIU-8 series analog gateway module board.

Status	Not wired	Wired	Off hook	Ring	call	PWR	RUN	After restoring factory keys
FXS	Green light is always on	Green light is always on	Green light flashes for 0.1s	The green light flashes for 0.1s (on for 0.1s and then stops for 0.1s)	The green light flashes quickly for 0.5s (on for 0.5s and then stops for 0.5s)	The green light is always on after powering on	Normal: Green light is on and flashing (1s) Abnormal: Steady on or off	Press for 8 seconds, the RUN light flashes quickly (on and off for 0.1 seconds), the module light flashes quickly (on and off for 0.1 seconds), and then enters the restart process. All lights except the PWR light go off, and the computer starts up and re-enters the initialization process.
FXO	The red light flashes slowly for 1s (on for 1s and then stops for 1s)	Red light always on	The red light flashes for 0.1s	The red light flashes for 0.1s (on for 0.1s and then stops for 0.1s)	The red light flashes quickly for 0.5s (on for 0.5s and then stops for 0.5s)	The green light is always on after powering on	Normal: Green light is on and flashing (1s) Abnormal: Steady on or off	

3. The AIU-8 module Web GUI can be accessed normally using a computer web browser.

5.3.4 AIU-16

OpenVox AIU-16 series analog gateway modules, each module provides 16 FXS channels.

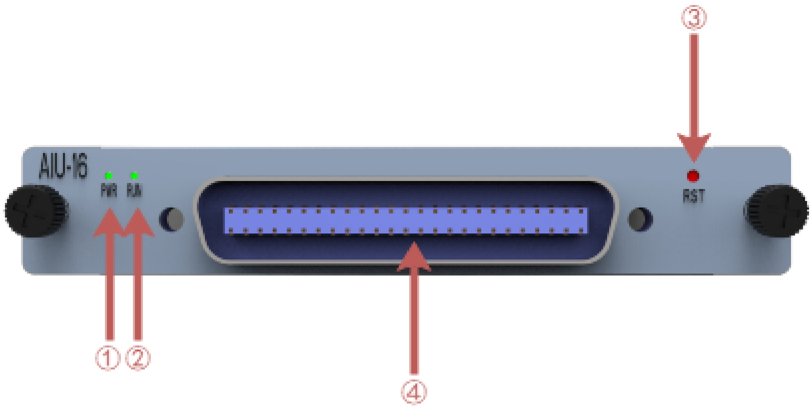
AIU-16 series analog gateway modules can bring you high-quality high-definition voice services, including G.711U, G.711A, GSM, G.722, G.729, ILBC and other codecs. The AIU-16 series analog gateway modules will be 100% compatible with Asterisk, 3CX, FreePBX, FreeSWITCH and VOS VoIP operating platforms to help users reduce telecommunications and communication costs.

Product Details

OpenVox AIU-16 Series Analog Gateway Module	
Product model	AIU-16
Number of channels	16
Power	12W
weight	189g
Size	158mm*100mm
Working environment temperature	0℃ ~ 50℃
Working environment humidity	10% ~ 90%

OpenVox AIU-16 Series Analog Gateway Module	
Storage environment temperature	-40℃ ~ 125℃

AIU-16 module port diagram



1. Power status indicator
2. Running status indicator light
3. reset button
4. Analog interface

Connect AIU-16 module

The AIU-16 series analog gateway module board is designed for easy configuration and installation. To connect the AIU-16 analog gateway module board is designed for easy configuration and installation, please follow the steps below.

1. Please ensure that a switch board is inserted into the UCP chassis, and plug the Ethernet cable into the LAN port of the switch board.
2. Insert the AIU-16 series analog gateway module board into the chassis slot.
3. Plug the power adapter into the power port of the chassis and connect it to the wall socket to power on.

Configure AIU-16 module

The default factory IP address is: 172.16.80.x

The AIU-16 series analog gateway module board can be configured through the PC's web browser. Please refer to the following steps:

1. Please use AC/DC with correct specifications to power your UCP chassis.
2. Open a web browser on your computer;
3. Enter the IP address of the AIU-16 module in the browser address bar;
4. Enter the administrator password to access the web configuration menu (by default, the administrator username and password are admin)

Note: Please ensure that your computer has an IP address in the 172.16.80.XXX range so that you can access the AIU-16's Web GUI

How does the AIU-16 module work properly

1. After the UCP chassis is powered on, the PWR power indicator light is always on.
2. According to different statuses, AIU-16 displays different lighting effects. Use the following figure to determine the status of the AIU-16 series analog gateway module board.

Status	Not wired	Wired	Off hook	Ring	call	PWR	RUN	After restoring factory keys
FXS	Green light is always on	Green light is always on	Green light flashes for 0.1s	The green light flashes for 0.1s (on for 0.1s and then stops for 0.1s)	The green light flashes quickly for 0.5s (on for 0.5s and then stops for 0.5s)	The green light is always on after powering on	Normal: Green light is on and flashing (1s) Abnormal: Steady on or off	Press for 8 seconds, the RUN light flashes quickly (on and off for 0.1 seconds), the module light flashes quickly (on and off for 0.1 seconds), and then enters the restart process. All lights except the PWR light go off, and the computer starts up and re-enters the initialization process.
FXO	The red light flashes slowly for 1s (on for 1s and then stops for 1s)	Red light always on	The red light flashes for 0.1s	The red light flashes for 0.1s (on for 0.1s and then stops for 0.1s)	The red light flashes quickly for 0.5s (on for 0.5s and then stops for 0.5s)	The green light is always on after powering on	Normal: Green light is on and flashing (1s) Abnormal: Steady on or off	

3. The AIU-16 module Web GUI can be accessed normally using a computer web browser.

5.3.5 DTU

DTU(L) series digital service boards are VoIP trunk voice service boards developed based on Asterisk open source software and specially designed for operators and call centers. This is a converged multimedia service board that can connect traditional telephone systems with IP networks to achieve seamless connection between VoIP, PBX and PSNT. The user-friendly interface and simple operation methods allow users to easily set up personalized services. Users can realize secondary development of business functions through AMI (Asterisk Management Interface).

DTU(L) series digital service boards support 1/2/4 optional T1/E1/PRI interfaces, with up to 30/60/120 concurrent calls. "L" means that the device does not have the hardware codec module V100. The device supports multiple codecs and signaling, including G.711A, G.711U, G.729, G.722, G.723 and GSM. It supports PRI/SS7/R2 protocols. DTU(L) series digital business boards have good processing capabilities and stability. We provide 1/2/4 T1/E1 protocols for your choice. DTU(L) series digital business boards are perfectly compatible with SIP servers, such as Asterisk, Elastix, trixbox, 3CX, FreeSWITCH and other VoIP operation platforms.

Connect to DTU digital service board

The DTU series digital business boards are designed for easy configuration and installation. To connect the DTU digital business boards, please follow the steps below:

```
1、Please ensure that a switch board is inserted into the UCP chassis, and plug the Ethernet
cable into the LAN port of the switch board.
2、Insert the DTU series digital service board into the chassis slot.
3、Plug the power adapter into the power port of the chassis and connect it to the wall
socket to power on.
```

Configure DTU digital service board

The default factory IP address is: 172.16.80.X (where X represents the slot number where the DTU service board is inserted into the chassis slot)

DTU series digital business boards can be configured through the web browser of PC. Please refer to the following steps: 1、Please use AC/DC with correct specifications to power your UCP chassis. 2、Open a web browser on your computer. 3、Enter the IP address of the DTU digital service board in the browser address bar. 4、Enter the administrator password to access the web configuration menu (by default, the administrator username and password are admin).

Note: Please ensure that your computer has an IP address in the 172.16.80.XXX range so that you can access the DTU Digital Business Board's Web GUI.

How to judge whether the DTU(L) digital service board is working normally

After the DTU series digital service board is inserted into the chassis, you can determine whether it is working properly through the following methods:

```
1、After the UCP chassis is powered on, the PWR power indicator light is always on.
2、When the DTU series digital service board system is running normally, the operating
status indicator SYS is in a slow flashing green state (the green light is on for 2 seconds
and flashes for 0.1 seconds).
3、Before connecting to the E1 line, the E1 port indicator light flashes red slowly (on for
0.5s and off for 0.5s).
4、After inserting the E1 cable, the E1 port indicator light is always green.
5、The Web GUI of the DTU digital business board can be accessed normally using a computer
web browser.
```

5.3.6 ACU

The ACU audio service board is a powerful voice access device that facilitates the connection between conference calls and conference room audio mixer systems. It realizes the reception and amplification of conference calls and transmits them through telephone lines. It meets the application needs of video and telephone conferencing in different industries, ensuring clear voice quality and powerful telephone access control capabilities. It not only supports PSTN access but also SIP line registration, making it the first choice for conference room applications that require high reliability.

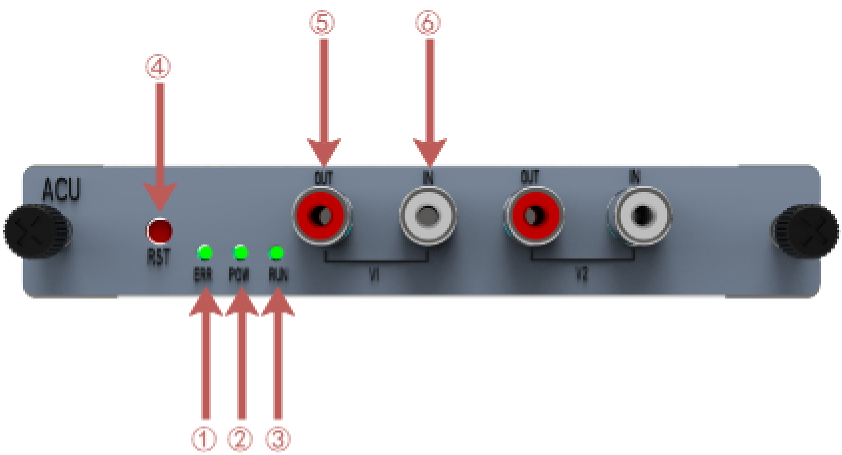
The ACU audio business board adopts a carrier-grade product design and has strong networking capabilities and sound processing capabilities. It adopts microcomputer chip technology and electronic switch technology. Each channel of control is independent of each other, and the operation of cutting in and out of audio signals is sensitive. It can realize simultaneous access of 2-way intercom. The infinite cluster board can be widely used in special command and dispatch systems in various industries.

The device provides two audio control interfaces. Users can connect the phone audio to the conference room through external IP phones, computers, central control and other equipment, so that the two conference rooms can share an audio access station device. The ACU audio business board is flexible and simple to deploy, and can be used with simple connections. Supports simultaneous access to PSTN, SIP, and GSM.

Product Details

OpenVox ACU audio business board	
port type	AV lotus head
port protocol	Standard SIP, TCP/IP
Size	160mm*100mm
weight	0.132kg
Capacity	2-way audio input and output
language encoding	G.711a、G.711U
Distortion	≤ 1%
signal-to-noise ratio	≥ 70dB (excluding telephone line noise)
side tone suppression	≥45dB
Maximum power	2W
Power specifications	12VDC
Minimum ventilation space	6.4cm
Working environment temperature	-20℃ ~ 60℃
Working environment humidity	8% ~ 90% no condensation

ACU service board port



1. fault light
2. power light

3. running light
4. reset button
5. audio output
6. Audio input

Note:

1. After the equipment is running normally, the power light will be steady green, the operation indicator light will flash green, and the fault light will remain temporarily useless.
2. Press and hold the RST key for more than 10 seconds to restore the temporary IP address 10.20.30.1. The original IP will be restored after power failure and restart.
3. V1 is the first audio channel, red is OUT which is the audio output, white is IN which is the audio input; V2 is the second channel.

Connect to the ACU business board

The ACU series audio service boards are designed to facilitate configuration and installation. To connect to the ACU, follow the steps below.

1. Please ensure that the CSU-F/G switching board is inserted into the UCP1600/2120/4131 series chassis, and plug the Ethernet cable into the LAN port of the switching board.
2. Insert the ACU series service board into the chassis slot.
3. Plug the power adapter into the power port of the chassis and connect it to the wall socket to power on.

Configure the ACU service board

The default factory IP address is: 10.20.40.40

The ACU audio service board can be configured through the PC's web browser. Please refer to the following steps:

1. Please use AC/DC with correct specifications to power your UCP chassis.
2. Open a web browser on your computer
3. Enter the IP address of the ACU service board in the browser address bar.
4. Enter the administrator password to access the web configuration menu (by default, the administrator username is admin and the password is 1)

NOTE: Please make sure your computer has an IP address in the 10.20.40.XXX range so that you can access ACU's Web GUI

5.3.7 RIU

The RIU wireless trunking service board is a powerful voice access device that facilitates the integration of the intercom trunking system and the telephone system. Users can conveniently call the intercom through the phone, or use the intercom to make calls. The system supports traditional PSTN phone lines and SIP-based VoIP phone lines. It is very convenient to deploy and use, and can be plug-and-play.

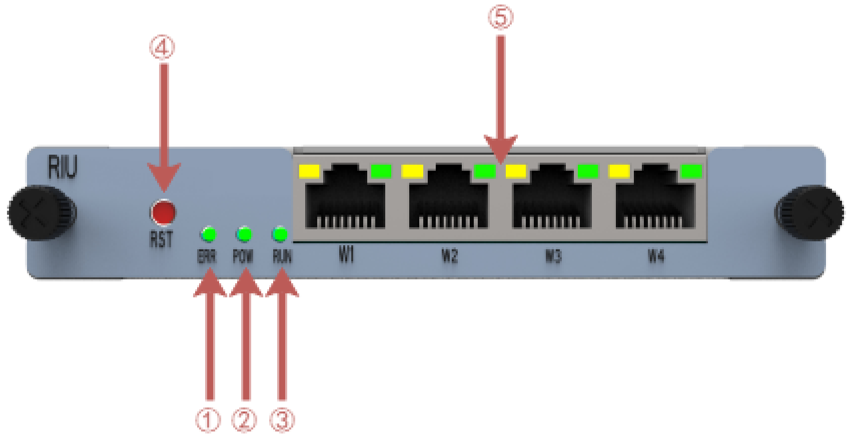
The wireless cluster business board adopts carrier-grade product design and has strong networking capabilities and sound processing capabilities. It adopts microcomputer chip technology and electronic switch technology. Each channel of control is independent of each other, and the audio signal operation is sensitive when cutting in and out. It can realize simultaneous access of 2-way intercom. It can be widely used in special command and dispatch systems in various industries. The device provides 2/4-way intercom interfaces, uses RJ45 plugs, and is equipped with professional intercom control cables. It is compatible with mainstream intercom handsets and car radios such as Motorola and Kenwood.

In addition, the wireless trunking business board has developed a voice algorithm based on experience in the multi-party voice field, making its call effect higher than similar products currently on the market. Its intercom gateway is flexible and simple to deploy and can be used with a simple connection. Supports PSTN and SIP, compatible with multiple models of mobile phones and radio stations.

Product Details

OpenVox RIU series wireless trunking service board	
Product model	RIU-4
port type	Support RJ45
port protocol	Standard SIP, TCP/IP
Size	160*100MM
weight	132g
Panel color	gray
Capacity	4 channels of radio or audio input and output
language encoding	G711、G729、G723
radio control	PTT、VOX、COR
Distortion	≤ 1%
signal-to-noise ratio	≥ 70dB (excluding telephone line noise)
side tone suppression	≥45dB
Maximum power	5W
Power specifications	12VDC
Minimum ventilation space	6.4CM
Working environment temperature	-20℃ ~ 60℃
Working environment humidity	8% ~ 90% no condensation

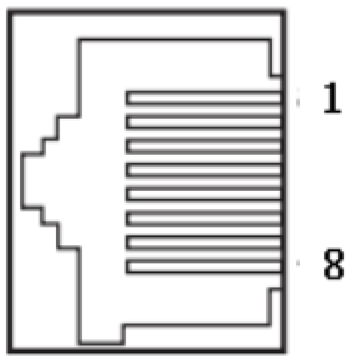
RIU service board port



1. Fault status indicator light
2. Power status indicator
3. Running status indicator light
4. reset button
5. W interface

Note:

1. After the equipment is running normally, the power light will be steady green, the operation indicator light will flash green, and the fault light will not light up.
2. Reset key: Short press to reset, long press for more than 5 seconds to turn off the watchdog, and the E light will turn on. Press and hold for more than 10 seconds to restore the temporary IP address 10.20.30.1, and restore the original IP after powering off and restarting.
3. The W interface is defined as follows:



1: PTT
2: GND
3: PTT_GND
4: COR
5: SPK
6: MIC
7: GND
8: PWR

Connect to RIU business board

The RIU wireless cluster service board is designed to facilitate configuration and installation. To connect to the RIU, follow the steps below.

1、Please ensure that the CSU-F/G switching board is inserted into the UCP1600/2120/4131 series chassis, and plug the Ethernet cable into the LAN port of the switching board. 2、Insert the RIU cluster board into the chassis slot. 3、Plug the power adapter into the power port of the chassis and connect it to the wall socket to power on.

Configure RIU module

The default factory IP address is: http://10.20.40.40

The RIU cluster board can be configured through the PC's web browser. Please refer to the following steps:

1、Please use AC/DC with correct specifications to power your UCP chassis. 2、Open a web browser on your computer; 3、Enter the IP address of the RIU cluster board in the browser address bar; 4、Enter the administrator password to access the web configuration menu (by default, the administrator username is admin and the password is 1)

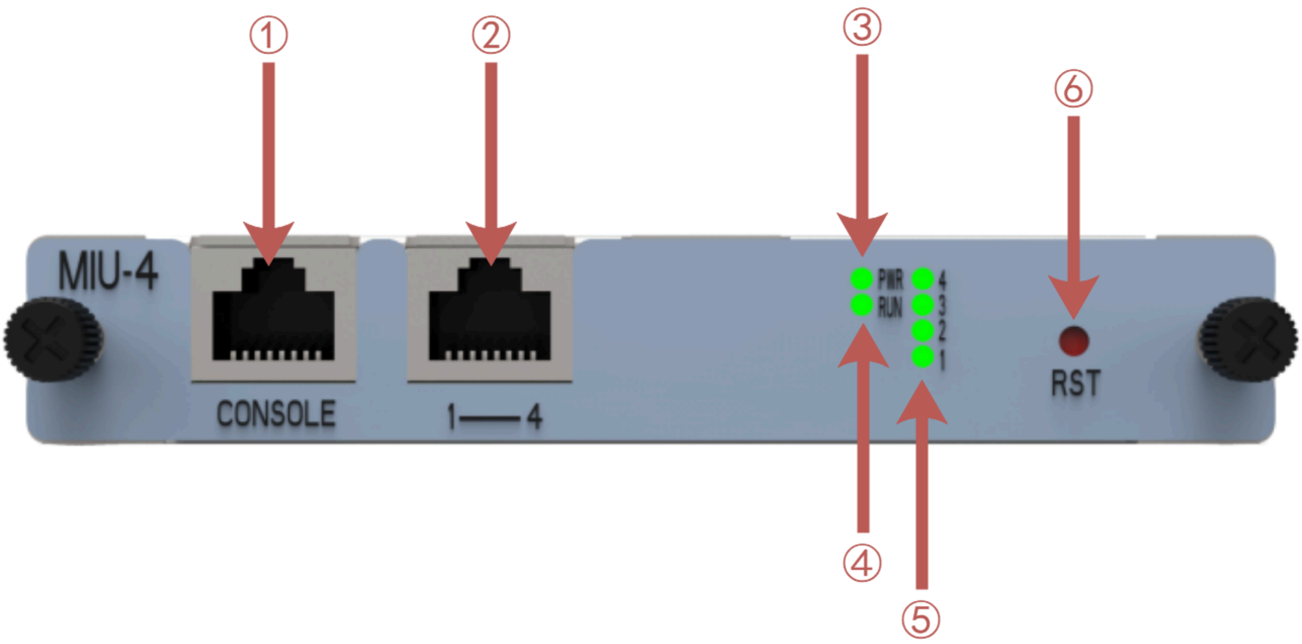
NOTE: Please make sure your computer has an IP address in the 10.20.40.XXX range so that you can access RIU's Web GUI

5.3.8 MIU

MIU-4/8 (Magnet Interface Unit) magnet relay gateway is a powerful voice access device, including MIU-4 and MIU-8, which provides the ability to access 4-way or 8-way magnet phones to the IP network. Using the standard SIP protocol, it can be connected to the standard SIP soft switching system to provide VoIP/MoIP solutions for magnet phone users. The device is easy to deploy quickly, has a high degree of security and strong anti-interference capabilities, and is widely used in military or other special command and dispatch systems and other fields.

The MIU-4/8 (Magnet Interface Unit) panel supports 1/2 magnet channel interface, Console interface, running status indicator light, power status indicator light, channel status indicator light and RST restart button. In terms of software docking, MIU uses standard SIP protocol, is compatible with mainstream IPPBX and SIP servers, and supports most VoIP operating system platforms, such as Asterisk, Issabel, 3CX, FreeSwitch, BroadSoft, VOS, etc.

MIU service board port



1. Console serial port
2. Magnet channel interface

3. Power status indicator
4. Running status indicator light
5. Channel status indicator
6. RST button

Connect MIU module

The MIU series magnet gateway module board is designed for easy configuration and installation. To connect the MIU magnet gateway module board, please follow the steps below.

1. Please ensure that a switch board is inserted into the UCP chassis, and plug the Ethernet cable into the LAN port of the switch board.
2. Insert the MIU series magnet gateway module board into the chassis slot.
3. Plug the power adapter into the power port of the chassis and connect it to the socket to power on.

Configure RIU module

The default factory IP address is: 172.16.80.x (x is determined according to the slot number where the MIU is inserted)

The MIU series magnet gateway module board can be configured through the PC's web browser. Please refer to the following steps:

1. Please use AC/DC with correct specifications to power your UCP chassis.
2. Open a web browser on your computer
3. Enter the IP address of the MIU module in the browser address bar
4. Enter the administrator password to access the web configuration menu (by default, the administrator username and password are admin)

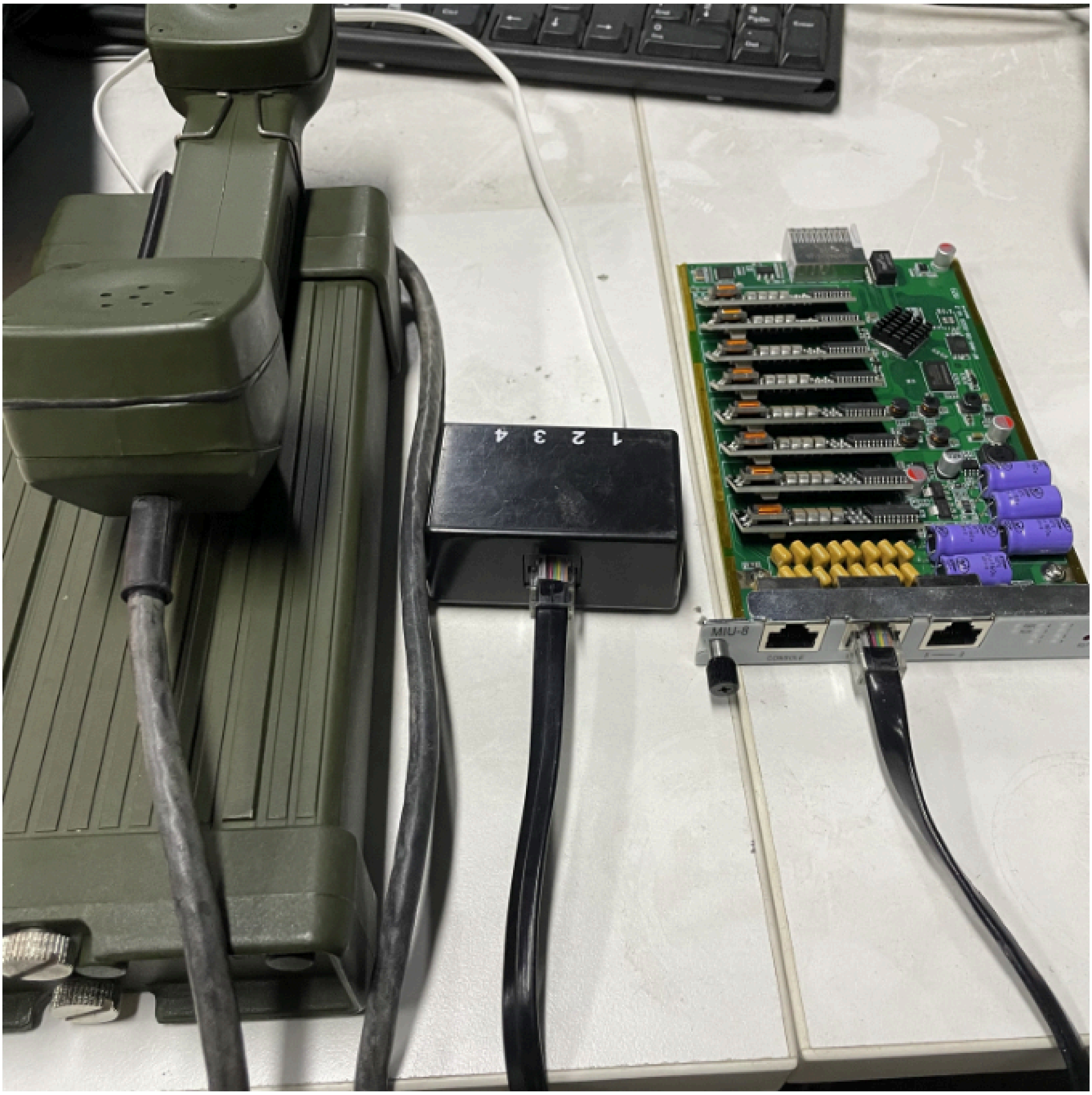
Note: Please make sure your computer has an IP address in the 172.16.80.XXX range so that you can access MIU's Web GUI

MIU module usage example

1. Installation preparation

Use a telephone junction box to connect the magnet phone to the MIU module


1
2
3
4



2. SIP call with magnet phone

①Create server-side SIP account 8002 ②Create 2 routes, namely SIP(8002)→mag, mag→SIP(8002)

	排序	优先级	规则名称	从	到	操作
		1	out	mag-1	sip-8002	
		2	in	sip-8002	mag-1	

SIP→Magnet Phone The soft phone registers SIP (8002) and calls 8001. The magnet phone will automatically connect after ringing 2 times. Press and hold the side button of the handset to start the call (the electronic magnet phone can start the call by picking up the phone).

Magnet phone→SIP The softphone registers SIP (8002), press the ring button twice on the magnet phone, the softphone receives the call, and you can start the call after the call is connected (the magnet phone still needs to hold down the side button or go off-hook)

3. Magnet phone calls to magnet phone

Create 2 routes, mag-1→mag-2, mag-2→mag-1

创建成功

☐	排序	优先级	规则名称	从	到	操作
☐		1	MAG1_2	mag-1	mag-2	
☐		2	MAG2_1	mag-2	mag-1	

Magnet phone 1→Magnet phone 2

Magnet phone A presses the ring button twice. Magnet phone B will automatically connect after ringing 2 times. Press and hold the side button of the earpiece to start the call (electronic magnet phone picks up the phone to start the call)

Magnet phone lighting effect description

Idle state: The power light is always on, the running light flashes slowly, and the channel light is off. Ringing status: The power light is always on, the running light flashes slowly, and the channel light is always on. Call status: power light is always on, running light flashes slowly, channel light flashes slowly

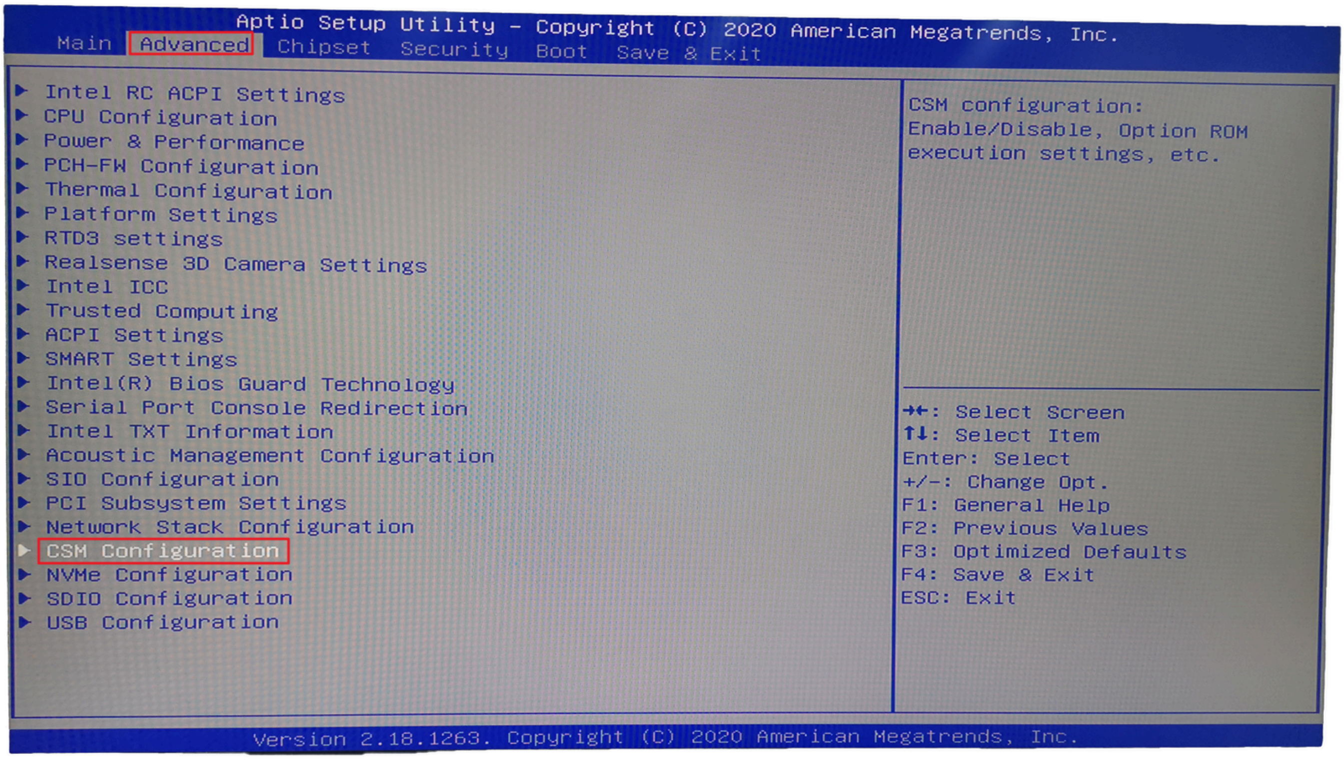
RST key reset: After pressing the RST key for about 5 seconds, the running light flashes quickly, then goes out, turns to steady light after a period of time, and finally resumes slow flashing, and the power light remains steady throughout the process.

5.3.9 RSU

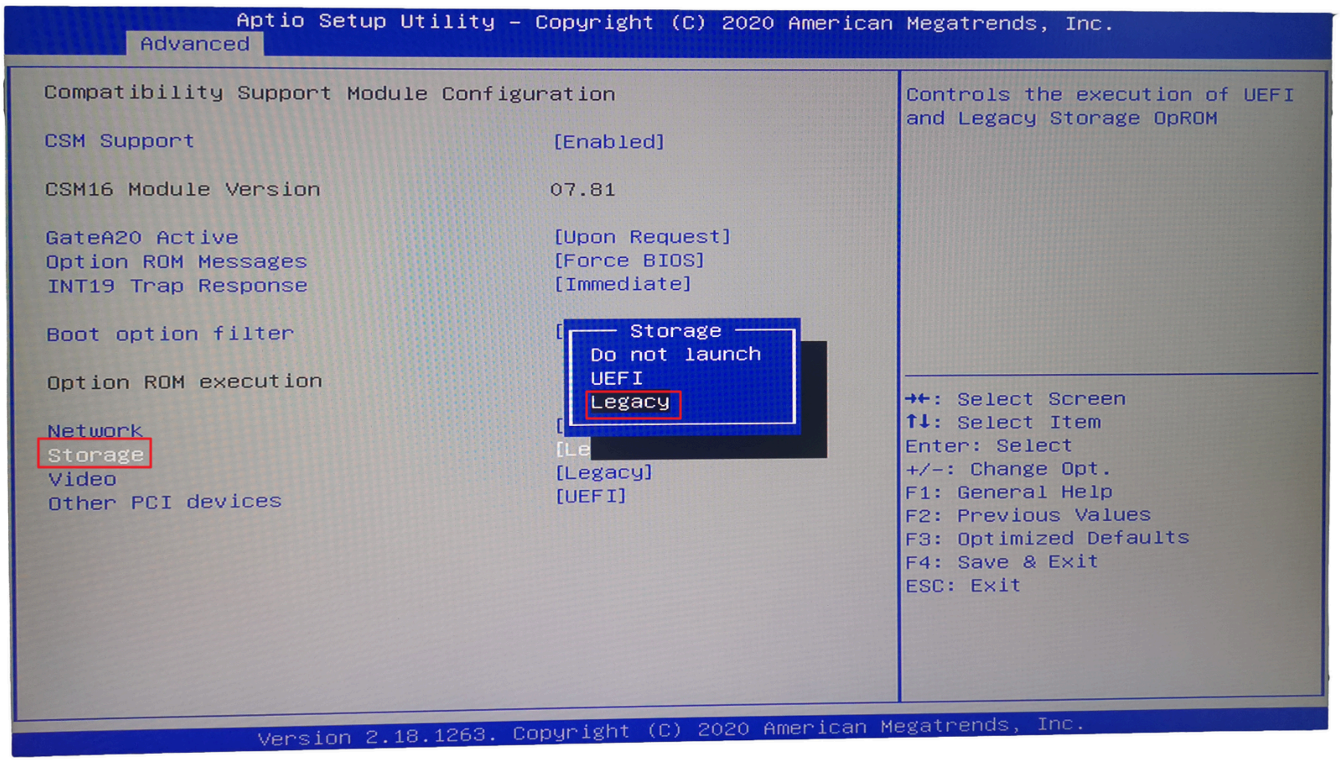
First, properly install the RAID card and two hard disks while disconnecting the power supply.

CSM Compatible Module Settings

During the power-on process, repeatedly press the BIOS shortcut key (press the Esc or Delete key) to interrupt the startup and enter the BIOS setting interface. Enter BIOS settings and select Advanced→CSM configuration.



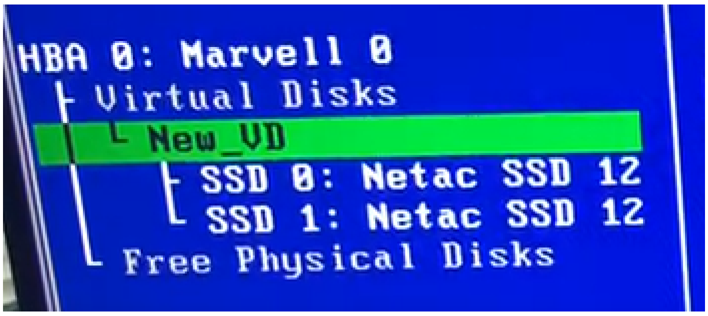
Confirm that the Storage parameter is set to legacy, save the settings and then restart.



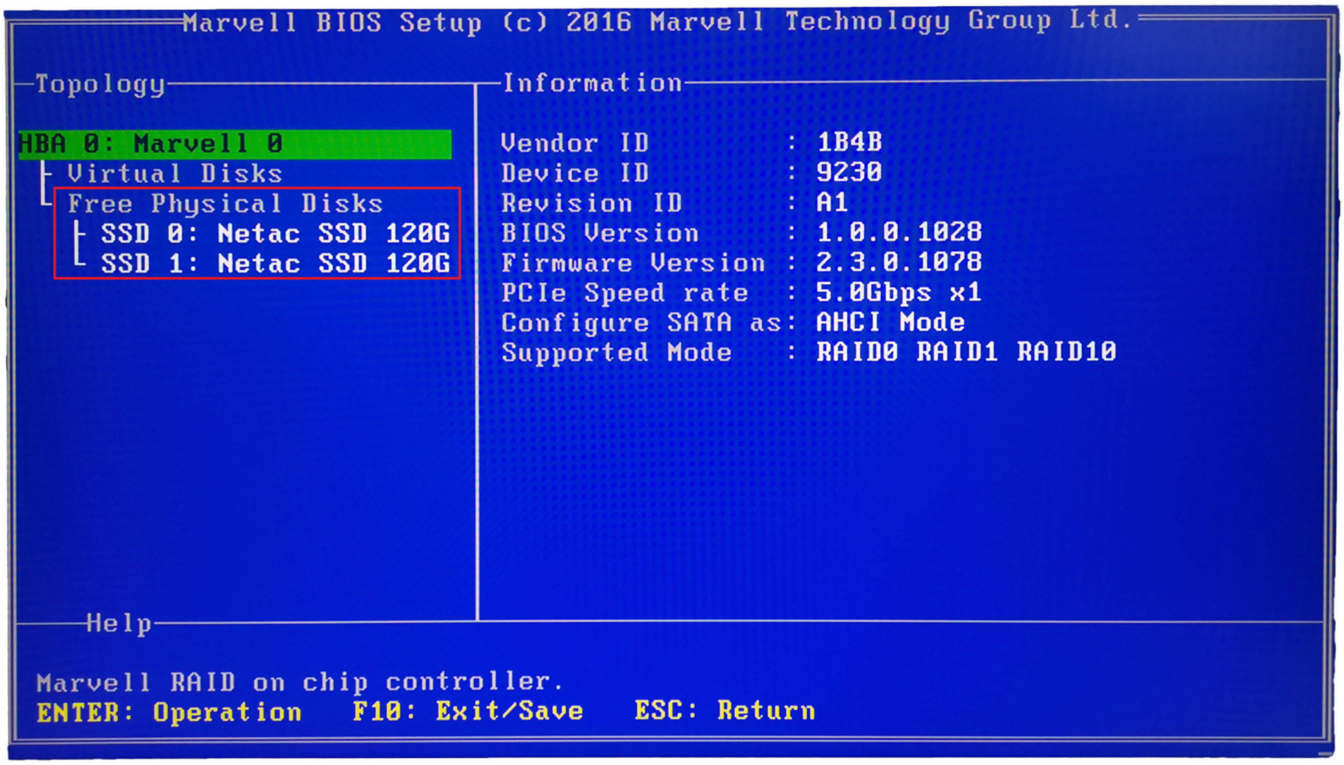
Configure RAID mode

After powering on, keep pressing Ctrl + m to enter the Raid card and start configuring Raid mode.

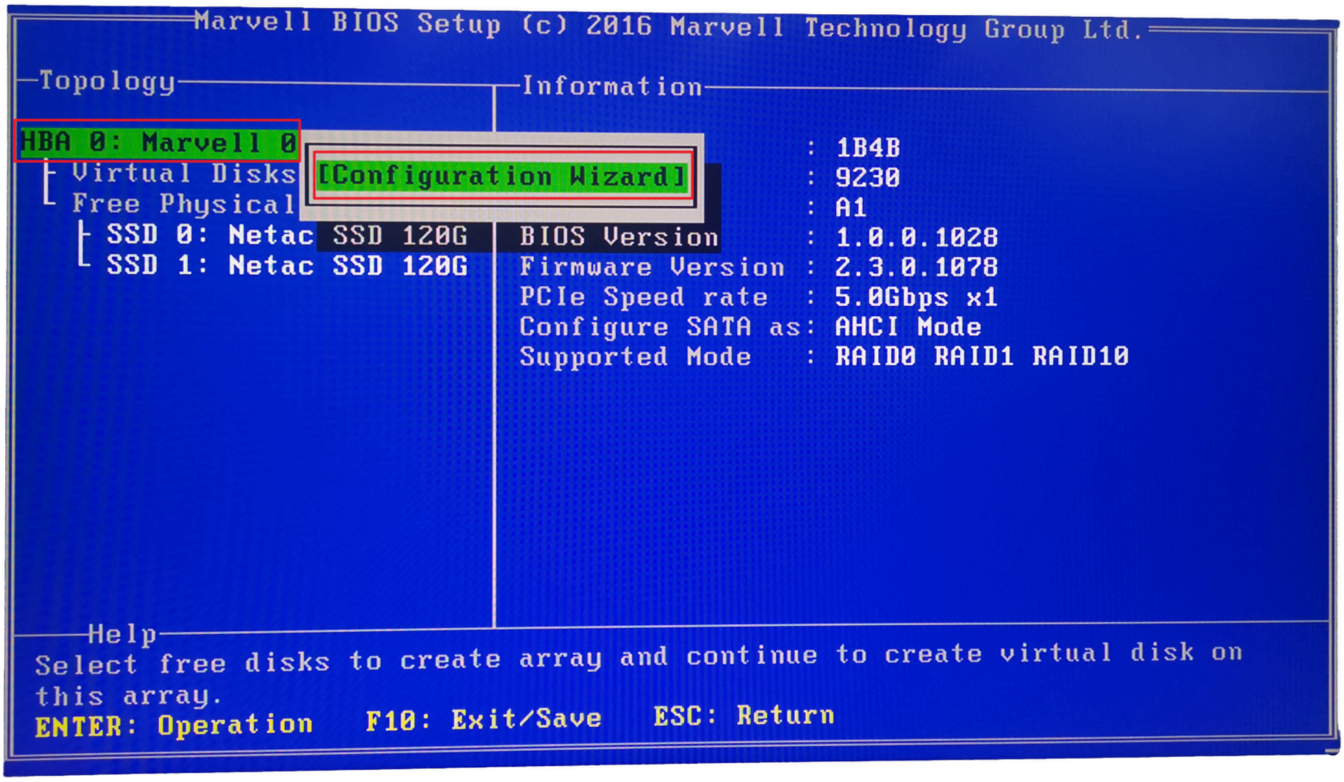
Enter the configuration interface, select HBA 0: Marvell 0 and press Enter. If the Configuration Wizard cannot select, you should see two hard drives in the New_VD directory.



Select New_VD and press Enter, select delete, and press Y to confirm. At this time, the two hard disks are transferred to Free Physical Disks.

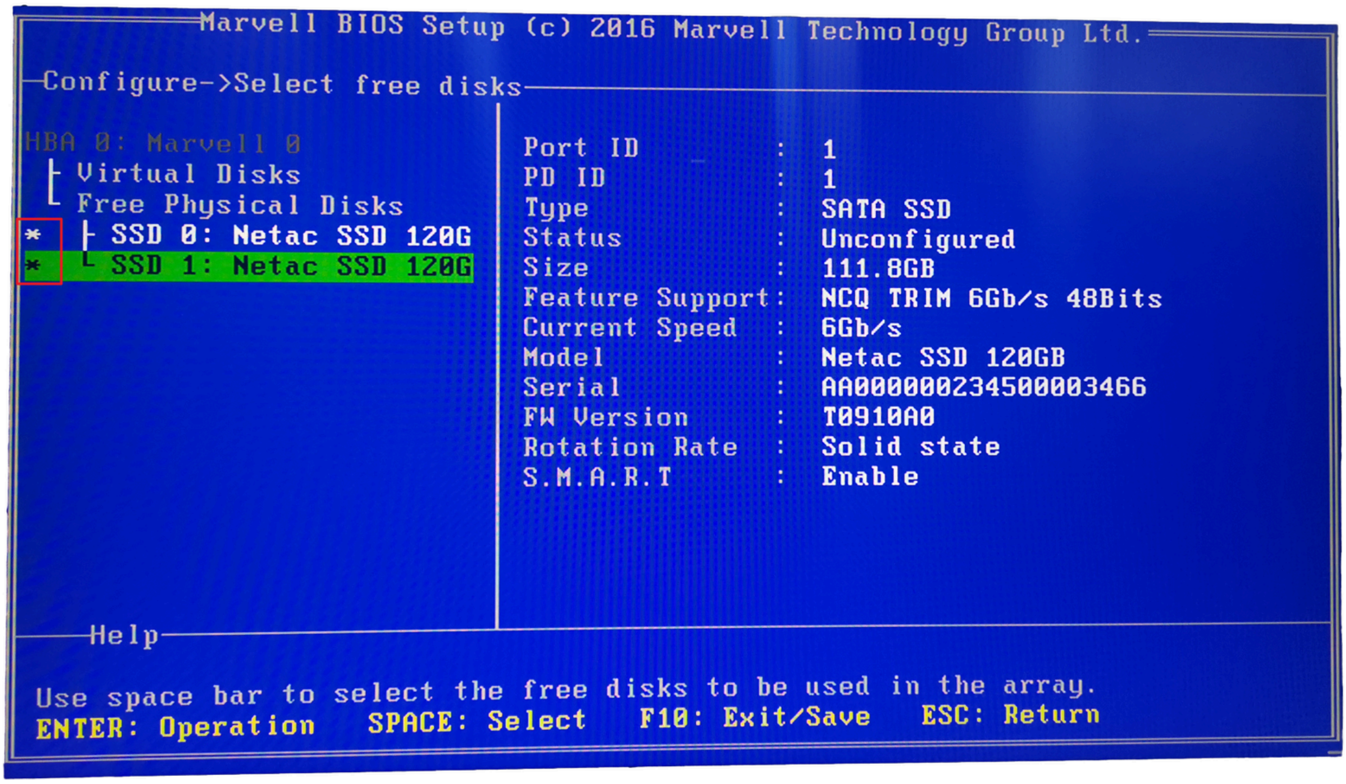


Select HBA 0: Marvell 0 and press Enter. Select Configuration Wizard and press Enter.



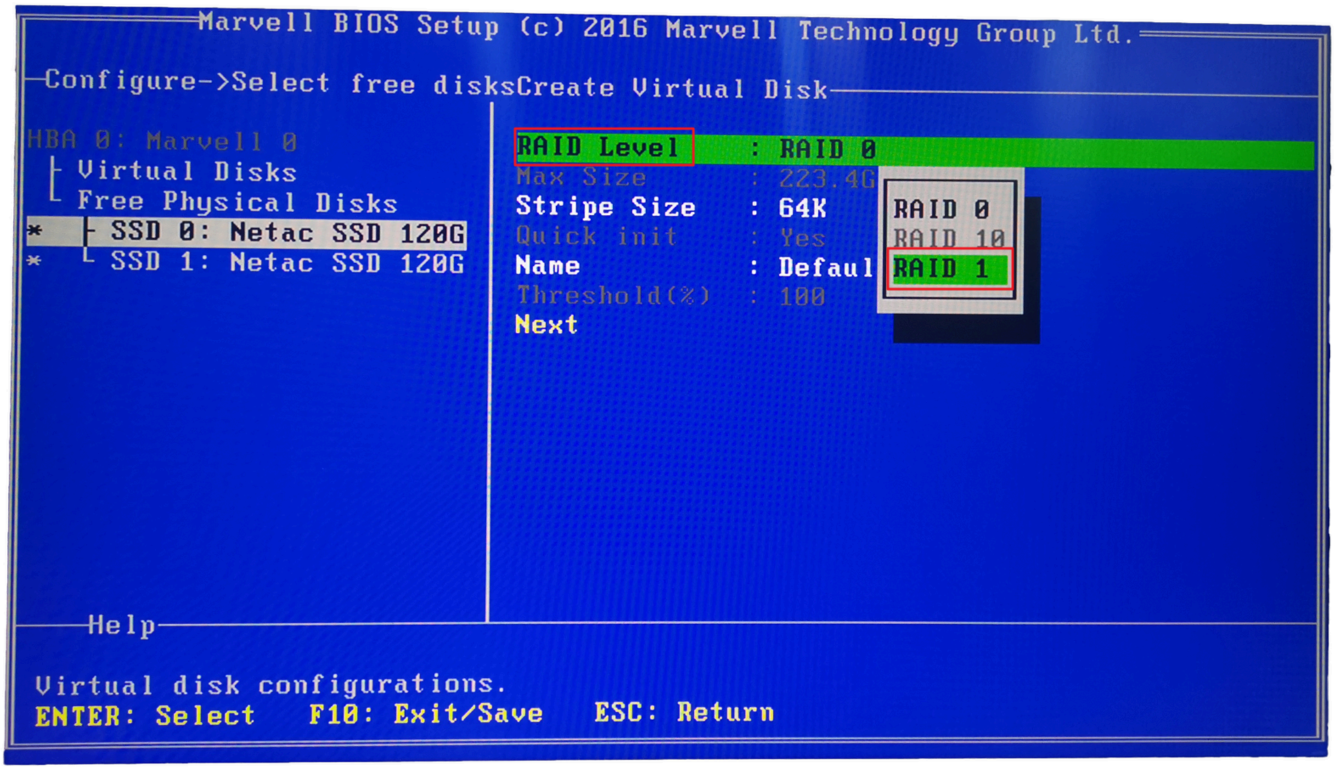
Select two hard drives in turn and press space to select.

After selection, there will be a * mark in front of the corresponding hard disk.

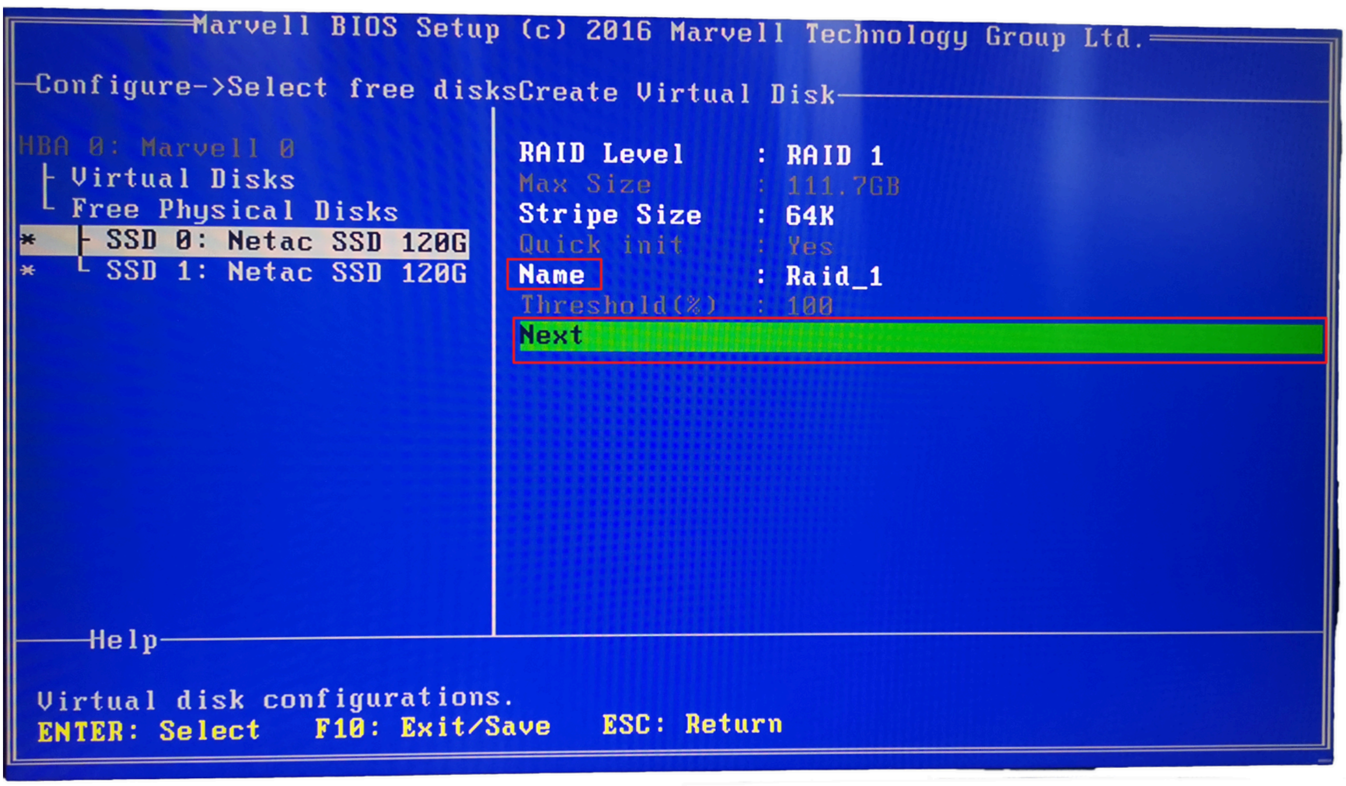


After selecting the hard disk, move the cursor to the hard disk numbered 0 and press Enter.

Press Enter at RAID Level on the right and select RAID 1.

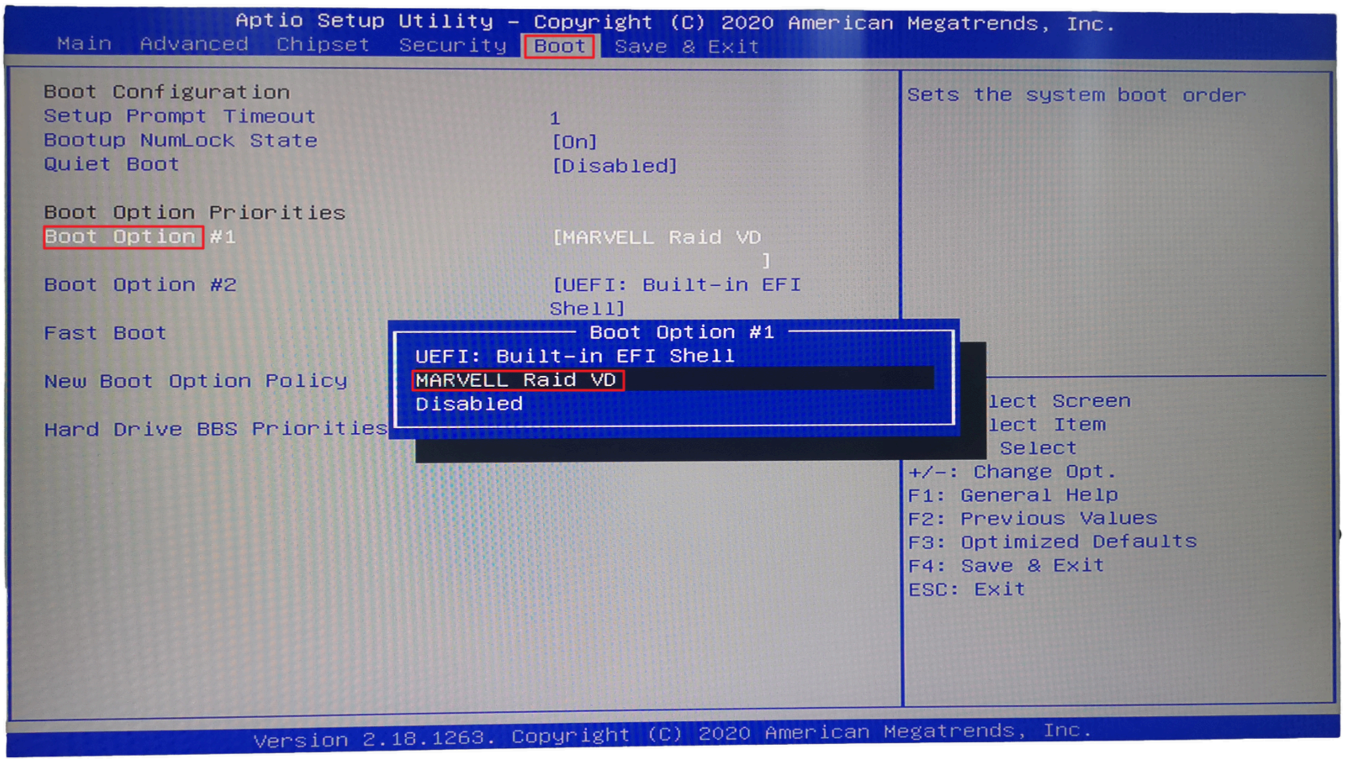


Name may be modified as appropriate.



Select Next and press Enter to confirm. F10 to save and exit, press Y to confirm.

After powering on, press the Esc or Delete key to enter the BIOS settings. In Boot -> Boot Option, you can see the MARCELL Raid VD option, indicating that the setting is successful. Just set this item as the preference.



6. Business board docking scenario

6.1 Business Board Interconnection Guide

Understand the contents of docking class documents and how to use the information provided in these documents to complete data configuration.

Content description of docking documents

Unified communications platform docking class documents include configuration examples. The content description of these configuration documents is shown in Table 1.

Table 1 Content description of configuration documents

Document name	Document content description
Configure instance	Describes the docking examples of the unified communications platform in various typical application scenarios, including OpenVox UC, 3CX, Asterisk, Freepbx and other examples.

How to use docking class documents

Based on the configuration documents provided by the unified communications platform, please complete the data configuration as follows.

- If there is a interconnection instance that matches the live network scenario, complete the data configuration according to the interconnection instance and the configuration guide.
- If there is no interconnection instance matching the live network scenario, complete the data configuration according to the configuration guide.

6.2 Business board interconnection instructions

6.2.1 WTU

This document mainly describes the detailed steps for interconnecting the WTU service board with the main control system 3CX, Asterisk16.15.1, FreePBX15, Freeswitch1.10.5, and Elastix4.0.

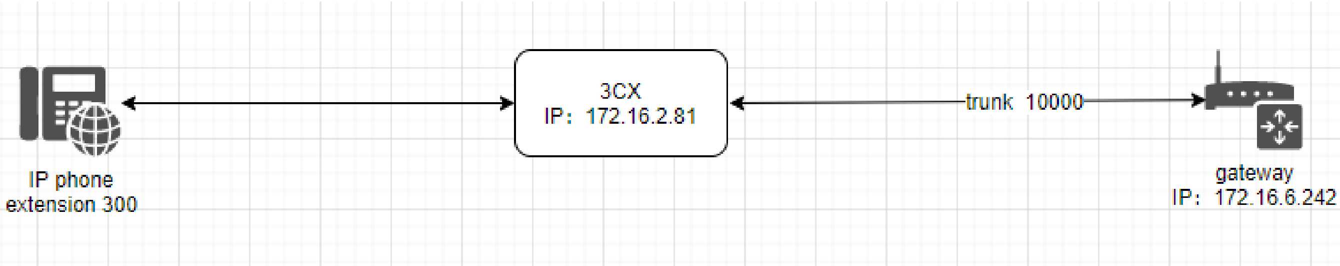
- The WTU business board is connected to the main control system 3CX Mainly describes the detailed steps for interconnecting the WTU service board and the main control system 3CX.
- WTU service board interconnects with Asterisk Mainly describes the detailed steps for interconnecting the service board with Asterisk16.15.1.
- WTU business board connects to FreePBX Mainly describes the detailed steps for connecting the service board to FreePBX15.
- WTU service board interconnects with FreeSwitch Mainly describes the detailed steps for interconnecting the service board with FreeSwitch1.10.5.
- WTU business board interconnection with Elastix Mainly describes the detailed steps for interconnecting the business board with Elastix4.0.

6.2.1.1 Interconnection between WTU service board and main control system 3CX

This document mainly describes the detailed steps for interconnecting the WTU service board and the main control system 3CX.

Follow the following steps to configure a two-way call between the phone and the gateway:

Outbound calls: sent from 3CX SIP extension 300 to the gateway through trunk 10000; Incoming call: From the outside call to the gateway, go to 3CX via SIP trunk 10000, and then send the call to 300 SIP extension through 3CX;

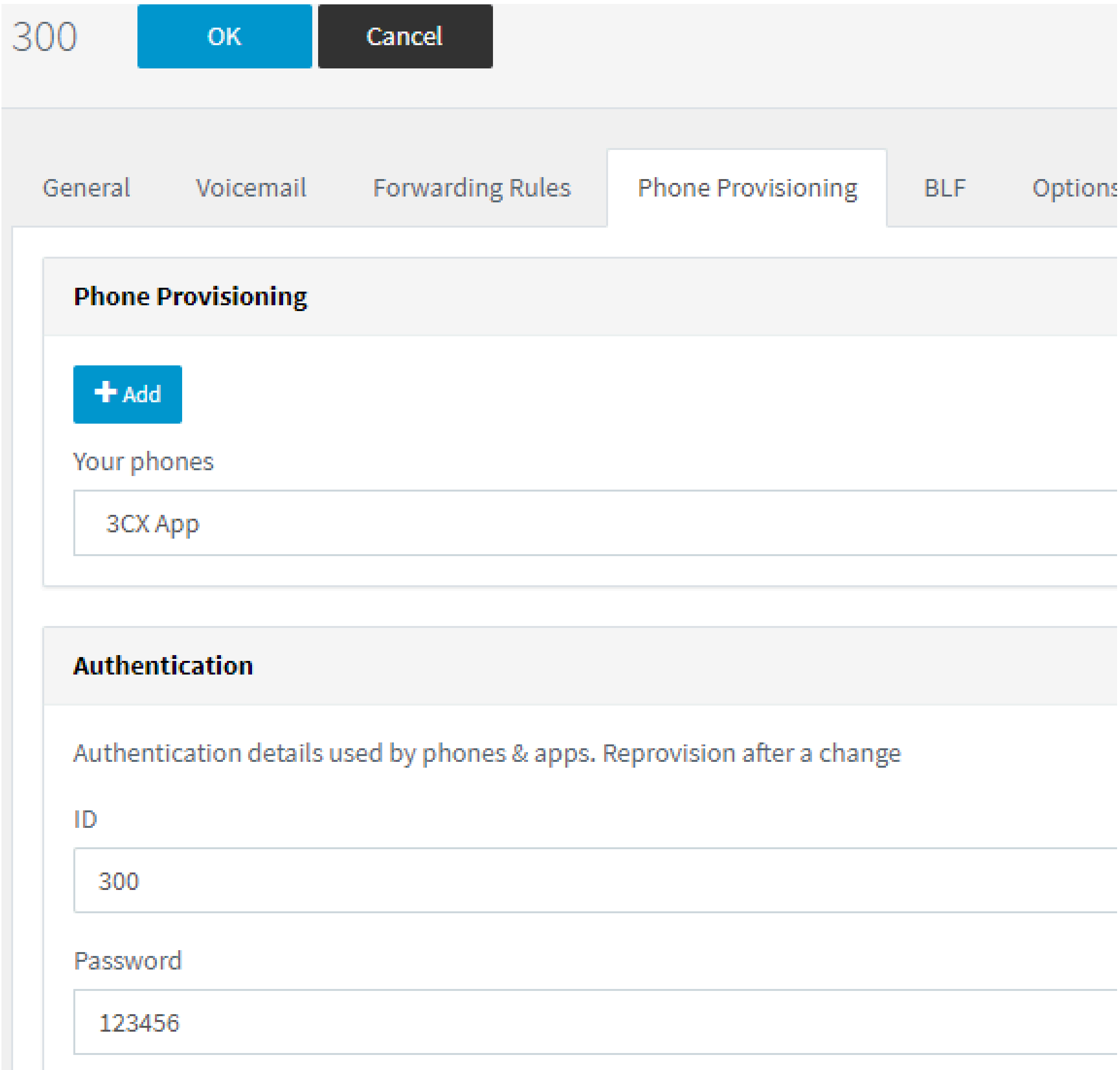


In the following steps, the parameters in the table are required configurations, and other parameters can be configured according to your own needs.

Step 1: Create a SIP extension (300) on the 3CX page

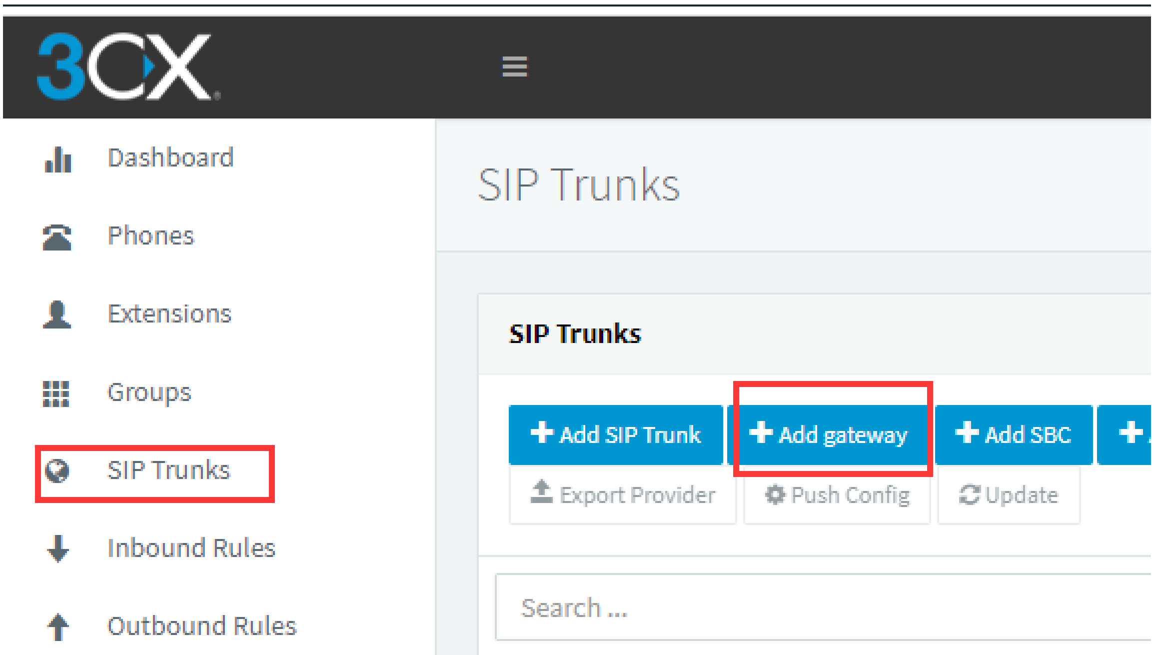
Click the Add button on the extension page, enter the extension number (300), and enter the ID and password for authentication on the phone configuration page.

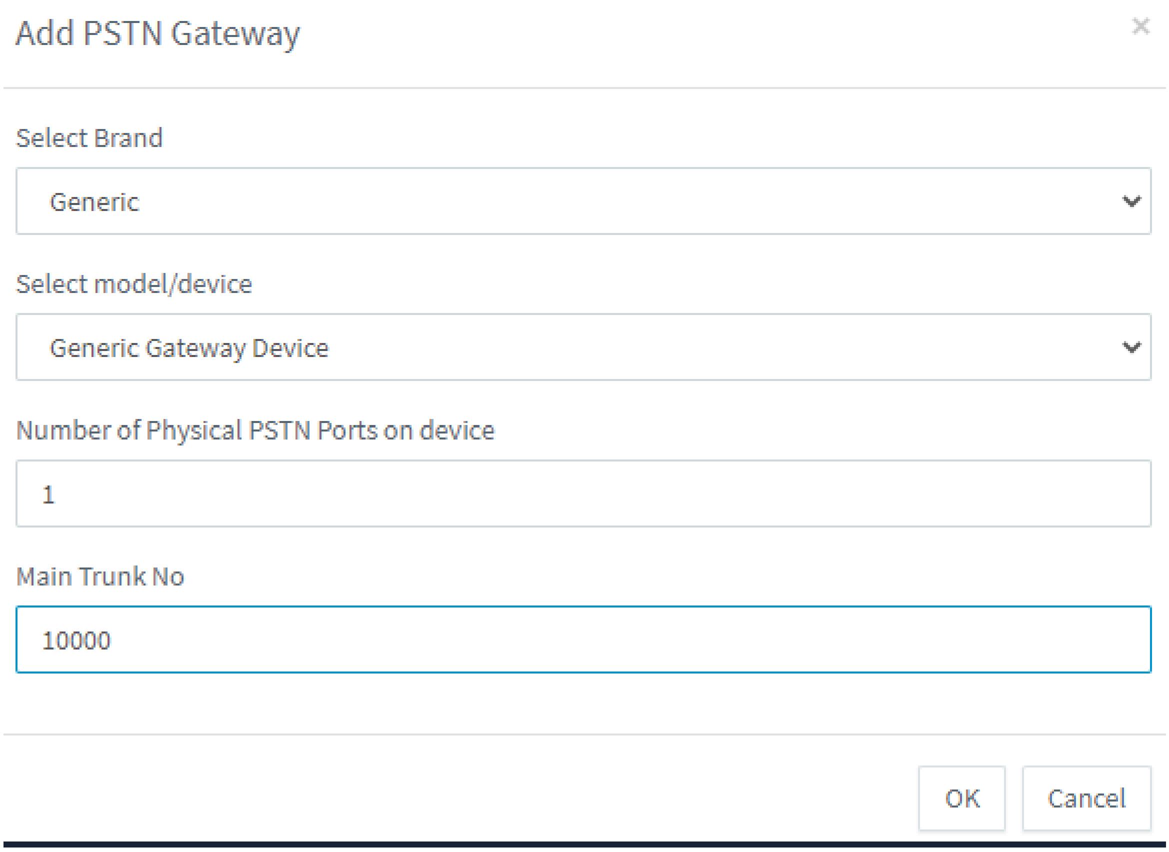
Parameter name	Parameter value
Extension display	300
Extension password	123456



Step 2: Create a SIP trunk on the 3CX page

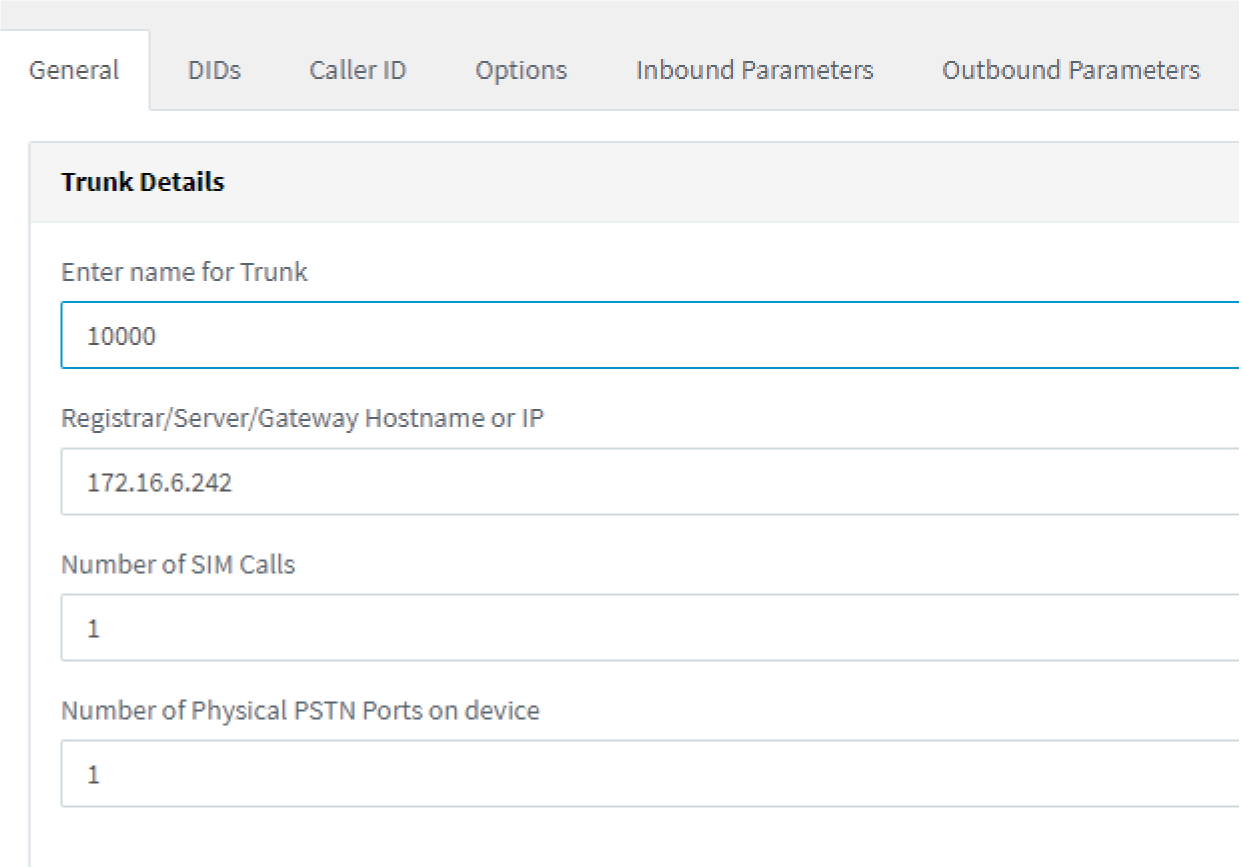
Please select "SIP Trunk—>Add Gateway" to create a SIP trunk:

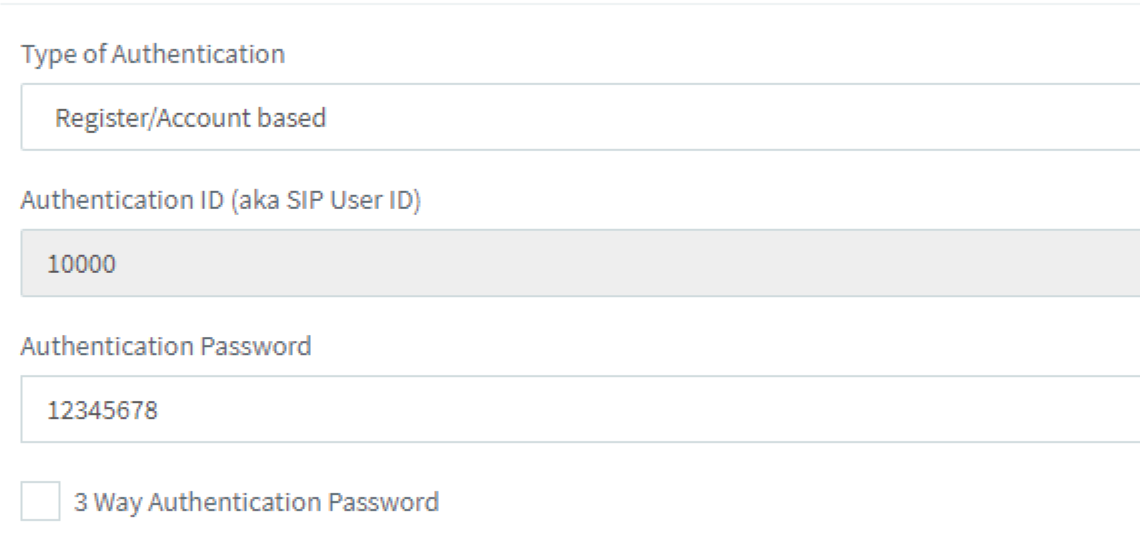




Then click the ok button

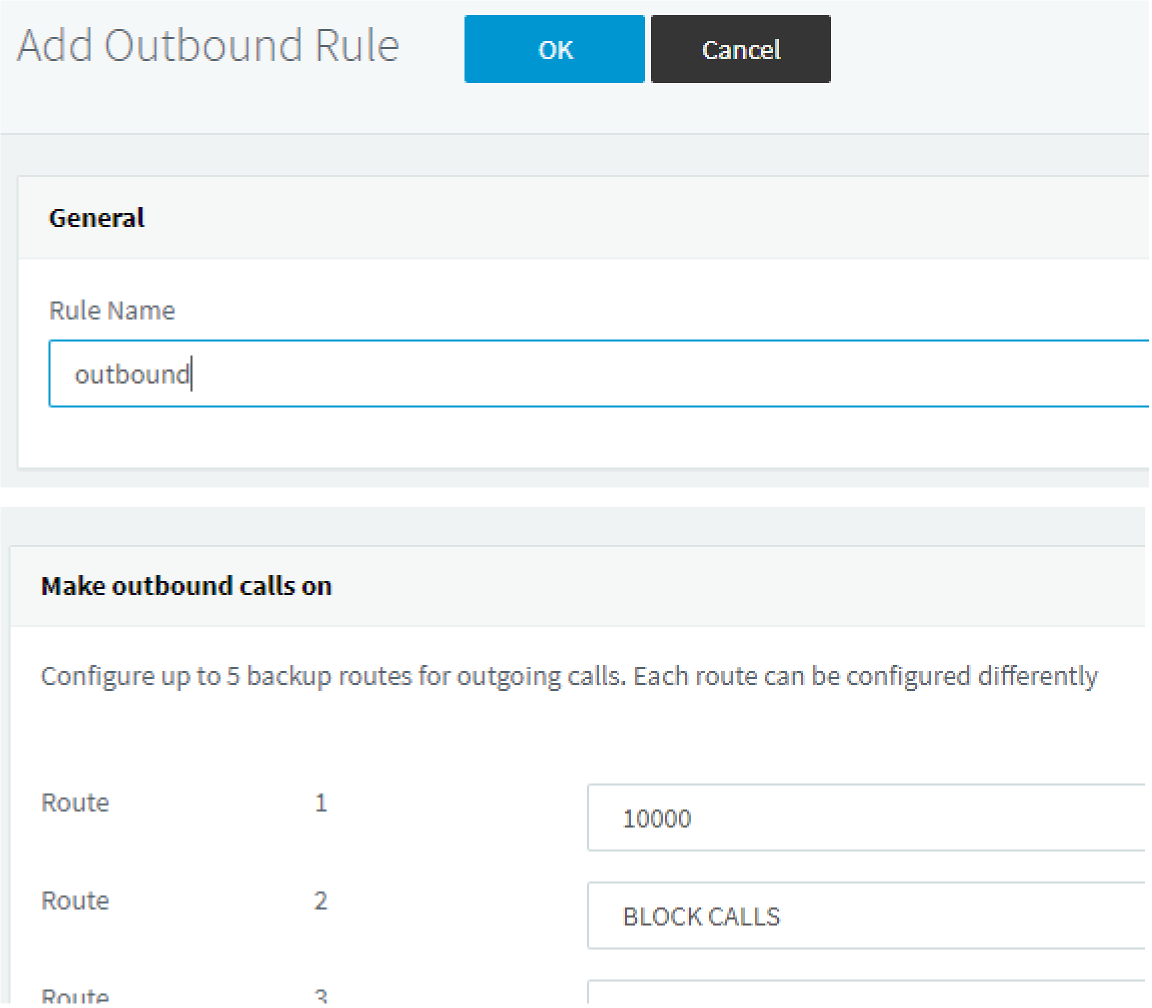
Parameter name	value
Trunk name	10000
Gateway IP address	172.16.8.242
Verification type	Based on registration/account
Verify ID	10000
Verify password	12345678





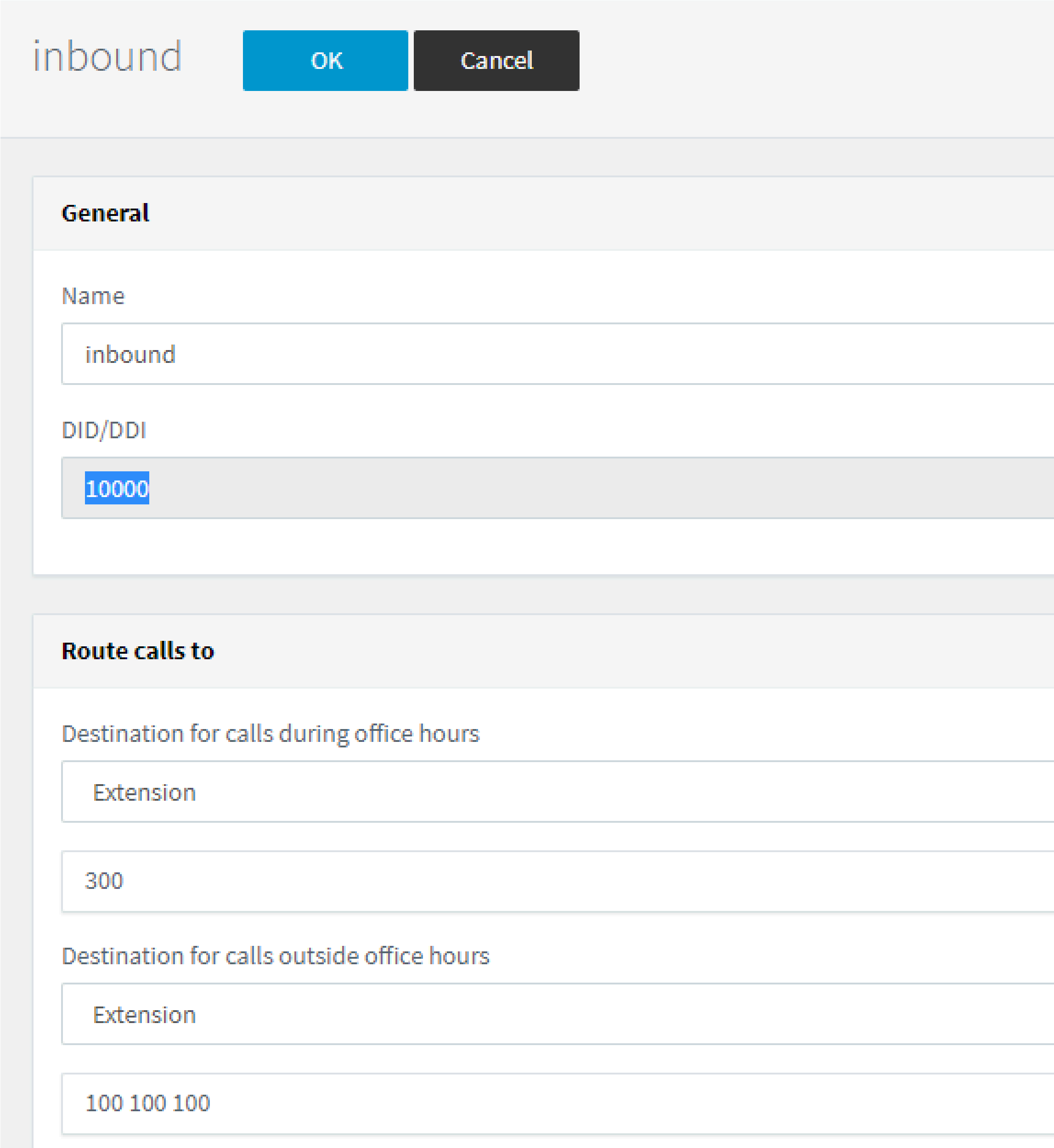
Step 3: Add incoming and outgoing routes in 3CX Outbound routing:

Parameter name	**Parameter value**
Rule name	outbound
routing	10000



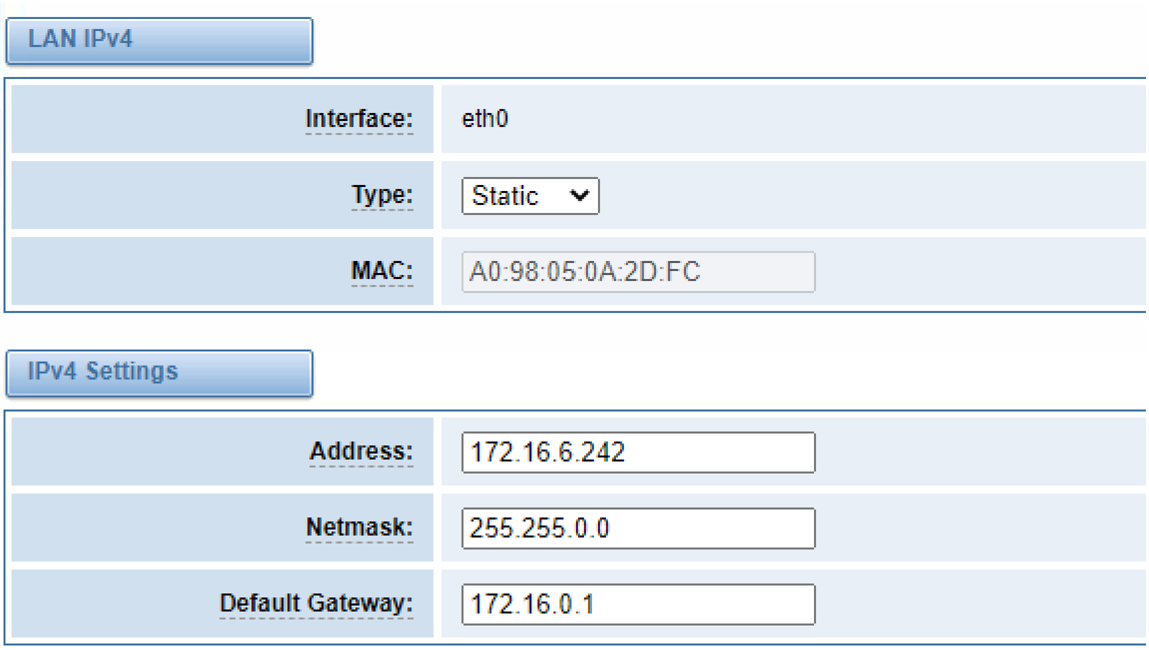
Incoming route:

Parameter name	Parameter value
Name	10000
Telephone assignment during working hours	Extension 300



Step 4: Set network parameters in the Web

Log in to the network in the browser and click "Network->LAN Settings" to set your network parameters. The picture below is an example for reference only.

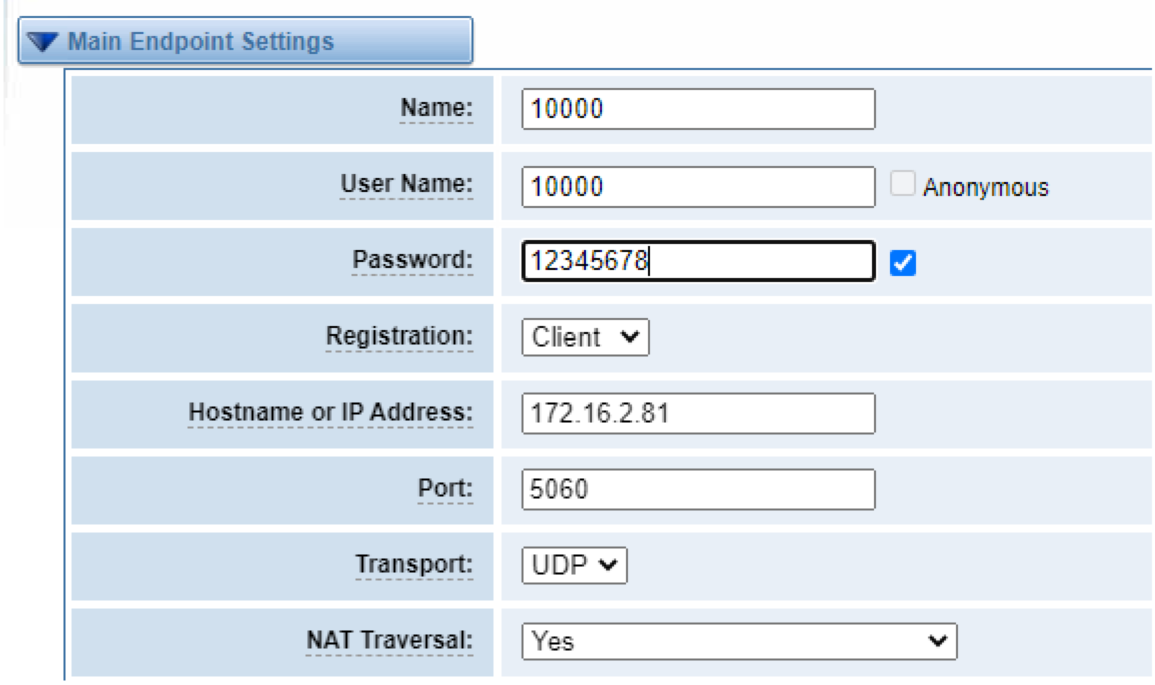


Step 5: Create SIP Endpoint in Web

Please select "VOIP->VOIP Settings->Add New SIP Endpoint" to set up SIP trunking. The image below shows details on how to set it up.

Parameter name	Parameter value
Name	10000
Username	10000
Password	12345678
Registration method	client
IP address	3CX IP: 172.16.2.81
port	5060

Regarding other parameters in SIP, please set them as per your requirement as there is no need to set them in a simple call.

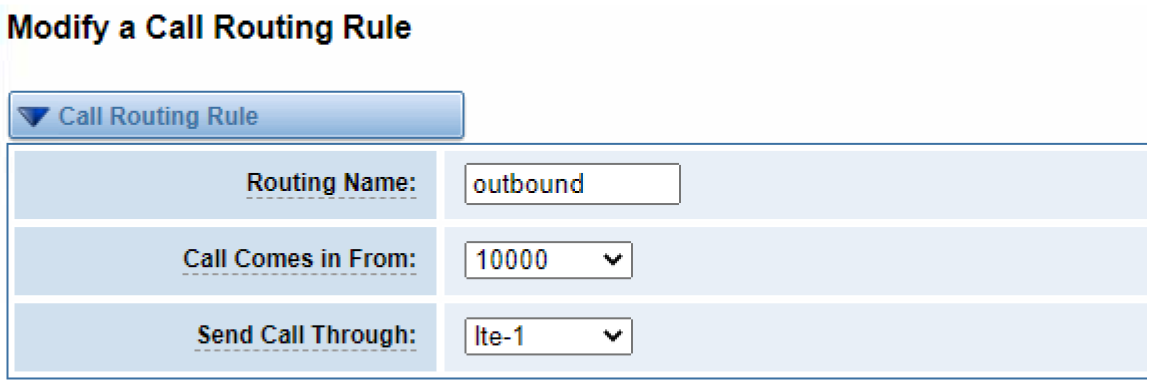


Step 6: Set up routing rules in the web

Click "Routing—>Call Routing Rules—>New Call Routing Rule" to set outbound and inbound routing rules, as shown below:

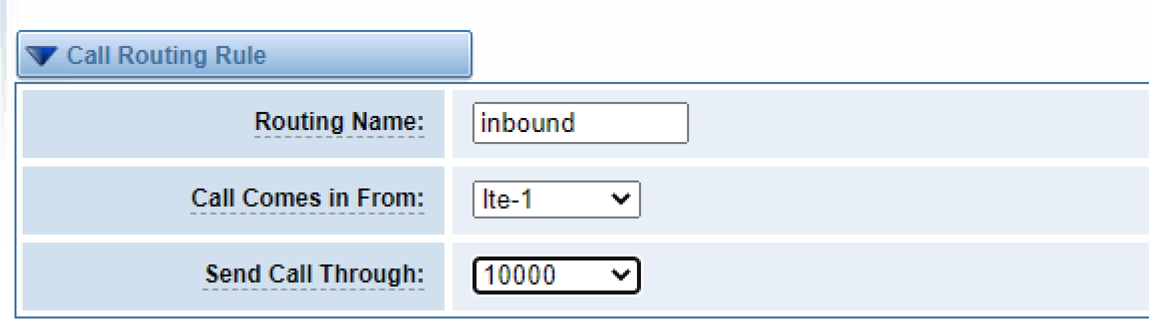
Outbound routing:

Parameter name	Parameter value
Route name	outbound
The call comes from	Trunk number: 10000
call delivery	lte-1



Inbound routing:

Parameter name	Parameter value
Route name	inbound
The call comes from	lte-1
call delivery	10000



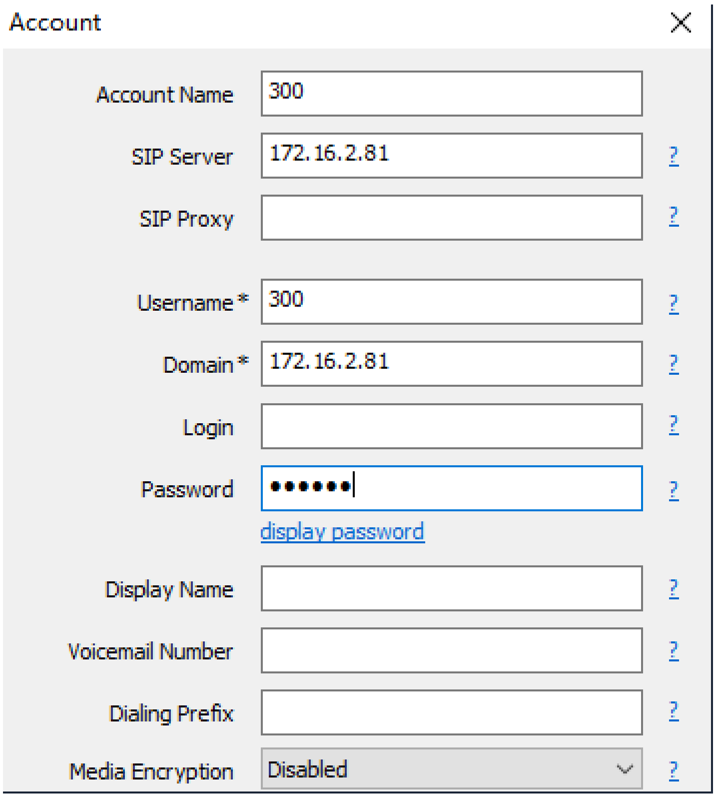
Step 7: Register extension (300) with softphone

SIP extensions (300) can be registered using SIP software such as Xlite, eyeBeam, microsip, etc. The picture below is an example of Xlite configuration 300.

Test call:

Incoming call test: Use your mobile phone to dial the number of port 1 on the gateway to see if 300 is ringing. Ringing indicates that the configuration is successful; if it is not ringing, please check the configuration.

Outbound test: Dial your mobile phone number on the registered 300 softphone. If your mobile phone rings, it means the configuration is successful; if it does not ring, it means the configuration failed. Please check the configuration.



6.2.1.2 Interconnecting WTU service board with Asterisk 16.15.1

This document mainly describes the detailed steps for interconnecting the business board with Asterisk.

Follow the following steps to configure a two-way call between the phone and the gateway:

- Outbound calls: from asterisk SIP extension 3002 to the gateway through trunk 1008;
- Incoming call: from the outside call to the gateway, via SIP trunk 1008 to asterisk, and then send the call to 3002 SIP extension through asterisk;

In the following steps, the parameters in the table are required configurations, and other parameters can be configured according to your own needs.

Step 1. Create SIP trunk in Asterisk server

Please add the following lines in sip.conf to create SIP trunk (1008) and SIP (3002):

```
[1008]
username=1008
secret=12345678
host=dynamic
port=5060
type=friend
context = from-gsm

[3002]
username=3002
secret=3002
host=dynamic
port=5060
type=friend
context = from-internal
```

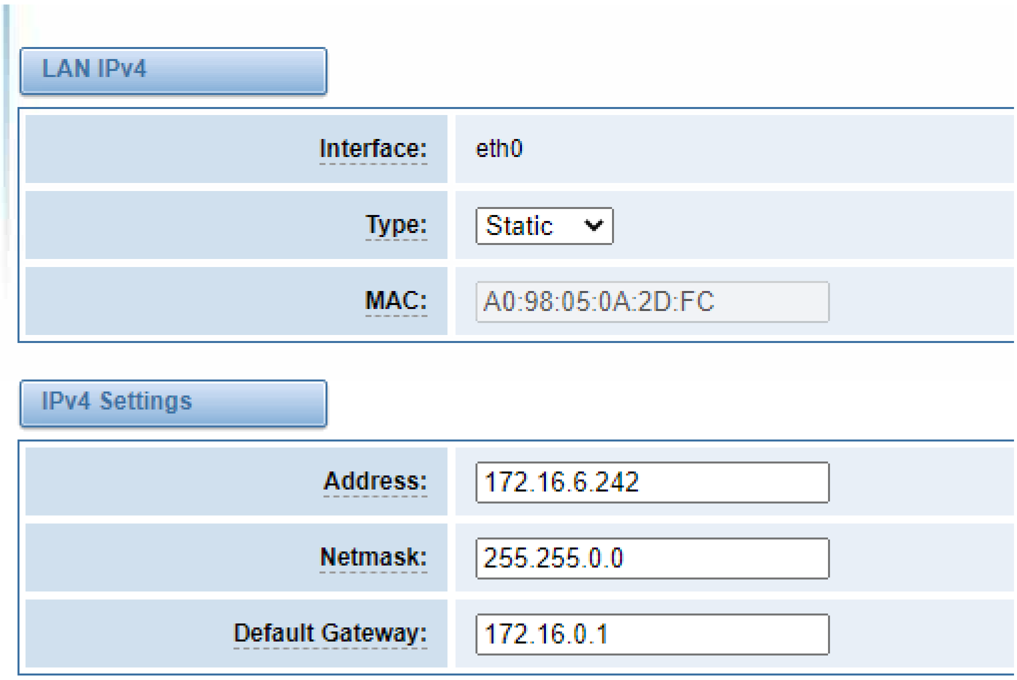
Step 2. Edit dialing rules in asterisk server

```
[from-internal]
exten => _x.,1,Dial(sip/1008/${EXTEN})
exten => _x.,n,hangup()

[from-gsm]
exten =>_x.,1,Dial(sip/3002)
exten =>_x.,n,Hangup()
```

Step 3. Set network parameters in Web

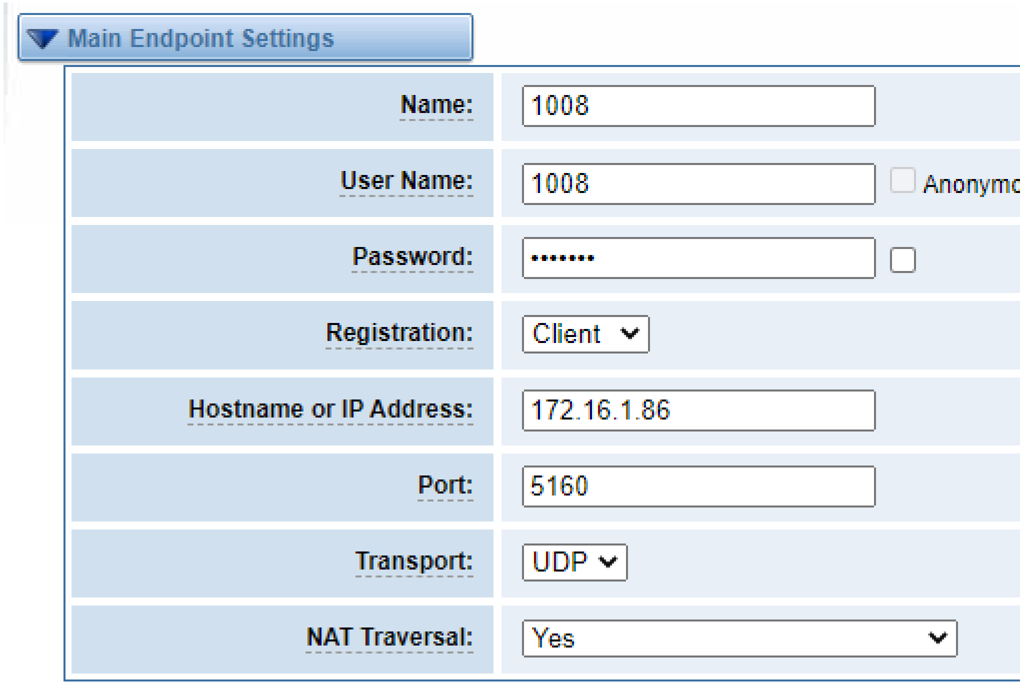
Log in to the network in the browser and click "Network->LAN Settings" to set your network parameters. The picture below is an example for reference only.



Step 4. Create SIP Endpoint in Web

Please select "VOIP->VOIP Settings->Add New SIP Endpoint" to set up SIP trunking. The image below shows details on how to set it up.

Parameter name	value
Trunk name	1008
Relay user name	1008
Relay password	12345678
Registration method	client
IP address	Server IP address (Asterisk IP address): 172.16.1.51
port	The port number used by chan_sip in Asterisk, the default is 5160
Transmission method	DefaultUDP
NAT penetration	The default is

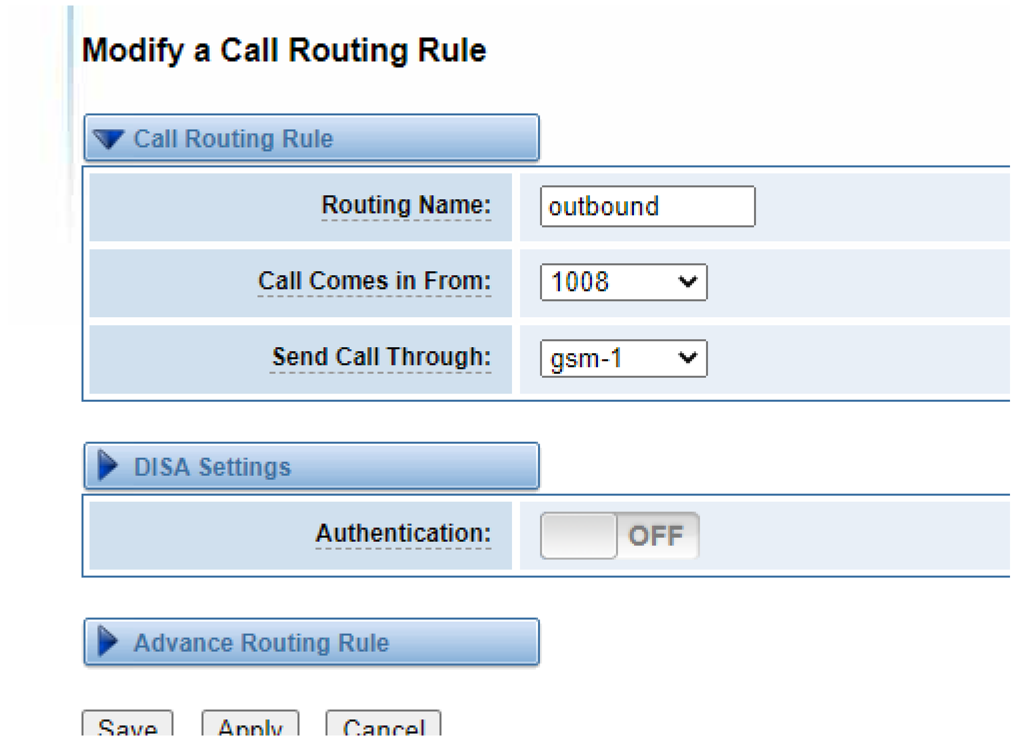


Step 5. Set Routing Rules in Web

Click "Routing—>Call Routing Rules—>New Call Routing Rule" to set outbound and inbound routing rules, as shown below:

Outbound routing:

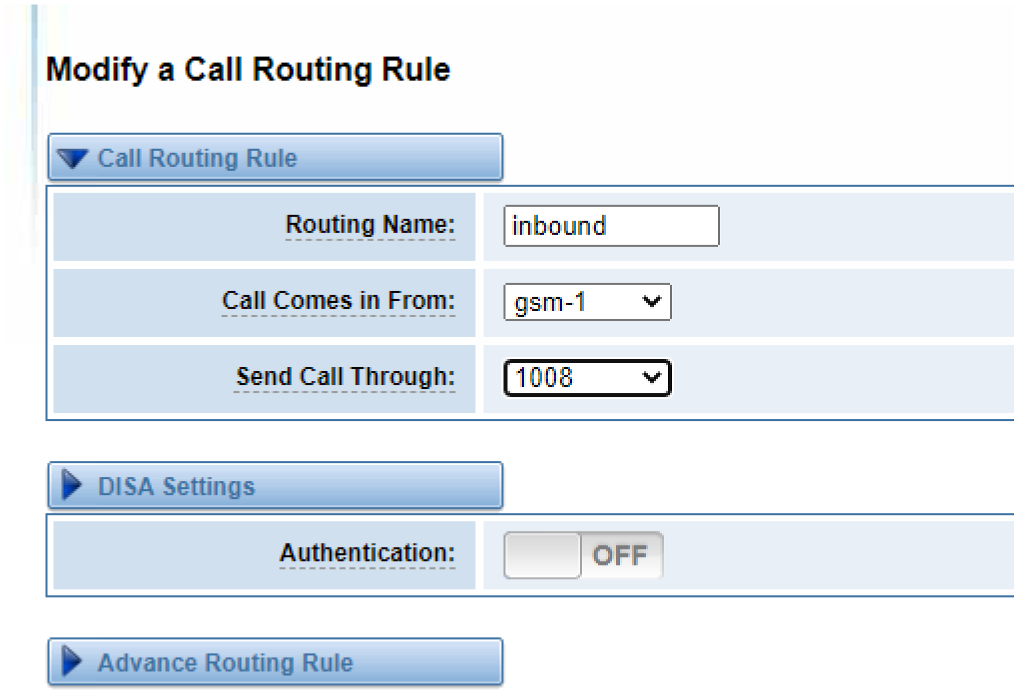
Parameter name	value
Route name	outbound
call from	Relay 1008
call delivery	Lte-1



Inbound routing:

Parameter name	value
Route name	inbound
call from	Lte-1
call delivery	Relay 1008

Please save and apply all settings



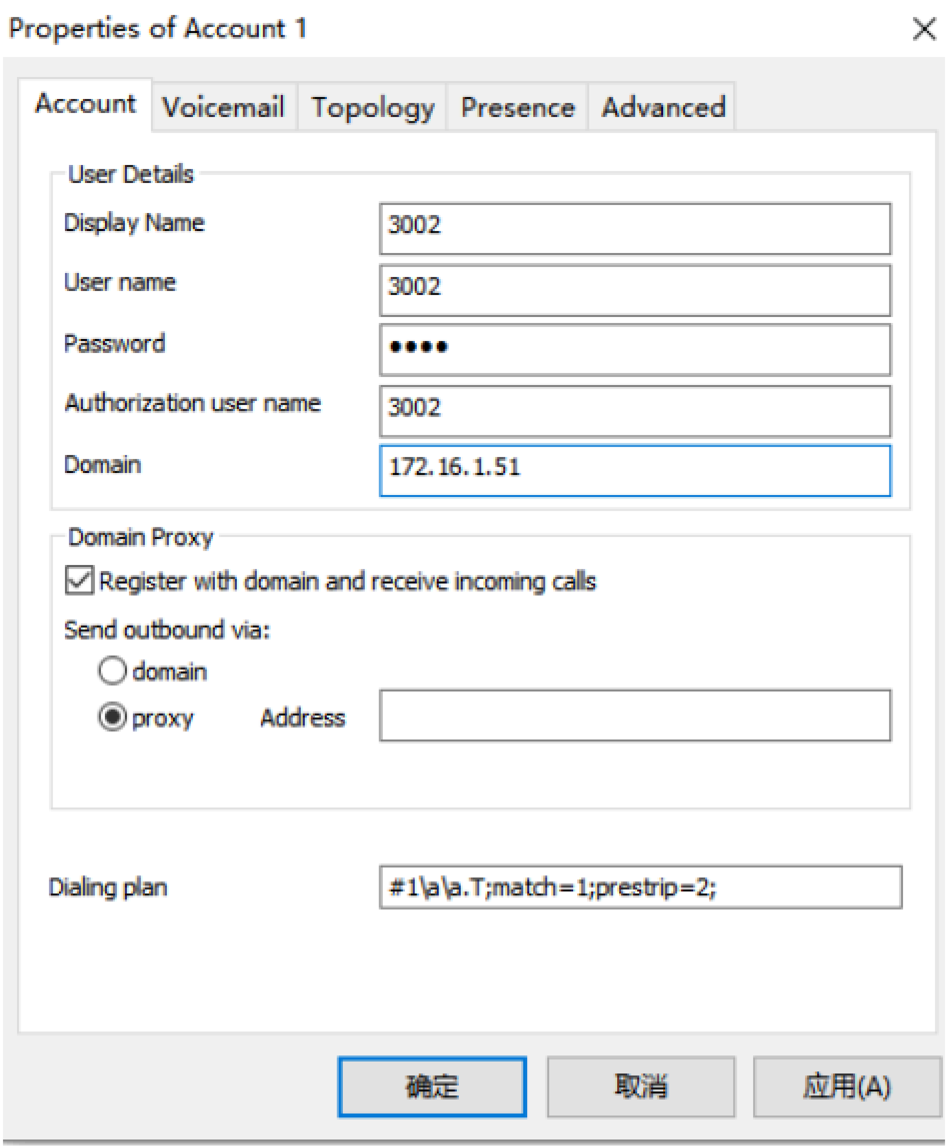
Step 6. Register SIP via software

Use Xlite, eyeBeam and other SIP software to register SIP (3002). After completing all the above steps, you can try making calls and sending text messages.

Test call:

Incoming call test: Use your mobile phone to dial the number of port 1 on the gateway and see if 3002 rings. Ringing indicates that the configuration is successful; if it does not ring, please check the configuration.

Outbound test: Dial your mobile phone number on the softphone registered with 3002. If your mobile phone rings, it means the configuration is successful; if it does not ring, it means the configuration failed. Please check the configuration.

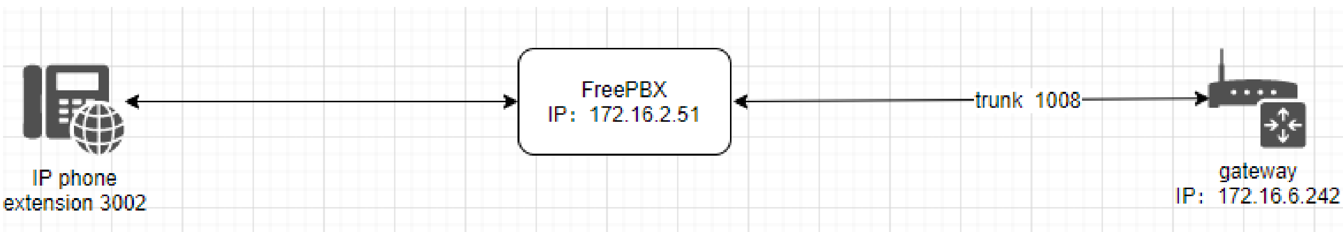


6.2.1.3 WTU service board interconnection with FreePBX 15

This document mainly describes the detailed steps for connecting the business board to FreePBX.

Follow the following steps to configure a two-way call between the phone and the gateway:

- Outbound calls: from FreePBX SIP extension 3002 to the gateway through trunk 1008;
- Incoming calls: Call from the outside to the gateway, go to FreePBX via SIP trunk 1008, and then send the call to SIP extension 3002 through FreePBX;

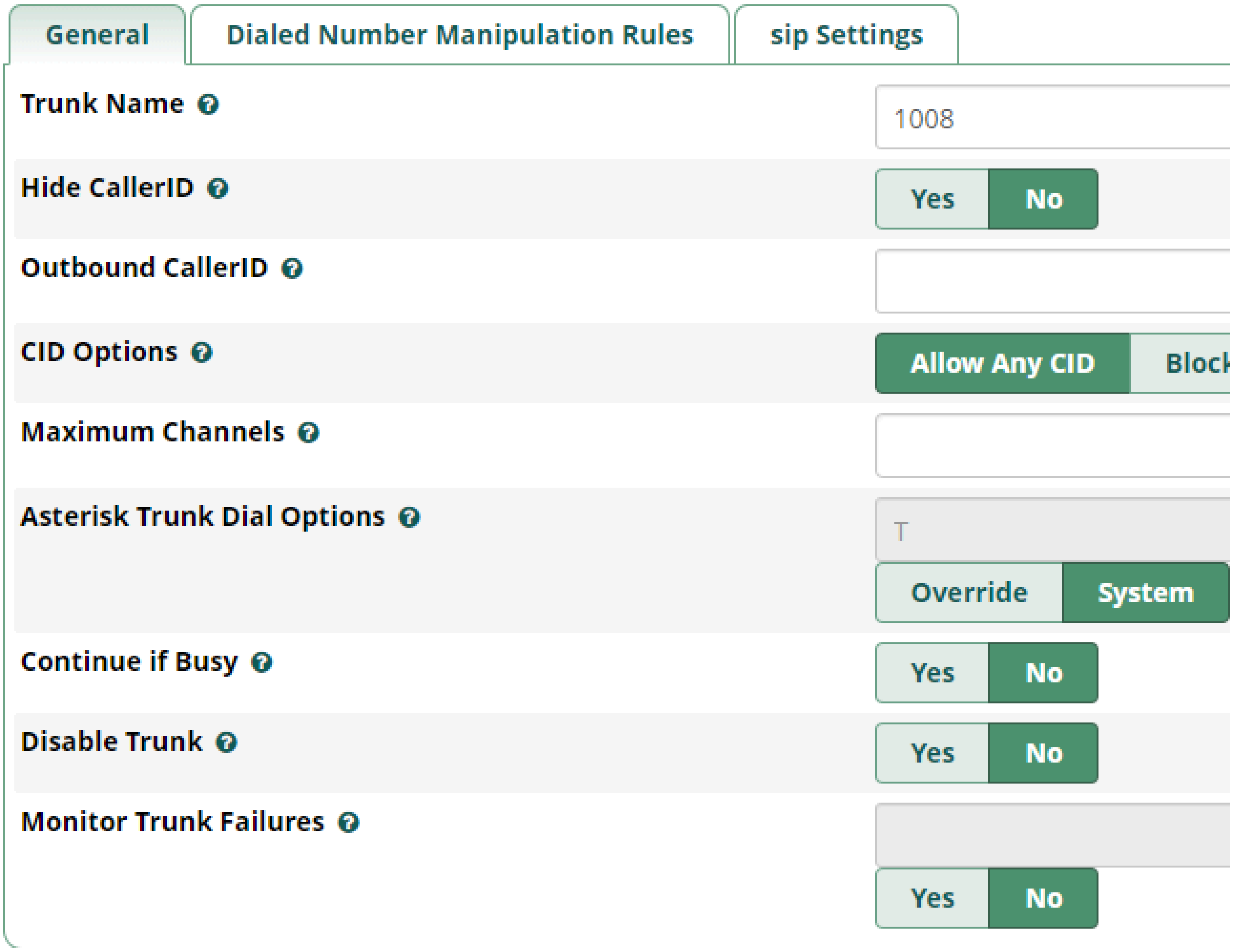


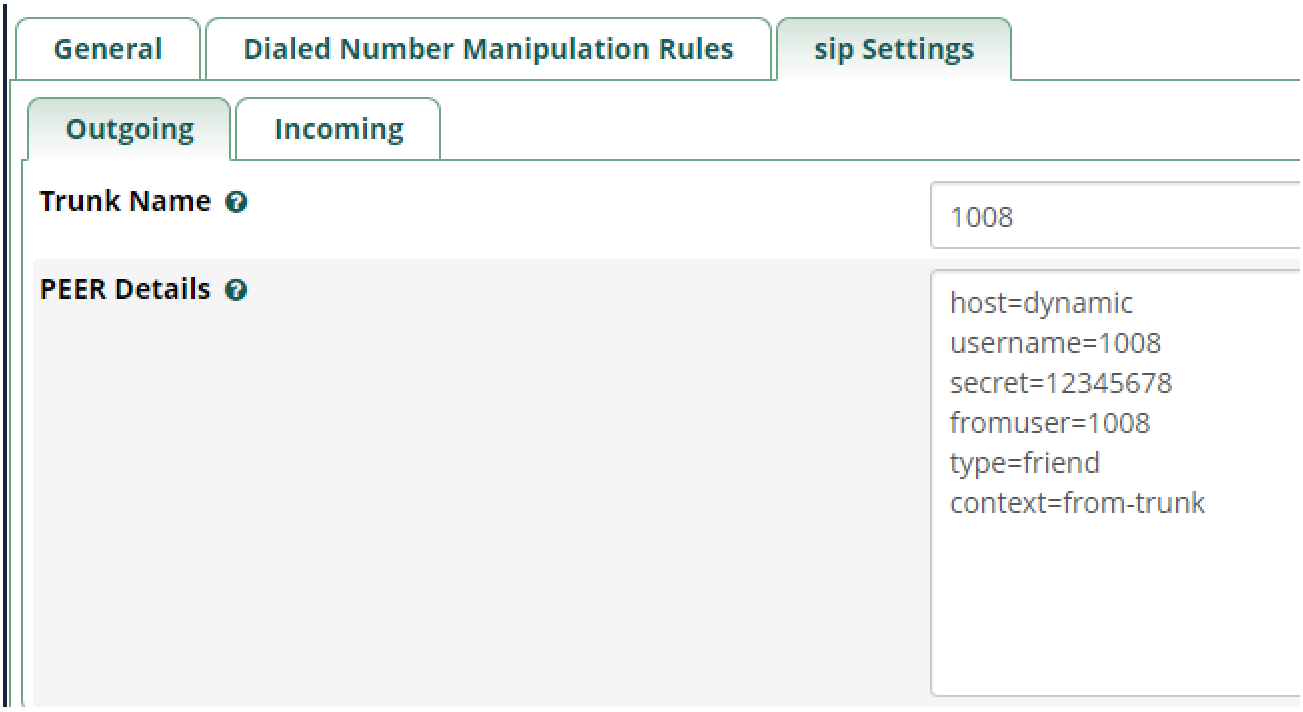
In the following steps, the parameters in the table are required configurations, and other parameters can be configured according to your own needs.

Step 1: Create sip trunk in Freepbx

Please log in to your freepbx server to create a SIP trunk (1008), click the Add Trunk button on the Communication Interface Connection--->Relay page, and perform the following configuration:

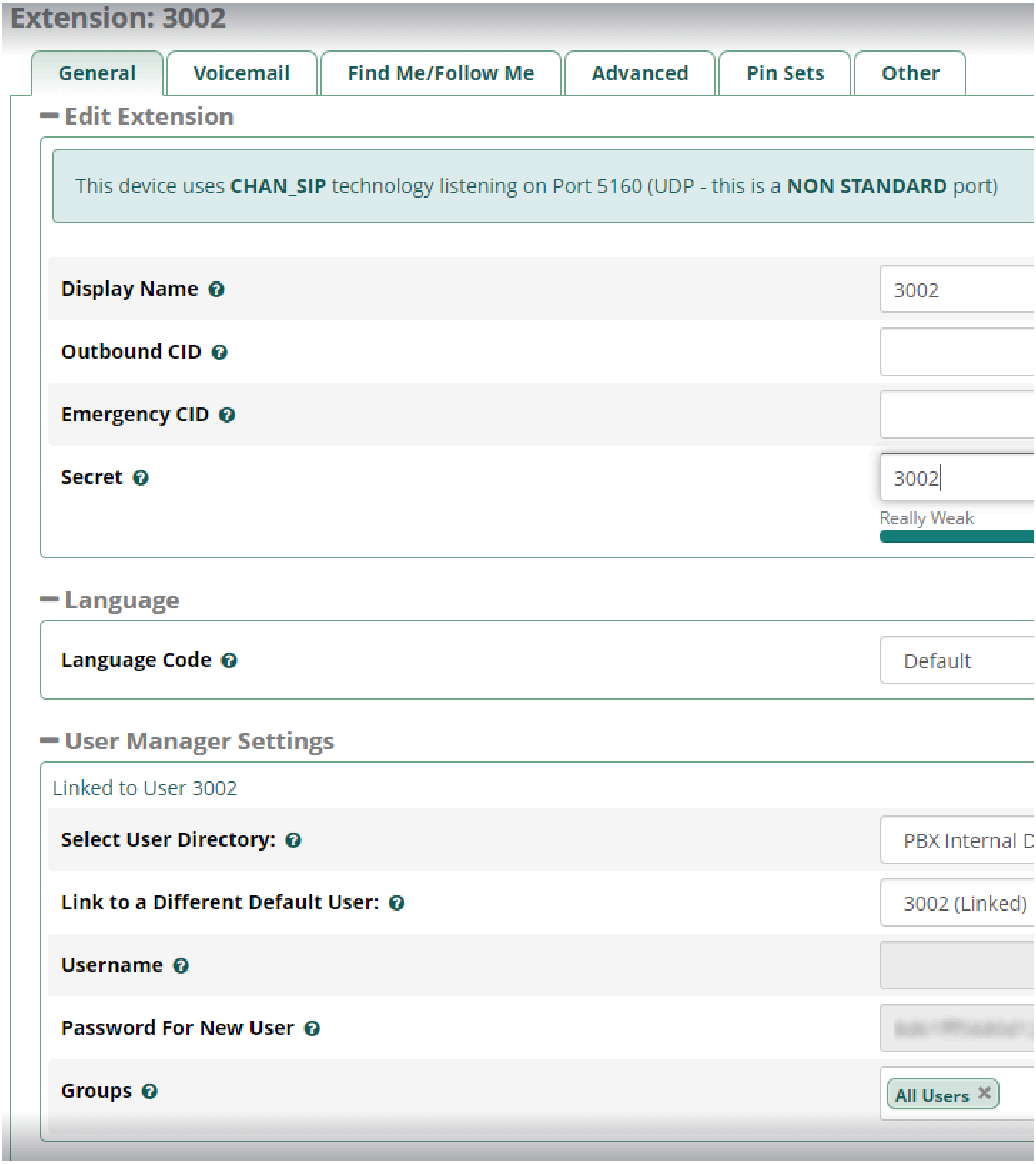
Parameter name	Parameter value
Trunk name	1008
host	dynamic
username	1008
secret	12345678
fromuser	1008
context	from-trunk





Step 2: Create sip extension (3002) on Freepbx

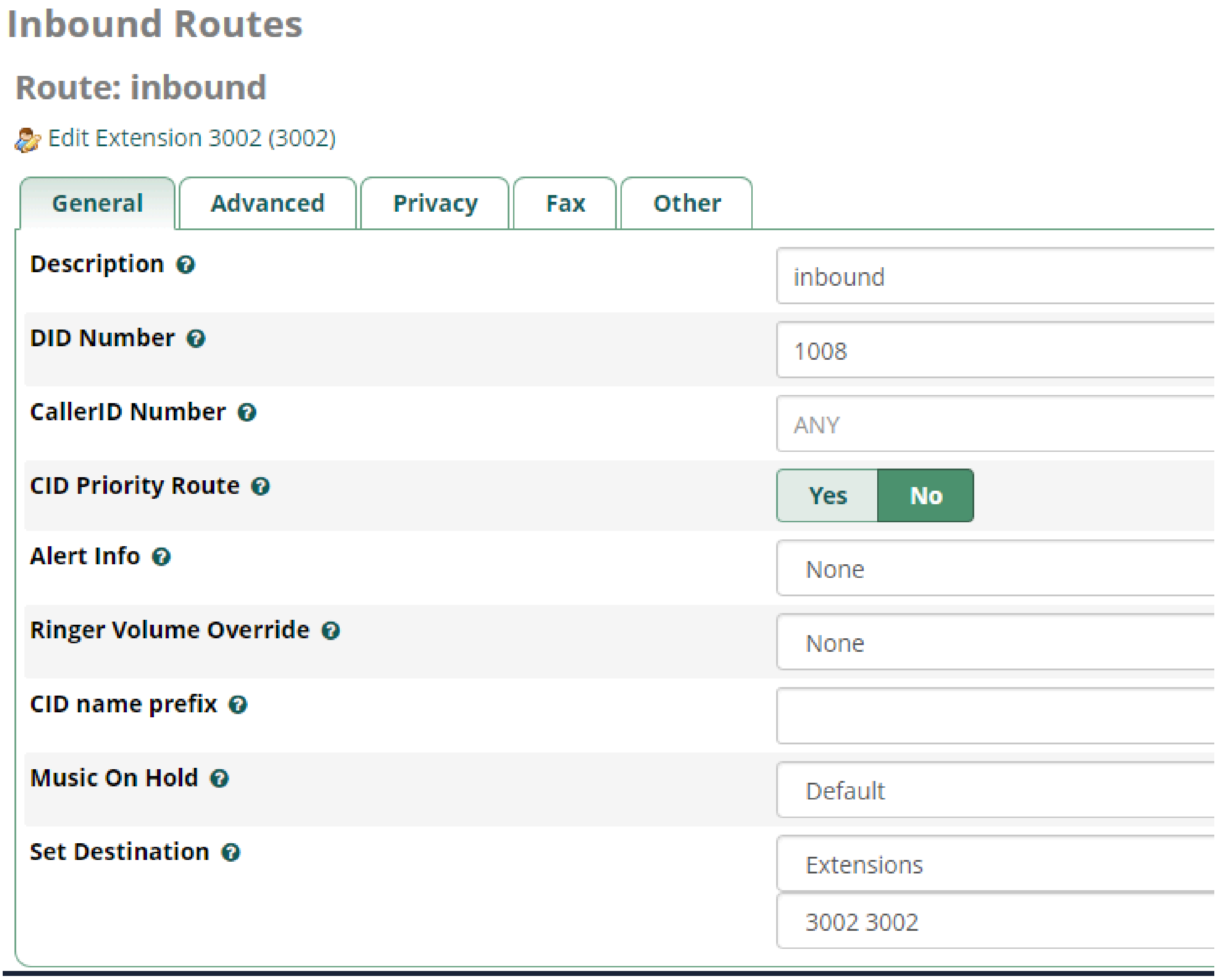
Parameter name	Parameter value
Extension display name	3002
Extension password	3002



Step 3: Add incoming and outgoing routes in Freepbx

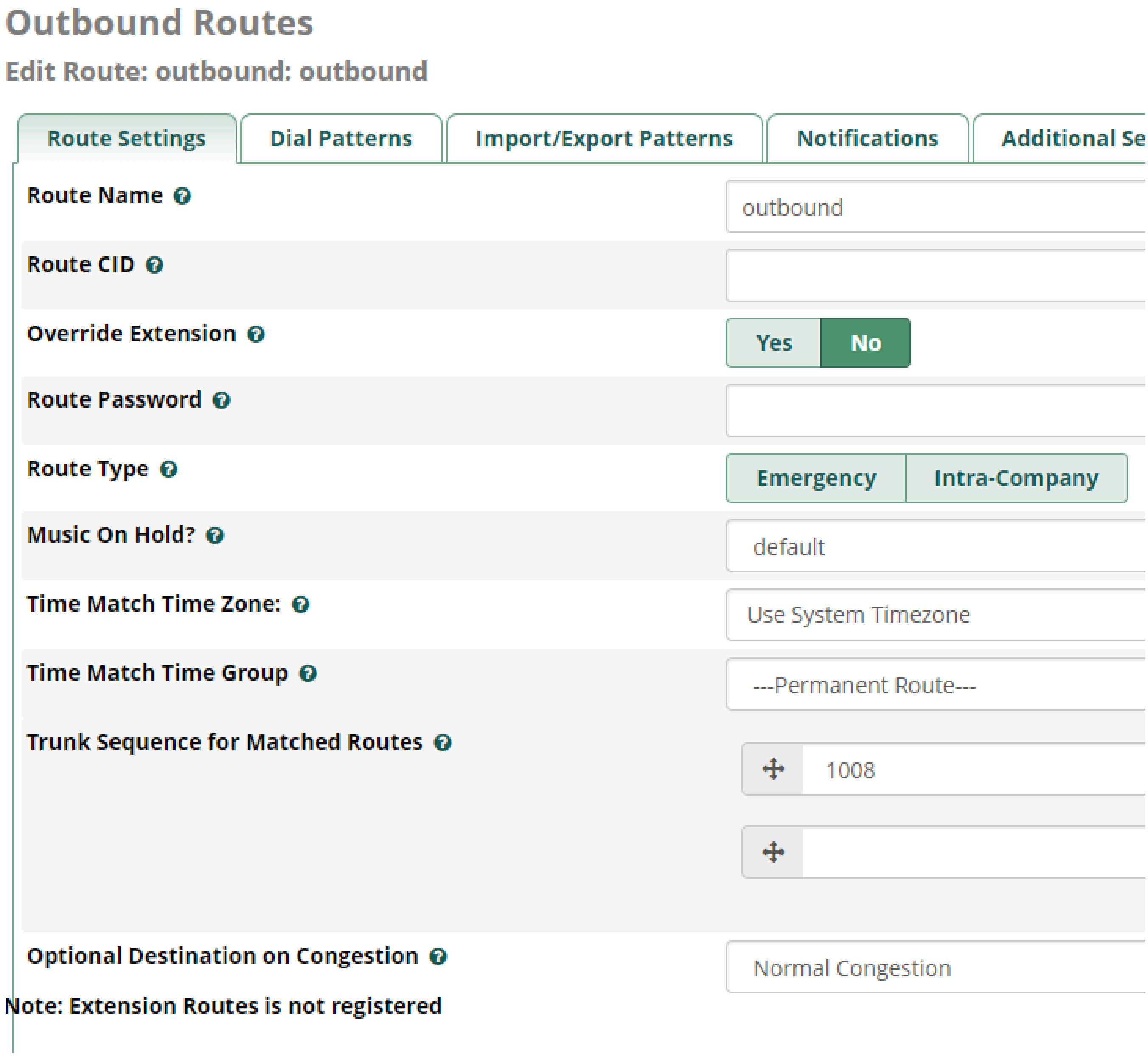
Incoming route:

Parameter name	Parameter value
Inbound route description	inbound
DID number	1008
Incoming call destination	Extension 3002



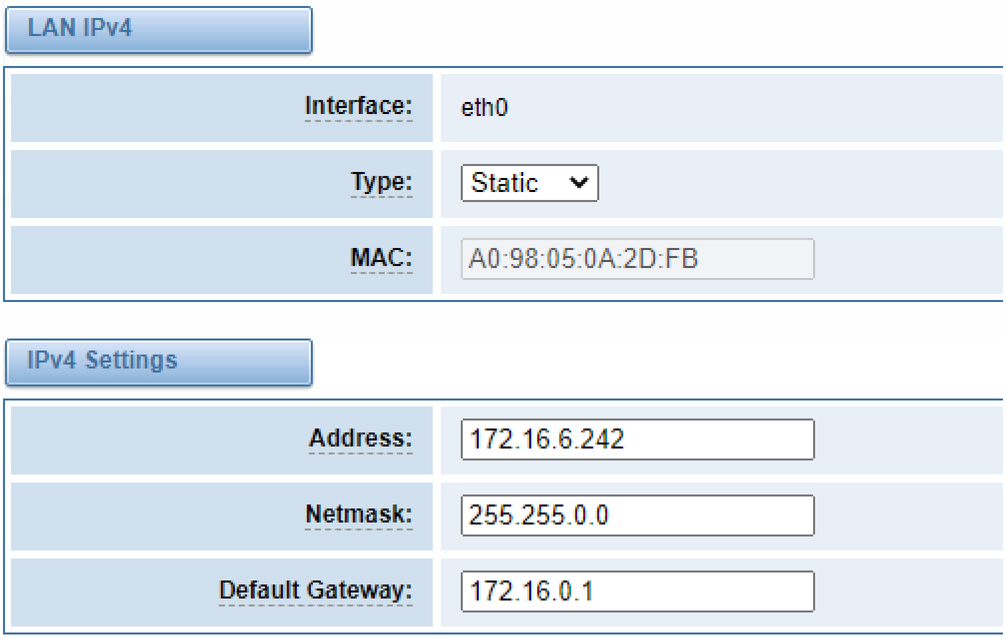
Outbound routing:

Parameter name	value
Outbound route name	outbound
Match relay order	1008



Step 4: Set network parameters in the Web

Log in to the network in the browser and click "Network->LAN Settings" to set your network parameters. The picture below is an example for reference only.

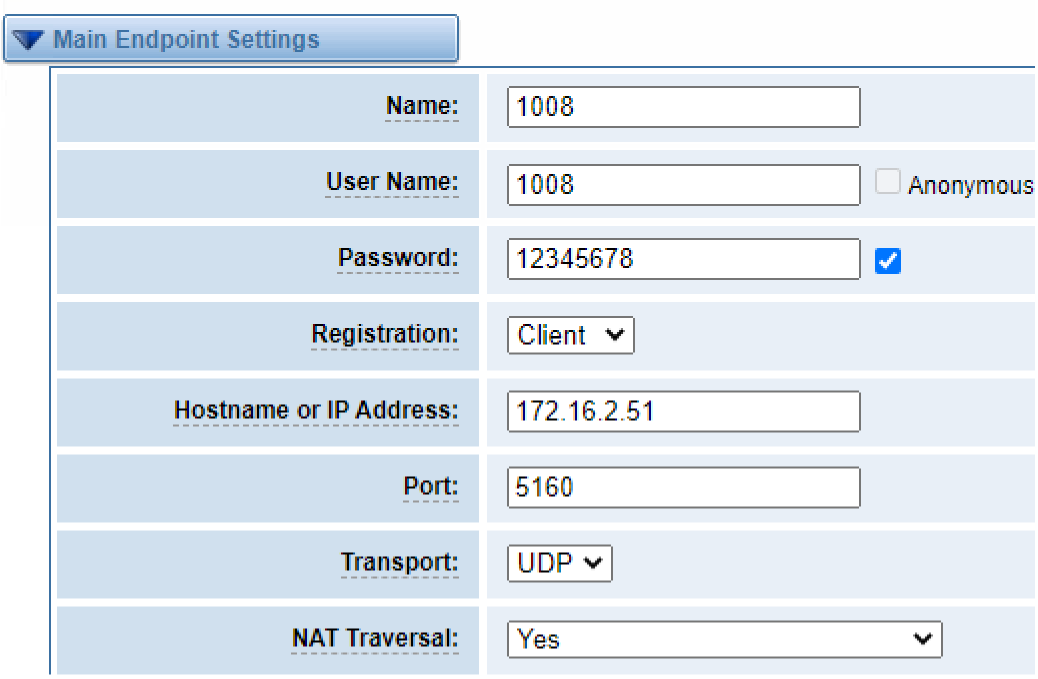


Step 5: Create SIP Endpoint in Web

Please select "VOIP->VOIP Settings->Add New SIP Endpoint" to set up SIP trunking. The image below shows details on how to set it up.

Regarding other parameters in SIP, please set them as per your requirement as there is no need to set them in a simple call.

Parameter name	value
Trunk name	1008
Relay user name	1008
Relay password	12345678
Registration method	client
IP address	Server's IP address (Freepbx's IP address): 172.16.1.51
port	The port number used by chan_sip in FreePBX, the default is 5160
Transmission method	DefaultUDP
NAT penetration	The default is

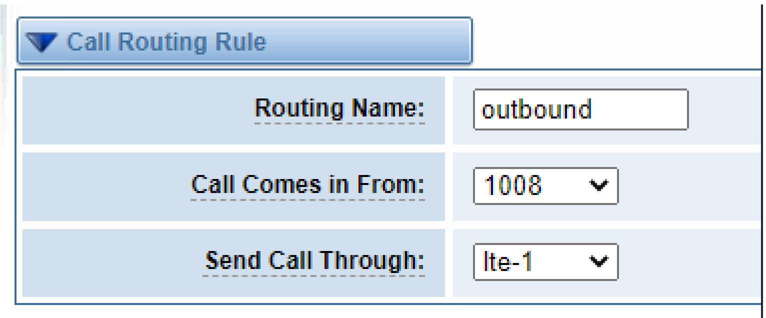


Step 6: Set up routing rules in the web

Click "Routing—>Call Routing Rules—>New Call Routing Rule" to set outbound and inbound routing rules, as shown below:

Outbound routing:

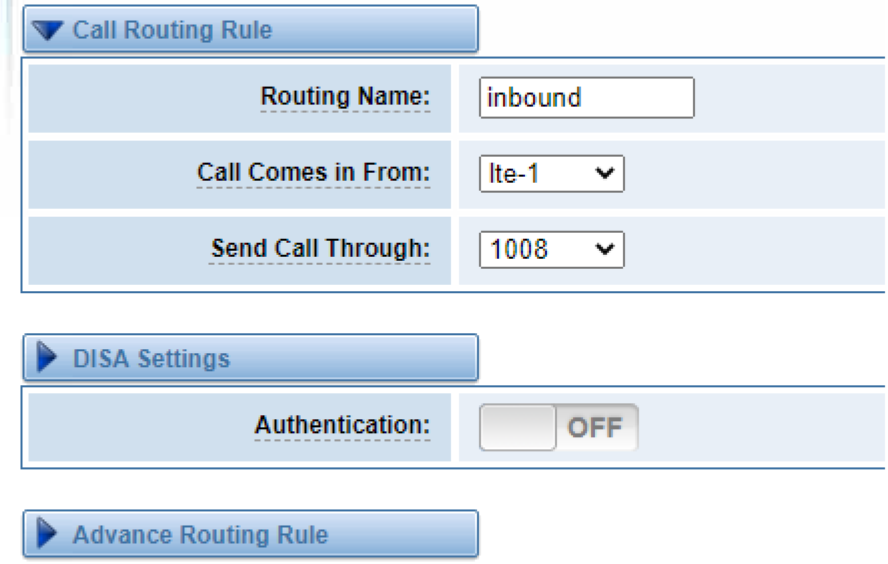
Parameter name	value
Route name	outbound
call from	Relay 1008
call delivery	Lte-1



Inbound routing:

Parameter name	value
Route name	inbound
call from	Lte-1
call delivery	Relay 1008

Please save and apply all settings.



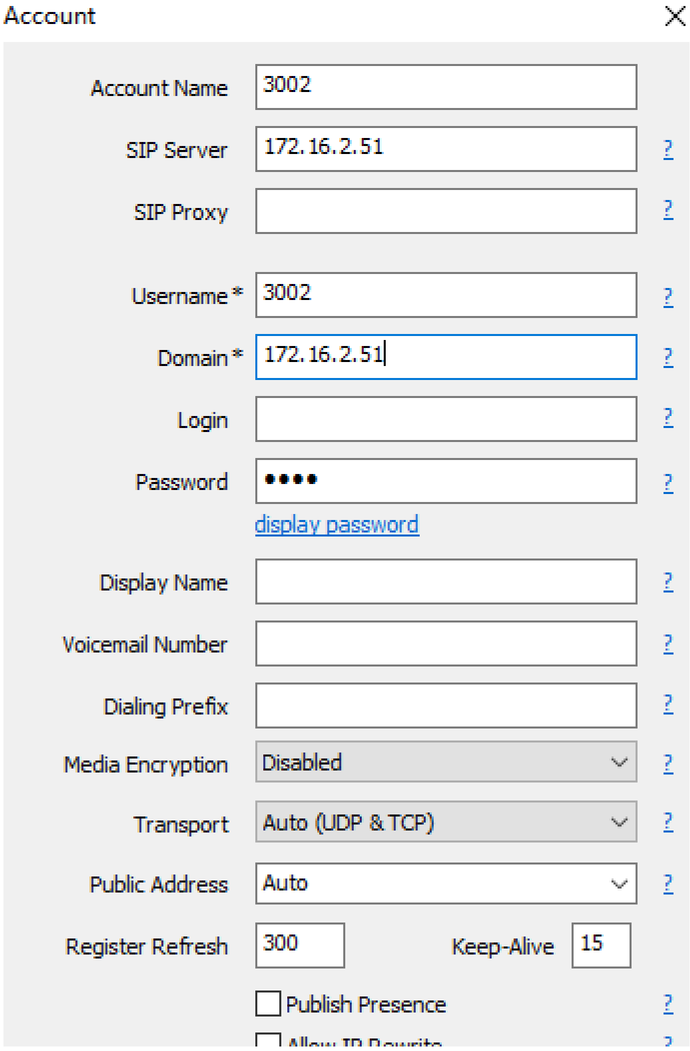
Step 7: Register extension (3002) with softphone

You can use Xlite, eyeBeam, microsip and other SIP software to register the SIP extension (3002). The picture below is an example of Xlite configuration 3002.

Test call:

Incoming call test: Use your mobile phone to dial the number of port 1 on the gateway and see if 3002 rings. Ringing indicates that the configuration is successful; if it does not ring, please check the configuration.

Outbound test: Dial your mobile phone number on the softphone registered with 3002. If your mobile phone rings, it means the configuration is successful; if it does not ring, it means the configuration failed. Please check the configuration.

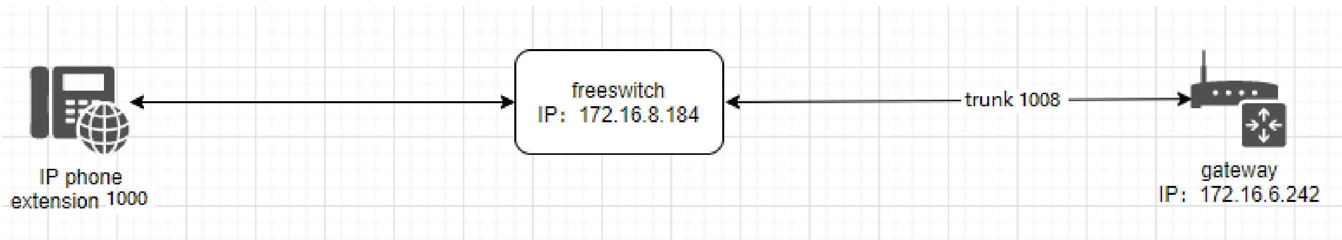


6.2.1.4 WTU service board interconnected with FreeSwitch 1.10.5

This document mainly describes the detailed steps for interconnecting the service board with FreeSwitch.

Follow the following steps to configure a two-way call between the phone and the gateway:

- Outbound calls: from FreeSwitch SIP extension 1000 to the gateway through trunk 1008;
- Incoming call: from the outside call to the gateway, via SIP 1020 to FreeSwitch, and then send the call to 1000 SIP extension through FreeSwitch;



In the following steps, the parameters in the table are required configurations, and other parameters can be configured according to your own needs.

Step 1: Create sip trunk in FreeSwitch

Enter the default configuration directory of FreeSWITCH, and then add the gateway configuration in /etc/freeswitch/directory/default/1008.xml: vi /etc/freeswitch/directory/default/1008.xml

```
<include>
  <user id="1008">
    <params>
      <param name="password" value="1008"/>
      <param name="vm-password" value="1008"/>
    </params>
    <variables>
      <variable name="toll_allow" value="domestic,international,local"/>
      <variable name="accountcode" value="1008"/>
      <variable name="user_context" value="default"/>
      <variable name="effective_caller_id_name" value="Extension 1008"/>
      <variable name="effective_caller_id_number" value="1008"/>
      <variable name="outbound_caller_id_name" value="$${outbound_caller_name}"/>
      <variable name="outbound_caller_id_number" value="$${outbound_caller_id}"/>
      <variable name="callgroup" value="techsupport"/>
    </variables>
  </user>
</include>
```

Parameter name	Parameter value
user id	1008
password	1008
effective_caller_id_name	Extension 1008
effective_caller_id_number	1008

Note: The above configuration is the default configuration and does not need to be modified.

Step 2: Create sip extension (1000) on FreeSwitch

```
<include>
  <user id="1000">
    <params>
      <param name="password" value="1000"/>
      <param name="vm-password" value="1000"/>
    </params>
    <variables>
      <variable name="toll_allow" value="domestic,international,local"/>
      <variable name="accountcode" value="1000"/>
      <variable name="user_context" value="default"/>
      <variable name="effective_caller_id_name" value="Extension 1000"/>
      <variable name="effective_caller_id_number" value="1000"/>
      <variable name="outbound_caller_id_name" value="$${outbound_caller_name}"/>
      <variable name="outbound_caller_id_number" value="$${outbound_caller_id}"/>
      <variable name="callgroup" value="techsupport"/>
    </variables>
  </user>
</include>
```

Parameter name	Parameter value
user id	1000
password	1000

Note: The above configuration is the default configuration and does not need to be modified.

Step 3: Add incoming and outgoing routes in FreeSwitch

Incoming route:

Edit the inbound routing file /etc/freeswitch/dialplan/public/00_inbound_did.xml

```
<include>
  <extension name="public_did">
    <condition field="destination_number" expression="1020">
      <!--
          If you're hosting multiple domains you will want to set the
          target_domain on these calls so they hit the proper domain after you
          transfer the caller into the default context.
          $${domain} is the default domain set from vars.xml but you can set it
          to any domain you have setup in your user directory.
      -->
      <action application="set" data="domain_name=$${domain}"/>
      <!-- This example maps the DID 5551212 to ring 1000 in the default context -->
      <action application="transfer" data="1000 XML default"/>
    </condition>
  </extension>
</include>
```

Parameter name	Parameter value
destination_number	1020
data	1000 XML default

Outbound routing:

Edit the outbound routing file /etc/freeswitch/dialplan/default.xml

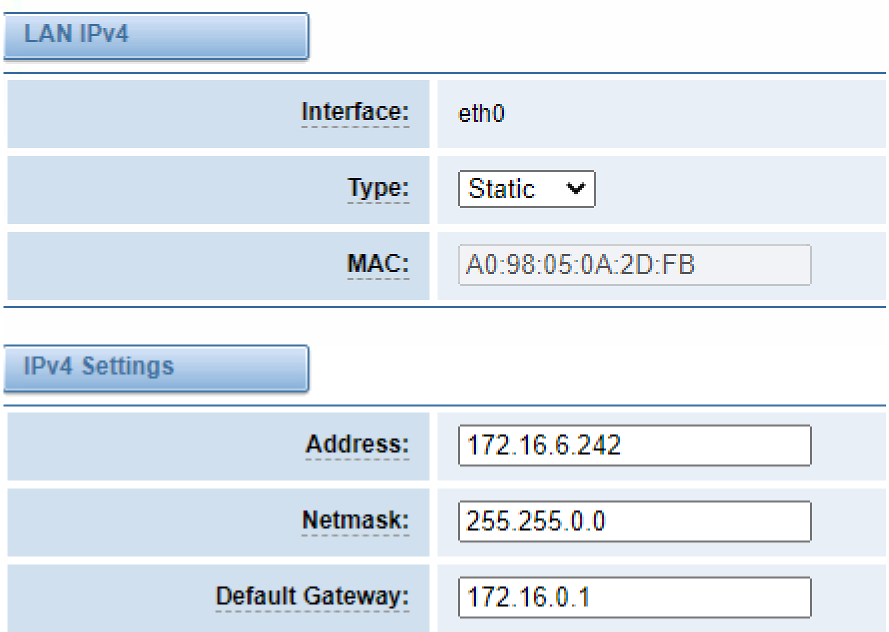
```
<extension name="outbound">
  <condition field="destination_number" expression="^9(\\d+)$">
    <action application="answer"/>
    <action application="set" data="ringback=${us-ring}"/>
    <action application="bridge" data="sofia/internal/$1@172.16.6.242"/>
  </condition>
</extension>
```

Parameter name	value
Outbound route name	outbound

Parameter name	value
expression	^9(\d+)$, numbers starting with 9 will match this route
data	sofia/internal/$1@172.16.6.242

Step 4: Set network parameters in the Web

Log in to the network in the browser and click "Network->LAN Settings" to set your network parameters. The picture below is an example for reference only.

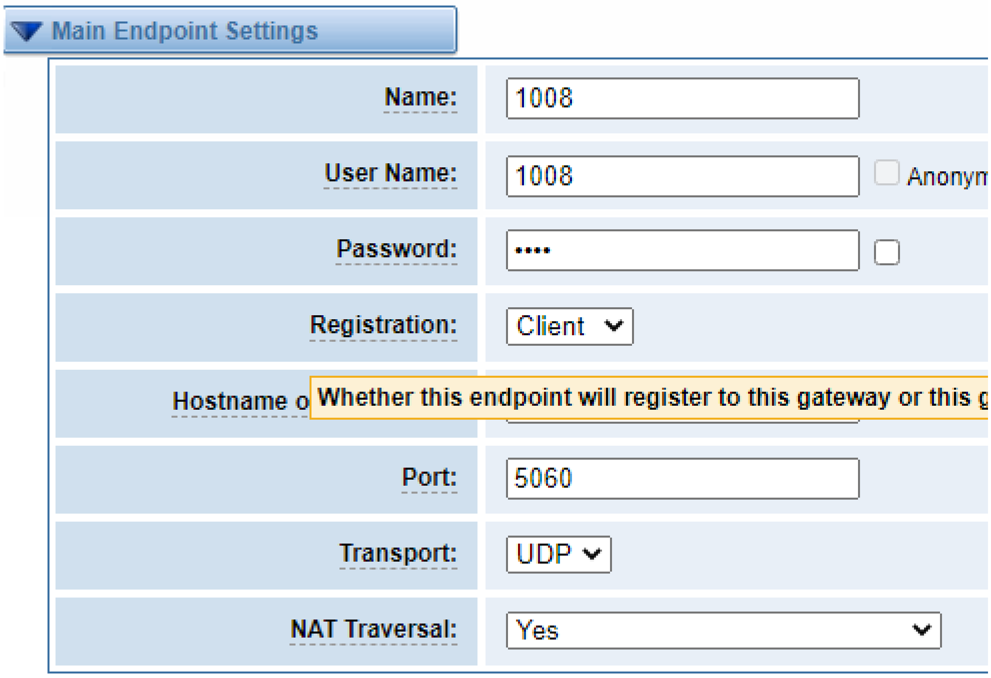


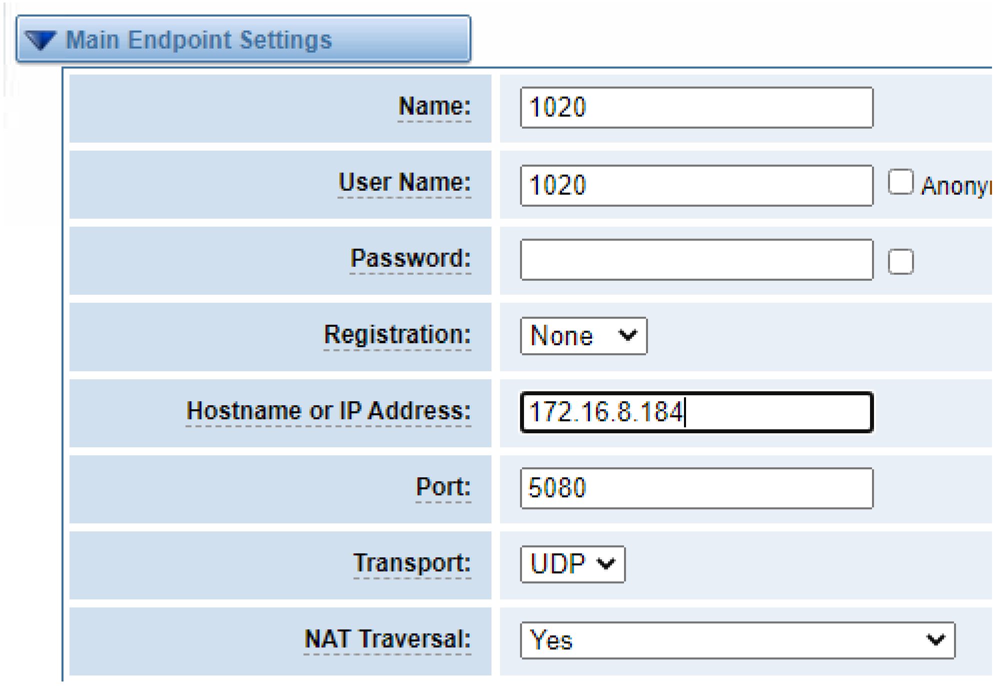
Step 5: Create SIP Trunk 1008 and Unauthenticated 1020 Endpoint in Web

Please select "VOIP->VOIP Settings->Add New SIP Endpoint" to set up SIP trunking. The image below shows details on how to set it up.

Regarding other parameters in SIP, please set them as per your requirement as there is no need to set them in a simple call.

Parameter name	value
Trunk name	1008
Relay user name	1008
Relay password	1008
Registration method	client
IP address	Server IP address (FreePBX IP address): 172.16.80.43
port	The default is 5060
Transmission method	DefaultUDP



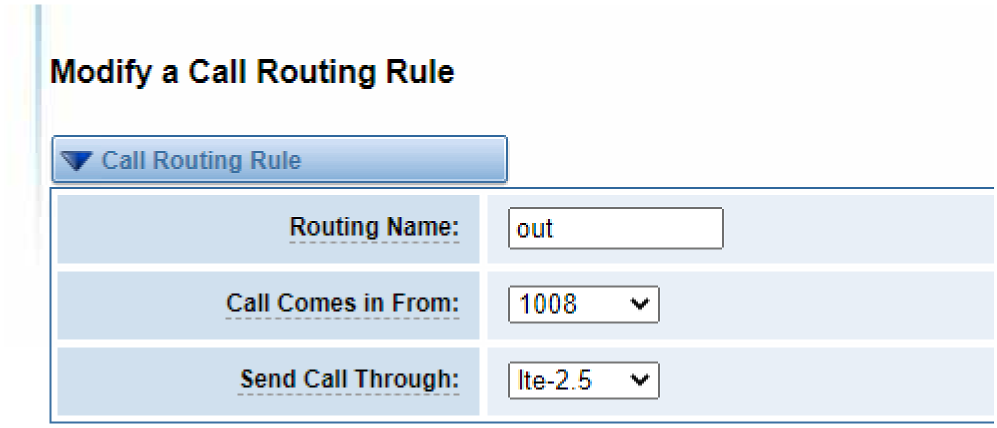


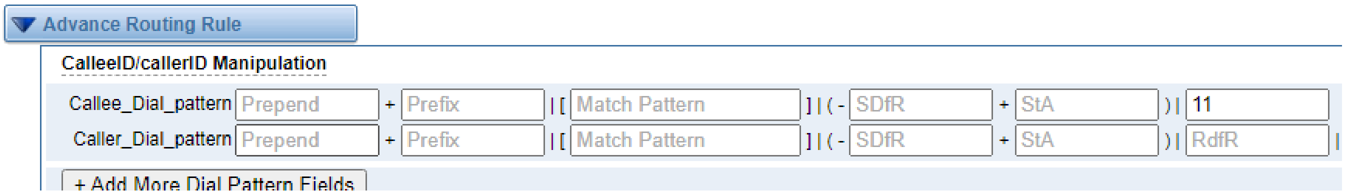
Step 6: Set up routing rules in the web

Click "Routing—>Call Routing Rules—>New Call Routing Rule" to set outbound and inbound routing rules, as shown below:

Outbound routing:

Parameter name	value
Route name	outbound
call from	Relay 1008
call delivery	Lte-2.5

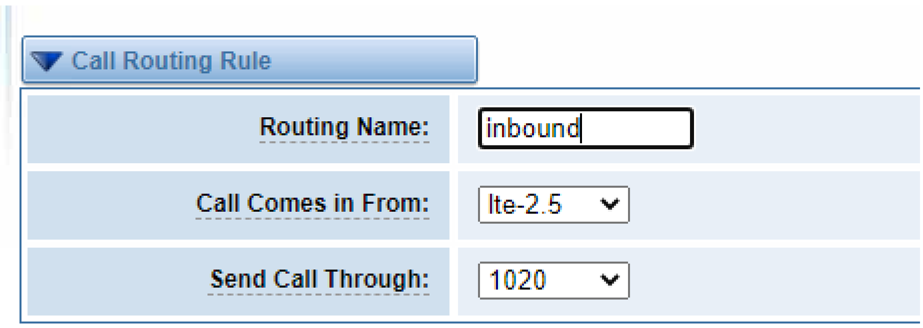




Inbound routing:

Parameter name	value
Route name	inound
call from	Lte-2.5
call delivery	1020

Please save and apply all settings.



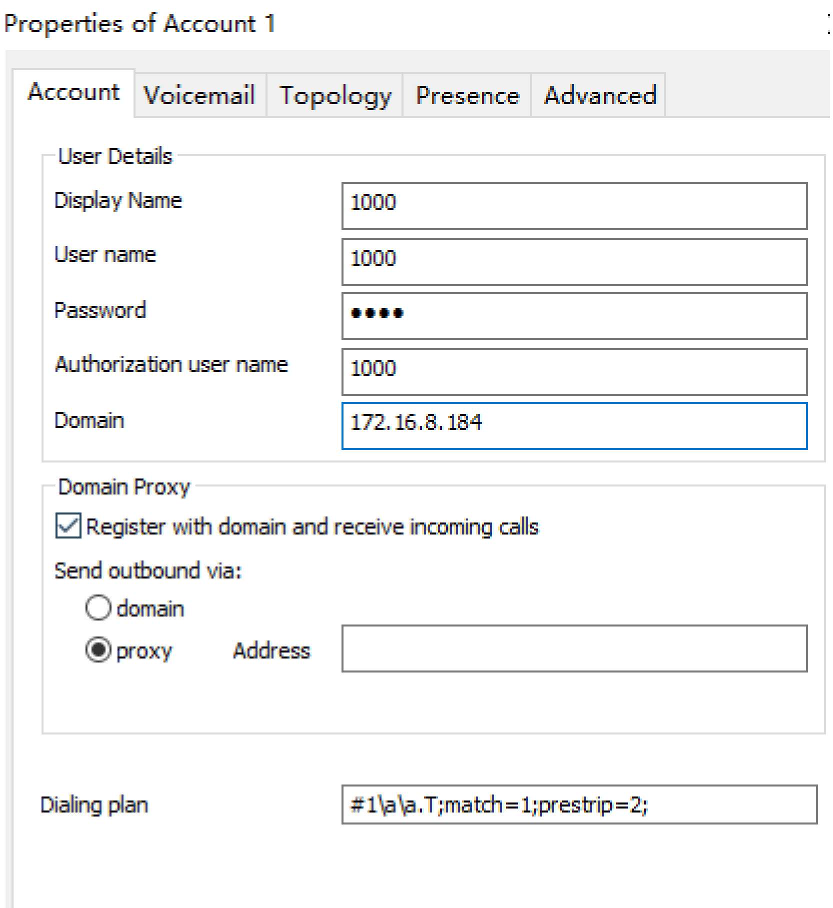
Step 7: Register extension (1000) with softphone

SIP extensions (1000) can be registered using SIP software such as Xlite, eyeBeam, microsip, etc. The picture below is an example of Xlite configuration 1000.

Test call:

Incoming call test: Use your mobile phone to dial the number of port 2.5 on the gateway to see if 1000 rings. Ringing indicates that the configuration is successful; if it does not ring, please check the configuration.

Outbound test: Dial your mobile phone number on the softphone registered with 1000. If your mobile phone rings, it means the configuration is successful; if it does not ring, it means the configuration failed. Please check the configuration.

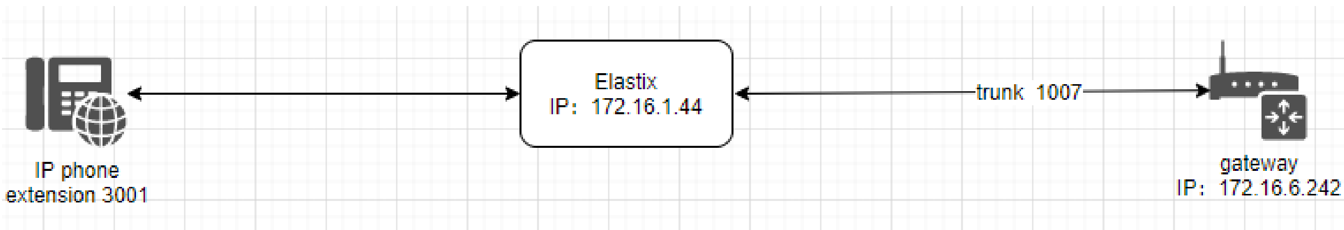


6.2.1.5 WTU service board interconnection with Elastix 4.0

This document mainly describes the detailed steps for interconnecting the business board with Elastix.

Follow the following steps to configure a two-way call between the phone and the gateway:

- Outbound calls: from FreePBX SIP extension 3002 to the gateway through trunk 1008;
- Incoming calls: Call from the outside to the gateway, go to FreePBX via SIP trunk 1008, and then send the call to SIP extension 3002 through FreePBX;



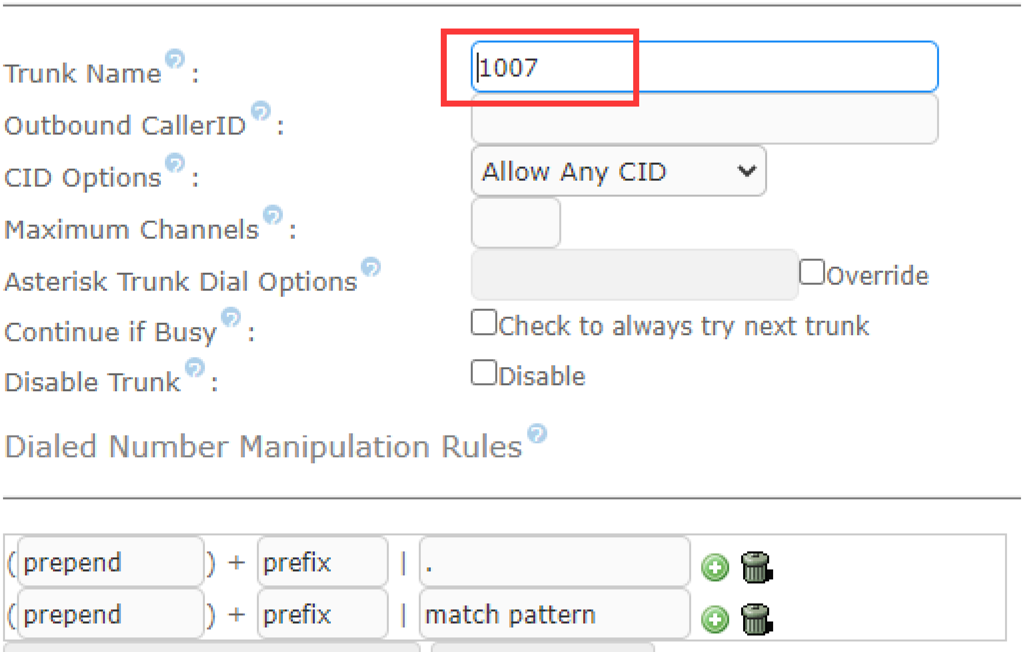
In the following steps, the parameters in the table are required configurations, and other parameters can be configured according to your own needs.

Step 1: Create SIP Trunk in Elastix Server

Please log in to your Elastix server to create a SIP trunk (1007). In the Elastix server Web, please select "PBX---->PBX Configuration---->Trunks---->Add SIP Trunk" to make the following settings:

Parameter name	Parameter value
Trunk name	1007

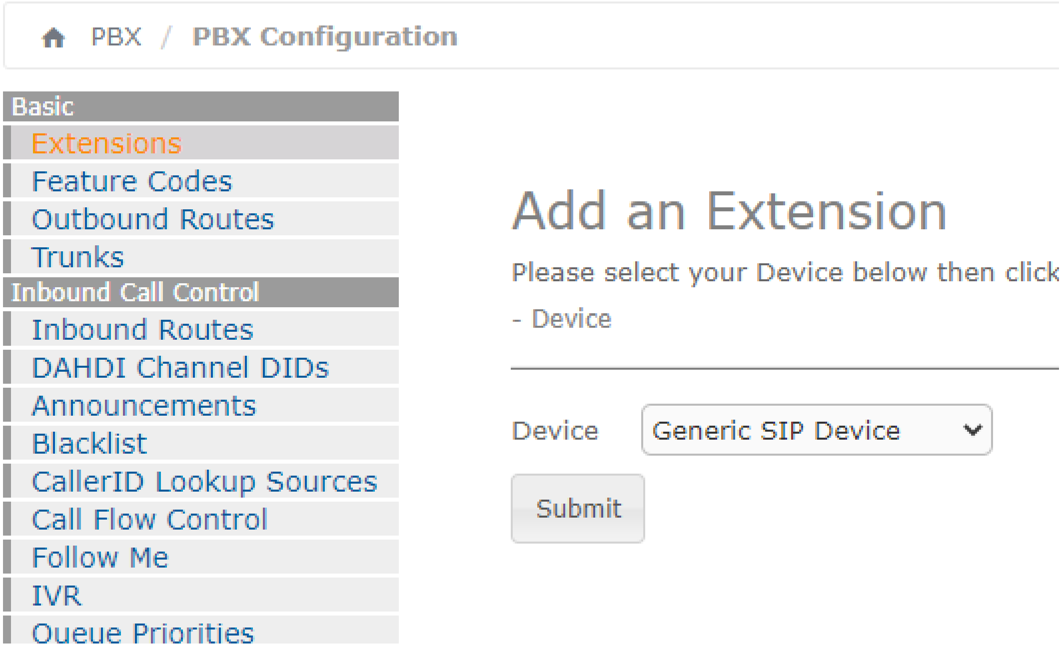
Parameter name	Parameter value
host	dynamic
username	1007
secret	123456
fromuser	1007
context	from-trunk
port	5060
type	friend

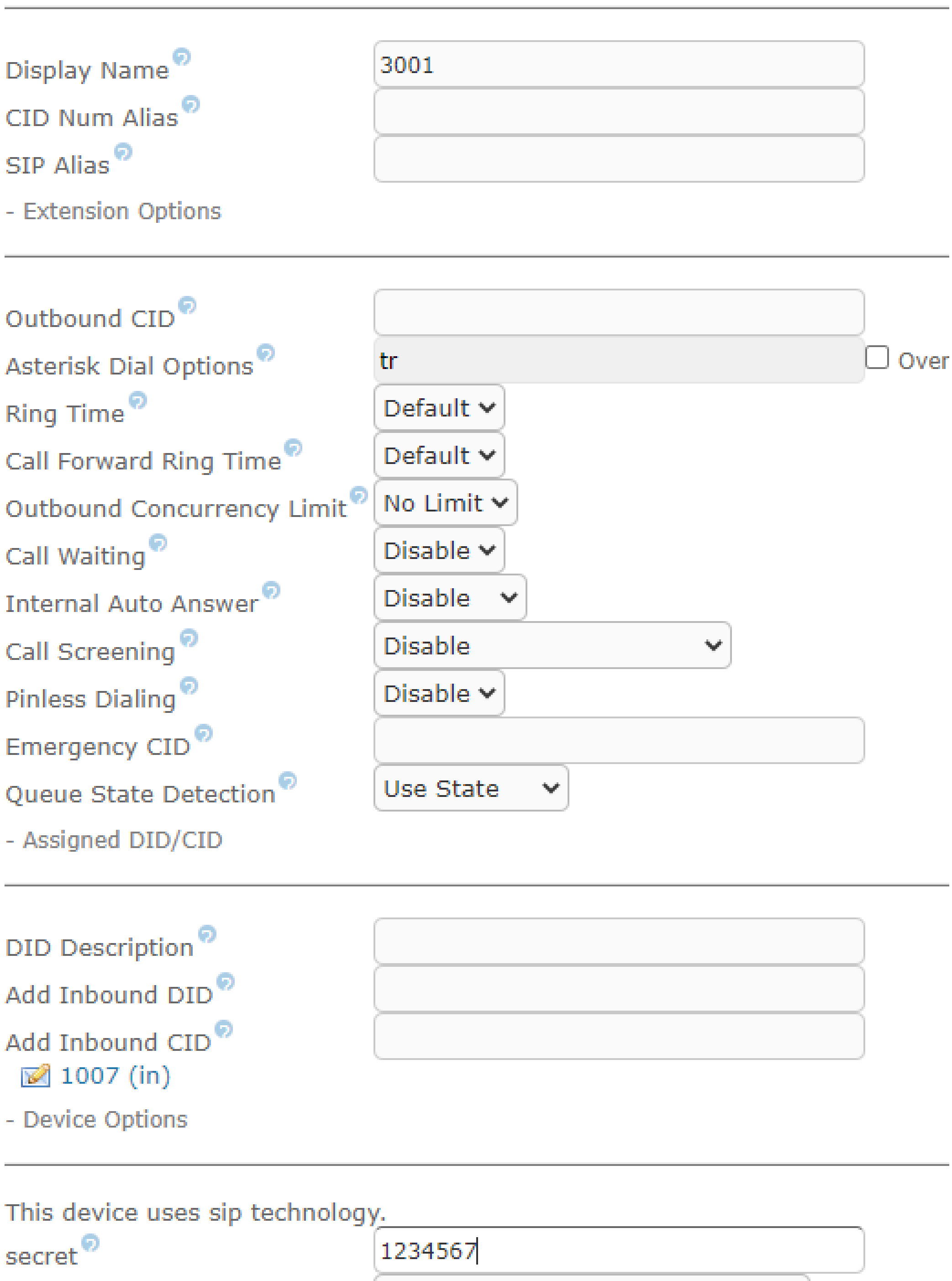


Step 2: A SIP extension (3001) in the Elastix server

In the "PBX---->PBX Configuration--->Extensions" page, select the common SIP device, click the Submit button to add the 3001 extension, and set the extension name and password.

Parameter name	Parameter value
display name	3001
secret	1234567

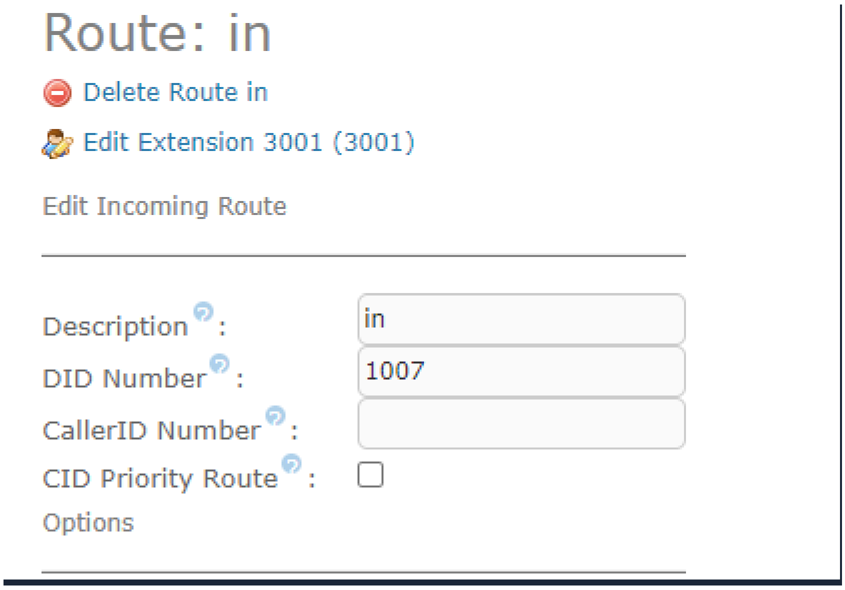




Step 3: Configure calling rules in Elastix

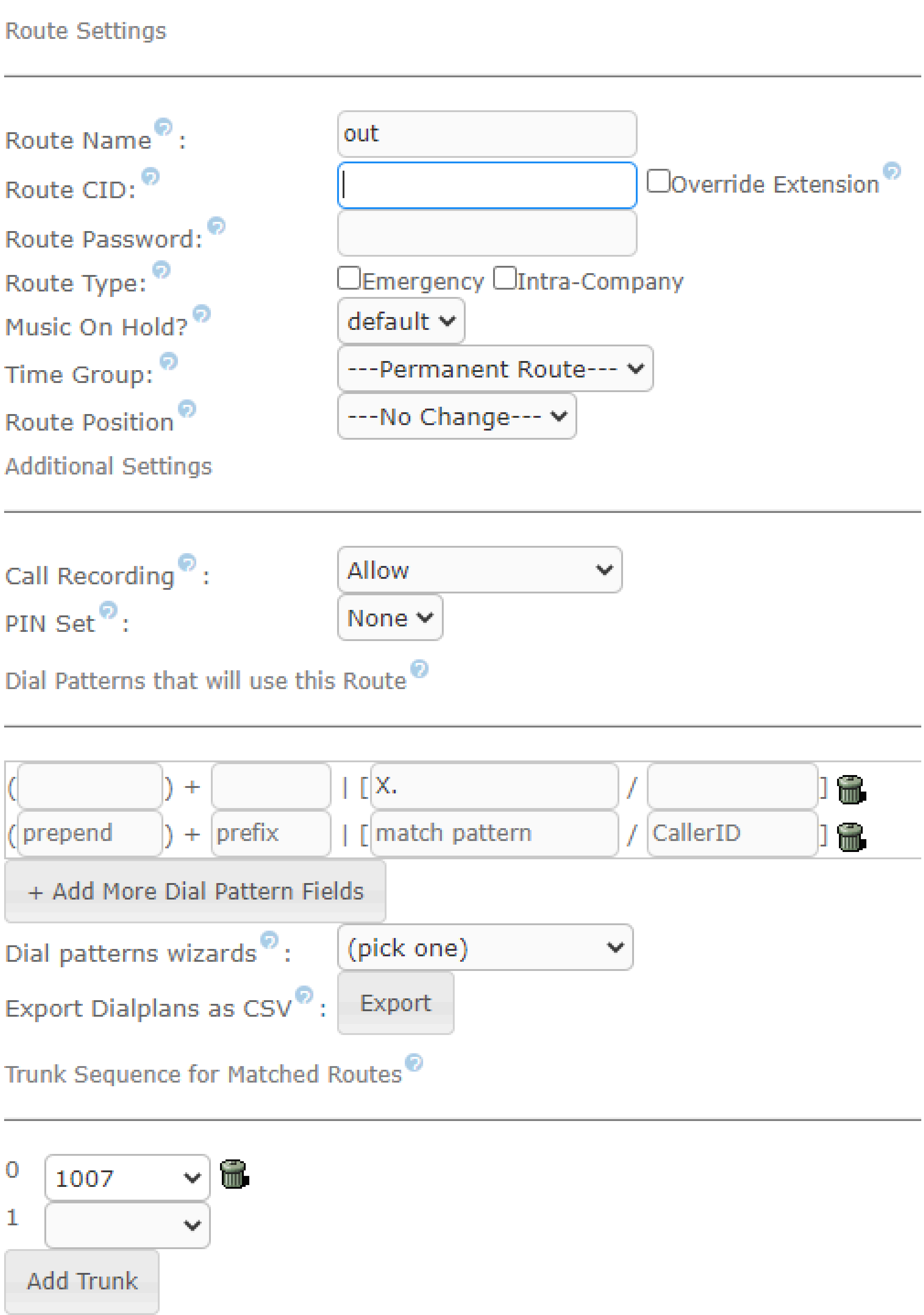
Inbound routing rules:

Parameter name	Parameter value
Description	in
DID number	1007
Grading	3001



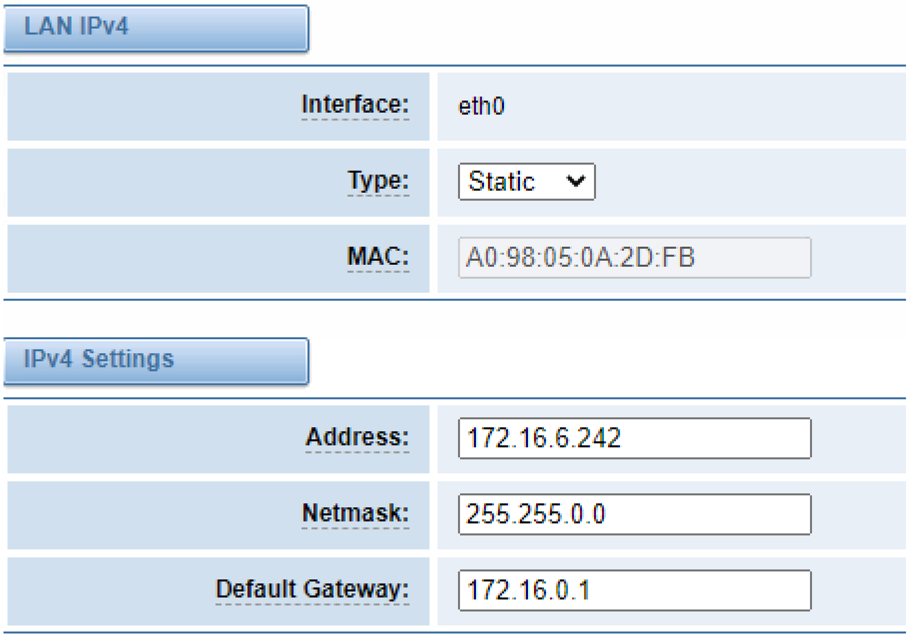
Outbound routing rules:

Parameter name	Parameter value
Line name	out
Add relay	1007



Step 4: Set network parameters in the Web

If your system topology is as shown in the picture, enter the default IP address of the gateway. Log in to the network in the browser and click "Network->LAN Settings" to set network parameters.

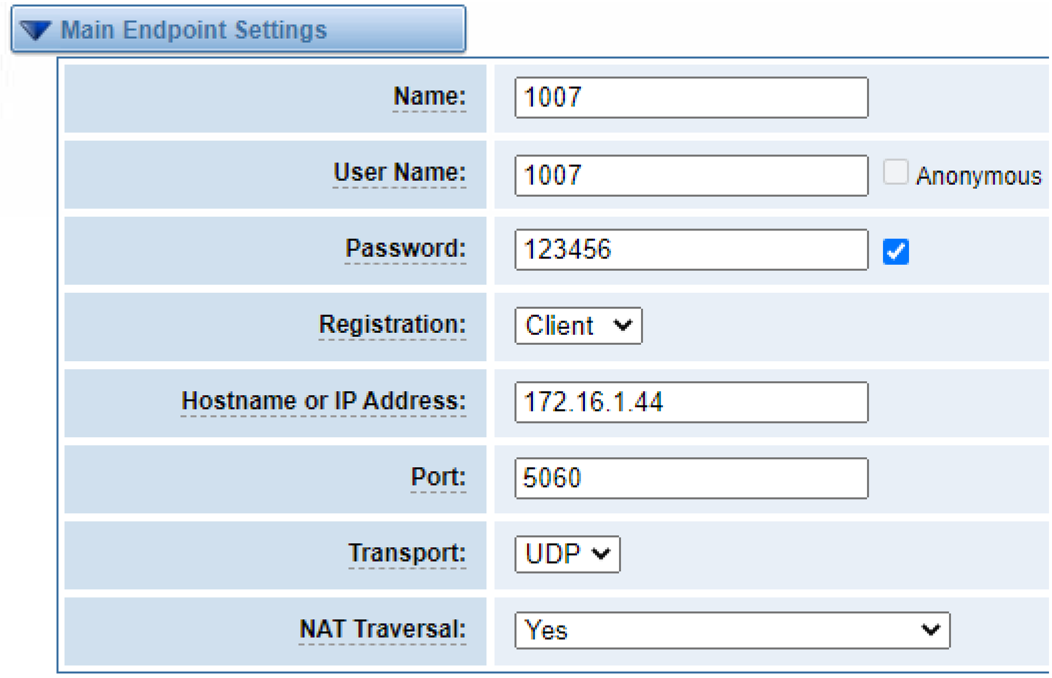


Step 5: Create SIP Endpoint in Web

Please select "VOIP->VOIP Settings->Add New SIP Endpoint" to set up SIP trunking. The image below shows details on how to set it up.

Parameter name	Parameter value
Name	1007
Username	1007
Password	123456
Registration method	client
IP address	172.16.1.44
port	5060

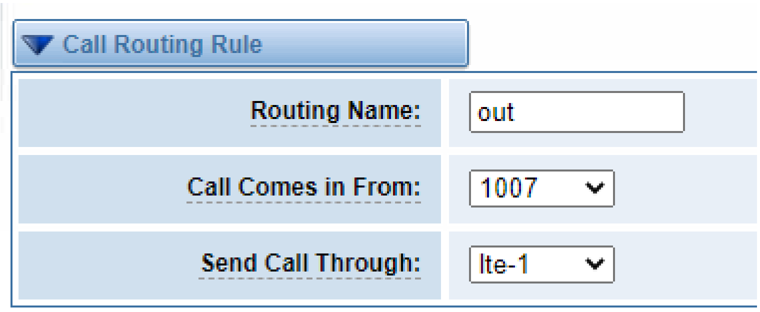
Regarding other parameters in SIP, please set them as per your requirement as there is no need to set them in a simple call.



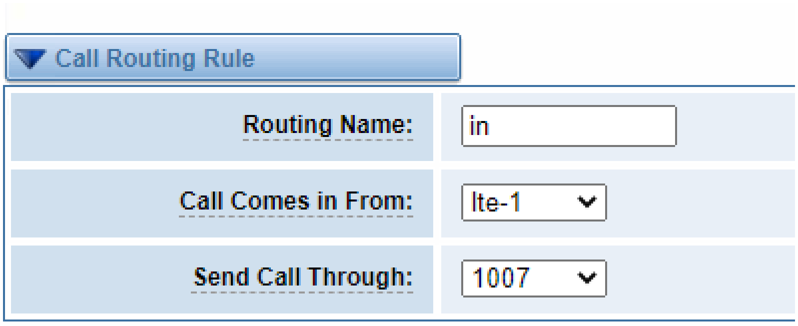
Step 6: Set up routing rules in the web

Click "Routing—>Call Routing Rules—>New Call Routing Rule" to set outbound and inbound routing rules, as shown below:

Outbound routing:

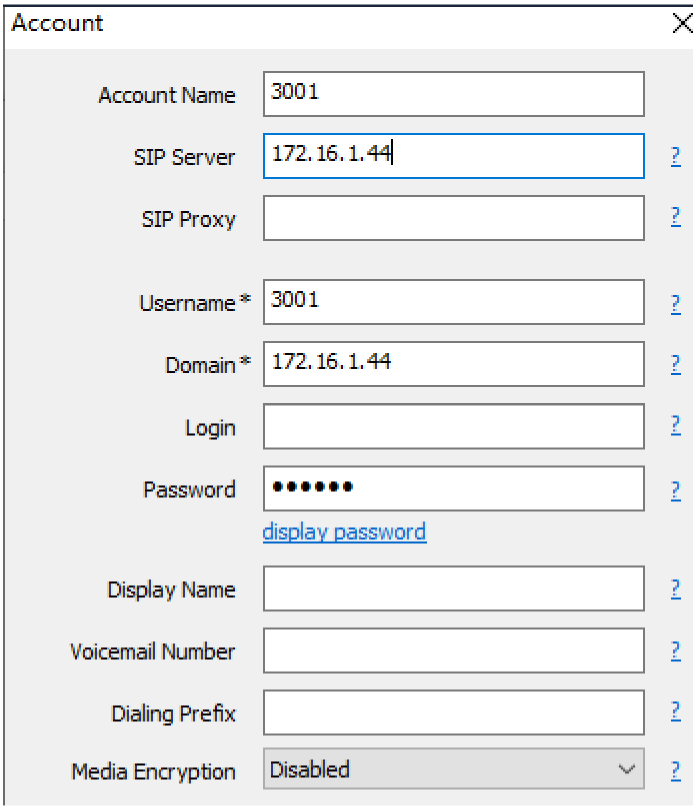


Inbound routing:



Please save and apply all settings.

Step 7: Register extension (3001) with softphone



You can use Xlite, eyeBeam, microsip and other SIP software to register the SIP extension (3001).

6.2.2 AIU-8

This document mainly describes the detailed steps for interconnecting the AIU-8 service board with the main control system 3CX, Asterisk16.15.1, FreePBX15, and OpenVOX UC.

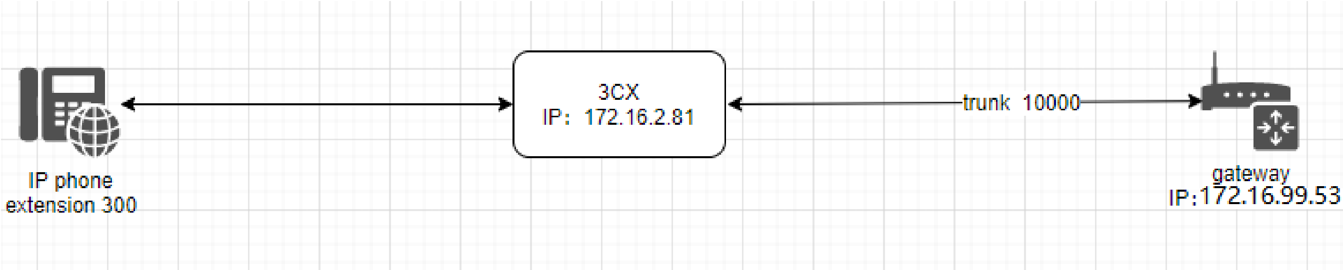
- AIU-8 FXO business board is connected to the main control system 3CX This article mainly introduces the detailed steps of connecting the AIU-8 FXO business board and the main control system 3CX.
- AIU-8 FXO business board is connected to the main control system Asterisk This article mainly introduces the detailed steps for connecting the AIU-8 FXO business board to Asterisk.
- AIU-8 FXO business board connected to FreePBX This article mainly introduces the detailed steps of connecting the AIU-8 FXO business board and Freepbx.
- AIU-8 FXO business board and main control system Open UC connection This article mainly introduces the detailed steps for connecting the AIU-8 FXO service board and UC.

6.2.2.1 AIU-8 FXO service board and main control system 3CX connection

This document mainly introduces the detailed steps for connecting the AIU-8 service board and the main control system 3CX.

Follow the steps below to configure two-way calling between the phone and the gateway.

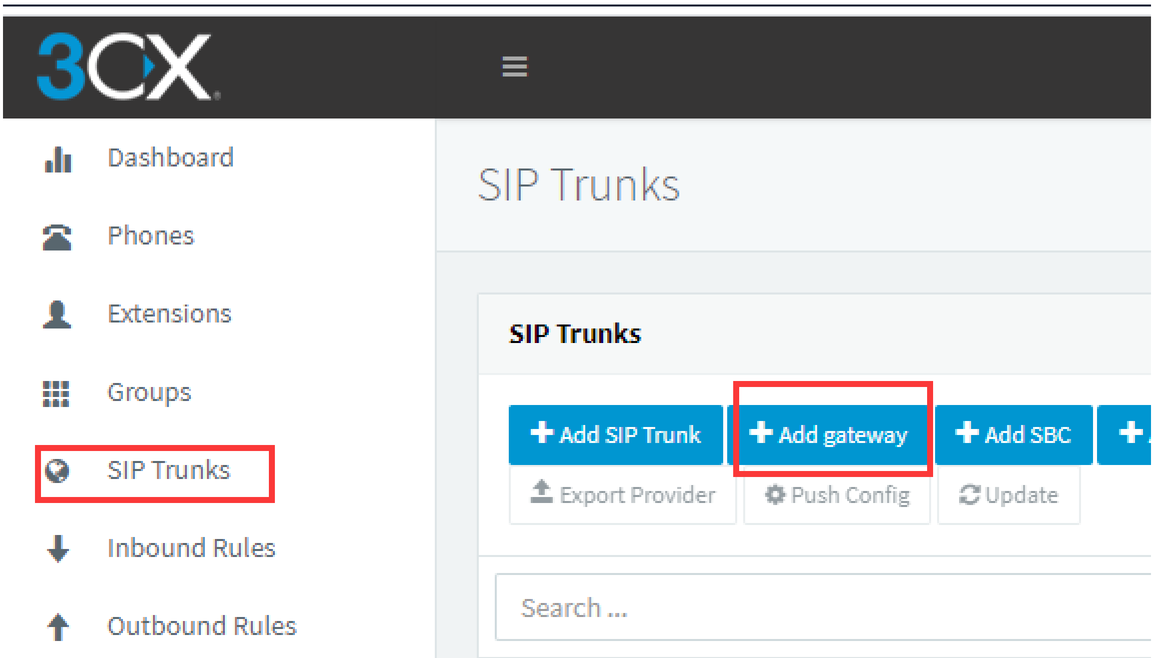
- Outbound calls: from 3CX SIP extension 300 through trunk 10000 calling the gateway.
- Incoming call: from outside call to gateway, via SIP trunk 10000 to 3CX, then send the call to 300SIP extension via 3CX.

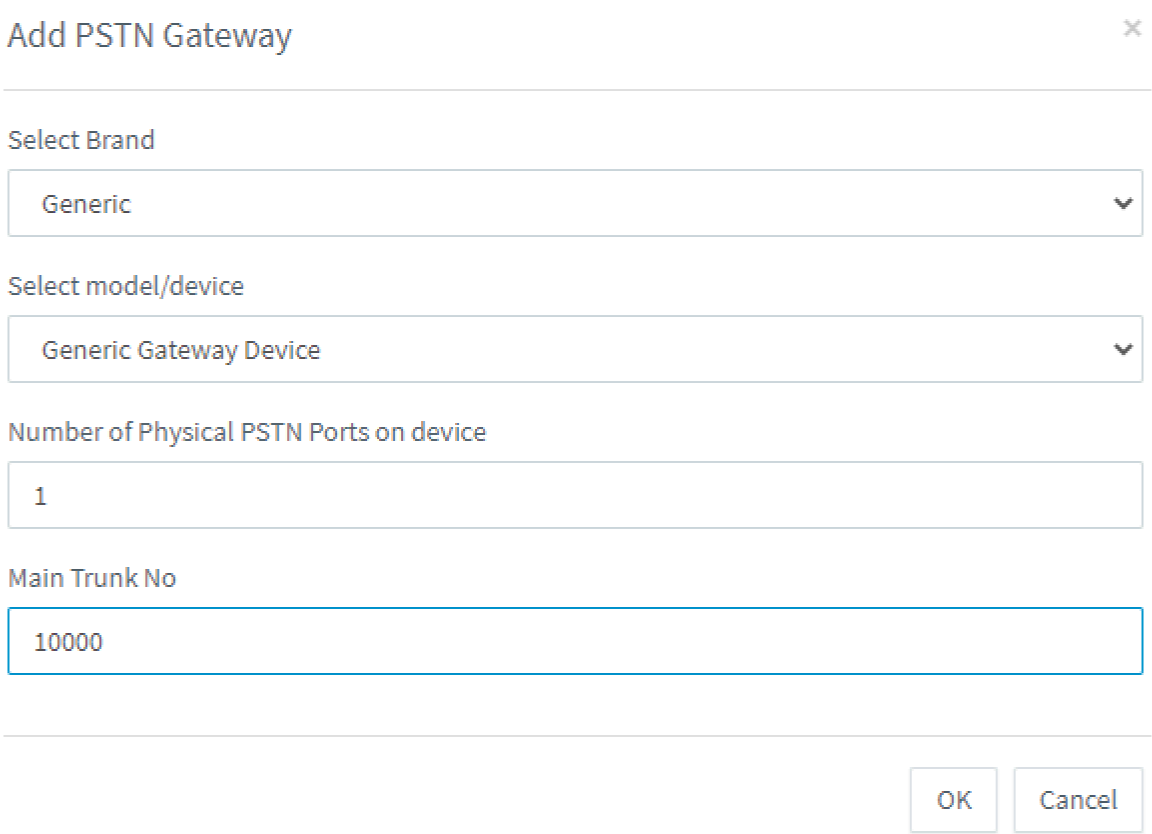


In the following steps, the following parameters must be configured, and other parameters can be configured according to your own needs.

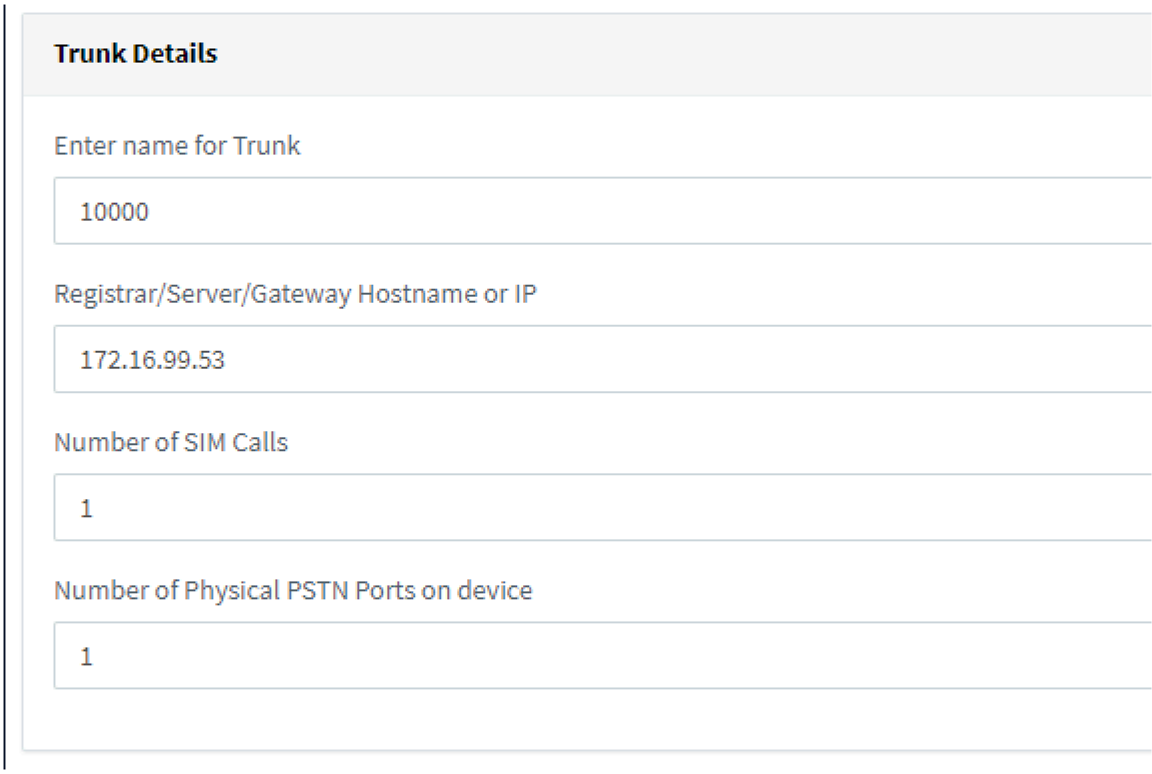
Step 1: Create a SIP trunk in the 3CX webpage

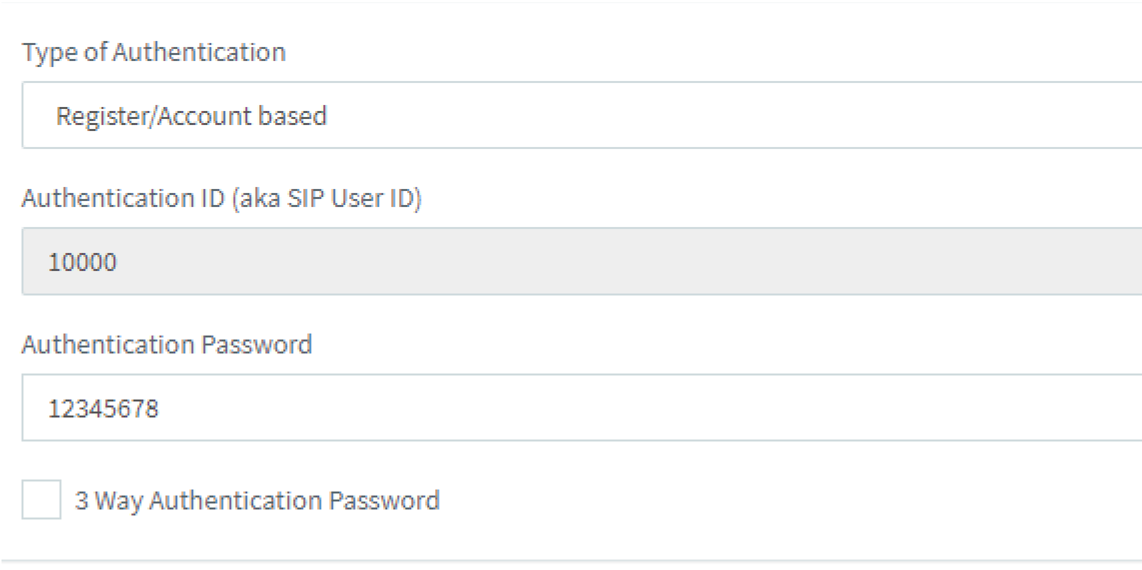
Please select "SIP Trunks->Add Gateway" to create a SIP trunk.



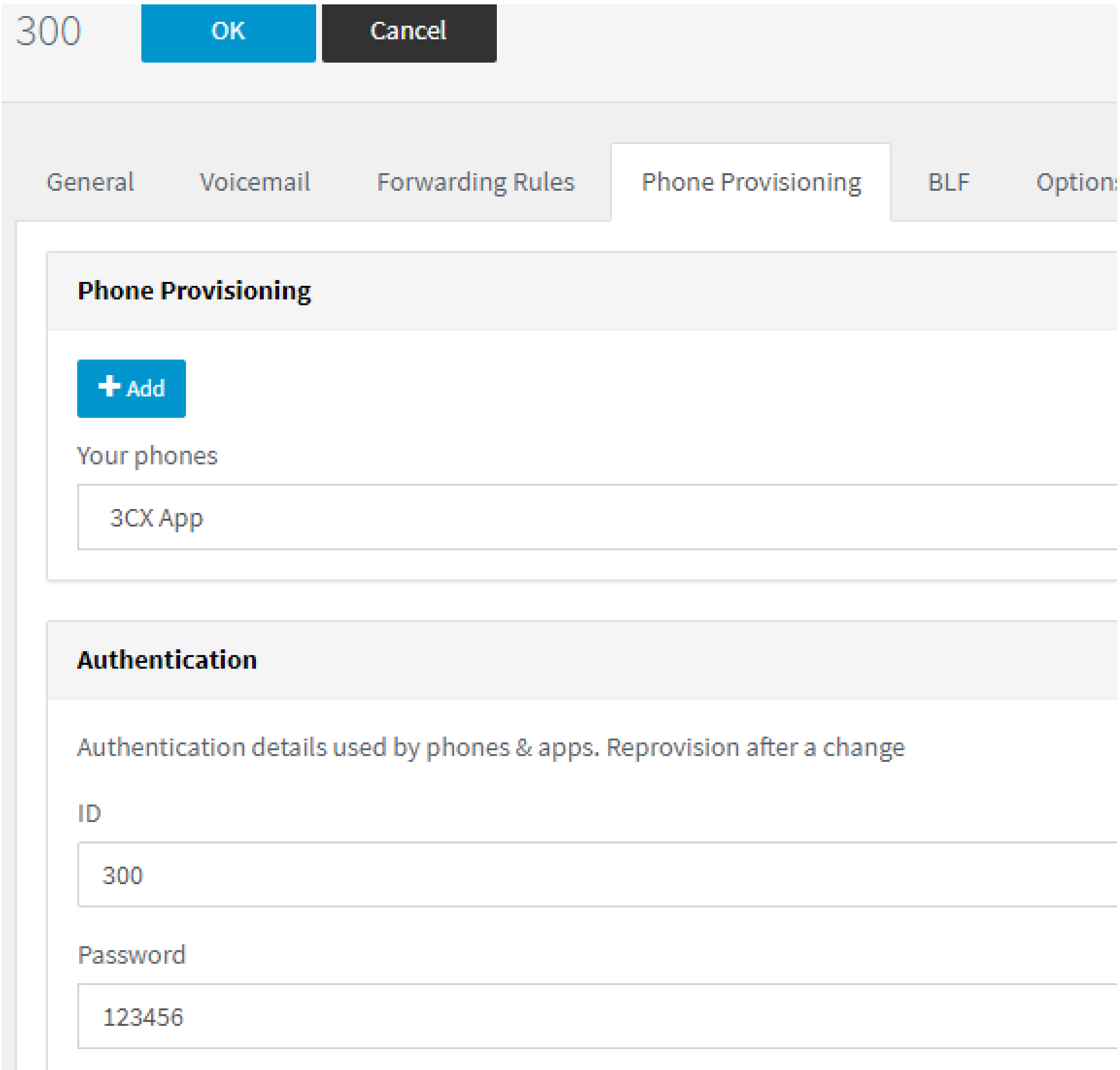


After that, please click the next button to go to the following page.





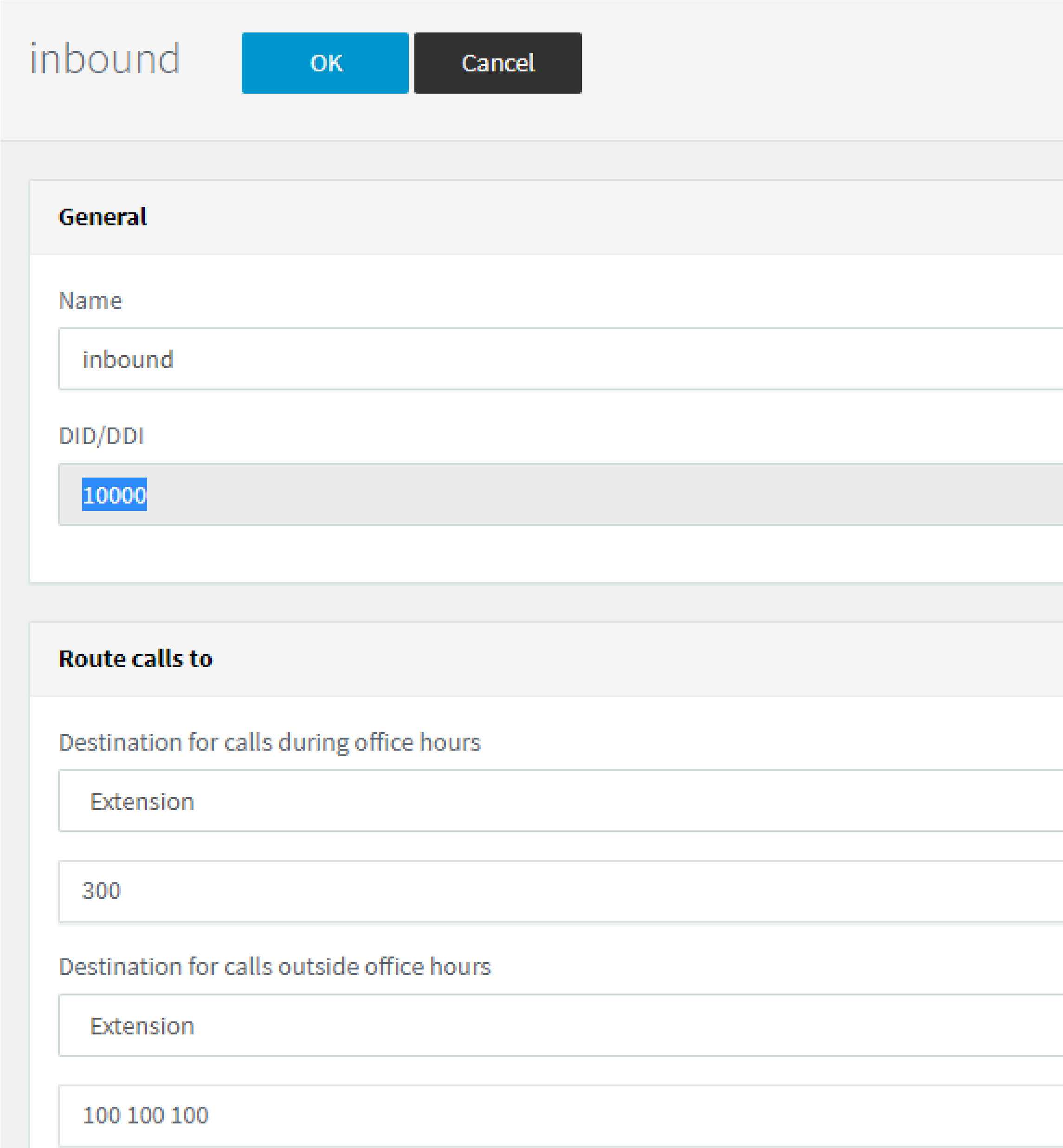
Step 2: Create extension 300



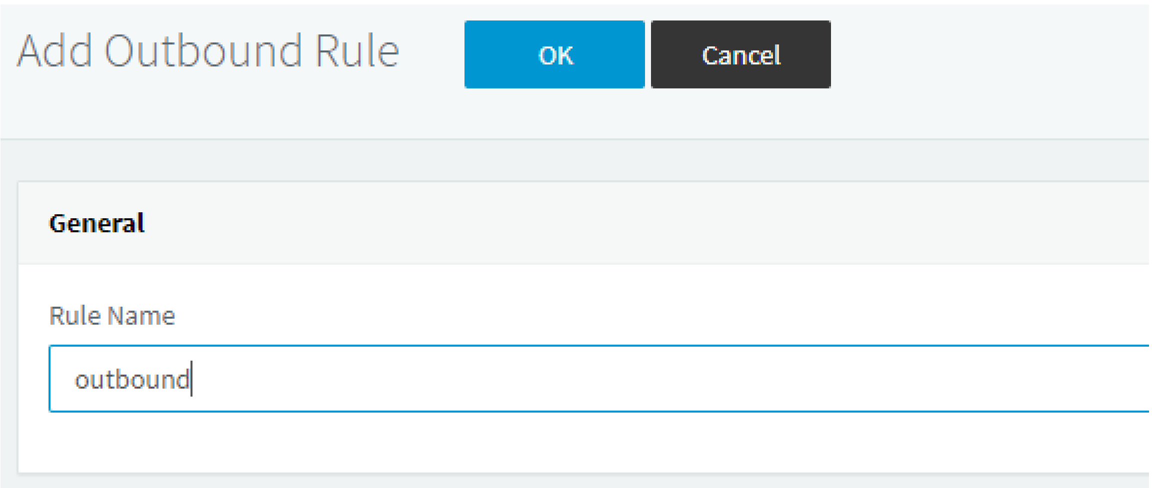
Step 3: Set up dialing rules in the 3CX webpage

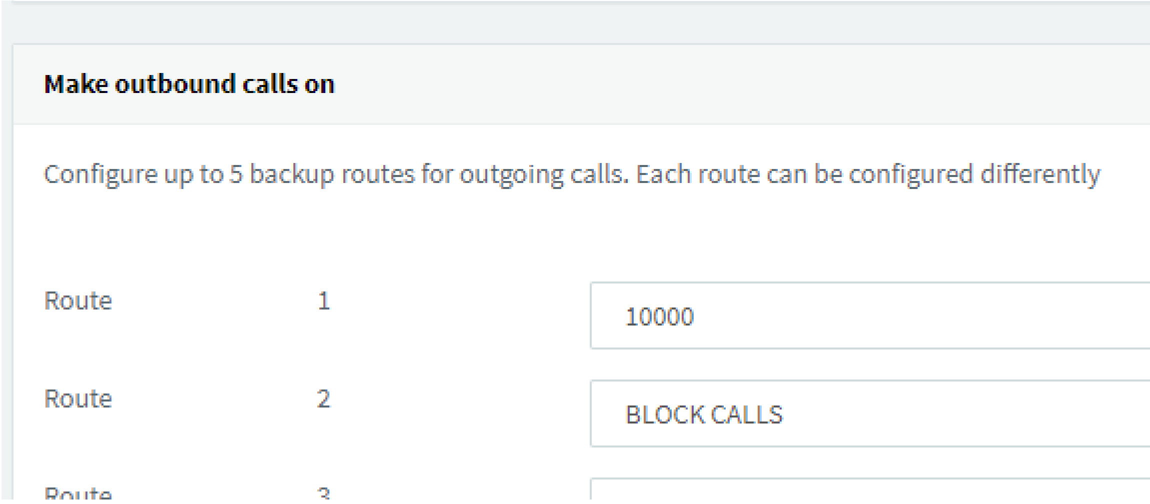
Click on Inbound Rules->Add DID to set Inbound Rules

Inbound routing rules



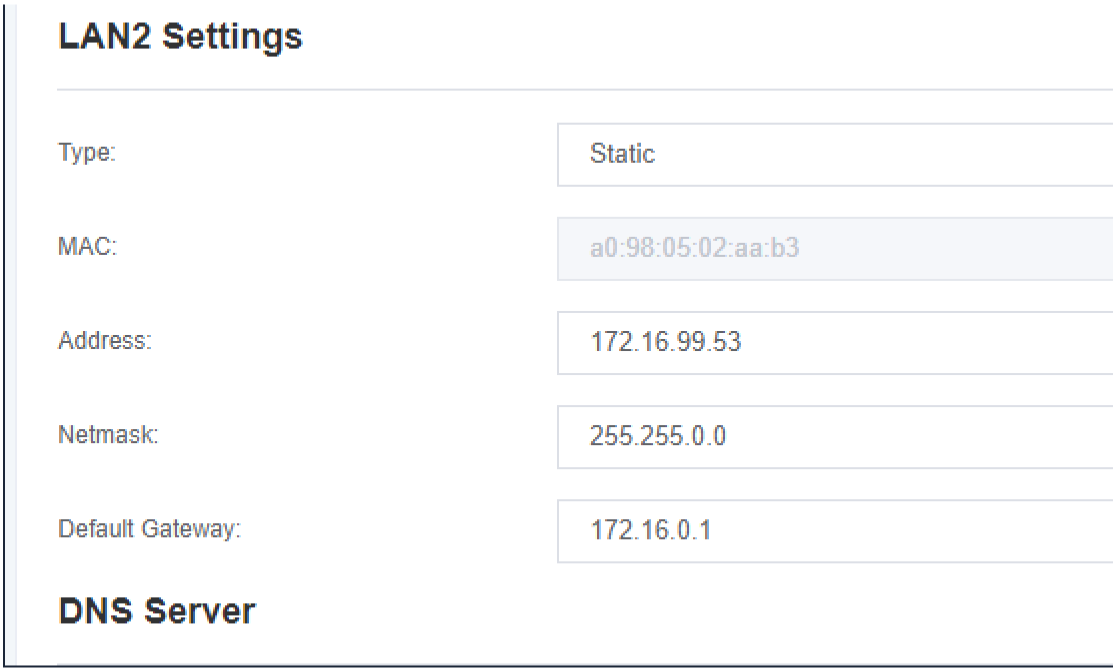
Click Outbound Call Rules→Add Outbound Call to set outbound call rules





Step 4: Set network parameters in the network

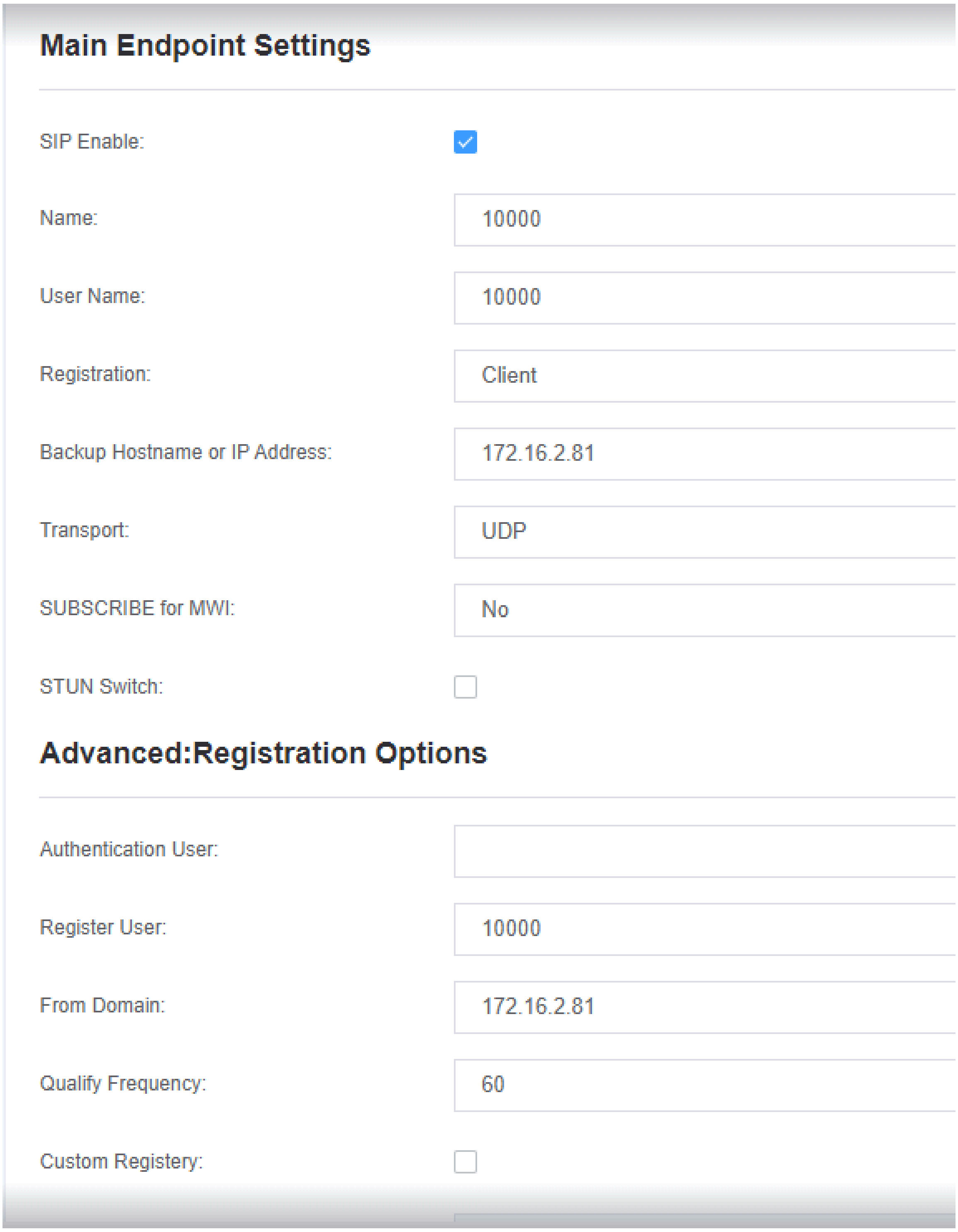
Log in to the network in the browser and set network parameters. The picture below is an example and is for reference only.

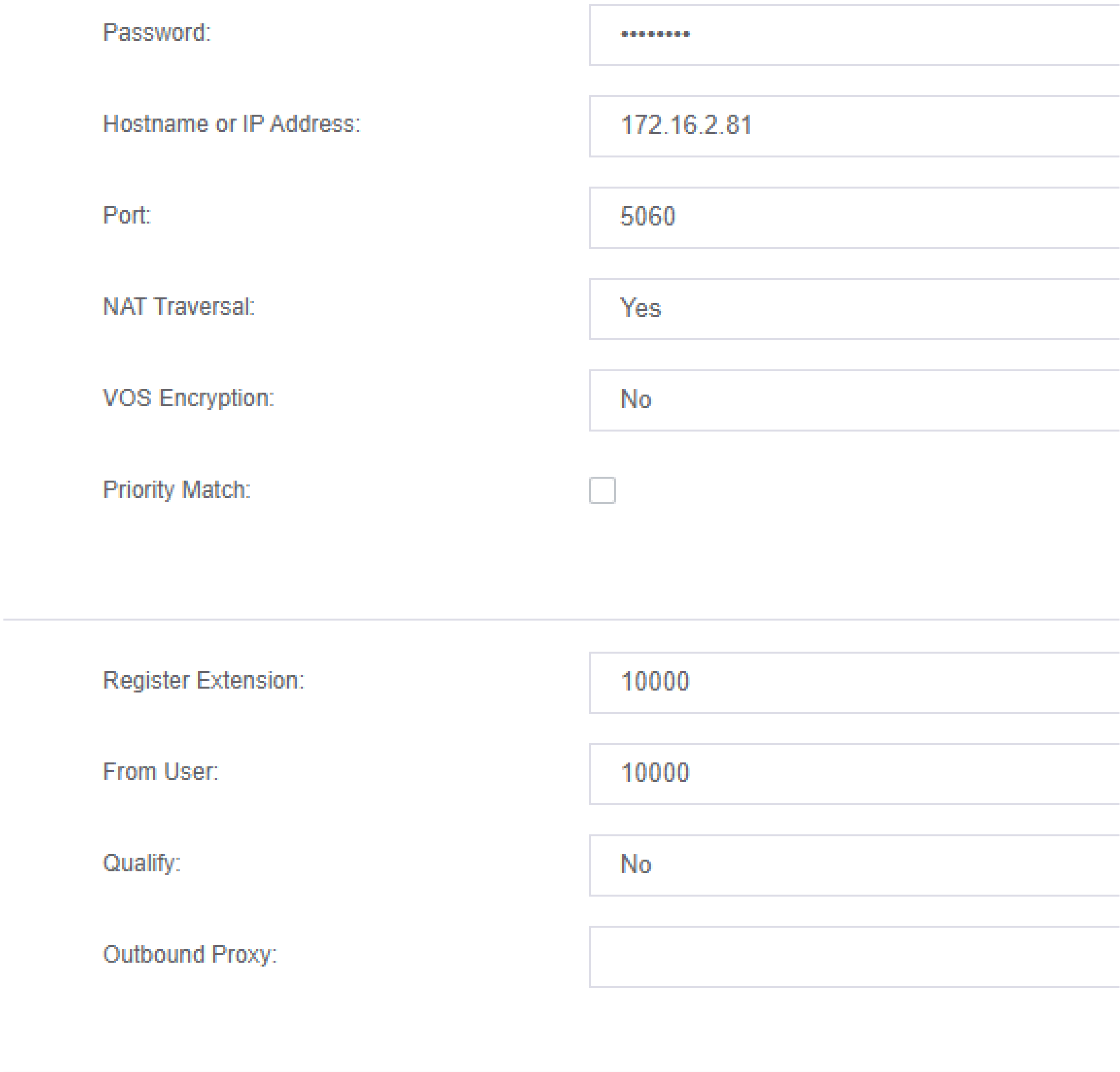


Save your changes. Please type your DNS server in "DNS Server Address".

Step 5: Create a SIP endpoint in the network

Please set up SIP trunk. The image below shows details on how to set it up.


Main Endpoint Settings
SIP Enable:
Name:
10000
User Name:
10000
Registration:
Client
Backup Hostname or IP Address:
172.16.2.81
Transport:
UDP
SUBSCRIBE for MWI:
No
STUN Switch:
Advanced:Registration Options
Authentication User:
Register User:
10000
From Domain:
172.16.2.81
Qualify Frequency:
60
Custom Registery:

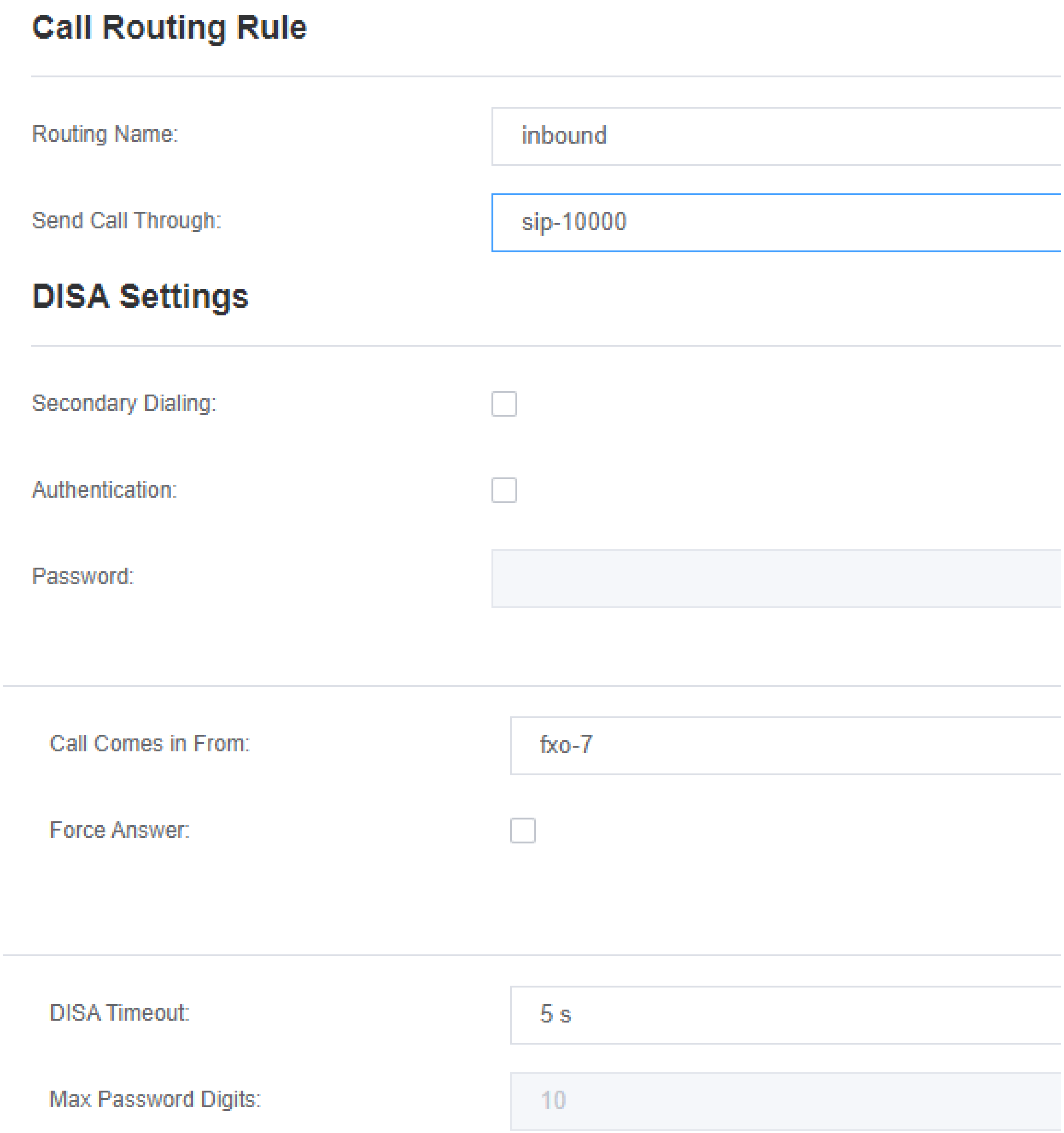


Regarding other parameters in SIP, please set them according to your requirements, because there is no need to set them in a simple call.

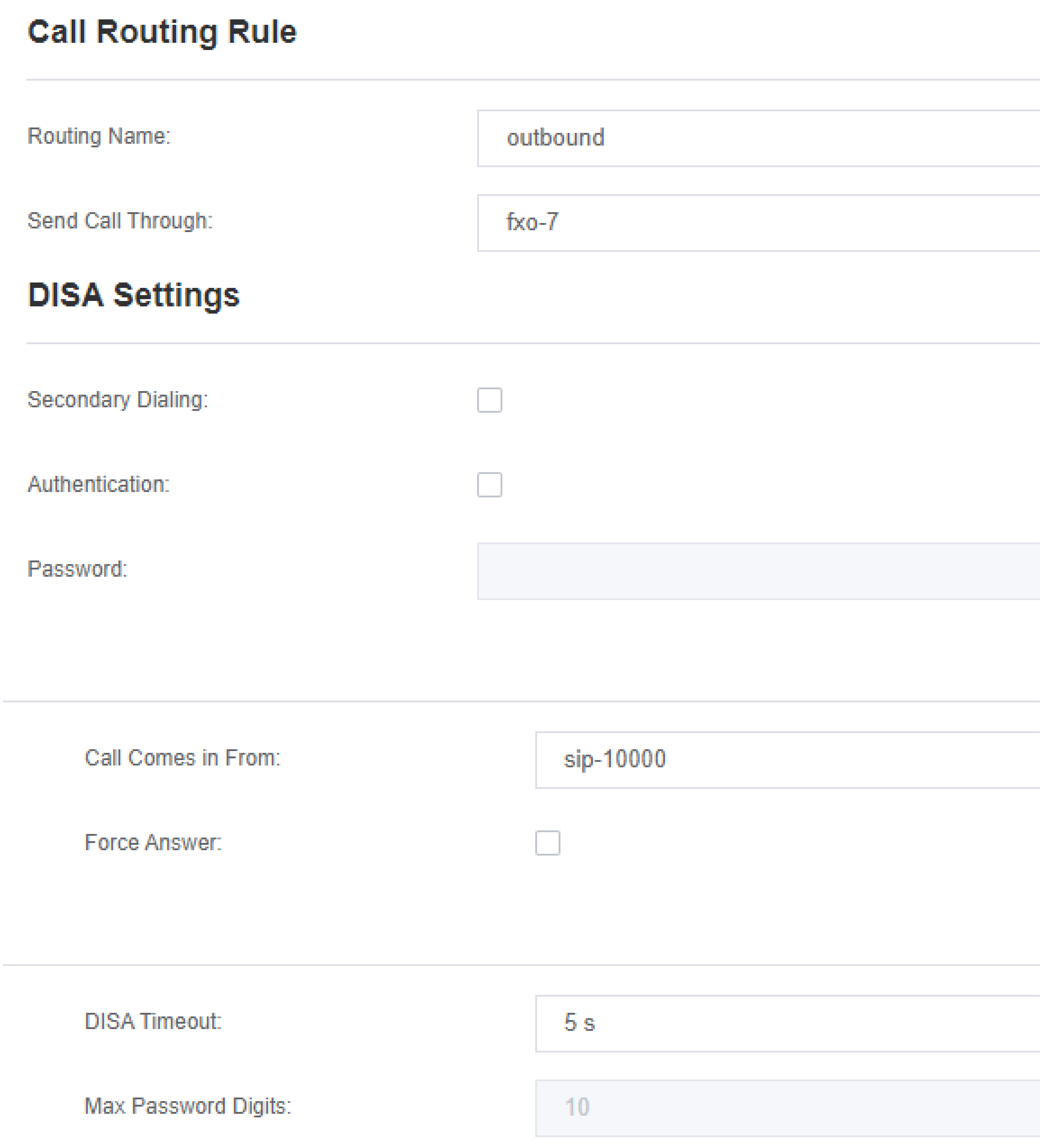
Step 6: Set up routing rules in the web

Click "ROUTING-> Call Routing Rules-> New Call Routing Rule" to set the routing rules for outbound and inbound calls, as shown below.

Inbound routing rules.



Outbound routing rules.



Please save all your changes to take effect.

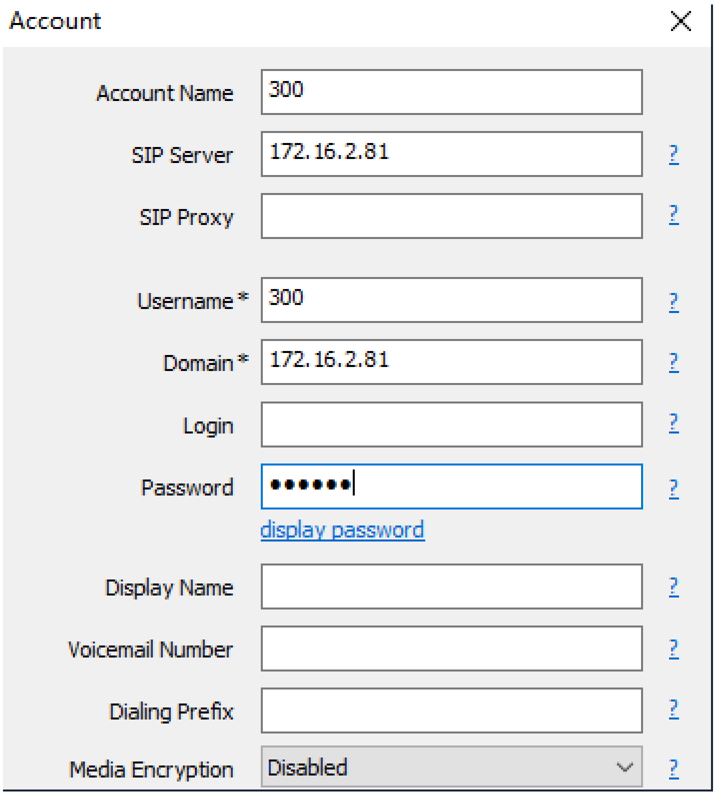
Step 7: Register a SIP extension through the software

Take advantage of SIP software, such as Xlite, eyeBeam, to register a SIP extension (300). After all the above steps, you can try making calls and sending text messages.

Test call.

Call test: Dial the number on the gateway through your mobile phone to see if 300 will ring. If it rings 300, it means your configuration is successful; otherwise, it means there is a problem with your configuration, please check.

Outbound call test: Dial your mobile phone number on the 300 extension registered in the software phone. If your phone rings, your configuration is OK; otherwise, please check your configuration.

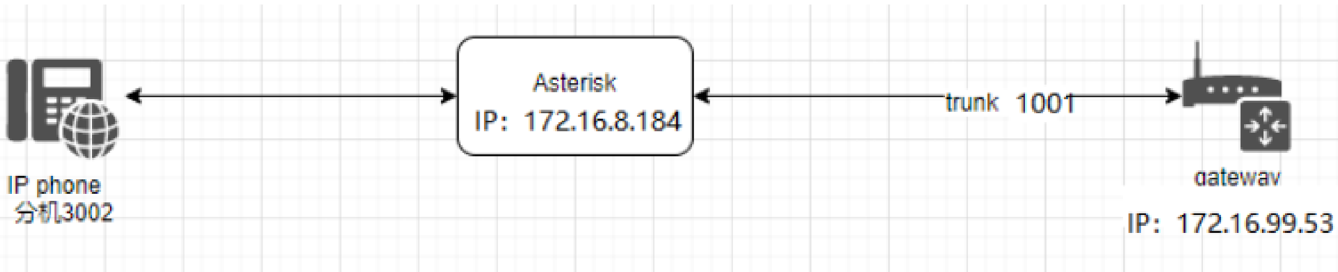


6.2.2.2 Connection between AIU-8 FXO service board and main control system Asterisk 16.15.1

This article mainly introduces the detailed steps for connecting the business board to Asterisk.

Follow the steps below to configure two-way calling between the phone and the gateway.

- Outbound calls: from Asterisk SIP extension 3002 through trunk 1008 to the gateway.
- Incoming call: Call from the outside to the gateway, call to Asterisk through SIP trunk 1008, and then send the call to 3002 SIP extension through Asterisk.



In the following steps, the following parameters are mandatory configurations, other parameters can be configured according to your needs.

Step 1: Create a SIP trunk on the Asterisk server

Please add the following lines in /etc/asterisk/sip.conf to create a SIP trunk (1001) and a SIP extension (3002).

```
[1001]
username=1001
secret=1001
host=dynamic
port=5060
type=friend
```

```
context = from-gsm

[3002]
username=3002
secret=3002
host=dynamic
port=5060
type=friend
context = from-internal
```

After editing, save and exit, and restart the SIP service of the Asterisk server.

Step 2: Edit Dial Rules in Asterisk

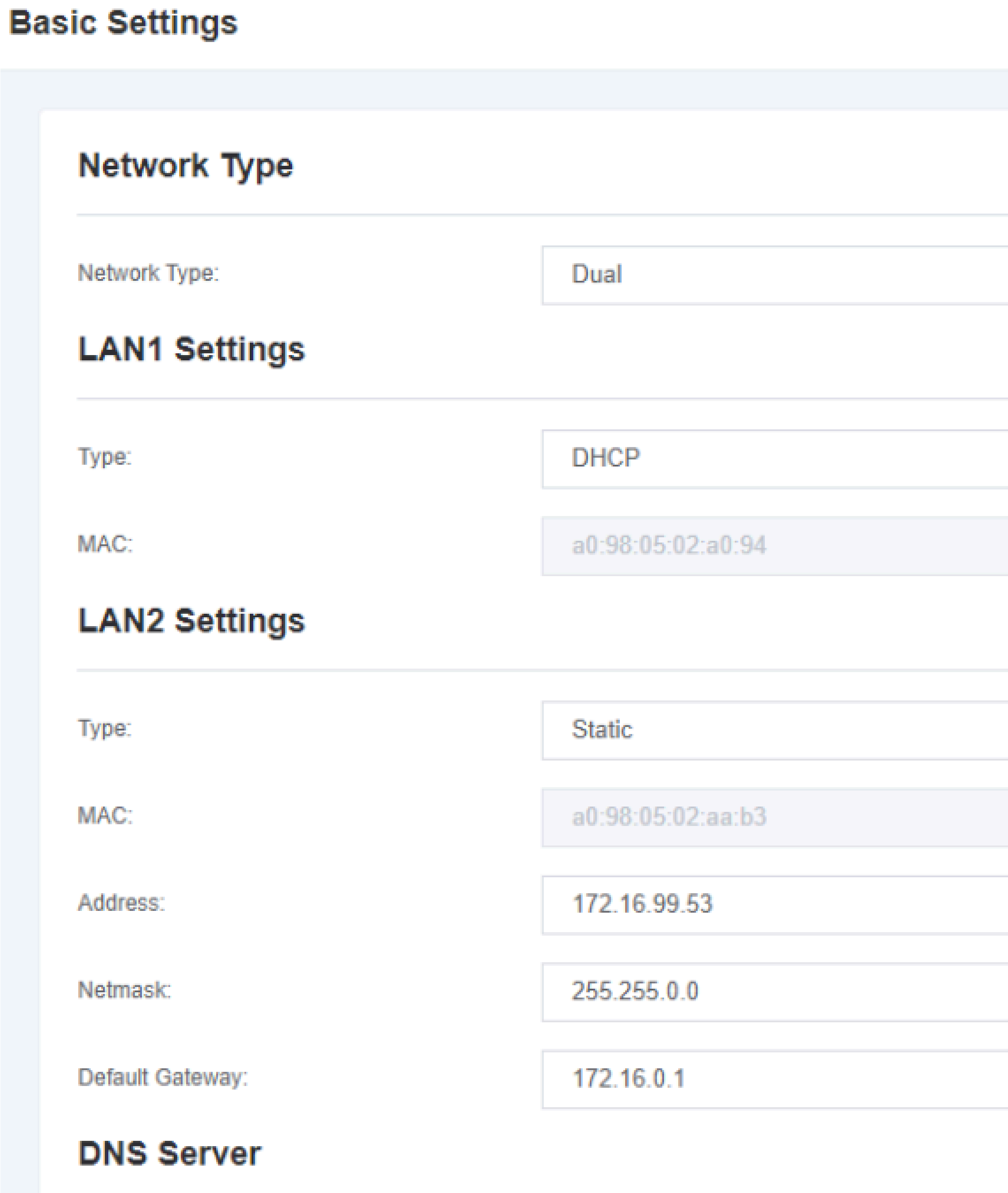
Edit the dialing rules in /etc/asterisk/extensions.conf.

```
[from-internal]
exten => _x.,1,Dial(sip/1001/${EXTEN})
exten => _x.,n,hangup()

[from-gsm]
exten =>_x.,1,Dial(sip/3001)
exten =>_x.,n,Hangup()
```

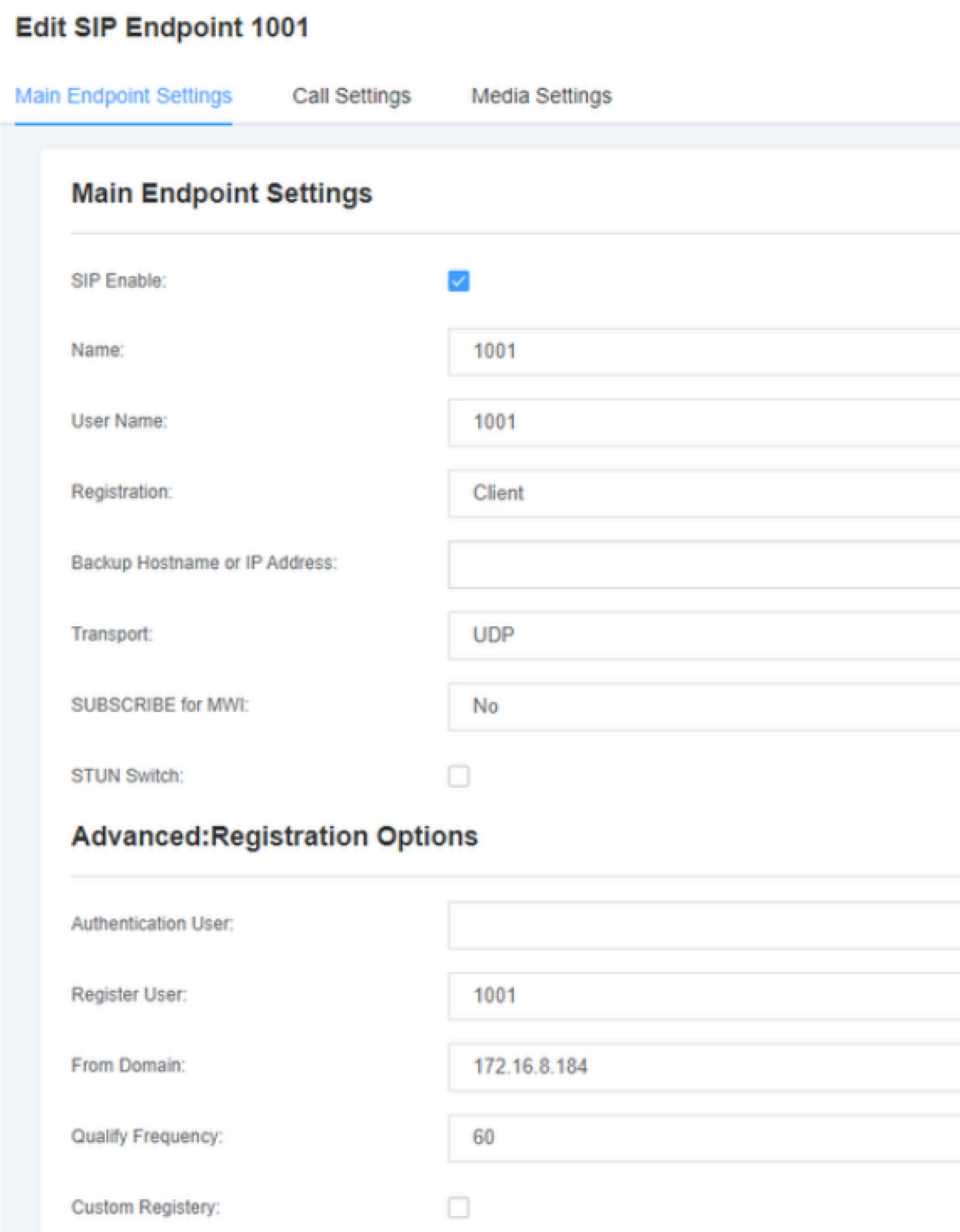
Step 3: Set network parameters in the network

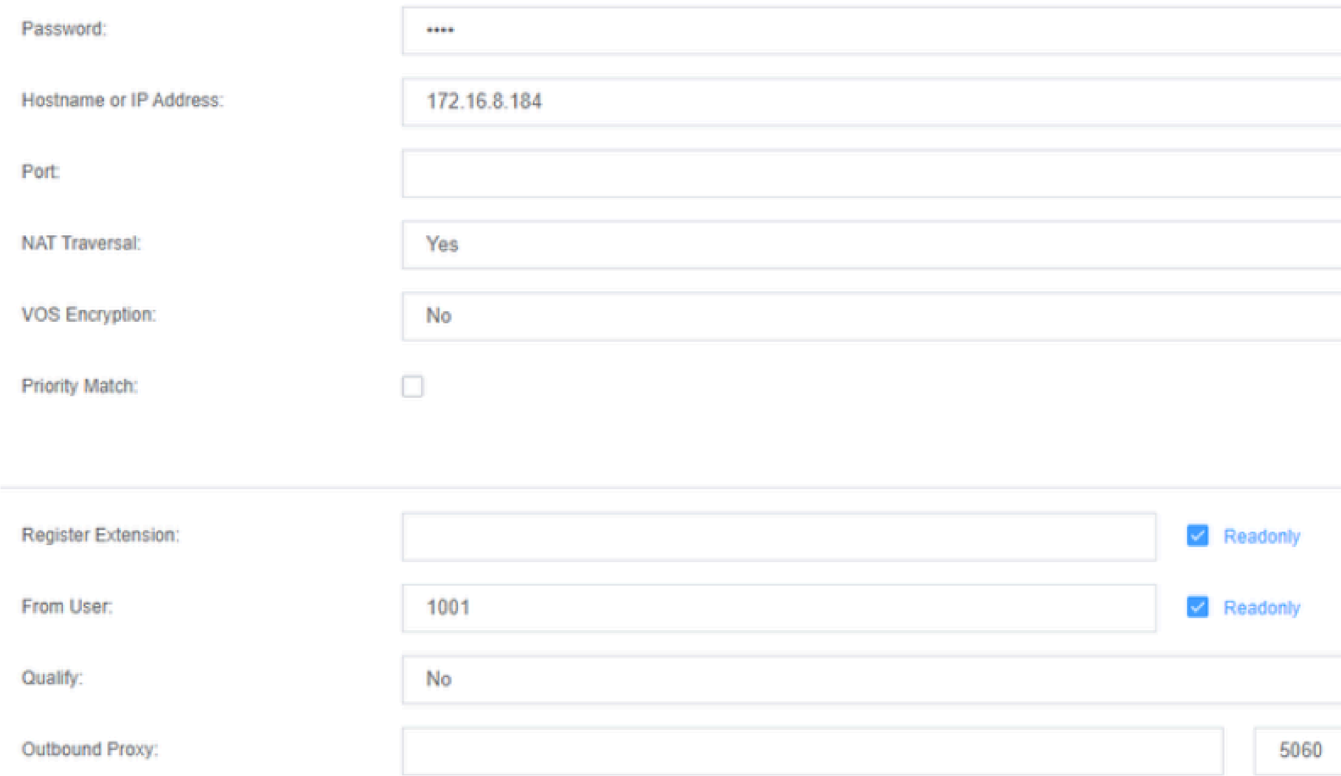
Log in to the network in the browser and set network parameters. The picture below is an example and is for reference only.



Step 4: Create a SIP endpoint in the network

Set up SIP trunk. The image below shows details on how to set it up.


Edit SIP Endpoint 1001
Main Endpoint Settings
Call Settings
Media Settings
Main Endpoint Settings
SIP Enable:
Name:
1001
User Name:
1001
Registration:
Client
Backup Hostname or IP Address:
Transport:
UDP
SUBSCRIBE for MWI:
No
STUN Switch:
Advanced:Registration Options
Authentication User:
Register User:
1001
From Domain:
172.16.8.184
Qualify Frequency:
60
Custom Registery:

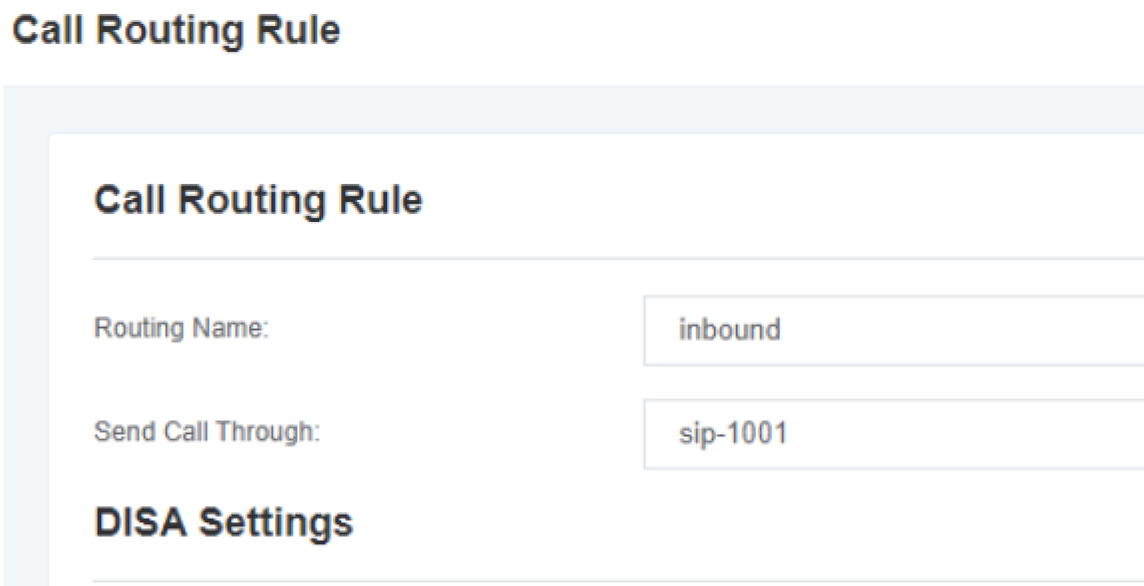


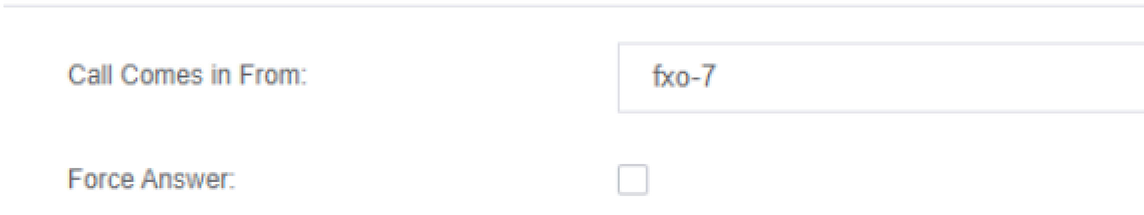
Regarding other parameters in SIP, please set them according to your requirements, because there is no need to set them in a simple call.

Step 5: Set routing rules in the network

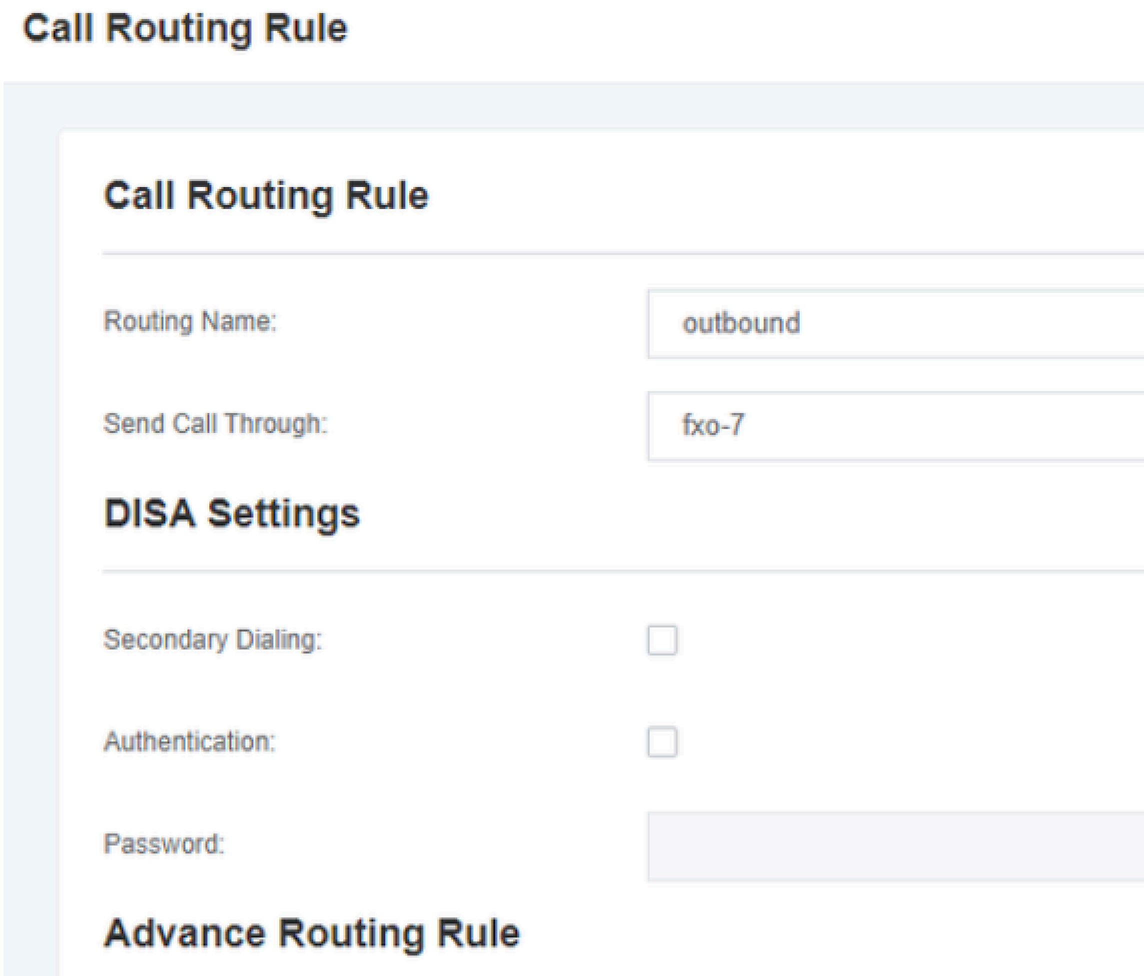
Set routing rules for outbound and inbound calls as shown below.

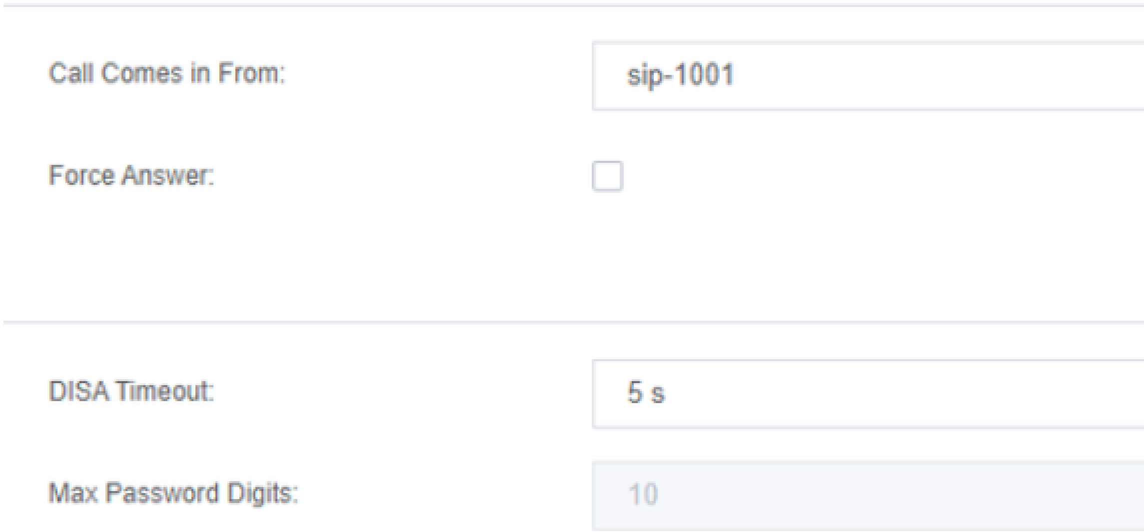
Inbound routing rule





Outbound routing rules

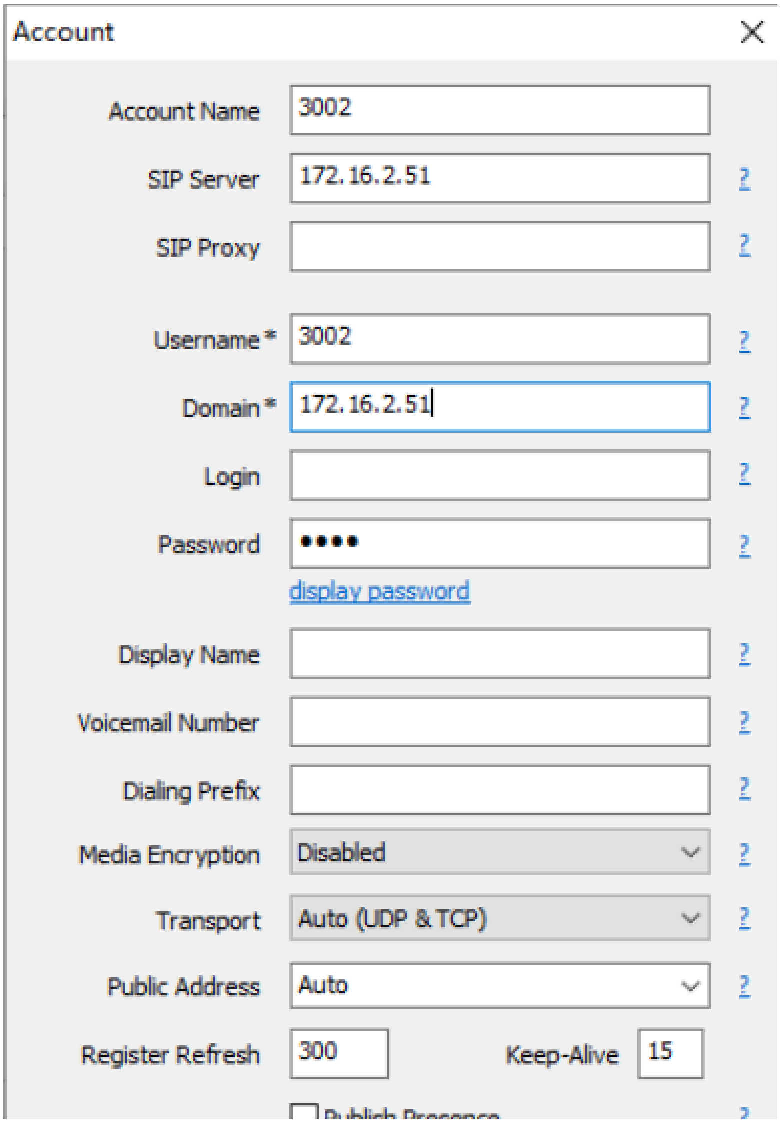




Please save all your changes to take effect.

Step 6: Register a SIP extension through the software

Use SIP software, such as Xlite, eyeBeam, to register a SIP extension (3001). After all the above steps, you can try making calls and sending text messages.



Use SIP software, such as Xlite, eyeBeam, to register a SIP extension (3002).

test call

Incoming call test: Dial the number of port 1 on the gateway through your mobile phone to see if 3002 will ring. If 3002 rings, it means your configuration is successful; otherwise, it means there is a problem with your configuration, please check the configuration.

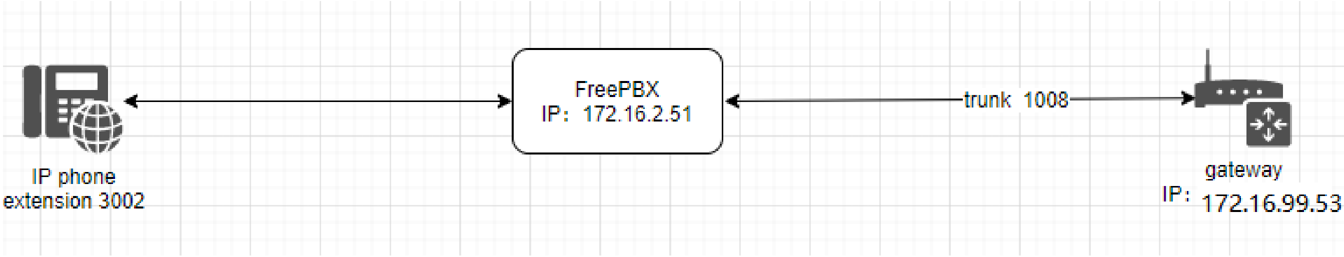
Outbound call test: Dial your mobile phone number on the 3002 extension registered in the software phone. If your phone rings, your configuration is OK; otherwise, please check your configuration.

6.2.2.3 AIU-8 FXO service board connected to FreePBX 15

This document mainly introduces the detailed steps for connecting the business board to Freepbx.

Follow the steps below to configure two-way calling between the phone and the gateway.

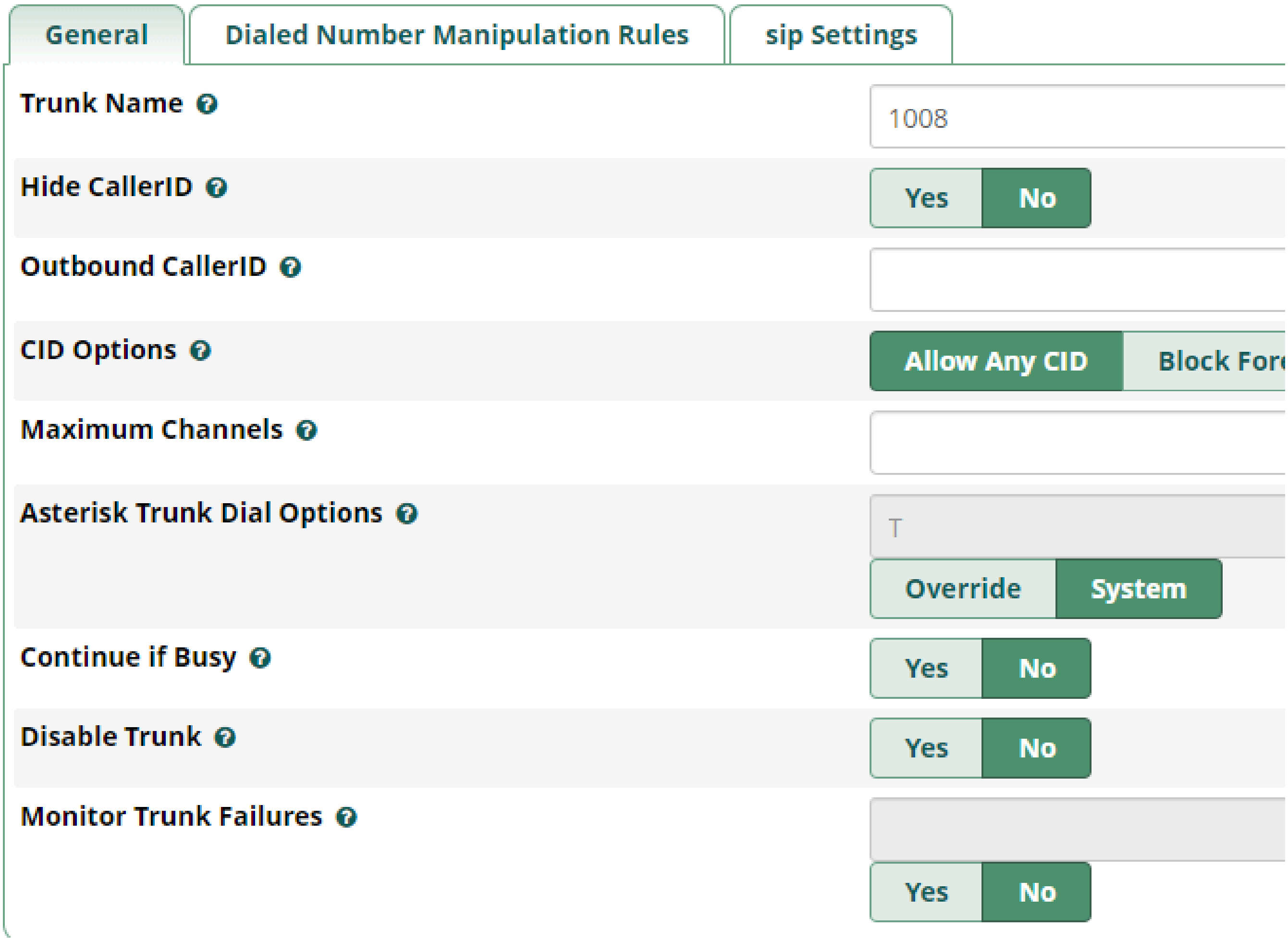
- Outbound calls: from Freepbx's SIP extension 3002 through trunk 1008 to the gateway.
- Incoming call: from outside call to gateway, through SIP trunk 1008 to Freepbx, then send the call through Freepbx to 3002 SIP extension.

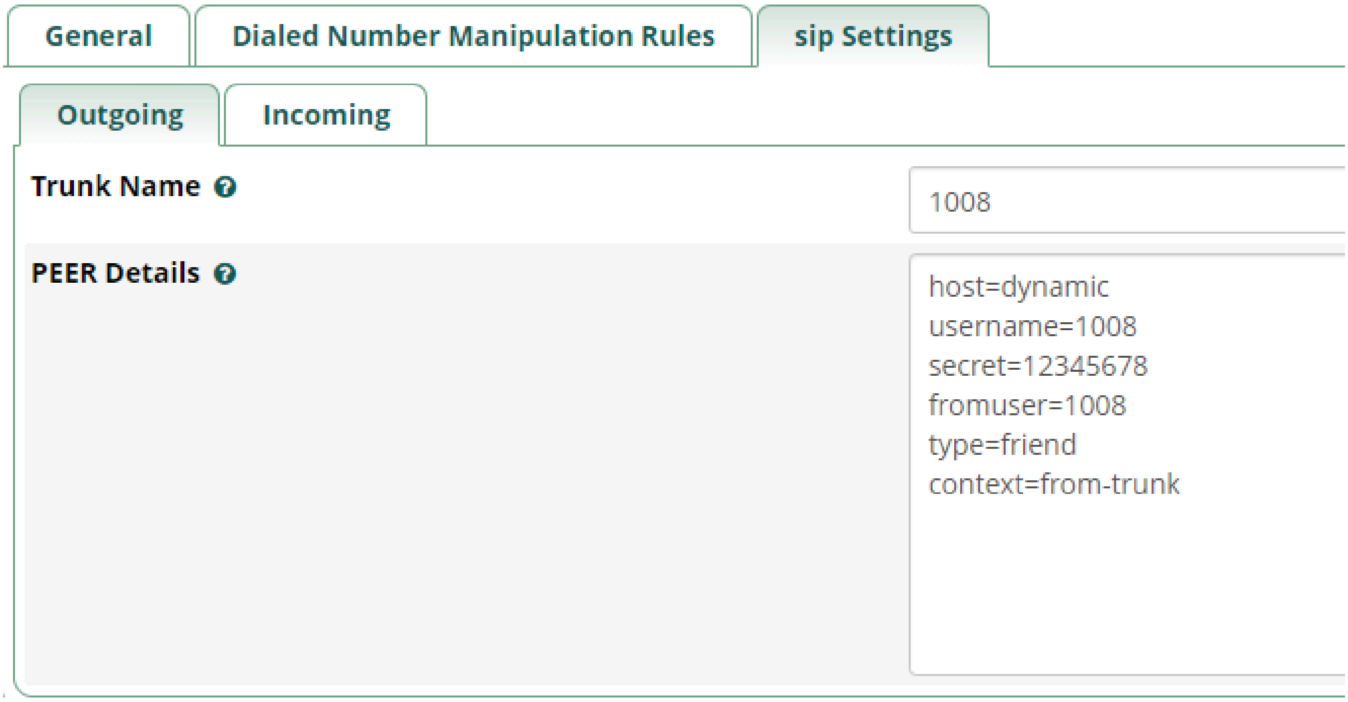


In the following steps, the following parameters must be configured, and other parameters can be configured according to your needs.

Step 1: Create a SIP trunk in Freepbx server

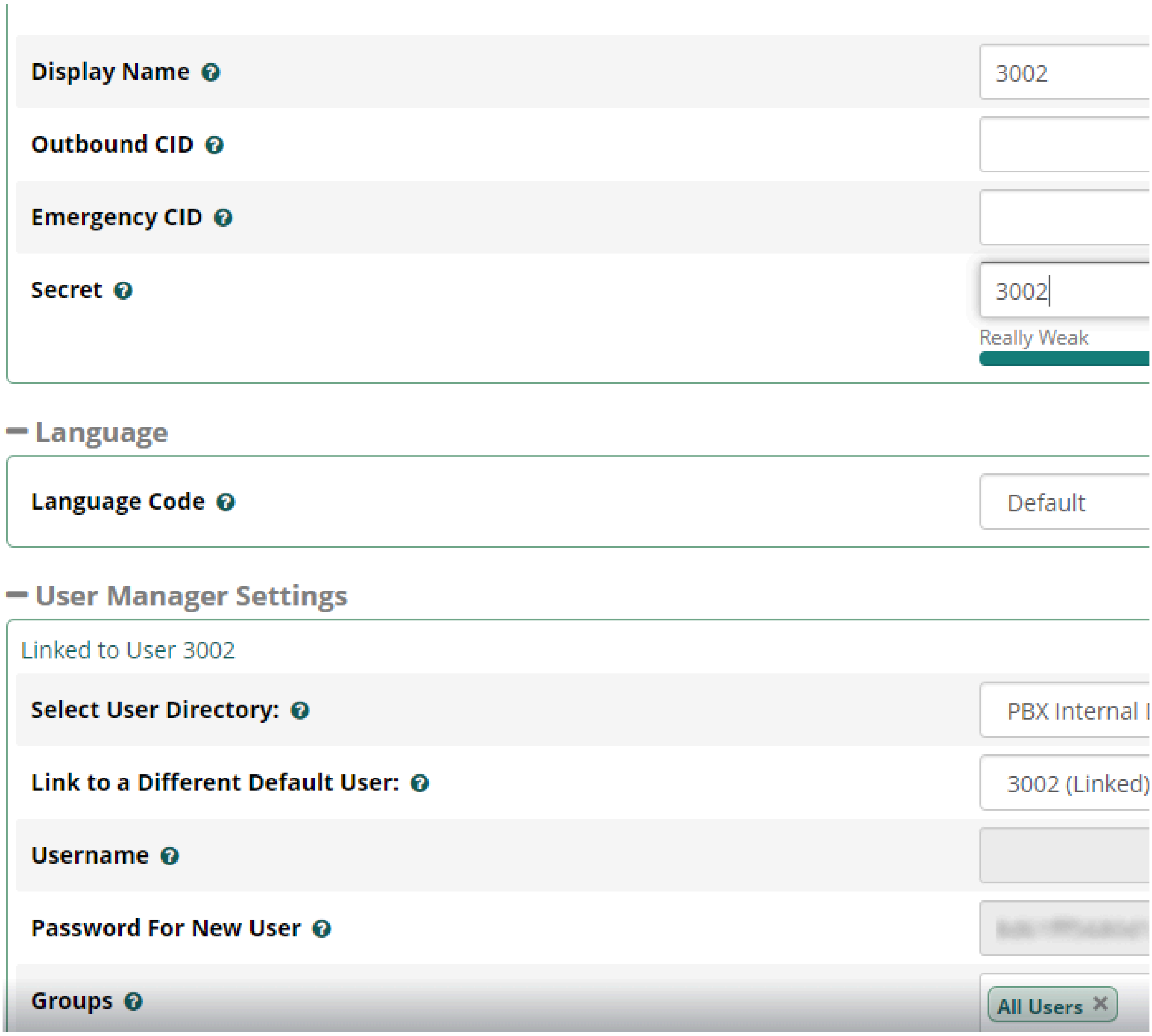
Please log in to your Freepbx server and create a SIP trunk (1008). On the Freepbx server web page, please select "Connectivity->Trunks->Add SIP (chan_sip)Trunk " to set it up, like this.





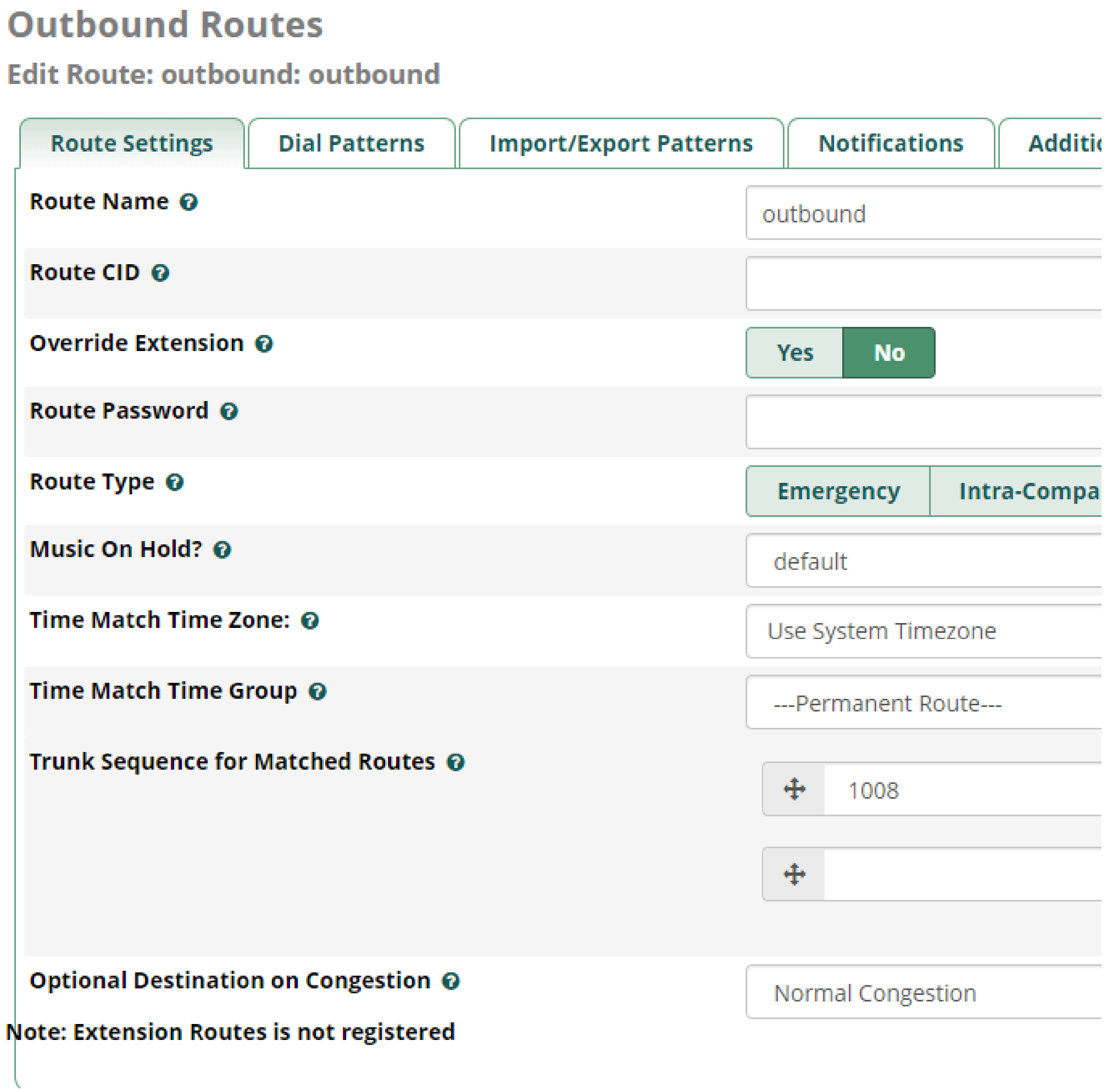
Step 2: Create a sip extension named 3002

Create a sip extension named 3002 in the web page "Application---->Extension".

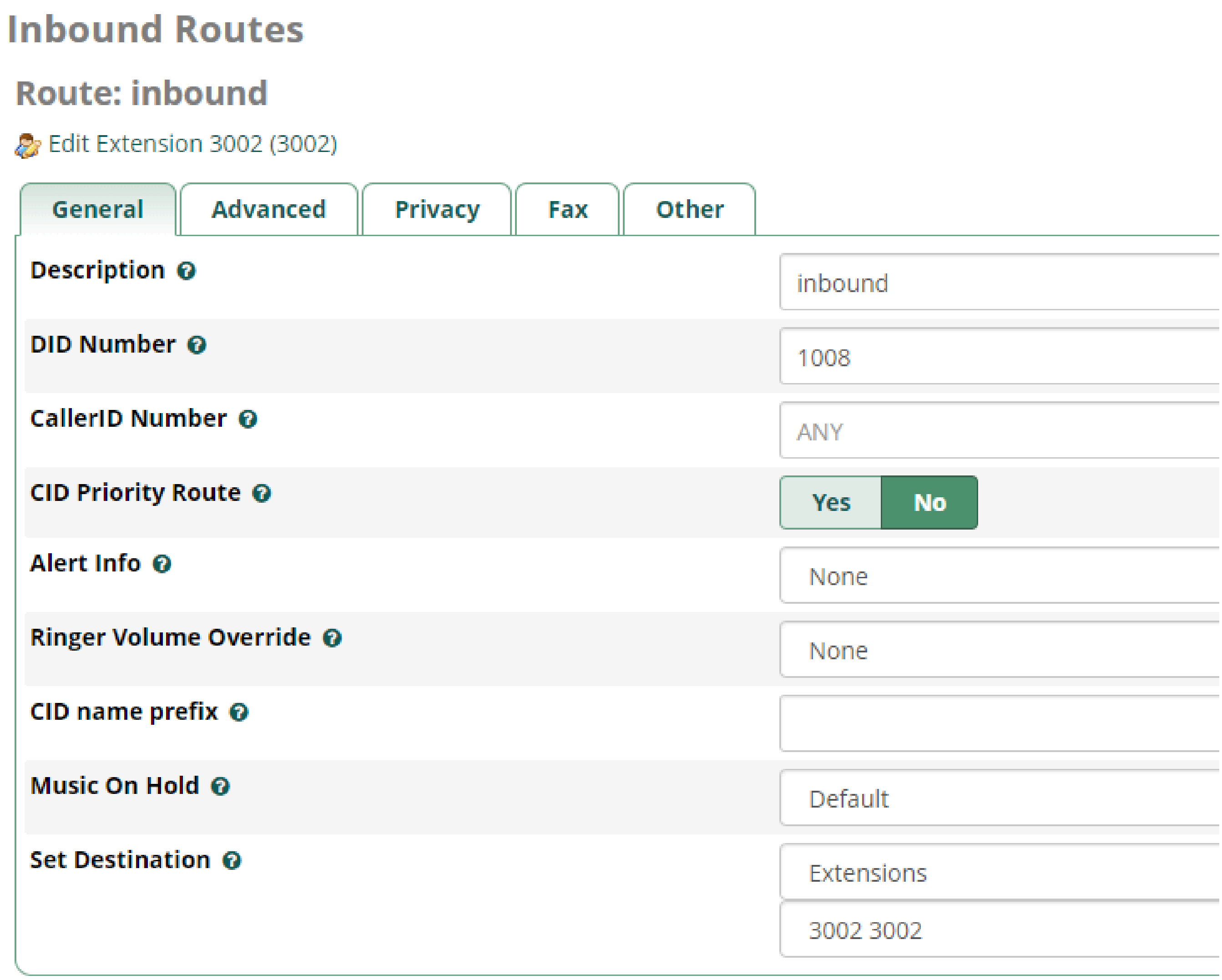


Step 3: Configure routing in Freepbx

Outbound routing rules:

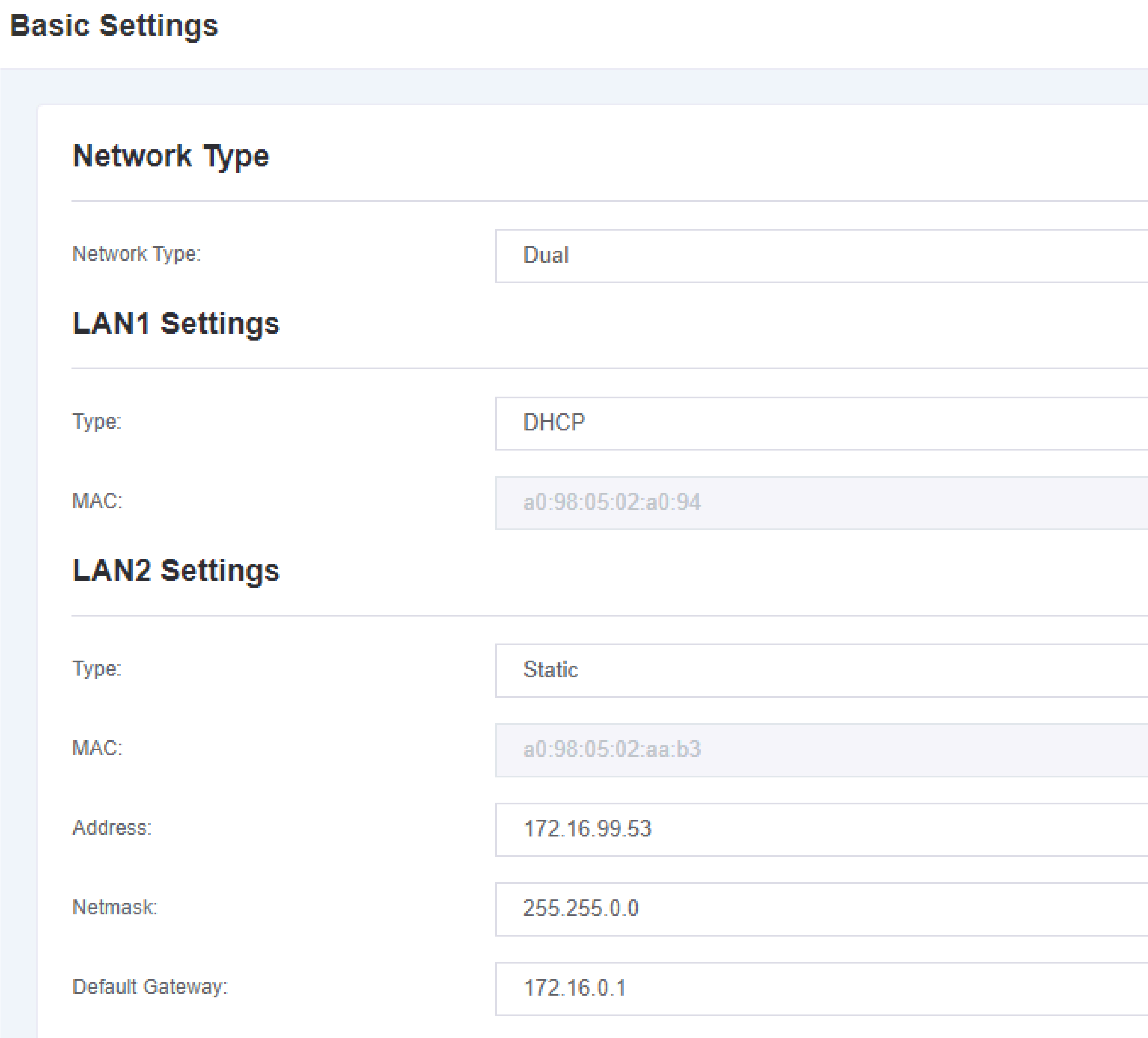


Inbound routing rules.



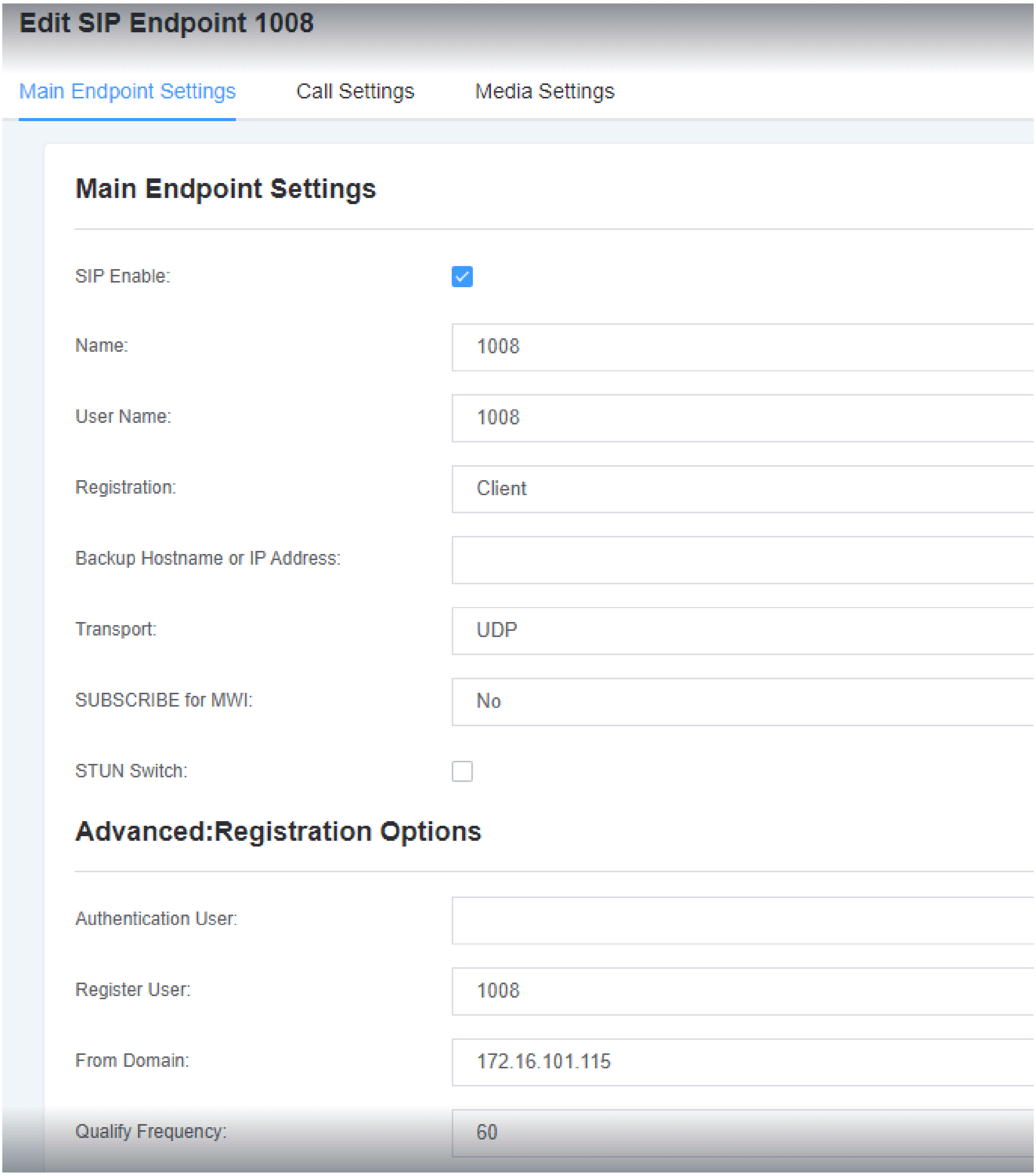
Step 4: Set network parameters in the network

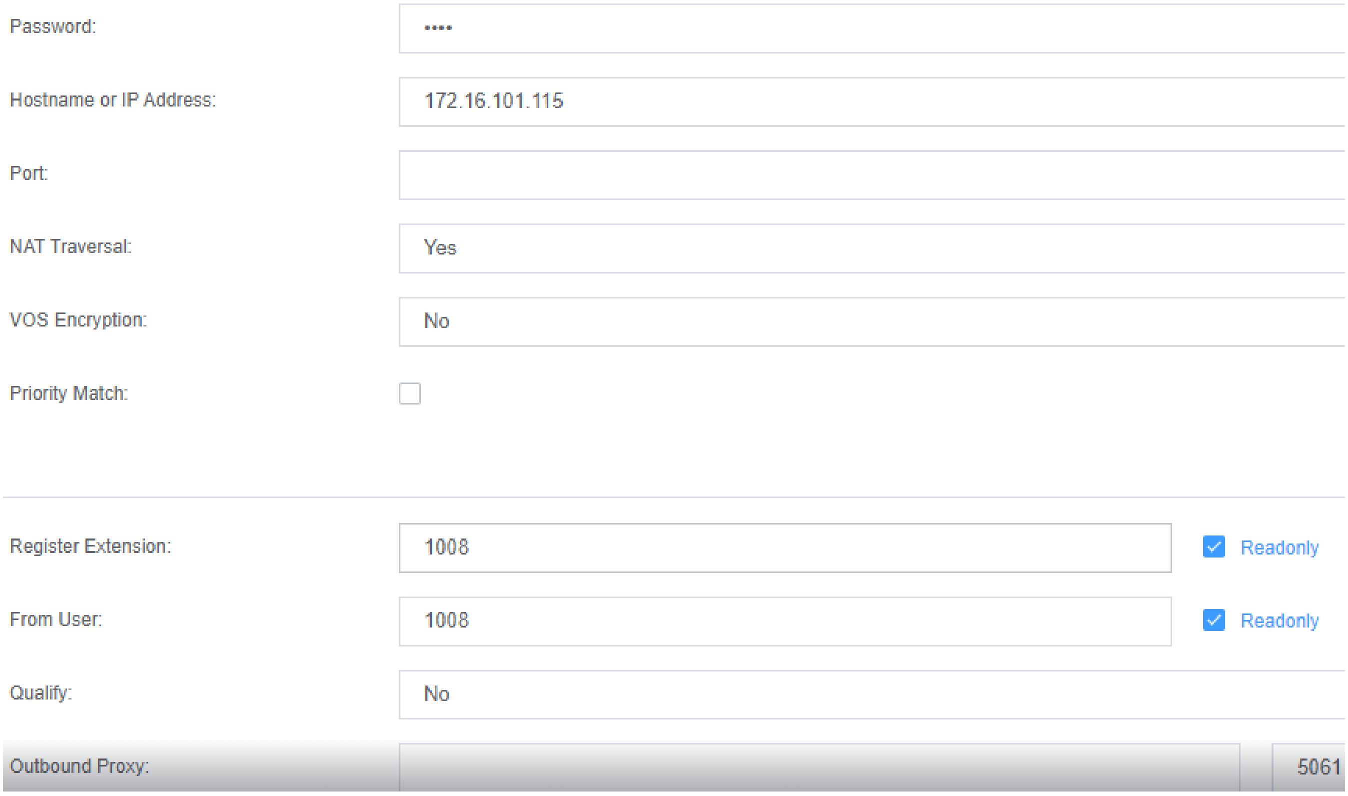
Log in to the network in the browser and set network parameters. The picture below is an example for reference only.



Step 5: Create a SIP endpoint in the network

Please set up SIP trunks. The image below shows details on how to set it up.


Edit SIP Endpoint 1008
Main Endpoint Settings
Call Settings
Media Settings
Main Endpoint Settings
SIP Enable:
Name:
1008
User Name:
1008
Registration:
Client
Backup Hostname or IP Address:
Transport:
UDP
SUBSCRIBE for MWI:
No
STUN Switch:
Advanced:Registration Options
Authentication User:
Register User:
1008
From Domain:
172.16.101.115
Qualify Frequency:
60

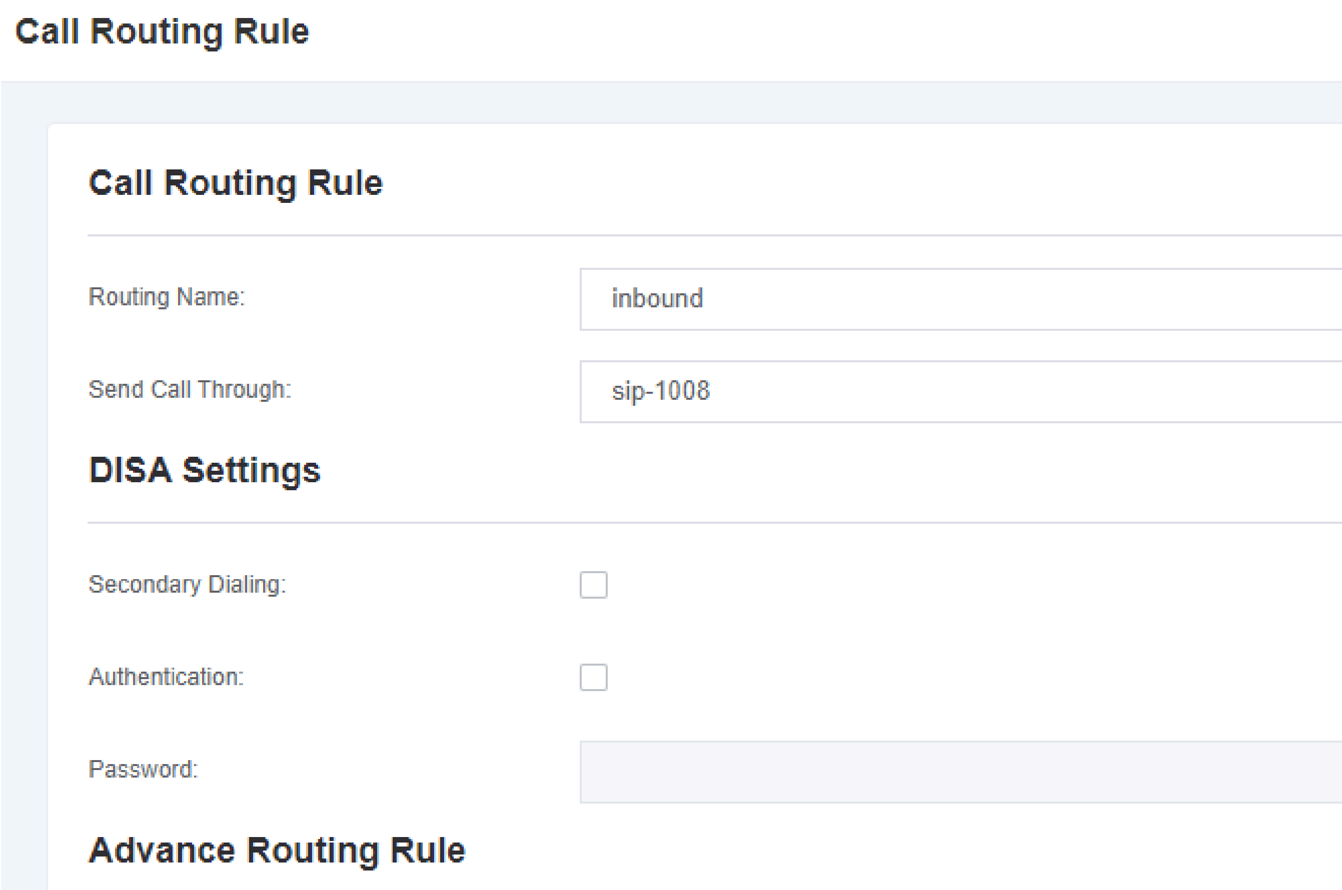


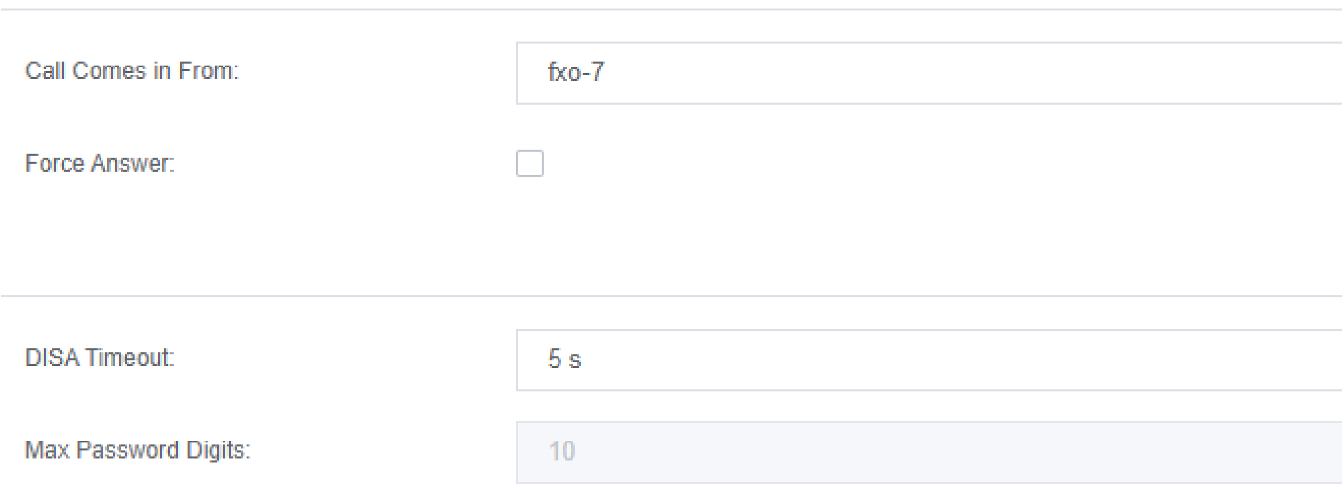
Regarding other parameters in SIP, please set them according to your own needs. There is no need to set them in a simple call.

Step 6: Set routing rules in the web page

Set up outbound and inbound routing rules as shown below.

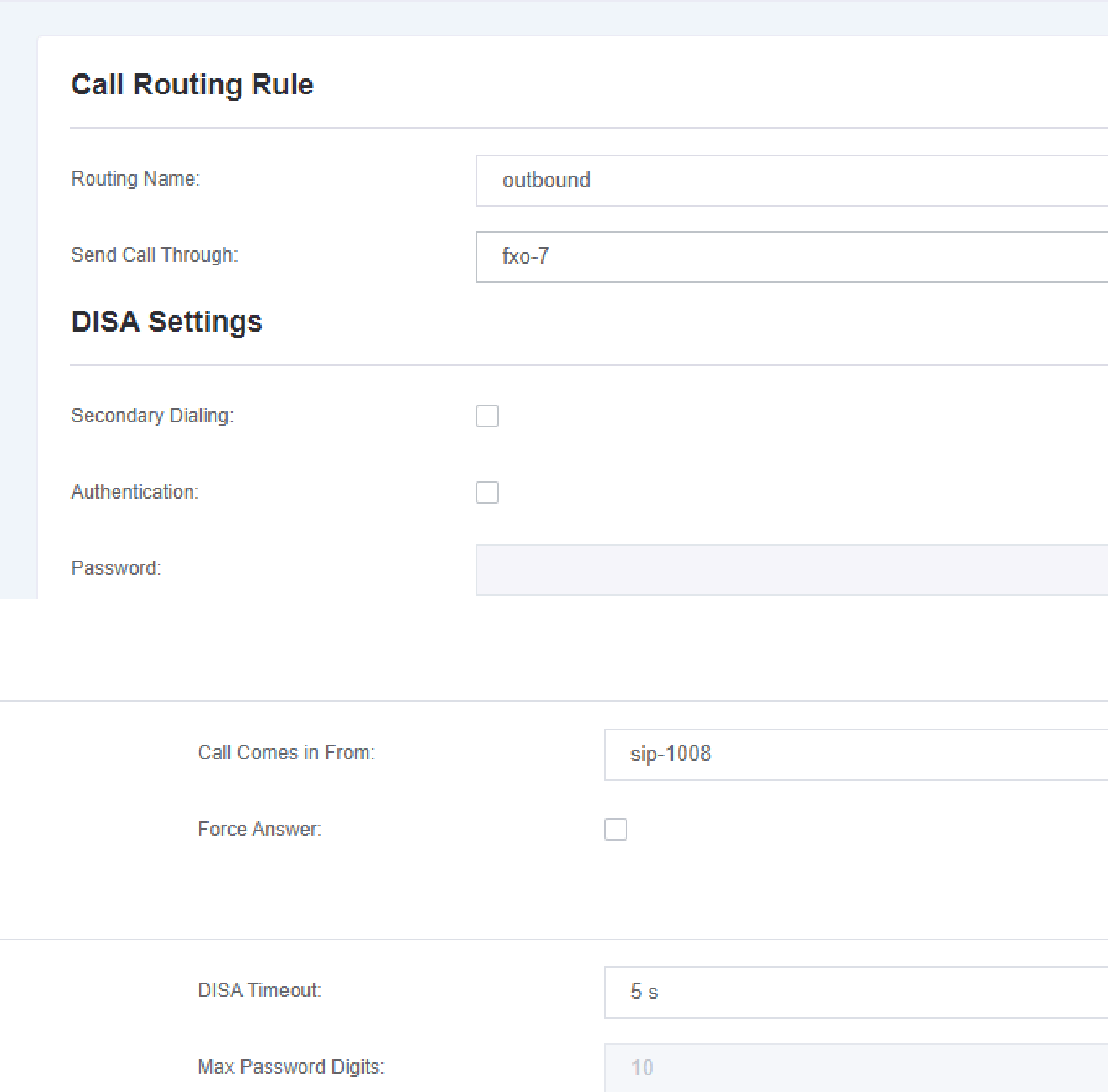
Inbound routing rules.





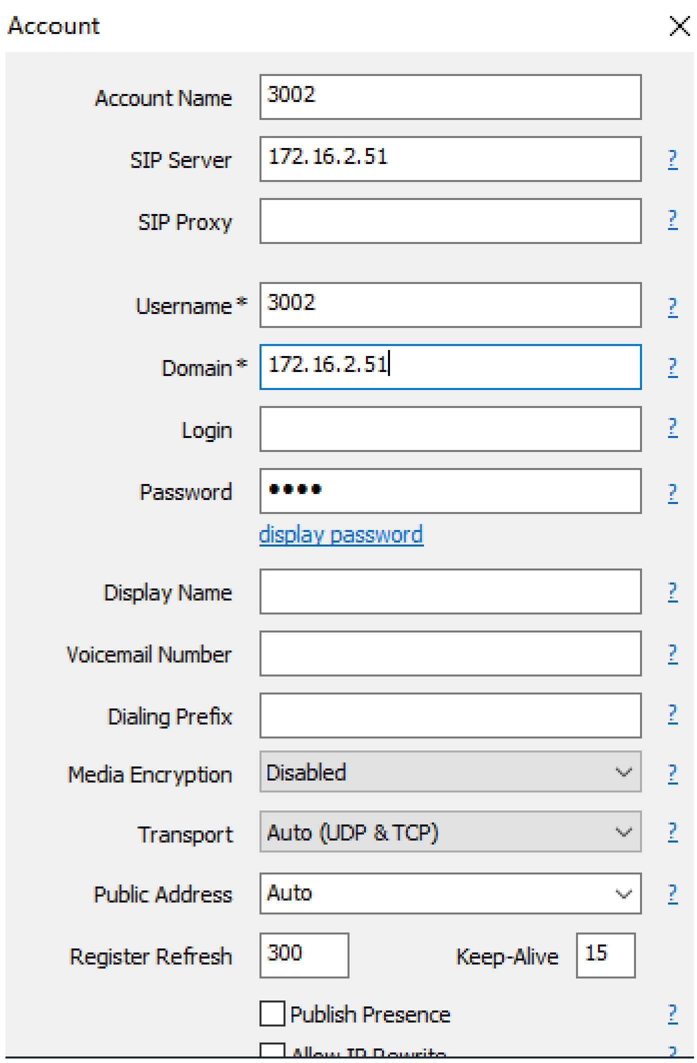
Outbound routing rules.

Call Routing Rule



Step 7: Register a SIP extension through the software

Take advantage of SIP software, such as Xlite, eyeBeam, to register a SIP extension (3002).



Use SIP software, such as Xlite, eyeBeam, to register a SIP extension (3002).

TEST CALL

Incoming call test: Dial the number of port 1 on the gateway through your mobile phone to see if 3002 will ring. If 3002 rings, it means your configuration is successful; otherwise, it means there is a problem with your configuration, please check the configuration.

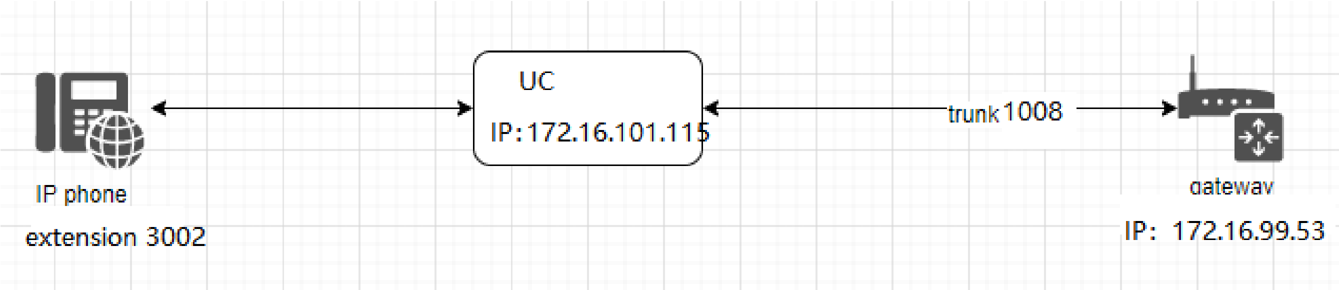
Outbound call test: Dial your mobile phone number on the 3002 extension registered in the software phone. If your phone rings, your configuration is OK; otherwise, please check your configuration.

6.2.2.4 AIU-8 FXO service board connected to OpenVox UC

This article mainly introduces the detailed steps for connecting the service board and UC.

Follow the steps below to configure two-way calling between the phone and the gateway.

- Outbound calls: from UC SIP extension 3002 to the gateway through trunk 1008.
- Incoming call: from outside call to gateway, through SIP trunk 1008 to UC, then send the call through UC to 3002 SIP extension.

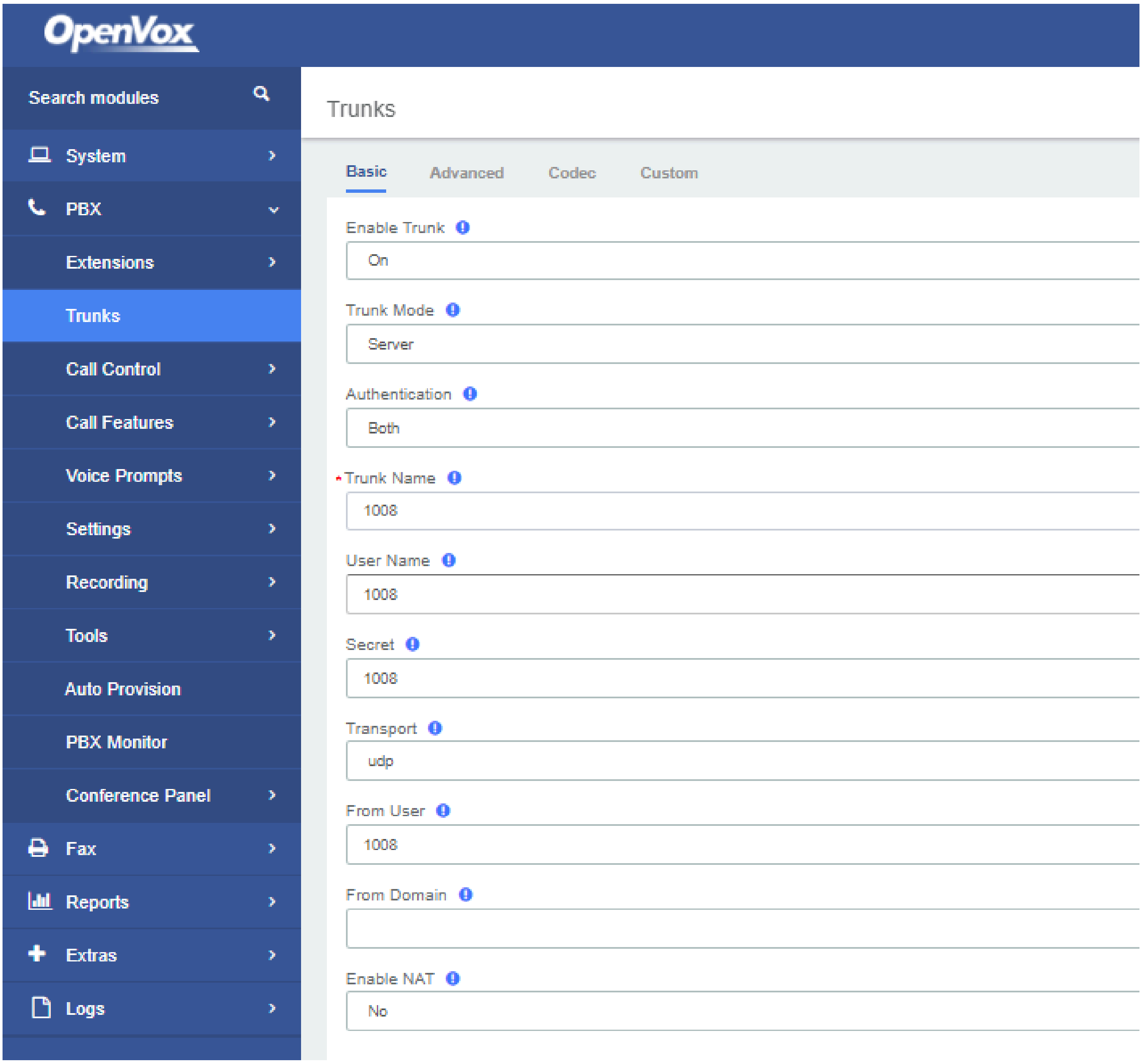


In the following steps, the following parameters are mandatory configurations, other parameters can be configured according to your needs.

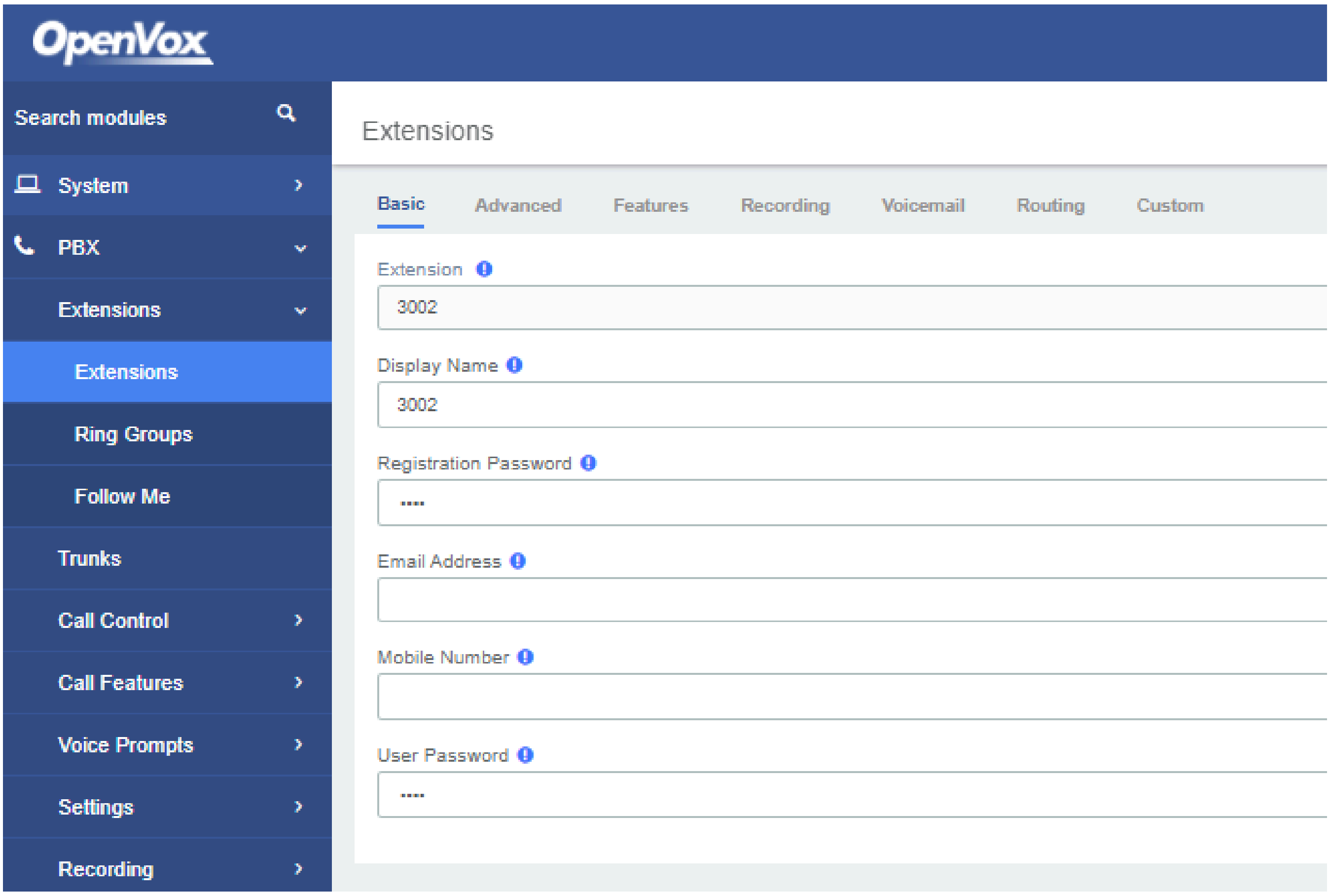
Step 1: Create a SIP Trunk in UC Server

Please create a SIP trunk (1008) and a SIP extension (3002).

trunk 1008

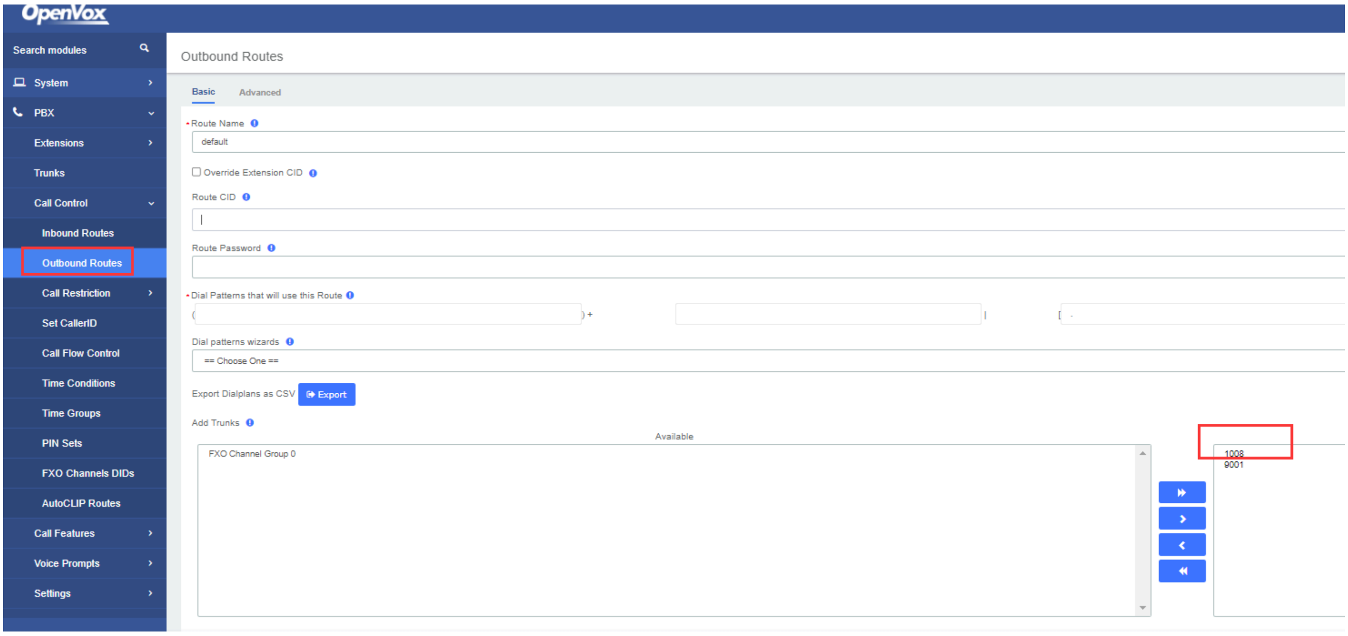


Extension 3002

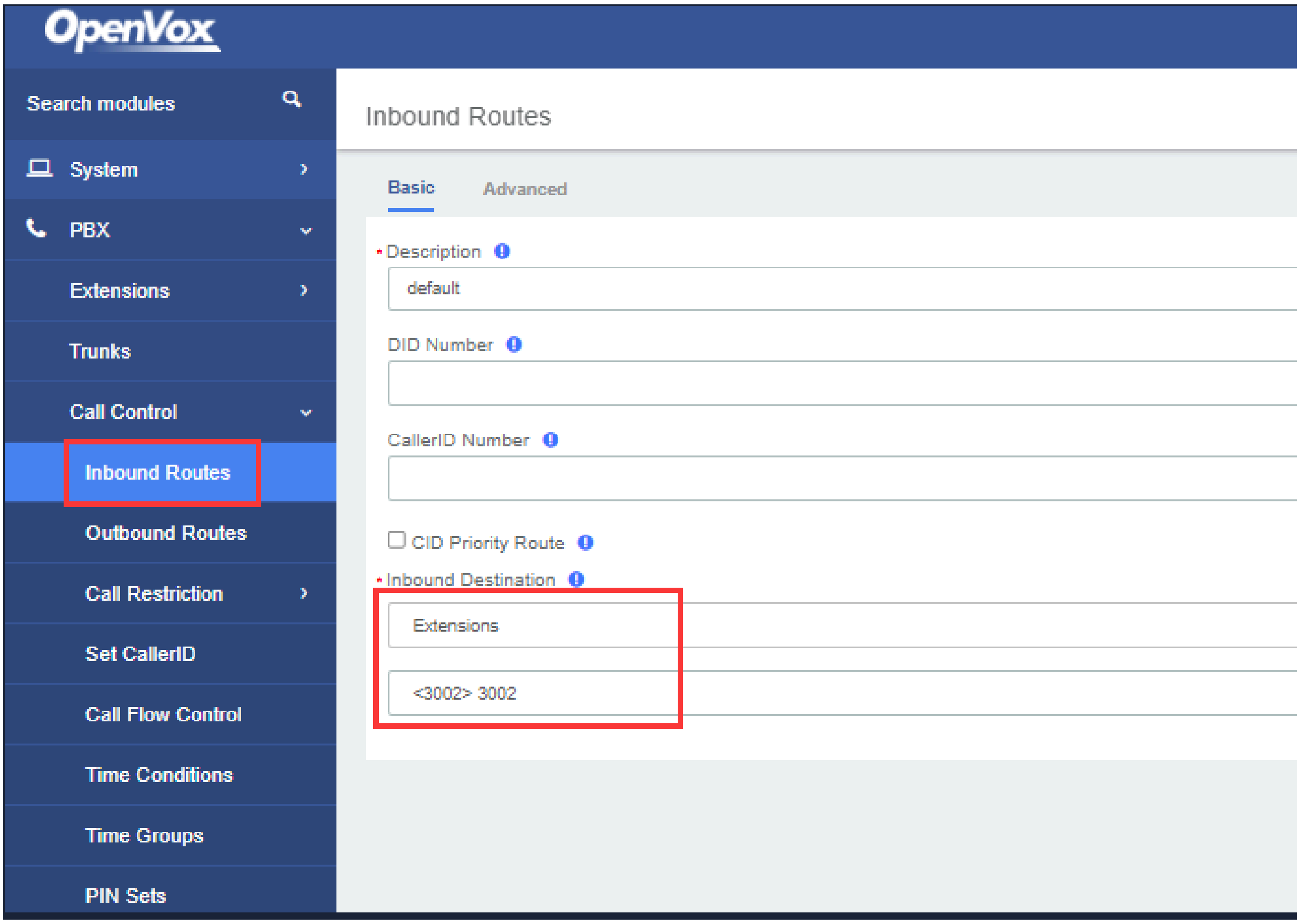


Step 2: Edit Dial Rules in UC

Outbound call routing rules

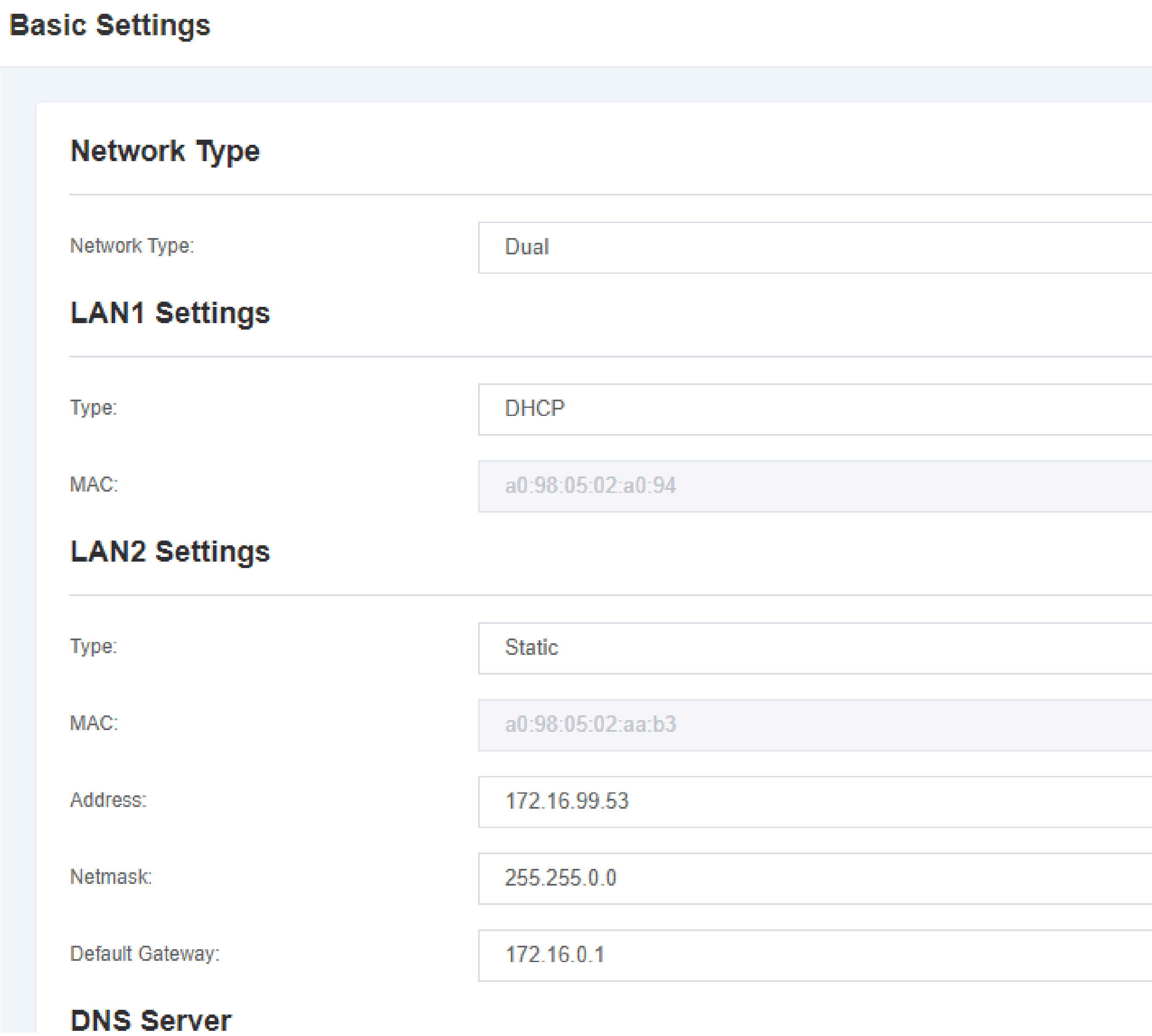


Inbound routing rules



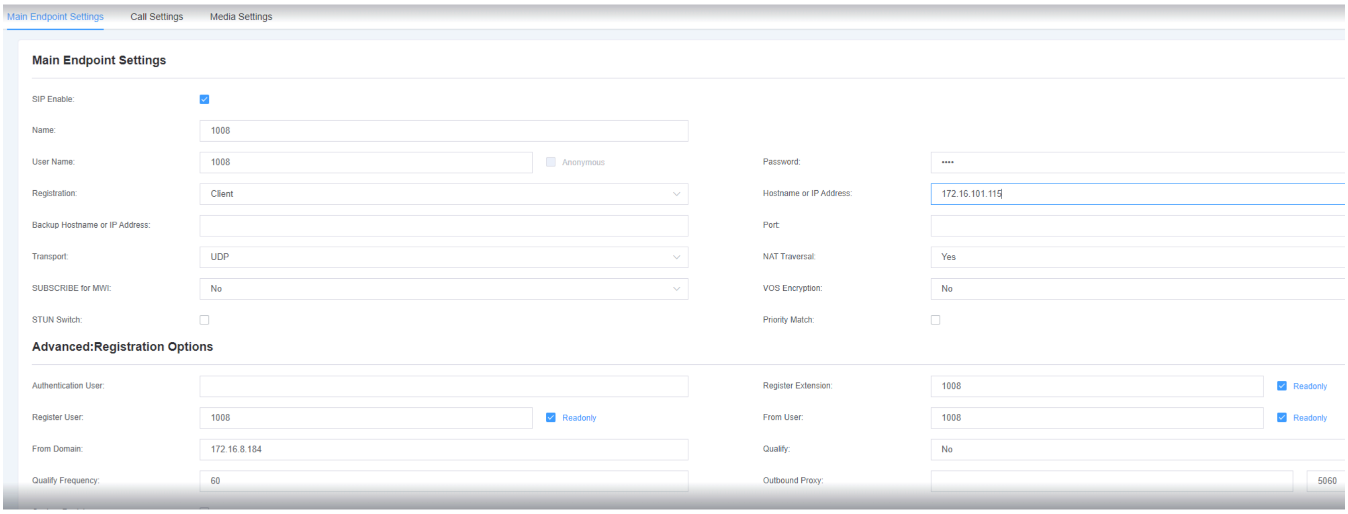
Step 3: Set network parameters in the network

Log in to the network in the browser and click to set network parameters. The picture below is an example and is for reference only.



Step 4: Create a SIP endpoint in the network

Please set up SIP trunk. The image below shows details on how to set it up.

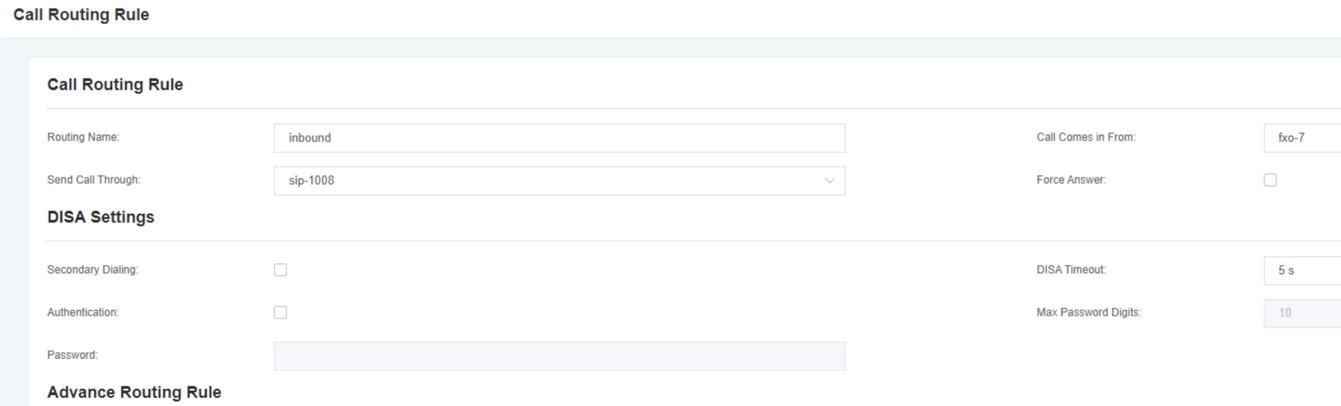


Regarding other parameters in SIP, please set them according to your requirements, because there is no need to set them in a simple call.

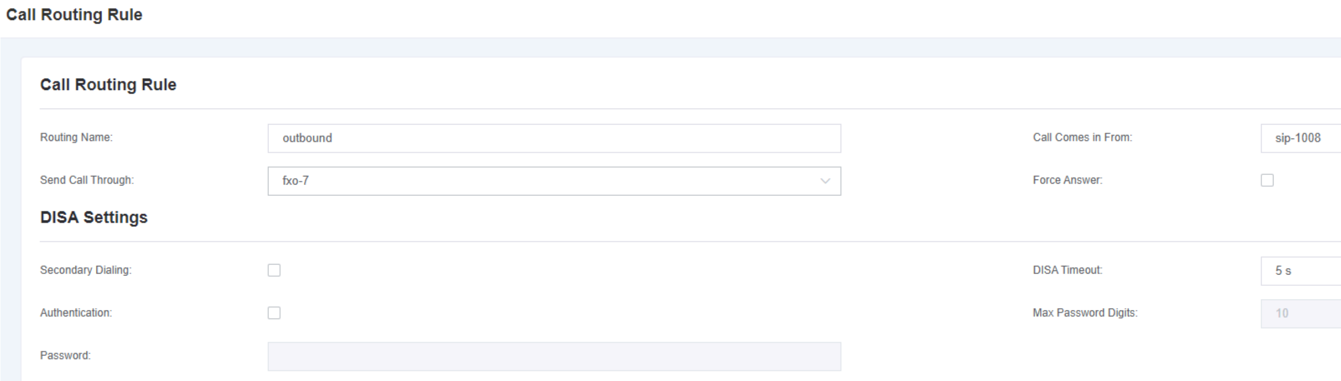
Step 5: Set routing rules in the network

Set routing rules for outbound and inbound calls as shown below.

Inbound routing rule:

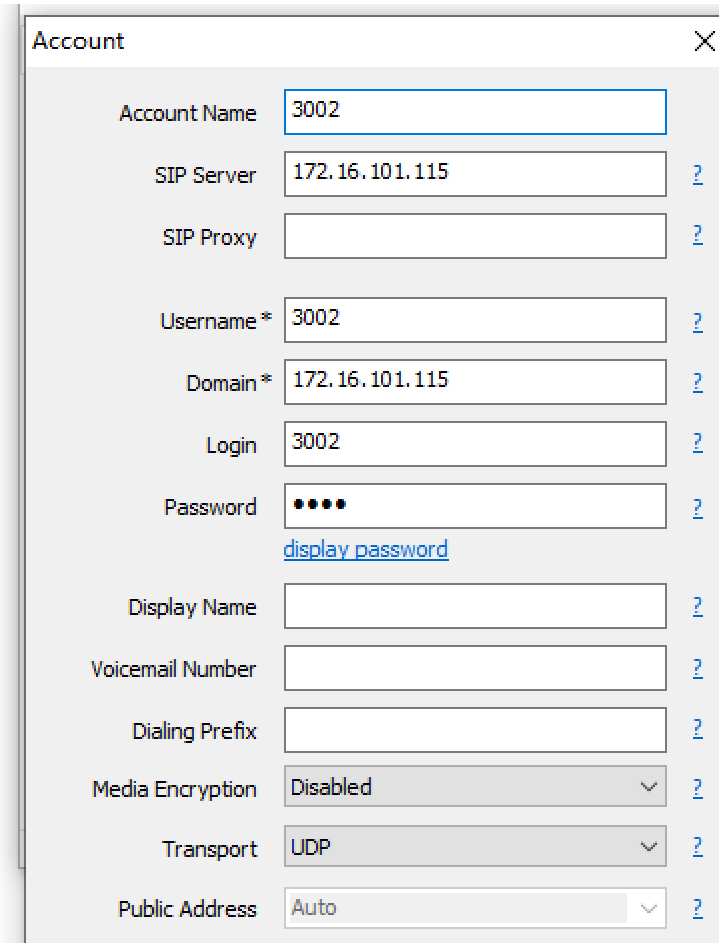


Outbound routing rule:



Please save all your changes to take effect.

Step 6: Register a SIP extension through the software



Use SIP software, such as Xlite, eyeBeam, to register a SIP extension (3002).

TEST CALL

Incoming call test: Dial the number of port 1 on the gateway through your mobile phone to see if 3002 will ring. If 3002 rings, it means your configuration is successful; otherwise, it means there is a problem with your configuration, please check the configuration.

Outbound call test: Dial your mobile phone number on the 3002 extension registered in the software phone. If your phone rings, your configuration is OK; otherwise, please check your configuration.

7. User manual

This section is a summary of user manual links for each business board:

7.1 MIU Magnet Gateway Business Board User Manual

MIU Magnet Gateway Business Board User Manual

7.2 AIU Analog Service Board User Manual

AIU Analog Service Board User Manual

7.3 DTU Digital Service Board User Manual

DTU Digital Service Board User Manual

7.4 ACU Audio Broadcast Service Board User Manual

ACU Audio Broadcast Service Board User Manual

7.5 RIU Wireless Trunking Service Board User Manual

RIU Wireless Trunking Service Board User Manual