

NEXO IP PBX 1024E1

MANUAL DEL USUARIO v1.0
User Manual



Preface

Welcome

Thanks for choosing the **NEXO IP PBX 1024E1**. We hope you will make full use of this rich-feature gateway. Contact us if you need any technical support: 0341 4820400

About This Manual

This manual provides information about the introduction of the IP PBX, and about how to **install**, configure or use the gateway. Please read this document carefully before **install** the IP PBX.

Intended Audience

This manual is aimed primarily at the following people:

- Users
- Engineers who install, configure and maintain the IP PBX.

Revision Record

| Document Name | Document Version | Firmware Version |
|------------------|------------------|------------------|
| User Manual V1.0 | V1.0 | |

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Product Introduction

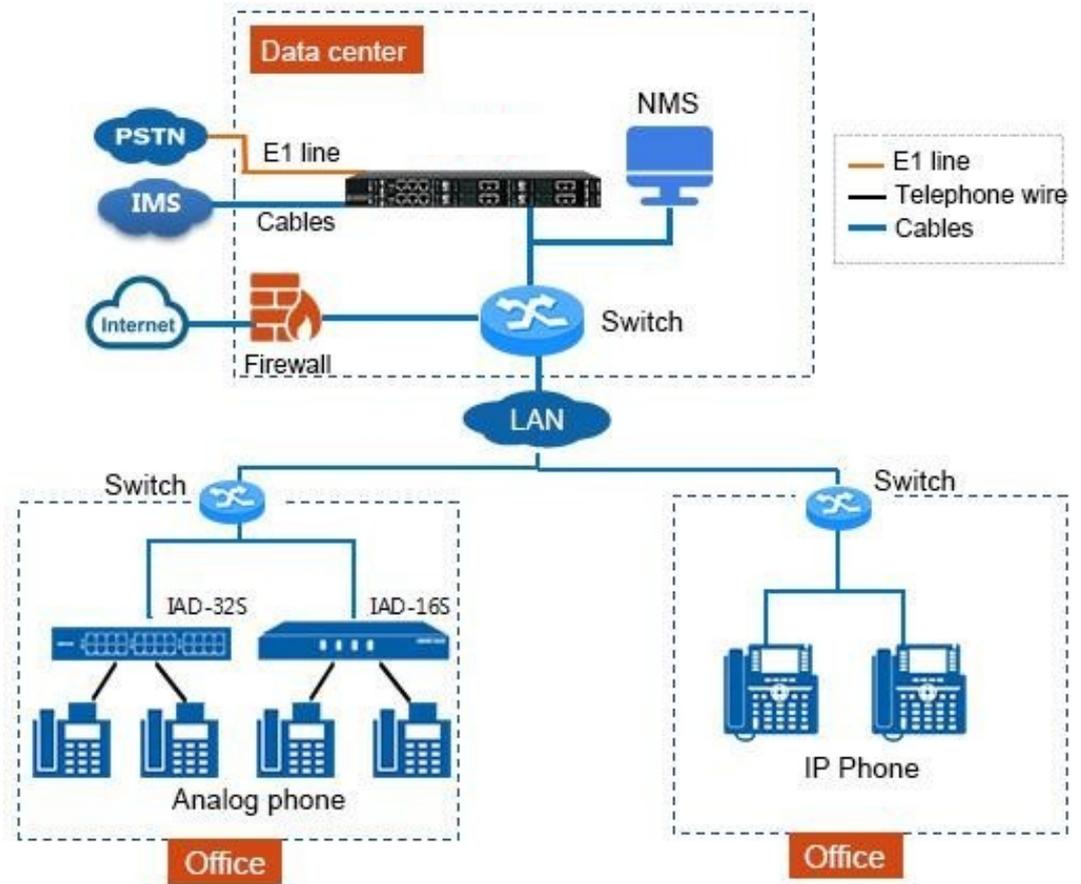
1.1 Overview

NEXO IP PBX 1024E1 is a new generation of large capacity unified communication. Based on a powerful hardware platform, It supports 1000 extensions and 200 concurrent calls which is integrated voice, video, paging, fax, conference, recording and other useful functions. It also provides four slots, which can be hot-plugged to install E1/T1 boards, FXS, and FXO boards, and can be flexibly configured and combined according to actual usage scenarios. It is not only suitable for helping to build the telephone system of large and medium-sized enterprises, but also to meet the needs of branch offices of large group enterprises and government agencies, and help enterprises and industry customers to establish convenient and efficient IP telephone system.

In addition, NEXO 1024E1 provides an intuitive and easy-to-use graphical interface for user management and maintenance. With the open API, users can easily integrate with the third-party CRM systems and dispatching platform to provide enterprises with convenient and intelligent one-stop telephone solutions.

1.2 Application Scenario

The application scenario of NEXO 1024E1 universal gateway is shown as follows: Figure 1- 1 Application Scenario of NEXO 1024E1

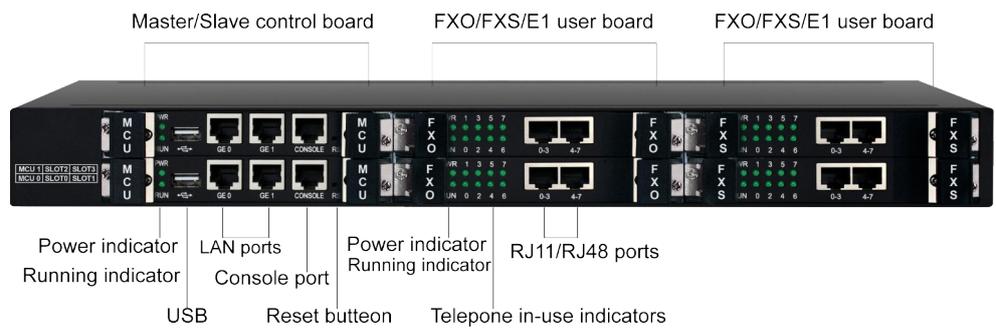


1.3 Product Appearance

Front View:



Back View:



Note:

The installation of three different types of user boards (FXO/FXS/E1) is supported. If the installation type is E1 user board, the RJ48 ports are connected to the E1 line; if the installation type is FXS user board, the RJ11 ports are connected to the telephone wire; if the installation type is FXO, the RJ11 ports are connected to the PSTN line.

| No. | Definition | Description |
|-----|--------------|---|
| 1 | Power switch | Power switch button |
| 2 | Power jack | Power interface: 100~240VAC, 50~60HZ |
| 3 | USB | Provide an outgoing USB interface, through which USB devices can be connected |
| 4 | Console port | Console port for debugging and configuring the device |
| 5 | CE0/CE1 | Connect to IP network via DSL modem or router or LAN switch |

1.4 Description of User Boards

| Type | Indicator | Definition | Status | Description |
|-----------------------------------|-----------|-------------------|---------------|--|
| Master/Slave control board | PWR | Power Indicator | Off | There is no power supply or power supply is abnormal. |
| | | | On | The device is powered on. |
| | RUN | Running Indicator | Flashing | The device is initialized successfully and is running normally |
| | | | On | The system is initializing. |
| | | | Off | The device is not running normally. |
| FXS/FXO user board | PWR | Power Indicator | On | The power supply is normal |
| | | | Off | The power supply is not normal |
| | RUN | Running Indicator | Off | The system is starting up |
| | | | Fast Flashing | Successful port registration |
| | | | Slow Flashing | All ports are registered |
| | | | On | Press the reset button to restore |
| | FXS/FXO | FXS/FXO Indicator | On | The FXS port is in off-hook (in-use) status. |
| | | | Off | The FXS port is in on-hook status. |

| | | | | |
|----------------------|-----|-------------------|---------------|--|
| E1 user board | PWR | Power Indicator | Off | There is no power supply or power supply is abnormal. |
| | | | On | The device is powered on. |
| | RUN | Running Indicator | Slow Flashing | The device is initialized successfully and is running normally |
| | | | On | The system is initializing. |
| | | | Off | The device is not running normally. |
| | E1 | E1 Indicator | On | E1 line is connected |
| | | | Off | E1 line E1 is disconnected |

1.5 Features & Functions

1.5.1 Key Features

- Supports up to 1000 SIP users and 200 concurrent calls
- Supports E1/T1, FXO and FXS ports with the flexible and alternative capability
- Distributed multi-core CPU, greatly improves the call processing capacity
- Flexible dial rules based on time, number or source IP etc.
- Supports Multi-level IVR, helps to build personalized voice navigation for enterprise
- Built-in VPN server/client
- Support voicemail/ Voice recording
- User-friendly web interface, classification of web user's rights

1.5.2 FXS

- Connector: RJ45
- Caller ID: Bellcore Type 1&2, ETSI,BT,NTT and DTMF
- Answer and Disconnect Signaling: Answer, Disconnect, Busy Tone
- Polarity Reversal
- Hook Flash

1.5.3 FXO

- Connector: RJ45
- Caller ID: FSK, DTMF
- Polarity Reversal
- Answer Delay

- Detection of Busy Tone
- Detection of No Current
- Auto Match of FXO Impedance

1.5.4 PSTN

- 4* E1/T1 Ports at max
- Interface: RJ48 (120Ohm)
- ISDN PRI:
- 23B+D(T1),30B+D(E1),NT or TE
- ITU-T Q.921, ITU-T Q.931, Q.Sig
- Signal 7/SS7:
- ITU-T, ANSI · ITU-CHINA
- MTP1/MTP2/MTP3, TUP/ISUP
- R2 MFC

1.5.5 Voice Capabilities

- VoIP Protocols: SIP over UDP/TCP/TLS,RTP/SRTP
- Codecs: G.711,G.723.1, G.729A/B, G.722,G.726, OPUS
- Silence Suppression
- Comfort Noise Generator(CNG)
- Voice Activity Detection(VAD)
- Echo Cancellation: G.168 with up to 128ms
- Dynamic Jitter Buffer
- Adjustable Gain Control
- Automatic Gain Control (AGC)
- Call Progress Tones: Dial Tone, Ring Back Tone, Busy Tone
- FAX: T.38 and Pass-through
- NAT: STUN/UPnP
- DTMF: RFC2833/Signal/Inband
- VPN Server/Client

1.5.6 PBX Services

- Call Forward (Unconditional/No Answer/Busy)
- Call Waiting
- Call Holding

- Call Transfer
- Hotline
- Do-not-disturb
- 3-Way Conference
- Ring Group
- Call Queue
- Routing Groups
- Caller/Called Number Manipulation
- Routing Based on Time Period
- Routing Based on Caller/Called Prefixes
- Routing Based on Source Trunks
- Dial Rules
- Failover Routing
- Multi-level IVR
- Auto-attendant Function
- CDRs
- Voicemail
- Voice Recording
- Up to 1000 SIP Extensions
- Up to 200 Concurrent Calls
- Paging
- Event Report
- Email Client
- Voicemail to Email

1.5.7 Environmental

- 1+1 Power Supply, 100-240VAC, 50-60 Hz
- Power Consumption:50W
- Operating Temperature: 0 °C ~ 45 °C
- Storage Temperature: -20 °C ~80 °C
- Humidity: 10%-90% Non-Condensing
- Dimensions (W/D/H): 437×345×49mm
- Unit Weight: 5.7 kg

1.5.8 Maintenance

- Web GUI Configuration
- Telnet Management
- Configuration Restore/Backup
- Multiple Languages Supported
- HTTP/TFTP/FTP Firmware Upgrade
- Auto Provision
- Syslog
- Ping and Tracert
- Traffic Statistics: TCP,UDP,RTP
- Network Capture
- NTP
- Classification of Web Users' Rights
- HTTP&HTTPS/NATS API

2

Installation Instructions

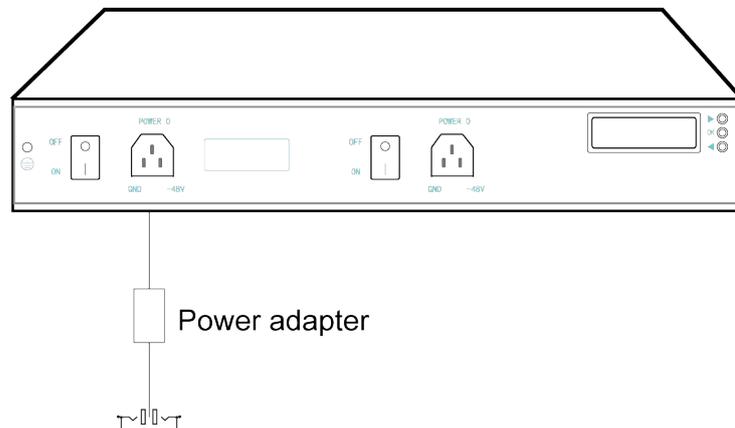
2.1 Installation Attentions

To avoid unexpected accident or device damage, please read the following instructions before you install the NEXO IP PBX 1024E1.

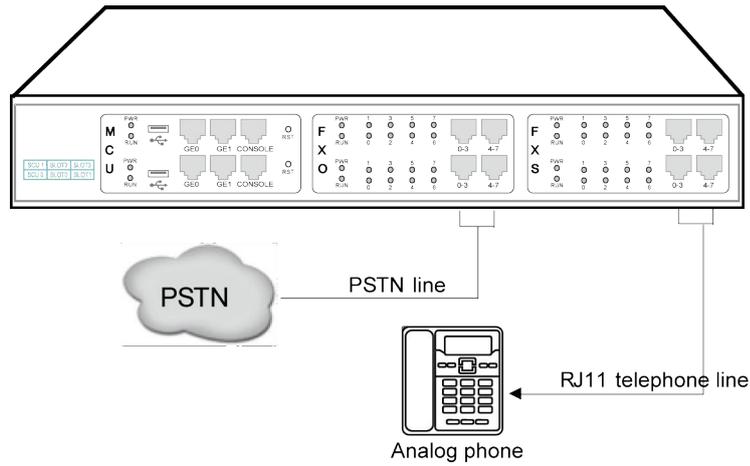
- The adapter of the gateway accepts DC220V 10A dual power input. Please ensure stable and safe power supply;
- To reduce the interference to telephone calls, please separate power cables from telephone lines;
- To guarantee stable running of the gateway, please make sure that there is enough network bandwidth;
- For better heat dissipation, please place the gateway on a level surface and do not pile up with other devices;

2.2 Installation Steps

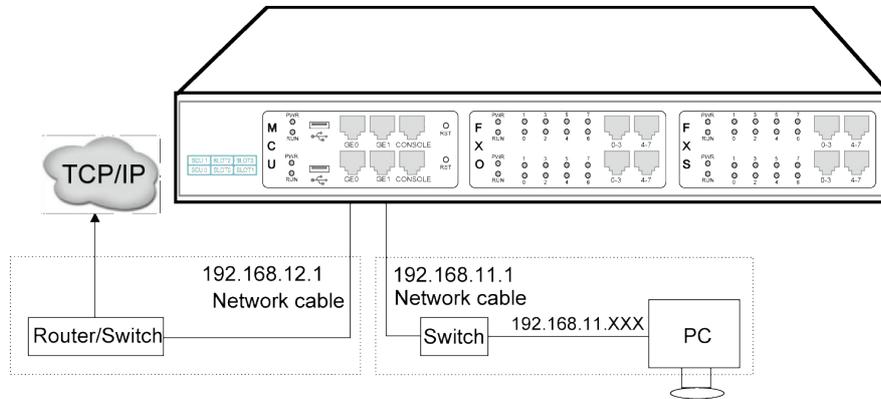
- 1) Connect the power adapter to the power jack;



- 2) Connect telephone line to the FXS port and connect PSTN line to the FXO port;

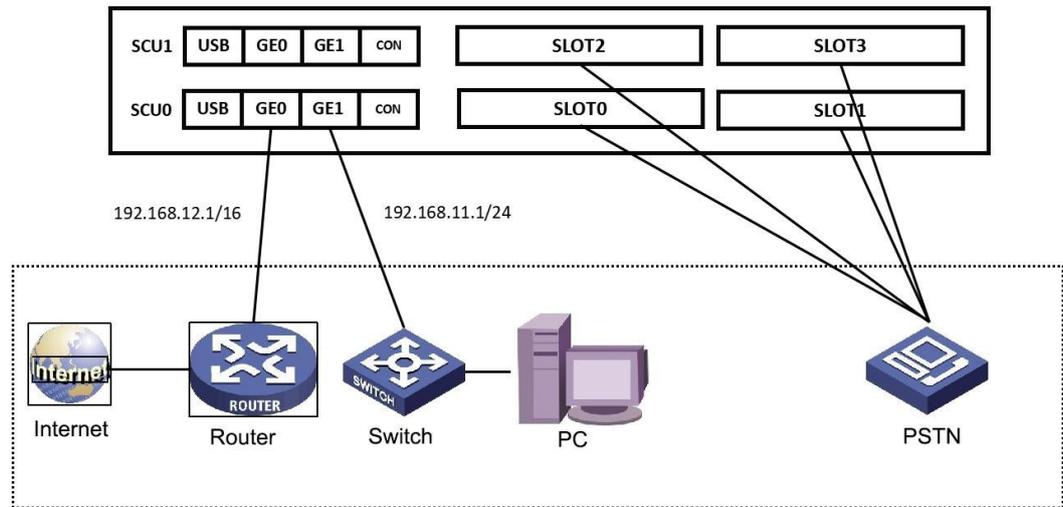


3) Connect network cable to the GE0 port, and the GE1 port is connected to the PC.



2.3 Network Connection

NEXO 1024E1 provides 3 RJ45 ports, of which GE1 port is the management port, which is mainly used for PC access and management equipment. GE0 port is a network service port, which is mainly used to access the network. GE0 port is a static IP address and needs to be configured with an IPv4 address in the same network segment as the uplink, as shown in Figure: 2.3 Network Connection.



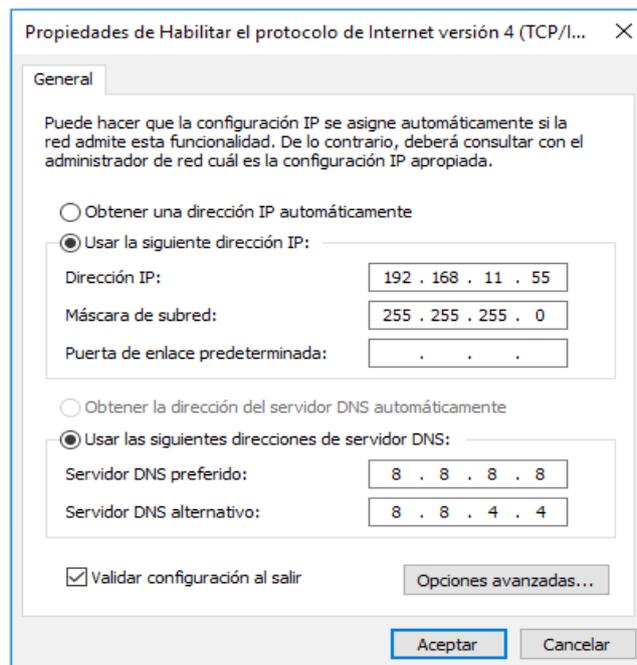
2.4 Connect Gateway to Network

2.4.1 Preparations for Login

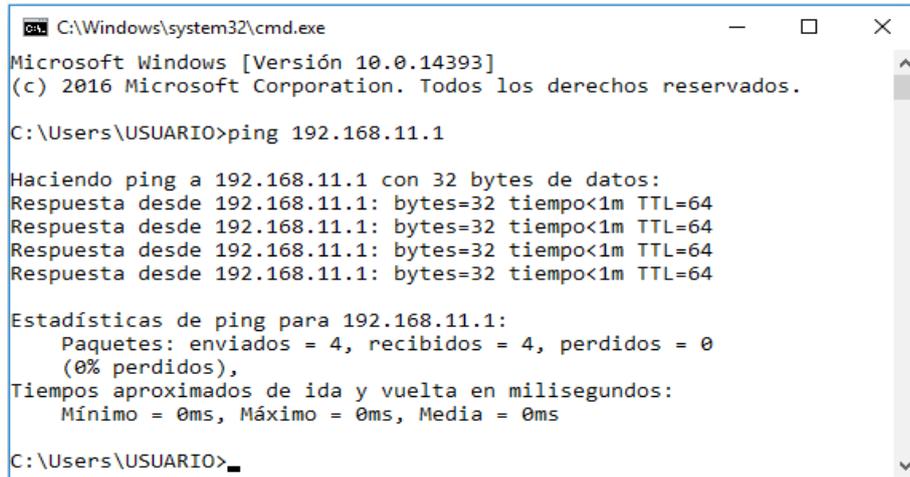
Modify the IP address of the PC to make it at the same network segment with the NEXO 1024E1 gateway, since the default IP address of GE1 port of the gateway is 192.168.11.1.

Take the Windows 7 operating system as an example, set the IP address of the local computer and the gateway to be in the same network segment.

Figure 2.4.1 Modify the local computer address.



Check the connectivity between the PC and the 1024E1. Click **Start** → **Run** of PC and enter **cmd** to execute 'ping 192.168.11.1' to check whether the IP address of GE1 port runs normally.



```
C:\Windows\system32\cmd.exe
Microsoft Windows [Versión 10.0.14393]
(c) 2016 Microsoft Corporation. Todos los derechos reservados.

C:\Users\USUARIO>ping 192.168.11.1

Haciendo ping a 192.168.11.1 con 32 bytes de datos:
Respuesta desde 192.168.11.1: bytes=32 tiempo<1m TTL=64

Estadísticas de ping para 192.168.11.1:
    Paquetes: enviados = 4, recibidos = 4, perdidos = 0
              (0% perdidos),
    Tiempos aproximados de ida y vuelta en milisegundos:
        Mínimo = 0ms, Máximo = 0ms, Media = 0ms

C:\Users\USUARIO>
```

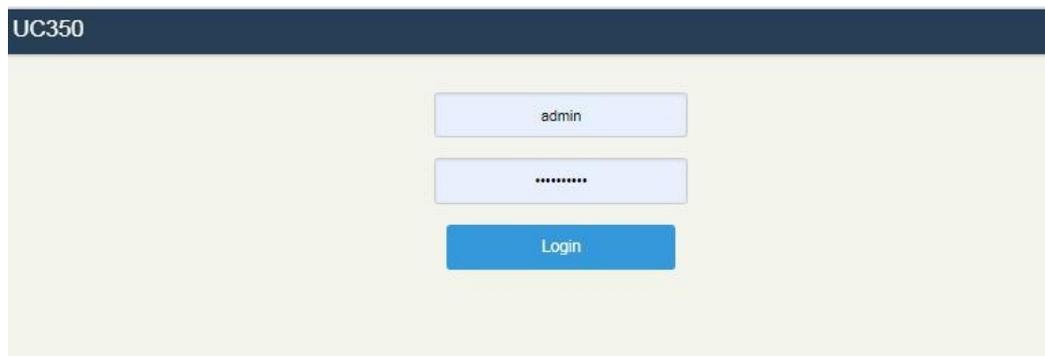
2.4.2 Log In Web

Open a browser and enter the IP address of GE1 port (the default IP is 192.168.11.1). Then the login GUI will be displayed.

You also can enter the IP address of GE0 port, but it's required to modify the IP address of PC to make it at the same network segment with that of GE0 port.

It is suggested that you should modify the username and password for security consideration.

Figure 2- 3 Login GUI of NEXO 1024E1



By default, the username is **admin**, while the password is **admin@123#**. After entering username and password, click **Login** to enter into the web interface.

Under some circumstances, login of the Web will be limited :

- For three consecutive login failures, you need to slide to validate your user account;
- Failing to log in the Web for ten times consecutively, the IP address of the NEXO 1024E1 device will be put into the blacklist, and you need to reset a new IP address for the

device ;

-
- Successful login or device restart will wipe out login failure records.

2.4.3 Modify the IP address of GE0 port

Connect the 1024E1 to the network according to the network connection diagram in section 2.3, but the GE0 port of the 1024E1 defaults to a static IP address: 192.168.12.1. At this time, you need to modify the IP address of the GE0 port so that the GE0 port and the upstream network are in the same network segment. There are two ways to modify the IP address of the GE0 port:

Method 1:

As shown in methods 2.4.1 and 2.4.2, log in to the web of the gateway and navigate to "Network -> Settings" to modify the IP address.

Figure 2.4-5 Modify IP address via web

The screenshot shows the 'Network / Setting' page for the GE0 port. The settings are as follows:

| Field | Value |
|----------------------|-------------|
| IP Address | 172.28.4.35 |
| Netmask | 255.255.0.0 |
| Default Gateway | 172.28.1.1 |
| Preferred DNS server | 172.28.1.1 |
| Alternate DNS server | 61.139.2.69 |
| MTU | 1500 |
| Metric | 9 |

Method 2:

Step 1: Connect the phone to the FXS port of the user board of the gateway with a telephone line.

Step 2: Modify the IP address of GE0 port by dialing *152, for example: dial *152*192*168*1*10# to set the IPv4 address to 192.168.1.10.

Step 3: Modify the gateway IP address of GE0 port by dialing *156, for example: dial *156*192*168*1*1# to set the IPv4 gateway to 192.168.1.1.

Step 4: Modify the mask of the GE0 port by dialing *153, for example: dial *153*255*255*0*0# to set the IPv4 mask to 255.255.0.0.

Step 5: Restart the device by dialing *111, the IP address of the GE0 port is the modified IP address.

3

Basic Operation

3.1 Methods to Number Dialing

There are two methods to dial telephone number or extension number:

- Dial the called number and wait for 4 seconds for dialing timeout, or dial the called number directly (the system will judge whether the dialing is completed according to Digitmap and Regular Expression dialplans).
- Press # after dialing the called number to end.

3.2 Call Holding

The current call can be held by pressing the "flash" key on the phone (if available), and then pressing the "flash" key again to resume the held call. If there is no "flash" key, you can use "hook flash" instead.

3.3 Call Waiting

When call waiting is enabled, if you hear the call waiting voice (three beeps of the FXS extension) during a call, it indicates that a new call is incoming. You can switch between the incoming call and the current call through the Flash key or hook flash.

3.4 Instruction of Hook Flash

A talks with B, A dials the number of C after pressing the hook flash, A and C talk, and the conversation between A and B is kept (B hears the waiting tone). At this time, A can switch to the call with B by hook flash and pressing the 1 key, switching to the call with C by hook flash and pressing the 2 button, and entering the three-way call by hook flash and pressing the 3 button.

3.5 Query IP Address and Restore Default Settings

After connecting the phone to the FXS port of the gateway with a telephone line, dial feature code *158 to query the IP address of the GE1 port, and dial feature code *159 to query the IP address of the GE0 port.

If you want to restore NEXO 1024E1 to default settings, you can press the **RST** button for 6 to 12 seconds or you can configure it on the Web interface.

After logging in to the Web, navigate to **System** → **Backup/Restore/Upgrade**, select the parts (system, network or service) that need to be restored in the "Restore to default settings" option,click **Reset** to restart the device, and the selected parts will be restored to default settings.

.Figure 3.5 Restore default settings

System / Backup/Restore/Upgrade

Upgrade Backup/Restore

Choose backup files and download System Network Service [Download](#)

Reset to defaults System Network Service [Reset](#)

Restore from the backup [Restore](#)

Restore to History Backup

| Index | User | Backup Time | |
|-------|-------|---------------------|---|
| 1 | admin | 2021-07-09 17:50:15 | ? ↶ ✖ |
| 2 | admin | 2021-07-09 16:47:22 | ? ↶ ✖ |
| 3 | admin | 2021-07-09 16:14:16 | ? ↶ ✖ |
| 4 | admin | 2021-07-09 16:13:59 | ? ↶ ✖ |
| 5 | admin | 2021-07-09 16:12:34 | ? ↶ ✖ |
| 6 | admin | 2021-07-09 16:00:52 | ? ↶ ✖ |
| 7 | admin | 2021-07-09 12:30:52 | ? ↶ ✖ |
| 8 | admin | 2021-07-09 11:52:27 | ? ↶ ✖ |
| 9 | admin | 2021-07-09 11:52:03 | ? ↶ ✖ |
| 10 | admin | 2021-07-09 11:23:42 | ? ↶ ✖ |

4

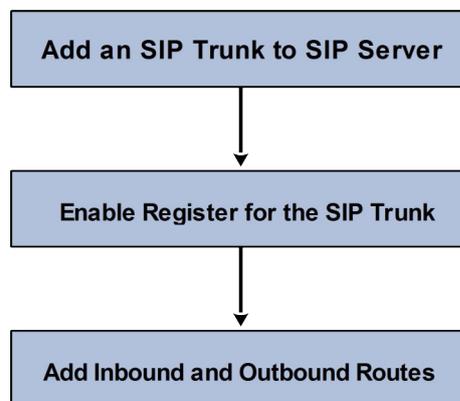
Configuration Wizard

4.1 Configuration Wizard

The following are the common ways to configure the NEXO 1024E1 gateway.

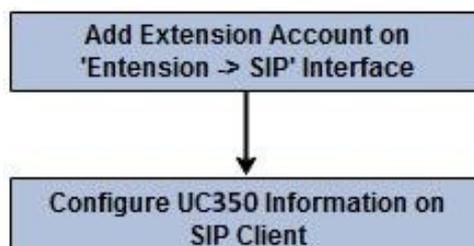
4.1.1 NEXO 1024E1 Regarded as Terminal and Registered to SIP Server

1. 1024E1 Registered to SIP Server

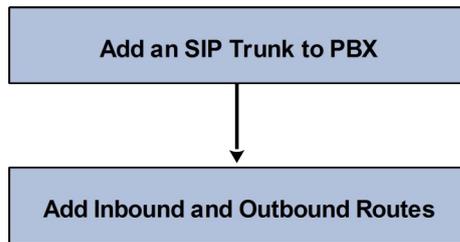


4.1.2 Other SIP Clients registered to 1024E1

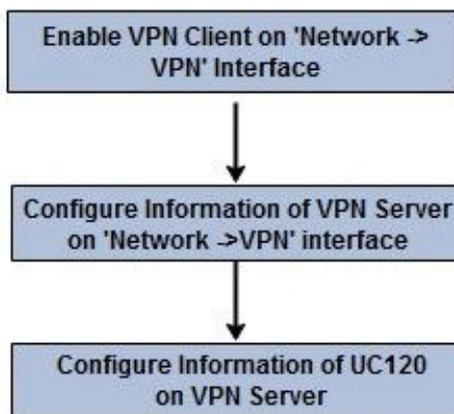
Under this mode, NEXO 1024E1 is regarded as an SIP Server. Create an extension account first on the **Extension** → **SIP** interface, and configure listening port on the **Profile** → **SIP** interface. Then, configure the server and account on SIP client.



4.1.3 NEXO 1024E1 Connected to PBX through Trunking



4.1.4 NEXO 1024E1 Serving as VPN Client



5

Configurations on Web Interface

5.1 Introduction to Web Interface

Modify the IP address of PC to make it at the same network segment with that of GE1 port of the NEXO 1024E1 gateway (the default IP of GE1 port is 192.168.11.1).

Open a web browser on the PC and then enter the IP address of GE1 port. Click **Login**, and the login GUI is displayed. The default username and password are **admin / admin@123#**.

The displayed login GUI is shown as follows:

Figure 5- 1 Introduction to login GUI

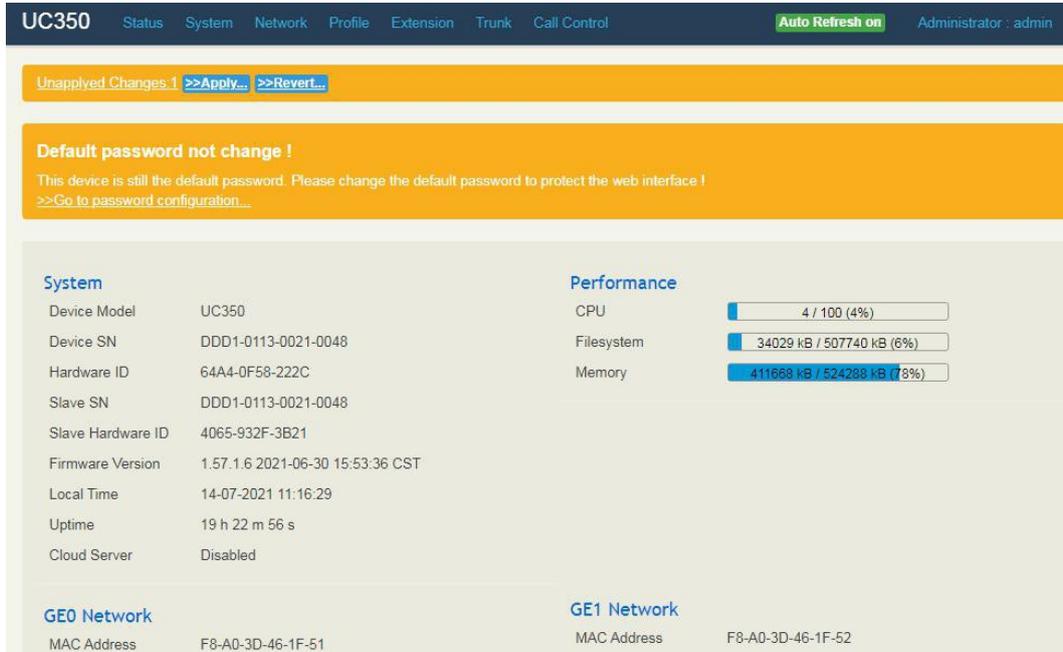


Table 5- 1 Introduction of Web Interface

| Index | Item | Description |
|-------|-----------------|--|
| 1 | NEXO 1024E1 | The name of the gateway; it can be edited on the System → Setting interface. |
| 2 | Menu Bar | The menu bar of NEXO 1024E1 |
| 3 | Unsaved Changes | All changes to the configuration of the gateway need to be saved. Click Apply to enter into the interface to save the changes; click Revert to return to original configuration. |
| 4 | Password | It will be prompted when the default password of the admin |

| | | |
|---|---------------------|---|
| | modification | account is not changed, and it will not be prompted after the default password is changed. |
| 5 | Detailed Interface | The detailed configuration interface or display interface |
| 6 | Auto Refresh Button | The button can be enabled or disabled. If it is enabled, the information on the Status → Overview/SIP/PSTN/Current Call interfaces will be refreshed automatically |
| 7 | User Role | The role of the current user logging into the Web. And the “exit” sign will pop up when the mouse moves over there. You can log out of the web from there |

5.2 Status

The ‘Status’ menu mainly displays all kinds of status information. It includes the following sub-menus: Overview, SIP, PSTN, Fail2ban, Current Call, CDRs, Service, performance and About etc.



5.2.1 Overview

Log in the Web interface of NEXO 1024E1, click **Status → Overview**, and the following interface will be displayed. On the interface, device model, firmware version, device running time, Mac address, IP address, performance parameters, etc.

Figure 5-2 Overview

System

Device Model: UC350
 Device SN: DDD1-0113-0021-0048
 Hardware ID: 64A4-0F58-222C
 Slave SN: DDD1-0113-0021-0048
 Slave Hardware ID: 4065-932F-3B21
 Firmware Version: 1.57.1.6 2021-06-30 15:53:36 CST
 Local Time: 2021-07-12 11:34:48
 Uptime: 58 m 21 s
 Cloud Server: Disabled

Performance

CPU: 93 / 100 (93%)
 Filesystem: 33881 kB / 807740 kB (6%)
 Memory: 405504 kB / 524288 kB (77%)

GE0 Network

MAC Address: F8-A0-3D-46-1F-51
 Type: Static
 IP Address: 172.28.4.35
 Netmask: 255.255.0.0
 Gateway: 172.28.1.1
 Preferred DNS server: 172.28.1.1
 Alternate DNS server: 61.139.2.69
 RX / TX (Per Second): 136.82 KB (246 Pkts.) / 18.00 KB (66 Pkts.)
 RX / TX (Total): 243.84 MB (626503 Pkts.) / 887.82 MB (220222 Pkts.)

GE1 Network

MAC Address: F8-A0-3D-46-1F-52
 Type: Static
 IP Address: 10.10.10.35
 Netmask: 255.255.255.0
 Gateway: 10.10.10.1
 Preferred DNS server: 172.28.1.1
 Alternate DNS server: 61.139.2.69
 RX / TX (Per Second): 4.18 KB (71 Pkts.) / 0 Bytes (0 Pkts.)
 RX / TX (Total): 8.45 MB (136829 Pkts.) / 39.78 KB (195 Pkts.)

5.2.2 SIP

Click **Status** → **SIP**, and the following interface will be displayed. On the interface, information of SIP profile, SIP Trunk and SIP extension is shown.

Figure 5- 3 Status of SIP Profile, SIP Trunk and SIP Extension

Status / SIP

SIP Extension

SIP Trunk

SIP Profile

Filter by Status Register Unregistered

| Index | Name | Extension | Online | Register Source | Status | Expires | Agent | Profile |
|-------|------|-----------|--------|--------------------|-----------------|---------|------------------|-------------------|
| 1 | 350 | 350 | 1 | 172.28.4.250:45066 | Registered(UDP) | 295 | MicroSIP/3.19.15 | 1-< GE0_Default > |
| 2 | 351 | 351 | 0 | | Unregistered | | | 1-< GE0_Default > |
| 3 | 352 | 352 | 0 | | Unregistered | | | 1-< GE0_Default > |
| 4 | 353 | 353 | 0 | | Unregistered | | | 1-< GE0_Default > |
| 5 | 354 | 354 | 0 | | Unregistered | | | 1-< GE0_Default > |
| 6 | 355 | 355 | 0 | | Unregistered | | | 1-< GE0_Default > |

Status / SIP

SIP Extension

SIP Trunk

SIP Profile

| Index | Name | Address | Transport | Reg | Heartbeat | Status | Call In(F/T) | Call Out(F/T) | Profile |
|-------|-----------|-------------------|-----------|-----|-----------|----------|--------------|---------------|--------------|
| 1 | trunk-200 | 172.28.4.9:5099 | UDP | off | off | NOREG/UP | 0/0 | 0/0 | 1-<GE0_De... |
| 2 | trunk-30 | 172.28.1.30:35060 | UDP | off | off | NOREG/UP | 0/0 | 0/0 | 1-<GE0_De... |
| 3 | trunk-32 | 172.28.1.32:25060 | UDP | off | off | NOREG/UP | 0/8093 | 0/0 | 1-<GE0_De... |
| 4 | trunk-66 | 172.28.1.66:65060 | UDP | off | off | NOREG/UP | 0/0 | 0/3340 | 1-<GE0_De... |
| 5 | UC2500 | 172.28.4.51:5062 | UDP | off | off | NOREG/UP | 0/1 | 0/0 | 1-<GE0_De... |

| Index | Name | Listening Addr | State | Current Call | Call In(F/T) | Call Out(F/T) |
|-------|-------------|-------------------|---------|--------------|--------------|---------------|
| 1 | GE0_Default | 172.28.4.35:35060 | RUNNING | 195 | 4754/8155 | 1/3402 |
| 2 | GE1_Default | 10.10.10.35:35060 | RUNNING | 0 | 0/0 | 0/0 |

Table 5- 2 Explanation of SIP Parameters

| Belong To | Parameter | Explanation |
|---------------|-------------------|---|
| Profile | Name | The name of the SIP profile |
| | Listening Address | The current listening address and port of SIP |
| | State | Green color means normal running, while red color means listening address and port of SIP is unavailable. There are two states :Running and Down |
| SIP Trunk | Heartbeat | If heartbeat is enabled, option message will be sent to peer device (the peer device is reachable) |
| | Status | Green color means available, while red color means abnormal, unavailable or prohibited. There are five statuses: Running, Reged/Up, Noreg/Up, Trying-Down, Fail-Wait |
| | Profile | The profile that is used by the SIP trunk |
| SIP Extension | Profile | The profile that is used by the SIP extension |
| | Status | SIP extension is registered or not. There are two statuses: Registered and Unregistered |

5.2.3 PSTN

On the **Status** → **PSTN** interface, the configuration status of the FXS port, FXO port and E1 port, the line state and hook status, and the channel status of the line and E1 port.

Figure 5.2-3 FXS port\FXO port status and E1 channel status

Figure 5- 4 Status of FXS and FXO

| FXS | | | | | |
|------|------|------------------|------------|------------|--|
| Slot | Port | Extension Number | Hook State | Line State | |
| 0 | 0 | 300 | ONHOOK | IDLE | |
| 0 | 2 | 301 | ONHOOK | IDLE | |
| 0 | 4 | 302 | ONHOOK | IDLE | |
| 0 | 6 | 303 | ONHOOK | IDLE | |
| 1 | 0 | 310 | ONHOOK | IDLE | |
| 1 | 2 | 311 | ONHOOK | IDLE | |
| 1 | 4 | 312 | ONHOOK | IDLE | |
| 1 | 6 | 313 | ONHOOK | IDLE | |

| FXO | | | | | |
|------|------|--------|------------|------------|--|
| Slot | Port | Number | Hook State | Line State | |
| 0 | 1 | | OFFLINE | IDLE | |
| 0 | 3 | | OFFLINE | IDLE | |
| 0 | 5 | | OFFLINE | IDLE | |
| 0 | 7 | | ONHOOK | IDLE | |
| 1 | 1 | | OFFLINE | IDLE | |
| 1 | 3 | | OFFLINE | IDLE | |
| 1 | 5 | | OFFLINE | IDLE | |
| 1 | 7 | | OFFLINE | IDLE | |
| 2 | 0 | | ONHOOK | IDLE | |
| 2 | 1 | | ONHOOK | IDLE | |

| E1 Trunk | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
|----------|------|-------------|----------------|---------|---|----------------|---|------------|---|-----------|---|-----------|----|-----------------------|----|-------------|----|------------|----|------|----|--------|----|---------|----|----------|----|------|----|------|----|---------|----|-------|--|---------|--|-----------|--|-----------|--|-----------|
| Slot | Port | Port Status | Channel Status | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| | | | 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 | 13 | 14 | 15 | 16 | 17 | 18 | 19 | 20 | 21 | 22 | 23 | 24 | 25 | 26 | 27 | 28 | 29 | 30 | 31 | | | | | | | | |
| NOTES | | Activated | | Disable | | Not Authorized | | LOS Alarmd | | RAI Alarm | | AIS Alarm | | ISDN/SS7 Signal Alarm | | Auto Closed | | Frame-Sync | | IDLE | | Signal | | Release | | Progress | | Ring | | Talk | | Release | | Fault | | Disable | | L-blocked | | R-blocked | | B-blocked |

5.2.4 Fail2ban

On the **Status** → **Fail2ban** interface, you can see currently-banned IP addresses and historic banned IP addresses. You can also unban those IP addresses that have been blocked before.

Fail2ban is a log-parsing application that monitors system logs for symptoms of an automated attack on your device. When an attempted compromise is located, using the defined parameters, Fail2ban will add a new rule to block the IP address of the attacker, either for a set amount of time or permanently. Fail2ban can also alert you through email that an attack is occurring.

Figure 5- 6 Banned IP Addresses

Status / Fail2ban

Current Ban List

| Index | IP | Ban time | Release time | Type | Action |
|-------|----|----------|--------------|------|--------|
|-------|----|----------|--------------|------|--------|

Operation History List

| Index | IP | Common Ban Duration | Type | Action | Operation time | Filter |
|-------|----|---------------------|------|--------|----------------|--------|
|-------|----|---------------------|------|--------|----------------|--------|

For the explanation of parameters related to fail2ban, please refer to the “Network ->Fail2ban” section.

5.2.5 Current Call

On **Status → Current Call** interface, the source, destination, calling number, called number, start time, answer time, state and duration of the current real-time call are shown. If there is no current call, no information will be shown

Figure 5- 8 Current Call Information

Status / Current Call

| Index | Src | Dest | Caller | Called | Start Time | Answer Time | State | Duration | Filter |
|-------|----------------------|----------------------|--------|--------|------------|-------------|---------|----------|--------|
| 1 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 663750 | 550786 | 11:44:15 | 11:44:16 | TALKING | 00:00:59 | |
| 2 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 669686 | 554286 | 11:44:18 | 11:44:20 | TALKING | 00:00:55 | |
| 3 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 663450 | 553466 | 11:44:21 | 11:44:23 | TALKING | 00:00:52 | |
| 4 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 660164 | 551262 | 11:44:21 | 11:44:24 | TALKING | 00:00:51 | |
| 5 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 665825 | 559498 | 11:44:23 | 11:44:25 | TALKING | 00:00:50 | |
| 6 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 660769 | 558112 | 11:44:24 | 11:44:26 | TALKING | 00:00:49 | |
| 7 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 667465 | 553224 | 11:44:25 | 11:44:27 | TALKING | 00:00:48 | |
| 8 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 668997 | 559200 | 11:44:25 | 11:44:27 | TALKING | 00:00:48 | |
| 9 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 660666 | 557733 | 11:44:26 | 11:44:27 | TALKING | 00:00:48 | |
| 10 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 669987 | 559775 | 11:44:26 | 11:44:28 | TALKING | 00:00:47 | |
| 11 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 669170 | 558491 | 11:44:28 | 11:44:30 | TALKING | 00:00:45 | |
| 12 | SIP Trunk/3/trunk-32 | SIP Trunk/4/trunk-66 | 663450 | 553466 | 11:44:28 | 11:44:30 | TALKING | 00:00:45 | |

5.2.6 Conference

The "Status -> Conference" interface displays the current conference status, conference call information, operation records, etc.

Status / Conference

| Name | Room | Total | Administrator | Start Time | Duration | Options |
|--------|------|-------|---------------|------------|----------|---------|
| 会议1 | 5678 | 0 | 0 | | | |
| Caller | | | Source | Join Time | Duration | Options |
| 会议2 | 6789 | 0 | 0 | | | |
| Caller | | | Source | Join Time | Duration | Options |
| 会议3 | 1234 | 0 | 0 | | | |
| Caller | | | Source | Join Time | Duration | Options |

5.2.7 Call Queue

The user can view the related information of the queue status on the "Status -> Call Queue" interface, the queue configuration change will restart the agent, and the dynamic agent will disappear.

Status / Call Queue

| Name | Number | Strategy | Agents Count | Waiting Calls | Answered Calls | Total Calls |
|-------|--------|----------|--------------|---------------|----------------|-------------|
| 呼叫队列2 | 89 | Linear | 1 | 0 | 0 | 0 |

Click the on the far left to view more detailed agent status.

Status / Call Queue

| Name | Number | Strategy | Agents Count | Waiting Calls | Answered Calls | Total Calls |
|------------------|--------------|-------------|--------------------|-----------------|----------------|-------------|
| 呼叫队列2 | 89 | Linear | 1 | 0 | 0 | 0 |
| Extension Number | Agent Status | Call Status | Last Call End Time | No Answer Calls | Answered Calls | |
| 310 | Available | Waiting | 0 | 0 | 0 | |

5.2.8 Parking Lot

You can use the parking feature to park a call, and then retrieve the call either from your phone or another phone. After you park a call, the call is placed on hold, you can continue the conversation after retrieving it.

On the **Status -> Parking Lot** interface, the numbers that are parked and the parking duration are shown.

Figure 5- 9 Call Parking Status

Status / Parking Lot

| Index | Parking Number | Source | Duration |
|-------|----------------|--------|----------|
|-------|----------------|--------|----------|

5.2.9 CDRs

Click **Status** → **CDRs**, and you can set query criteria to query the CDRs (Call Detailed Records) that you want on the displayed interface. Meanwhile, you are allowed to clear CDRs or export CDRs through clicking the **Empty** or **Export** button. The maximum number of CDRs that can be saved is 5000.

CDRs cannot be saved on the **Status** → **CDRs** interface unless the CDRs function has been enabled on the **System** → **Setting** interface.

Figure 5- 10 CDRs

The screenshot shows the 'Status / CDRs' interface. It features a 'CDRs Query Param' section with fields for Start Date (2021, 7, 1), End Date (2021, 7, 12), Caller, Source (Any), Min Duration, Called, Destination (Any), and Max Duration. There are 'Query' and 'Reset' buttons. Below is a 'CDRs List' section with 'Empty' and 'Export' buttons. A table displays the following data:

| Index | Caller | Source | Called | Destination | Start Time | End Time | Duration | Hangup By | Codec | Hangup Cause | Filter |
|-------|--------|--------------------|--------|-------------------|-------------------|-------------------|----------|-----------|-------|-----------------|--------|
| 1 | 668189 | SIP Trunk/trunk-32 | 553845 | SIP Trunk/trun... | 2021-07-12 11:... | 2021-07-12 11:... | 0 | Called | G729 | Temporary Fa... | |
| 2 | 668181 | SIP Trunk/trunk-32 | 553083 | SIP Trunk/trun... | 2021-07-12 11:... | 2021-07-12 11:... | 36 | Caller | G723 | Normal Clearing | |
| 3 | 664100 | SIP Trunk/trunk-32 | 553626 | SIP Trunk/trun... | 2021-07-12 11:... | 2021-07-12 11:... | 31 | Caller | G729 | Normal Clearing | |
| 4 | 669098 | SIP Trunk/trunk-32 | 556712 | SIP Trunk/trun... | 2021-07-12 11:... | 2021-07-12 11:... | 30 | Caller | PCMA | Normal Clearing | |
| 5 | 663962 | SIP Trunk/trunk-32 | 557685 | SIP Trunk/trun... | 2021-07-12 11:... | 2021-07-12 11:... | 37 | Caller | PCMA | Normal Clearing | |
| 6 | 665403 | SIP Trunk/trunk-32 | 556827 | SIP Trunk/trun... | 2021-07-12 11:... | 2021-07-12 11:... | 0 | Called | G729 | DESTINATIO... | |

5.2.10 Service

Click **Status** → **Service**, and the service status of NEXO 1024E1 is displayed. This function is enabled by default. The Web, SSH and Telnet service can be disabled and their ports can be modified on the **Network** → **Access Control** interface. The remote proxy and NATS server are enabled by default, and can be disabled and modified on the "System -> Cloud Service" interface; other services cannot be disabled. If no running status is shown, it means exception has occurred on NEXO 1024E1

Besides, if syslog is disabled on the **System** → **Setting** interface, the logs cannot be uploaded to the server, but log service is still running.

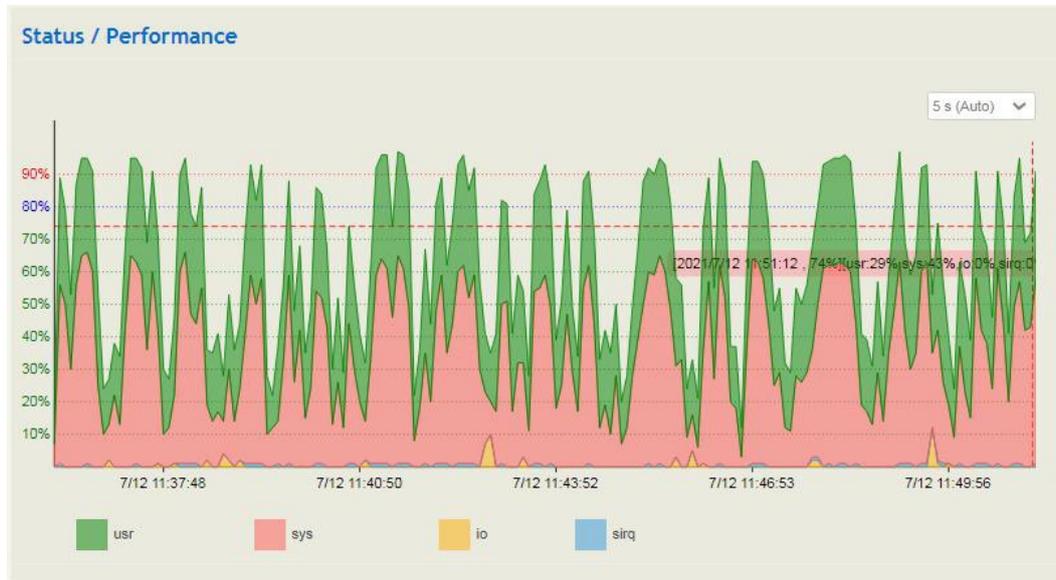
Figure 5- 11 Service Status



5.2.11 Performance

On the **Status ->Performance** Interface displays the performance statistics of the system. The parameters of the device such as usr, sys, io, and sirq are displayed at the 5th second, 1 minute, 5 minutes, and 1 hour.

Figure 5- 12 System Performance



5.2.12 About

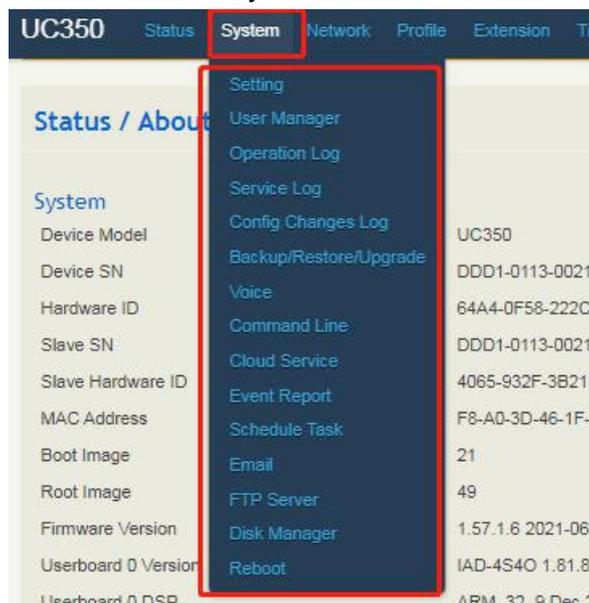
On the **Status -> About** interface, the device model, device SN, hardware ID, MAC address, boot image, root image, firmware Version of the NEXO 1024E1 device are displayed.

Figure 5- 13 Device and Company Information

| Status / About | |
|---------------------|--|
| System | |
| Device Model | UC350 |
| Device SN | DDD1-0113-0021-0048 |
| Hardware ID | 64A4-0F58-222C |
| Slave SN | DDD1-0113-0021-0048 |
| Slave Hardware ID | 4065-932F-3B21 |
| MAC Address | F8-A0-3D-46-1F-51 |
| Boot Image | 21 |
| Root Image | 49 |
| Firmware Version | 1.57.1.6 2021-06-30 15:53:36 CST |
| Userboard 0 Version | IAD-4S4O 1.81.80.02 PCB 3 LOGIC 0 BIOS 1, 2021-06-24 12:25:47 (13 / 14 / 12) |
| Userboard 0 DSP | ARM_32_9 Dec 29 2018 17:01:36 |
| Userboard 1 Version | IAD-4S4O 1.81.80.02 PCB 3 LOGIC 0 BIOS 1, 2021-06-24 12:25:47 (13 / 14 / 12) |
| Userboard 1 DSP | ARM_32_9 Dec 29 2018 17:01:36 |
| Userboard 2 Version | IAD-8O 1.81.80.02 PCB 2 LOGIC 0 BIOS 1, 2021-06-24 12:25:47 (13 / 14 / 12) |
| Userboard 2 DSP | ARM_32_9 Dec 29 2018 17:01:36 |
| Userboard 3 Version | 02.06.25.03 2021-06-07 17:44:26(25/39) |
| Userboard 3 DSP | [dsp0.succ-128][dsp1.succ-128] |
| License | |
| Version | 1.0.0.3 |
| SN | 2 |
| Valid Period | 90 d |
| Max Concurrency | 150 |
| SIP Extensions | 1000 |
| Remain | 35 d |

5.3 System

Configurations for timezone, login username & password, other user name, provision, operation log, service log, upgrade/backup/restore, Command Line, cloud server and device reboot can be carried out in the System section.



5.3.1 Setting

On the **System** → **Setting** interface, you can modify the device name, set a new timezone, synchronize local time and enable CDRs, Syslog as well as built-in NTP server.

Figure 5- 12 Basic Setting

The screenshot displays the 'System / Setting' configuration page, divided into three main sections:

- General:**
 - Hostname: UC350
 - Language: English
 - Timezone: Asia/Beijing
 - Local Time: 2021-07-12 11:55:28 (with a 'Sync with browser' button)
 - Date Format: YYYY-MM-DD
 - CDRs: Enable
 - Hover Prompt: Disable
- Log:**
 - Service Log Level: Warning
 - Enable Syslog:
 - Log Server IP Address: 172.28.88.187
 - Log Server Port: 514
- Time Synchronization:**
 - Enable builtin NTP server:
 - NTP server candidates:
 - 0.pool.ntp.org (with a red 'x' icon)
 - 1.pool.ntp.org (with a red 'x' icon)
 - 2.pool.ntp.org (with a red 'x' icon)
 - 3.pool.ntp.org (with a red 'x' icon and a green '+' icon)

At the bottom of the 'Time Synchronization' section, there are buttons for 'Cancel', 'Save', and 'Reset'.

Figure 5- 4 Explanation of Basic Setting Parameters

| Parameter | Explanation |
|------------|--|
| Hostname | The name of the gateway. After it is configured, the name will be displayed on the left of the menu bar. |
| Timezone | You can choose a time zone you want. The default value is UTC (Universal Time Coordinated) |
| Local Time | The current time based on current time zone. It is synchronized with NTP. |
| CDRs | If it is enabled, CDRs will be saved automatically. 5000 CDRs call be saved at most and they can be queried on the Status → CDRs |

| | |
|---|--|
| | interface. If it is disabled, CDRs will not be saved |
| Service Log Level | There are eight levels, including Debug, Info, Notify, Warning, Error, Critical, Alert and Emergency |
| Enable Syslog | Whether to enable syslog |
| Time Synchronization | If NTP server is enabled, the NEXO 1024E1 can be synchronized with the world standard time. Meanwhile, you're able to add or reduce NTP servers. Please consult local telecom operators or surf the internet for the address of NTP servers. |
|  | Delete a NTP Server |
|  | Add a NTP Server |

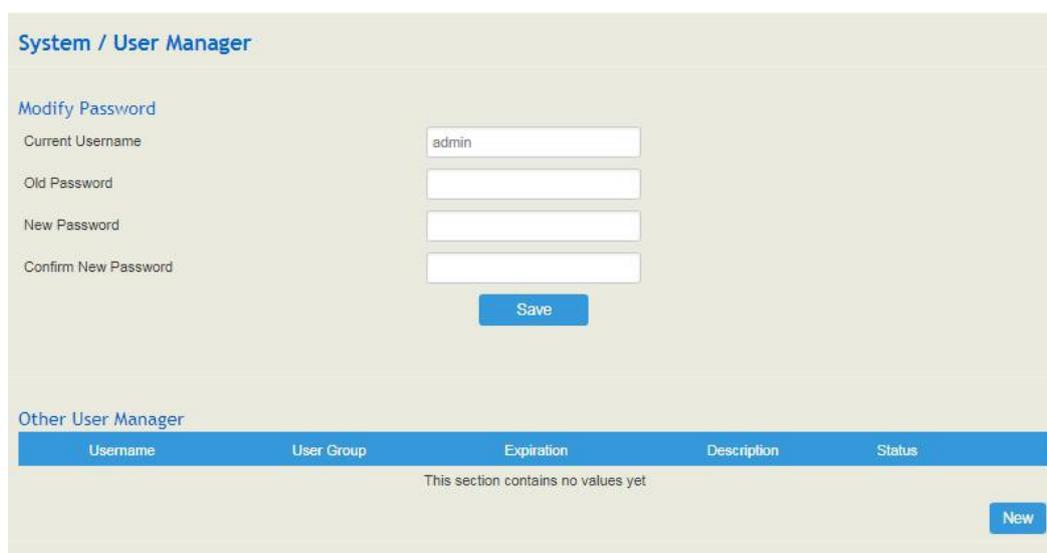
5.3.2 User Manager

Click **System** → **User Manager**, and you can modify the username name and password for logging in the NEXO 1024E1 gateway. Factory defaults for username name and password are **admin** and **admin@123#** respectively, so it is advised to modify them for security consideration.

The above mentioned username and password are also used to log in Web Interface, Telnet and SSH.

The super administrator of the device can add different users to the device and assign different roles for them, like observer, operator and administrator. Different roles can be allocated with different permissions to the functions.

Figure 5- 14 Modify Username ,Password and Manager Users



System / User Manager

Modify Password

Current Username:

Old Password:

New Password:

Confirm New Password:

Other User Manager

| Username | User Group | Expiration | Description | Status |
|-------------------------------------|------------|------------|-------------|--------|
| This section contains no values yet | | | | |

Figure 5- 15 Add New User

Figure 5- 5 Explanation of Provision Parameters

| Parameter | Explanation |
|-----------------------|--|
| Name | The name of the new user. After it is established, the name and the password will be used to log into the web interface of the device. |
| User Group | You can choose a role for the new user, such as administrator, operator and observer. The default value is administrator. |
| New Password | Setting the login password for the new user. The password needs to consist of 8 to 32 characters. |
| Expiration | The expiry date when the user cannot log in the device any more. |
| Status | Choose enable or disable. |
| Web Access Permission | The permissions to view status, system, network, profile, extension, trunk and call control. |

5.3.3 Operation Log

The logs tracing the operations carried out on the Web can be queried on the **System** → **Operation Log** interface. You are allowed to set query criteria to query the logs that you want and to export the logs through clicking the **Export** button at the top-right corner.

Figure 5- 17 Operation Logs

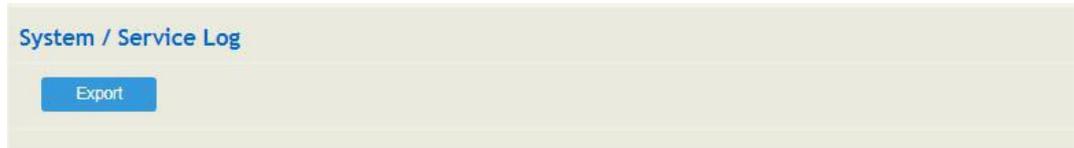
| Index | Time | Level | Access Source | Operation | Page |
|-------|-------------------------|-------|--------------------|-------------------|--------------------------|
| 100 | 2021-07-12 Mon 14:08:42 | Info | 172.19.1.74:59195 | View | system/security |
| 99 | 2021-07-12 Mon 14:08:42 | Info | 172.19.1.74:59194 | Cancel New Config | system/security/user/add |
| 98 | 2021-07-12 Mon 14:07:58 | Info | 172.19.1.74:59189 | Add New Config | system/security/user/add |
| 97 | 2021-07-12 Mon 14:07:02 | Info | 172.19.1.74:59174 | View | system/security |
| 96 | 2021-07-12 Mon 14:07:00 | Info | 172.19.1.74:59173 | View | status |
| 95 | 2021-07-12 Mon 14:06:59 | Info | 172.19.1.74:59173 | Login Succ | |
| 94 | 2021-07-12 Mon 13:51:44 | Info | 172.28.4.250:55216 | View | status |
| 93 | 2021-07-12 Mon 13:41:44 | Info | 172.28.4.250:58266 | View | status/currentcall |
| 92 | 2021-07-12 Mon 13:41:38 | Info | 172.28.4.250:50523 | View | status |
| 91 | 2021-07-12 Mon 13:41:38 | Info | 172.28.4.250:50523 | Login Succ | |
| 90 | 2021-07-12 Mon 13:41:35 | Info | 172.28.4.250:50523 | View | status |

Note: Operation logs are generally used to locate faults by device manufacturer.

5.3.4 Service Log

Service logs (the running logs of NEXO 1024E1) can be exported on the **System → Service Log** interface. Those logs are used for analyzing where a fault has occurred on the gateway.

Figure 5- 18 Service Log



5.3.5 Config Changes Log

On the **System → Config Changes Log** interface, the configurations changed by administrator on the Web of the gateway are recorded.

Figure 5- 19 Config Changes Log

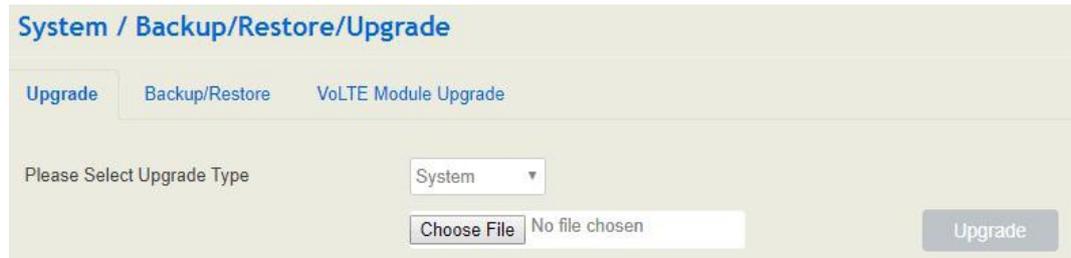


5.3.6 Backup/Restore/Upgrade

On the **System** → **Backup/Restore/Upgrade** interface, you can back up or restore configuration data, and can upgrade NEXO 1024E1 to a new version. But you need to restart the device for the change to take effect after executing restore or upgrade.

Figure 5- 19 Backup/Restore/Upgrade

Figure 5- 20 Upgrade the Device



Note: the file you choose to be upgraded on the above interface is a local file, while the version file upgraded through the Provision function is a file from http/ftp/tftp server.

Figure 5- 21 Back up configurations

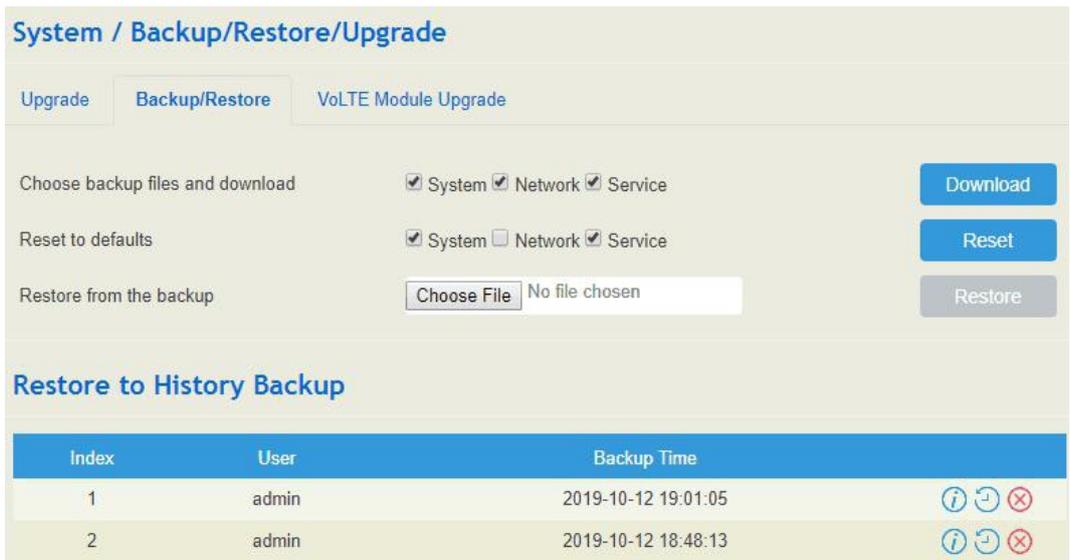


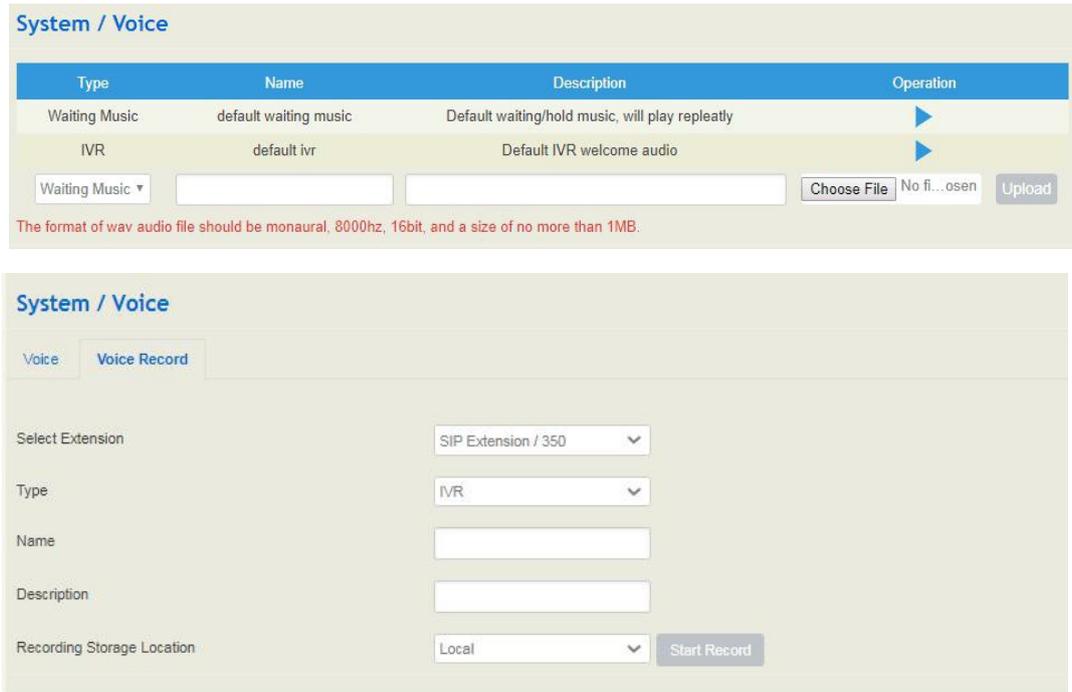
Table 5- 7 Explanation of Backup/Restore/Upgrade Button

| | |
|----------|--|
| Download | You can download the configuration data to be backed up. Select any of the checkboxes on the left of System, Network and Service, and then click Download |
| Reset | Select any of the checkboxes on the left of System, Network and Service, and then click Reset , and configurations related to the selected part will be restored to factory defaults. |
| Restore | Choose a backup file, and then click Restore . |
| Upgrade | Choose a file to be upgraded , and then click Upgrade |

5.3.7 Voice

On the **System** → **Voice** interface, you can upload an IVR file according to your needs. At present, only wav audio file is allowed. The format of the uploaded wav audio file must be: monaural, 8000hz, 16bit, and size of no more than 1M.

Figure 5- 23 Upload IVR File



5.3.8 Command Line

On the **System** → **Command Line** interface, some commonly-used command lines can be directly selected in the draw-down box, and therefore user has no need to enter command lines on Telnet. In this way, the efficiency of problem diagnostics is greatly improved.

Commonly-used command lines include fxo config, fxo status, fxs config, fxs status, gsm status, gsm bcch, gsm oper, sip status, sip profile and so on.

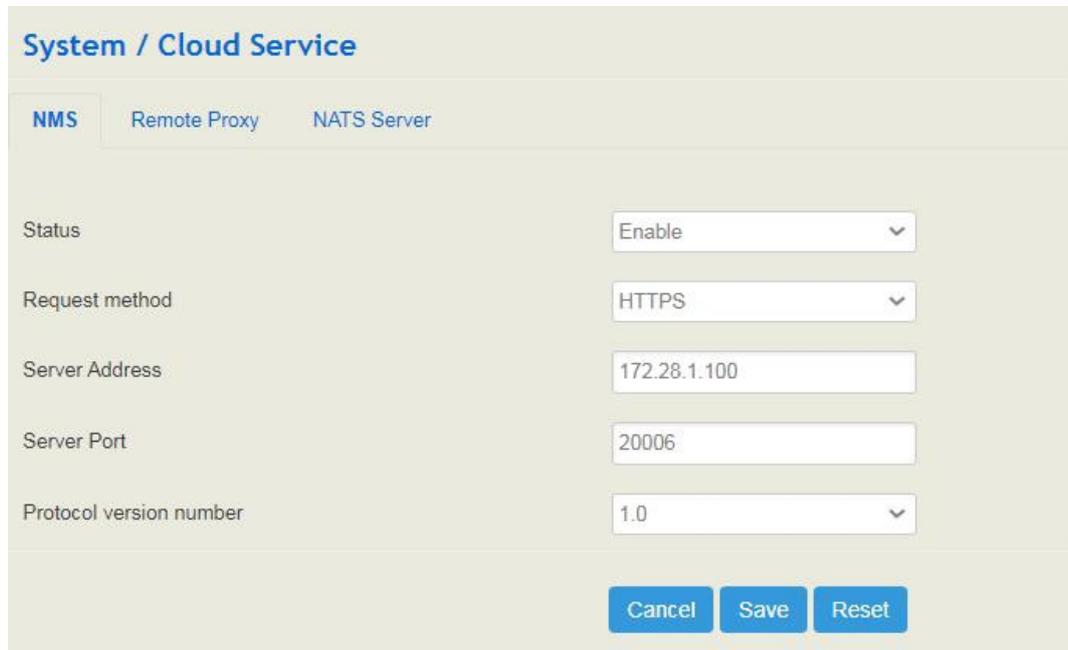
Figure 5- 24 Command Line

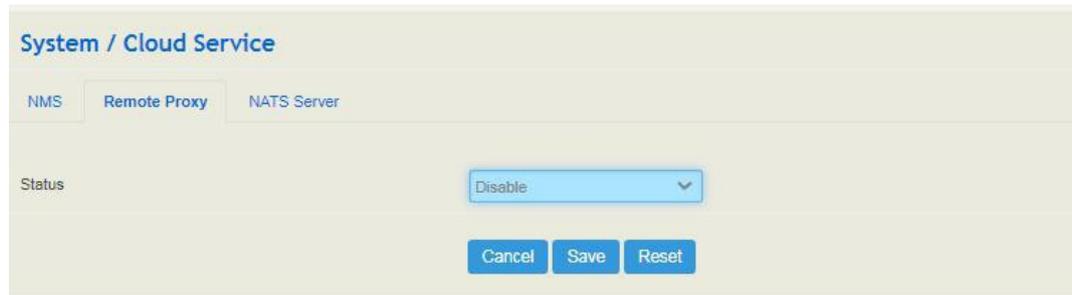


5.3.9 Cloud Service

Cloud service is mainly used to centrally manage all kinds of devices. Through cloud service, you can query the system status of a device, upgrade at batch, log in and configure remotely. NEXO 1024E1 provides two cloud services:remote proxy and NATS server, enter the IP address, service port and password of the Cloud server, and then the gateway will connect to the cloud server.

Figure 5- 26 Cloud Server





NATS Server:

NEXO 1024E1 can work as a NATS client to send messages to a NATS server, and then the NATS server will open related ports to facilitate the connection with those clients or servers of users.

How NATS works:

Operating mode

Both NATS client and management server need to connect to the NATS server;

The management server subscribes to from the NATS server, and the NATS client is responsible for publishing messages. After receiving the subject published by the client, the NATS server forwards the message to the management server. Subscription subject message flow: management server -> NATS server; reporting Message flow: NATS client -> NATS server -> management server;

NATS clients can also subscribe to subjects and receive messages published by the management server, but this way of working is unavailable in the event reporting function.

Configuration steps:

1) Configure NATS server on the web interface (NEXO 1024E1 acts as a client):

On the "System -> Cloud Server" interface, enable the NATS server and configure other parameters, such as IP address, port (when TLS is disabled, the port defaults to 4222, you can choose not to fill in), user name and password (can be empty), enable heartbeat etc.

Figure: NATS server configuration

The screenshot shows the 'System / Cloud Service' configuration interface. The 'NATS Server' tab is selected. The configuration fields are as follows:

- Status: Disable
- Server Address: [Empty text box]
- Username: [Empty text box]
- Password: [Empty text box]
- Heartbeat: Disable
- TLS Verification: Disable
- TLS Skip Server Verification: Disable
- Server Certificate: 选择文件 未选择任何文件
- Client Certificate: 选择文件 未选择任何文件
- Client Key: 选择文件 未选择任何文件

At the bottom, there are 'Save' and 'Reset' buttons.

(3) Configure the same NATS server information on the management server;

(4) Enter the command line on the management server and enter the command to subscribe to the subject; command format: `nats_client subscribe xxxxxx` (example: `nats_client subscribe *.server.register`)

The following are the three subjects that the management server must subscribe to:

Register subject(used to receive the registration information of the NATS client): `nats-client subscribe *.server register`

Event subject (used to receive event messages from NATS clients): `nats_client subscribe *.server.event`

Heartbeat subject (used to receive heartbeat messages from NATS clients): `nats_client subscribe *.server.heartbeat`

Note: After subscribing , you can use the `nats-client status` command to view the registration status and the subjects that have been subscribed. The command format for unsubscribing subjects is: `nats-client unsubscribe *.server.register`.

(5) Enable NATS report on the "System -> Event Reporting" interface of NEXO

1024E1. Figure: Enable NATS repor

System / Event Report

System SIP Recording Log

Event Type

| | |
|---------------|-------------------------------------|
| Device Status | <input checked="" type="checkbox"/> |
| URL Report | <input type="checkbox"/> |
| NATS Report | <input checked="" type="checkbox"/> |
| Call Status | <input type="checkbox"/> |
| CDRs Info | <input type="checkbox"/> |

Save Reset

To make NATS report work normally, the following three points need to be met:

- 1) The NATS client (NEXO 1024E1) has enabled the NATS report;
- 2) The connection between the NATS client and the NATS server has been established;;
- 3) The NATS client has been registered to the management server.

Note: You can view the status of the NATS client by entering "nats-client status" on the "System -> Command Line" interface. If the status displays "NATS Status: OK", it means that the NATS client and NATS server are successfully connected.

Figure: Enter the command to view the status of the NATS client

System / Command Line

nats-client status Execute Save Empty

```

Tue Jul 13 2021 10:32:42 GMT+0800 (中国标准时间) >> nats-client status
connect to nats client timeout

Tue Jul 13 2021 10:32:39 GMT+0800 (中国标准时间) >> nats-client status
connect to nats client timeout

```

5.3.10 Event Report

NEXO 1024E1 allows the following events to be reported through NATS or URL: device startup, call status, Register or deregister SIP extension, availability or unavailability of SIP trunks, off-hook or on-hook of FXS phone, FXO status and update of CDR information.

For event report through NATS, please refer to the configuration steps of NATS in the Cloud Server section.

For event report through URL, please see the following example:

- 1) On the "System -> Event Report" interface, select the events to be reported and the reporting method(URL);

The screenshot shows the 'System / Event Report' configuration page. The 'SIP' tab is active. Under 'Event Type', the following options are visible:

- SIP Extension Register/Unregister:
- URL Report: (The input field for the URL is highlighted with a red border.)
- Json Format:
- Parameter List:
- NATS Report:
- SIP Trunk Available/Unavailable:

The parameter list includes:

- \$username : Username
- \$network_address : SIP Extension Register Address, IP:Port
- \$agent : SIP Agent
- \$sip_status : SIP Extension Status, REGISTER/UNREGISTER
- \$sn : Device SN
- \$smac : MAC Address
- \$ip : WAN IP address(Route Mode) or LAN IP address(Bridge Mode)
- \$time : Local Date/Time, YYYY-MM-DD HH:MM:SS
- \$epochtime : Unix epoch time

Buttons for 'Save' and 'Reset' are located at the bottom right of the form.

- 2) Input the URL.

Format: `http://ip:port/event?key1=$value1&key2=$value2`

Example: `http://172.18.111.65:8080/sip?sn=$sn&mac=$username&key=$sip_status`

Event refers to startup, callstatus, sip, siptrunk, fxs, fxo, gsm, volte, vpn and cdr, while value refers to the parameter that needs to be reported. Key can be defined by yourself, but it's generally the same with value.

Figure 5- 27 Input URL

System / Event Report

System SIP Recording Log

Event Type

SIP Extension Register/Unregister

URL Report

Json Format

Parameter List

\$susexname : Username
 \$network_address : SIP Extension Register Address, IP:Port
 \$agent : SIP Agent
 \$sip_status : SIP Extension Status, REGISTER/UNREGISTER
 \$sn : Device SN
 \$mac : MAC Address
 \$ip : WAN IP address(Route Mode) or LAN IP address(Bridge Mode)
 \$time : Local Date/Time, YYYY-MM-DD HH:MM:SS
 \$epochtime : Unix epoch time

NATS Report

SIP Trunk Available/Unavailable

Save Reset

- 3) Use a softphone to register to an extension of NEXO 1024E1, and then the registration or deregistration of the softphone will be reported to NEXO 1024E1 through the URL.
- 4) On the System→Event Report→Log interface, you can view the report information. Figure:View report log

System / Event Report

System SIP Recording Log

Only latest 100 records provided to show, if want to see more, you can export it ! Export

| Index | Time | Type | URL Info | Report Status | Filter |
|-------|---------------------|-----------|---|-----------------------|--------|
| 1 | 2021-07-12 14:39:12 | Recording | http://172.28.88.187:8080/recordings/recordings333[20210... | Timeout | |
| 2 | 2021-07-12 13:08:57 | Recording | http://172.28.88.187:8080/recordings/recordings333[20210... | 301 Moved Permanently | |
| 3 | 2021-07-12 13:08:56 | Recording | http://172.28.88.187:8080/recordings/recordings333[20210... | 301 Moved Permanently | |
| 4 | 2021-07-12 13:08:54 | Recording | http://172.28.88.187:8080/recordings/recordings333[20210... | 301 Moved Permanently | |
| 5 | 2021-07-12 13:08:52 | Recording | http://172.28.88.187:8080/recordings/recordings333[20210... | 301 Moved Permanently | |
| 6 | 2021-07-12 13:08:49 | Recording | http://172.28.88.187:8080/recordings/recordings333[20210... | 301 Moved Permanently | |
| 7 | 2021-07-12 13:08:43 | Recording | http://172.28.88.187:8080/recordings/recordings333[20210... | 301 Moved Permanently | |
| 8 | 2021-07-12 13:08:39 | Recording | http://172.28.88.187:8080/recordings/recordings333[20210... | 301 Moved Permanently | |
| 9 | 2021-07-12 13:08:37 | Recording | http://172.28.88.187:8080/recordings/recordings333[20210... | 301 Moved Permanently | |

5.3.11 Schedul Task

On the **System → Schedule Task** interface, you can set a scheduled to reboot the NEXO 1024E1 device, record backup, and back up CDRs, logs or configurations.

Figure 5- 28 Configure scheduled task

The screenshot shows a web interface for configuring a scheduled task. The title is "System / Schedule Task". There are five tabs: "Reboot", "Record Backup", "CDR Backup", "Config Backup" (which is active), and "Log Backup". Below the tabs, there are four configuration items: "Status" with a dropdown menu set to "Disable"; "Interval" with a dropdown set to "1" and the unit "Day"; "Execution Time" with two dropdowns set to "0" for "Hour" and "0" for "Min"; and "Backup to Server" with an unchecked checkbox. At the bottom, there are two buttons: "Save" and "Reset".

5.3.12 Email

On the **System** → **Email** interface, you can configure a email client on NEXO 1024E1, which can be used to connection test for sending and receiving mail. But on top of that, SMTP, IMAP and POP 3 services need to be enabled for the email client.

When the email client is used with SMS routing, email and SMS are bound, which brings great convenience, for example, you can receive an email , although someone is sending you an SMS message. Meanwhile, logs will be generated can be viewed on the **System** → **Email** → **Log** interface.

Figure 5- 29 Configure Email Client

System / Email

Configuration
Log

Status Enable

Username

Password

Send
 Receive

Send(SMTP)

Server Address

Port

TLS Enable

Email Address

Receive

Protocol IMAP

Server Address

Port

TLS Enable

Folder

Message Query Interval(min)

Message Valid Time Range Within 5 minutes

Numbers of Message Per Receive

| | |
|------------------------------|---|
| Username | Enter the address of email client |
| password | The password or authorization code of the email client |
| Server Address | The Address of the SMTP server, supported by the email client |
| Protocol | Choose IMAP or POP3. When POPS is selected, TLS port is 995 by default. |
| Message Query Interval (min) | The time interval to check whether there is a new email. |
| Message Valid Time Range | Only those emails received during this time range are addressed. |

| | |
|-------------------------------|--|
| Number of Message Per Receive | The maximum number of emails that are received at one time. If the number exceeds, they will be received in batches. |
|-------------------------------|--|

5.3.13 FTP Server

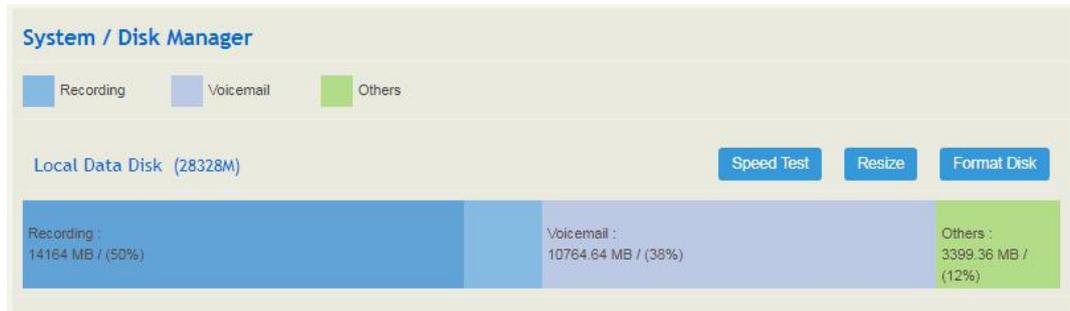
On the **System** → **FTP Server** interface, you can enable the FTP server function of NEXO 1024E1 and configure related parameters such as username, password and access permissions. You can connect FTP clients to this FTP server and then access those files (like recording files and system logs) that are open on the NEXO 1024E1 through the 21 port.

Figure 5- 30 Configure FTP Server

5.3.14 Disk Manager

On the **System** → **Disk Manager** interface, you can view the memory usage of device. The memory are divided into three categories, including voicemail(38%), recording (50%) and Others(12%). You can also re-division the proportion of each category or execute formatting on this interface.

Figure 5- 31 Disk Manager

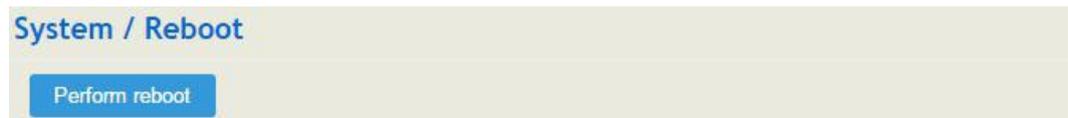


Note: NEXO 1024E1 currently does not support U disk/SD card.

5.3.15 Reboot

On the **System** → **Reboot** interface, you can click **Perform Reboot** to reboot the NEXO 1024E1 gateway. After the device is rebooted, those configurations that have been saved will remain unchanged.

Figure 5- 32 Reboot Device



5.4 Network

NEXO 1024E1 provides three RJ45 ports, namely GE0, GE1 and Console ports. GE0 is a network interface for access to Ethernet; GE1 is a management port for PC access to management equipment; a console port is used for access commands line.



5.4.1 Setting

On the **Network** → **Setting** interface, you can set the IP address of GE0 port.

The GE0 port of NEXO 1024E1 can only be a static IP address (an IP address outside the 192.168.12.1/24 network segment), and the IP address of the GE1 port is 192.168.11.1.

Static IP Address:

Static IP address is a semi-permanent IP address and remains associated with a single computer over an extended period of time. This differs from a dynamic IP address, which is assigned *ad hoc* at the start of each session, normally changing from one session to the next. If you choose static IP address, you need to fill in the following information:

- IP Address: the IP address of the GE0 port of the NEXO 1024E1;
- Netmask: the net mask of the router connected the NEXO 1024E1;
- Default Gateway: the gateway IP address of the router connected the NEXO 1024E1;
- Use custom DNS server: the IP address of the DNS server

Figure 5- 33 Static IP address of GE0 port

The screenshot displays a 'Network / Setting' configuration page. It is divided into two sections: 'GE0' and 'GE1'. Each section contains several input fields for network parameters. At the bottom of the page are three buttons: 'Cancel', 'Save', and 'Reset'.

| Port | IP Address | Netmask | Default Gateway | Preferred DNS server | Alternate DNS server | MTU | Metric |
|------|-------------|---------------|-----------------|----------------------|----------------------|------|--------|
| GE0 | 172.28.4.35 | 255.255.0.0 | 172.28.1.1 | 172.28.1.1 | 61.139.2.69 | 1500 | 9 |
| GE1 | 10.10.10.35 | 255.255.255.0 | 10.10.10.1 | 172.28.1.1 | 61.139.2.69 | 1000 | 10 |

The GE0 port of NEXO 1024E1 is a static IP address: 192.168.12.1/24. The PC can be connected to the GE1 port with a network cable and enter 192.168.12.1 in the browser to access the NEXO 1024E1 Web interface (the PC needs to be configured with a static IP address, Such as: 192.168.12.100).

5.4.2 Access Control

The access ports of Web, Telnet and SSH, as well as relevant on-off controls, can be configured on the Network → Access Control interface.

Figure 5- 38 Access Control

The screenshot shows the 'Network / Access Control' configuration page. It is divided into sections for 'Web Server', 'HTTP', and 'SSH'. Under 'Web Server', there is a section for 'HTTP' with 'Enable' checked, 'HTTP Port' set to 80, and 'HTTPS Port' set to 443. Below that is a section for 'SSH' with 'Enable' checked and 'Port' set to 22. At the bottom are 'Cancel', 'Save', and 'Reset' buttons.

5.4.3 Firewall

You can choose to enable the firewall function and adds filtering rules such as protocol/IP address/port /MAC address to accept or reject packets that meet the filtering rules from passing through the firewall.

Figure 5- 39 Firewall



Configuration Procedures:

- 1) Select **On** in the drop-down box on the right of **Filter Rules Control**
- 2) Select filter action, accept or reject;
- 3) Click the **New** button;
- 4) Fill in information of filter rule (Filtering rule information: IP, port, mac address can all be empty, no judgment will be made when it is empty, otherwise all three must be meet after configuration);
- 5) Click the **Save** button to save the configuration.

Note:



: Edit information for the corresponding filter rule.



: Delete the corresponding filter rule.

/*: Information of Source or Destination is not completely filled in.

Figure 5- 40 Create Filter Rule

Table 5- 12 Explanation of Parameters for Filter Rule

| | |
|------------------|---|
| Source IP | The source IP address that you want NEXO 1024E1 to accept or reject. It is the source IP address of the message; it can also be a string of IP addresses, for example, 172.16.11.1/15. |
| Source Port | The source port of host which the accepted or rejected IP address belongs to |
| Source MAC | The source mac of the host which the accepted or rejected IP address belongs to |
| Destination IP | The destination IP address that you want NEXO 1024E1 to accept or reject. It is the destination IP address of the message; it can also be a string of IP addresses, for example, 152.16.11.11/19. |
| Destination Port | The destination port of host which the accepted or rejected IP address belongs to |
| Action | Choose accept or reject |

5.4.4 Diagnostics

On the **Network** → **Diagnostics** interface, you can use three network utilities including Ping, Traceroute and Nslookup to diagnose the network, and can capture data packages of the available network ports.

Figure 5- 45 Network Diagnostics

The screenshot displays the 'Network / Diagnostics' interface. It is divided into two main sections: 'Network Utilities' and 'Network Capture'.
The 'Network Utilities' section contains three input fields, each followed by a button: 'Ping', 'Traceroute', and 'Nslookup'.
The 'Network Capture' section includes several configuration options:
- 'Network Interface': A dropdown menu currently set to 'GE0'.
- 'Logical Type': A dropdown menu currently set to 'OR'.
- 'Source IP': An empty text input field.
- 'Source Port': An empty text input field.
- 'Destination IP': An empty text input field.
- 'Destination Port': An empty text input field.
- 'Protocol': A set of radio buttons for 'TCP', 'UDP', 'ICMP', and 'ARP', all of which are currently unselected.
At the bottom of the 'Network Capture' section is a blue 'Start' button.

● Ping

Ping is used to examine whether a network works normally through sending test packets and calculating response time.

Instructions for using Ping:

1. Enter the IP address or domain name of a network, a website or a device in the input box of Ping, and then click **Ping**.
2. If related messages are received, it means the network works normally; otherwise, the network is not connected or is connected faultily.

● Traceroute

Traceroute is used to determine a route from one IP address to another.

Instruction for using Traceroute:

1. Enter the IP address or domain name of a destination device in the input box of Traceroute, and then click **Traceroute**.
2. View the route information from the returned message.

- **Nslookup**

Nslookup (Name Server Lookup) is a network command-line tool to obtain domain name of internet or to diagnose the problems of DNS.

Instruction for using Nslookup:

1. Enter a domain name and then click **Nslookup**.
2. View the DNS information from the returned message.

- **Network Capture**

On the following interface, you can capture data packages of the available network ports. You can also set source IP, source port, destination IP or destination port to capture the packages that you want.

There is a "and"/" or "logical type. The "and" relationship can only capture a one-way message, and "or" relationship to fetch the interaction message between a particular IP.

Note: If there are multiple source or destination IP addresses, please use ‘|’ to separate them, for example, 172.16.115.12|172.16.115.15.

After package capturing is completed, save the captured packages on a computer and then use a tool to analyze them.

5.4.5 DDNS

On the "Network -> DDNS" interface, users can use NEXO 1024E1 as a dynamic domain name client and configure parameters to map the IP address of the network where NEXO 1024E1 is located to the domain name resolution server.

DDNS (Dynamic Domain Name Server) is to map the dynamic IP address to a static domain name resolution server, and the client program will update the currently obtained dynamic IP address to the domain name resolution when the user connects to the network.

Figure 5-4-5 Dynamic domain name configuration

Network / DDNS

DDNS Service:

Service Providers List:

Domain:

Username:

Password:

IP Source:

IP Check URL:

IP Check Period(m):

Force Update Interval(h):

Retry Interval When Fail(s):

Table: Explanation of Parameters for DDNS

| | |
|--------------------------|--|
| Service Providers List | Dynamic domain name service providers |
| Domain | Domain name applied for on the service provider website |
| Username | The user name when applying for a domain name on the service provider website |
| Password | The password when applying for a domain name on the service provider website |
| IP Source | The external address and the device address can be selected, the external address is the current network export public network IP address, and the device address is the GE0 port IP address |
| IP Check URL | Server address that detects whether the IP address is updated |
| IP Check Period(m) | Check whether the IP address has changed detection period |
| Force Update Interval(h) | Force update within the configured time interval and report the IP address to the DDNS server |

| | |
|-----------------------------|---|
| Retry Interval When Fail(s) | Set the retry interval when updating the IP address fails |
|-----------------------------|---|

5.4.6 Static Route

On the **Network** → **Static Route** interface, you can configure static routes for the network.

Figure 5- 52 Create New Static Route

The screenshot shows a web-based configuration interface for creating a new static route. The title is 'Network / Static Route / New'. The form contains the following fields and values:

- Index:** 1
- Name:** Static Route-1
- Target IP:** 192.168.1.102
- Netmask:** 255.255.255.0
- Gateway:** 172.16.1.5
- Interface:** WAN
- Status:** Enable

At the bottom of the form, there are three buttons: 'Cancel', 'Save', and 'Reset'.

Table 5- 16 Explanation of Parameters for Static Route

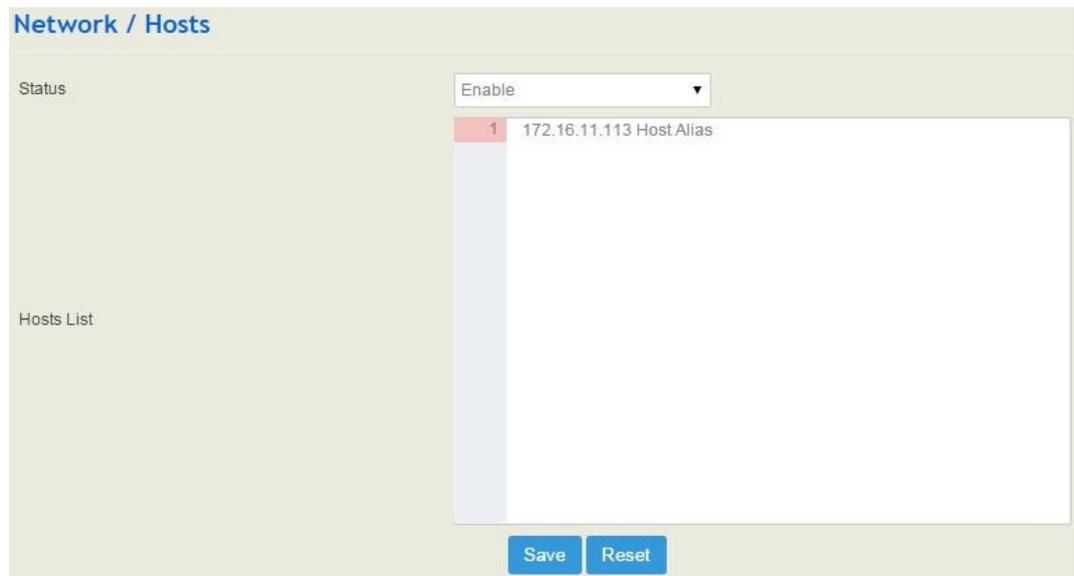
| | |
|-----------|---|
| Name | The name of the static route |
| Target IP | The destination IP address of the static route |
| Netmask | The netmask of the static route, default: 255.255.255.0 |
| Gateway | The IP address of the outbound gateway of the static route |
| Interface | The outbound interface of the static route, namely WAN port or LAN port |
| Status | The static route is enabled or disabled |

5.4.7 Hosts

On the **Network** → **Hosts** interface, you can add a host file. After being enabled the hosts file, you can visit the corresponding host by inputting the alias or domain name of the host. The format of the hosts file is as follows: IP address host alias/domain name.

The hosts file contains the mapping relationship between IP address and host name/alias /domain name. And the mapping relationship allows quick and convenient access to the host.

Figure 5- 53 Enable Hosts File



5.4.8 Fail2ban

Fail2ban is used to scan system logs and update firewall rules to reject the IP addresses that show malicious signs (for example, too many login failures) for a specified amount of time.

On the Network → Fail2ban interface, you can configure rules for Fail2ban. Fail2 ban is generally targeted SSH and SIP. The Fail2ban of NEXO 1024E1 only works on the GE0 port, and does not disable the IP of the 192.168.11.1/24 network segment of the GE1 port.

Figure 5- 54 Configure Fail2ban Rules

Network / Fail2ban

SSH

Status

Ban Duration(second)

Max Retry Duration(second)

Max Retry

White List +

Black List +

SIP

Status

Ban Duration(second)

Max Retry Duration(second)

SIP Register Max Retry

SIP Invite Max Retry

White List +

Black List +

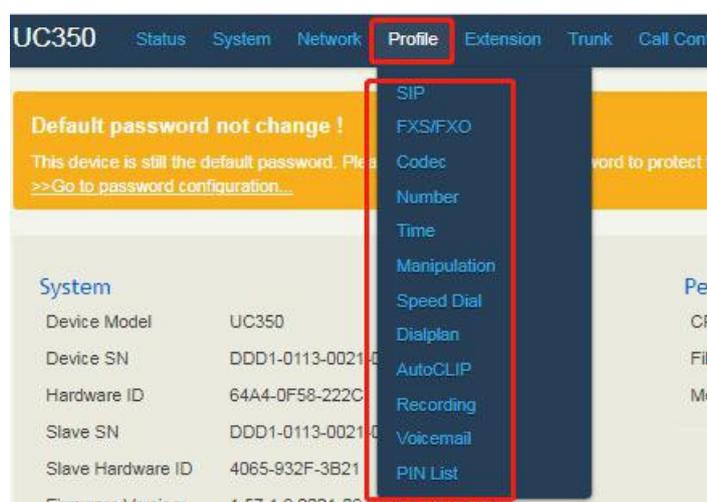
| SSH/SIP | |
|------------------------------|--|
| Ban Duration (Second) | The time period during which the IP addresses that conform to the banning rule or are in the blacklist are prohibited. Range: 60-315360000 seconds |
| Max Retry Durations (second) | The time period during which the maximum retries have been executed and then the corresponding IP address will be banned. For example, if this parameter is set as 60 seconds and the maximum number of retries is set as 10, an IP address will be banned in case that it has tried 10 times during 60 seconds. Range: 5-3600 |
| Max Retry | The maximum number of retries during a specific time. For example, if this parameter is set as 10 and the max retry duration is set as 60 seconds, an IP address will be banned in case that it has tried 10 times during 60 seconds. Range: 5-3600 |
| White List | Those IP addresses that are in the white list will not be banned by Fail2ban. |

| | |
|------------|---|
| Black List | Those IP addresses that are in the black list will not be banned by Fail2ban. |
|------------|---|

Note: If the SSH/SIP sent by an IP has an exception such as "The network is reachable but not responding", you can go to the **Status → Fail2ban** interface to check whether the IP address is banned or not.

5.5 Profile

The Profile menu includes the following sub-menus: SIP, FXS/FXO, Codec, Number, Time, Manipulation, Dialplan, Recording, Voicemail etc.



5.5.1 SIP

On the **Profile → SIP** interface, you can set SIP information such as listening port, which will be used in extension and trunk. Multiple SIP profiles can be configured for one NEXO 1024E1 device, so you can choose different SIP profiles according to different needs.

Figure 5- 55 Configure SIP Profile

Profile / SIP / New

| | |
|---|-----------------|
| Index | 3 |
| Name | |
| Local Listening Interface | GE0 |
| Local Listening Port | 5080 |
| Progress Timeout(s) | 50 |
| DTMF Send Type | RFC2833 |
| RFC2833-PT | 101 |
| Detect Inband When Call in IVR | Off |
| Process DTMF as Hold/Unhold | Off |
| PRACK | Off |
| Session Timer | Off |
| Extension Register Lock | Off |
| Trunk Reg Num to the Same Addr per Second | 1 |
| Caller Number Source | From: User Part |
| Called Number Source | To: User Part |
| Inbound Codec Negotiation Priority | Remote |

| | | |
|------------------------------|-----------------------------|---|
| Inbound Codec Profile | 1-< default > | ▼ |
| Outbound Codec Profile | 1-< default > | ▼ |
| CNG(Comfort Noise Generator) | On | ▼ |
| Detect Extension is Online | Off | ▼ |
| Ignore ACK | Off | ▼ |
| BLF | On | ▼ |
| CID Header | Off | ▼ |
| Allow Unknown Call | Off | ▼ |
| Inbound Source Filter | 0.0.0.0/0 | ⊕ |
| QoS | Off | ▼ |
| Signal Encryption | Off | ▼ |
| RTP Encryption | Off | ▼ |
| User Agent | Hostname / Full Firmware \t | ▼ |
| Timer T1(ms) | 500 | |
| Timer T2(ms) | 4000 | |
| Timer T4(ms) | 4000 | |
| Timer T1X64(ms) | 32000 | |
| <p>Cancel Save Reset</p> | | |

| | |
|------------------------------|---|
| Inbound Codec Profile | 1-< default > |
| Outbound Codec Profile | 1-< default > |
| CNG(Comfort Noise Generator) | On |
| Bypass Media(SIP to SIP) | Off |
| Proxy Media(SIP to SIP) | Off |
| Detect Extension is Online | Off |
| Ignore ACK | Off |
| BLF | On |
| Allow Unknown Call | Off |
| Inbound Source Filter | 0.0.0.0/0  |
| QoS | Off |
| User Agent | Hostname / Full Firmware Ver |
| Encryption | Off |
| Timer T1(ms) | 500 |
| Timer T2(ms) | 4000 |
| Timer T4(ms) | 4000 |
| Timer T1X64(ms) | 32000 |

Table 5- 19 Explanation of Parameters for SIP Profile

| | |
|---------------------------|--|
| Name | The name of the SIP profile |
| Local Listening Interface | The local listening interface of this SIP profile. It can be GE0 port and GE1 port. If the SIP profile is used by a SIP trunk, the interface filled in here is the listening port for the SIP trunk. |
| Local Listening Port | The local listening port of this SIP profile. If the SIP profile is used by a SIP trunk, the port filled in here is the listening port for the SIP trunk. |
| Progress Timeour(s) | If the parameter is set as 50 seconds, it means that the call will be considered as timeout in case that no one answers the call during 50 seconds. |

| | |
|---|--|
| DTMF Send Type | DTMF is short for Dual Tone Multi Frequency. There are three DTMF modes, including SIP Info, INBAND, RFC2833 |
| RFC2833-PT | RFC2833 payload coding |
| Detect Inband When Call in IVR | After enabling, the DTMF sent by the caller inband is supported in the IVR |
| Process DTMF as Hold/Unhold | By default, this parameter is off. When it is set as on, DTMF will be addressed as call hold/unhold. |
| PRACK | Provisional Response ACKnowledgement |
| Session Timer | <p>Session Expires: The validity period of a SIP session. When a SIP session times out, an invite message needs to be sent to refresh the session, otherwise, the session ends; It is 1800 seconds by default</p> <p>Min Session Expires: the minimum validity period to respond to a SIP session.</p> <p>Session Refresh Method: re-INVITE or UPDATE</p> |
| Extension Register Lock | When enabled, only the first successfully registered client is allowed to register |
| Trunk Reg Num to the Same Addr per Second | When multiple trunk are registered to the same address, please set the interval for sending register messages during registration, |
| Caller Number Source | <p>From: User Part : to obtain the caller number from the user part contained in the 'From' field.</p> <p>From: Display Name: to obtain the caller number from the display name contained in the 'From' field.</p> <p>To: User Part: to obtain the caller number from the user part contained in the 'To' field.</p> <p>Contact: User Part: to obtain the caller number from the user part contained in the 'Contact' field.</p> |
| Called Number Source | <p>From: User Part : to obtain the called number from the user part contained in the 'From' field.</p> <p>From: Display Name: to obtain the called number from the display name contained in the 'From' field.</p> <p>To: User Part: to obtain the called number from the user part contained in the 'To' field.</p> <p>Contact: User Part: to obtain the called number from the user part contained in the 'Contact' field.</p> |
| Inbound Codec Negotiation Priority | <p>To take the remote device or the local device as priority for inbound codec negotiation</p> <p>Assume local device supports PCMA, PCMU, G.729 and</p> |

| | |
|------------------------------|---|
| | G.723, while the remote device supports G.723 and G.729 If remote device is taken as codec negotiation priority, G.723 will be the codec mode, since the remote device supports G.723 and G.729 and G.723 is prior to G.729 |
| Inbound Codec Profile | The codec supported by SIP for inbound calls |
| Outbound Codec Profile | The codec supported by SIP for outbound calls |
| CNG(Comfort Noise Generator) | This function is used to generate background noise for the call when there is a short silence during the call, which sounds very comfortable |
| Detect Extension is Online | Whether to detect the SIP extension using this SIP profile is online or not |
| Ignore ACK | After enabling, the gateway will not retransmit <i>200 OK</i> if the remote end does not send an ACK, otherwise it will retransmit at intervals. |
| BLF | After enabling, you can monitor the working status of other extension through the preset indicator lights on a specified extension. The indicator lights will show different states according to the status of the monitored number. |
| CID Header | Add the CID header to the invite message sent by the gateway |
| Allow Unknown Call | If this function is enabled, incoming calls from unknown sources are allowed. Unknown sources are those IP addresses that do not fall into the source range configured for SIP trunks or SIP extensions |
| Inbound Source Filter | The source of inbound calls, which is allowed. It can be an IP address or a network segment. If it is a network segment, the format is 172.16.0.0/16 or 172.16.0.0/255.255.0.0, which means calls from the network segment of 172.16 is allowed to come in. 0.0.0.0 means calls of any source is allowed to come in |
| QoS | Whether to enable QoS. QoS is a technology used to solve network delay or congestion |
| Signal Encryption | After enabling, the gateway will transmit signaling via TLS |
| RTP Encryption | Select encrypted SRTP for RTP stream transmission. SRTP is a secure real-time transmission protocol to ensure the security of voice communication, SRTP is a secure real-time transmission protocol to ensure the security of voice communication |
| User Agent | Then content of the 'user agent' field in SIP packets |
| Timer T1 | The value of timer T1 in SIP protocol. Default value is 500ms |
| Timer T2 | The value of timer T2 in SIP protocol. Default value is 4000ms |

| | |
|------------------|--|
| TimerT4 | The value of timer T4 in SIP protocol. Default value is 5000ms |
| Timer T1X64 (ms) | The value of timer T1X64 in SIP protocol. Default value is 32000ms |

Description for SIP server

SIP server is the main part of the VoIP network, responsible for establishing all SIP calls. SIP server is also called SIP proxy server or registration server. Both IPPBX and softswitch can act as the role of SIP server. and NEXO 1024E1 is set in the "Extension-SIP" interface to act as this role.

Note:For more details on how to configure SIP server, please refer to **5.7.1**

5.5.2 FXS/FXO

On the **Profile → FXS/FXO** interface, you can configure the driving parameters of FXS port and FXO port, including tone standard, dial timeout, ring timeout, hook-flash detection, DTMF parameters, CID-related parameters, impedance, dialplan and so on.

Figure 5- 56 FXS/FXO Profile

The screenshot displays two configuration tables. The first table, titled 'Profile / FXS', lists two profiles with columns for Index, Slot, Name, Tone Group, Digit Timeout(s), Dial Timeout(s), Call Out Ring Timeout(s), and Call In No Answer Timeout(s). The second table, titled 'Profile / FXO', lists one profile with columns for Index, Slot, Name, Tone Group, Digit Timeout(s), Dial Timeout(s), Detect Caller ID, and Dial Mode. Both tables include information icons, edit icons, and delete icons for each row, and a 'New' button at the bottom right of each table.

| Profile / FXS | | | | | | | |
|---------------|-------|---------|------------|------------------|-----------------|--------------------------|------------------------------|
| Index | Slot | Name | Tone Group | Digit Timeout(s) | Dial Timeout(s) | Call Out Ring Timeout(s) | Call In No Answer Timeout(s) |
| 1 | 1 2 3 | default | China | 5 | 10 | 55 | 55 |
| 2 | 0 | slot0 | China | 5 | 10 | 55 | 55 |

| Profile / FXO | | | | | | | |
|---------------|---------|---------|------------|------------------|-----------------|---------------------------|-----------|
| Index | Slot | Name | Tone Group | Digit Timeout(s) | Dial Timeout(s) | Detect Caller ID | Dial Mode |
| 1 | 0 1 2 3 | default | China | 5 | 15 | Detect before ring/5000ms | DTMF |

Click , and corresponding configuration interface will pop up.

Figure 5- 57 Configure FXS Parameters

Profile / FXS / Edit

| | |
|------------------------------|--|
| Index | 1 |
| Slot | <input checked="" type="checkbox"/> Slot 1 <input checked="" type="checkbox"/> Slot 2 <input checked="" type="checkbox"/> Slot 3 |
| Name | <input type="text" value="default"/> |
| Tone Group | <input type="text" value="China"/> ▼ |
| Digit Timeout(s) | <input type="text" value="5"/> |
| Dial Timeout(s) | <input type="text" value="10"/> |
| Call Out Ring Timeout(s) | <input type="text" value="55"/> |
| Call In No Answer Timeout(s) | <input type="text" value="55"/> |
| Flash Detection | <input checked="" type="checkbox"/> |
| Min Time (ms) | <input type="text" value="100"/> |
| Max Time (ms) | <input type="text" value="400"/> |
| FlashHook Operation Mode | <input type="text" value="Mode 1"/> ▼ |
| DTMF Parameters | |
| DTMF Send Interval(ms) | <input type="text" value="100"/> |
| DTMF Duration(ms) | <input type="text" value="100"/> |
| DTMF Gain | <input type="text" value="0dB"/> ▼ |
| CID Send Mode | <input type="text" value="DTMF"/> ▼ |

| | |
|--|---------|
| Impedance | 600 Ohm |
| REN(Ringer Equivalency Number) | 1 |
| Send Polarity Reverse | On |
| Long Line Support | On |
| Call Waiting Tone | |
| Call Waiting Tone Duration(ms) | 800 |
| Call Waiting Tone Gap(ms) | 2000 |
| Call Waiting Tone Repeat Count | 5 |
| Auto Gain Control | Off |
| Dialplan | Off |
| <input type="button" value="Cancel"/> <input type="button" value="Save"/> <input type="button" value="Reset"/> | |

Table 5- 20 Explanation of FXS Parameters

| | |
|-------------------------------|--|
| Name | The name of this FXS profile |
| Tone Group | The national standard of dialing tone, busy tone and ring tone; default value is China |
| Digit Timeout (s) | The timeout value for dialing a digit of a telephone number; When the time of dialing a digit exceeds this value, the system will think the dialing has completed; Default value is 4 seconds |
| Dial Timeout (s) | The timeout value for dialing the first telephone number after off-hook; Default value is 10 seconds |
| Call Out Ring Timeout (s) | The timeout value for the ringing of the analog phones of the FXS port when when calling out |
| Call In No Answer Timeout (s) | The timeout value for ending a call which goes out through the FXS port, when nobody answers the call. |
| Flash Detection | Whether to enable flash-hook detection; If flash detection is not enabled, the press on flash-hook will be ignored and won't be processed. |
| Min Time(ms)/ Max Time(ms) | Min Time: when flash-hook detection is enabled, if the time of the press on the flash-hook is less than this minimum time, the press will be ignored and won't be processed. Max Time: when flash-hook detection is enabled, if the time of the press on the flash-hook is longer than this maximum time, the |

| | |
|--------------------------------|---|
| | phone will be hanged up. |
| DTMF Send Interval(ms) | The minimum interval between the sending of two DTMF tone DTMF: Dual Tone Multi Frequency |
| DTMF Gain | Signal gain of DTMF |
| DTMF Duration (ms) | The minimum duration of a DTMF tone |
| Impedance | The impedance (SLIC) matched with analog phones |
| REN(Ringer Equivalency Number) | The equivalent number of ringing phones. It is used to determine how many devices can be connected by telephone lines. Range: 1 to 4 |
| Send Polarity Reverse | If polarity reverse is on, call tolls will be calculated based on the changes in voltage. If polarity reverse is off, you need to set the time for offhook detect and call tolls will be calculated starting from the set time. |
| Dialplan | The rules for dialing. The NEXO 1024E1-1G1S1O device supports regular expression. Please make reference to Profile → Dianplan section. |

Figure 5- 58 Configure FXO Parameters

Profile / FXO / Edit

| | |
|---------------------------------|---|
| Index | 1 |
| Slot | <input checked="" type="checkbox"/> Slot 0 <input checked="" type="checkbox"/> Slot 1 <input checked="" type="checkbox"/> Slot 2 <input checked="" type="checkbox"/> Slot 3 |
| Name | <input type="text" value="default"/> |
| Tone Group | <input type="text" value="China"/> ▼ |
| Digit Timeout(s) | <input type="text" value="5"/> |
| Dial Timeout(s) | <input type="text" value="15"/> |
| Call Out Ring Timeout(s) | <input type="text" value="55"/> |
| Call In No Answer Timeout(s) | <input type="text" value="55"/> |
| Detect Caller ID | <input type="text" value="Detect before ring"/> ▼ |
| Dial Mode | <input type="text" value="DTMF"/> ▼ |
| One Stage Dialing | <input type="text" value="On"/> ▼ |
| Add # As Ending Key | <input type="text" value="Off"/> ▼ |
| Delay Offhook(ms) | <input type="text" value="500"/> |
| Dial Delay(ms) | <input type="text" value="400"/> |
| Detect Polarity Reverse | <input type="text" value="On"/> ▼ |
| Delay Time after FXO Offhook(s) | <input type="text" value="61"/> |

DTMF Parameters

DTMF Send Interval(ms)

DTMF Duration(ms)

DTMF Gain

Onhook when

BusyTone Detect

Current Detected

Current Disconnect Threshold

DC Impedance

BusyTone Detect Parameters

Busy Tone Cadence

Detect Tone counts

Detect Tone Delta(ms)

On->Off Energy Threshold

Off->On Energy Threshold

Table 5- 21 Explanation of FXO Parameters

| | |
|-------------------------------|---|
| Name | The name of this FXO profile |
| Tone Group | The national standard of dialing tone, busy tone and ring tone; default value is China |
| Digit Timeout (s) | The timeout value for dialing a digit of a telephone number; When the time of dialing a digit exceeds this value, the system will think the dialing has completed; Default value is 4 seconds |
| Dial Timeout (s) | The timeout value for dialing the first telephone number after off-hook; Default value is 10 seconds |
| Call Out Ring Timeout (s) | The timeout value for the ringing of the analog phones of the FXS port when calling out |
| Call In No Answer Timeout (s) | The timeout value for ending a call which goes out through the FXS port, when nobody answers the call. |

| | |
|------------------------------|---|
| Detect Caller ID | Detect before ring: the CID will be shown before ringing; otherwise, CID will be displayed after ringing; Detect after ring: the CID will be shown after ringing; otherwise, CID will be displayed before ringing Off: the CID will not be shown |
| Dial Delay(ms) | The delay time of dialing. Default value is 400ms |
| Detect Polarity Reverse | Whether to enable 'detect polarity reverse'. If 'detect polarity reverse' is on, call tolls will be calculated based on the changes in voltage. If 'detect polarity reverse' is off, you need to set the time for offhook delay and call tolls will be calculated starting from the set time. |
| DTMF Send Interval(ms) | The minimum interval between the sending of two DTMF tone DTMF: Dual Tone Multi Frequency |
| DTMF Gain | Signal gain of DTMF |
| DTMF Duration (ms) | The minimum duration of a DTMF tone |
| Current Disconnect Threshold | This current threshold is used to determine whether a phone is onhook. |
| Detect Tone counts | Set the number of busy notes to check |
| Detect Tone Delta | Set the error size to check the busy tone |
| Dialplan | The rules for dialing. The NEXO 1024E1 device supports regular expression. Please make reference to Profile → Dianplan section. |

5.5.3 Codec

NEXO 1024E1 supports voice codec and video codec, including G729, G723, PCMU PCMA, eat. You can adjust the priority of these codec according to you needs.

Figure 5- 59 Add or Delect Codec Profile

Profile / Codec / New

Index: 2

Name:

Audio Codec: PCMA 20ms

Video Codec: Off

Buttons: Cancel, Save, Reset

 : Edit codec profile.

 : Delete the corresponding codec profile or a codec mode.

 : Create a new codec profile.

5.5.4 Number

On the **Profile → Number** interface, you can set a prefix for calling numbers or called numbers. When the prefix of a calling number or a called number matches the set prefix, the call will be passed to choose a route.

Figure 5- 60 Number Profile

Profile / Number

| Index | Name | Caller Prefix | Caller Length | Called Prefix | Called Length |
|-------|----------|---------------|---------------|---------------|---------------|
| 1 | Number 1 | 0755 | * | * | * |

Buttons: Edit, Delete, New

 : Edit number profile.

 : Delete the corresponding number profile

Click , and you will see the following interface:

Figure 5- 61 Create Number Profile

Profile / Number / New

Index: 1

Name: test

Caller Number

Length: 5

Prefix:

| | |
|---|---|
| 1 | # |
| 2 | * |

Called Number

Length: 5

Prefix:

| | |
|---|---|
| 1 | # |
| 2 | * |

Cancel Save Reset

Table 5- 22 Explanation of Number Parameters

| | |
|-------------------------|--|
| Name | The name of the number profile |
| Prefix of Caller Number | The prefix of the calling number. It supports multiple prefixes, multiple rules for "or" relationships .It supports regular expression |
| Prefix of Called Number | The prefix of the called number. It supports regular expression.It Supports multiple prefixes, multiple rules for "or" relationships. |
| Length | The length of the calling number or called number. For example, : 4 6 7 means the calling number or called number must be 4 digits, 6 digits or 7 digits except the prefix |

Regex (Regular Expression) Syntax

Table 5- 3 Explanation of frequently-used metacharacters in Regex

| | |
|------|--|
| ^ | Matches the starting position in a number string. For example, ^134 matches the numbers starting with 134 |
| \$ | Matches the ending position of a string. For example, 2\$ matches the numbers ending with 2. |
| | Separates alternate possibilities. For example, 2 3 4 means 2,3or 4. |
| \ | Marks the next character as a special character, a literal, a backreference, or an octal escape |
| [] | Matches a single character that is contained within the bracket. For example, [123] matches 1, 2, or 3. [0-9] matches any digit from "0" to "9". |
| [^] | Matches any one character except those enclosed in []. For example, [^9] matches any character except 9. |
| . | Matches any single character except the newline character. For example, 3.4 matches 314, 324, 334, 344. |
| ? | Indicates there is zero or one of the preceding element. For example, colou?r matches both color and colour |
| * | Indicates there is zero or more of the preceding element. For example, ab*c matches ac, abc, abbc, abbbc, and so on. |
| + | Indicates there is one or more of the preceding element. For example, ab+c matches abc, abbc, abbbc, and so on, but not ac |
| \d | Mark any digit, equal to [0-9] |
| \D | Mark any character that is not a digit, equal to [^0-9] |
| \s | Mark any blank character such as a space or a tab. |
| \S | Mark any character that is not a blank character |

Examples:

| | |
|---------------------|--|
| ^0755 | Matches the phone numbers with starting digits of 0755. |
| ^0755 ^8899 ^0110 | Matches the phone numbers with starting digits of 0755, 8899 or 0110. |
| ^[1][358][0-9]{9}\$ | Matches the phone numbers with the first digit as 1, the second digit as 3, 5 or 8, the left nine digits as any of 0 to 9. |

Note: the matching of number prefix also supports some digits that are not conform to the format of regular expression. For example, 0755 matches the numbers starting with 0755, and 0755|8899|0110 matches the numbers starting with 0755, 8899 or 0110.

5.5.5 Time

On the **Profile → Time** interface, you can set a time period for calls to choose routes. If the local time when a call is initiated falls into the set time period, the call will be passed to choose the corresponding route.

Click the **New** button, and you will see the following interface:

Figure 5- 63 CreateTime Profile

The screenshot shows a web interface titled "Profile / Time / New". It contains the following fields and controls:

- Index:** A dropdown menu with the value "1" selected.
- Name:** A text input field.
- Date Period:** A text input field with a green plus icon to its right.
- Weekday:** A row of checkboxes for "Mon", "Tue", "Wed", "Thu", "Fri", "Sat", and "Sun".
- Time Period:** A text input field with a green plus icon to its right.
- Buttons:** "Cancel", "Save", and "Reset" buttons are located at the bottom right of the form.

Table 5- 23 Explanation of Time Parameters

| | |
|-------------|--|
| Name | The name of the number profile |
| Date Period | Configure the starting date and ending date of a period : Add a date period : Delete a date period |
| Weekdate | Choose a weekdate |
| Time Period | Choose the starting time and ending time of a day |

5.5.6 Manipulation

Number manipulation refers to the change of a called number or a caller number during calling process when the called number or the caller number matches the preset rules.

Click the **New** button, and you will see the following interface:

Figure 5- 64 Create Manipulation Profile

Profile / Manipulation / New

Index: 1

Name: [Text Input]

Caller:

Delete Prefix Count: [Text Input]

Delete Suffix Count: [Text Input]

Add Prefix: [Text Input]

Add Suffix: [Text Input]

Replace by: [Text Input]

Called:

Buttons: Cancel, Save, Reset

Table 5- 24 Explanation of Manipulation Parameters

| | |
|-------------------------------------|---|
| Name | The name of this manipulation profile |
| Delete Prefix Count | The number of digits that are deleted from the left of the caller number or calling number |
| Delete Suffix Count | The number of digits that are deleted from the right of the caller number or calling number |
| Add Prefix | The prefix added to the caller number or the calling number |
| Add Suffix | The suffix added to the caller number or the calling number |
| Replace by | The number which replace the caller number or the calling number |
| <input checked="" type="checkbox"/> | If the checkbox on the right of Caller is selected, it means the caller number will be manipulated; if the checkbox on the right of Called is selected, it means the called number will be manipulated. |

Note: During number manipulation, deletion rules are carried out first, followed by adding rules. If 'Replace by' has been set, deletion rules and adding rules are invalid.

5.5.7 Speed Dial

On the **Profile → Speed Dial** interface, you can set one-digit or two-digit speed dial numbers for FXS/SIP calls. For example, if the short number (speed dial number) is set as 1, the long number is set as 8000, and this speed dial profile is applied to an FXS/SIP extension, the FXS/SIP extension only needs to dial 1 and the call will be directed to the extension number of 8000.

Figure 5- 65 Add Speed Dial Profile

The screenshot displays a web-based configuration page titled "Profile / Speed Dial / New". The page contains the following elements:

- Index:** A dropdown menu with the value "1" selected.
- Name:** An empty text input field.
- Abbreviated Number Table:** A table with four columns: "Name", "Short Number", "Long Number", and "Status".

| Name | Short Number | Long Number | Status |
|----------------------|----------------------|----------------------|---|
| <input type="text"/> | <input type="text"/> | <input type="text"/> | Enable <input type="button" value="⊕"/> |
- Buttons:** "Cancel", "Save", and "Reset" buttons are located at the bottom right of the form.

5.5.8 Dialplan

Dialplan is used for number dialing of calls through FXS ports. It supports Regular Expression (Regex) and DigitMap.

Figure 5- 66 Add Dialplan

Profile / Dialplan / New

Index

Name

Dialplan

Regex (Regular Expression) Syntax

| | |
|------|--|
| ^ | Matches the starting position in a number string. For example, ^134 matches the numbers starting with 134 |
| \$ | Matches the ending position of a string. For example, 2\$ matches the numbers ending with 2. |
| | Separates alternate possibilities. For example, 2 3 4 means 2,3or 4. |
| \ | Marks the next character as a special character, a literal, a backreference, or an octal escape |
| [] | Matches a single character that is contained within the bracket. For example, [123] matches 1, 2, or 3. [0-9] matches any digit from "0" to "9". |
| [^] | Matches any one character except those enclosed in []. For example, [^9] matches any character except 9. |
| . | Matches any single character except the newline character. For example, 3.4 matches 314, 324, 334, 344. |
| ? | Indicates there is zero or one of the preceding element. For example, colou?r matches both color and colour |
| * | Indicates there is zero or more of the preceding element. For example, ab*c matches ac, abc, abbc, abbbc, and so on. |
| + | Indicates there is one or more of the preceding element. For example, ab+c matches abc, abbc, abbbc, and so on, but not ac |
| /d | Mark any digit, equal to [0-9] |
| /D | Mark any character that is not a digit, equal to [^0-9] |
| /s | Mark any blank character such as a space or a tab. |

| | |
|----|--|
| /S | Mark any character that is not a blank character |
|----|--|

Examples of Regex Syntax:

| | |
|---------------------|--|
| ^0755 | Matches the phone numbers with starting digits of 0755. |
| ^0755 ^8899 ^0110 | Matches the phone numbers with starting digits of 0755, 8899 or 0110. |
| ^[1][358][0-9]{9}\$ | Matches the phone numbers with the first digit as 1, the second digit as 3, 5 or 8, the left nine digits as any of 0 to 9. |

DigitMap Syntax:

| | | |
|-------------------|-------|---|
| Supported Objects | Digit | 0-9 |
| | T | Timer |
| | DTMF | A digit, a timer, or one of the symbols of A, B, C, D, #, or * |
| Range | [] | One or more DTMF symbols enclosed in the [], but only one DTMF symbol can be selected |
| Range | () | One or more expressions enclosed the (), but only one can be selected |
| Separator | | Separate expressions or DTMF symbols. |
| Subrange | - | Two digits separated by hyphen (-) which matches a ny digit between and including the two digits. |
| Wildcard | x | Matches any digit of 0 to 9 |
| Modifiers | . | Matches 0 or more times of the preceding element |
| Modifiers | ? | Matches 0 or 1 times of the preceding element |

Examples of DigitMap Syntax

| | |
|---------------------------|---|
| (13 15 18)xxxxxxxx | Matches the phone numbers with stating digits as 13, 15 or 18 and the left nine digits as any of 0 to 9 |
| [2-8] xxxxxx 13xxxxxxxx | Matches the phone numbers starting with any digit of 2 to 8 and the left six digits as any of 0 to 9; or matches the phone numbers starting with 13 and the left nine digits as any of 0 to 9 |

5.5.9 AutoCLIP

AutoCLIP is mainly used to SIP trunks, FXO trunks and VoLTE trunks and it helps record the outgoing and incoming calls of a trunk.

Figure 5- 67 Configure AutoCLIP Rule

| | |
|----------------------|--|
| Index | The index of AutoCLIP profile |
| Name | The name of AutoCLIP profile |
| Record Strategy | You can choose missed calls or all calls. If missed calls is selected, NEXO 1024E1 will record the missed calls of the trunk. If all calls are selected, all the calls going through the trunk will be recorded |
| Record Expire (hour) | The validity period of a record. For example, if this parameter is set as 2 hours, the record will be valid in 2 hours since the record is generated. During the validity period, if there is coming call for the extension number contained in the record, the call will directly led to the extension without routing. |
| Delete Used Record | By default, this parameter is disabled. If this parameter is selected, those records that have been used to match extension number or trunk will be deleted. |
| Match Outgoing Trunk | By default, this parameter is enabled. If this parameter is enabled, those calls going through the trunks in the |

| | |
|--|---------------------------------------|
| | record can coming in without routing. |
| Enable Number Matching Rules When It Fails | Enable number matching rules |

5.5.10 Recording

On the **Profile** → **Recording** interface, you can choose SD card or Udisk as master/slave storage location.

How to Record Calls:

Configure a recording profile (or choose one of the two default recording profiles), and then add it to a SIP/FXS route. When there are calls going through the route and match the recording profile, the calls will be recorded.

Figure 5- 68 Add recording profile

The screenshot shows the 'Profile / Recording' configuration page. It has two tabs: 'Configuration' and 'Recording List'. The 'Recording List' tab is active, displaying a table with the following data:

| Index | Name | Strategy | Recording Direction | Stereo | Min Duration(s) | Silence Detect | |
|-------|---------------|-------------------------------|---------------------|--------|-----------------|----------------|---|
| 1 | auto_record | Auto Recording After Answer | Inbound & Outbound | Off | 1 | Off-/- |   |
| 2 | manual_record | Manual Recording After Answer | Inbound & Outbound | Off | 1 | Off-/- |    |

At the bottom right of the table, there is a blue 'New' button.

Figure 5- 69 View Recording List

The screenshot shows the 'Profile / Recording' configuration page, specifically the 'Recording List' tab. The page contains a form for editing a recording profile. The fields and their values are as follows:

- Index:** 2
- Name:** manual_record
- Strategy:** Manual Recording After Answer
- Stop Recording Manually:** On
- Recording Direction:** Inbound & Outbound
- Stereo:** Off
- Min Duration(s):** 1
- Silence Detect:** Off

At the bottom of the form, there are three buttons: 'Cancel', 'Save', and 'Reset'.

| | |
|---------------------|--|
| Index | The index of the recording profile. Range: 1-32 |
| Name | The name of the recording profile, used to identify the recording profile |
| Strategy | Auto Recording after Answer: start recording after the callee pick up the phone. Ban Recording: either caller or callee enables his function, and then the call in both directions will not be recorded. Manual Recording after Answer: press *3 to start recording after the callee answers the call. |
| Recording Direction | Inbound & Outbound: If this recording profile is added to FXS/SIP extension, both inbound and outbound calls will be recorded. Inbound: If this recording profile is added to FXS/SIP extension, only inbound calls will be recorded. Outbound: If this recording profile is added to FXS/SIP extension, only outbound calls will be recorded. Note: If this recording profile is added to routing, this parameter is invalid and all calls going through the routing will be recorded. |
| Stereo | When enabled, the file size will be twice that of mono for the same call duration |
| Min Duration | If the actual recording time is shorter than this value, the recording file will not be saved. |
| Silence Detect | When silence is detected, no recording will be done during muting |

You can click **Recording List** to view the recording files which show the caller/called number, recording duration and so on. You can also play, download or delete the recording files on this interface.

| Index | Time | Caller | Source | Called | Destination | Duration | Operation |
|-------|---------------------|--------|--------------------|--------|--------------------|----------|-----------|
| 1 | 2021-07-13 10:47:39 | 660800 | SIP Trunk/trunk-32 | 556796 | SIP Trunk/trunk-66 | 00:49 | |
| 2 | 2021-07-13 10:47:35 | 663587 | SIP Trunk/trunk-32 | 557444 | SIP Trunk/trunk-66 | 00:31 | |
| 3 | 2021-07-13 10:47:31 | 662387 | SIP Trunk/trunk-32 | 550754 | SIP Trunk/trunk-66 | 00:41 | |
| 4 | 2021-07-13 10:47:28 | 664960 | SIP Trunk/trunk-32 | 558659 | SIP Trunk/trunk-66 | 01:02 | |
| 5 | 2021-07-13 10:47:27 | 665349 | SIP Trunk/trunk-32 | 558820 | SIP Trunk/trunk-66 | 01:03 | |
| 6 | 2021-07-13 10:47:26 | 665343 | SIP Trunk/trunk-32 | 552304 | SIP Trunk/trunk-66 | 00:46 | |
| 7 | 2021-07-13 10:47:24 | 669992 | SIP Trunk/trunk-32 | 559662 | SIP Trunk/trunk-66 | 00:37 | |
| 8 | 2021-07-13 10:47:24 | 663923 | SIP Trunk/trunk-32 | 559065 | SIP Trunk/trunk-66 | 00:34 | |
| 9 | 2021-07-13 10:47:19 | 665528 | SIP Trunk/trunk-32 | 553334 | SIP Trunk/trunk-66 | 00:42 | |
| 10 | 2021-07-13 10:47:17 | 664958 | SIP Trunk/trunk-32 | 551454 | SIP Trunk/trunk-66 | 00:35 | |

5.5.11 Voicemail

On the **Profile** → **Voicemail** interface, you can configure the location, number and duration of a voice mail.

How to use voice mail:

Navigate to **Extension** → **SIP** or **Extension** → **FXS** interface, enable the voice mail function, and the voice mail will be activated when the call times out..

Figure 5- 70 Configure Voicemail Profile

Profile / Voicemail

Configuration
Message List

| | |
|------------------------------|---|
| Max Messages Per User | <input type="text" value="50"/> |
| Maximum of Login Attempts | <input type="text" value="3"/> |
| Maximum of Operation Failure | <input type="text" value="3"/> |
| Min Message Time(sec) | <input type="text" value="3"/> ▼ |
| Max Message Time(min) | <input type="text" value="2"/> ▼ |
| Auto Play New Message | <input checked="" type="checkbox"/> |
| Play CID Number | <input checked="" type="checkbox"/> |
| Play from Latest Message | <input type="checkbox"/> |
| Play Message Date | <input type="text" value="Before Playing Message"/> ▼ |

| | |
|------------------------------|--|
| Max Message Per User | If this maximum number of messages is reached, a prompt voice “the mail box is full” will be played. |
| Maximum of Login Attempts | If this maximum number of attempts (by dialing *170*2 to log in the voice mailbox) is reached, the call will hang up. |
| Maximum of Operation Failure | When a call enters into the voice mailbox and the caller dial inexistent DTMF repeatedly, the caller will be forced to log out the voice mailbox after the repetition times exceed this value. |
| Min Message Time (second) | The minimum duration of a voice mail |
| Max Message Time (second) | The maximum duration of a voice mail. |
| Auto Play New Message | If this parameter is on, new messages will be played automatically. If it is off, a prompt voice “please dial 1 to listen to new message” will be given. |
| Play CID Number | If this parameter is on, the caller number will be played together with messages. |
| Play from Latest Message | If this parameter is on, the latest messages will be played first. |
| Play Message Date | When to play message date. You can choose ‘Before Playing Message’, ‘After Playing Message’ and ‘Never’. |

You can click **Message List** to view the voicemail files which show the caller/called number, message duration and so on. You can also play, download or delete the message files on this interface.



The screenshot shows a web interface titled "Profile / Voicemail". It has two tabs: "Configuration" and "Message List", with "Message List" being the active tab. Below the tabs is a table with the following columns: Index, Time, Caller, Source, Called, Destination, Message Type, Duration, and Operation. The table is currently empty, displaying the text "No result found" in the center.

| Index | Time | Caller | Source | Called | Destination | Message Type | Duration | Operation |
|-----------------|------|--------|--------|--------|-------------|--------------|----------|-----------|
| No result found | | | | | | | | |

5.6 Extension

5.6.1 SIP

On the **Extension** → **SIP** interface, you can configure the SIP accounts registered in the NEXO 1024E1 by SIP clients (hereby NEXO 1024E1 is regarded as a SIP server).

Figure 5- 71 Configure SIP Extension

Extension / SIP / Edit

SIP Extension SIP Phone

| | |
|--|------------|
| Index | 1 |
| Name | 350 |
| Extension | 350 |
| Password | ***** |
| Classification Tag | |
| Outbound CID | |
| DID | |
| Max Concurrent Register | 1 |
| Max Concurrent Call | 1 |
| Ring Timeout(s) | 50 |
| Original Called Number Location(Send INVITE) | Off |
| Register Source | Any |
| Register User Agent | Any |
| Call Pickup | Ring Group |
| Call Waiting | Off |
| Do Not Disturb | Off |

| | |
|----------------------------|--|
| Call Forward Unconditional | Off |
| Call Forward Unregister | Off |
| Call Forward Busy | Off |
| Call Forward No Reply | Off |
| NAT | Off |
| Call In Filter | Off |
| Call Out Filter | Off |
| Speed Dial | Off |
| Allow Being Monitored | <input type="checkbox"/> |
| Monitor Mode | Disable |
| Voicemail | On |
| Password | ***  |
| Message Forward Email | <input type="checkbox"/> |
| Recording Profile | 1-< auto_record > |
| SIP Profile | 1-< GE0_Default > |
| Status | Enable |

Table 5- 25 Explanation of Parameters for SIP Extension

| | |
|-------------------------|---|
| Name | The name of this SIP extension |
| Extension | The SIP account of the extension registered in NEXO 1024E1 by a SIP client |
| Password | The password of the SIP account registered in NEXO 1024E1 by a SIP client |
| Classification Tag | Labels for extension classification |
| Outbound CID | After the outgoing caller number is configured, the caller number dialed from the SIP extension is replaced with the number configured here |
| DID | Direct Inward Dialing; if the called number is same with DID, the call will be directly forwarded to the extension, rather than choosing a route. Users can set multiple DID. |
| Max Concurrent Register | Number of clients that can register online at the same time |

| | |
|--|---|
| Max Concurrent Call | The number of concurrent calls that can be made at the same time |
| Ring Timeout(s) | The ringing timeout period for incoming calls to this extension, the default value is 50. If the extension does not go off-hook within 50s after ringing, the device will initiate disconnection |
| Original Called Number Location(Send INVITE) | When sending INVITE, configure the location of the original called number |
| Register Source | If Any is chosen, all SIP clients are allowed to register the SIP account of this extension; if Specified is chosen, only the SIP client with the specified IP address or network segment is allowed to register the SIP account of this extension. For example, 172.16.0.0/16 means the register source is 172.16 |
| Register User Agent | Filter the useragent field in the register message during registration |
| Call Pickup | After configuration, the designated call can be picked up (ring group/local extension, the default is the ring group) |
| Call Waiting | If a calling party places a call to a called party which is otherwise engaged, and the called party has the call waiting feature enabled, the calling party will hear an IVR voice. |
| Do Not Disturb | If Do Not Disturb feature is enabled, calls cannot reach the called party. |
| Call Forward Unconditional | If Call Forward Unconditional feature is enabled, all coming calls will be forwarded to a preset number. |
| Call Forward Unregister | When the SIP extension is not registered, you can transfer all the calls to the set number |
| Call Forward Busy | If Call Forward Busy feature is enabled, new coming call will be forwarded when the corresponding local port is busy. |
| Call Forward No Reply | If Call Forward No Reply feature is enabled, calls will be forwarded when nobody answer the calls during a specified period. |
| NAT | If NAT is enabled, the IP address of SIP extension in LAN will be turned into the outbound IP address of public network, thus making NAT traversal possible |
| Call In Filter | When you breathe in to SIP, you match the relevant filter conditions |
| Call Out Filter | When the SIP is called out,The filter conditions are matched |
| Speed Dial | Configuration for Speed dial |
| Voicemail | Choose to on or off the voice mail |
| Password | Configure the password for logging in to the extension's voice mail |
| Message Forward Email | Configure the e-mail address for voice mail messages, and make sure that the e-mail is normally |
| Recording Profile | When recording is enabled, FXS calls will be recorded according to the recording rules |

| | |
|-------------|--|
| SIP Profile | The SIP profile that is selected for the extension |
| Status | If it is enabled, this SIP extension is registered to NEXO 1024E1; Otherwise the SIP extension is not registered |

5.6.2 FXS

On the **Extension** → **FXS** interface, you can configure the parameters of the FXS extension.

Figure 5- 72 Configure Parameters of FXS Extension

Extension / FXS / New

| | | |
|----------------------------|---------------------------------|---|
| Slot | 3 | ▼ |
| Port | 0 | ▼ |
| Extension | <input type="text"/> | |
| DID | <input type="text"/> | + |
| Hot Line | Off | ▼ |
| Ring Timeout(s) | <input type="text" value="50"/> | |
| Call Pickup | Ring Group | ▼ |
| Call Waiting | Off | ▼ |
| Do Not Disturb | Off | ▼ |
| Call Forward Unconditional | Off | ▼ |
| Call Forward Busy | Off | ▼ |
| Call Forward No Reply | Off | ▼ |
| Call In Filter | Off | ▼ |
| Call Out Filter | Off | ▼ |
| Speed Dial | Off | ▼ |
| Allow Being Monitored | <input type="checkbox"/> | |
| Monitor Mode | Disable | ▼ |

| | |
|------------------------------|------------------|
| Voicemail | Off |
| Recording Profile | Off |
| Work Mode | Voice |
| Voice Output Mod | Telephone |
| Gain Configure Mode | General Settings |
| TX Gain(IP->PSTN) | +4dB |
| RX Gain(PSTN->IP) | 0dB |
| CID Send Timing | Send After RING |
| Delay Timeout After Ring(ms) | 1000 |
| Status | Enable |

Table 5- 26 Explanation of Parameters for FXS Extension

| | |
|----------------------------|--|
| Extension | The extension account of FXS port, which is used to register |
| DID | Direct Inward Dialing; if the called number is same with DID, the call will be directly forwarded to the extension, rather than choosing a route. |
| Hot line | If hotline is enabled, calls will directly go to the hotline number |
| Ring Timeout(s) | The ringing timeout period for incoming calls to this extension, the default value is 50. If the extension does not go off-hook within 50s after ringing, the device will initiate disconnection |
| Call Pickup | After configuration, the designated call can be picked up (ring group/local extension, the default is the ring group) |
| Call Waiting | If a calling party places a call to a called party which is otherwise engaged, and the called party has the call waiting feature enabled, the calling party will hear an IVR voice. |
| Do Not Disturb | If 'Do Not Disturb' feature is enabled, calls cannot reach the called party. |
| Call Forward Unconditional | If 'Call Forward Unconditional' feature is enabled, all coming calls will be forwarded to a preset number. |
| Call Forward Busy | If 'Call Forward Busy' feature is enabled, new coming call will be forwarded when the corresponding local port is busy. |
| Call Forward No Reply | If 'Call Forward No Reply' feature is enabled, calls will be forwarded when nobody answer the calls during a specified period. |

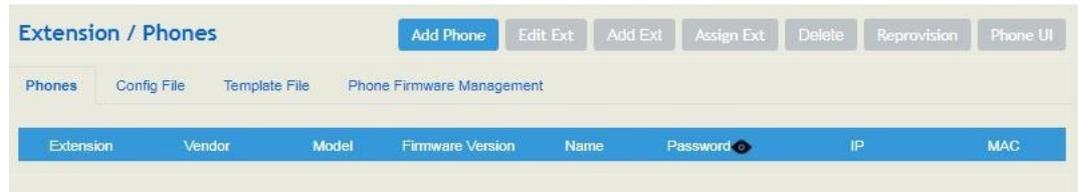
| | |
|------------------------------|---|
| Input Gain | The receiving gain of the FXS port |
| Output Gain | The sending gain of the FXS port |
| Work Mode | The working mode of the FXS port, including Voice and POS |
| Call In Filter | When a call is given to the FXS port of NEXO 1024E1, the call will not be connected to the FXO port if it is in the blacklist |
| Call Out Filter | When a call goes out from the FXS port of NEXO 1024E1, the call cannot go out if it is in the blacklist |
| Speed Dial | Configuration for Speed dial |
| Voicemail | Choose to on or off the voice mail |
| Recording Profile | When recording is enabled, FXS calls will be recorded according to the recording rules |
| Work Mode | Configure the working mode of the FXS port |
| Voice Output Mode | Configure the voice output mode of the FXS port |
| Gain Configure Mode | Select the gain configuration mode of the FXS port (general settings and advanced settings), TX gain and RX gain are newly added to advanced settings than general settings. |
| TX Gain(IP→PSTN) | The volume level of the remote end during a call; that is, adjusting the "TX gain" will affect the volume of the sound heard by the remote end. |
| RX Gain(PSTN→IP) | The volume level of the user during the call; that is, adjusting the "RX gain" will affect the level of sound you hear. |
| CID Send Timing | Set the phone caller ID display before ringing or after ringing |
| Delay Timeout After Ring(ms) | CID sending timing is configured when sending after ringing. Set how long the phone will ring before sending CID. When "CID send timing" is configured to "send after ring", you need to configure the delay timeout, that is, how long to ring before sending the CID |
| Status | If it is on, this FXS extension can be used, otherwise, the FXS extension is unavailable. |

5.6.3 Phones

On the "Extension -> Phone" interface, the user can deliver the configuration to the phone according to the template file.

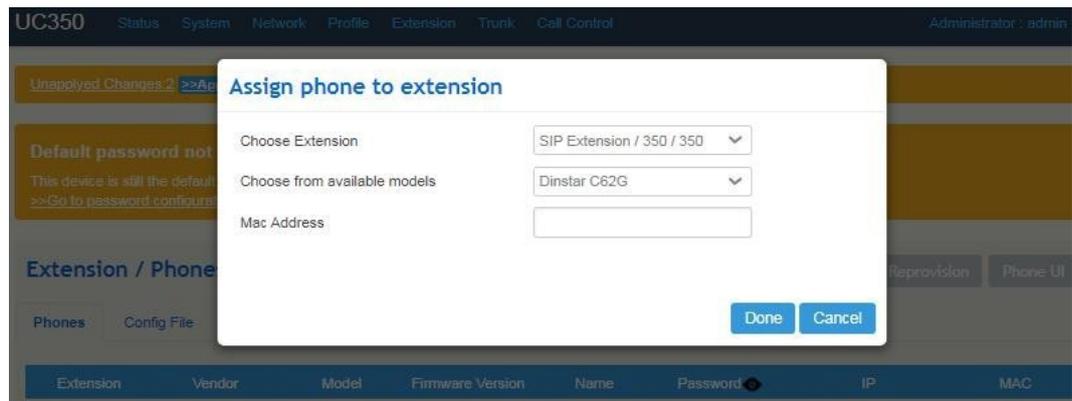
After enabling PNP, the phone will periodically send a Subscribe message to the multicast address. If the PBX receives the multicast message, it will list the phone models in the PBX configuration list.

Figure 5-6-3 Phones



If the phone is in the configuration list of UC, after selecting, you can assign the phone to an existing extension, or you can create a new extension through "Add phone".

Figure: Add phone



At this time, UC will automatically generate a configuration file suitable for the phone model, and send a SIP NOTIFY message to the phone, carrying the download address of the configuration file in the body, and notify the phone to download. After the phone receives it, you can use the assigned extension to register.

5.6.4 Ring Group

On the **Extension → Ring Group** interface, you can group FXS extension and SIP extension(s) together and set strategy for choosing the FXS extension and which SIP extension to ring under a ring group. The ring group function is widely used in call centers.

Figure 5- 73 Configure Ring Group

Table 5- 27 Explanation of Parameters for Ring Group

| | |
|----------------------------|---|
| Name | The name of this ring group |
| Members Select | Select the FXS extension and an SIP extension or several SIP extensions;  : Add an extension to the ring group  : Delete an extension from the ring group |
| Strategy | The strategies for choosing which SIP extension to ring, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random |
| Ring Group Number | The number of the ring group; it is generally the same with DID. |
| DID | Same with Ring Group Number; it is optional to fill in |
| Ring Time (5-60s) | The duration of ring when there is a coming call. Range: 5s to 60s |
| When No Answer Transfer To | When none of the members in the ring group answer, you can transfer the call to a specified extension or hang up |

Note: If ring group function has been set, the call forwarding function is unavailable.

5.6.5 Paging Group

On the **Extension → Paging Group** interface, you can group SIP extensions into a paging group and then if there calls given from FXS/FXO/SIP to the paging group, the calls will be led to one extension of the paging group according to the preset strategy.

Figure 5- 74 Configure Paging Group

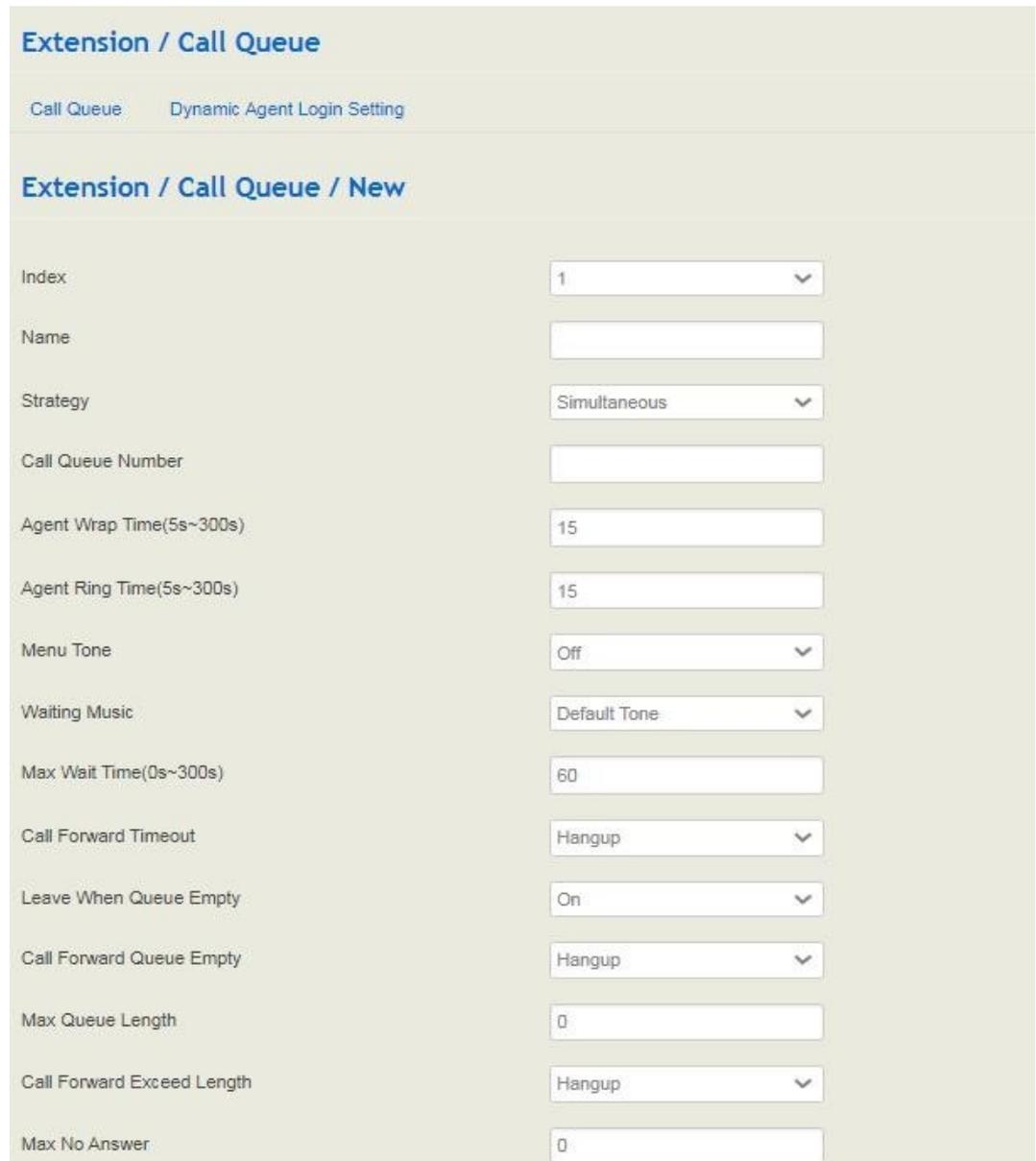
| | |
|---------------------|---|
| Name | The name of this paging group |
| Paging Group Number | The number of the paging group. When there calls given from FXS/FXO/SIP to this number, the calls will be led to one extension of the paging group according to the preset strategy. |
| Strategy | Include one-way paging and two-way intercom. one-way paging : members of the paging group only can listen to the voice of presenter and cannot answer the call. two-way intercom : members of the paging group can have conversation with the presenter, but members cannot talk to each other. |
| Members Select | Select the SIP extensions that are added into the paging group. An SIP extension cannot exist in two paging groups at the same time. Click  to add an SIP extension to the paging group; Click  to delete an SIP extension from the paging group. |

5.6.6 Call Queue

On the "Extension -> Call Queue" interface, the user can add the local extension to a queue. When calling into the call queue, the system will transfer the call to the queue member/agent to answer the call according to the strategy.

For example, when a large number of customers call in at the same time, and the customer service staff is limited, queue the incoming and play a voice waiting tone or custom music file. At the same time, the agent can answer the call according to the preset call queue strategy.

Figure 5-6-6 New call queue



| Extension / Call Queue | |
|------------------------------|-----------------------------|
| Call Queue | Dynamic Agent Login Setting |
| Extension / Call Queue / New | |
| Index | 1 |
| Name | |
| Strategy | Simultaneous |
| Call Queue Number | |
| Agent Wrap Time(5s~300s) | 15 |
| Agent Ring Time(5s~300s) | 15 |
| Menu Tone | Off |
| Waiting Music | Default Tone |
| Max Wait Time(0s~300s) | 60 |
| Call Forward Timeout | Hangup |
| Leave When Queue Empty | On |
| Call Forward Queue Empty | Hangup |
| Max Queue Length | 0 |
| Call Forward Exceed Length | Hangup |
| Max No Answer | 0 |

Enable Position Announcement Off ▼

Members Select FXS Extension / 300 ▼ +

Cancel
Save
Reset

| | |
|--------------------------|---|
| Name | The name of the call queue |
| Strategy | <p>Calls into the queue, the agents ring according to the strategy.</p> <p>Simultaneous: The agents ring together</p> <p>Sequential Mode: When there is no incoming call, a new user calls in, each time it will ring sequentially from the first agent)</p> <p>Random: one is randomly selected for ringing</p> <p>Memory rotation mode : When there is no incoming call , a new user calls in, and the ringing starts from the next agent who hangs up last before</p> <p>Max idle time : Idle time, namely the time from the end of the agent's last call to the present; ringing in the order from longest to shortest time</p> <p>Min talk time :The ringing starts from the least to the most according to the times of calls</p> |
| Call Queue Number | The number of the call queue can be called into the queue |
| Agent Wrap Time(5s-300s) | The interval time between the next ringing after the agent call hangs up |
| Agent Ring Time(5s-300s) | If the ringing exceeds the time, it will call to the next agent |
| Menu Tone | The first menu tone the remote end hears when calling in |
| Waiting Music | The remote end waits for the agent to answer the waiting tone after calling in |
| Max Wait Time(0s-300s) | The longest time the caller waits. The caller will exit after this time. 0 means no limit, but it should be noted that this time is not necessary. For example, an agent is ringing and the caller has reached the timeout. The caller will wait until the agent answers or hang up after the timeout. |
| Call Forward Timeout | If the caller times out, other actions can be configured |
| Leave When Queue Empty | If there is no agent in the queue, it will exit the queue (if there is ON-Break, it is still an agent), and the call transfer will be performed when the queue is empty |

| | |
|------------------------------|---|
| Max Queue Length | How many users are waiting, those connected are not counted, 0 means no limit, hang up if the maximum number of queues is exceeded |
| Max No Answer | If the times that the agent does not answer is exceeded, it will enter On-Break state, in this state, it will not be ringing again until the agent answer |
| Enable Position Announcement | Timely notify the user of the waiting position in the queue, the first one does not notify |

5.7 Trunk

5.7.1 SIP

SIP trunk can realize the connection between NEXO 1024E1 and IPPBX or SIP servers.

Figure 5- 75 Configure SIP Trunk

Trunk / SIP / New

| | |
|--------------------------|----------------------|
| Index | 6 |
| Name | <input type="text"/> |
| Address | <input type="text"/> |
| Port | <input type="text"/> |
| Outbound Proxy | <input type="text"/> |
| Port | <input type="text"/> |
| Transport | UDP |
| Register | Off |
| From Header User Part | Caller's Number |
| From Header Display Name | Caller's Number |
| From Header Host | Local Address |
| Heartbeat | Off |
| AutoCLIP Profile | Off |
| DNIS | Off |
| SIP Profile | 1-< GE0_Default > |
| Outbound Codec Profile | 1-< default > |
| Extra Param | <input type="text"/> |

Table 5- 28 Explanation of Parameters for SIP Trunk

Status

| | |
|----------------|---|
| Name | The name of the SIP trunk |
| Address | The IP address or domain name of the peer SIP devices or servers |
| Port | The SIP listening port of the peer SIP devices or servers; 5060 is the default port |
| Outbound Proxy | If outbound proxy is used, enter the IP address or domain name of the |

| | |
|--------------------------|---|
| | proxy server |
| Port | If outbound proxy is used, enter the listening port of the proxy server |
| Transport | Transport protocol: TCP or UDP |
| Register | If it is on, the SIP trunk will send register request to the peer device |
| From Header Display Name | Choose the registered username or the true caller ID for the 'from header' of the invite message when a call goes out. |
| Heartbeat | If heartbeat in on, heartbeat (options) messages will be sent to examine the connection with servers; The default value is 'Off' |
| AutoCLIP Profile | AutoCLIP is mainly used to SIP trunks / FXO trunks and it helps record the outgoing and incoming calls of a trunk. |
| SIP Profile | he SIP profile of the SIP Trunk; make reference to Profile → SIP section |
| Status | If it is enabled, it means the SIP Trunk can be used; otherwise, the SIP trunk is unavailable |

Note:

If NEXO 1024E1 is regarded as a terminal and intends to register to a server, you need to configure a SIP trunk connecting NEXO 1024E1 and the server, and then enable register for the SIP trunk.

If the FXS of NEXO 1024E1 intends to register to a server, you need to configure a SIP trunk connecting NEXO 1024E1 and the server, then enable register for the port and designate the SIP trunk to it.

5.7.2 FXO

FXO Trunk interconnects the PSTN with NEXO 1024E1. Calls from the PSTN can come into the gateway and calls can go out from the gateway to search telephone numbers under the PSTN.

Different from the FXO ports of other gateways, the FXO port of NEXO 1024E1 only allows one-time dialing, which means called numbers needs to be dialed directly for calls that go out from the FXO port.

Figure 5- 76 Configure FXO Trunk

Table 5- 29 Explanation of Parameters for FXO Trunk

| | |
|---------------------|--|
| Port | The FXO portl number |
| Autodial Number | The autodial number of the FXO port when there are incoming calls |
| AutoCLIP Profile | AutoCLIP is mainly used to SIP trunks, FXO trunks ans VoLTE trunks and it helps record the outgoing and incoming calls of a trunk. |
| Work Mode | Configure the working mode of the FXO port |
| Voice Output Mode | Configure the voice output mode of the FXO port |
| Gain Configure Mode | Select the gain configuration mode of the FXO port (general settings and advanced settings), TX gain and RX gain are newly |

| | |
|------------------|---|
| | added to advanced settings than general settings. |
| TX Gain(IP→PSTN) | The volume level of the remote end during a call; that is, adjusting the "TX gain" will affect the volume of the sound heard by the remote end. |
| RX Gain(PSTN→IP) | The volume level of the user during the call; that is, adjusting the "RX gain" will affect the level of sound you hear. |
| Impedance | Set the value of impedance(SLIC) for the remote FXS port |
| Hybrid | Set the value of hybrid for the remote FXS port |
| Status | If it is on, this FXO trunk can be used, otherwise, the FXO trunk is unavailable. |

- **FXO Automatch Impedance :**

The FXO Trunk interconnects the PSTN. When the value of impedance at both ends does not match, the automatch impedance function can be used to automatically adapt the impedance of the remote FXS port to ensure stable communication.

Conditions:

Only the ports in the online can be tested.

Steps:

- 1) Navigate to "**Trunk > FXO > Automatch Impedance**";
 - 2) Configure the mode and parameters of automatch impedance , and click "Start";
 - 3) After the test is completed, the Impedance and Hybrid values are displayed.
-

Figure 5- 77 FXO Automatch Impedance

The screenshot shows the 'Trunk / FXO' configuration interface with the 'Automatch Impedance' tab selected. The configuration includes the following parameters:

- FXO:** Slot 1/Port 1/OFFLINE (dropdown menu)
- Digit Timeout(s):** 9 (text input)
- Automatch Mode:** Simple (dropdown menu)
- Current Impedance:** 600 Ohm (display field)
- Current Transhybrid Balancing Param:** 0 (display field)
- DTMF:** 1234567890 (text input)

Buttons for 'Start' and 'Save' are visible at the bottom right of the configuration area.

Explanation of Parameters for Automatch Impedance

| | |
|-------------------------------------|--|
| FXO | Select FXO port |
| Digit Timeout(s) | Set a time for dial timeout |
| Automatch Mode | Automatch mode: simple/standard/accurate (The higher the mode, the higher the accuracy and the longer it takes). |
| Current Impedance | Display the current impedance value of the FXO port (just display, cannot be modified) |
| Current Transhybrid Balancing Param | Display the current hybrid of the FXO port ((just display, cannot be modified) |
| DTMF | When automatch impedance, the DTMF value sent by the local FXO port is 1234567890 by default, which can be modified. |

- **Busytone Learning**

The FXO Trunk interconnects the PSTN. When the busy tone at both ends does not match, it may cause problems such as incorrect or unsuccessful detection of the busy tone. At this time, you can use the busy tone learning function to automatically adapt to the busy tone of the remote FXS port ensures that the busy tone can be correctly identified.

Conditions:

Only the ports in the online can be tested.

Steps:

1) Navigate to "**Trunk > FXO > Busytone Learning**";

Figure 5- 77 FXO Busytone Learning

The screenshot shows a web interface for configuring FXO Busytone Learning. At the top, there are three tabs: 'FXO', 'Automatch Impedance', and 'Busytone Learning'. The 'Busytone Learning' tab is active. Below the tabs, there are several input fields and buttons:

- FXO:** A dropdown menu showing 'Slot 1/Port 1/OFFLINE'.
- Current Cadence:** A text input field containing '360,340,0,0,0,0,0,0'.
- Destination Number:** A text input field containing '1234567890#' and a blue 'Start' button to its right.
- Original Cadence:** A greyed-out text input field.
- Automatch Optimum Cadence:** A white text input field.
- Save:** A grey 'Save' button at the bottom right.

Explanation of Parameters for Busytone Learning

| | |
|---------------------------|--|
| FXO | Select FXO port |
| Current Cadence | Display the current FXO trunk busy tone cadence |
| Destination Number | The destination number during busy tone learning |
| Original Cadence | Previous busy tone cadence data |
| Automatch Optimum Cadence | Optimal cadence after FXO trunk busy tone learning |

- 2) After selecting an online port, filling in the destination number, click "start";
- 3) Busy tone learning takes about 30-60s. After the learning is completed, the progress bar displays 100%.
- 4) Learning is completed, the Optimal cadence will be displayed. It will only take effect after saving.

5.7.3 E1

There are two types of E1 trunk: SS7 and PRI. The two different trunk are as follows.

- **SS7 Trunk**

Navigate to "Trunk >E1", select SS7 in the type bar.

Figure 5- 77 SS7 Trunk

Trunk / E1

[E1 Trunk](#) [E1 Param](#)

Trunk / E1 / New

| | |
|-------------------|------------------|
| Slot | 3 |
| Type | SS7 |
| E1 No | 0 |
| SS7 Trunk | |
| Protocol | ITU |
| Protocol Type | ISUP |
| SPC Format | Hex |
| OPC | 1 |
| DPC | 2 |
| Support APC | Disable |
| Network Indicator | National Network |
| Sending SLTM | Enable |
| SS7 MTP Link | |
| Channel No | 16 |
| Caller Type | Not Configured |

| | |
|--|----------------|
| Callee Type | Not Configured |
| OrgCallee Type | Not Configured |
| Numbering Plan | ISDN |
| Calling Presentation | Allowed |
| Screening indicator | User Provided |
| Called Stop sending | Disable |
| Calling Stop sending | Disable |
| SS7 CIC | |
| Start CIC No | 0 |
| AutoCLIP Profile | Off |
| <input type="button" value="Cancel"/> <input type="button" value="Save"/> <input type="button" value="Reset"/> | |

Explanation of Parameters for SS7 Trunk

| | |
|-------------------|---|
| Type | Type: SS7 and PRI |
| Protocol | Protocol standard SPC types: ITU (14 bit), ANSI (24 bit), ITU-CHINA (24 bit) |
| Protocol Type | SS7 service types: ISUP (ISDN user side) and TUP (Telephone side) |
| SPC Format | SPC format: Hexadecimal system and 14bit (3-8-3) |
| OPC | OPC: Original Point Code, usually uniformly assigned by the operator |
| DPC | DPC: Destination Point Code, usually uniformly assigned by the operator |
| Support APC | APC is required when it is enabled, and the format is the same as the SPC format. Enter the STP code provided by the operator |
| Network Indicator | Display the network indicator of SS7, including :domestic network, domestic network backup, international network, and international network backup; the default is domestic network(mainly used in China, the United States and Japan), and "international network" is usually used exchange in the office, others are according to the environment. |
| Sending SLTM | SLTM: Signalling Link Test Message |
| Channel No | The channel for establishing link 7, usually channel No. 16 or No. 1, and the default channel No. 16 |
| Caller Type | Configure the caller number type (not configured/ international/ |

| | |
|----------------------|--|
| | domestic/ user) |
| Callee Type | Configure the callee number type (not configured/ international/ domestic/ user) |
| OrgCallee Type | Configure the original callee number type (not configured/ international/ domestic/ user) |
| Numbering Plan | Configure the number plan (ISDN/ data/ telegram/ special) |
| Calling Presentation | Calling presentation (allowed /limited/ invalid/ not configured) |
| Screening Indicator | Configure screening indicator (user-provided/ network-provided) |
| Called Stop Sending | After enabled, the called number with the suffix F |
| Calling Stop Sending | After enabled, the calling number with the suffix F |
| Start CIC No | The start CIC No of E1 port |
| AutoCLIP Profile | AutoCLIP is mainly used to SIP trunks, FXO trunks and VoLTE trunks and it helps record the outgoing and incoming calls of a trunk. |

- **PRI Trunk**

Navigate to "Trunk >E1", select PRI in the type bar.

Figure 5- 77 PRI Trunk

Trunk / E1

E1 Trunk E1 Param

Trunk / E1 / New

Slot: 3

Type: PRI

PRI Trunk

Protocol: ISDN

Switch Side: User Side

Alerting Indication: ALERTING

PRI Parameter

Calling Party Numbering Plan: ISDN/Telephony numbering |

Calling Party Number Type: Unknown

Screening Indicator for Displaying Caller Number: User-provided, not screened

Screening Indicator for No Displaying Caller Number: User-provided, not screened

Called Party Numbering Plan: ISDN/Telephony numbering |

Called Party Number Type: Unknown

Information Transfer Capability: Speech

AutoCLIP Profile: Off

Explanation of Parameters for PRI Trunk

| | |
|------------------------------|---|
| Type | Type: SS7 and PRI |
| Protocol | There are two types of PRI protocol: ISDN and QSIG |
| Switch Side | "User side" and "Network side" can be chosen. When implementing a PRI circuit, the nature of E1 in the network must be different on the receiving and sending sides |
| Alerting Indication | Configure alerting indication (Alerting and progressing) |
| Calling Party Numbering Plan | 6 options, the default is "ISDN/telephone numbering plan" |

| | |
|---|--|
| Calling Party Number Type | 6 types of calling party numbers can be selected |
| Screening Indicator for Displaying Caller Number | 4 options, the default is "user-provided ,not screened" |
| Screening Indicator for No Displaying Caller Number | 4 options, the default is "user-provided ,not screened" |
| Called Party Numbering Plan | 6 options, the default is "ISDN/telephone numbering plan" |
| Called Party Number Type | 6 options, the default is "unknown" |
| Information Transfer Capability | Support voice and 3.1khz voice |
| AutoCLIP Profile | AutoCLIP is mainly used to SIP trunks, FXO trunks and VoLTE trunks and it helps record the outgoing and incoming calls of a trunk. |

- **E1 Param**

Navigate to "Trunk >E1 >E1 Param".

Figure 5- 77 E1 Trunk

Explanation of Parameters for E1Trunk

| | |
|--------------|--|
| PCM Mode | PCM (Pulse Code Modulation) Mode: ALAW or Mu LAW |
| Frame Format | The frame format of the E1 port is: DF, MF-CRC4, MF, the default is DF |

5.8 Call Control

This section is to configure routes or route groups for incoming and outgoing calls through NEXO 1024E1, as well as IVR, SMS and so on.

5.8.1 Setting

Figure 5- 79 Basic Setting of Call Control

Call Control / Setting

Voice

Disconnect call when no RTP packet

Period without RTP packet(10s~300s)

Tone

Waiting Music

Table 5- 30 Explanation of Parameters for Call Control

| | |
|------------------------------------|--|
| Disconnect call when no RTP packet | If it is enabled, and no RTP packets are received within the preset time, calls will be disconnected |
| Waiting Music | Select the waiting music |

5.8.2 Route Group

On the **Call Control** → **Route Group** interface, you can group SIP trunks, SIP extensions, FXS extension and FXO trunk together according to your needs and set strategy for choosing which trunk or extension as the destination route under a route group.

Figure 5- 80 Create Route Group

The screenshot shows the 'Call Control / Route Group / New' configuration page. It features the following elements:

- Index:** A dropdown menu with the value '3' selected.
- Name:** An empty text input field.
- Members Select:** A section containing two lists:
 - Source list 0/70:** A list of items with checkboxes, including 'SIP Trunk / trunk-200', 'SIP Trunk / trunk-30', 'SIP Trunk / trunk-32', 'SIP Trunk / trunk-66', 'SIP Trunk / UC2500', and several 'SIP Extension' entries with their respective extension numbers.
 - Target list 0/0:** An empty list box.
 - Navigation arrows (right-pointing and left-pointing) are positioned between the two lists.
 - Vertical scroll arrows are on the right side of the target list.
- Strategy:** A dropdown menu set to 'Sequence(Ascending)'.
- Buttons:** 'Cancel', 'Save', and 'Reset' buttons at the bottom.

Table 5- 31 Explanation of Parameters for Route Group

| | |
|----------------|--|
| Name | The name of the route group |
| Members Select | Select FXS extension, SIP extension, SIP trunk, FXO trunk or GSM trunk |
| Strategy | The strategies for choosing which route under the route group as the destination route, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random |

5.8.3 Route

On the **Call Control** → **Route** interface, you can configure routes for incoming calls and outgoing calls.

Figure 5- 81 Create a Route

Call Control / Route / New

Priority

Name

Condition

Source

Number Profile

Caller Number Prefix

Called Number Prefix

Time Profile

Action

Callback

Distinctive Ringtone(Alert-Info)

Manipulation

Destination

Password Type

Recording Profile

Failover Action

Table 5- 32 Explanation of Route Parameters

| | |
|----------------|---|
| Priority | The priority for choosing the route; the higher value, the lower priority |
| Name | The name of the route |
| Condition | The condition under which the route will be used |
| Source | The source of the call; it can be the FXS extension, SIP extension, FXO trunk ,GSM trunk, a customized source or any |
| Number Profile | The profile of the caller number and the called number; please make reference to the Profile → Number section.The default |

| | |
|----------------------|---|
| | value is 'Off' Note: it cannot be simultaneously used with the following parameters of 'caller number prefix' and 'called number prefix' |
| Caller Number Prefix | The prefix of caller number; it supports regular expression |
| Called Number Prefix | The prefix of called number; it supports regular expression |
| Time Profile | The profile of time during which the route can be used; make reference to the Profile → Time section |
| Action | Include manipulating number and sending call to destination |
| Callback | After enabling, the caller who configures this route will directly hang up after the incoming call, and then initiate a call to the called after the waiting time expires, and then initiate a call to the caller after the called picks up |
| Manipulation | If it is on, the caller number or called number of the route will be manipulated; make reference to the Profile → Manipulation section |
| Destination | The destination of the route |
| Recording Profile | Record according to the configured rules |
| Failover Action | The processing when a call through this route fails |

5.8.4 Feature Code

NEXO 1024E1 provides convenient telephone functions. Connect a telephone to the FXS port and dial a specific feature code, and you can query corresponding information.

The following is the corresponding function of each feature code:

Figure 5- 82 Feature Code

Call Control / Feature Code

Feature Code Service

On

Save

| Index | Feature | Key | Description | Status |
|-------|--|-------|--|---|
| 1 | Inquiry IP | *158 | Inquiry IP | Enabled   |
| 2 | Inquiry Phone Number | *114 | Inquiry Phone Number | Enabled   |
| 3 | Configure IP Address | *152* | Set IPv4 Address 192.168.1.10 by dial *152*192*168*1*10# | Enabled   |
| 4 | Configure Gateway | *156* | Set IPv4 Gateway 192.168.1.1 by dial *156*192*168*1*1# | Enabled   |
| 5 | Configure Subnet Mask | *153* | Set IPv4 Netmask 255.255.0.0 by dial *153*255*255*0*0# | Enabled   |
| 6 | Restart Device | *111 | Restart Device | Enabled   |
| 7 | Call Waiting Activate | *51 | Enable Call Waiting service | Enabled   |
| 8 | Call Waiting Deactivate | *50 | Disable Call Waiting service | Enabled   |
| 9 | Call Forwarding Uncondition Activate | *72* | Enable Call Forwarding Uncondition service.Example:*72*8000,set... | Enabled   |
| 10 | Call Forwarding Uncondition Deactivate | *73 | Disable Call Forwarding Uncondition service | Enabled   |
| 11 | Call Forwarding Busy Activate | *90* | Enable Call Forwarding Busy service.Example:*90*8000,set the ca... | Enabled   |
| 12 | Call Forwarding Busy Deactivate | *91 | Disable Call Forwarding Busy service | Enabled   |
| 13 | Call Forwarding No Reply Activate | *92* | Enable Call Forwarding No Reply service.Example:*92*8000,set th... | Enabled   |
| 14 | Call Forwarding No Reply Deactivate | *93 | Disable Call Forwarding No Reply service | Enabled   |
| 15 | DND Activate | *78 | Enable Do Not Disturb service | Enabled   |
| 16 | DND Deactivate | *79 | Disable Do Not Disturb service | Enabled   |
| 17 | Call Pickup | ** | Pick up the ringing extension, Example:**8000, pick up the extensi... | Enabled   |
| 18 | Voicemail Service | *170* | *170*1# - Leave messages, *170*2# - Play messages | Enabled   |
| 19 | Callback Service | *163 | Callback the last received call | Enabled   |
| 20 | Recording Service | *3 | Start or stop recording when manual recording | Enabled   |
| 21 | Call Park | *4 | Example: *4, you can park another part during the call. *4100, you ... | Enabled   |
| 22 | Call Monitor | *164* | *164*1 - Listen Mode, *164*2 - Whisper Mode, *164*3 - Barge-in M... | Enabled   |
| 23 | Auto Answer | *5 | Make an intercom with a specific extension user. Example: dial *51... | Enabled   |

Note:

- 1) All feature codes are enabled by default.
- 2) Any feature code can be customized and edited

5.8.5 IVR

On the **Call Control** → **IVR** interface, you can carry out specific configurations for the IVR which has been uploaded from the **System** → **Voice** interface. IVR is often used for voice prompts in call centers.

Figure 5- 83 IVR Setting

Callcontrol / IVR / New

Index: 1

Name:

Menu Tone: Off

Repeat Loops: 3

Enable Direct Extension: Off

Select Invalid Times: 3

Select Invalid Tone: Off

Destination Invalid Times: 3

Destination Invalid Tone: Off

Response Timeout(s): 10

Digit Timeout(s): 3

Response Timeout Tone: Off

Exit Tone: Off

Status: Enable

Menu

| DTMF | Tone | Destination |
|------|------|-------------|
| 0 | Off | Extension |

SIP Extension / 350

Buttons: Cancel, Save, Reset

Table 5- 33 Explanation of IVR Parameters

| | |
|-------------------------|---|
| Name | The name of the IVR |
| Menu Tone | When a call comes to the IVR, the menu tone heard |
| Enable Direct Extension | Whether to allow direct dialing of extensions during the playing of IVR |
| Repeat Loops | If it is set as '3', the call will be hanged up after the IVR has been repeated for three times during timeout. |
| Select Invalid Times | When a call comes to the IVR, according to the voice prompt, if you receive two dialings that do not match the DTMF , then the dialing is invalid, and the invalid prompt tone is played. When the invalid times is exceeded, the voice prompt: Goodbye |

| | |
|---------------------------|---|
| Select Invalid Tone | When an invalid dial is received, an invalid tone will be played |
| Destination Invalid Times | It takes effect when the direct extension is enabled. When you call into the IVR, and the entered number does not exist, the destination invalid prompt will be played. When the time of entries exceeds the set value, the voice prompt: Goodbye |
| Destination Invalid Tone | When receiving an invalid destination dial, the invalid tone will be played |
| Response Timeout(s) | When a call comes to the IVR, according to the voice prompt, the second dial is not received within the set time, the response is timed out, and the timeout tone is played |
| Digit Timeout(s) | The timeout for dialing DTMF |
| Response Timeout Tone | When the second dialing timeout, the timeout will be played after being enabled |
| Exit Tone | When exiting IVR, the exit tone will be played |
| Status | If it is disabled, the IVR cannot be seen in the destination of route. |
| Menu | <p>DTMF: It can be 0-9 quick-dial numbers, *, #, others or timeout.</p> <p>Destination: the destination of the IVR; it can be an extension or a trunk.</p> <p>For example, if DTMF is configured as 1,2,3 and others, and the telephone key that is pressed is not 1, 2 or 3, the IVR will choose the destination of 'others'.</p> <p>When the the playing of the IVR times out, and user does not press any telephone key, the IVR will choose the destination of 'timeout'.</p> <p>When the destination is a trunk, user does not need to pre-configure the called number, and the system will prompt the user to dial the called number.</p> |

5.8.6 Conference

On the "Call Control -> Conference" interface, users can create a conference room, and the caller can create a multi-party conference by dialing the number of the conference room.

Figure 5-8-6 Conference

Call Control / Conference / New

| | |
|----------------------------------|---|
| Index | 4 ▼ |
| Name | <input style="width: 90%;" type="text"/> |
| Number | <input style="width: 90%;" type="text"/> |
| Public Mode | <input type="checkbox"/> |
| Password | <input style="width: 90%;" type="password"/> 👁 |
| Administrator Password | <input style="width: 90%;" type="password"/> 👁 |
| Quiet Mode | <input type="checkbox"/> |
| Wait For Administrator | <input type="checkbox"/> |
| Play Waiting Music when Idle | <input type="checkbox"/> |
| Enable Menu | <input checked="" type="checkbox"/> |
| Invite Member or Conference Room | <input type="checkbox"/> |

Cancel
Save
Reset

Table 5- 33 Explanation of Parameters for Conference

| | |
|------------------------------|---|
| Name | The name of the conference room |
| Number | Conference room number, the extension can join the conference by dialing this number |
| Public Mode | No password is required to join the conference in public mode |
| Password | The password for users to join the conference in non-public mode |
| Administrator Password | Administrator password: the password for the administrator to join the conference in non-public mode. A conference can have multiple administrators, and the administrator password cannot be blank |
| Quiet Mode | When the quiet mode is enabled, the conference will not hear any voice |
| Wait For Administrator | Once enabled, the conference will only start after the administrator join the conference, otherwise it will be idle |
| Play Waiting Music When Idle | When the conference is in idle, the waiting tone will be played after being enabled |
| Enable Menu | After activation, everyone can use the menu through DTMF |

| | |
|----------------------------------|--|
| Invite Member Or Conference Room | After enabling the menu, non-administrators can invite members to join the conference room |
|----------------------------------|--|

Table: Conference Menu

| DTMF | Description | Notes |
|------|---|---|
| 1 | Invite members | Non-administrators need to enable configuration |
| 2 | Invite members, need to be confirmed by the invitee | Non-administrators need to enable configuration |
| 3 | Initiate a conference | Non-administrators need to enable configuration |
| 4 | Decrease the volume of the handset | |
| 6 | Increase the volume of the handset | |
| 7 | Decrease the volume of the microphone | |
| 9 | Increase the volume of the microphone | |
| * | Mute | |
| 0 | All non-administrators are muted | Administrator permissions |
| # | Exit all non-administrators from the conference | Administrator permissions |

Menu instructions:

Invite members: Invite multiple SIP extensions

- 1) After pressing 1, it will prompt to enter the number and the extension number;
- 2) The extension rings;
- 3) After the extension is connected, join the conference as a non-administrator.

Invite members (requires confirmation):

- 1) After pressing 2, it will prompt to enter the number and the extension number;
- 2) The extension rings;
- 3) After the extension is connected, you hear the prompt that you will join the conference, press 1 to join the conference as a non-administrator, press 2 or other to hang up.

Invite a conference: The conference room is activated

- 1) After pressing 3, it will prompt to enter the conference room number;
-

- 2) If there is a password, you will be prompted to enter the conference room password;
- 3) Connect to the meeting.

5.8.7 SMS Route

NEXO 1024E1 allows SMS to be sent between SIP clients, On the **Call Control → SMS Route** interface, you can establish route for these SMS.

Figure 5- 84 Create SMS Route

The screenshot shows a web interface for creating a new SMS route. The title is "Call Control / SMS Route / New". The form is organized into sections: "From" and "To".

- Priority:** A dropdown menu with the value "32".
- Name:** An empty text input field.
- From Section:**
 - Source:** A dropdown menu with the value "SIP Extension / 350".
 - Content Has the Words:** An empty text input field.
- To Section:**
 - Action:** A dropdown menu with the value "Forward".
 - Destination:** A dropdown menu with the value "SIP Extension / 350".
 - Add Prefix in Content:** A dropdown menu with the value "From \${from_user} :".
 - Add Suffix in Content:** A dropdown menu with the value "None".

At the bottom of the form, there are three buttons: "Cancel", "Save", and "Reset".

Table 5- 34 Explanation of SMS Route Parameters

| | |
|-----------------------|--|
| Priority | The priority for the SMS route; the higher value, the lower priority |
| Name | The name of the SMS route |
| Source | The source of the SMS route. It can be a trunk or an extension. It also can be a LTE SMS and USSD. |
| Content Has the Words | Match key words in text message content |
| Action | The text message action can choose whether to forward or reply |

| | |
|-----------------------|--|
| Destination | The destination of the SMS route. It can be a trunk or an extension. |
| Add Prefix in Content | The prefix of the SMS content. It is generally 'none', which means there is no prefix to be matched. |
| Add Suffix in Content | The suffix of the SMS content. It is generally 'none', which means there is no suffix to be matched. |

5.8.8 Diagnostics

In case that call cannot be connected or voice has quality problem, you can enter the **Call Control** → **Diagnostics** interface to collect fault-related information and then send it to technical support to locate fault.

Operation Procedures:

1. Select the module that need to be traced. For example, if a call from SIP to FXS has voice problem, you can select SIP message, SIP Stack and Voice, and then click the **Start** button.
2. Give a call, and come back to the **Call Control** → **Diagnostics** interface after the call ends. Then click Stop and download the tracing file.
3. In order to locate faults more quickly, you sometimes need to enter into the **System** → **Service Log** interface, click export, and then send this exported file and the tracing file to technical support,

Figure 5- 87 Call Tracing for Diagnostics

The screenshot shows the 'Call Control / Diagnostics' interface. At the top, there are two tabs: 'Call Trace' (selected) and 'SIP Test'. Below the tabs, there are two sections: 'Select the module you want to trace' and 'Select the userboard you want to trace'. In the first section, three checkboxes are checked: 'SIP Stack', 'SIP Message', and 'Voice'. In the second section, there are three rows for 'Userboard 0', 'Userboard 1', and 'Userboard 2'. Each row has an unchecked checkbox, the label 'Userboard X', and a 'Port' field with a dropdown menu showing '0'. At the bottom right, there is a blue 'Start' button.

6

Appendix

| Abbreviation | Explanation |
|--------------|---|
| ARP | Address Resolution Protocol |
| CID | Caller Identity |
| DNS | Domain Name System |
| DDNS | Dynamic Domain Name Service |
| DHCP | Dynamic Host Configuration Protocol |
| DMZ | Demilitarized Zone |
| DND | Do NOT Disturb |
| DTMF | DTMF : Dual Tone Multi Frequency |
| FTP | File Transfer Protocol |
| HTTP | Hypertext Transfer Protocol |
| LAN | Local Area Network |
| L2TP | L2TP: Layer 2 Tunneling Protocol |
| PPTP | PPTP: Point-to-Point Tunneling Protocol |
| MAC Address | Media Access Control Address |
| NAT | Network Address Translation |
| Ping | Packet Internet Grope |
| SIP | Session Initiation Protocol |
| TCP | Transmission Control Protocol |
| UDP | User Datagram Protocol |
| RTP | Real Time Protocol |
| PPPOE | Point-to-point Protocol over Ethernet |
| QoS | Quality of Service |
| UPnP | Universal Plug and Play |
| VLAN | Virtual Local Area Network |

| Abbreviation | Explanation |
|---------------------|-----------------------------------|
| NTP | Network Time Protocol |
| STUN | Simple Traversal of UDP over NAT |
| PSTN | Public Switched Telephone Network |
| WLAN | Wireless Local Area Network |