KENWOOD

NEXEDGE®

Application Notes

Telephone Interconnect Gateway For NEXEDGE

KPG-1014GW R1.00

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Contents

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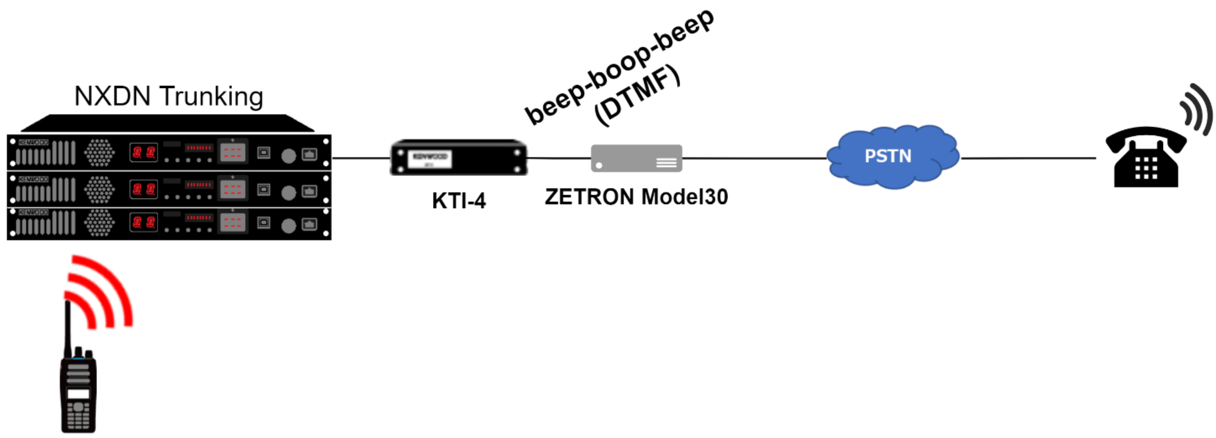
1. Introduction

1.1. From analog phones to digital IP phones

In NEXEDGE Gen1 Trunking system up to now, the KTI-4 was used to connect to analog telephone networks. The KTI-4 played a crucial role in adapting to the telephone infrastructure of the time by generating DTMF signals based on input from radios to initiate calls, and converting audio signals between analog and digital formats during communication. However, as telephone networks evolved from analog to digital and eventually to IP-based systems, more flexible and advanced connection methods became necessary. In response to this shift, the KPG-1014GW Telephone Interconnect Gateway was introduced. This document provides Basic configuration instructions for integrating this gateway into an NXDN Trunking System to enable connectivity between modern digital telephone networks and radio systems.

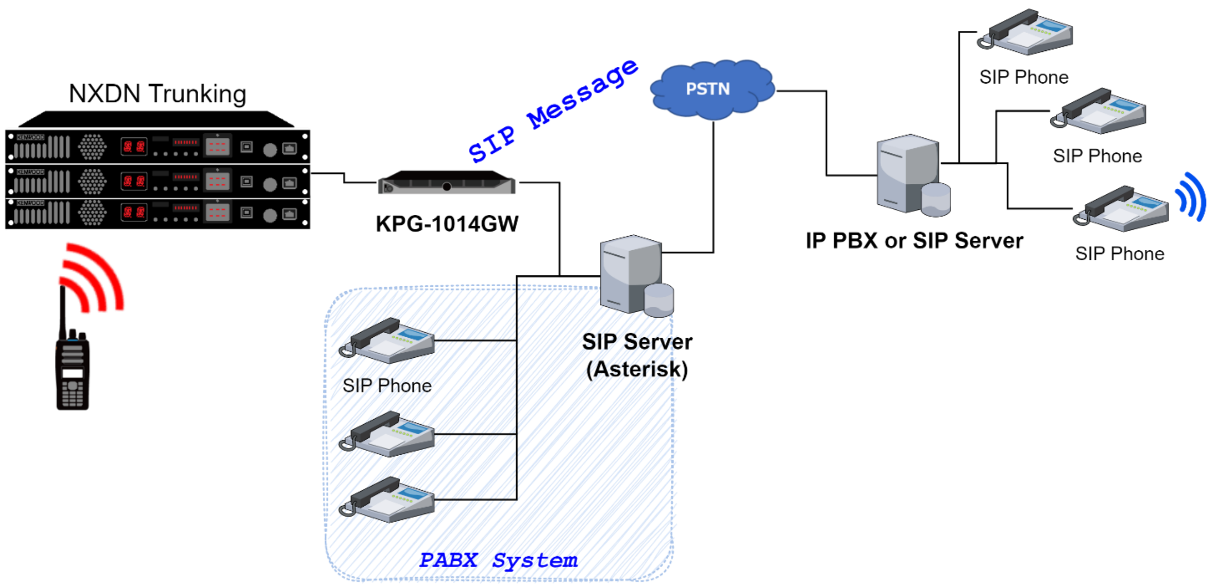
1.2. System Overview (NEXEDGE Gen1 Trunking System)

The diagram below illustrates the system configuration for connecting an NXDN Trunking System to an analog telephone network. Based on call requests from the radio, the KTI-4 generated outgoing call signals, which were sent as DTMF tones through the ZETRON Model 30 to the Public Switched Telephone Network (PSTN). Once the call was connected, the KTI-4 converted analog audio to digital and vice versa, enabling real-time communication between the radio system and the telephone line.



The diagram below illustrates the connection configuration between the NXDN Trunking System and the IP telephone network in a digital telephony environment.
Call requests from radios are converted into SIP protocol by the KPG-1014GW Telephone Interconnect Gateway and sent to an Asterisk-based SIP server. This SIP server functions as a private branch exchange (PABX), connecting to multiple SIP phones. Furthermore, the

SIP server can interconnect with external IP PBXs or SIP servers via the Public Switched Telephone Network (PSTN), enabling broader communication across various networks. This configuration allows for real-time communication between radios and IP phones, providing a flexible and highly scalable communication infrastructure.



1.3. KPG-1014GW Telephone Interconnect Gateway

1.3.1. Features

The KPG-1014GW Telephone Interconnect Gateway enables integration between radios and IP telephone networks by linking the Telephone User ID configured in the NEXEDGE Gen1 Trunking System with a corresponding SIP User ID. Since the existing settings on the Gen1 system remain unchanged, no firmware modifications or updates are required. The KPG-1014GW connects to the IP telephone network using the SIP User ID registered on the SIP server. This allows IP phone terminals such as SIP phones to initiate both Individual Calls and Group Calls to radio users.

1.3.1.1. Licenses

Authentication is performed using the license file generated by the license management client (KPT-300LMC). The software licenses handled by the Telephone Interconnect Gateway are as follows:

License	Type	Description
KWD-1014SP	Software Upgrade License	・Up to 10 calls are available at the same time for a call using a SIP Phone. ・Additional installation is available. ・The maximum number of installations: 10

2. Basic Configuration: Gen1 Single-Site System

2.1. Sample Configuration

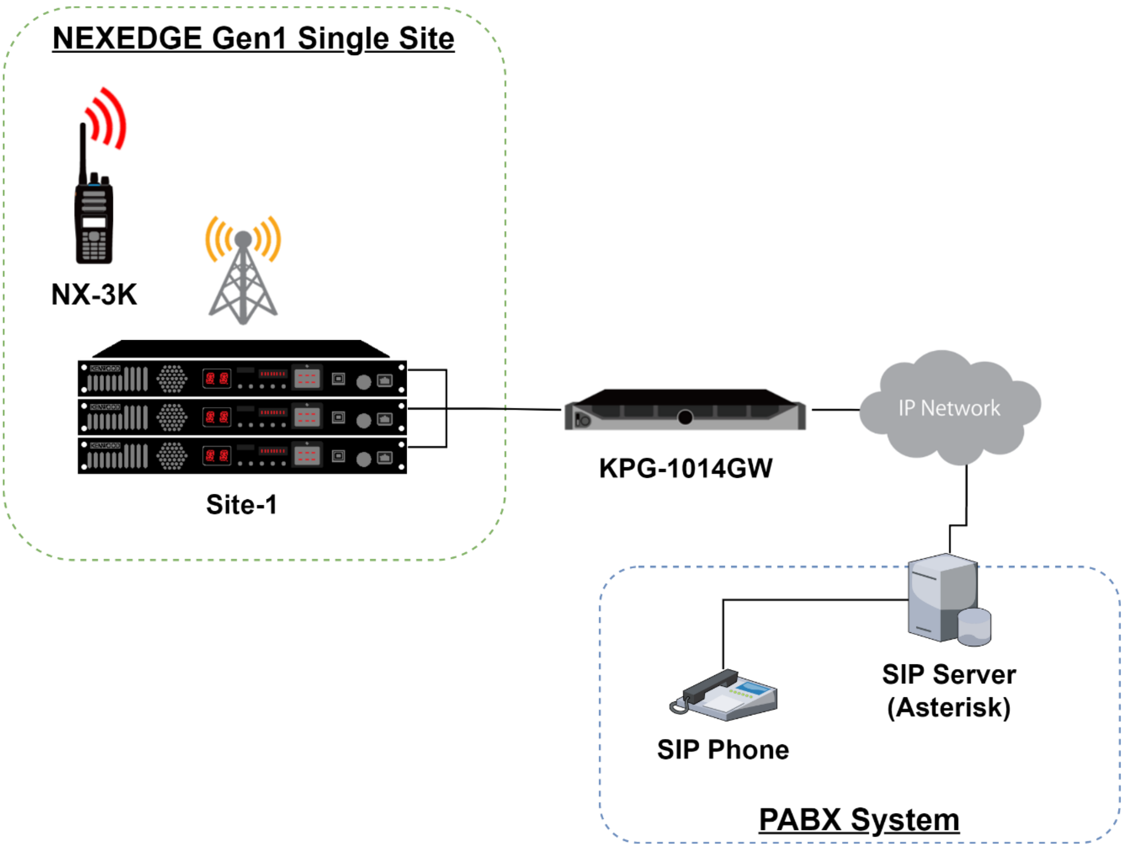
In this chapter, it will explain the system configuration where the KPG-1014GW Telephone Interconnect Gateway is installed in a Gen1 NEXEDGE Single-Site System, enabling communication with PABX telephones, along with its configuration procedures.

2.1.1. System Structure

The system configuration described in this chapter is as follows. The NEXEDGE Gen1 Trunking System is structured in the simplest single-site configuration.

When the NX-3000 SERIES makes a call request to a telephone within the NEXEDGE Single Site Trunking System, the system forwards the call request to the KPG-1014GW, which in turn relays it to the PABX via the SIP server and calls the PABX telephone.

Conversely, when a PABX telephone wishes to communicate with an NX-3000 SERIES, it sends a call request to the SIP server, which then relays the request through the KPG-1014GW to the NEXEDGE Gen1 Trunking System, which calls the target NX-3000 SERIES.



2.1.2. Product and Version information

The products and their firmware versions used in this chapter are as follows:

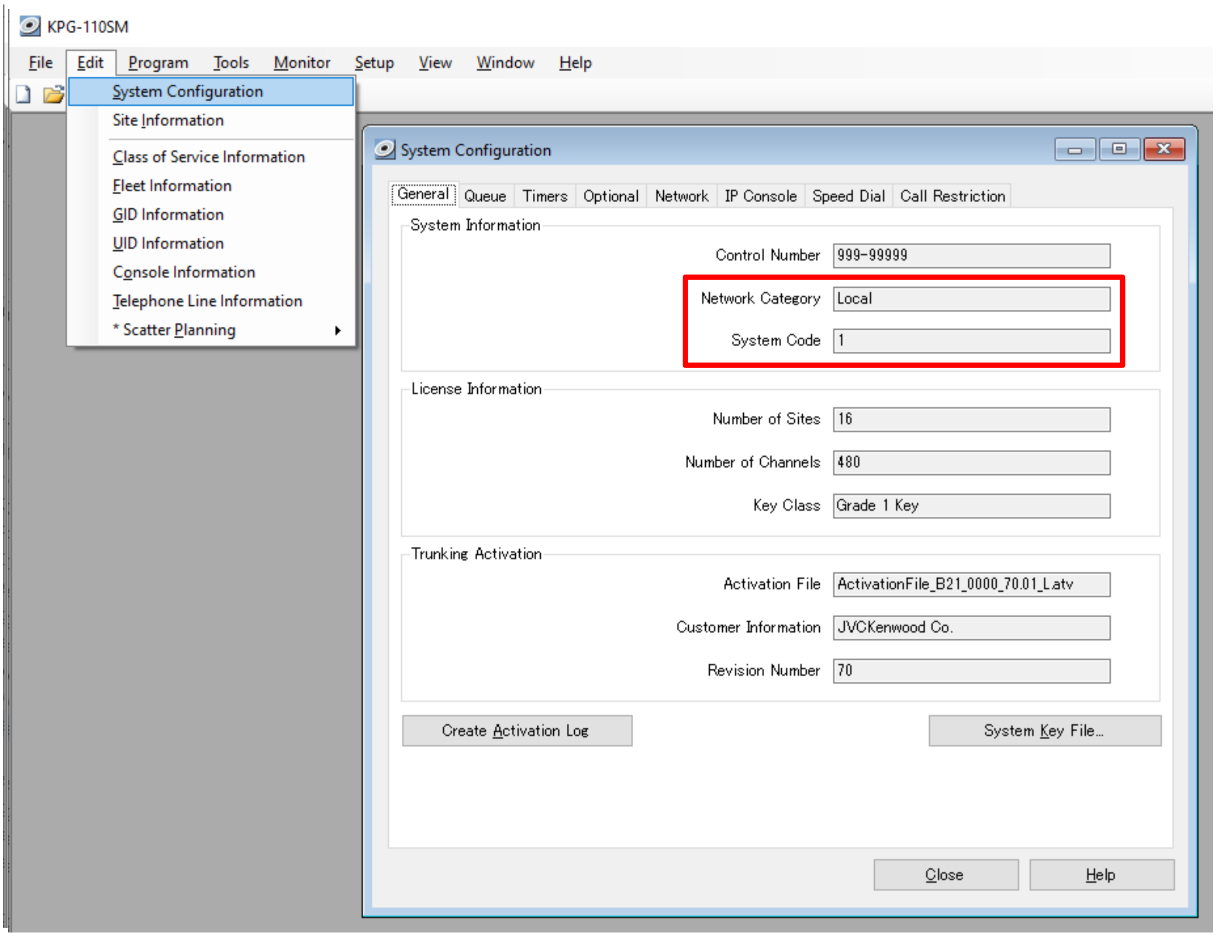
Products	Version
NX-3300	4.24
NXR-5800	V5.42
KPG-1014GW	1.00

2.1.3. NEXEDGE Gen1 Trunking system Configuration

In this chapter, we introduce the necessary configuration information for integrating the KPG-1014GW Telephone Interconnect Gateway into the NEXEDGE Gen1 Trunking System to enable communication between PABX telephones and radios.

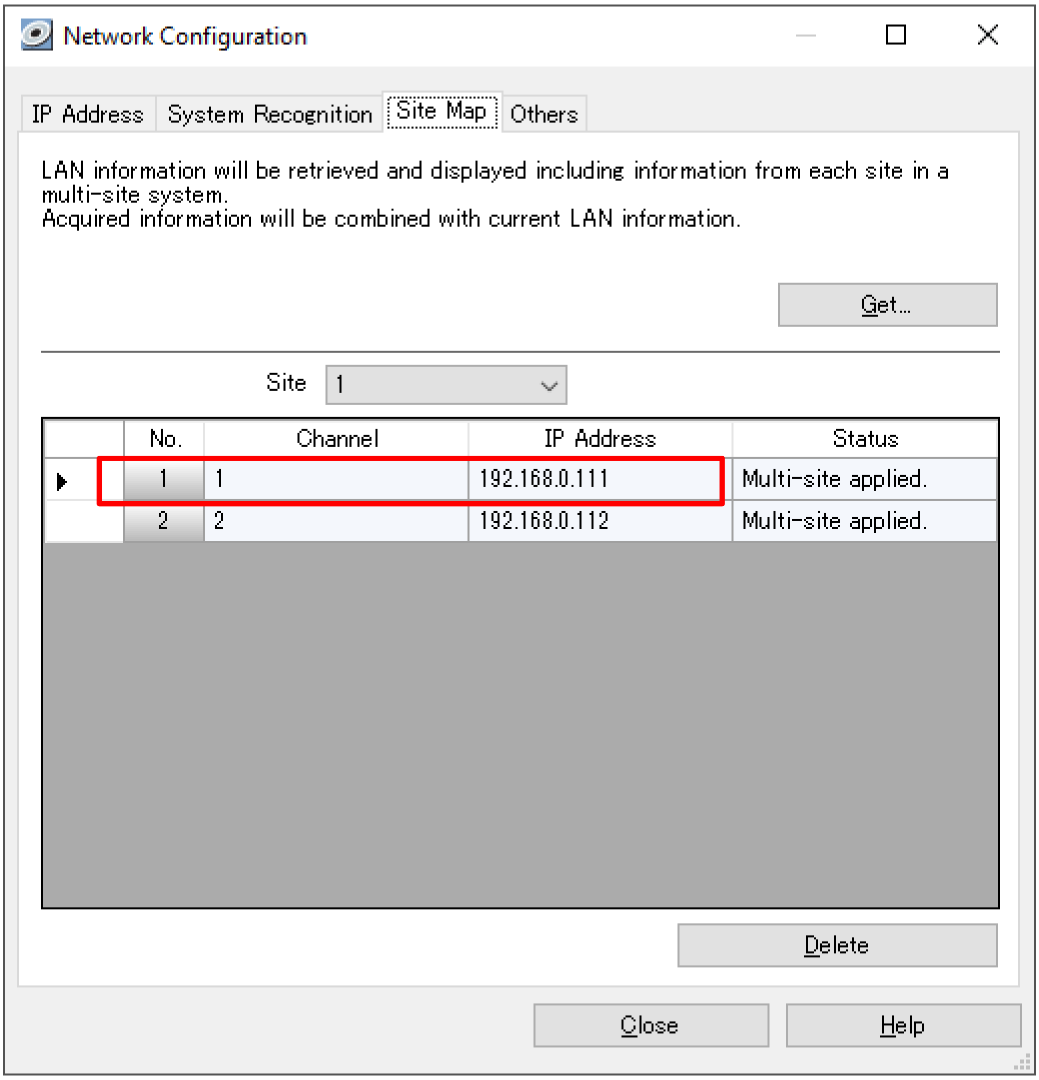
2.1.3.1. KPG-110SM: Edit > System Configuration > General > System Information

Launch KPG-110SM and open the "System Configuration" screen. Select the "General" tab, where the "Network Category" and "System Code" displayed under "System Information" are essential parameters for configuring the KPG-1014GW.



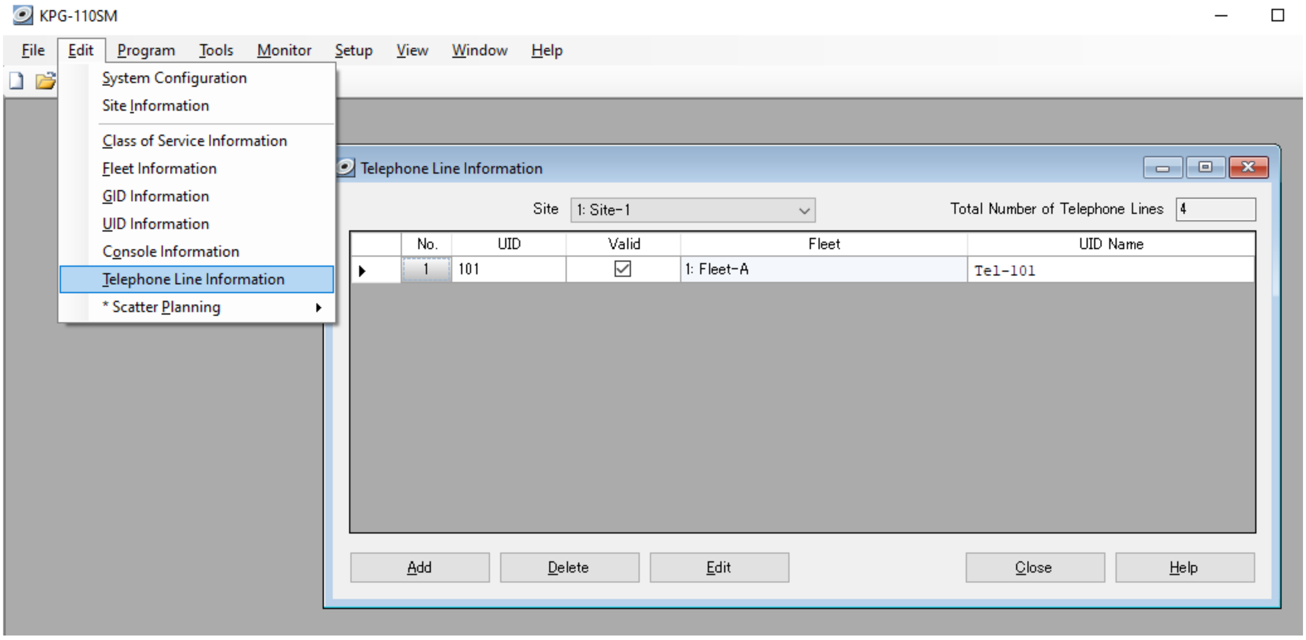
2.1.3.2. KPG-110SM: Tools > Network Configuration > Site Map

The NEXEDGE Gen1 Trunking system has an access point for connecting and communicating with the KPG-1014GW Telephone Interconnect Gateway. Among the channels that make up a site in the NEXEDGE Gen1 Trunking system, the IP address assigned to the channel with the lowest channel number serves as this access point. Since the KPG-1014GW Telephone Interconnect Gateway uses this IP address to establish call sessions with the NEXEDGE Gen1 Trunking System, it is an important parameter when configuring the KPG-1014GW Telephone Interconnect Gateway.



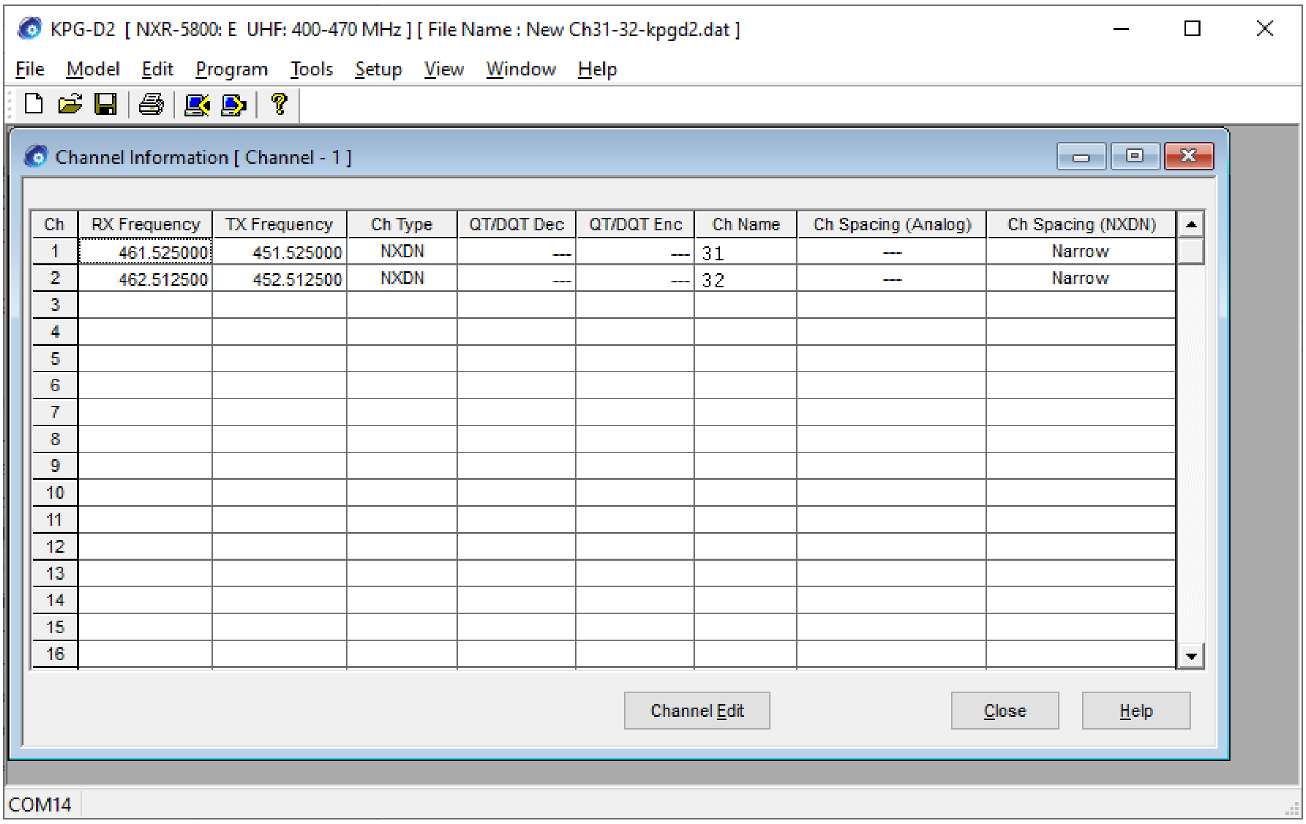
2.1.3.3. KPG-110SM: Edit > Telephone Line Information

In the NEXEDGE Gen1 Trunking System, the UID can be assigned to the KTI-4 module. When the KTI-4 receives a call request from the PSTN analog line, it uses the assigned UID to make a proxy call request to the NEXEDGE Trunkig System. Similarly, when a radio terminal makes a call request to the Telephone Line, both the terminal and the system use this UID for the call. The KPG-1014GW uses this existing telephone UID to relay call requests from the external SIP server to the PABX/PSTN and vice versa.



2.1.3.4. KPG-D2: Ch Spacing (NXDN)

The NEXEDGE Gen1 Trunking System offers two channel spacing types: Narrow and Very Narrow. While the voice encode/decode format used is identical in both cases, the data format little bit differs, so this setting must be uniform across the system. The KPG-1014GW Telephone Interconnect Gateway uses this information to convert the audio format for telephone use. Therefore, the channel spacing setting must match between the NEXEDGE Gen1 Trunking System and the KPG-1014GW Telephone Interconnect Gateway.



2.1.3.5. Summary of Gen1 System information required for KPG-1014GW configuration

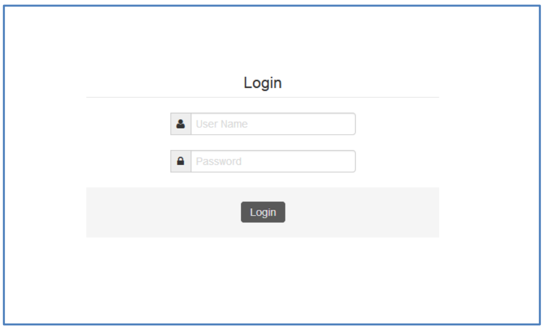
In the previous sections, we introduced the NEXEDGE Gen1 Trunking System configuration items that are closely related to the setup of the KPG-1014GW Telephone Interconnect Gateway. The following table summarizes the sample parameters for the NEXEDGE Gen1 Trunking system used in this chapter. These values will be used as reference examples in the next chapter when explaining the configuration steps for the KPG-1014GW.

<table>
<tr><th colspan="2">Gen1 System Information</th><th>Tools</th><th>Note</th></tr>
<tr><td>Network Category</td><td>Local</td><td rowspan="2">KPG-110SM</td><td rowspan="2">It can be found in the General TAB of the “System Configuration Window”.</td></tr>
<tr><td>System Code</td><td>1</td></tr>
<tr><td>Ch Spacing (NXDN)</td><td>Narrow</td><td>KPG-D2</td><td>The data written in Channel can be checked by reading the data in KPG-D2.</td></tr>
<tr><td>IP Address (Minimum-number of Channel)</td><td>192.168.0.111</td><td rowspan="2">KPG-110SM</td><td>IP address of the Channel with the lowest number configured in the site.
It can be found in the Site Map TAB in Network Configuration.</td></tr>
<tr><td>Telephone UID</td><td>101</td><td>The Telephone Line Information box allows you to see the Telephone UID number set for your system.</td></tr>
</table>

2.1.4. KPG-1014GW Configuration

2.1.4.1. Login to the Web Tool

To configure the KPG-1014GW Telephone Interconnect Gateway It must first log in to the Web Tool. The default IP address for accessing the Web Tool via a web browser is set to "192.168.0.1".



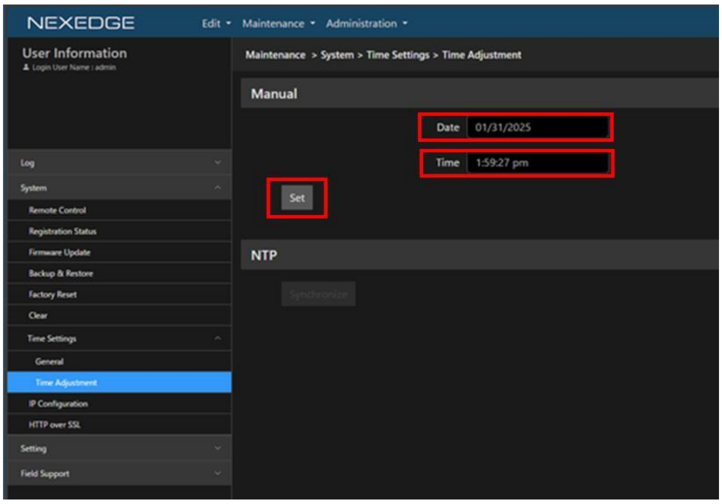
The initial User Name and Password for the login screen are set as shown in the table below.

Web Tool Access Default Parameter	
IP Address	192.168.0.1
User Name	Admin
Password	(blank)

Upon the first login, you will be prompted to change the admin password. After changing it, use the new password for all subsequent logins. Be sure to remember the password you set to avoid losing access.

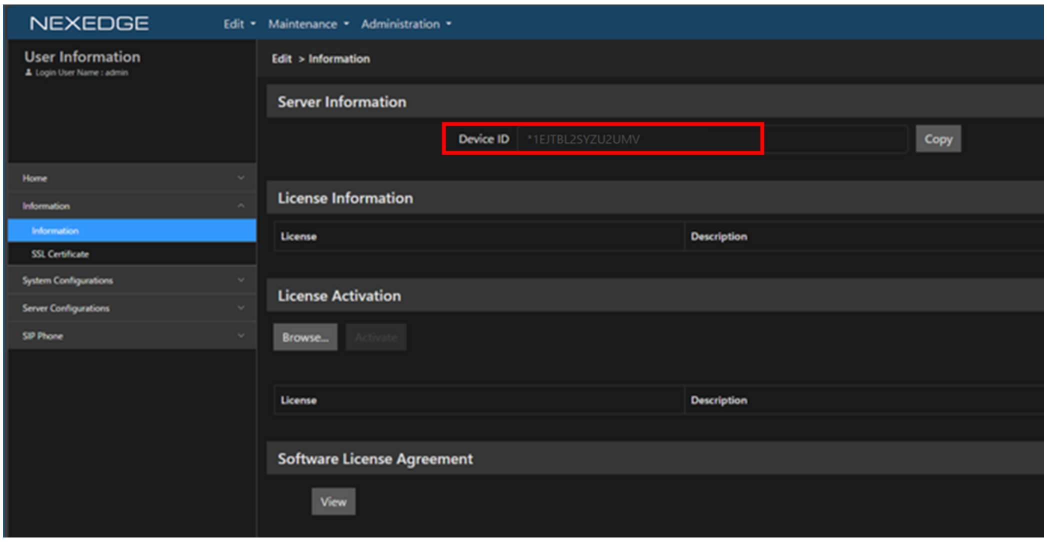
2.1.4.2. Maintenance ＞ System > Time Adjustment

After logging in, configure the time settings for the KPG-1014GW. Since the time settings are important for timestamps in logs and other records, it is recommended to set them during the initial setup. From the top menu, select Maintenance, then **"System > Time Adjustment"**, and set the correct date and time.

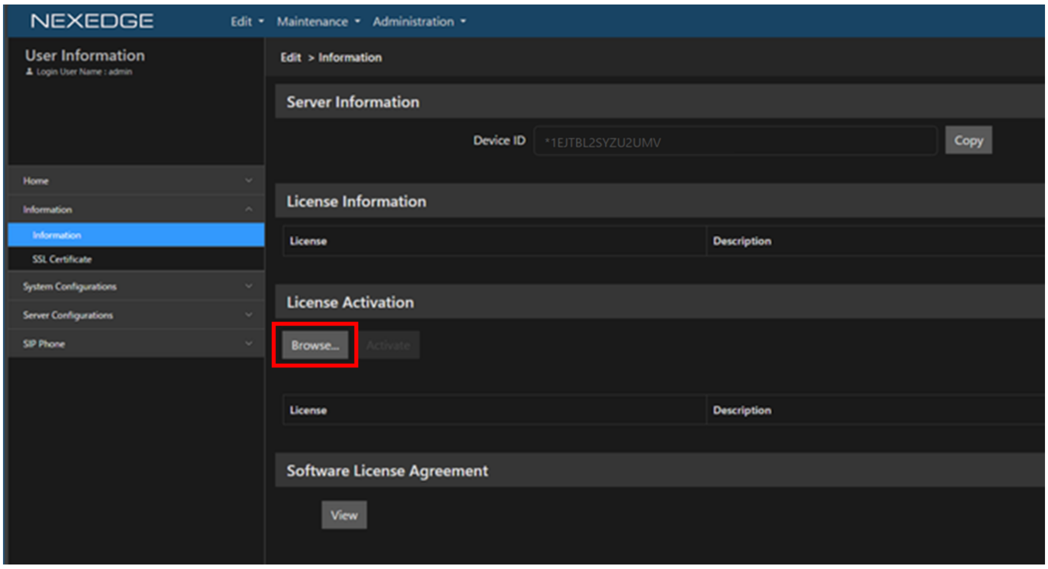


2.1.4.3. Edit > Information > License Activation

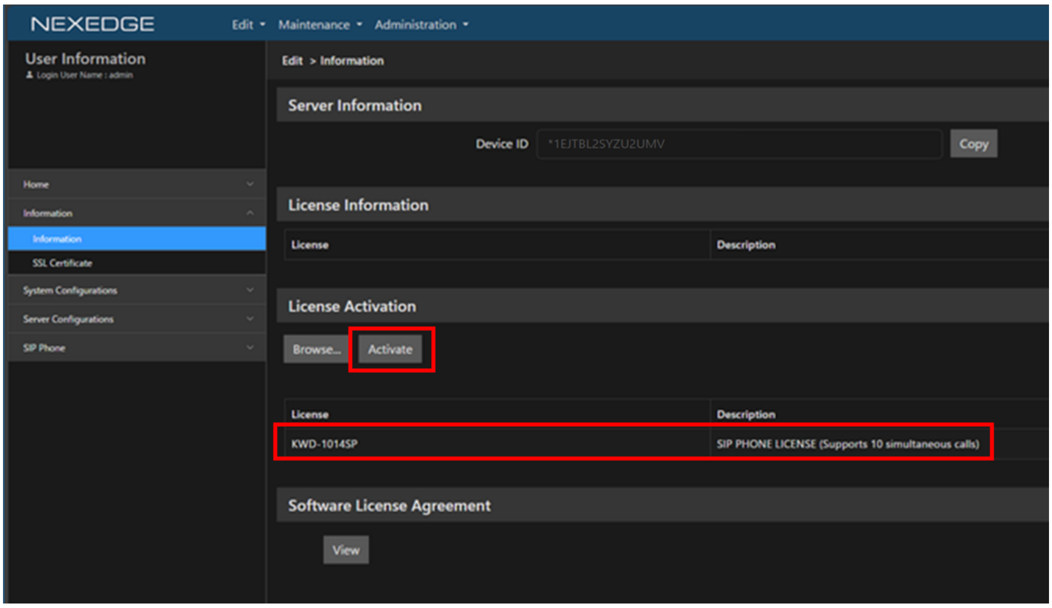
From the top menu of the Web Tool, select **"Edit > Information"**. On the displayed screen, obtain the Device ID.



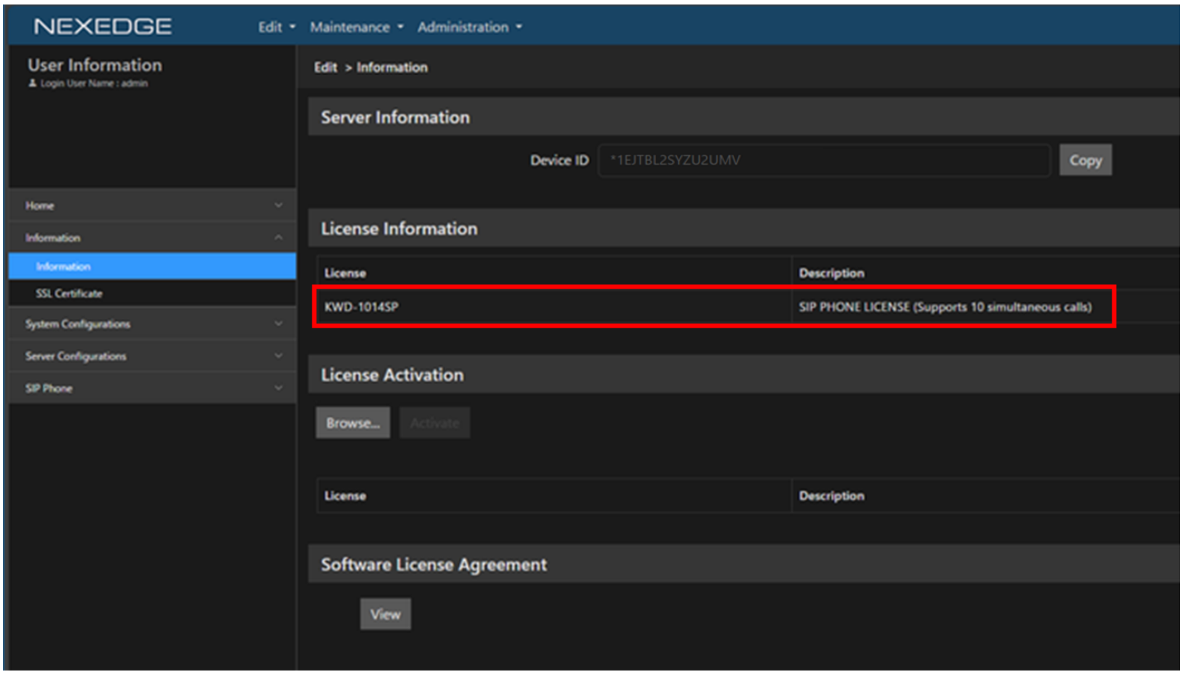
Using the obtained Device ID, perform offline authentication for the Level License with KPT-300LMC and acquire the Authentication File. Upload the Authentication File to the KPG-1014GW using the Web Tool. Select **"Edit > Information"** from the top menu, and when the License Activation section appears, click the **"Browse..."** button and select the Authentication File.



If the Authentication File is successfully loaded, the license name and description will be displayed in the License Activation section. Confirm the contents and if correct, click **"Activate"**.



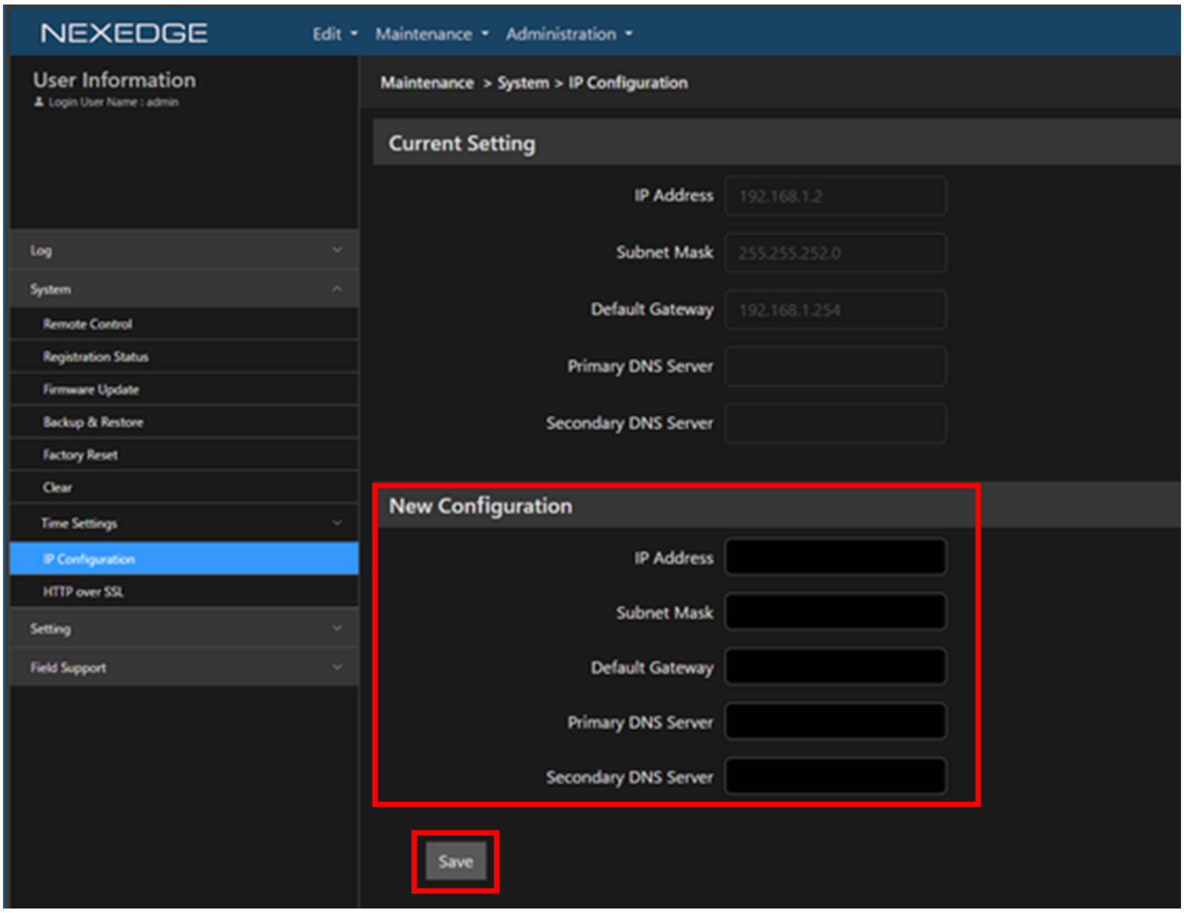
Once activation is successful, the license information will be displayed. Verify that the license information is displayed correctly.



With the license installed, the system can handle up to 10 concurrent calls with telephones.

2.1.4.4. Maintenance > System > IP Configuration

Change the IP Address of the KPG-1014GW Telephone Interconnect Gateway so that it can be used in the IP network for actual operation. Select **"Maintenance > System > IP Configuration"** and enter the necessary parameters under **"New Configuration"**. After entering the settings, click **"Save"** to apply the configuration. The KPG-1014GW will restart using the new IP Address. After restarting, use the new IP Address to log back into the Web Tool.

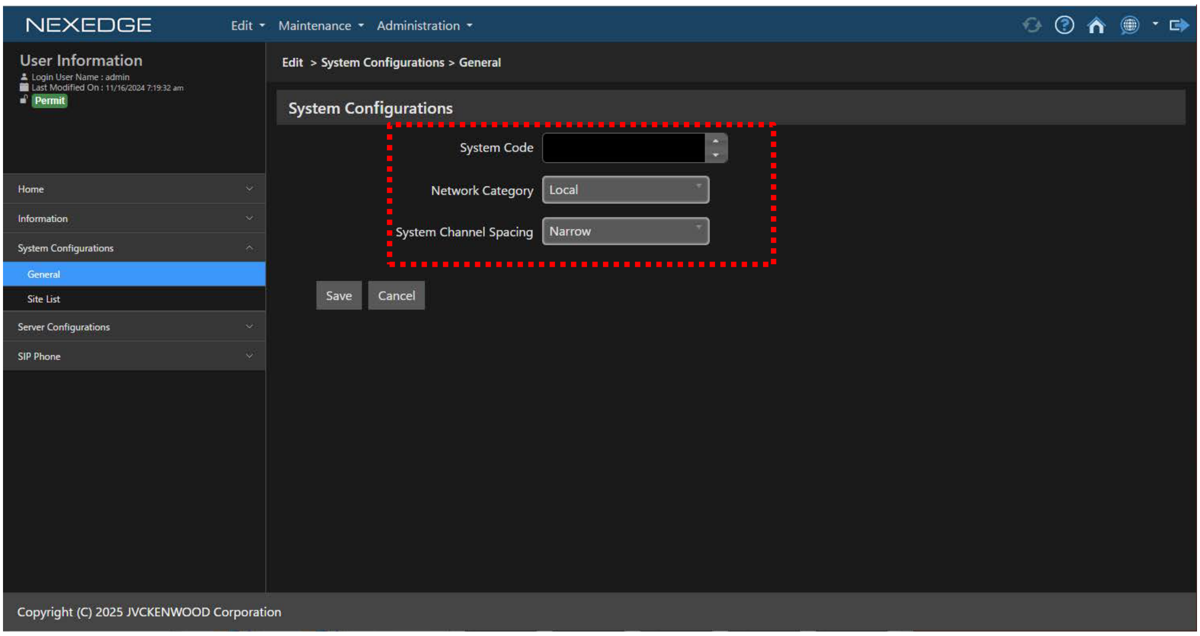


With the above steps, the basic setup required for the KPG-1014GW Telephone Interconnect Gateway — in other words, the initial configuration necessary for normal startup — has been completed. In the following chapters, we will proceed with the configuration required to connect the KPG-1014GW to the NEXEDGE Gen1 Trunking System.

2.1.4.5. Edit > System Configuration > General

The table in"2.1.3.5 Summary of Gen1 System information required for KPG-1014GW configuration" lists the key NEXEDGE Gen1 Trunking system parameters used in the sample configuration for this chapter. These values will be referenced in the explanations that follow. The parameters required in this section are indicated with red underlines in the table.

Gen1 System Information	
Network Category	Local
System Code	1
Ch Spacing (NXDN)	Narrow
IP Address (Minimum-number of Channel)	192.168.0.111
Telephone UID	101



2.1.4.5.1. System Configurations > System Code

Select **"System Configurations > General"** from the left menu and enter the system code of the NEXEDGE Gen1 Trunking System to connect to in the box labeled "System Code".

2.1.4.5.2. System Configurations > Network Category

Select **"System Configurations > General"** from the left menu and select the "Network Category" set for the NEXEDGE Gen1 Trunking System to connect to in the box labeled **"Network Category"** from the pull-down menu.

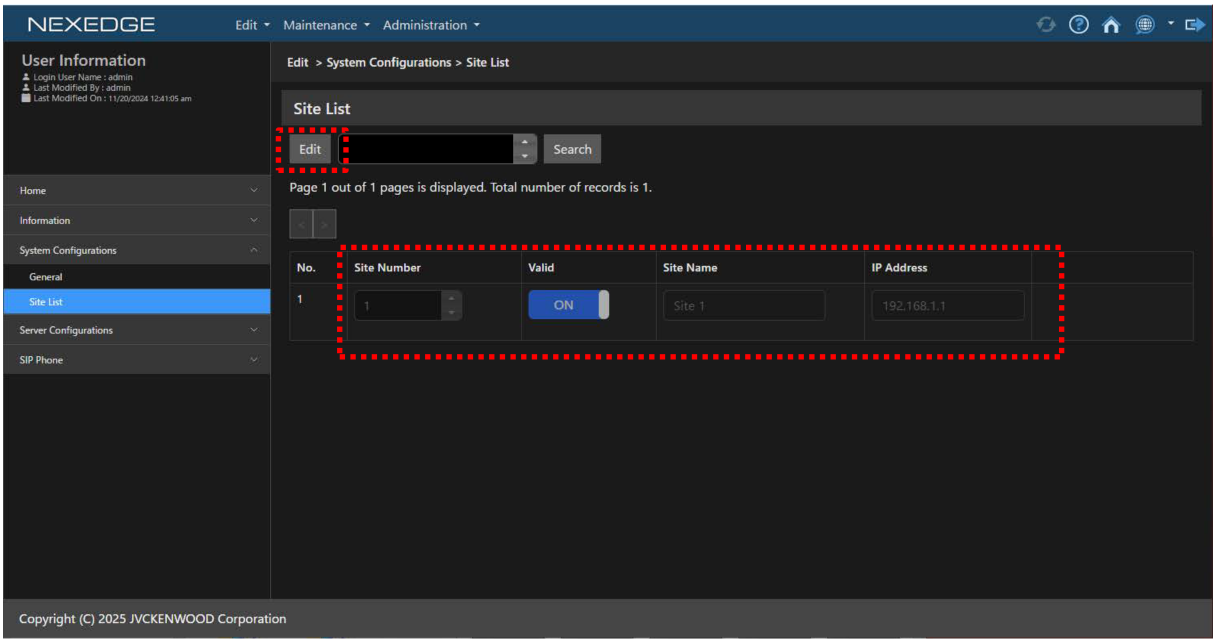
2.1.4.5.3. System Configurations > System Channel Spacing

Select **"System Configurations > General"** from the left menu and select the **"System Channel Spacing"** used by the NEXEDGE Gen1 Trunking System to connect to in the box labeled **"System Channel Spacing"** from the pull-down menu.

2.1.4.6. Edit > System Configuration > Site List

As with the previous section, the values required in this section are the parameters underlined in red in the table below. To establish a call session with Telephone, the KPG-1014GW must make a call request to the NEXEDGE Gen1 Trunking System. The NEXEDGE Gen1 Trunking System exchanges messages to establish a call session with the KPG-1014GW using the IP Address set for the channel with the smallest site channel number. On the KPG-1014GW, select **"System Configurations > Site List"** from the Menu on the left, press the **"Edit"** button to edit the table, and edit the Channel information that the KPG-1014GW will access.

Gen1 System Information	
Network Category	**Local**
System Code	**1**
Ch Spacing (NXDN)	**Narrow**
IP Address (Minimum-number of Channel)	**192.168.0.111**
Telephone UID	**101**



2.1.4.6.1. Site List > Edit

Select **"System Configuration > Site List"** and click **"Edit"** at the top of the screen to open the editing screen.

2.1.4.6.2. Site List > Site Number

In the editing screen, enter the **"Site Number"** of the NEXEDGE Trunking to be connected.

2.1.4.6.3. Site List > Valid

In the editing screen, set **"Valid"** to On. (If you will not be accessing, for example if you will not be connecting temporarily, set Valid to Off.)

2.1.4.6.4. Site List > Site Name

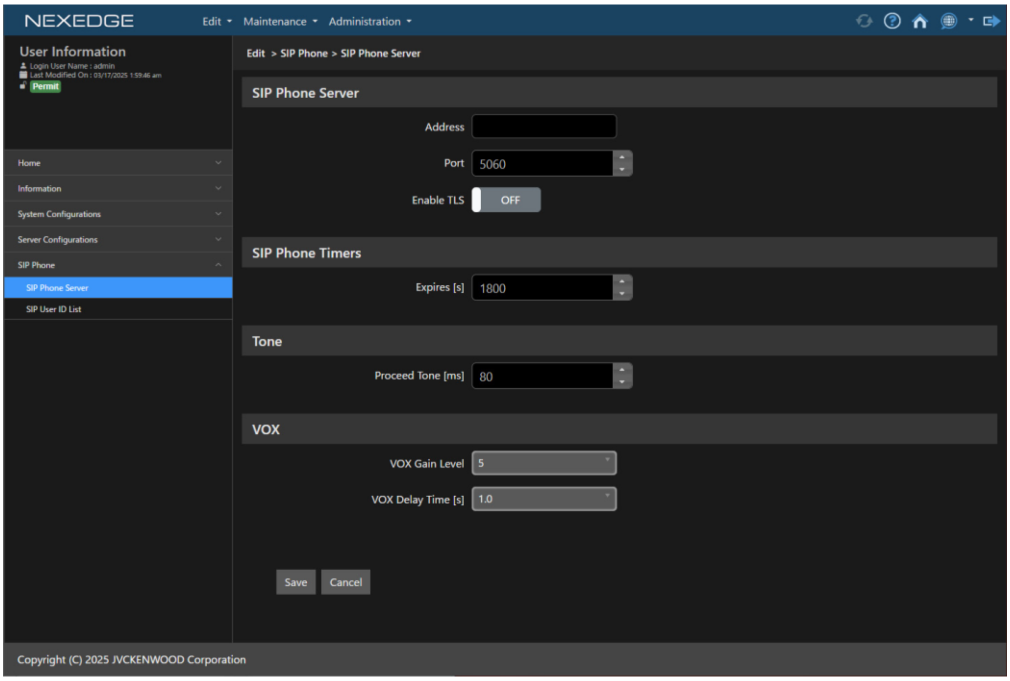
In the editing screen, enter the site name in text.

2.1.4.6.5. Site List > IP Address

Enter the **IP Address** of the Repeater with the smallest channel number of the site specified in **"Site Number".**

2.1.4.7. Edit > SIP Phone > SIP Phone Server

To establish a call line with the PABX Telephone, the KPG-1014GW Telephone Interconnect Gateway must connect to the SIP Server that manages the PABX. Select "Edit > SIP Phone > SIP Phone Server" and configure the SIP Phone Server to which the KPG-1014GW Telephone Interconnect Gateway connects.



2.1.4.7.1. SIP Phone Server > Address

The KPG-1014GW Telephone Interconnect Gateway uses the SIP Protocol to establish a call session with an external SIP Server. The IP Address set here is the IP Address of the SIP Phone Server where the SIP User ID used by the KPG-1014GW Telephone Interconnect Gateway is preregistered. The KPG-1014GW Telephone Interconnect Gateway can use the SIP Phone Server by registering the IP Address set here using the **"SIP User ID"** and **"SIP Authentication Password"** described in a later section.

2.1.4.7.2. SIP Phone Server > Port

When communicating, the SIP Phone Server waits for communication on the port number set on the server itself. Therefore, the port number set here is the **"Port"** number on which the destination SIP Phone Server is waiting. Usually, the default setting (Port No. = 5060) is used, but for security reasons, a different port number may be used. In that case, check with your system administrator to set the port number used by the SIP Server.

2.1.4.7.3. SIP Phone Server > Enable TLS

In some cases, encrypted communication (TLS) is used for communication with the SIP Server. If you set **"Enable TLS"** to "On", "TLS" will be used for communication with the SIP Server.

2.1.4.7.4. SIP Phone Timers > Expires [s]

The SIP Phone Server has an expiration date for the registration of the registered SIP User ID. When this expiration date passes, the SIP Phone Server deletes the registration information of the target SIP User ID. Therefore, the target SIP User ID must be re-registered before this expiration date expires if you want to use the SIP Phone Server. This item sets the registration expiration date of the SIP User ID for the SIP Phone Server, and the KPG-1014GW Telephone Interconnect Gateway will re-register the SIP User ID before the expiration time is reached.

2.1.4.7.5. Tone > Proceed Tone [ms]

Sets the length of **the tone** that the telephone will play when the terminal finishes sending.

2.1.4.7.6. VOX > VOX Gain Level

Sets the input sensitivity of the telephone voice during VOX operation during a Group Call call. Used to determine that voice input from the telephone has ceased. Sets the input sensitivity of the telephone voice during VOX operation during a Group Call from 1 (low sensitivity) to 10 (high sensitivity).

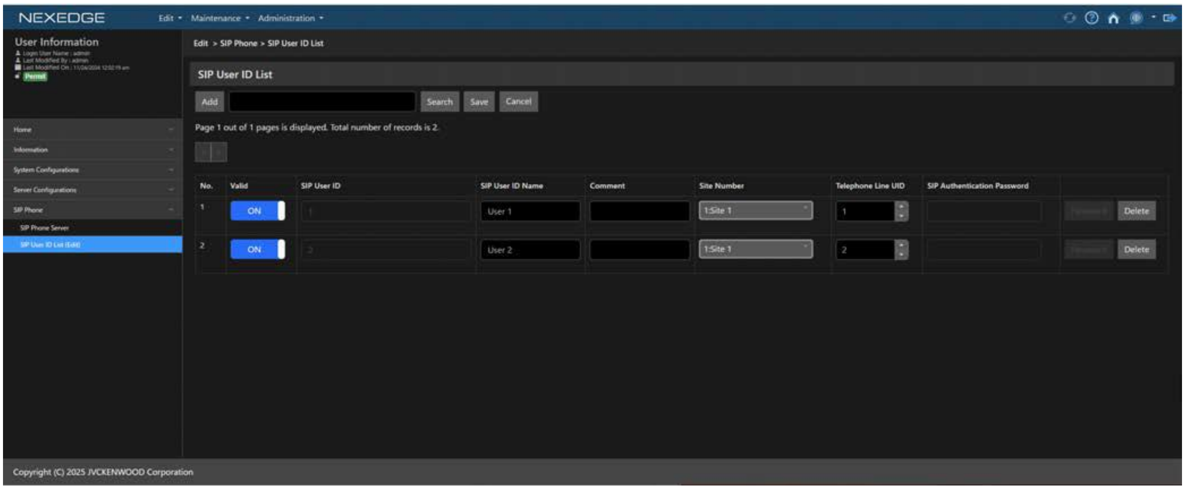
2.1.4.7.7. VOX > VOX Delay Time [s]

When VOX is operating during a Group Call call, this sets the transmission time to be maintained after the voice input from the phone is no longer heard. This is used to prevent the transmission audio from being interrupted. When VOX is operating during a Group Call call, after the voice input from the phone is no longer heard, transmission is maintained for the time set in VOX Delay Time.

2.1.4.8. SIP Phone > SIP User ID List

In the NEXEDGE Gen1 Trunking System, a dedicated Telephone UID is configured for telephones. This section explains how to link the SIP User ID used on the SIP Phone Server with the Telephone UID set in the NEXEDGE Gen1 Trunking System.
You can add or edit SIP User IDs, set their validity, assign names and comments for identification, and configure the Site Number to match the one set in the Site List.

Gen1 System Information	
Network Category	**Local**
System Code	**1**
Ch Spacing (NXDN)	**Narrow**
IP Address (Minimum-number of Channel)	**192.168.0.111**
Telephone UID	**101**



2.1.4.8.1. SIP User ID List > Add/Edit

There is an Add button at the top of the screen. Pressing it will add a row for the SIP User ID to the list.

2.1.4.8.2. SIP User ID List > Valid

Set whether or not to use the set SIP User ID. (ON: Use allowed, Off: Use prohibited)

2.1.4.8.3. SIP User ID List > SIP User ID

Set the SIP User ID that the KPG-1014GW Telephone Interconnect Gateway will use when accessing the SIP Phone Server. The SIP User ID set here will be used to register with the SIP Phone Server, so this User ID must be registered in the SIP Phone Server beforehand.

2.1.4.8.3.1. SIP User ID List > SIP User ID Name

Set any name for the SIP User ID to identify and manage the SIP User ID.

2.1.4.8.4. SIP User ID List > Comment

Comment for the registered SIP User ID. It can record information about the SIP User ID.

2.1.4.8.5. SIP User ID List > Site Number

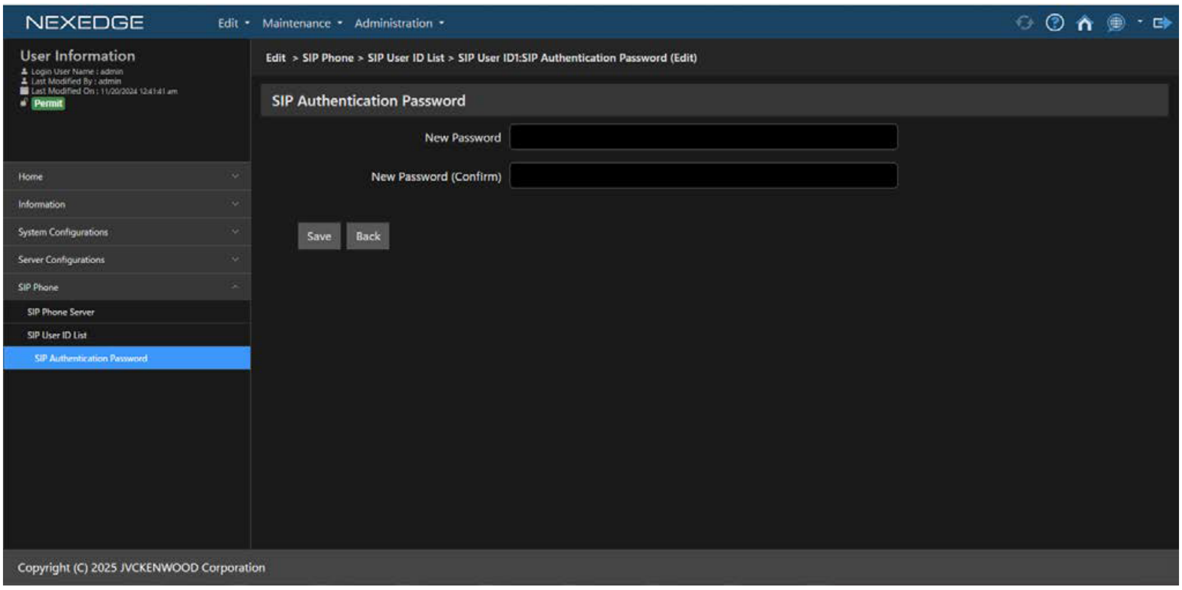
It can display or set the site number of the system corresponding to the Telephone Line UID set in the NEXEDGE Gen1 Trunking System. The setting must match the Site Number of the NEXEDGE Gen1 Trunking System set in the Site List above.

2.1.4.8.6. SIP User ID List > Password

Pressing the Password button will take you to the screen where you can set the “Password” that the SIP User ID will use to access the SIP Phone Server. Only Log In Users with at least edit permissions can press this button.

2.1.4.8.7. SIP Phone > SIP User ID List > SIP Authentication Password

It can set the password for the SIP User ID that the KPG-1014GW Telephone Interconnect Gateway will use when registering with the SIP Phone Server. Enter the Password that is SIP User ID for selected in the "New Password" box. The previously set password will be deleted if it leave the password field blank and save it.



3. Appendix

3.1. Constructing Asterisk SIP Server

Step-1: Preparing Boot USB Memory for Debian OS Install

Click the following link to install Debian 12:

Step by Step Debian 12 Installation

Note:

Before starting the installation, make sure to either assign a static IP address for the Asterisk SIP Server or ensure that an IP address will be assigned via DHCP. Also, during the installation process, some update packages may be downloaded from the Internet, so be sure to use a network connection that has Internet access.

Step-2: Install Asterisk Server to Debian Server

Asterisk will be installed by following the instructions provided in

GitHub – FreePBX Installation Script for Debian

In this document, however, we modify the downloaded installation script to change the version of Asterisk. Please update the following line(s) in the script accordingly. (The version of Asterisk used in this document is “**20**”)

```
ASTVERSION=21 (or ASTVERSION=22)
```

Change the above text to "20" as shown below.

```
ASTVERSION=20
```

When executing the installation, run the installation script with the option to skip FreePBX installation and the option to bypass version checks.

Note:

The following example assumes that the installation script has been downloaded to the /tmp/ directory.

```
bash /temp/sng_freepbx_debian_install.sh --nofreepbx --skipversion
```

After the installation script has finished running, install the additional configuration file packages.

```
apt-get install asterisk20-configs
```

Step-3: Configuring Asterisk Server

Use a console terminal software such as PuTTY to log in to the Asterisk server via SSH. Once logged in, use a text editor such as **"vi"** or **"nano"** to edit the configuration files, specifically **"<u>pjsip.conf</u>"** and **"<u>extensions.conf</u>"**, as shown below.

<u>Sample Configuration: pjsip.conf</u>

```
;********************************************************************************
;*
;*   pjsip.conf
;*   2025/05/16
;*
;********************************************************************************
[transport-udp]
type = transport
protocol = udp
bind = 0.0.0.0:5060

[transport-tls]
type=transport
protocol=tls
bind=0.0.0.0:5061
cert_file=/etc/asterisk/keys/asterisk.crt
priv_key_file=/etc/asterisk/keys/asterisk.key
method=sslv23

;********************************************************************************
;*
;*   Soft Phone
;*
;********************************************************************************
;Soft Phone 1
[200]
type = aor
max_contacts = 1
remove_existing = yes

[200]
type = auth
username = 200
password = pass

[200]
type = endpoint
context = default
disallow = all
allow = ulaw
allow = alaw
use_ptime = yes
```

```
asymmetric_rtp_codec = yes
direct_media = no
language = en
auth = 200
outbound_auth = 200
aors = 200

;Echo Test
[202]
type = aor
max_contacts = 1
remove_existing = yes

[202]
type = auth
username = 202
password = pass

[202]
type = endpoint
context = default
disallow = all
allow = ulaw
allow = alaw
use_ptime = yes
asymmetric_rtp_codec = yes
direct_media = no
language = en
auth = 202
outbound_auth = 202
aors = 202

;Soft Phone 2
[203]
type = aor
max_contacts = 1
remove_existing = yes

[203]
type = auth
username = 203
password = pass

[203]
type = endpoint
context = default
disallow = all
allow = ulaw
allow = alaw
use_ptime = yes
asymmetric_rtp_codec = yes
direct_media = no
language = en
auth = 203
outbound_auth = 203
aors = 203


;*******************************************************************************
;*
;*   NXDN Telephone Interconnect Gateway
;*
;*******************************************************************************
```

```
;KPG-1014GW SIP User ID
[600]
type = aor
max_contacts = 1
maximum_expiration = 43200
remove_existing = yes

[600]
type = auth
username = 600
password = pass

[600]
type = endpoint
context = default
dtmf_mode = inband
disallow = all
allow = ulaw
allow = alaw
use_ptime = yes
asymmetric_rtp_codec = yes
direct_media = no
language = en
auth = 600
outbound_auth = 600
aors = 600


[601]
type = aor
max_contacts = 1
maximum_expiration = 43200
remove_existing = yes

[601]
type = auth
username = 601
password = pass

[601]
type = endpoint
context = default
dtmf_mode = inband
disallow = all
allow = ulaw
allow = alaw
use_ptime = yes
asymmetric_rtp_codec = yes
direct_media = no
language = en
auth = 601
outbound_auth = 601
aors = 601


[602]
type = aor
max_contacts = 1
maximum_expiration = 43200
remove_existing = yes

[602]
type = auth
```

```
username = 602
password = pass

[602]
type = endpoint
context = default
dtmf_mode = inband
disallow = all
allow = ulaw
allow = alaw
use_ptime = yes
asymmetric_rtp_codec = yes
direct_media = no
language = en
auth = 602
outbound_auth = 602
aors = 602
```

Sample Configuration: extensions.conf

```
;*******************************************************************************
;*
;*   extensions.conf
;*   2025/05/16
;*
;*******************************************************************************
[general]
writeprotect=no
priorityjumping=no

;*******************************************************************************
;*
;*   Variable Definition
;*
;*******************************************************************************
[globals]

;Asterisk Ext Phone
PABX_PHONE1=200
PABX_PHONE2=202
PABX_PHONE3=203

;KPG-1014GW SIP User ID
KPG-1014GW_SIPUSERID1=600
KPG-1014GW_SIPUSERID2=601
KPG-1014GW_SIPUSERID3=602

;*******************************************************************************
;*
;*   PJSIP Header Definition
;*
;*******************************************************************************
[pabx-individual]
exten => _X.,1,Set(PJSIP_HEADER(add,JKC-Header-Calltype)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DstId)=${EXTEN})
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-SrcType)=pabx)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DelayTime)=0)
```

```
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-MdDelayTime)=400)
exten => _X.,n,Return

[pabx-group]
exten => _X.,1,Set(PJSIP_HEADER(add,JKC-Header-Calltype)=8)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DstId)=${EXTEN})
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-SrcType)=pabx)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DelayTime)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-MdDelayTime)=400)
exten => _X.,n,Return

[pstn-individual]
exten => _X.,1,Set(PJSIP_HEADER(add,JKC-Header-Calltype)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DstId)=${EXTEN})
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-SrcType)=pstn)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DelayTime)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-MdDelayTime)=400)
exten => _X.,n,Return

[pstn-group]
exten => _X.,1,Set(PJSIP_HEADER(add,JKC-Header-Calltype)=8)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DstId)=${EXTEN})
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-SrcType)=pstn)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DelayTime)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-MdDelayTime)=400)
exten => _X.,n,Return

[default]
;*-------------------------------------------------------------------------------*
;*
;*   System   NXDN Gen1 Trunking System
;*   From     SU
;*   To       PABX
;*
;*-------------------------------------------------------------------------------*
;REQ PABX_PHONE1 Call PABX_PHONE1
exten => ${PABX_PHONE1},1,Answer(500)
same => n,Dial(PJSIP/${PABX_PHONE1},30,rH)
same => n,Playtones(busy)
same => n,Busy(3)

;REQ PABX_PHONE2 Call PABX_PHONE2
exten => ${PABX_PHONE2},1,Answer(500)
same => n,Dial(PJSIP/${PABX_PHONE2},30,rH)
same => n,Playtones(busy)
same => n,Busy(3)

;REQ PABX_PHONE3 Call PABX_PHONE3
exten => ${PABX_PHONE3},1,Answer(500)
same => n,Dial(PJSIP/${PABX_PHONE3},30,rH)
same => n,Playtones(busy)
same => n,Busy(3)


;*-------------------------------------------------------------------------------*
;*   System   NXDN Gen1 Trunking System
;*   From     PABX
;*   To       SU
;*
;*   JKC Extension Header      JKC-Header-Calltype: 0 Individual / 8 Group
;*                             JKC-Header-DstId: xxx...
;*                             JKC-Header-SrcType: pabx
;*                             JKC-Header-DelayTime: 0
```

```
;*                            JKC-Header-MdDelayTime: 400
;*
;*   Dialing Example          Dial SIP User ID of Telephone Interconnect GW
;*                            IVR "Hellow World"
;*                            IVR "Please dial the umber wish to Call?"
;*                                    Dial "1" To NXDN UID
;*                                    DIal "2" To NXDN GID
;*                            IVR "Please dial the umber wish to Call?"
;*                                    Dial "NXDN ID"
;*------------------------------------------------------------------------------*
exten => _6XX,1,Set(CALLED_SIP_USER_ID=${EXTEN})
exten => _6XX,n,Goto(Asterisk-pabx-ivr,s,1)

[Asterisk-pabx-ivr]
exten => s,1,Ringing()
exten => s,n,Wait(3)
exten => s,n,Answer()
exten => s,n,Playback(hello-world)
exten => s,n(menu_question_type),Background(vm-enter-num-to-call)
exten => s,n,WaitExten(10)
exten => s,n,Goto(1,1)

;Input 1 (Individual)
exten => 1,1,Goto(pabx-IVR-Individual,1,1)
;Input 2 (Group)
exten => 2,1,Goto(pabx-IVR-Group,2,1)

[pabx-IVR-Individual]
exten => 1,1,Playback(message-number)
exten => 1,2,Playback(digits/1)
exten => 1,3(menu_question_type),Background(vm-enter-num-to-call)
exten => 1,4,WaitExten(10)

exten => _XXXXX,1,NoOp("Extension "${EXTEN})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,Dial(PJSIP/${CALLED_SIP_USER_ID},20,hftb(pabx-individual^${EXTEN}^1))
same => n,Hangup

[pabx-IVR-Group]
exten => 2,1,Playback(message-number)
exten => 2,2,Playback(digits/2)
exten => 2,3(menu_question_type),Background(vm-enter-num-to-call)
exten => 2,4,WaitExten(10)

exten => _XXXXX,1,NoOp("Extension "${EXTEN})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,Dial(PJSIP/${CALLED_SIP_USER_ID},20,hftb(pabx-group^${EXTEN}^1))
same => n,Hangup
```

Step-4: Applying Asterisk Server Configuration

To apply configuration changes in Asterisk, you first need to launch the Command Line Interface (CUI). You can access Asterisk's interactive shell by running the following command in the terminal:

```
asterisk -r
```

Once inside the CUI, execute the following command to reload PJSIP-related settings:

```
pjsip reload
```

Next, to apply changes to the dial plan, run:

```
dialplan reload
```

After confirming that the settings have been successfully applied, type exit to leave the CUI. Then, stop and restart Asterisk to reload the entire configuration in a clean state by executing the following commands:

```
sudo systemctl stop asterisk
```

```
sudo systemctl start asterisk
```

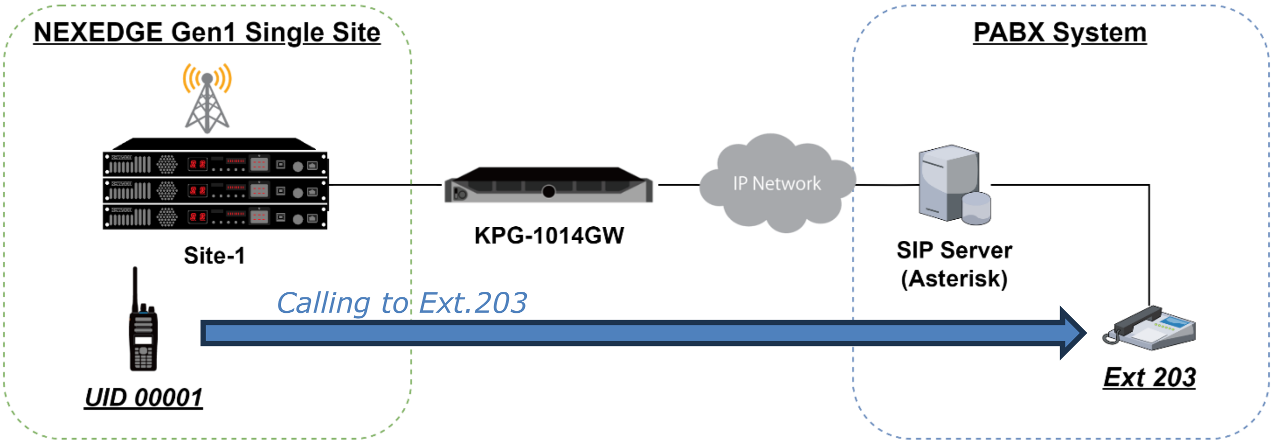
This will restart Asterisk based on the latest configuration files, ensuring stable operation.

Step-5: Call Procedure

This section explains the call procedure between a radio and a SIP phone using the Asterisk server configured in the previous steps.

<u>From Radio to SIP Phone</u>

The following example describes how to make a call from a radio to a SIP phone using the configuration shown below.



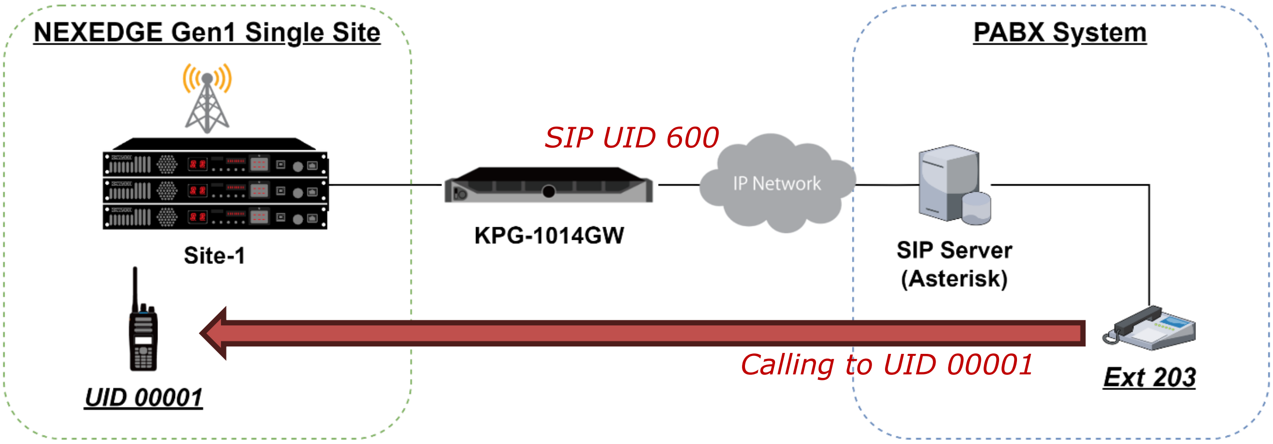
The simplest method is to configure **"Auto Dial"** in the radio's **"Menu"** and register the extension number (e.g., Ext. 203) in the **"DTMF > Auto Dial List"**. This allows easy call initiation.

1. Select **"Auto Dial"** from the **"Menu"**.
2. A list will appear; select **Ext. 203** and press **"Call (Menu)"**.
3. **"Phone Call"** will be displayed, and the SIP phone will start ringing.
4. When the SIP phone is answered, the call will begin.
5. To end the call from the radio, press **"Clear (Home)"**.

Note:
If you are using a keypad-equipped radio and want to manually enter the SIP phone's extension number, make sure to enable both **"Manual Dialing"** and **"Store and Send"** under **"DTMF > Encode/Decode"**. With these settings enabled, selecting Auto Dial will bring up a number entry screen, allowing you to input the number directly from the keypad.

From SIP Phone to Radio(Individual Call)

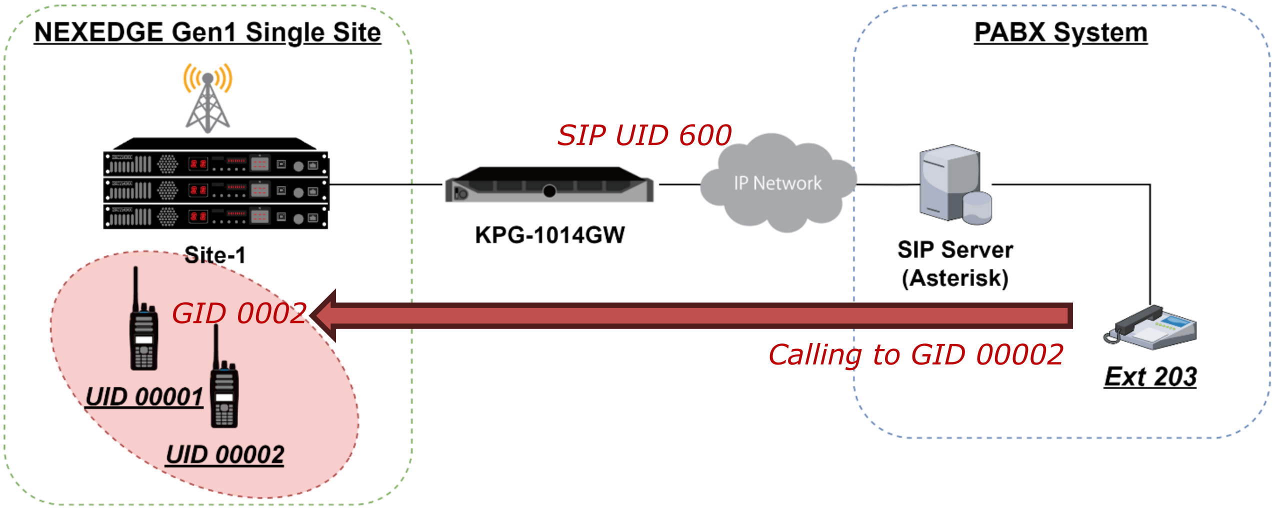
The following example describes how to make an individual call from a SIP phone to a radio using the configuration shown below.



1. On the SIP phone, dial **"600"**, which is the SIP UID assigned to the NEXEDGE Gen1 Trunking system (equivalent to an extension number).
2. You will hear a ringback tone from the handset, and once connected, a voice prompt will say, **"Please enter the number you wish to call."**
3. Press **"1"** at this point to select an Individual Call.
4. After pressing "1," another prompt will say, **"Message number is 1. Please enter the number you wish to call."** Enter the radio UID **"00001" (must be 5 digits)**.
5. You will hear a ringback tone again, and the radio will also start ringing.
6. The radio user presses the **PTT** (Push-To-Talk) button to begin the conversation.
7. To end the call, either hang up the SIP phone or press the **"Clear"** button on the radio.

From SIP Phone to Radio(Group Call)

The following example describes how to make a group call from a SIP phone to a radio using the configuration shown below.



1. On the SIP phone, dial **"600"**, which is the SIP UID assigned to the NEXEDGE Gen1 Trunking system (equivalent to an extension number).
2. You will hear a ringback tone from the handset, and once connected, a voice prompt will say, **"Please enter the number you wish to call."**
3. Press **"2"** at this point to select a Group Call.
4. After pressing "2," another prompt will say, **"Message number is 2. Please enter the number you wish to call."** Enter the Group ID **"00002" (must be 5 digits)**.
5. As soon as the GID is entered, the SIP phone will begin the call.
6. The radio will ring once and then the call will start.
7. **The call can only be disconnected from the SIP phone.** Simply hang up the handset to end the call.

3.2. Improved call procedure

In the previous chapter, when making a call from a SIP Phone to a Radio, it was necessary to enter the Radio's UID as a 5-digit number. By slightly modifying the settings in Asterisk's **"extensions.conf"**, the procedure can be changed as follows:

From SIP Phone to Radio (Pattern A)

Using "#" and "*", the system can distinguish between Individual Calls and Group Calls, and the call can be initiated using the following steps:

1. On the SIP phone, dial **"600"**, which is the SIP UID assigned to the NEXEDGE Gen1 Trunking system (equivalent to an extension number).
2. You will hear a ringback tone from the handset, and once connected, a voice prompt will say, "Please enter the number you wish to call."
3. Once connected, for an Individual Call, **enter the desired "UID No." followed by "#".** For a Group Call, **enter the desired "GID No." followed by "*"**.
4. For an Individual Call, a ringback tone will be heard on the Radio side, and the call will begin when the PTT is pressed. For a Group Call, the group call will start immediately.

From SIP Phone to Radio (Pattern B)

In advance, assign telephone numbers to Radios as follows:

- If the Unit ID is 1, assign 1601; if the Unit ID is 2, assign 1602. That is, for Unit IDs 1 to 99, assign "16 + Unit ID" as the extension number.

- If the Group ID is 1, assign 2601; if the Group ID is 2, assign 2602. That is, for Group IDs 1 to 99, assign "26 + Group ID" as the extension number.

In this case, the call can be initiated using the following steps:

1. To call a Radio, dial the assigned number from the SIP Phone. For example, to make an Individual Call to Radio with UID No. 1, dial "1601". To make a Group Call to GID No. 2, dial "2602".
2. For an Individual Call, a ringback tone will be heard on the Radio side, and the call will begin when the PTT is pressed. For a Group Call, the group call will start immediately.

Sample Configuration: extensions.conf

The following is a sample configuration (extensions.conf) that implements the two patterns described above.

```
;*********************************************************************************
;*
;*   extensions.conf
;*   2025/06/20
;*
;*********************************************************************************
[general]
writeprotect=no
priorityjumping=no

;*********************************************************************************
;*
;*   Variable Definition
;*
;*********************************************************************************
[globals]

;Asterisk Ext Phone
PABX_PHONE1=200
PABX_PHONE2=202
PABX_PHONE3=203

;KPG-1014GW SIP User ID
KPG-1014GW_SIPUSERID1=600
KPG-1014GW_SIPUSERID2=601
KPG-1014GW_SIPUSERID3=602

;*********************************************************************************
;*
;*   PJSIP Header Definition
;*
;*********************************************************************************
[pabx-individual]
exten => _X.,1,Set(PJSIP_HEADER(add,JKC-Header-Calltype)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DstId)=${EXTEN})
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-SrcType)=pabx)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DelayTime)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-MdDelayTime)=400)
exten => _X.,n,Return

[pabx-group]
exten => _X.,1,Set(PJSIP_HEADER(add,JKC-Header-Calltype)=8)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DstId)=${EXTEN})
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-SrcType)=pabx)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DelayTime)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-MdDelayTime)=400)
exten => _X.,n,Return

[pstn-individual]
exten => _X.,1,Set(PJSIP_HEADER(add,JKC-Header-Calltype)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DstId)=${EXTEN})
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-SrcType)=pstn)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DelayTime)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-MdDelayTime)=400)
exten => _X.,n,Return
```

```
[pstn-group]
exten => _X.,1,Set(PJSIP_HEADER(add,JKC-Header-Calltype)=8)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DstId)=${EXTEN})
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-SrcType)=pstn)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-DelayTime)=0)
exten => _X.,n,Set(PJSIP_HEADER(add,JKC-Header-MdDelayTime)=400)
exten => _X.,n,Return

[default]
;*----Virtual Endpoints -> NXR Endpoints+UID,GID ---*
include => virtual-endpoints

;*--------------------------------------------------------------------------------*
;*
;*   System   NXDN Gen1 Trunking System
;*   From     SU
;*   To       PABX
;*
;*--------------------------------------------------------------------------------*
;REQ PABX_PHONE1 Call PABX_PHONE1
exten => ${PABX_PHONE1},1,Answer(500)
same => n,Dial(PJSIP/${PABX_PHONE1},30,rH)
same => n,Playtones(busy)
same => n,Busy(3)

;REQ PABX_PHONE2 Call PABX_PHONE2
exten => ${PABX_PHONE2},1,Answer(500)
same => n,Dial(PJSIP/${PABX_PHONE2},30,rH)
same => n,Playtones(busy)
same => n,Busy(3)

;REQ PABX_PHONE3 Call PABX_PHONE3
exten => ${PABX_PHONE3},1,Answer(500)
same => n,Dial(PJSIP/${PABX_PHONE3},30,rH)
same => n,Playtones(busy)
same => n,Busy(3)

;*--------------------------------------------------------------------------------*
;*
;*   System   NXDN Gen1 Trunking System
;*   From     PABX
;*   To       SU
;*
;*   JKC Extension Header      JKC-Header-Calltype: 0 Individual / 8 Group
;*                             JKC-Header-DstId: xxx
;*                             JKC-Header-SrcType: pabx
;*                             JKC-Header-DelayTime: 0
;*                             JKC-Header-MdDelayTime: 400
;*
;*
;*   Dialing Example           Dial SIP User ID of Telephone Interconnect GW
;*                             IVR "Hellow World"
;*                             IVR "Please dial the umber wish to Call?"  *
;*                                     Dial "UID No." + " * " To NXDN UID
;*                                     DIal "GID No." + " # " To NXDN GID"
;*
;*--------------------------------------------------------------------------------*
exten => _6XX,1,Set(CALLED_SIP_USER_ID=${EXTEN})
exten => _6XX,n,Goto(Asterisk-pabx-ivr,s,1)

[Asterisk-pabx-ivr]
exten => s,1,Ringing()
exten => s,n,Wait(3)
```

```
exten => s,n,Answer()
exten => s,n,Playback(hello-world)
exten => s,n(menu_question_type),Background(vm-enter-num-to-call)
exten => s,n,WaitExten(10)
exten => s,n,Goto(1,1)

;Entering only the required number of digits (Example for Individual: 1#)
exten => _X#,1,Goto(pabx-IVR-Individual-Call,0000${EXTEN},1)
exten => _XX#,1,Goto(pabx-IVR-Individual-Call,000${EXTEN},1)
exten => _XXX#,1,Goto(pabx-IVR-Individual-Call,00${EXTEN},1)
exten => _XXXX#,1,Goto(pabx-IVR-Individual-Call,0${EXTEN},1)
exten => _XXXXX#,1,Goto(pabx-IVR-Individual-Call,${EXTEN},1)

;Entering only the required number of digits (Example for Group: 1*)
exten => _X*,1,Goto(pabx-IVR-Group-Call,0000${EXTEN},1)
exten => _XX*,1,Goto(pabx-IVR-Group-Call,000${EXTEN},1)
exten => _XXX*,1,Goto(pabx-IVR-Group-Call,00${EXTEN},1)
exten => _XXXX*,1,Goto(pabx-IVR-Group-Call,0${EXTEN},1)
exten => _XXXXX*,1,Goto(pabx-IVR-Group-Call,${EXTEN},1)

[pabx-IVR-Individual-Call]
exten => _XXXXX#,1,Set(DEST_ID=${EXTEN:0:5})
same => n,NoOp("Extension "${DEST_ID})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,Dial(PJSIP/${CALLED_SIP_USER_ID},20,hftb(pabx-individual^${DEST_ID}^1))
same => n,Hangup

[pabx-IVR-Group-Call]
exten => _XXXXX*,1,Set(DEST_ID=${EXTEN:0:5})
same => n,NoOp("Extension "${DEST_ID})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,Dial(PJSIP/${CALLED_SIP_USER_ID},20,hftb(pabx-group^${DEST_ID}^1))
same => n,Hangup

;*------------------------------------------------------------------------------*
;*
;*   System   NXDN Gen1 Trunking System
;*   From     PABX
;*   To       SU      (ID = 5 digit)
;*
;*   JKC Extension Header      JKC-Header-Calltype: 0 Individual / 8 Group
;*                             JKC-Header-DstId: xxx...
;*                             JKC-Header-SrcType: pabx
;*                             JKC-Header-DelayTime: 0
;*                             JKC-Header-MdDelayTime: 400
;*
;*   Dialing Example           Dial SIP User ID of Telephone Interconnect GW
;*                             Individual Call : 16xx "xx=UID" To NXDN UID"
;*                             Group Call      : 26xx "xx=GID" To NXDN GID"
;*------------------------------------------------------------------------------*
[virtual-endpoints]
; Individual-Virtual-Call (ex.:1601 -> SIP E-Point=600 UID=00001)
exten => _16XX,1,NoOp(Individual-Virtual-Call: ${EXTEN})
same => n,Set(UID=${EXTEN:2}) ; Get xx (ex.:01)
same => n,Set(FULL_UID=000${UID}) ; 5 digit zero padding(ex.:00001)
;same => n,Gosub(pabx-individual,${FULL_UID},1)
;same => n,Dial(PJSIP/600)
same => n,Answer(500)
same => n,Wait(0.1)
```

```
same => n,Playback(ascending-2tone)
same => n,Dial(PJSIP/${KPG-1014GW_SIPUSERID1},20,hftb(pabx-individual^${FULL_UID}^1))
same => n,Hangup

; Group-Virtual-Call(ex.:2601 -> SIP E-Point=600 GID=00001)
exten => _26XX,1,NoOp(Group-Virtual-Call: ${EXTEN})
same => n,Set(GID=${EXTEN:2})
same => n,Set(FULL_GID=000${GID})
;same => n,Gosub(pabx-group,${FULL_GID},1)
;same => n,Dial(PJSIP/600)
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,Dial(PJSIP/${KPG-1014GW_SIPUSERID1},20,hftb(pabx-group^${FULL_GID}^1))
same => n,Hangup
```

Revision History

Version	Date of issue	Note
1.00	16 May 2025	First Release
1.01	20 June 2025	Added improvements to Call Procedure to the Appendix