KENWOOD

Application Note

KAIROS DMR System

July 2020 Update

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Revision History

Version	Date of issue	Note
1.00	July. 30. 2020	First edition

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Contents

1. Overview

This document is described additional functionality of KAIROS DMR System on end of July 2020.

1.1. Outline Features

New features of July 2020 Version up are below.

- KPG-1013GW IP Gateway can connect to SIP Server, and NX-3000/5000 series support PABX and PSTN service.
- "Call interruption (From KAS-20C)" function can be used on DMR Tier II Conventional KAIROS System.
- "Individual Call between Consoles" can be used. A Console can call another Console via IP Gateway server.
- SNMP Manager software can get SNMP MIB information from IP Gateway server.

1.2. Version Information

Each product version of 2020 Version up are below. This document is used these versions to build the environments of new features.

<table>
<tr><th>Products</th><th colspan="2">Supported Version in this Release</th></tr>
<tr><td>NX-5000 series</td><td colspan="2">V4.30</td></tr>
<tr><td>NX-3000 series</td><td colspan="2">V3.30</td></tr>
<tr><td rowspan="6">KAIROS System</td><td>Main Firmware</td><td>1.6.B.2_stable</td></tr>
<tr><td>Controller Firmware</td><td>1.0.6.0_stable</td></tr>
<tr><td>DSP Firmware</td><td>4.64</td></tr>
<tr><td>PLD Firmware</td><td>4.13</td></tr>
<tr><td>Operating System Firmware</td><td>01.08</td></tr>
<tr><td>KAIROS Manager</td><td>1.7.10</td></tr>
<tr><td>IP Gateway (KPG-1013GW)</td><td colspan="2">KPG-1013GW Version 1.20</td></tr>
<tr><td>KAS-20C/KAS-20S</td><td colspan="2">KAS-20S/KAS-20C Version 3.10</td></tr>
</table>

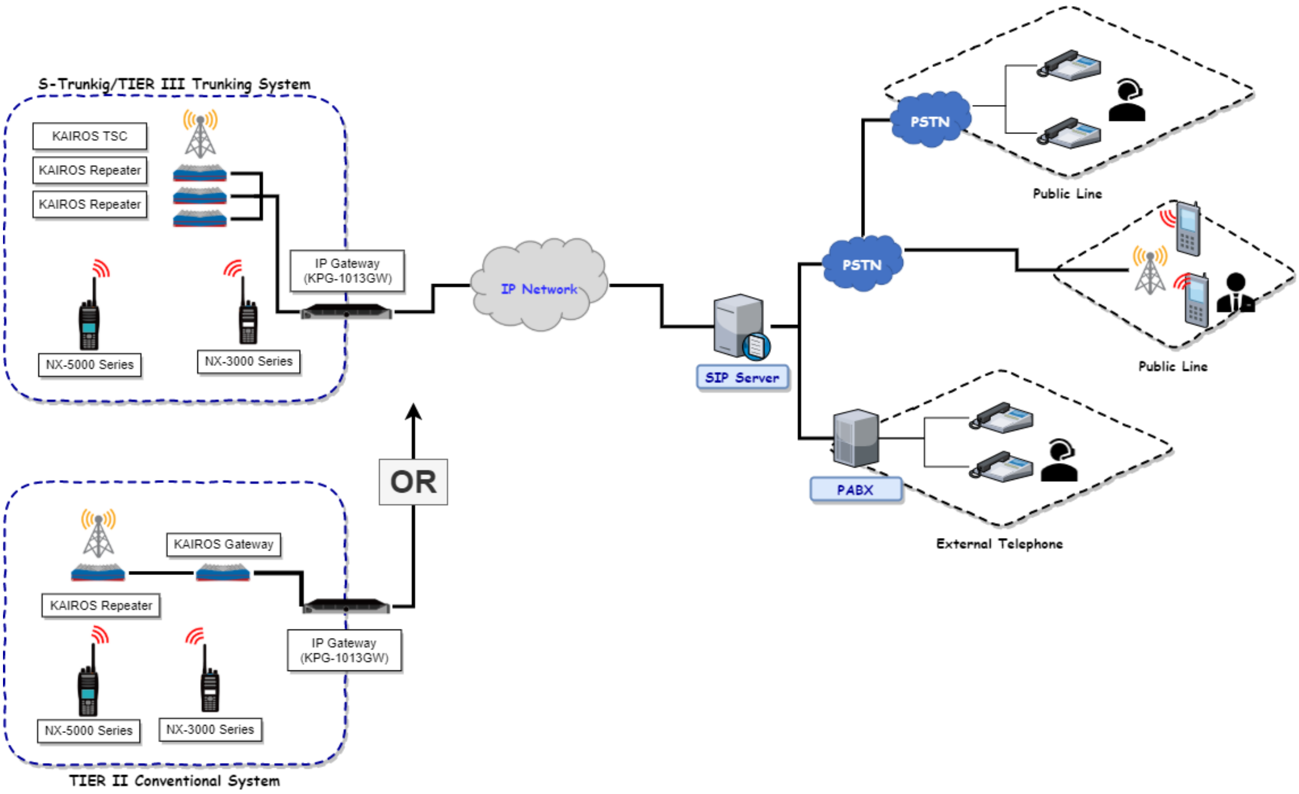
1.3. Document to reference

Before configuring the contents of this document, if you build the system with using Application Note "**AN-19-0013**" in advance, you can understand and configure with this document contents quickly.

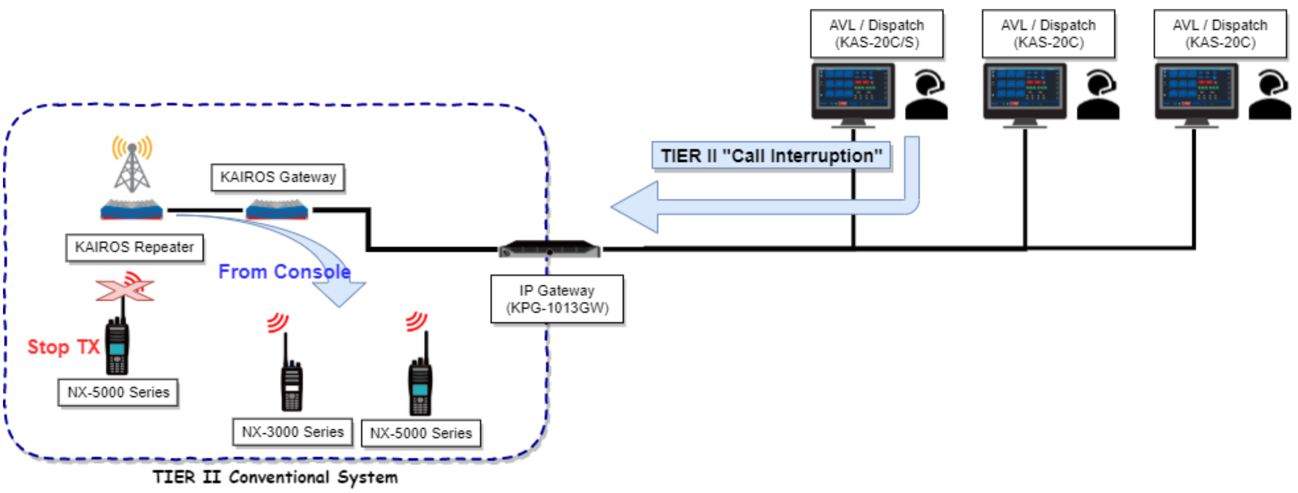
1.4. System Diagram

1.4.1. Supported to the SIP Server System (for PSTN/PABX)

To connect PABX/PSTN Phone system with KAIROS DMR based Trunking System (S-Trunking/TIER III Trunking), IP Gateway Server and SIP Server are needed as new items to construct. KAIROS Gateway is required to KAIROS TIER II Conventional System in addition to IP Gateway and SIP Server.

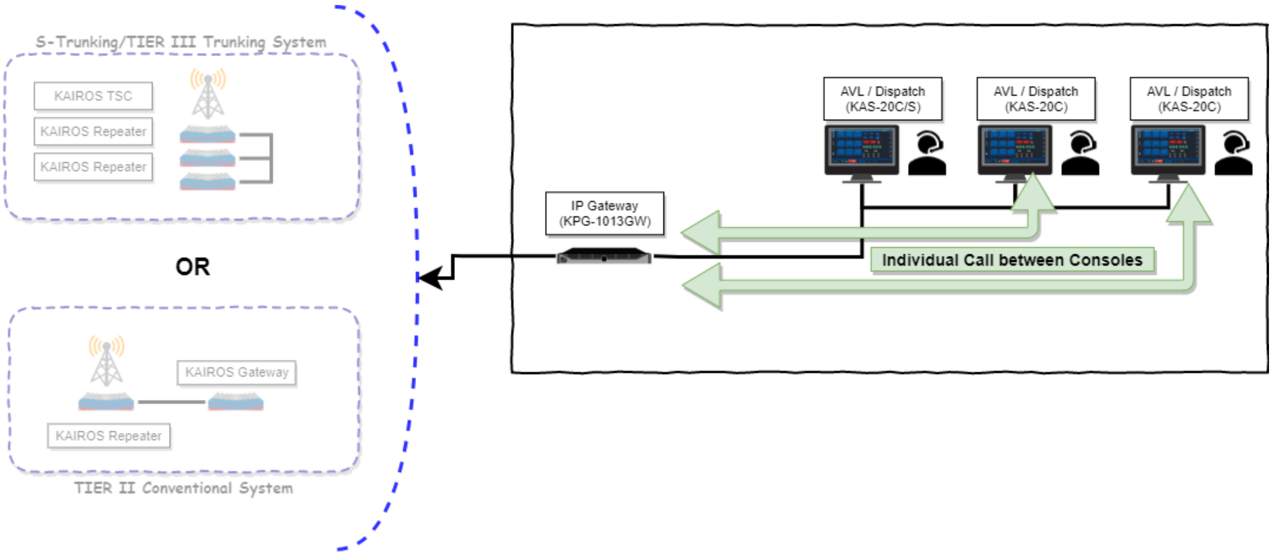


1.4.2. Call Interruption (for KAIROS TIRE II Conventional)



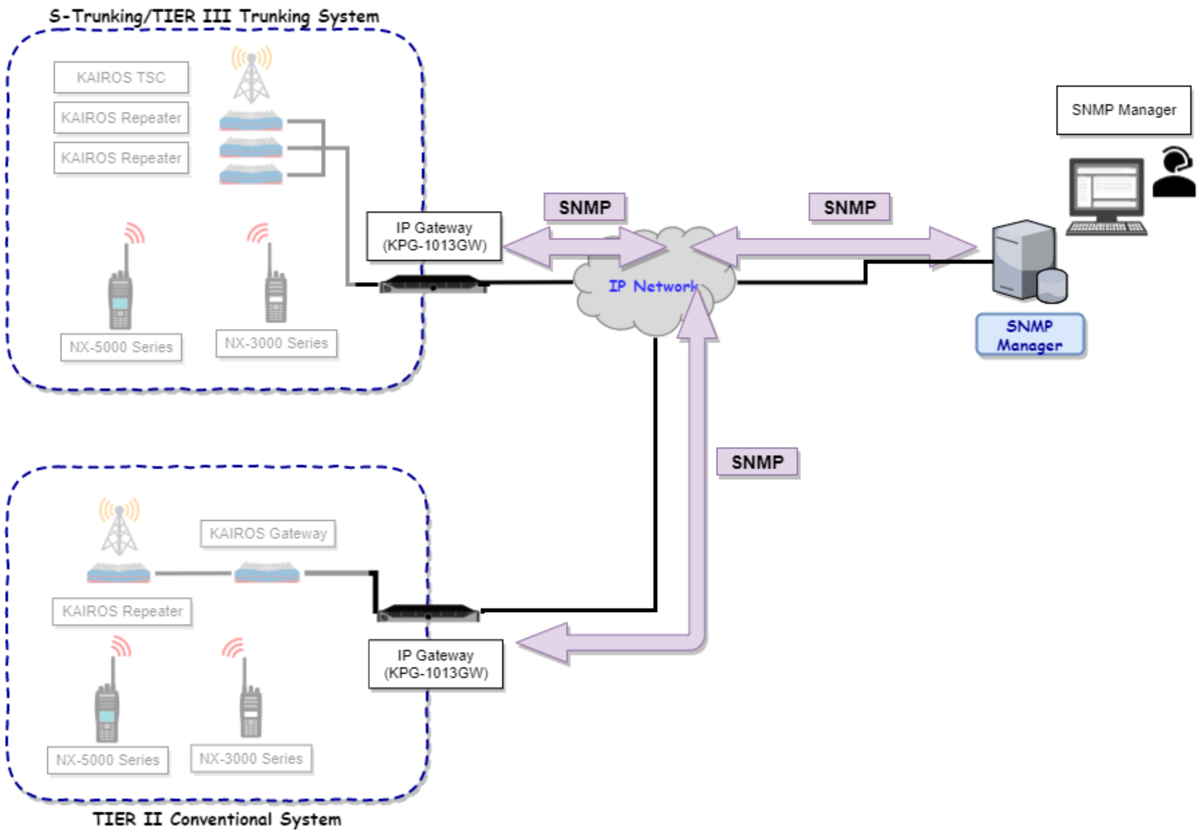
On the KAIROS TIER II Conventional System can interrupt the call of using the slot. All items are needed updating to the new version for this feature.

1.4.3. Individual Call between Consoles



Individual call between consoles in the KAIROS DMR system is managed by IP Gateway.

1.4.4. SNMP Feature (for IP Gateway Server)



System Administrator can manage the IP Gateway with SNMP Manager software.

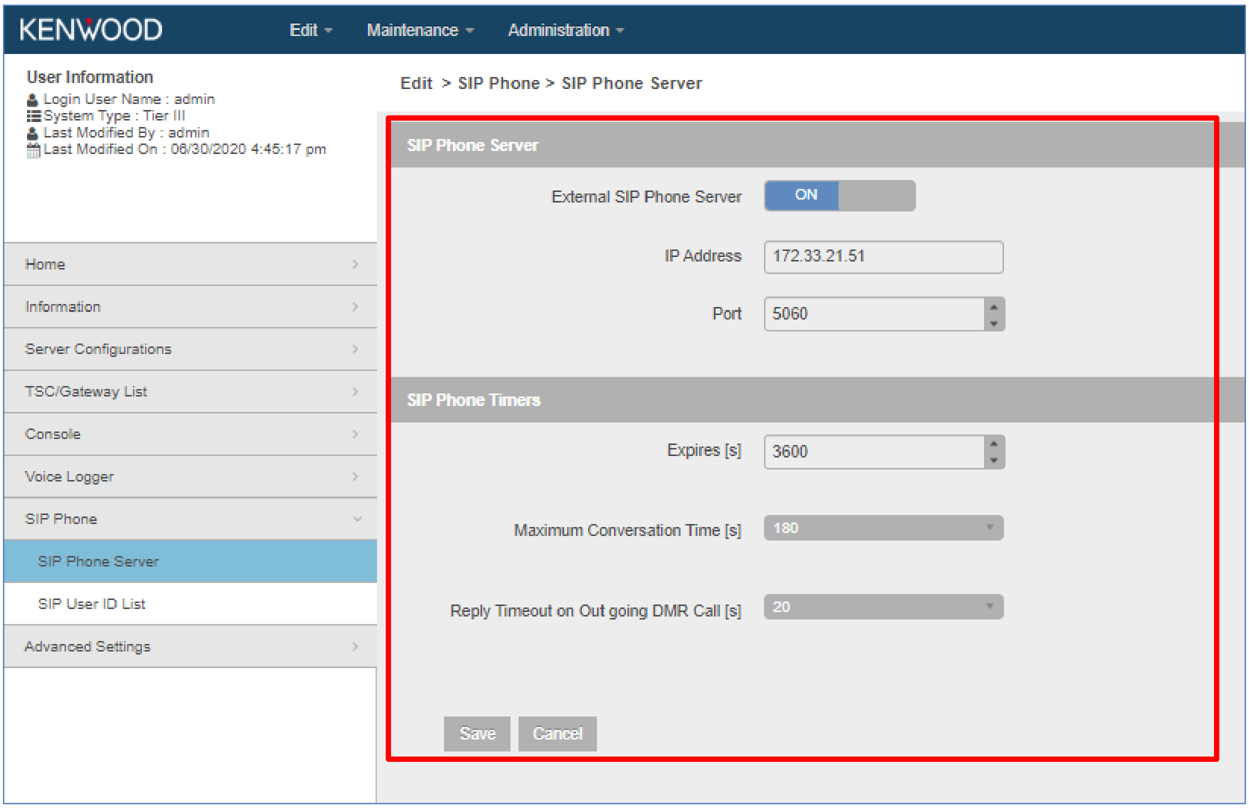
2. Basic Configuration: Connecting to the PSTN/PABX via IP Gateway

2.1. KPG-1013GW IP Gateway

2.1.1. SIP Configuration of IP Gateway for KAIROS DMR System

Before configuration of PABX/PSTN, IP Gateway should be constructed for DMR Based Truking System (S-Trunking/TIER III Trunking System) or TIER II Conventional System. (Refer to the Application Note "AN-19-0013".) This document is described that is specialized for PABX/PSTN configurations.

2.1.2. Edit>SIP Phone>SIP Phone Server



2.1.2.1. External SIP Phone Server

If the Asterisk SIP server is also installed the PC server where the IP gateway is installed, set this switch to "OFF". If the IP Gateway server is not installed with the SIP Asterisk, set this switch to "ON".

2.1.2.2. IP Address

This edit box is activated when the "External SIP phone server" is ON. Enter the IP address of the SIP server to which the IP gateway server connects.

2.1.2.3. Port

This edit box is activated when the "External SIP phone server" is ON. Enter the Port number used when connecting to SIP Server (*1). If you do not know this number, ask the system administrator of the SIP Server for the port number used for SIP.

*1: When External SIP Server is Asterisk, the default port is "5060".

2.1.2.4. Expires [s]

The IP Gateway Server registers with the SIP Server and keep the connection. However, since this connection is periodically disconnected, it will need to re-register. "Expires[s]" is expiration of registration time (*2). Before the number of seconds set in Expires to keep the connection alive by re-register.

*2: Default time is "3600" sec.

2.1.2.5. Maximum Conversation Time [s]

As configuring the "Maximum Conversation Time [s]" (*3), it can prevent the call line from being occupied for a long time after a call is established between the Radio and the phone via the IP Gateway. The IP Gateway automatically hangs up after the time set here. This parameter is for TIER II Conventional only.

*3: Unlimited : There is no limit on call time.

120-600[s] : After the call is established, when the Radio or Console releases the PTT, the call will be disconnected if this time has expired. If the time expires while the PTT is pressed, the call will be disconnected when the PTT is released.

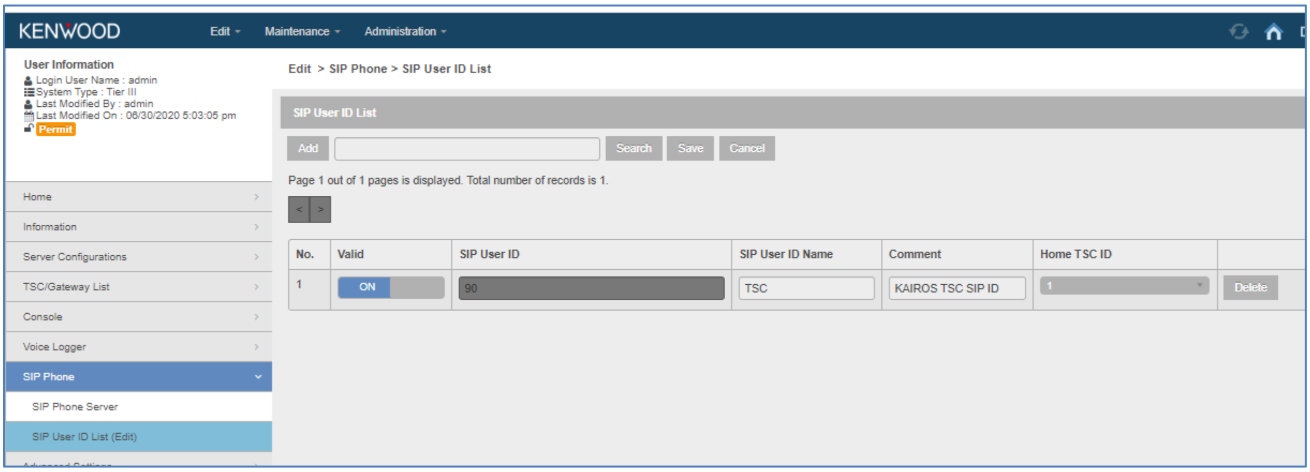
2.1.2.6. Reply Timeout on Outgoing DMR Call [s]

As configuring the "Reply Timeout on Outgoing DMR Call [s]" (*4), it is started this timer when the Radio is stopped response and detected the no voice packets. If voice packets do not detect until expired this timer, the call line is disconnected. This parameter is for TIER II Conventional only.

*4: Unlimited : There is no limit on call time.

20-120[s] : After the call is established, If it detected no voice from the Radio, the call will be disconnected when does not detect voice packets until this time has expired.

2.1.3. Edit>SIP Phone>SIP User ID List (DMR Based Trunking)



When user selects the menu in Web Interface, the "Edit" button is displayed at the top. Press the "Edit" button and the "Add" button will appear as above. Click the "Add" button to add a new SIP User ID. A row is added to the table and the following items can be set.

2.1.3.1. Valid

It is used when user wants to stop the configured SIP ID. By setting Valid OFF, the corresponding SIP ID cannot be used. (Common setting item with TIER II Conventional)

2.1.3.2. SIP User ID

It can configure the User ID to connect to the SIP Server. The IP Gateway uses this User ID to register to the SIP Server. Therefore, this ID must be registered in SIP Server in advance as a SIP user. Contact System administrator and get the SIP User ID in advance. (Common setting item with TIER II Conventional)

2.1.3.3. SIP User ID Name

It can enter a name that indicates the SIP User ID. This setting makes it easier to identify and manage within the Web Interface. (Common setting item with TIER II Conventional)

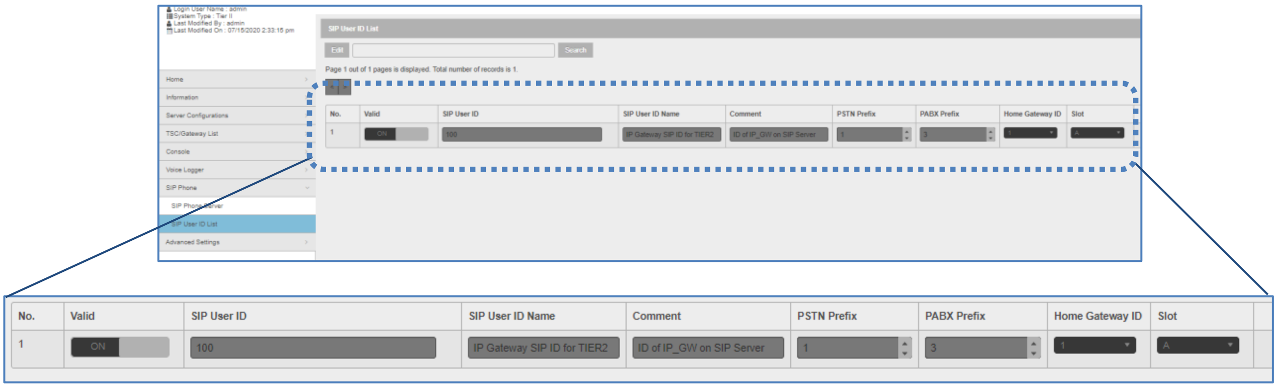
2.1.3.4. Comment

It can enter any comments. This setting makes it easier to manage simple information within the Web Interface. (Common setting item with TIER II Conventional)

2.1.3.5. Home TSC ID

It can configure the DMR Based Trunking System (S-Trunking/TIER III Trunking System) that user wants to connect to SIP Server. Select the table line number of the corresponding TSC that is configured in the "TSC/Gateway List" in advance.

2.1.4. Edit>SIP Phone>SIP User ID List (DMR TIER II Conventional)



2.1.4.1. Valid, SIP User ID, SIP User ID Name, Comment

This is the same setting item as the DMR Based Trunking System. (Refer to DMR Based Trunking System setting item)

2.1.4.2. PSTN Prefix/ PABX Prefix

It can configure the Prefix Number for connecting to PSTN and PABX. When the Radio makes a call, this prefix number must be added to the Phone number. (ex. Extension Phone number "200". PABX Prefix is "3". In this case, call number is "3200".)

2.1.4.3. Home Gateway ID

It can configure the TIER II Conventional System that user wants to connect to SIP Server. Select the table line number of the corresponding “KAIROS Gateway” that is configured in the "TSC/Gateway List" in advance.

2.1.4.4. Slot

It can select the Slot for the PSTN/PABX communication. It can select from "Slot A" or "Slot B".

2.2. KAIROS DMR Repeater (DMR Based Trunking)

Before configuring of the PABX/PSTN settings, KAIROS DMR Based Trunking System should be configured. (Refer to “AN-19-0013”.) This document describes for PSTN/PABX and provides important confirmation points after configured the system.

2.2.1. KA-NMS

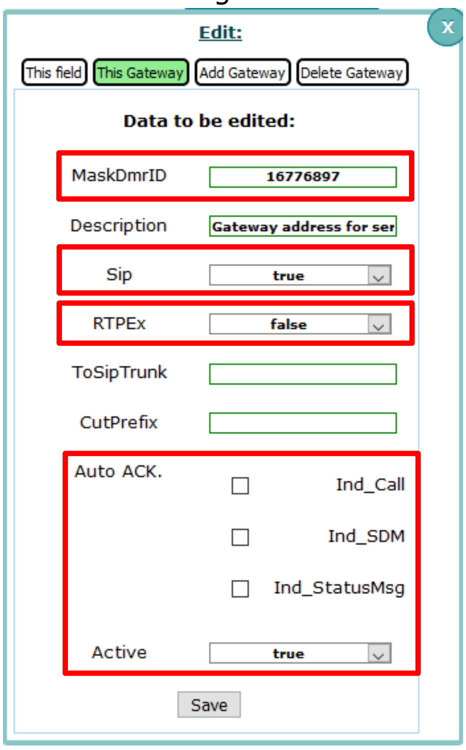
2.2.1.1. Gateways

KAIROS TSC recognizes PABX and PSTN System as unique DMR IDs. In case of DMR Based Trunking System (S-Trunking, TIER III Trunking), the following DMR IDs are assigned as Gateway. (These ID's are already registered to the Gateway table on the NMS.)

・PABX : 16776897[DEC]（FFFEC1[HEX]）
・PSTN : 16776896[DEC]（FFFEC0[HEX]）

➢ **PABX/PSTN**

It should be configured below values.



MaskDmrID: The following IDs can be selected from the rows in the table.
(PABX) “16776897”.
(PSTN) “16776896”.

Description: It is recommended to describe so that it is identified as PABX (or PSTN).

Sip:
(PABX/PSTN) Select “true”.

RTPEx:
(PABX/PSTN): Select “false”.

ToSipTrunk: (blank)

CutPrefix: (blank)

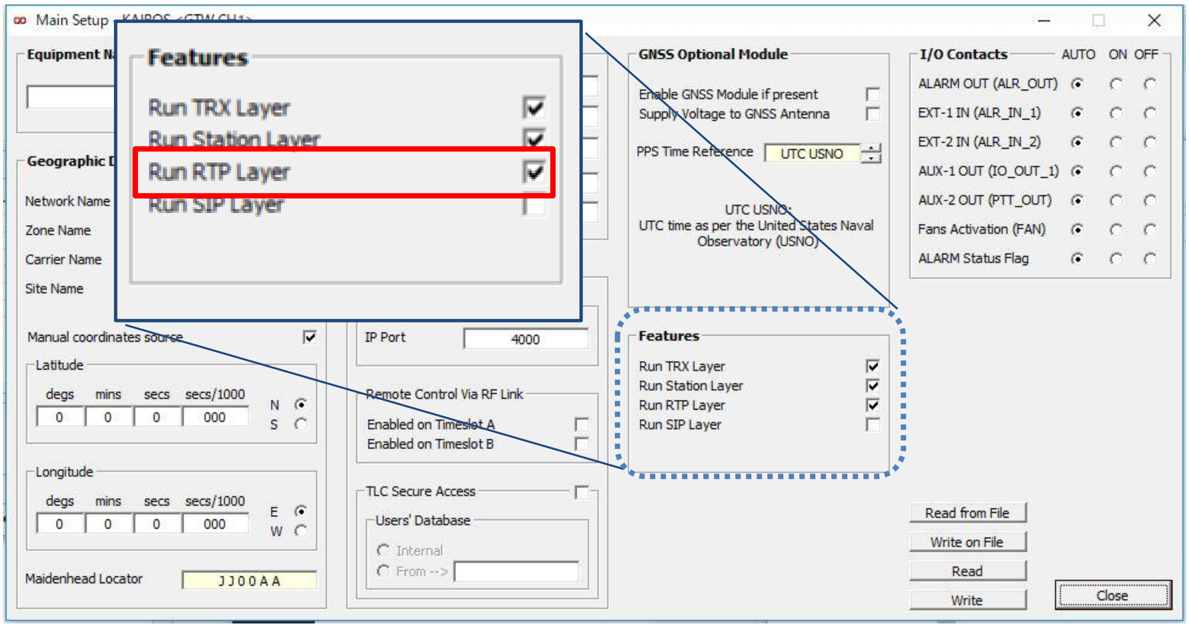
Auto ACK.:
(PABX/PSTN) Set to all “Un-check”.

Active:
(PABX/PSTN): Select “true”.

2.2.2. KAIROS Manager Settings

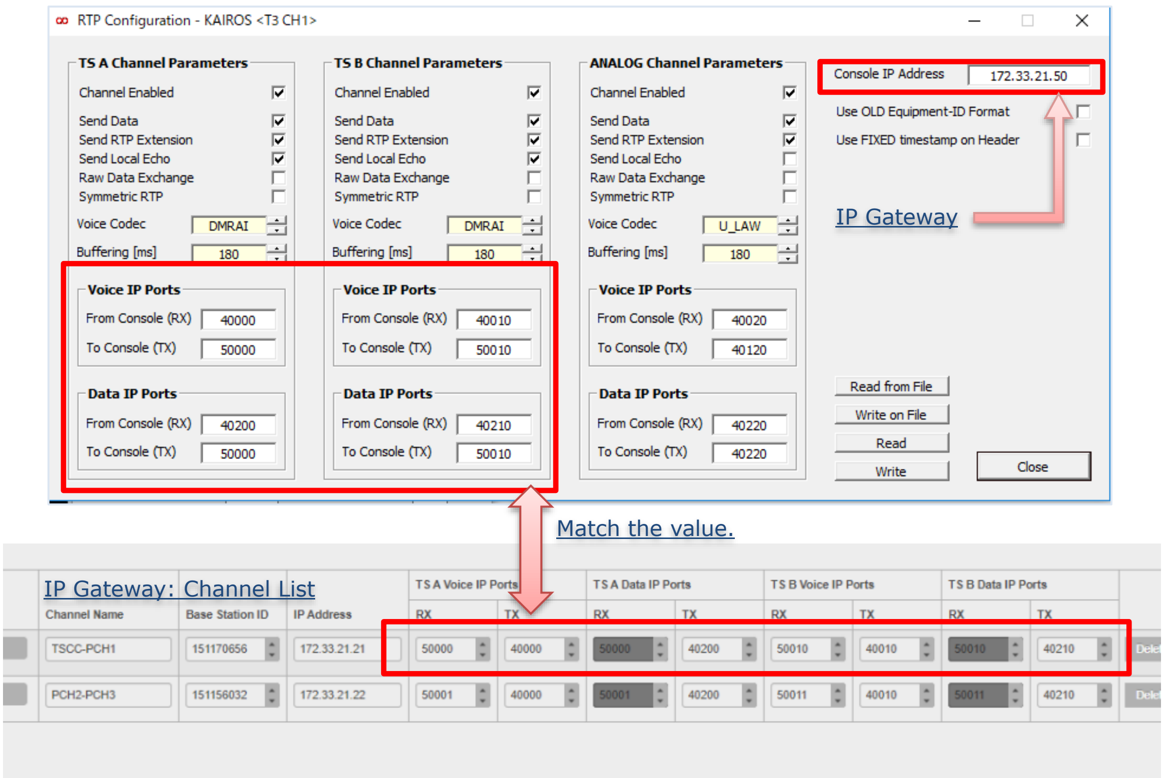
2.2.2.1. Main setup

Check the below red rectangle.



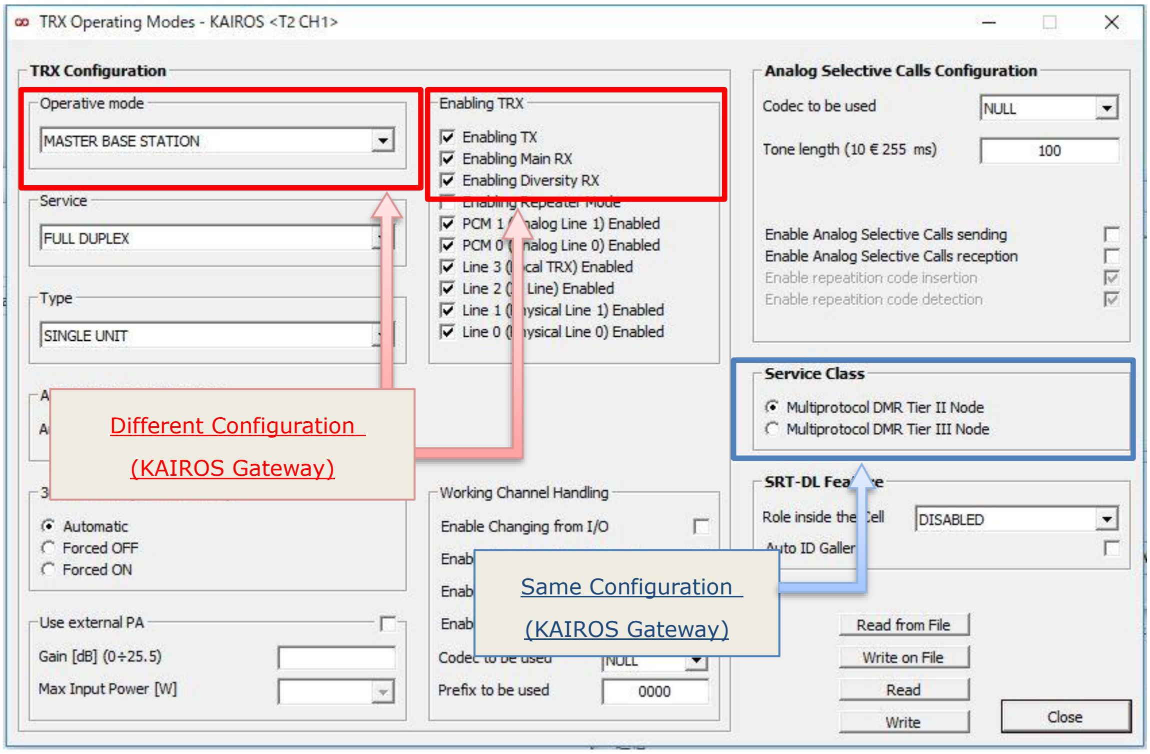
2.2.2.2. RTP Configuration

Check the below rectangle and compare with IP Gateway "Channel List".



2.3. KAIROS DMR Repeater & Gateway (DMR TIER II Conventional)

Before configuring of the PABX/PSTN settings, KAIROS TIER II Conventional System should be configured with KAIROS Master Repeater and KAIROS Gateway. (Refer to "AN-19-0013".) This document provides important confirmation points after configured the system. (It uses **KAIROS Manager only**.)



2.3.1. TRX Operating Modes

Master Repeater

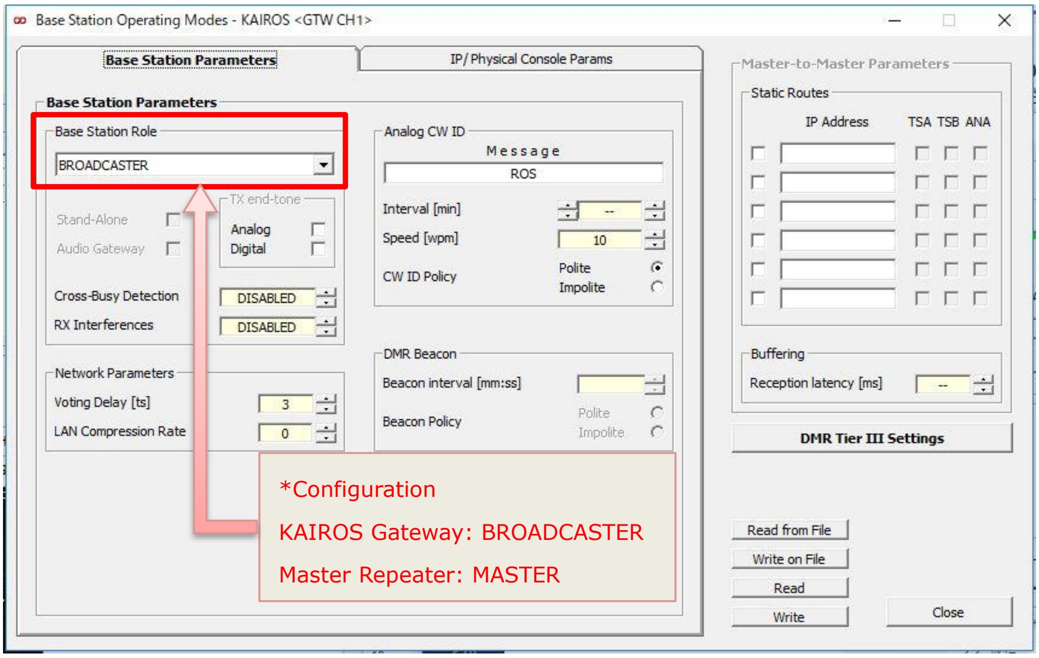
Operative mode: "MASTER BASE STATION" must be selected.
Service Class: Checked "Multiprotocol DMR Tier II Node"
Enabling TRX: Checked "Enabling TX", "Main RX" and "Enabling Diversity RX"

KAIROS Gateway

Please confirm whether the KAIROS Gateway configurations are below.
Operative Mode: Selected "SLAVE BASE STATION".
Service Class: Checked "Multiprotocol DMR Tier II Node".
Enabling TRX: Unchecked "Enabling TX", "Main RX" and "Enabling Diversity RX".

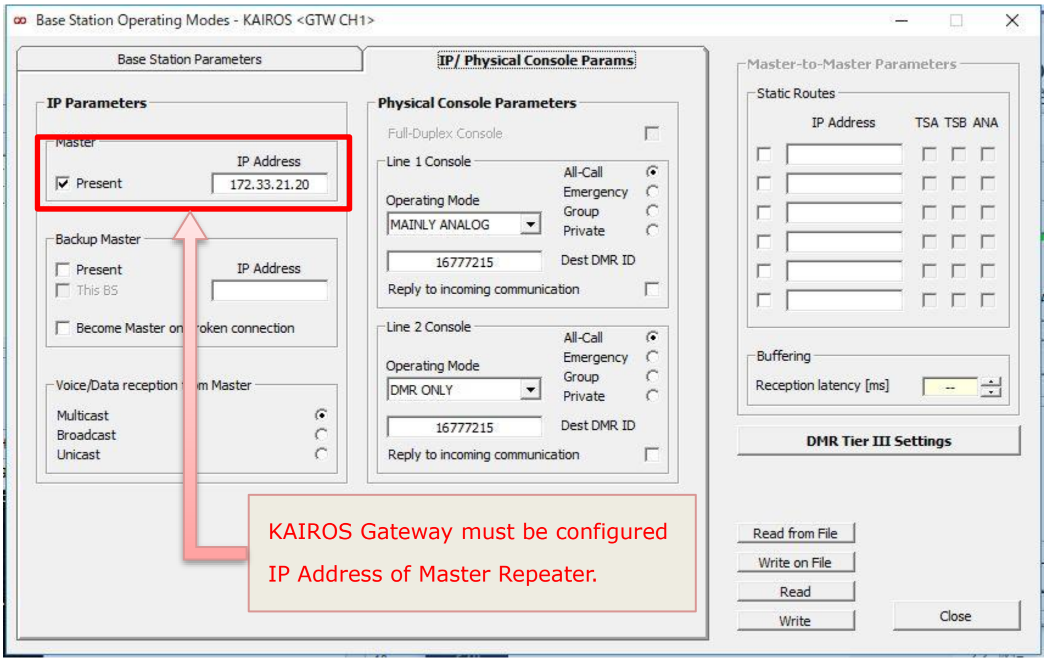
2.3.2. Base Station Operating Modes

➢ Base Station Parameters



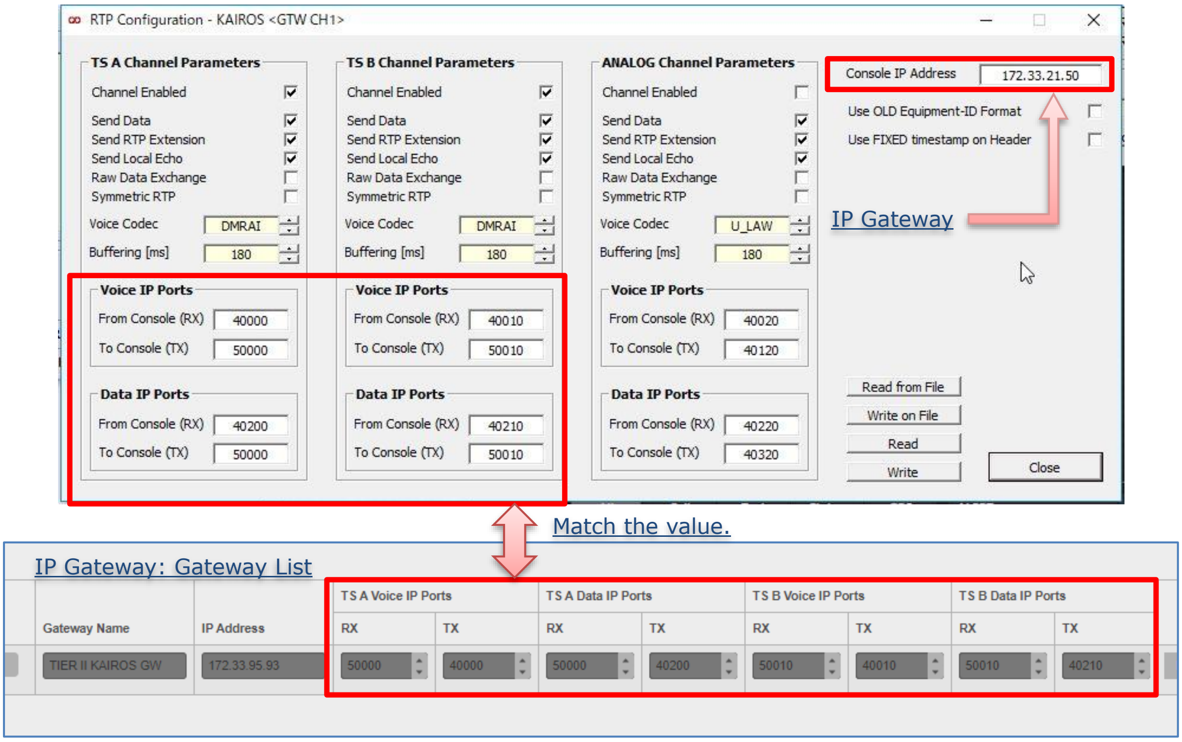
Check the parameter of "Base Station Role". Please confirm the KAIROS Gateway should be configured to "BROADCASTER". Master Repeater should be configured to "MASTER".

➢ IP / Physical Console Params



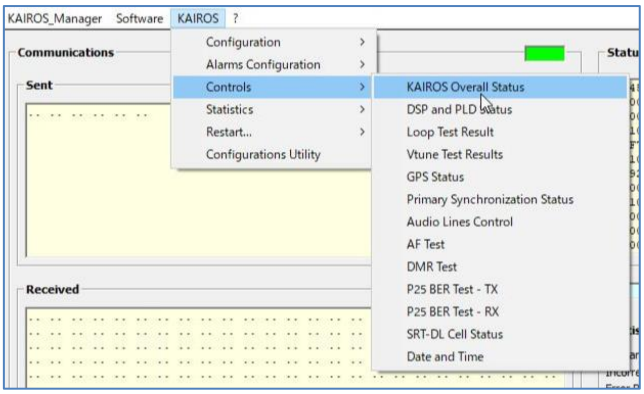
KAIROS Gateway must be configured IP Address of Master Repeater. (Master Repeater is not able to set the value of this box.)

2.3.3. RTP Configuration



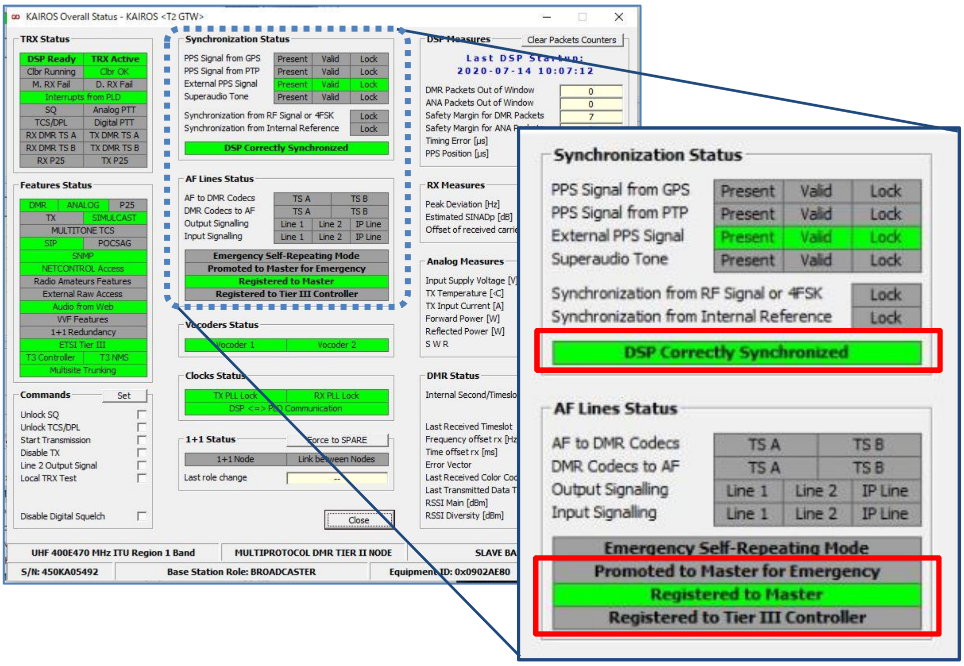
“RTP Configuration” should be configured to KAIROS Gateway only. Voice Packets are received and transmitted on KAIROS Gateway with this RTP Ports. KAIROS Gateway port numbers must be configured correspond value of the port numbers in "Gateway List" of IP Gateway.

2.3.4. KAIROS Overall Status



Finally, it should check the status in case of KAIROS TIER II system (especially KAIROS Gateway). Please select "KAIROS->Controls->KAIROS Overall Status” on the menu of KAIROS Manager.

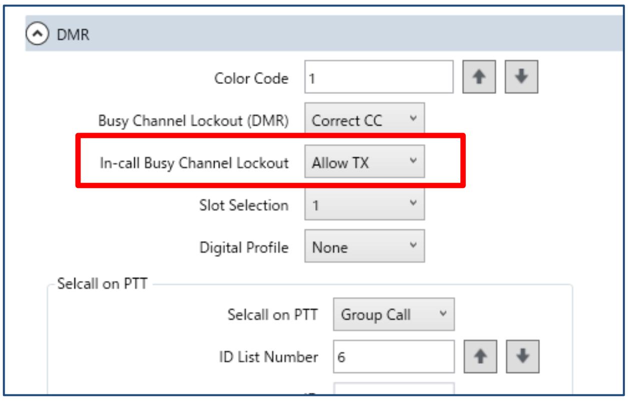
Below is sample status image of KAIROS Gateway. Please confirm the two items in the zoom up image. First, whether the sync status is colored to green (that mean is in sync). Second, whether KAIROS Gateway registered to the Master Repeater. The sample image shows that KAIROS Gateway registered to Master Repeater.



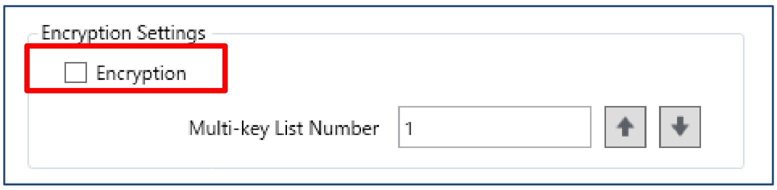
2.4. NX-3000/5000 Series

2.4.1. NX-5000 series (KPG-D1)

2.4.1.1. Transceiver Settings > Zone/Channel > Channel Edit > DMR

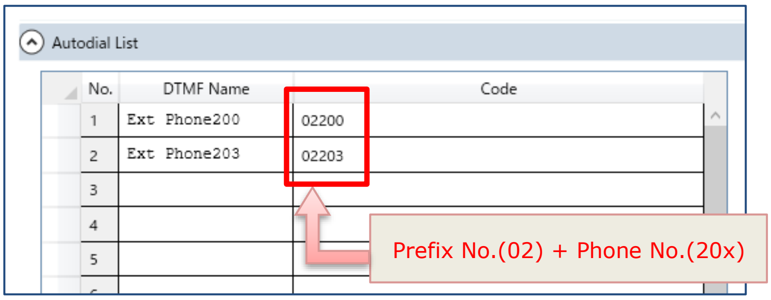


NX-5000 series should be selected “Allow TX“ at the item of ”In-call Busy Channel Lockout”. In addition, “Encryption” should be unchecked.



Note: To connect the radio to PSTN/PABX, it is necessary to configure as the above. <u>This is a common setting for Trunking System and Conventional System.</u>

2.4.1.2. Transceiver Settings > DTMF > Autodial List



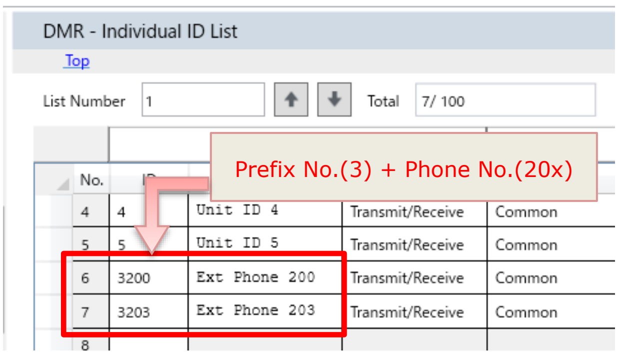
On DMR Based Trunking System (S-Trunking/TIER III Trunking System), if the user wants to make a call from radio, select to the phone number that is set in the "Autodial List". On the column of "Code" should be filled the telephone number that is configured with "Prefix No. + Phone No.". Prefix numbers are static assigning below.

PSTN Prefix Number	01
PABX Prefix Number	02

Note: It recommend assigning "Autodial" function to an empty Key or add it to the menu so that the user can easily select the "Autodial List". <u>This is a configuration for Trunking System only.</u> On the TIER II Conventional System is different procedure for making a call.

2.4.1.3. Transceiver Settings > DMR > Individual ID List

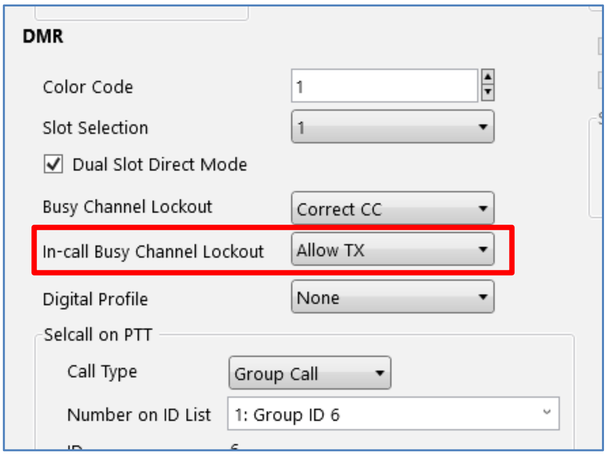
"Autodial List" cannot use on the TIER II Conventional System. "Individual ID list" can be used instead of "Autodial List". If the user wants to make a call, Phone number with Prefix number should be listed in this table in advance. (eg. If the PABX prefix number is "3" and extension phone number is "200", ID number should be "3200".)

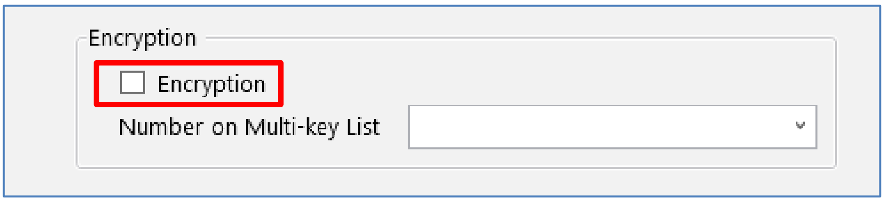


PSTN Prefix Number	Value of the "PSTN Prefix (SIP User List)" on the IP Gateway
PABX Prefix Number	<u>Above sample is "3"</u> Value of the "PABX Prefix (SIP User List)" on the IP Gateway

2.4.2. NX-3000 Series (KPG-D3)

2.4.2.1. Radio Configuration > System Properties > Systems > Personalities

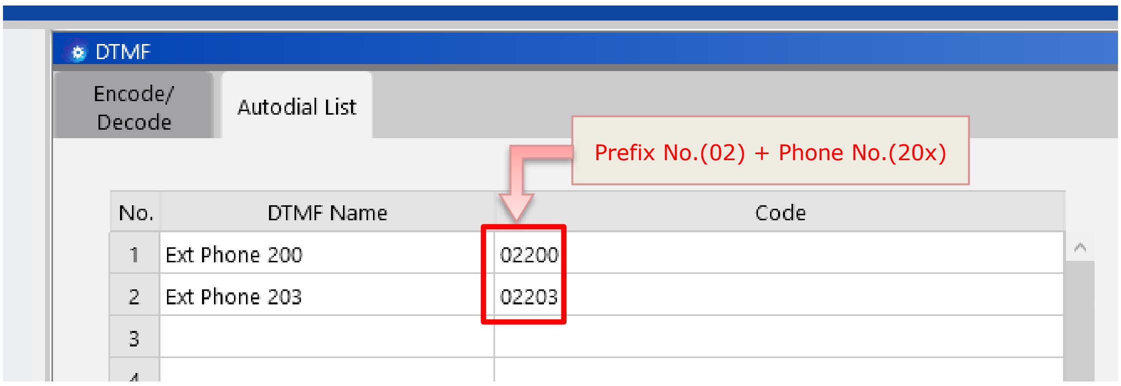




It is same configuration that NX-5000 series is configured.

Note: To connect the radio to PSTN/PABX, it is necessary to configure as the above. These are common setting for Trunking System and Conventional System.

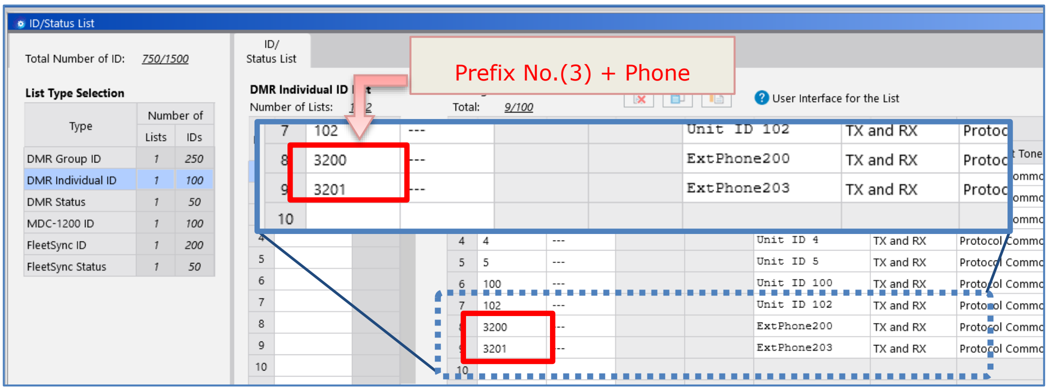
2.4.2.2. Radio Configuration > Protocol Option > DTMF > Autodial List



It is same configuration that NX-5000 series is configured.

Note: It recommend assigning "Autodial" function to an empty Key or add it to the menu so that the user can easily select the "Autodial List". This is a configuration for Trunking System only. On the TIER II Conventional System is different procedure for making a call.

2.4.2.3. Radio Configuration > System Properties > ID/Status List



It is same configuration that NX-5000 series is configured.

Note: This is a configuration for Conventional System only.

2.5. How to make a call

2.5.1. From PABX to the Radio

If telephone user wants to talk to the radio, telephone user should enter combined number that is the Prefix number and UID (or GID) number. If the Prefix number for the Individual Call is "11" and UID is "3", telephone user should enter "113". (Group Call can be selected as the Prefix. see "Appendix".) When the radio receives a call from the telephone, the ring tone is output from the speaker. Pressing PTT can establish the call.

Note: This is common procedure to call from the telephone to the KAIROS DMR system (DMR Based Trunking System/TIER II Conventional System). If telephone user from PSTN wants to talk to the radio, call the telephone number which is set in the IP Gateway at first. The next procedure is the same as PABX call procedure.

2.5.2. From the Radio to PSTN/PABX

➢ KAIROS DMR Based Trunking System (S-Trunking/TIER III Trunking System)

The radio user selects the phone number from the "Autodial List" and press the “Page” button (start Paging Call). Then, calling tone is output from the speaker. The call will be established when the phone user answers. To hung up the call press “Clear” key on the radio.

➢ KAIROS TIER II Conventional

The radio user selects a phone number entry from the Individual Call list or enter the Individual Call destination manually. After that, the radio user presses the “Page” button (start Paging Call). Then, calling tone is output from the speaker. The call will be established when the phone user answers. To hung up the call press “Clear” key on the radio.

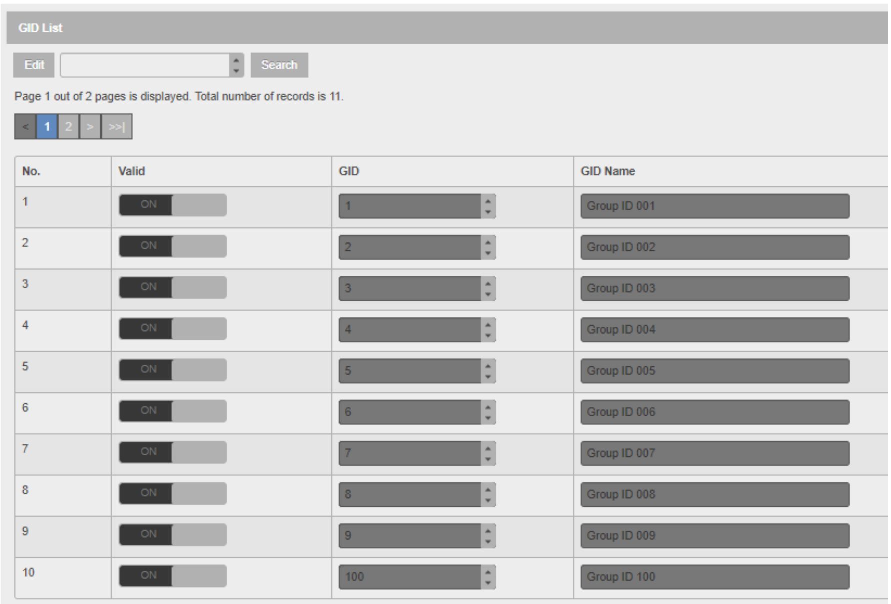
3. Basic Configuration: Call Interruption (Tier II Conventional)

3.1. KPG-1013GW IP Gateway

Before configuring of "Call Interruption" function, IP Gateway should be constructed for TIER II Conventional System. (Refer to the Application Note “AN-19-0013”.)

3.1.1. Edit > Console > GID List

“GID List” should be configured for Consoles.



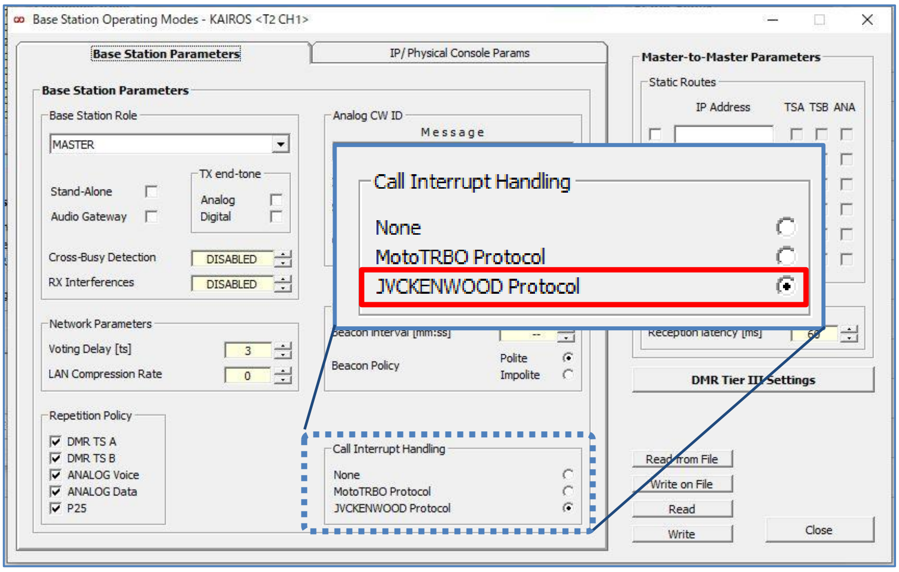
3.2. KAIROS DMR Repeater and KAIROS Gateway

3.2.1. KAIROS Repeater & Gateway

Before configuring of "Call Interruption" function, KAIROS Repeater and KAIROS Gateway should be constructed for TIER II Conventional System. (Refer to the Application Note “AN-19-0013”.)

3.2.1.1. Base Station Operating Modes

Please select the menu on KAIROS Manager "KAIROS > Configuration > Base Station Operating Modes" in the Master Repeater. "Base Station Parameters" Tab can configure for the "Call Interrupt Handling". Select protocol is "JVCKENWOOD Protocol".

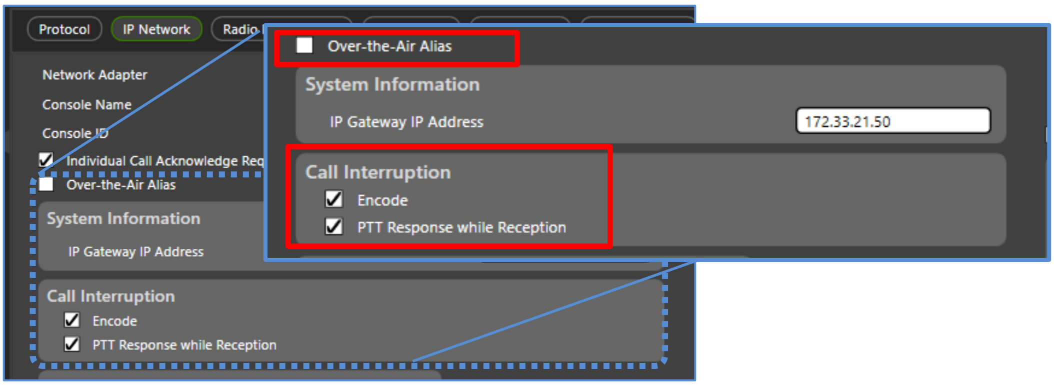


3.3. KAS-20C/KAS-20S Dispatch Console System

Before configuring of "Call Interruption" function, KAS-20 should be constructed for TIER II Conventional System. (Refer to the "Basic Configuration" in Application Note "AN-19-0013".)

3.3.1. Call Interruption: Encode

Select "Setup >Setting" and press "IP Network" where "Protocol" is "DMR: Conventional". There is a "Call Interruption" setting item in the middle of the screen. Configure the Check to "Encode". In addition, configure the Uncheck to "Over-the-Air Alias". Because TIER II "Call Interruption" function and "Over the Air Alias" function cannot be used at the same time.



➢ Example of "Call Interruption on the KAS-20C"
For example, when a Group Call is received as shown below, the "Call Interruption" icon appears on the Tile. (The red square on the left Tile.) User clicks this icon if user wants to interrupt this call. When the user clicks this Icon, System tries to interrupt the call. (Tile in the middle) When the call has be interrupted, the Compleate screen appears (Tile on the right) and you can press the PTT to talk.

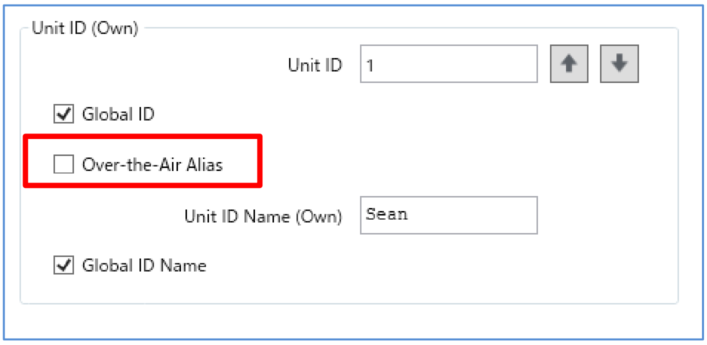
3.3.2. Call Interruption: PTT Response while Reception
When "Call Interruption" is enabled and user wants to interrupt a call, user cannot interrupt the call without clicking the icon that appears in the Tile. However, <u>if "PTT Response while Reception" is checked, user can omit the operation</u>. It selects the received Tile and presses the PTT button, then it starts the interrupt operation described above.

3.4. NX-3000/5000 Series

3.4.1. NX-5000 series (KPG-D1)

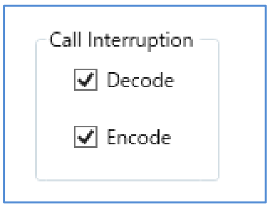
3.4.1.1. Transceiver Settings > Personal > System Information

➢ **Over-the-Air Alias**
It should be unchecked “Over-the-Air Alias” setting. If it is checked, the radio will not be able to receive the "Interrupt Request" then it will not be able to use the "Call Interruption" function.



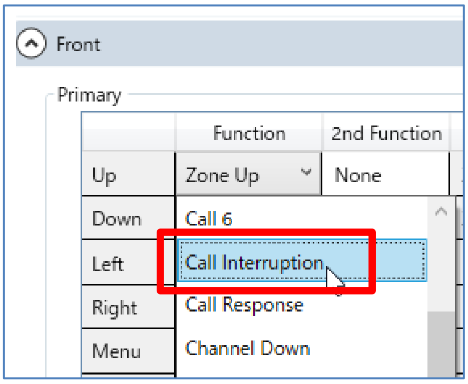
3.4.1.2. Transceiver Settings > Zone/Channel > Channel Edit > DMR

➢ **Call interruption**



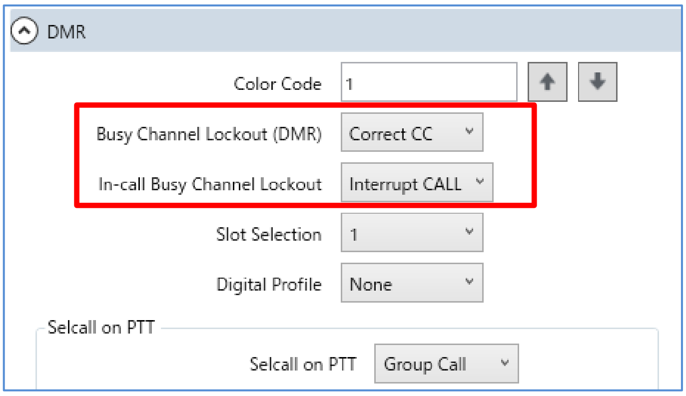
"Call Interruption" can be configured in the item of "Channel Edit". In order to use "Call Interruption" function, "Decode" should be checked. When "Decode" is enabled, the configured radio can recognize the "Interrupt request" message and stop transmission. When "Encode" is enabled, the configured radio can send "Interrupt request" message, the transmitting radio can be interrupted.

Note: User should be assigned "Call Interruption" with "Key Assignment". when the user wants to interrupt the call, push this "Call Interruption" Key.



➢ **Busy Channel Lockout / In-call Busy Channel Lockout**

Radio should not interfere with other Radio calls, so it should be configured "Busy Channel Lockout". It recommends configuring "Carrier Only" or "Correct CC". If it is configured the "In-call Busy Channel Lockout" to "Interrupt CALL", Radio can transmit the "Interrupt Request" during the in-call by simply pressing the PTT.



Busy Channel Lockout (DMR): "Carrier Only" or "Correct CC"
In-call Busy Channel Lockout: "Follow BCL" or "Interrupt CALL"

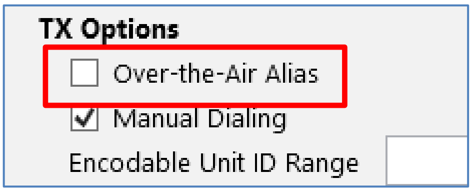
3.4.2. NX-3000 Series (KPG-D3)

3.4.2.1. Radio Configuration > System Properties > System

➢ **DMR (TAB) > TX Options**

Over-the-Air Alias

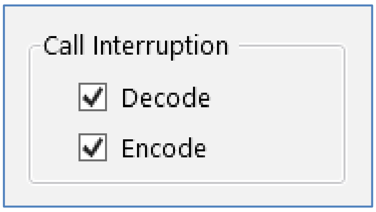
It is same configuration that NX-5000 series is configured.



➢ **Personalities (TAB) > DMR**

Call interruption

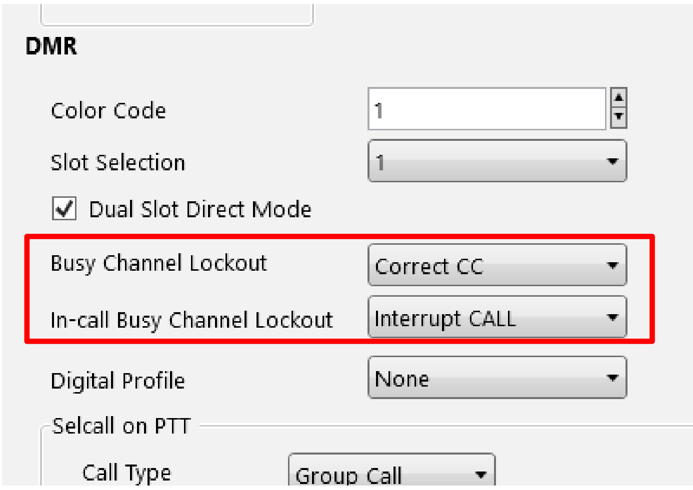
It is same configuration that NX-5000 series is configured.



Note: User should be assigned "Call Interruption" with "Key Assignment". when user wants to interrupt the call, push this "Call Interruption" Key.

Busy Channel Lockout / In-call Busy Channel Lockout

It is same configuration that NX-5000 series is configured.



Busy Channel Lockout: “Carrier Only” or “Correct CC”

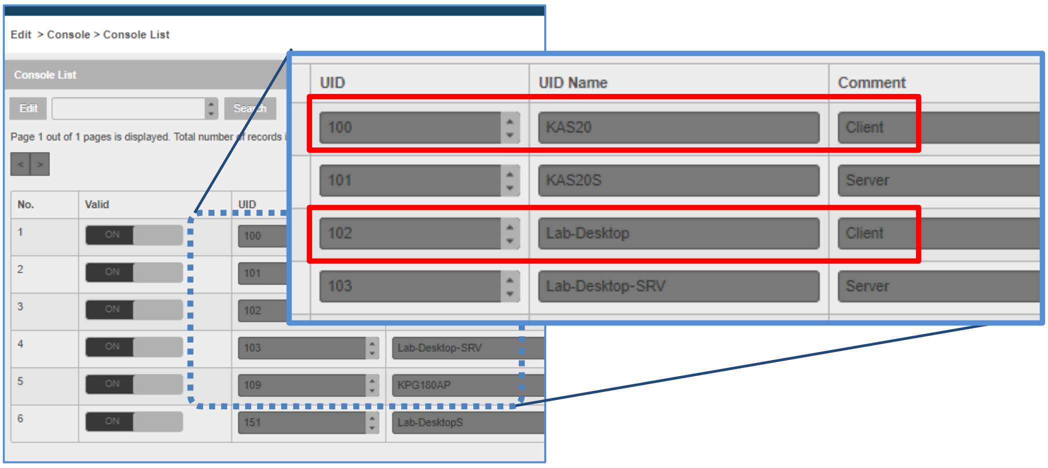
In-call Busy Channel Lockout: “Follow BCL” or “Interrupt CALL”

4. Basic Configuration: Individual Call between Consoles

4.1. KPG-1013GW IP Gateway

IP Gateway should be updated to the new version in advance, and it should be constructed for DMR Based Truking System (S-Trunking/TIER III Trunking System) or TIER II Conventional System. (Refer to the Application Note “AN-19-0013".)

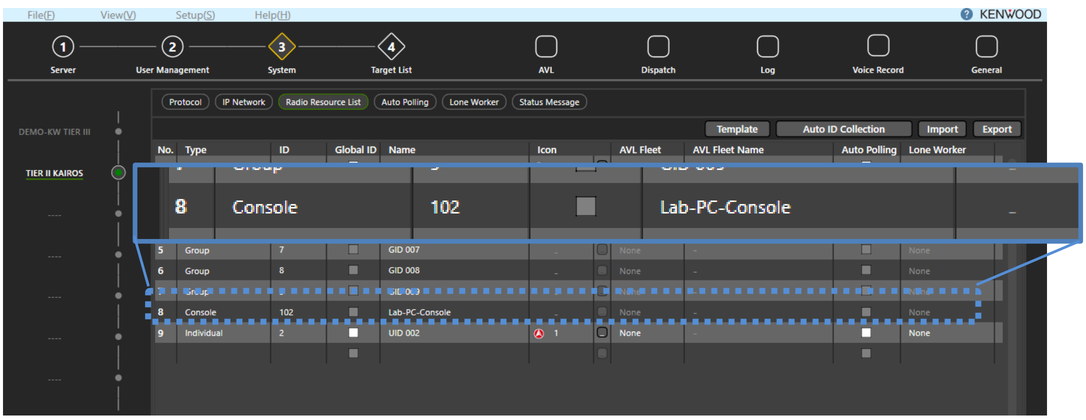
4.1.1. Edit > Console >Console List



IP Gateway can connect Individual call between KAS-20C consoles that these are registered in the "Console List".

4.2. KAS-20C/KAS-20S Dispatch Console System

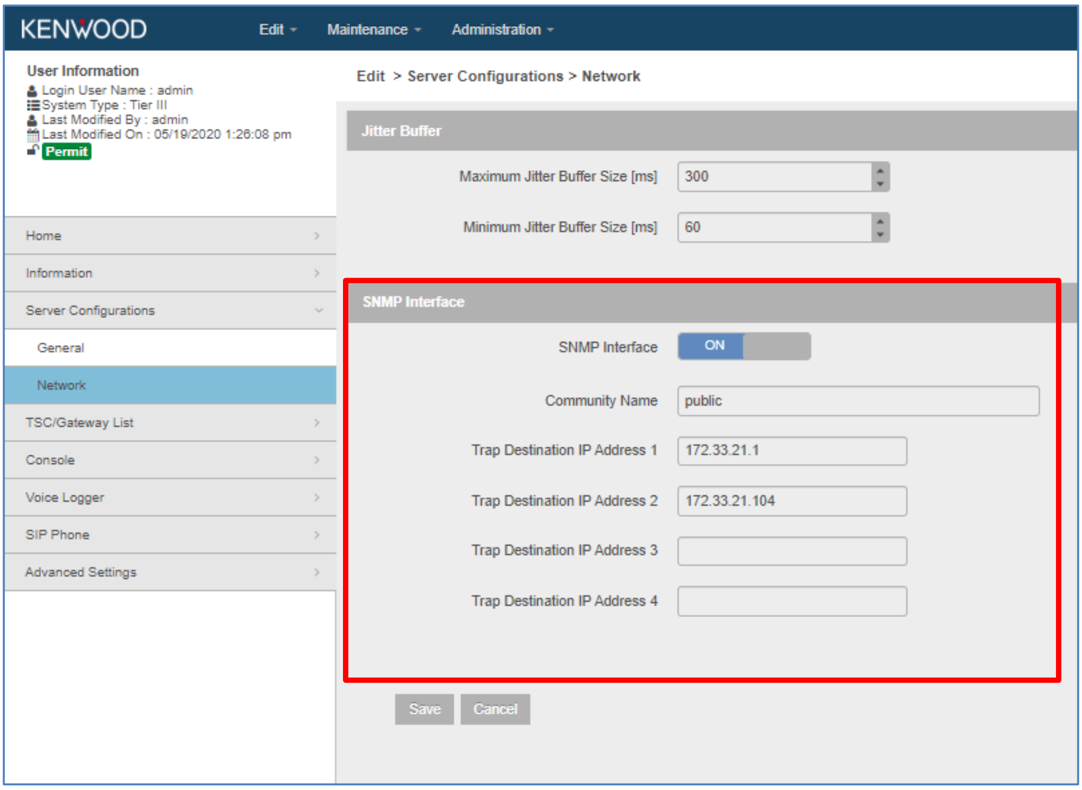
If dispatch user wants to call individually another KAS-20C console, the UID of the KAS-20C console should be registered in the "Radio Resource List" of KAS-20C. After saved, when the user selects the tile of KAS-20C and press PTT, KAS-20C can call to the target console. The call is established when the target console answered.



5. SNMP Feature (for IP Gateway)

5.1. KPG-1013GW IP Gateway

5.1.1. Edit>Server Configuration>Network



5.1.1.1. SNMP Interface

If the user wants to use the SNMP manager to acquire the SNMP MIB information of IP Gateway, "SNMP Interface" is configured "ON". The SNMP manager polls periodically requesting to get the SNMP MIB information of the IP Address set in the IP Gateway.

5.1.1.2. Community Name

If IP Gateway is received SNMP request from SNMP manager, IP Gateway is confirmed the "Community Name" that is involved the request message. If all strings are matched, IP Gateway will be able to send MIB information to the SNMP manager. Therefore, the SNMP Manager must set the same value as "Community Name" set in the IP Gateway.

5.1.1.3. Trap Destination IP Address

The trap information is transmitted from the IP Gateway to the SNMP manager when the IP Gateway has some trouble. Destination for sending the trap information is SNMP manager, and IP Gateway can be assigned four IP Address that is the destination of these.

6. Appendix

6.1. Construction of SIP Asterisk server

In this document, it is built a SIP Asterisk Server using DELL PowerEdge R340 Server. This chapter describes an example of SIP server construction using FreePBX Distro and Asterisk Ver16. And this SIP Server is used the chapter "Sample configuration".

Step1: Install FreePBX Distro

- Download the ISO image from the following URL.
  https://www.freepbx.org/downloads/
  Note: This document is used "SNG7-FPBX-64bit-2002-2.iso".
- Write iso file to the USB memory for booting. (Writing with software for creating the startup memory. There is some Free software for this type of memory making.)
- Allow USB Boot on the server BIOS settings.
- Insert the USB memory into the USB port on the front panel and restart to start the installation.
- At the end of installation, set the "root" password to finish.

Note: Startup after installation is configured with "DHCP network configuration". It is recommended that the Server should be configured a static IP address, and that it is installed using the network which can connect to the Internet.

Step2: Change the Network configuration

If the IP address has been assigned in advance by the network administrator, it is edited the file to assign the IP address. However, confirm to the network administrator whether the Server can connect to the Internet.

- Login to the Server using "root" user.
- And edit the file of "/etc/sysconfig/network-scripts/ifcfg-eth0".
- Below is sample setting of Network configuration.

```
BOOTPROTO="static"              # Change to the "dhcp" to "static"
DNS1=xxx.xxx.xxx.xxx            # This is necessary for "Internet connection"
IPADDR=172.33.21.51             # Asterisk Server IP Address
NETMASK=255.255.0.0             # subnet mask
GATEWAY=172.33.21.254           # default gateway
ONBOOT=yes
IPV6INIT=no
IPV6_AUTOCONF=no
IPV6_DEFROUTE=no
IPV6_FAILURE_FATAL=no
```

Note: If Proxy setting is required, add it to "/etc/yum.conf". Internet connection is necessary for the command "yum".

- Restart the Network service of the server by below commands.

```
systemctl␣restart␣network↲
```

Step3: Uninstall "freepbx" package

- Enter the following command as the "root" user to uninstall the "sangoma-pbx" and "freepbx" packages.

```
rm␣/etc/yum/protected.d/sangoma-pbx.conf↲          <- Remove protection of" sangoma-pbx"
yum␣remove␣sangoma-pbx␣freepbx↲
```

```
[root@freepbx ~]#
[root@freepbx ~]# rm /etc/yum/protected.d/sangoma-pbx.conf
rm: remove regular file  e/etc/yum/protected.d/sangoma-pbx.conf f? y_
```

```
[root@freepbx protected.d]# yum remove sangoma-pbx freepbx
Loaded plugins: fastestmirror, versionlock
Resolving Dependencies
--> Running transaction check
---> Package freepbx.x86_64 0:14.1-1.sng7 will be erased
---> Package sangoma-pbx.noarch 0:2002-2.sng7 will be erased
--> Finished Dependency Resolution

Dependencies Resolved

==========================================================================================================
 Package                   Arch                Version                  Repository                   Size
==========================================================================================================
Removing:
 freepbx                   x86_64              14.1-1.sng7              @anaconda/2002              1.1 G
 sangoma-pbx               noarch              2002-2.sng7              @anaconda/2002               15 k

Transaction Summary
==========================================================================================================
Remove  2 Packages

Installed size: 1.1 G
Is this ok [y/N]: y
Downloading packages:
Running transaction check
Running transaction test
Transaction test succeeded
Running transaction
  Erasing    : sangoma-pbx-2002-2.sng7.noarch                                                          1/2
  Erasing    : freepbx-14.1-1.sng7.x86_64                                                              2/2
  Verifying  : freepbx-14.1-1.sng7.x86_64                                                              1/2
  Verifying  : sangoma-pbx-2002-2.sng7.noarch                                                          2/2

Removed:
  freepbx.x86_64 0:14.1-1.sng7                          sangoma-pbx.noarch 0:2002-2.sng7

Complete!
[root@freepbx protected.d]# _
```

Step4: Install Asterisk package

- Enter the following command as the "root" user to install the "asterisk16-configs" package.

```
yum␣install␣asterisk16-configs-16.4.1 ↲
```

```
[root@freepbx protected.d]# yum install asterisk16-configs-16.4.1
Loaded plugins: fastestmirror, versionlock
Determining fastest mirrors
sng-base                                                          | 3.6 kB  00:00:00
sng-epel                                                          | 2.9 kB  00:00:00
sng-extras                                                        | 3.4 kB  00:00:00
sng-pkgs                                                          | 3.4 kB  00:00:00
sng-updates                                                       | 3.4 kB  00:00:00
(1/6): sng-base/7/x86_64/group_gz                                 | 166 kB  00:00:00
(2/6): sng-extras/7/x86_64/primary_db                             | 205 kB  00:00:00
(3/6): sng-pkgs/7/x86_64/primary_db                               | 708 kB  00:00:00
(4/6): sng-base/7/x86_64/primary_db                               | 6.0 MB  00:00:00
(5/6): sng-epel/7/x86_64/primary_db                               | 7.5 MB  00:00:00
(6/6): sng-updates/7/x86_64/primary_db                            | 6.5 MB  00:00:00
Resolving Dependencies
--> Running transaction check
---> Package asterisk16-configs.x86_64 0:16.4.1-1.sng7 will be installed
--> Finished Dependency Resolution

Dependencies Resolved

=====================================================================================
 Package                Arch          Version               Repository          Size
=====================================================================================
Installing:
 asterisk16-configs     x86_64        16.4.1-1.sng7         sng-pkgs           221 k

Transaction Summary
=====================================================================================
Install  1 Package

Total download size: 221 k
Installed size: 763 k
Is this ok [y/d/N]: y
Downloading packages:
asterisk16-configs-16.4.1-1.sng7.x86_64.rpm                       | 221 kB  00:00:00
Running transaction check
Running transaction test
Transaction test succeeded
Running transaction
  Installing : asterisk16-configs-16.4.1-1.sng7.x86_64                            1/1
warning: /etc/asterisk/res_fax.conf created as /etc/asterisk/res_fax.conf.rpmnew
warning: /etc/asterisk/smdi.conf created as /etc/asterisk/smdi.conf.rpmnew
warning: /etc/asterisk/statsd.conf created as /etc/asterisk/statsd.conf.rpmnew
  Verifying  : asterisk16-configs-16.4.1-1.sng7.x86_64                            1/1

Installed:
  asterisk16-configs.x86_64 0:16.4.1-1.sng7

Complete!
[root@freepbx protected.d]# _
```

Step5: Auto start configuration for Asterisk service

- Enter the following command as the "root" user. it makes to start automatically "Asterisk" service when the server power on.

```
chkconfig␣asterisk␣on↲
```

Step6: Configuration for "sip.conf"

- Login to the Server as "root" user.
- Change the directory as below.
- Edit the file of "sip.conf" with text editor.

```
cd␣/etc/asterisk/ ↲
```

```
[root@freepbx /]#
[root@freepbx /]# cd /etc/asterisk/
[root@freepbx asterisk]# vi sip.conf_
```

- Below is sample configure of "sip.conf" for TIER II Conventional and DMR Based Trunking.

```
;*-------- sip.conf: Sample ---------
[general]
maxexpirey=43200
context=default
bindport=5060
bindaddr=0.0.0.0
disallow=all
allow=ulaw:60
allow=alaw:60
language=en
type=friend

;-------- PSTN: Sample ---------
;register => xxxxxxxx:yyyyyyyy@xxxxxxxxxxx
;[------]
;type=friend
;username=xxxxxxxx
;fromuser=xxxxxxxx
;secret=yyyyyyyy
;host=xxxxxxx
;context=incoming
;insecure=port,invite
;canreinvite=no
;fromdomain=xxxxxxxx
;dtmfmode=rfc2833
;allowsubscribe=no

;****(to the next page....)
```

```
;****(from the previous page….)
;[xxxxxxxx]
;type=friend
;defaultuser=xxxxxxxx
;secret=yyyyyyyy
;canreinvite=no
;host=dynamic

;-------- PABX Phone(Soft phone): Sample ---------
;PABX Phone 1
[200]
type=friend
defaultuser=200
secret=pass
canreinvite=no
host=dynamic

;PABX Phone 2
[203]
type=friend
defaultuser=203
secret=pass
canreinvite=no
host=dynamic

;-------- IP Gateway: Sample ---------
;IPGW For Tier II KAIROS Slot A
[100]
type=friend
defaultuser=100
canreinvite=no
host=dynamic
dtmfmode=inband

;IPGW For S-Trunking & Tier III KAIROS TSC
[90]
type=friend
defaultuser=90
canreinvite=no
host=dynamic
dtmfmode=inband

;****(End of file)
```

Step7: Configuration for “extensions.conf”

- Edit the file of “extensions.conf” with text editor. There is the same directory path of “extensions.conf” where the directory path of “sip.conf”.

- There are two sample configurations of “extensions.conf” for TIER II Conventional and DMR Based Trunking. "sip.conf" can be used for all SIP phone systems in this Appendix, but "extensions.conf" should be used separately for TIER II Conventional and DMR Based Trunking.

For TIER II Conventional System

```
;------- extensions.conf for TIER II Conventional System -------
[general]
writeprotect=no
priorityjumping=no

;------- Variable Definition -------
[globals]
PSTN_SIP_ACCOUNT=xxxxxxxx
PSTN_PHONE1=xxxxxxxxxx
PABX_PHONE1=200
PABX_PHONE2=203
IPGW_SIPUSERID_SLOTA=100

;------- TIER II Conventional (Radio/PABX Phone -> PABX Phone) -------
[default]
;REQ PABX_PHONE1 Call PABX_PHONE1
exten => ${PABX_PHONE1},1,Answer(500)
same => n,Dial(SIP/${PABX_PHONE1},30,rH)
same => n,Playtones(busy)
same => n,Busy(3)

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

;****(to the next page….)
```

```
;****(from the previous page….)

;------- TIER II Conventional (PABX Phone -> Radio) -------
;IndividualCall
exten => _11.,1,NoOp("Extension "${EXTEN})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,SIPAddHeader(RA-Header-Calltype: 0)
same => n,SIPAddHeader(RA-Header-DstDmrId: ${EXTEN:2})
same => n,SIPAddHeader(JKC-Header-SrcType: pabx)
same => n,Dial(Sip/${IPGW_SIPUSERID_SLOTA},20,hft)
same => n,Hangup

;Group Call
exten => _22.,1,NoOp("Extension "${EXTEN})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,SIPAddHeader(RA-Header-Calltype: 8)
same => n,SIPAddHeader(RA-Header-DstDmrId: ${EXTEN:2})
same => n,SIPAddHeader(JKC-Header-SrcType: pabx)
same => n,Dial(Sip/${IPGW_SIPUSERID_SLOTA},20,hft)
same => n,Hangup

;------- TIER II Conventional (Radio -> PSTN Phone) -------
exten => 331,1,Answer(500)
same => n,Dial(SIP/${PSTN_PHONE1}@fusion,20,ftrH)
same => n,Playtones(busy)
same => n,Busy(6)

;------- TIER II Conventional (PSTN Phone -> Radio) -------
[incoming]
exten => ${PSTN_SIP_ACCOUNT},1,Goto(Asterisk-ivr,s,1)


;****(to the next page….)
```

```
;****(from the previous page….)

[Asterisk-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)

;IVR Change Connection
;Input 1 (Soft Phone)
exten => 1,1,Goto(IVR-SoftPhone,1,1)
;Input 2 (Soft Phone)
exten => 2,1,Goto(IVR-DMR-Individual,2,1)
;Input 3 (DMRID Group)
exten => 3,1,Goto(IVR-DMR-Group,3,1)

[IVR-SoftPhone]
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 => _X.,1,dial(SIP/${EXTEN})

[IVR-DMR-Individual]
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 => _XXXXXXXX,1,NoOp("Extension "${EXTEN})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,SIPAddHeader(RA-Header-Calltype: 0)
same => n,SIPAddHeader(RA-Header-DstDmrId: ${EXTEN})
same => n,SIPAddHeader(JKC-Header-SrcType: pstn)
same => n,Dial(Sip/${IPGW_SIPUSERID_SLOTA},20,hft)
same => n,Hangup

;****(to the next page….)
```

```
;****(from the previous page....)

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

exten => _XXXXXXXX,1,NoOp("Extension "${EXTEN})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,SIPAddHeader(RA-Header-Calltype: 8)
same => n,SIPAddHeader(RA-Header-DstDmrId: ${EXTEN})
same => n,SIPAddHeader(JKC-Header-SrcType: pstn)
same => n,Dial(Sip/${IPGW_SIPUSERID_SLOTA},20,hft)
same => n,Hangup

;****(end of file)
```

For DMR Based Trunking System (S-Trunking/TIER III Trunking System)

```
;------- extensions.conf for DMR Based Trunking System -------
[general]
writeprotect=no
priorityjumping=no

;------- Variable Definition -------
[globals]
PSTN_SIP_ACCOUNT=xxxxxxxx
PABX_PHONE1=200
PABX_PHONE2=203
IPGW_SIPUSERID_TSC=90

;****(to the next page....)
```

```
;****(from the previous page….)

;------- S-Trunking/TIER III Trunking (Radio/PABX Phone -> PABX Phone) -------
[default]
;REQ PABX_PHONE1 Call PABX_PHONE1
exten => ${PABX_PHONE1}, 1,Dial(SIP/${PABX_PHONE1},30,rH)
same => n,Playtones(busy)
same => n,Busy(3)

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

;------- S-Trunking/TIER III Trunking (PABX Phone -> Radio) -------
;IndividualCall
exten => _11.,1,NoOp("Extension "${EXTEN})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,SIPAddHeader(RA-Header-Calltype: 0)
same => n,SIPAddHeader(RA-Header-DstDmrId: ${EXTEN:2})
same => n,SIPAddHeader(JKC-Header-SrcType: pabx)
same => n,Dial(Sip/${IPGW_SIPUSERID_TSC},20,hft)
same => n,Hangup

;Group Call
exten => _22.,1,NoOp("Extension "${EXTEN})
same => n,Answer()
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,SIPAddHeader(RA-Header-Calltype: 8)
same => n,SIPAddHeader(RA-Header-DstDmrId: ${EXTEN:2})
same => n,SIPAddHeader(JKC-Header-SrcType: pabx)
same => n,Dial(Sip/${IPGW_SIPUSERID_TSC},20,hft)
same => n,Hangup

;****(to the next page….)
```

```
;****(from the previous page….)
;------- S-Trunking/TIER III Trunking (Radio -> PSTN Phone) -------
exten => _xxx.,1,Dial(SIP/${EXTEN}@xxxxxx,20,Hft)
same => n,Playtones(busy)
same => n,Busy(6)

exten => _yyy.,1,Dial(SIP/${EXTEN}@xxxxxx,20,Hft)
same => n,Playtones(busy)
same => n,Busy(6)

exten => _zzz.,1,Dial(SIP/${EXTEN}@xxxxxx,20,Hft)
same => n,Playtones(busy)
same => n,Busy(6)

;------- S-Trunking/TIER III Trunking (PSTN Phone -> Radio) -------
[incoming]
exten => ${PSTN_SIP_ACCOUNT},1,Goto(Asterisk-ivr,s,1)

[Asterisk-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)

;IVR Change Connection
;Input 1 (Soft Phone)
exten => 1,1,Goto(IVR-SoftPhone,1,1)
;Input 2 (Soft Phone)
exten => 2,1,Goto(IVR-DMR-Individual,2,1)
;Input 3 (DMRID Group)
exten => 3,1,Goto(IVR-DMR-Group,3,1)

[IVR-SoftPhone]
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 => _XXX,1,dial(SIP/${EXTEN})

;****(to the next page….)
```

```
;****(from the previous page….)

[IVR-DMR-Individual]
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 => _XXXXXXXX,1,NoOp("Extension "${EXTEN})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,SIPAddHeader(RA-Header-Calltype: 0)
same => n,SIPAddHeader(RA-Header-DstDmrId: ${EXTEN})
same => n,SIPAddHeader(JKC-Header-SrcType: pstn)
same => n,Dial(Sip/${IPGW_SIPUSERID_TSC},20,hft)
same => n,Hangup

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

exten => _XXXXXXXX,1,NoOp("Extension "${EXTEN})
same => n,Answer(500)
same => n,Wait(0.1)
same => n,Playback(ascending-2tone)
same => n,SIPAddHeader(RA-Header-Calltype: 8)
same => n,SIPAddHeader(RA-Header-DstDmrId: ${EXTEN})
same => n,SIPAddHeader(JKC-Header-SrcType: pstn)
same => n,Dial(Sip/${IPGW_SIPUSERID_TSC},20,hft)
same => n,Hangup

;****(End of file)
```

Note: Below table are shown the numbering rule to call that each device call using this Asterisk Server configuration.

	Individual Call	Group Call	Note
PABX Phone-> Radio	Prefix (11) +UID No.	Prefix (22) +GID No.	This is defined "extensions.conf" on the above.

	PABX Telephone Call	Note
[TIER II Conventional] Radio -> PABX Phone	Prefix (2) +Phone No.	PABX Prefix number is configured on IP Gateway.
[DMR Based Trunking] Radio -> PABX Phone	Prefix (02) +Phone No.	PABX Prefix number is static number on these systems.

Step8: Reload modules (sip, extensions)/Restart Asterisk service

- Start up the "Asterisk CLI (Asterisk Command Line Interface)" as "root" user.

```
asterisk␣-r ↲
```

```
[root@freepbx asterisk]#
[root@freepbx asterisk]# asterisk -r
Asterisk 16.4.1, Copyright (C) 1999 - 2018, Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=========================================================================
Connected to Asterisk 16.4.1 currently running on freepbx (pid = 1614)
freepbx*CLI>
```

- Reload configured file of "sip.conf".

```
sip␣reload ↲
```

```
freepbx*CLI>
freepbx*CLI>
freepbx*CLI> sip reload
freepbx*CLI>
```

- Reload configured file of "extensions.conf"

```
dialplan␣reload ↲
```

```
freepbx*CLI>
freepbx*CLI>
freepbx*CLI> dialplan reload
Dialplan reloaded.
freepbx*CLI>
```

- Quit Asterisk CLI

```
quit ↲
```

```
freepbx*CLI> quit
Asterisk cleanly ending (0).
Executing last minute cleanups
[root@freepbx asterisk]# _
```

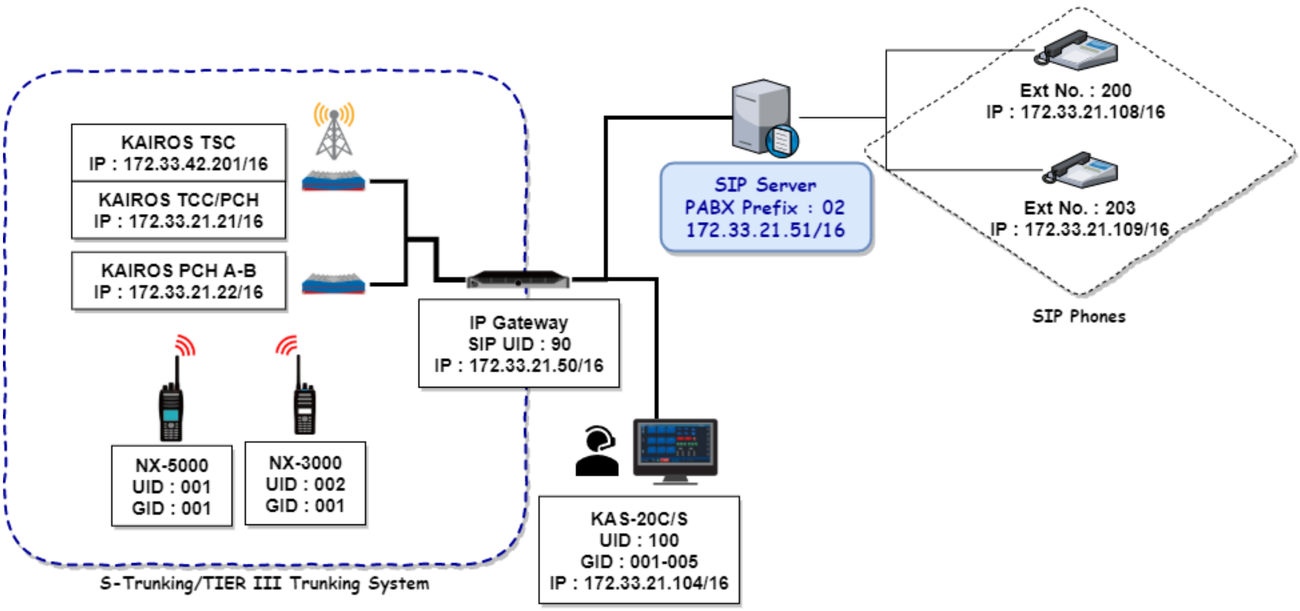
- Restart Asterisk service

```
systemctl␣restart␣asterisk.service ↲
```

```
[root@freepbx asterisk]#
[root@freepbx asterisk]#
[root@freepbx asterisk]# systemctl restart asterisk.service_
```

6.2. Sample Configuration

6.2.1. KAIROS DMR Based Trunking System with PABX Phone System



Step1: Construction of KAIROS DMR base Trunking System
This sample is constructed "Single Site Trunking with IP Gateway". Refer to "AN-19-0013", if you need more information for construct the DMR Based Trunking System with IP Gateway.

Step2: Construction of SIP Asterisk server
Refer to "6.1 Construction of SIP Asterisk server". It is configured for the System that is described this Appendix. And it must use "extensions.conf" file that is for "DMR Based Trunking System".

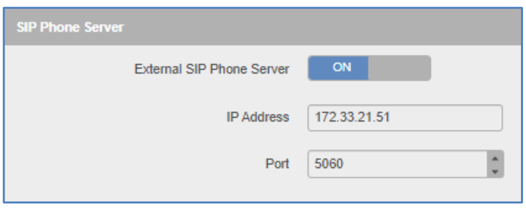
Step3: Configuration of IP Gateway

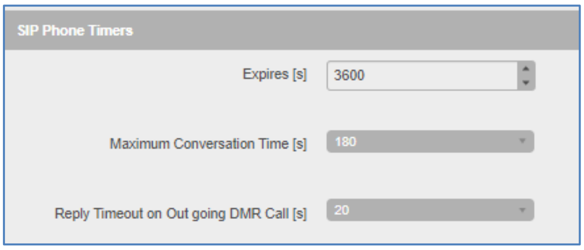
Refer to “ 2.1 KPG-1013GW IP Gateway ”. This sample is configured below parameters.

- TSC IP Address: 172.33.42.201 (Port: 23132)

No.	TSC ID	Valid	TSC Name	TSC IP Address	Port	Number Of Channel
1	1	ON	KENWOOD	172.33.42.201	23132	2

- External SIP Phone Server IP Address: 172.33.21.51 (Port: 5060)





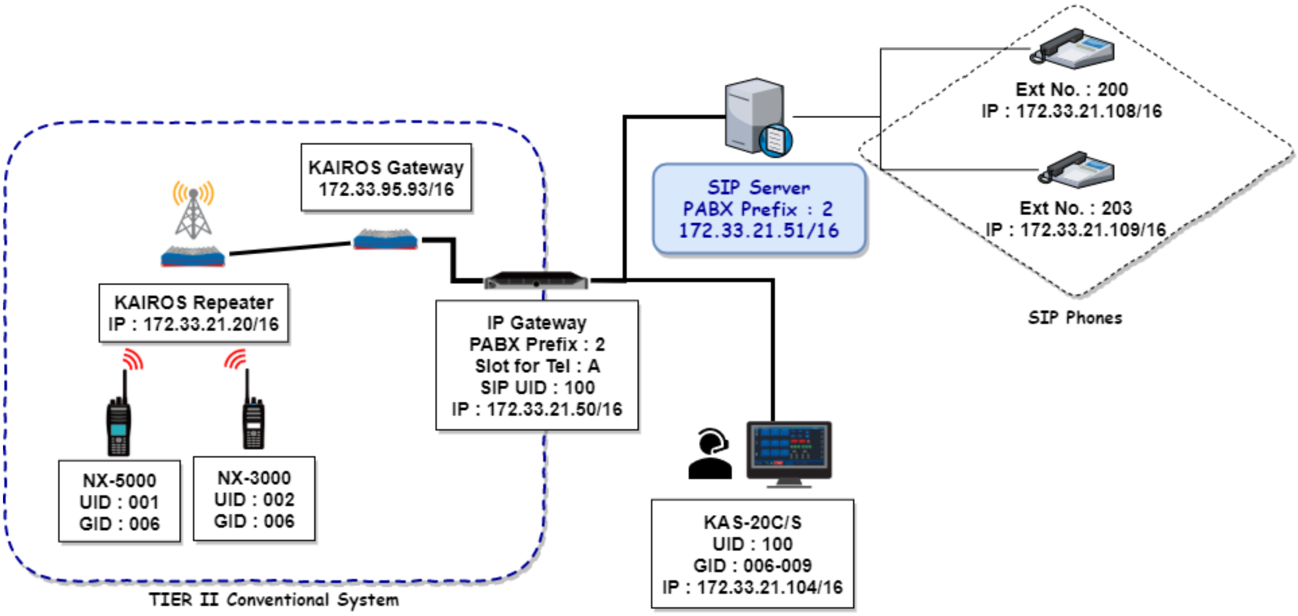
- SIP User ID: 90

No.	Valid	SIP User ID	SIP User ID Name	Comment	Home TSC ID
1	ON	90	TSC	KAIROS TSC SIP ID	1

Step4: Configuration of KAIROS NMS

Refer to “ 2.2.1 KA-NMS ”. Configuration is the same value for this Sample System.

6.2.2. KAIROS TIRE II Conventional with PABX Phone System



Step1: Construction of TIER II Conventional System with KAIROS Gateway

This sample is constructed "TIER II Conventional with IP Gateway". If you need more information for construct TIER II Conventional System with IP Gateway, refer to "AN-19-0013" in addition. In addition, refer to " 2.3 KAIROS DMR Repeater & Gateway (DMR TIER II Conventional) ". This chapter shows important confirmation points about configured for the KAIROS Gateway.

Step2: Construction of SIP Asterisk server

Refer to "6.1 Construction of SIP Asterisk server". It is configured for the System that is described this Appendix. And it must use "extensions.conf" file that is for "TIER II Conventional System".

Step3: Configuration of IP Gateway

Refer to " 2.1 KPG-1013GW IP Gateway ". This sample is configured below parameters.

- KAIROS Gateway IP Address: 172.33.95.93
- External SIP Phone Server IP Address: 172.33.21.51 (Port: 5060)
- SIP User ID: 100
- PABX Prefix: "2"
- For the telephone call Slot: "A"

No.	Valid	SIP User ID	SIP User ID Name	Comment	PSTN Prefix	PABX Prefix	Home Gateway ID	Slot
1	ON	100	IP Gateway SIP ID for TIER2	ID of IP_GW on SIP Server	1	2	1	A