KENWOOD

# Application Note

SIP Features for KAIROS

KAIROS Series V1.6.5.0

KAIROS Manager V1.7.1

| | |
|---|---|
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JVCKENWOOD

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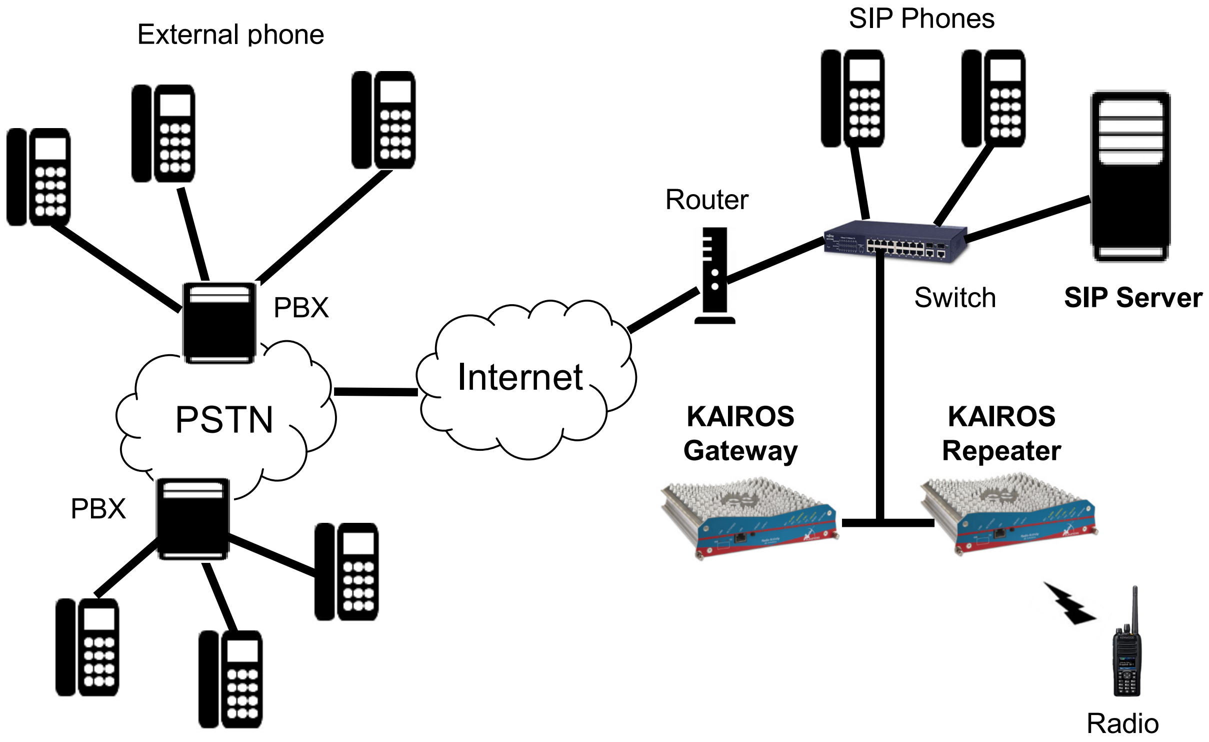
# 1 Outline

This document describes about SIP connection with Radio Activity's KAIROS. It guides to construct a voice call system with SIP phone/External phone and Radio through KAIROS Gateway and SIP server using Asterisk, one of SIP server software, as reference.

## 1.1 Regarding KAIROS's SIP feature




KAIROS Gateway is required to use SIP feature in a KAIROS system. It manages voice call connection between a SIP server and a radio system. Connecting PSTN via Internet, it is possible to connect with an external phone, too.

## 1.2 Necessary Requirement for SIP Feature

Following devices and software are required to use KAIROS SIP feature.

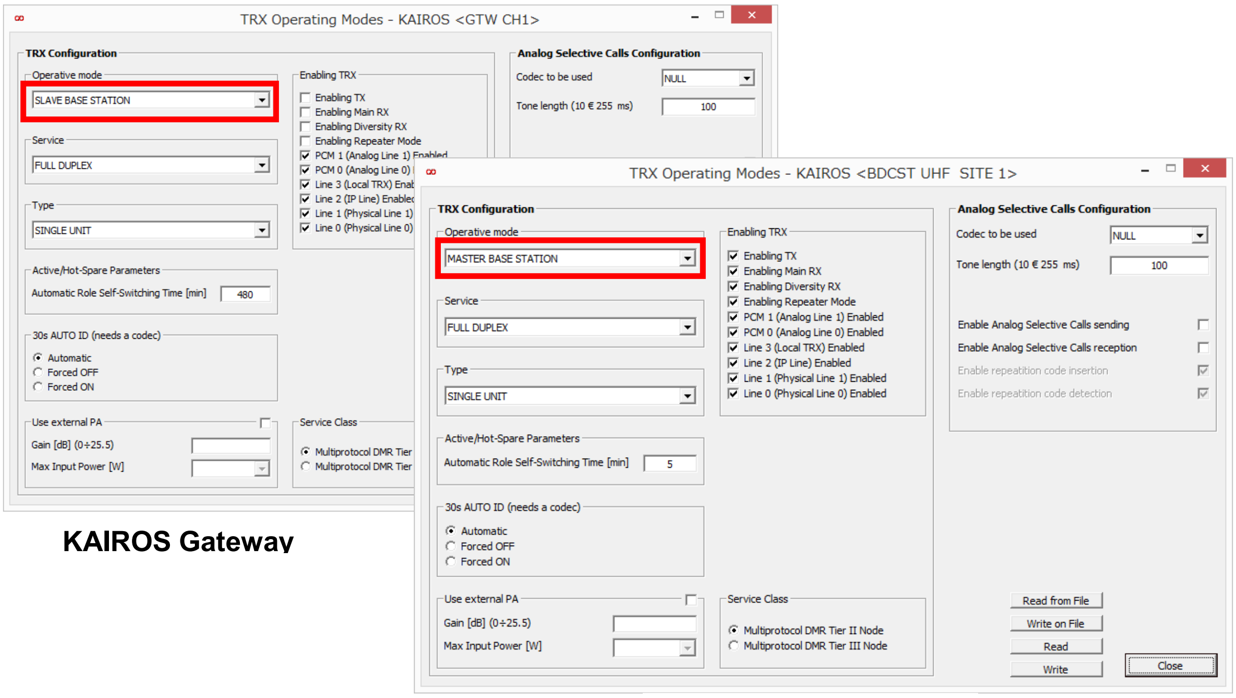
| Production Name | Description | Remarks |
|---|---|---|
| KA-TI-02 | KAIROS Gateway | Gateway to connect with a device like Dispatcher.<br>AMBE codec option board is required for SIP Gateway. |
| KA-080/160/350<br>/450/500/900 | Repeater | |
| KAIROS Manager | Configuration Software | Software in order to configure KAIROS Series |

KAIROS Gateway and KAIROS Repeater need registration and synchronization each other. Follow below steps on KAIROS Manager to configure.

Configure Master to KAIROS Repeater and Slave to KAIROS Gateway for both registration and synchronization.

## ●Registration setting

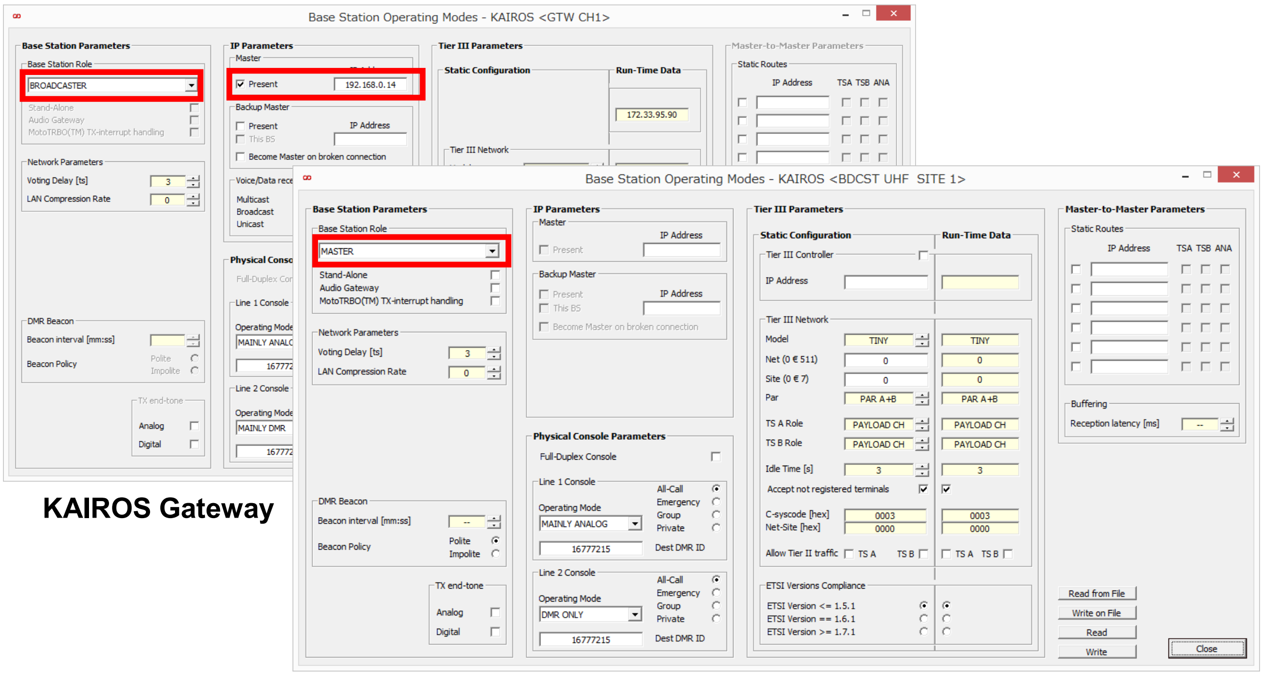
Open the TRX Operating Modes Window from “KAIROS” -> “Configuration” on KAIROS Manager, and configure as below.




KAIROS Gateway

KAIROS Repeater

Open the Base Station Operating Modes Window, and configure Master and Slave.




KAIROS Gateway

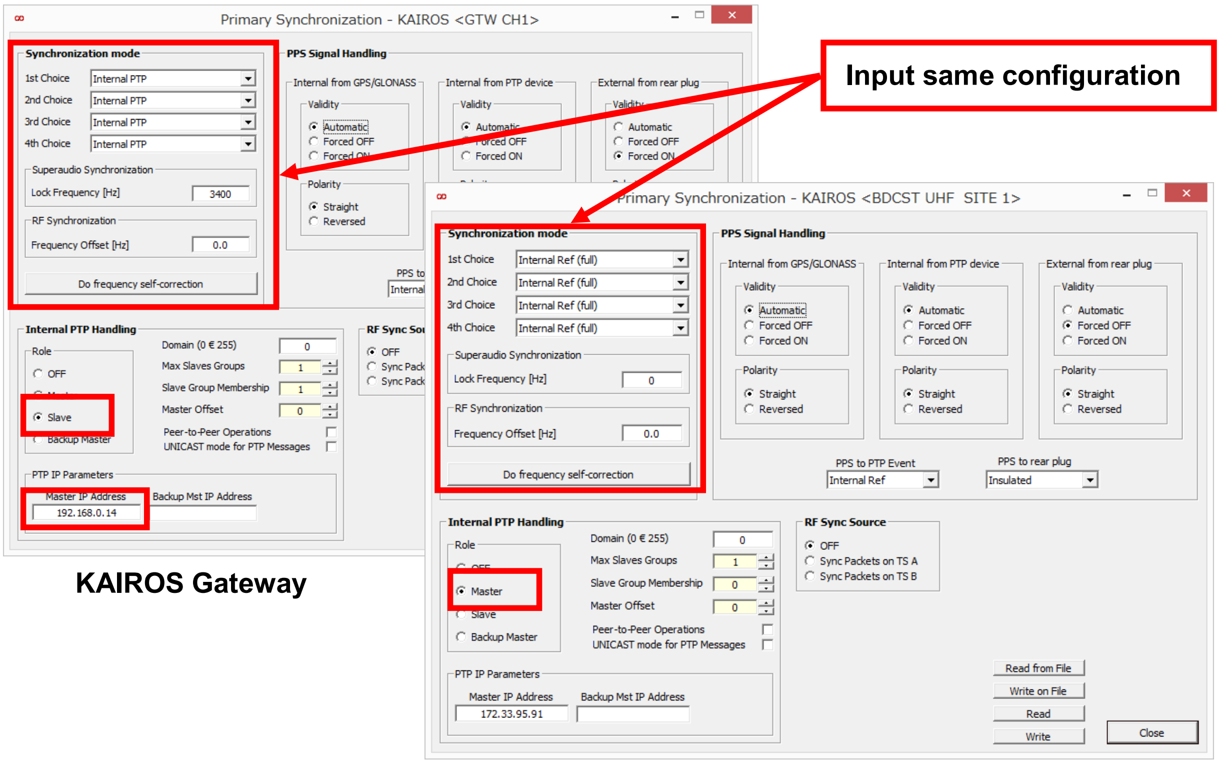
KAIROS Repeater

Enter IP address of KAIROS Repeater as Master at the field of IP Parameters for KAIROS Gateway.

## ●Synchronization setting

Open the Primary Synchronization Window for synchronization selection.

There is several synchronization selection. In this time, use PTP(Precision Time Protocol), then select ”Internal Ref(full)” for KAIROS Repeater and ”Internal PTP” for KAIROS Gateway.




**KAIROS Gateway**

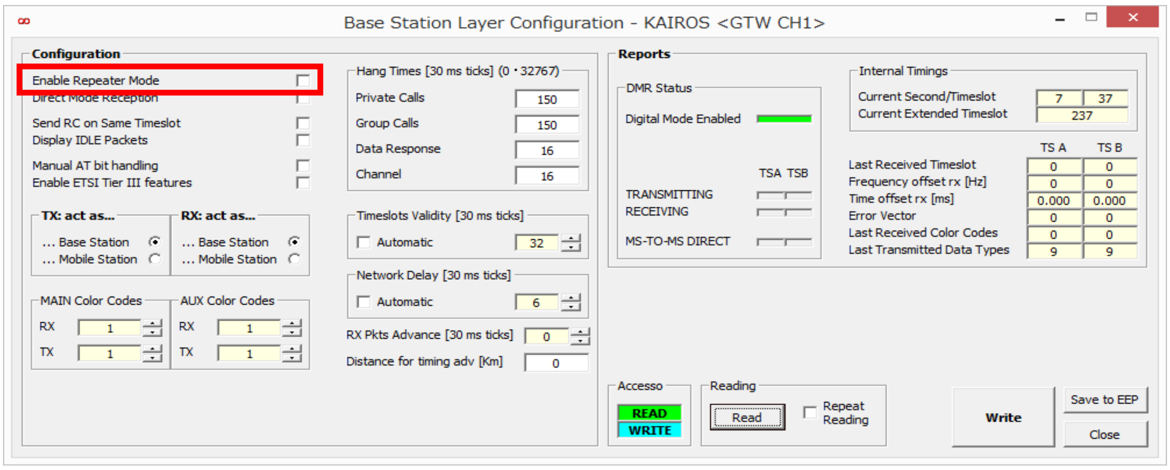
**KAIROS Repeater**

**Note:**

It has to select Slave from Internal PTP handling configuration and enter IP address for KAIROS Gateway. For details of further configuration, refer to ENB52-KAIROS User Manual and ENB56-KAIROS OPERATIONAL MANUAL, PTP Usage and Applications documents provided by Radio Activity.

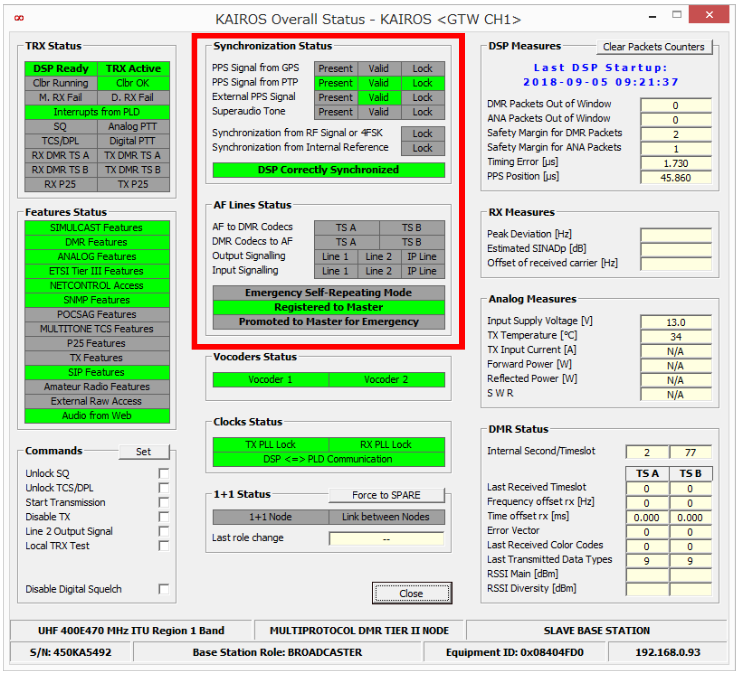
## ●Other setting

The Enable Repeater Mode checkbox should be unchecked on the Base Station Layer Configuration Window for KAIROS Gateway.




## ●Confirmation

On KAIROS Manager connecting to KAIROS Gateway, open the KAIROS Overall Status window from “KAIROS”->”Controls”.



If all configurations in red square is correct and shows green, it means synchronization is working well. That is all for registration and synchronization of KAIROS Gateway and KAIROS Repeater.

**Note:**

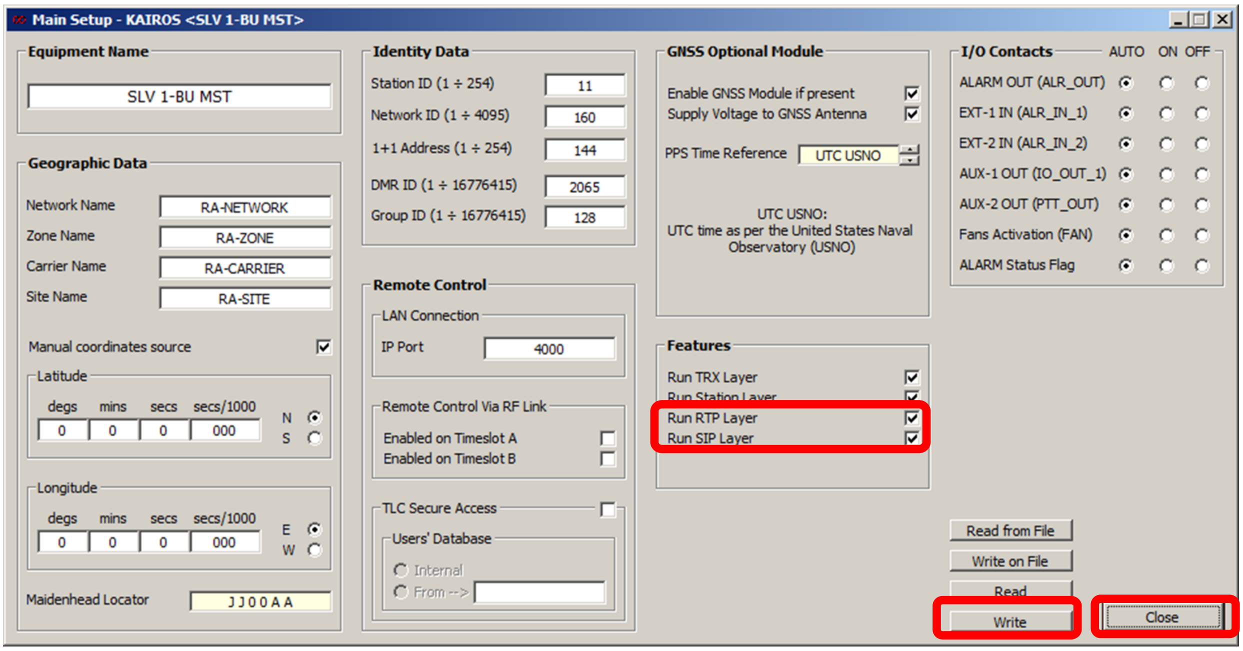
In advance, it should be completed configuration to be able to communicate with radios through KAIROS repeater.

# 2 KAIROS Manager Setting

This chapter describes KAIROS Gateway configuration to work with SIP Gateway.

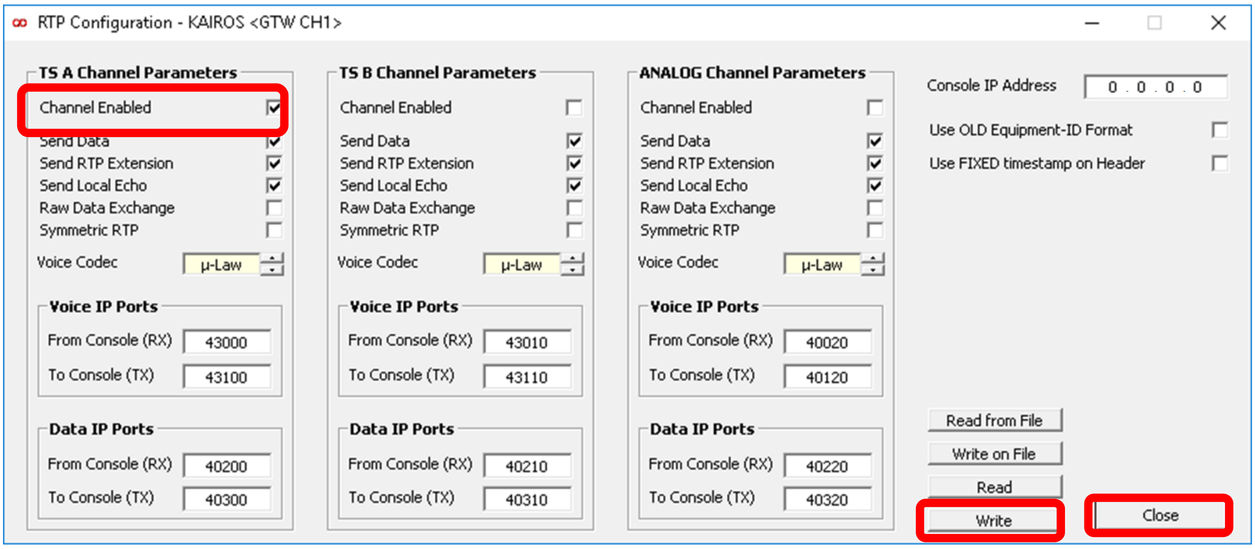
### 2.1.1 Main Setup Window

Open the Main Setup window on KAIROS Manager.



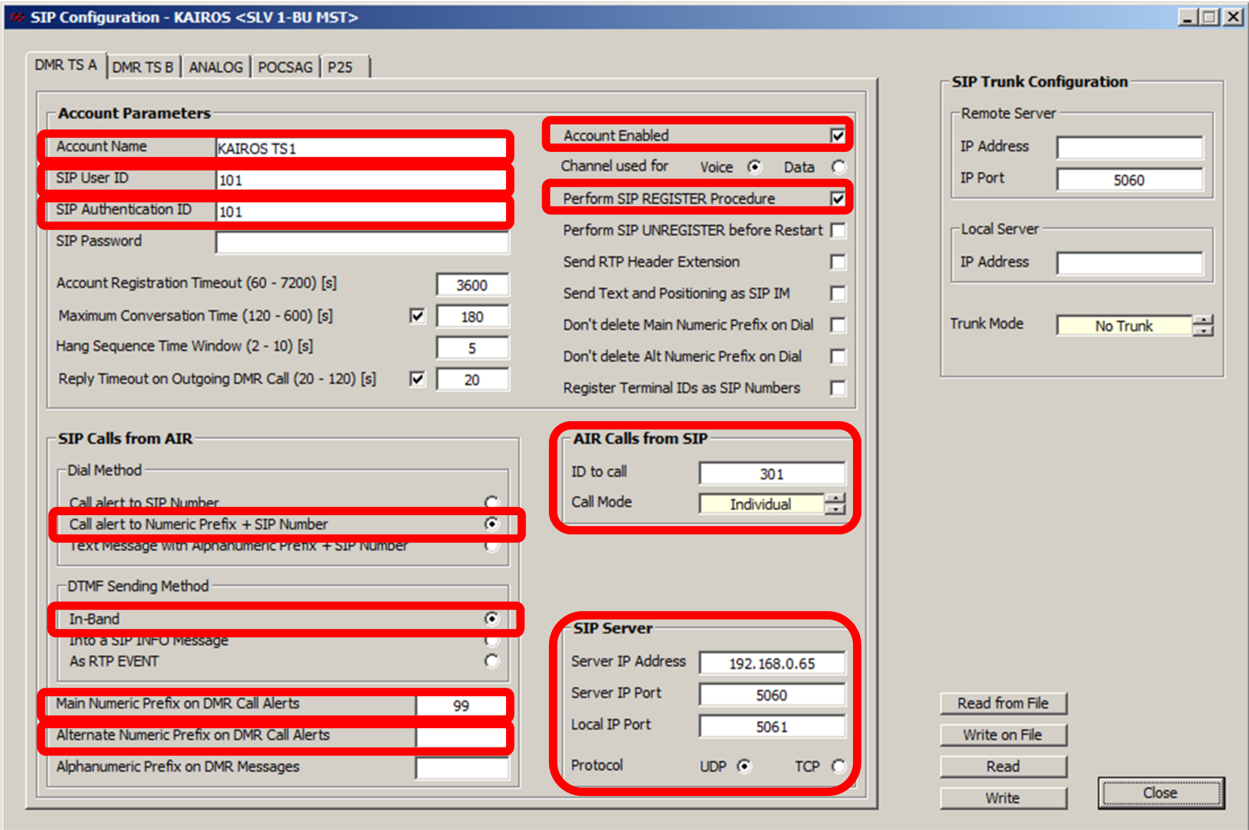

SIP Gateway and RTP Gateway which manage RTP voice packet are required for SIP feature. Enable both checkboxes of Run RTP Layer and Run SIP Layer, then start writing and close the window.

### 2.1.2 RTP Configuration Window




It is able to configure to each Time Slot (TS), A and B on the window. In this case, RTP Gateway is actives to TS A. Enable the checkbox for Channel Enabled, then start writing and close the window.

### 2.1.3 SIP Configuration Window




This is SIP Configuration window. The underlined items are mandatory.

- <u>"Account Name"</u>
  Account name for the TS

- <u>"SIP User ID"/"SIP Authentication ID"</u>
  Extension number assigned to each TS. Both fields should be same number.
  In above Example, TS A extension number is "101". Dialing extension 101 from a SIP phone is able to call on Time Slot A.

- "SIP Password"
  Password for registration to SIP server, but it is not available now.

- "Maximum Conversation Time"
  Maximum time to call.

- "Hang Sequence Time Window"
  The time to start Hang Sequence.

- <u>Account Enabled</u>
  Select whether use SIP Gateway. Check this box.
  **<u>Note:</u>** In advance, enter Server IP Address in SIP Sever to active.

- <u>Perform SIP REGISTER Procedure</u>
  Select whether apply SIP register procedure. Check this box.

**SIP Calls from AIR**

- Dial Method

  Dialing to call from a radio to a SIP phone. Select “Call Alert No Numeric Prefix + SIP Number”.

- DTMF Sending Method

  Exchange DTMF tone between a radio and a SIP phone. Select In-Band.

・AIR Calls from SIP

Select from Individual or Group for the call which a SIP Gateway calls with extension number assigned to the TS. The example is ID 301 Individual Call

・SIP Server

Enter SIP server IP address and Port number, and select Protocol.
As default setting, configure as below. (Each TS has default Local IP Port.)

| | | |
|---|---|---|
| Server IP Port | | 5060 |
| Local IP Port | TS A | 5061 |
| | TS B | 5062 |
| Protocol | | UDP |

・Main Numeric Prefix on DMR Call Alerts

If a radio calls another extension number, Add Prefix to the extension number.
For example, when a radio calls extension number 200, dial UID 99200. In this case, dialing UID has to be maximum 6 digits including Prefix.

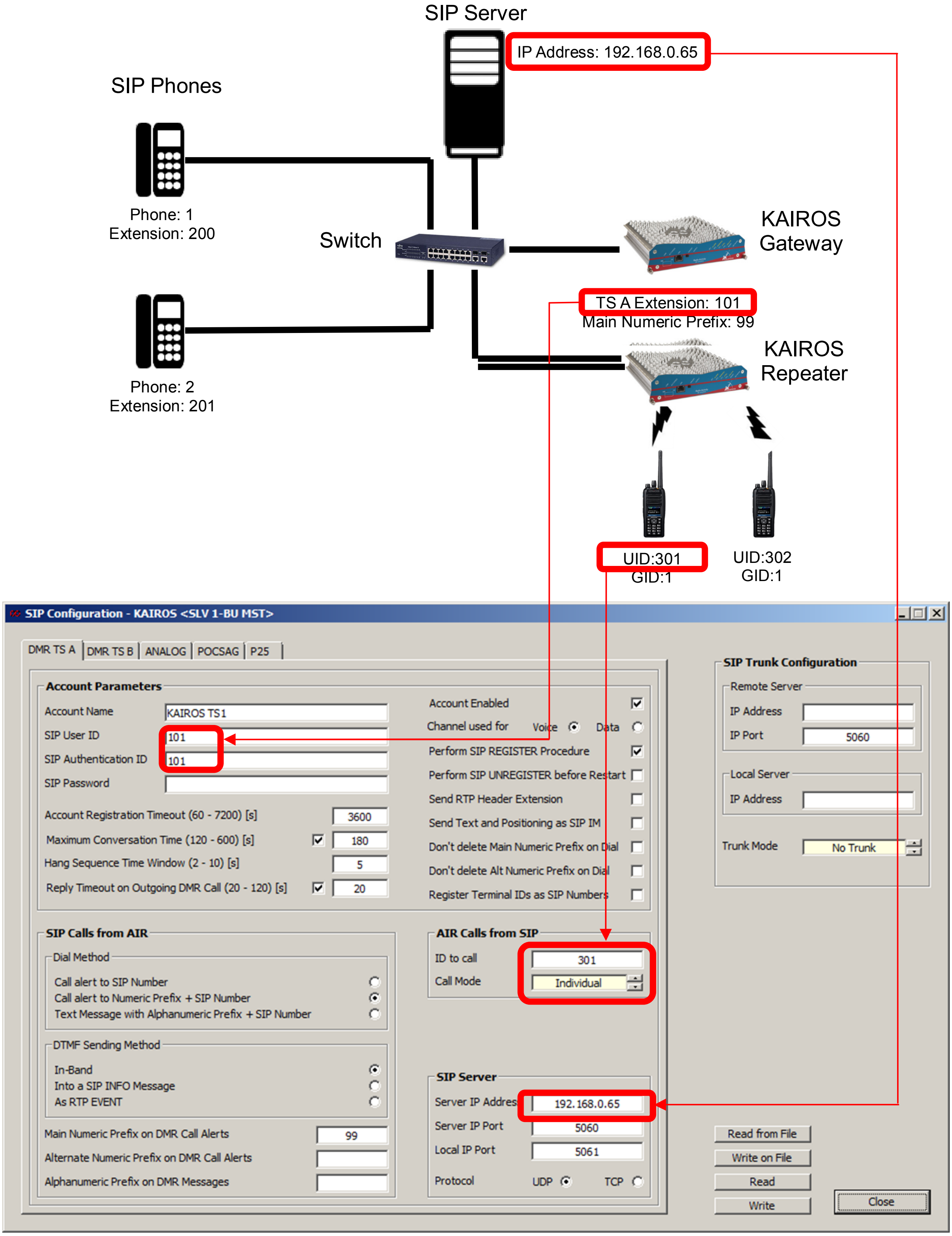
・Alternate Numeric Prefix on DMR Call Alerts

It is able to configure different prefix with Main Numeric Prefix on DMR Call Alerts.

Configuration of SIP Gateway has to be same with connected SIP server. For details, refer 4.Appendix A.

# 3 Sample Configuration

This chapter describes the sample of configuration for each parameter.
Refer each parameter of following system and the parameter of SIP Gateway in KAIROS manager.

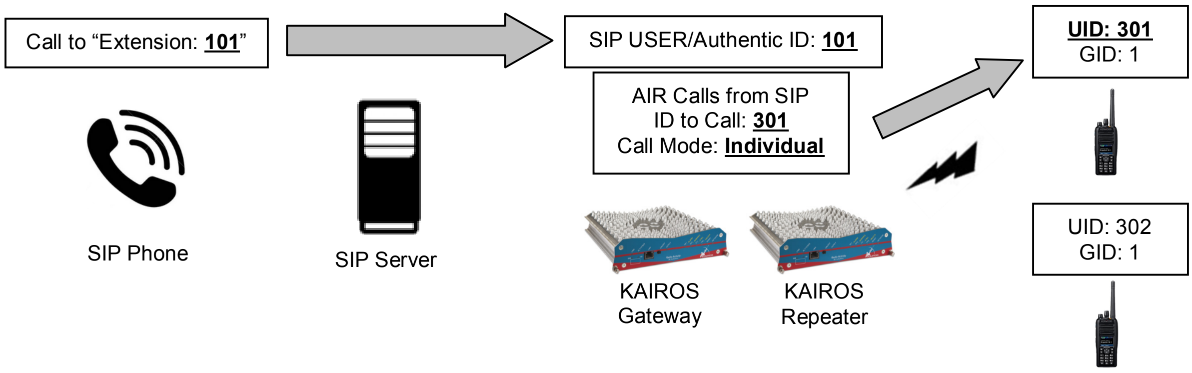


●Call from a SIP phone to a Radio

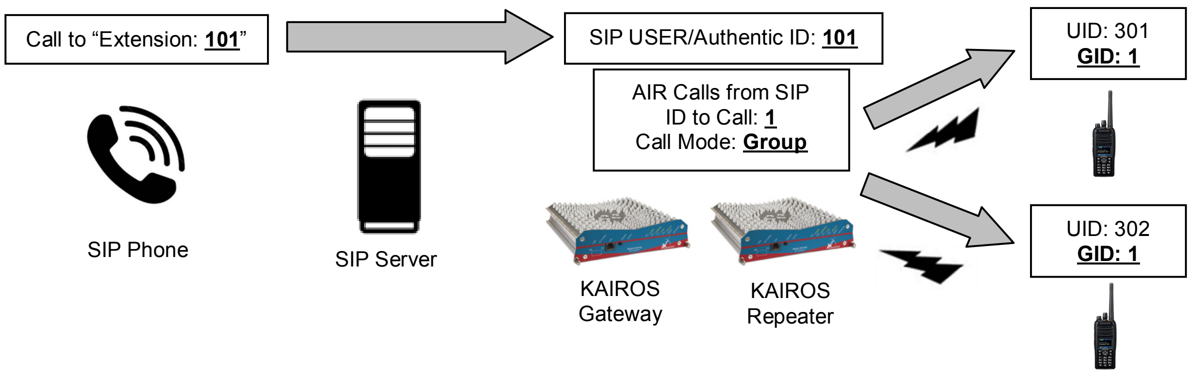
Configure ” AIR Calls from SIP” in the SIP Configuration Window of KAIROS Manager.

Case 1:

When a SIP phone calls to TS A Extension: 101, a radio (UID: 301) emits Call Alert tone. Pressing Radio PTT button to start a voice call.



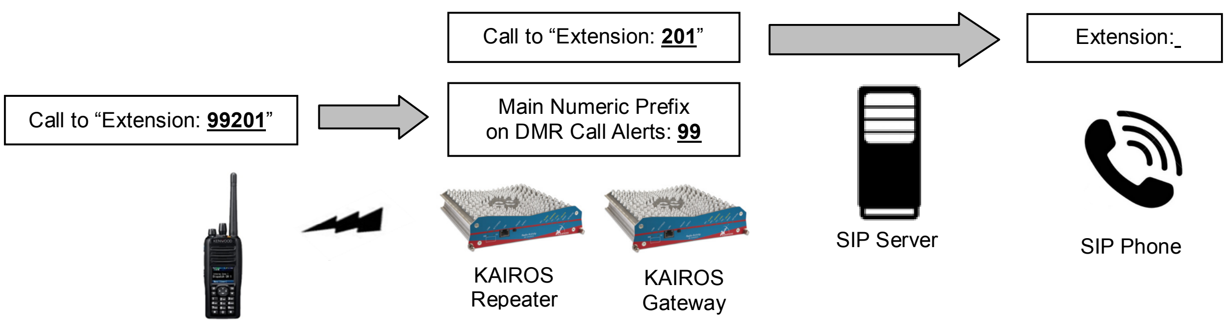

Case 2: In case of Group call




In this case, all radios emits Call Alert tone. When one of the radios presses the PTT button, stop alerting and start a voice call.

●Call from a Radio to a SIP phone

For example, to call a SIP phone assigned Extension: 201, a radio uses Paging Call.




To start Paging Call, prefix 99 should be added to UID number. In the above case, UID is Prefix”99”+Extension”201”=”99201”. SIP Gateway received “99201” as UID recognizes Extension: 201 by prefix and calls Extension: 201. By picking up the phone handset of the phone, the call from Radio to SIP phone is established.

**Reference:**

・As described above, KAIROS Manager is able to assign only one call destination to each TS.
And also, it is possible to specify multiple call destinations by using Radio Activity SIP Proprietary Headers. Edit Radio Activity proprietary header information to the extensions.conf which is a configuration file in Asterisk SIP server . For detail, refer 4.4.2.extensions.conf.

●Terminate a call

・by SIP phone
Hanging up the phone handset to terminate the call

・by Radio
Repeat PTT ON and OFF three times during Hang Sequence timer configured on SIP Configuration window to terminate the call. If Auto Reset Timer is configured, it is not able to re-call during the time. Therefore, shorter Auto Reset Timer could be recommended.

# 4 Appendix A

This chapter describes an example of SIP server, Asterisk setup procedure and installation of necessary software as we investigated and explains Extension and External phone setting, too.

## 4.1 What is Asterisk?

Asterisk is open source IP-PBX software developed by digium in USA. It is distributed as GPL2 and ported in Linux, FreeBSD, Solaris, MacOS, and Windows.

Asterisk has several functions related to telephone not only extensions but also voice recording and voice mail, voice conference and connecting PSTN, etc...

## 4.2 Install Asterisk

CentOS7, one of Linux distribution was used to investigate setup procedure in this time. Asterisk does not need a high spec PC to install, but we used 16GB hard disk in order to use database software.

Following example shows installation of OS, network configuration and so on, you may skip if not needed. Remind you that the required software package may depends on Linux OS version or Base environment, etc.

### 4.2.1. OS installation

Get "CentOS-7-x86_64-DVD-1804.iso" from the link and refer installed PC condition.

https://www.centos.org/download/

Base environment: Server (use GUI)

Add-on: KDE

### 4.2.2. Network configuration

It needs internet connect to start following procedure.

It must configure NIC (Network Interface Card) and proxy settings of "yum" and "wget".

**Administrator login level is required to proceed setting.**

Open console window and enter command as below.

・yum

```
vi /etc/ yum.config
```

As next, edit Proxy URL your internet connection.

proxy = http://proxy.xxxxxxx

・wget

```
vi /etc/wgetrc
```

As next, edit Proxy URL your internet connection.
http_proxy= http://proxy.xxxxxxx
https_proxy= https://proxy.xxxxxxx

### 4.2.3. Install necessary software for compilation

```
yum -y install zip unzip bzip2 gcc gcc-c++ make wget subversion libxml2-devel ncurses-devel openssl-devel vim-enhanced kernel-devel
```

### 4.2.4. Add EPEL and REMI as yum repository

```
yum install epel-release
```

```
wget http://rpms.famillecollet.com/enterprise/remi-release-7.rpm
```

```
rpm -Uvh remi-release-7.rpm
```

```
yum update
```

It takes about 25 minutes for “yum update”.

### 4.2.5. Stop security function

Stop security function temporarily to avoid interfering during installation. After completing all installation, it should be returned before starting operation if you need.

・Stop SELinux

```
vi /etc/selinux/config
```

As next, disable SELINUX like following.
SELINUX=disabled

・Stop Firewall

```
systemctl stop firewalld
```

```
systemctl disable firewalld
```

### 4.2.6. Reboot

Enter reboot command to reflect configuration.

```
reboot
```

After rebooting, keep login with administrator level for further configuration.

### 4.2.7. Download and unzip Asterisk

After downloading Asterisk via internet, unzip compressed file.

```
cd /usr/src
```

```
wget http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-15-current.tar.gz
```

```
tar zxvf asterisk-15*.tar.gz
```

### 4.2.8. Install necessary library for Asterisk scripts

```
cd /usr/src/asterisk-15*/contrib/scripts
```

```
./install_prereq install
```

### 4.2.9. Install library which is not available repository

Download and install the library which is not available repository.

```
yum install -y https://dl.fedoraproject.org/pub/epel/6/x86_64/Packages/l/libresample-0.1.3-12.el6.x86_64.rpm
https://dl.fedoraproject.org/pub/epel/6/x86_64/Packages/l/libresample-devel-0.1.3-12.el6.x86_64.rpm
```

### 4.2.10. Install by using Asterisk script

```
./install_prereq install-unpackaged
```

In case an error occur and stop installation during procedure, subversion proxy setting might be inactive.

```
vi ~/.subversion/servers
```

After the command, check proxy settings as below.

[global]
http-proxy-host = proxy.xxxxx (remove "http://")
http-proxy-port = xxxx (Port number of proxy server)

### 4.2.11. Install DAHDI

DAHDI is Digium Asterisk Hardware Device Interface to manage connection between kernel module and external phone line.

External connecting card is not used this time, but we install DAHDI in order to install necessary library for Asterisk.

The latest version 2.11.1 of DAHDI is not installed correctly, we install the previous version.

```
cd /usr/src/
```

```
wget https://downloads.asterisk.org/pub/telephony/dahdi-linux-complete/dahdi-linux-complete-2.10.2+2.10.2.tar.gz
```

```
tar zxvf dahdi-linux-complete-2.10.2+2.10.2.tar.gz
```

```
cd dahdi-linux-complete-2.10.2+2.10.2
```

```
make && make install && make config
```

### 4.2.12. Add Dummy Setting in DAHDI Modules File

```
vi /etc/dahdi/modules
```

After executing above command, add following two lines at the end of line.

# DAHDI Dummy
dahdi_dummy

### 4.2.13. Confirm whether DAHDI can be executed

```
service dahdi start
```

Edit the command to start DAHDI and shows following.

```
Starting dahdi (via systemctl):                    [  OK  ]
```

Next, edit command to stop DAHDI.

```
service dahdi stop
```

### 4.2.14. Install Asterisk

Change directory

```
cd
```

```
cd /usr/src/asterisk-15*/
```

Start scripts

```
./bootstrap.sh
```

```
./configure
```

Edit commands to build and install

```
make && make install && make samples && make config && make install-logrotate
```

```
ldconfig
```

### 4.2.15. Start Asterisk and confirm whether Asterisk console is available

```
service asterisk start
```

```
asterisk -vvvvvr
```

If success, following window will be shown.

```
[root@localhost admin]# asterisk -vvvcr
Asterisk 15.5.0, Copyright (C) 1999 - 2016, 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 15.5.0 currently running on localhost (pid = 1703)
  == Using SIP RTP CoS mark 5
    -- Executing [202@default:1] Dial("SIP/201-00000000", "SIP/202,30,r") in new stack
  == Using SIP RTP CoS mark 5
    -- Called SIP/202
    -- SIP/202-00000001 is ringing
  == Spawn extension (default, 202, 1) exited non-zero on 'SIP/201-00000000'
localhost*CLI> quit
Asterisk cleanly ending (0).
Executing last minute cleanups
[root@localhost admin]# vi /etc/asterisk/extensions.conf
[root@localhost admin]# vi /etc/asterisk/extensions.conf
[root@localhost admin]# service dahdi start
Starting dahdi (via systemctl):                            [  OK  ]
[root@localhost admin]#
[root@localhost admin]# service dahdi stop
Stopping dahdi (via systemctl):                            [  OK  ]
[root@localhost admin]# asterisk -vvvvvr
Asterisk 15.5.0, Copyright (C) 1999 - 2016, 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 15.5.0 currently running on localhost (pid = 1703)
localhost*CLI>
```

Enter command to exit Asterisk console.

```
exit
```

### 4.2.16. Auto Execution Setting

Enter following command in order to execute automatically from next time.

```
chkconfig asterisk on
```

Complete Asterisk installation for SIP server, and continue Database installation in Section 4.4.

## 4.3 Install Maria DB

Asterisk uses Data Base to manage voice communication. This chapter describes installation and initial settings of MariaDB which is one of Data Base software.

**Please configure as administrator from this.**

### 4.3.1. Uninstall Old Version MariaDB

MariaDB is available to install by using yum, but it makes old version be installed. So after deleting old version, re-install.

The delete command is following.

```
yum remove mariadb mariadb-libs
```

### 4.3.2. Add yum Repository

Register other repository in order to install new MariaDB. Following example shows that the repository (MariaDB10.1) for CentOS7 is added.

```
vi /etc/yum.repos.d/MariaDB.repo
```

After enter the command, enter below contents.

```
[mariadb]
name = MariaDB
baseurl = http://yum.mariadb.org/10.1/centos7-amd64
gpgkey=https://yum.mariadb.org/RPM-GPG-KEY-MariaDB
gpgcheck=1
```

### 4.3.3. Install New Version MariaDB

Enter following command to install. As option ” '--enablerepo=mariadb”, the repository which is set at 4.3.2 comes to be available.

```
yum -y install --enablerepo=mariadb MariaDB-common MariaDB-devel MariaDB-shared MariaDB-compat MariaDB-server
MariaDB-client
```

### 4.3.4. Initial Setting

Here, configure some settings by making a copy the minimum configuration sample of MariaDB.

```
cp -p /usr/share/mysql/my-small.cnf /etc/my.cnf.d/server.cnf
```

Enter “y” to make a copy.

```
cp: Overwite `/etc/my.cnf.d/server.cnf'?
```

Modify each setting.

```
vi /etc/my.cnf.d/server.cnf
```

Enter the command and add each lines at the part of” #### added” in following example.

This example shows that the destination of saving Data Base is ”datadir=/var/lib/mysql”.

```
[client]
port = 3306
socket = /var/lib/mysql/mysql.sock
default-character-set = utf8            #### added

[mysqld]
port = 3306
socket = /var/lib/mysql/mysql.sock
skip-external-locking
key_buffer_size = 16K
max_allowed_packet = 1M
table_open_cache = 4
sort_buffer_size = 64K
read_buffer_size = 256K
read_rnd_buffer_size = 256K
net_buffer_length = 2K
thread_stack = 240K
datadir=/var/lib/mysql                  #### added
character-set-server = utf8             #### added

server-id = 1

[mysqldump]
```

### 4.3.5. Activate and Start Service

Enter following command to automatically start when OS is started next time.

```
systemctl enable mariadb.service
```

```
systemctl start mariadb.service
```

### 4.3.6. Checking

Confirm whether MariaDB works correctly. Enter following command.

```
mysql -uroot -p
```

At initial setting, there is no password. Click enter with blank.

Next, following is shown if correct.

```
[root@localhost admin]# mysql -uroot -p
Enter password:
Welcome to the MariaDB monitor.  Commands end with ; or \g.
Your MariaDB connection id is 2
Server version: 10.1.35-MariaDB MariaDB Server

Copyright (c) 2000, 2018, Oracle, MariaDB Corporation Ab and others.

Type 'help;' or '\h' for help. Type '\c' to clear the current input statement.

MariaDB [(none)]>
```

Continue configuring Data Base for Asterisk.

### 4.3.7. Configuration of Data Base for Asterisk

#### 4.3.7.1. Make Data Base

Make Data Base as the name "asteriskdb", and configure.

Enter following command.

```
CREATE DATABASE asteriskdb;
```

If correct, "Query OK, x row affected (x.xx sec)" is shown.

```
MariaDB [(none)]> CREATE DATABASE asteriskdb;
Query OK, 1 row affected (0.03 sec)
```

Next, enter following command in order.

```
GRANT ALL PRIVILEGES ON asteriskdb.* TO asteriskdb@localhost IDENTIFIED BY 'ASTERISK' WITH GRANT OPTION;
```

**ASTERISK** which is shown using bold font and underline shows the password of this Data Base.

Please execute following.

```
FLUSH PRIVILEGES;
```

```
USE asteriskdb;
```

If correct, following is shown.

```
MariaDB [(none)]> USE asteriskdb;
Database changed
MariaDB [asteriskdb]>
```

```
exit
```

Exit MariaDB.

#### 4.3.7.2. Make Table Data

Make a table data for Asteriskdb. Enter following command.

First, use a prepared table as SQL in Asterisk.

```
mysql -uroot -Dasteriskdb -p < /usr/src/asterisk-15.*/contrib/realtime/mysql/mysql_config.sql
```

```
mysql -uroot -Dasteriskdb -p -e "drop table alembic_version;"
```

```
mysql -uroot -Dasteriskdb -p < /usr/src/asterisk-15.*/contrib/realtime/mysql/mysql_cdr.sql
```

```
mysql -uroot -Dasteriskdb -p -e "drop table alembic_version;"
```

```
mysql -uroot -Dasteriskdb -p < /usr/src/asterisk-15.*/contrib/realtime/mysql/mysql_voicemail.sql
```

Next, make a table which is not prepared as SQL.
For CEL (Channel Event Logging) table data. Enter the command.

```
mysql -uroot -Dasteriskdb -p
```

Next, enter like followings. Complete entering till end of right edge and the enter key, ”->” will be shown. Continue entering the command.

```
CREATE TABLE IF NOT EXISTS cel (
  id int(11) NOT NULL auto_increment,
  eventtype varchar(30),
  eventtime datetime,
  cid_name varchar(80),
  cid_num varchar(80),
  cid_ani varchar(80),
  cid_rdnis varchar(80),
  cid_dnid varchar(80),
  exten varchar(80),
  context varchar(80),
  channame varchar(80),
  src varchar(80)
  dst varchar(80),
  channel varchar(80),
  dstchannel varchar(80),
  appname varchar(80),
  appdata varchar(80),
  amaflags int(11),
  accountcode varchar(20),
  uniqueid varchar(32),
  userfield varchar(255),
  linkedid varchar(32),
  peer varchar(80),
  userdeftype varchar(255),
  eventextra varchar(255),
  PRIMARY KEY(id),
  KEY uniqueid_index (uniqueid),
  KEY linkedid_index (linkedid)
);
```

Next,

```
exit
```

Exit to complete making table data for CEL.

Next, make a table data for Queue Log. Enter following command same as making a table data for CEL.

```
mysql -uroot -Dasteriskdb -p
CREATE TABLE queue_log (
  id bigint(255) unsigned NOT NULL AUTO_INCREMENT,
  time varchar(26),
  callid varchar(40),
  queuename varchar(20),
  agent varchar(20),
  event varchar(20),
  data varchar(100),
  data1 varchar(40),
  data2 varchar(40),
  data3 varchar(40),
  data4 varchar(40),
  data5 varchar(40),
  created timestamp NOT NULL DEFAULT CURRENT_TIMESTAMP,
  PRIMARY KEY (id),
  KEY queue (queuename),
  KEY event (event)
);
exit
```

Making all of table data is completed.

#### 4.3.7.3. Make View Data

Make “sigregs” view data to Asteriskdb data base.

```
mysql -uroot -Dasteriskdb -p
CREATE VIEW `sipregs` AS SELECT
   `id`
  ,`name`
  ,`ipaddr`
  ,`port`
  ,`regseconds`
  ,`defaultuser`
  ,`fullcontact`
  ,`regserver`
  ,`useragent`
  ,`lastms`
FROM `sippeers`;
exit
```

### 4.3.7.4. ODBC Setting

This section describes configuration of ODBC (Open Database Connectivity). ODBC is common interface for integrating access to each data base.

First, confirm whether MySQL is registered in /etc/odbcinst.ini.

```
vi /etc/odbcinst.ini
```

Next, confirm whether there is following description. If there is not, add description.

```
[MySQL]
Description=ODBC for MySQL
Driver=/usr/lib/libmyodbc5.so
Setup=/usr/lib/libodbcmyS.so
Driver64=/usr/lib64/libmyodbc5.so
Setup64=/usr/lib64/libodbcmyS.so
FileUsage=1

[MySQL ODBC 5.3 Unicode Driver]
Driver=/usr/lib64/libmyodbc5w.so
UsageCount=1

[MySQL ODBC 5.3 ANSI Driver]
Driver=/usr/lib64/libmyodbc5a.so
UsageCount=1
```

Next, add the configuration in order to connect to Asterisk in /etc/odbc.ini. If odbc.ini doesn't exist, make the file.

```
vi /etc/odbc.ini
```

Entering vi command, add below description.

```
[ASTERISK_DB]
Description = ASTERISK_DB
Driver = MySQL ODBC 5.3 Unicode Driver
SERVER = 127.0.0.1
UID = asteriskdb
PWD = Password
DATABASE = asteriskdb
Port = 3306
```

”**Password**” has been configured in section 4.3.7.1.

## 4.4 Configuration of Asterisk

This chapter describes an example of description for Asterisk's sip.conf and extensions.conf files.
SIP server connects to SIP devices when following settings are correct.

- Extensions should be registered.
- Action for registered extension should be defined when it receives.

Note that the unregistered extension doesn't work connection processing.

### 4.4.1 sip.conf

It locates at /etc/asterisk/sip.conf.
Register extensions number of SIP devices which SIP server manages in sip.conf file.
Following shows an example based on Sample system in Chapter3. Refer following URL for detail of Asterisk sip.conf.

https://www.voip-info.org/asterisk-config-sipconf/

```
[general]
context=default
allowguest=yes
allowoverlap=no
bindaddr=0.0.0.0
tcpenable=no
transport=udp
srvlookup=no
disallow=all
allow=ulaw
dtmfmode = info

[101]
type=friend
defaultuser=101
canreinvite=no
host=dynamic

[200]
type=friend
defaultuser=200
canreinvite=no
host=dynamic

[201]
type=friend
defaultuser=201
canreinvite=no
host=dynamic
```

### 4.4.2 extensions.conf

It locates at /etc/asterisk/extensions.conf.

This file has descriptions of action when registered extension is called by other extension, or registered extension calls to other extension (This is called “Dial plan”.).

Basic form is ”exten => extensions, priority, application”. Following is a sample of ”Dial” application which shows calling to other and connecting each channels.

exten => 201,1,Dial(SIP/201,30,r)

When extension:201 is called

Priority

Execute “Dial” application
SIP/201 ⇒ Specify Extension:201 SIP device
30 ⇒ Call time(second)
r ⇒ Make listen Ring Back Tone to calling side when calling

In case there is multiple applications, they will start sequentially referring priority.

In detail of Asterisk’s extensions.conf, please refer following URL.

https://www.voip-info.org/asterisk-config-extensionsconf/

#### 4.4.2.1. Basic Configuration

Following describes an example for Sample system in Chapter 3. In this case, the extension number specified to TS calls UID/GID configured in ”AIR Calls from SIP”.

```
[default]
exten => 101,1,Dial(SIP/101,30,r)
exten => 101,2,Hangup()

exten => 200,1,Dial(SIP/201,30,r)
exten => 200,2,Hangup()

exten => 201,1,Dial(SIP/202,30,r)
exten => 201,2,Hangup()
```

#### 4.4.2.2. Advanced Configuration

In basic configuration, only one call destination can be assigned to TS.

But, by using Radio Activity’s proprietary header, it is possible to configure any destinations.

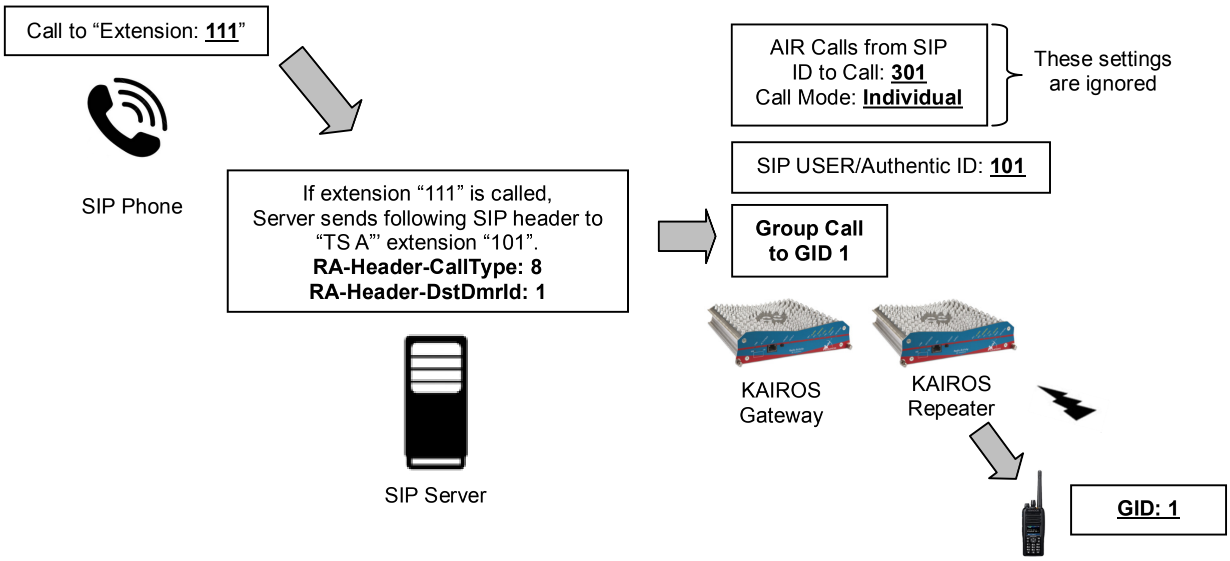
Following shows an example of extensions.conf.

```
exten =>111,1,SIPAddHeader(RA-Header-CallType: 8)
exten =>111,n,SIPAddHeader(RA-Header-DstDmrId: 1)
exten =>111,n,Dial(Sip/101,20,ft)
exten =>111,n,Hangup
```

Order to call "Call Type:8(Group Call), ID: 1" to Repeater

When extension:111 is called

Extension specified to TS of Repeater




Call Type is specified like following.

| Call Type | Description |
|---|---|
| 0x00 | Private Call |
| 0x08 | Group Call |

**Note:**

Extension: 111 must be added same as other extensions in sip.conf file.

```
[111]
type=friend
defaultuser=111
canreinvite=no
host=dynamic
```

### 4.4.3 Reflect and Confirm Configuration Files

It has to reflect configuration when the files are modified.

Entering following commands on terminal window, Asterisk CLI will be shown.

```
asterisk -vvvcr
```

```
[root@localhost admin]# asterisk -vvvcr
Asterisk 15.5.0, Copyright (C) 1999 - 2016, 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 15.5.0 currently running on localhost (pid = 1703)
localhost*CLI>
```

If above figure is shown, Asterisk is working correctly. On this window, enter following commands.

```
sip reload
```

```
dialplan reload
```

This makes each configurations be reflected.

Below shows the list of all extensions which is configured.

```
sip show peers
```

```
localhost*CLI> sip show peers
Name/username              Host                                    Dyn Forceport Comedia    ACL Port     Status      Description
101/101                    192.168.0.40                             D  Auto (No)  No            57224    Unmonitored
111/111                    (Unspecified)                            D  Auto (No)  No            0        Unmonitored
200/200                    (Unspecified)                            D  Auto (No)  No            0        Unmonitored
201/201                    (Unspecified)                            D  Auto (No)  No            0        Unmonitored
202/202                    (Unspecified)                            D  Auto (No)  No            0        Unmonitored
222/202                    (Unspecified)                            D  Auto (No)  No            0        Unmonitored
6 sip peers [Monitored: 0 online, 0 offline Unmonitored: 1 online, 5 offline]
localhost*CLI>
```

It is described connection to the extension: 101 having IP address ”192.168.0.40”.
It can confirm whether the registered SIP devices (including KAIROS) are available or not.

Exit CLI.

```
quit
```

### 4.4.4 Connect to Outside Line (Public Line)

In the case of connecting to outside line (public line), it is necessary subscribing IP telephone network service in advance.

These services vary from each country. Following describes an example of the connection in Japan. Service and account information used this time are following.

・Service Information

| | |
|---|---|
| Service Name | Fusion IP-Phone Smart |
| Company Name | Rakuten |

・Account Information

| Name | Value | Remarks |
|---|---|---|
| Phone Number | 0505329xxxx | |
| Domain Name | smart.0038.net | |
| SIP Account | 5329xxxx | without “050” from Phone Number |
| SIP Account Password | yyyyyyyy | |

Add following settings at ”[general]” section in sip.conf.

pedantic=yes ; <- **If you need to connect internet**

register => 5329xxxx:yyyyyyyy@smart.0038.net

SIP Account

Domain Name

SIP Account Password

Furthermore, add ”[fusion]” section.

[fusion]
type=friend
username=5329xxxx
fromuser=5329xxxx
secret=yyyyyyyy
host=smart.0038.net
fromdomain=smart.0038.net
context=incoming
insecure=port,invite
canreinvite=no
dtmfmode=rfc2833

SIP Account

SIP Account Password

Domain Name

Described in extensions.conf separately

The sip.conf sample added these lines is below.

```
[general]
context=default
allowguest=yes
allowoverlap=no
bindaddr=0.0.0.0
tcpenable=no
transport=udp
srvlookup=no
pedantic=yes
disallow=all
allow=ulaw
dtmfmode = info
;fusion register
register => 5329xxxx:yyyyyyyy@smart.0038.net
[fusion]
type=friend
username=5329xxxx
fromuser=5329xxxx
secret=yyyyyyyy
canreinvite=no
context=incoming
insecure=port,invite
host=smart.0038.net
fromdomain=smart.0038.net
dtmfmode=rfc2833
allowsubscribe=no
[101]
type=friend
defaultuser=101
canreinvite=no
host=dynamic
[200]
type=friend
defaultuser=200
canreinvite=no
host=dynamic
[201]
type=friend
defaultuser=201
canreinvite=no
host=dynamic
```

Next, add Send/Receive configuration in extensions.conf. Section name is ”[incoming]”.

・Receive configuration

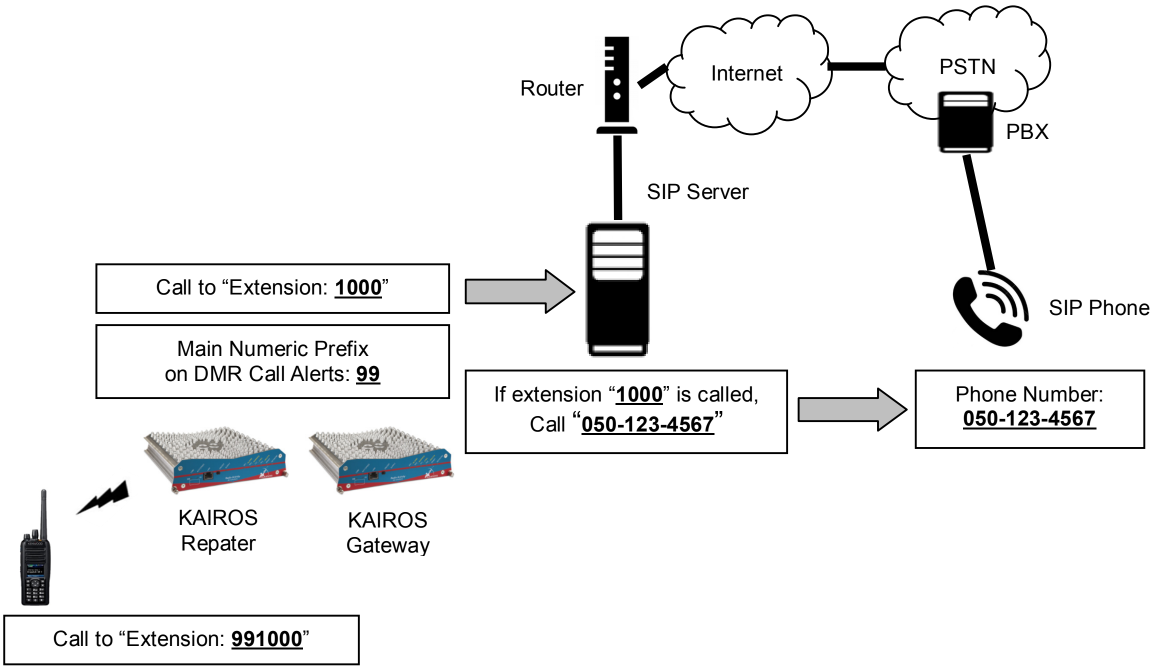
```
[incoming]
exten => 5329xxxx,1,Dial(SIP/01,30)
```

SIP Account

Above example shows that extension: 01 is called when the registered outside line account ” 5329xxxx” receives a call.

・Send Configuration

Following example shows configuration which Radio calls to outside line.
This example is based on following diagram.




In the case of a call from a radio to outside line, it is also necessary to add prefix which is configured in SIP Gateway. This time, UID: 1000 is assigned to connect with outside line.
The contents should be described in extensions.conf as below.

```
exten => 1000,1,Set(CALLERID(num)= 5329xxxx)
exten => 1000,1,Set(CALLERID(name)=5329xxxx)
exten => 1000,n,Dial(SIP/0501234567@fusion)
```

SIP Account

UID for connecting outside line

Outside line Num which is called

The description following to @ in the third line should be section name ”fusion” added in sip.conf.

The extensions.conf sample is below.

```
[default]
exten => 101,1,Dial(SIP/101,30,r)
exten => 101,2,Hangup()
exten => 200,1,Dial(SIP/200,30,r)
exten => 200,2,Hangup()
exten => 201,1,Dial(SIP/201,30,r)
exten => 201,2,Hangup()

exten => 1000,1,Set(CALLERID(num)=5329xxxx)
exten => 1000,2,Set(CALLERID(name)=5329xxxx )
exten => 1000,3,Dial(SIP/0501234567@fusion)

[incoming]
exten => 5329xxxx,n,Dial(SIP/101,20,ft)
```

## 4.5 Regarding SIP phone and Other SIP Devices

・SIP phone corresponding Asterisk was used in this investigation. It should be reminded that SIP phone must be chosen one which your SIP server is supporting.

・Always IP device has to configure below.

1. Extension of SIP device itself.
2. SIP server's IP Address

## 4.6 Reference

References: Following each files are issued by Radio Activity.

"AN_0015 - KA_SIP releases ***.pdf"

"ENB52-KAIROS user manual ***.pdf"

"PTP usage and applications ***.pdf"