KENWOOD

KPG-1013GW IP Gateway

Initial Setting Guide (Appendix)

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Chapter Contents

1. About Asterisk Setting Guide

1.1. Outline

This manual shows the procedure to make a call to a SIP phone by installing Asterisk on the PC Server where the IP Gateway behaves. Asterisk is open source IP-PBX software developed and released by the US company Digium.

1.2. Notice

The key points if the IP Gateway and Asterisk are made coexistent are as follows:

- The ports used in Asterisk are for SIP use (60520) and for RTP use (63600 to 63999), and these are treated as fixed values in the IP Gateway.
- The various logs to be recorded in Asterisk cannot be acquired from the IP Gateway.
- OSS Redundancy (Redundancy by using OSS) cannot be used.

1.3. Condition

The target for the coexistence of the IP Gateway and Asterisk satisfies the following conditions:

Condition	Requirements/ Minimum Conditions	Remarks
Hardware	The 1 unit of PC Server where the IP Gateway can behave	
Operating System	CentOS 7.6	
Asterisk	Limited to the distribution for coexistence specified by KENWOOD	Asterisk 16 is used.
IP Gateway Firmware Version	V1.20.00 or later	

2. Outline of the Process Steps

3.1 IP Gateway Preparation

| The IP Gateway where Asterisk coexists is prepared.

3.2 Package Installation Required for Asterisk

| The package required for Asterisk is installed.

3.3 Configuring Asterisk

The various configurations to make Asterisk behave are configured.

3. Details of the Process Steps

3.1. IP Gateway Preparation

Prepare the IP Gateway where Asterisk coexists according to Chapter 4 of IP Gateway Initial Setting Guide (procedure up to 4.4 IP Gateway Application Install).

3.2. Package Installation Required for Asterisk

3.2.1. Connecting Monitor/ Keyboard/ Mouse to the PC Server

Connect Monitor/ Keyboard/ Mouse to the PC Server, and login by a user with root authorization.

3.2.2. Insertion of the Asterisk Installation Media to the PC Server

Insert the Asterisk installation media (CD/ DVD/ USB memory) to the PC Server. After installation media insertion, the installation media icon automatically appears on the desktop.

<u>In the case of USB memory</u>

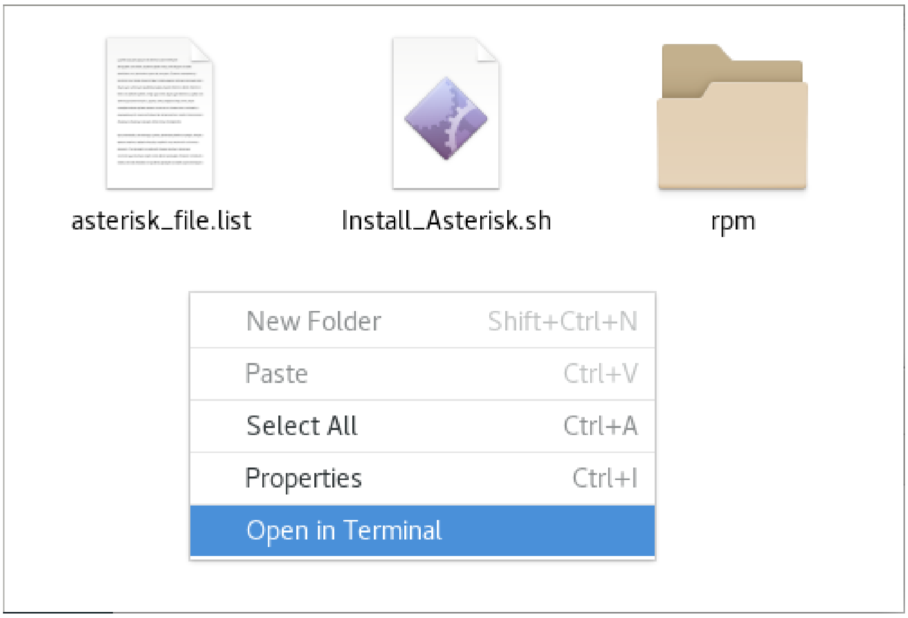
By double-clicking this icon, the ISO file appears on the file manager.

Double-click "Install_Asterisk_V10000.iso".

If mounting occurs, the "CDROM" icon of the following diagram is generated on the desktop:

3.2.3. Displaying the Contents of the Mounted ISO File on the File Manager and Opening the Console

By double-clicking the "CDROM" icon, the installer file appears on the file manager. Open the console by right-clicking on the following folder of the file manager and selecting "Open in Terminal".



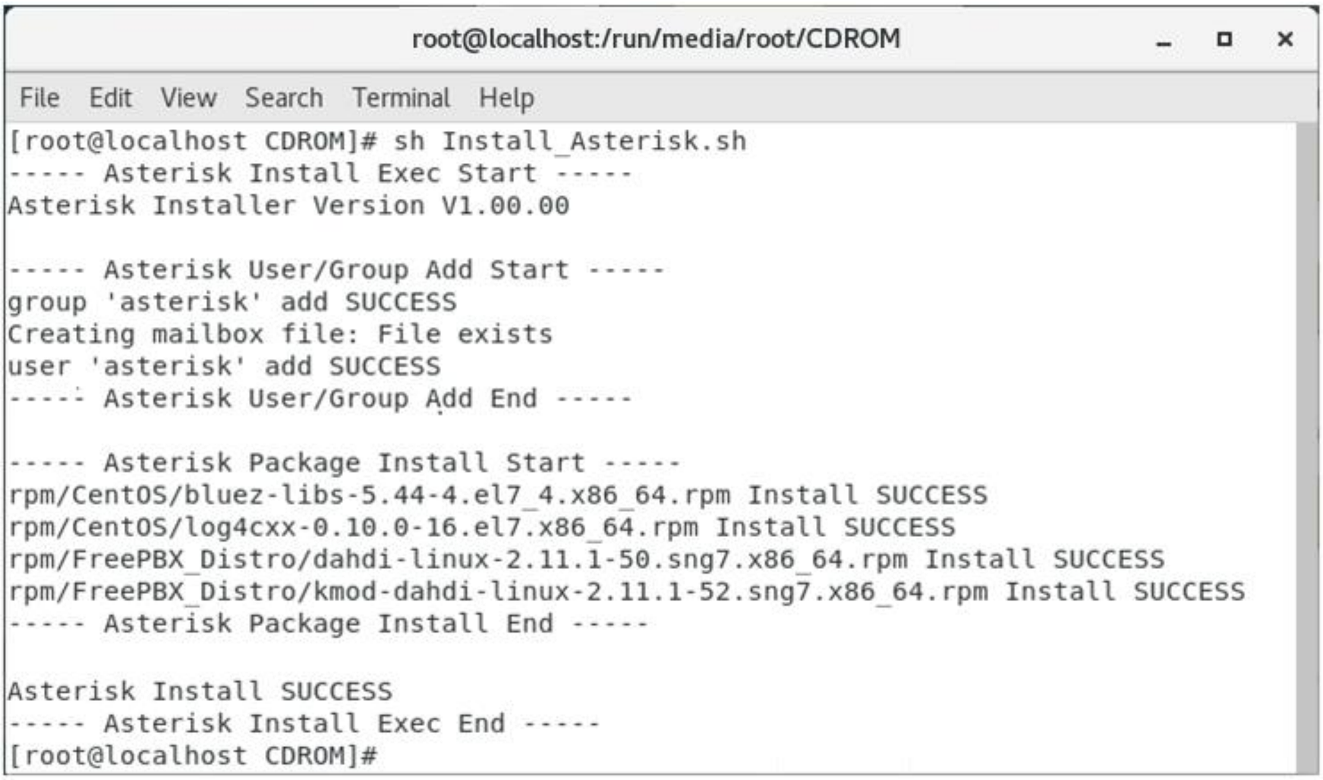
3.2.4. Executing the Installation Script (GUI Operation Unavailable)

Execute "Install_Asterisk.sh" in the ISO file by using the following command on the console.

[root@localhost CDROM]# **sh Install_Asterisk.sh**

If Asterisk installation is successful, the following sample is displayed:

<Sample>



3.2.4.1. Confirming the Installation Script Execution Result

Find the log file "asterisk. install.log" at /var/log/*, and confirm the contents.

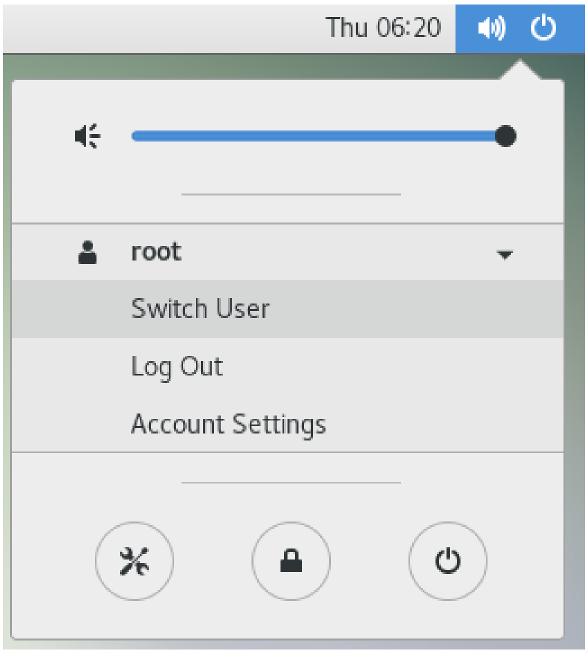
***Confirmation of a successful result**

The result is successful if "Asterisk Install SUCCESS" is written at the very bottom of the log.
If "ERROR: ..." is written midway through the log, the process failed there.

3.3. Configuring Asterisk

Login to the PC Server as the admin user.

By clicking the power symbol at the desktop top-right and using "Switch User", the user can also be switched.



3.3.1. Configuring sip.conf

Define the Asterisk configuration, and the information of SIP Phone to be connected to Asterisk and the outside line service. Because an explanation about the configuration is described in the original "sip.conf" file, copy the file for backup by executing the following command.

```
[admin@localhost ~]# cp -a /etc/asterisk/sip.conf /etc/asterisk/original.sip.conf
```

To change the configuration, edit the "sip.conf" file by using the vi editor. Open the file by executing the following command.

```
[admin@localhost ~]# vi /etc/asterisk/sip.conf
```

Delete all lines of the "sip.conf" file by executing the following command.

```
:%d
```

Change the command mode to the entry mode by using the [i] key, and enter the following sample. Always change the parts indicated in **red bold** to match the information of the SIP Phone and outside line service used, and the configuration of the IP Gateway used. Make changes as required for other configuration contents after referring to the original "sip.conf" file. The parts indicated in blue are explanations, and not configuration contents.

<Sample>

```
; ---- SIP Configuration for Asterisk -------------

[general]                 ← Configures the options of the entire "sip.conf" file.
maxexpirey=43200          ← Configures 43200 [s] for the maximum number of seconds for which the
Register request will be enabled to match the value that can be configured in the IP Gateway.
context=default
bindport=60520            ← Changes the SIP port used in Asterisk to 60520.
bindaddr=0.0.0.0
disallow=all
allow=ulaw:60             ← Configures 60 [msec] for the RTP (ulaw) sent from Asterisk.
allow=alaw:60             ← Configures 60 [msec] for the RTP (alaw) sent from Asterisk.
language=en
type=friend

; ---- PSTN -------------  ← Configure to match the information of the outside line service.
                            (xxxxxxxx: user name/ yyyyyyyy: password/ zzzzzzzz: host)

;PSTN register
register => xxxxxxxx:yyyyyyyy@zzzzzzzz

[------]                  ← Configure the identifier of the outside line service.
                            Used for the "extensions.conf" file configuration.
type=friend
username=xxxxxxxx
fromuser=xxxxxxxx
secret=yyyyyyyy
host=zzzzzzzz
context=incoming
insecure=port,invite
canreinvite=no
fromdomain=zzzzzzzz
dtmfmode=rfc2833
allowsubscribe=no

[xxxxxxxx]
type=friend
defaultuser=xxxxxxxx
secret=yyyyyyyy
canreinvite=no
host=dynamic

; ---- PABX Phone -------------     ← Configure to match the information of the PABX Phone.
                                      (XXX: ID/ YYY: password)

;PABX Phone 1                       ← Configure for only the number of PABX Phones to be used.
[XXX]
type=friend
defaultuser=XXX
secret=YYY
canreinvite=no
host=dynamic

;PABX Phone 2
[XXX]
type=friend
defaultuser=XXX
secret=YYY
canreinvite=no
host=dynamic

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```

(Continued from the previous part)

```
; ---- IP Gateway -------------  ← Configure to match the configuration of the IP Gateway. (XXX : SIP User ID)

;IPGW For KAIROS
[XXX]
type=friend
defaultuser=XXX
canreinvite=no
host=dynamic
dtmfmode=inband
```

After the editing ends, change the entry mode to the command mode by using the [Esc] key, and save by the following command and end the editing.

```
:wq
```

3.3.2. Configuring rtp.conf

To change the range of the RTP port used in Asterisk, edit the "rtp.conf" file by using the vi editor. Open the file by executing the following command.

```
[admin@localhost ~]# vi /etc/asterisk/rtp.conf
```

Change the command mode to the entry mode by using the [i] key, and edit the following parts indicated in **red bold**. The parts indicated in blue are explanations, and not configuration contents.

<Sample>

```
;
; RTP Configuration
;
[general]
;
; RTP start and RTP end configure start and end addresses
;
; Defaults are rtpstart=5000 and rtpend=31000
;
rtpstart=63600          ← Change the RTP port used in Asterisk to 63600 to 63999.
rtpend=63999

(Omitted)
```

After the editing ends, change the entry mode to the command mode by using the [Esc] key, and save by the following command and end the editing.

```
:wq
```

3.3.3. Configuring extensions.conf

Define the dial plan of the Asterisk transmission behavior and Asterisk receipt behavior when communicating between a telephone and the transceiver or between telephones. Because an explanation about the configuration is described in the original "extensions.conf" file, copy the file for backup by executing the following command.

```
[admin@localhost ~]# cp -a /etc/asterisk/extensions.conf /etc/asterisk/original.extensions.conf
```

To change the configuration, edit the "extensions.conf" file by using the vi editor. Open the file by executing the following command.

```
[admin@localhost ~]# vi /etc/asterisk/extensions.conf
```

Delete all lines of the "extensions.conf" file by executing the following command.

```
:%d
```

Change the command mode to the entry mode by using the [i] key, and enter the following sample. Always change the parts indicated in **red bold** to match the information of the SIP Phone and outside line service used, and the configuration of the IP Gateway used. Make changes as required for other configuration contents after referring to the original "extensions.conf" file. The parts indicated in blue are explanations, and not configuration contents.

<Sample of Tier II System>

```
; ---- extensions.conf - the Asterisk dial plan -------------

[general]                  ← Configures the options of the entire "extensions.conf" file.
writeprotect=no
priorityjumping=no

; ---- Variable Definition -------------

[globals]                       ← Define the values used in the "extensions.conf" file.
PSTN_SIP_ACCOUNT= XXX           ← SIP account of the outside line service
PSTN_PHONE1= XXX                ← Phone number of the PSTN Phone
PABX_PHONE1= XXX                ← ID of the PABX Phone
PABX_PHONE2= XXX                ← ID of the PABX Phone
IPGW_SIPUSERID=XXX              ← SIP User IDconfigured in the IP Gateway

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```

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; ---- System Tier II (Transceiver/ PABX Phone → PABX Phone) -------------
↓ Dial plan when a PABX Phone is called from the Transceiver/ PABX Phone

[default]
↓ Configuration to connect to PABX_PHONE1 if PABX_PHONE1 is the receiver

```
;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)
```

↓ Configuration to connect to PABX_PHONE2 if PABX_PHONE2 is the receiver

```
;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)
```

; ---- System Tier II (PABX Phone → Transceiver) -------------
↓ Dial plan when the Transceiver is called from the PABX Phone
To call a phone by using the following configuration, enter the Prefix number (XX/ YY) ＋ target DMR ID.

```
;IndividualCall
exten => _XX.,1,NoOp("Extension "${EXTEN})                ← Desired Prefix number
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: ${EXTEN:2})   ← The number of Prefix digits = 2 (Can be changed.)
same => n,SIPAddHeader(JKC-Header-SrcType: pabx)
same => n,Dial(Sip/${IPGW_SIPUSERID},20,hft)
same => n,Hangup

;Group Call
exten => _YY.,1,NoOp("Extension "${EXTEN})                ← Desired Prefix number
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: ${EXTEN:2})   ← The number of Prefix digits = 2 (Can be changed.)
same => n,SIPAddHeader(JKC-Header-SrcType: pabx)
same => n,Dial(Sip/${IPGW_SIPUSERID},20,hft)
same => n,Hangup
```

; ---- System Tier II (Transceiver → PSTN Phone) -------------
↓ Dial plan when a PSTN Phone is called from the Transceiver
In the following configuration, a call can be made to the outside line of PSTN_PHONE1 by entering the Prefix number (PSTN Prefix configured in the IP Gateway) ＋ ZZZ.

```
exten => ZZZ,1,Answer(500)                        ←Desired number
same => n,Dial(SIP/${PSTN_PHONE1}@------,20,ftrH)  ← Configures the identifier of the outside line service configured in the "sip.conf" file.
same => n,Playtones(busy)
same => n,Busy(6)
```

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; ---- System Tier II (PSTN Phone → Transceiver) -------------

↓ Dial plan when the Transceiver is called from the PSTN Phone
In the following configuration, Asterisk requests the number entry of [1]/ [2]/ [3] by a voice prompt if the Transceiver is called from an outside line.

[1]: To PABX Phone
When number entry is requested by a voice prompt, enter the ID of the PABX Phone.
[2]: To DMR ID Individual (Transceiver)
When number entry is requested by a voice prompt, enter the 8-digit DMR ID.
[3]: To DMR ID Group
When number entry is requested by a voice prompt, enter the 8-digit DMR ID.

```
[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 (PABX Phone)
exten => 1,1,Goto(IVR-PABXPhone,1,1)
;Input 2 (DMRID Individual)
exten => 2,1,Goto(IVR-DMR-Individual,2,1)
;Input 3 (DMRID Group)
exten => 3,1,Goto(IVR-DMR-Group,3,1)

[IVR-PABXPhone]
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},20,hft)
same => n,Hangup
```

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```
[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},20,hft)
same => n,Hangup
```

<Sample of Tier III / S-Trunking System>

```
; ---- extensions.conf - the Asterisk dial plan -------------

[general]                  ← Configures the options of the entire "extensions.conf" file.

writeprotect=no
priorityjumping=no

; ---- Variable Definition -------------

[globals]                  ← Define the values used in the "extensions.conf" file.
PSTN_SIP_ACCOUNT= XXX      ← SIP account of the outside line service
PABX_PHONE1= XXX           ← ID of the PABX Phone
PABX_PHONE2= XXX           ← ID of the PABX Phone
IPGW_SIPUSERID= XXX        ← SIP User IDconfigured in the IP Gateway

; ---- SysSystem Tier III/ S-Trunking (Transceiver/ PABX Phone → PABX Phone) -------------
↓ Dial plan when a PABX Phone is called from the Transceiver/ PABX Phone

[default]
↓ Configuration to connect to PABX_PHONE1 if PABX_PHONE1 is the receiver

;REQ PABX_PHONE1 Call PABX_PHONE1
exten => ${PABX_PHONE1}, 1,Dial(SIP/${PABX_PHONE1},30,rH)
same => n,Playtones(busy)
same => n,Busy(3)

↓ Configuration to connect to PABX_PHONE1 if PABX_PHONE1 is the receiver

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

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(Continued from the previous part)

; ---- System Tier III/ S-Trunking (PABX Phone → Transceiver) -------------
↓ Dial plan when the Transceiver is called from the PABX Phone
To call a phone by using the following configuration, enter the Prefix number (XX/ YY) ＋ target DMR ID.

```
;IndividualCall
exten => _XX.,1,NoOp("Extension "${EXTEN})                    ← Desired Prefix number
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: ${EXTEN:2})    ← The number of Prefix digits = 2 (Can be changed.)
same => n,SIPAddHeader(JKC-Header-SrcType: pabx)
same => n,Dial(Sip/${IPGW_SIPUSERID},20,hft)
same => n,Hangup

;Group Call
exten => _YY.,1,NoOp("Extension "${EXTEN})                    ← Desired Prefix number
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: ${EXTEN:2})    ← The number of Prefix digits = 2 (Can be changed.)
same => n,SIPAddHeader(JKC-Header-SrcType: pabx)
same => n,Dial(Sip/${IPGW_SIPUSERID},20,hft)
same => n,Hangup
```

; ---- System Tier III/ S-Trunking (Transceiver → PSTN Phone) -------------
↓ Dial plan when a PSTN Phone is called from the Transceiver
In the following configuration, transmission is made by using the Prefix number (045/ 080) + the phone number.
The Prefix number is configured based on the environment.
The identifier of the outside line service configured in sip.conf is configured in "------".

```
exten => _045.,1,Dial(SIP/${EXTEN}@------,20,Hft)
same => n,Playtones(busy)
same => n,Busy(6)

exten => _080.,1,Dial(SIP/${EXTEN}@------,20,Hft)
same => n,Playtones(busy)
same => n,Busy(6)
```

; ---- System Tier III/ S-Trunking (PSTN Phone → Transceiver) -------------
↓ Dial plan when the Transceiver is called from the PSTN Phone
In the same manner as with System Tier II, Asterisk requests the number entry by a voice prompt if the Transceiver is called from an outside line.

```
[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)
```

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```
(Continued from the previous part)

;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})

[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},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},20,hft)
same => n,Hangup
```

After the editing ends, change the entry mode to the command mode by using the [Esc] key, and save by the following command and end the editing.

```
:wq
```

3.3.4. Configuring indications.conf

To change the tone, edit the "indications.conf" file by using the vi editor. Open the file by executing the following command.

```
[admin@localhost ~]# vi /etc/asterisk/indications.conf
```

Change the command mode to the entry mode by using the [i] key, and edit the following parts indicated in **red bold**. In the following sample, a default US tone is changed to an Italy tone. When configuring to another country, make the change after referring to the "indications.conf" file. The parts indicated in blue are explanations, and not configuration contents.

<Sample>

```
(Omitted)

[general]
;country=us        ; default location     ← Disable the configuration by adding ";" at the beginning.
country=it         ; Italy                ← Configure a tone by changing to the desired country.

(Omitted)
```

After the editing ends, change the entry mode to the command mode by using the [Esc] key, and save by the following command and end the editing.

```
:wq
```

3.3.5. Configuring features.conf

To change the disconnection by DTMF in Asterisk to "***", edit the "features.conf" file by using the vi editor. Open the file by executing the following command.

```
[admin@localhost ~]# vi /etc/asterisk/features.conf
```

Change the command mode to the entry mode by using the [i] key, and edit the following parts indicated in **red bold**. The parts indicated in blue are explanations, and not configuration contents.

<Sample>

```
(Omitted)

[featuremap]
;blindxfer => #1
disconnect => ***                         ← Delete ";" at the beginning, and change to "***".
;automon => *1
;atxfer => *2
;parkcall => #72
;automixmon => *3

(Omitted)
```

After the editing ends, change the entry mode to the command mode by using the [Esc] key, and save by the following command and end the editing.

```
:wq
```

3.3.6. Reflecting the Configuration

To reflect the configuration changes in Asterisk, start Asterisk CLI by executing the following command.

```
[admin@localhost ~]# sudo asterisk -r
```

As entering the password of the user who has executed a command is requested when the command is executed, enter the password.

<Sample>

```
[sudo] password for admin:
Asterisk 16.6.2, 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 16.6.2 currently running on localhost (pid = 32098)
localhost*CLI>
```

If the "sip.conf" file is changed, reflect the configuration change by executing the following command.

```
localhost*CLI> sip reload
```

If the "extensions.conf" file is changed, reflect the configuration change by executing the following command.

```
localhost*CLI> dialplan reload
```

After the configuration changes are reflected in Asterisk, close Asterisk CLI by executing the following command.

```
localhost*CLI> quit
```

Restart Asterisk by executing the following command.

```
[admin@localhost ~]# sudo systemctl restart asterisk.service
```