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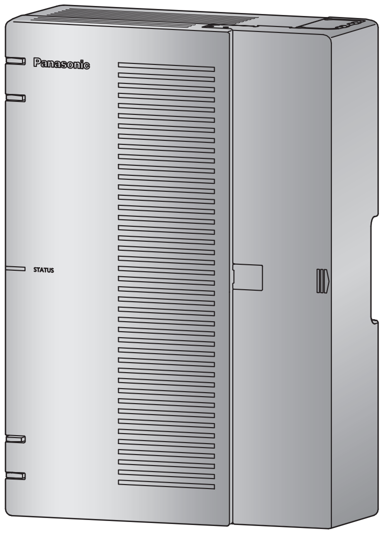


Feature Manual

Hybrid IP-PBX

Model No. KX-HTS824
KX-HTS32

Thank you for purchasing this Panasonic product.
Please read this manual carefully before using this product and save this manual for future use.
In particular, be sure to read "1.1.1 For Your Safety" before using this product.

KX-HTS Series: PJMPR Software File Version 002.00000 or later

Manuals and supporting information are provided on the Panasonic Web site at:
https://panasonic.net/cns/pcc/support/pbx/

Introduction

About this Feature Manual

This Feature Manual is designed to serve as a feature reference for the Panasonic Hybrid IP-PBX.facilities. It explains what this PBX can do, and how to obtain the most out of its many features and facilities.

The Structure of this Manual

This manual contains the following sections:

Section 1, For Your Safety

Provides details about safety precautions for preventing personal injury and/or damage to property.

Section 2, Feature Numbering Lists

Provides details about the feature number lists for user and manager.

Section 3, Call Control Features

Provides details about the call handling features.

Section 4, Flexible Button Features

Provides details about the features of flexible buttons.

Section 5, Voice Mail Features

Provides details about the features of the Voice Mail system.

Section 6, Networking Features

Provides details about the networking features.

Compatible Telephones and Devices

- The button on the DSS Console which is connected to the KX-HDV series can be used as a Flexible Button. (Refer to 4 Flexible Button Features)
- A list of telephone and device types that are compatible with this PBX can be found in the Web site as below:
  https://panasonic.net/cns/pcc/support/pbx/

Functional Limitation

- Depending on the PBX's software version, some features may not function. For details about which versions support these features, consult your dealer.
- When a user makes a call to an external line, if the called party does not answer to the call within 180 seconds, the call will be disconnected. Analog trunk lines have to be set enable reverse detection. It will be able to detect answering to call on the analog trunk line. For more detail, refer to "2.4.1 PBX Configuration—[3-1] Trunk—Port-Analog - Basic-Reverse Detection" in the Programming Item List.

Other Information

Trademarks

- All trademarks identified herein are the property of their respective owners.

Note

- The contents of this manual apply to PBX with a certain software version, as indicated on the cover of this manual. To confirm the software version of your PBX, refer to "4.1 Maintenance-Version Information-Main Unit Version" **in the Programming Item List.**
- Some optional hardware, software, and features are not available in some countries/areas, or for some PBX models. Please consult your certified Panasonic dealer for more information.
- Product specifications are subject to change without notice. In some cases, additional information, including updates to this and other manuals, is included in the Maintenance Console's Information before programming.

- Throughout this manual, phone displays and other displays are shown in English. Other languages may be available, depending on the country or area.
- In this manual, the suffix of each model number (e.g., KX-HTS824**SX**) is omitted unless necessary.
- This PBX supports SIP (Session Initiation Protocol) phones. However, some PBX features may not be available for SIP phones, depending on your telephone type.

List of Abbreviations

A

AA
Automated Attendant

ACD
Automatic Call Distribution

ARS
Automatic Route Selection

C

CDR
Call Detail Record

CLI
Calling Line Identification

CLIP
Calling Line Identification Presentation

CLIR
Calling Line Identification Restriction

COS
Class of Service

CPC
Calling Party Control

D

DDI
Direct Dialing In

DID
Direct Inward Dialing

DIL
Direct In Line

DISA
Direct Inward System Access

DN
Directory Number

DND
Do Not Disturb

DSCP
DS Code Point
*This word is used in the Programming Item List.

DSS
Direct Station Selection

DTMF
Dual Tone Multi-Frequency

F

FWD
Call Forwarding

I

IVR
Interactive Voice Response

L

LED
Light Emitting Diode

M

MOH
Music On Hold

O

OGM
Outgoing Message

P

PIN
Personal Identification Number

R

RSSI
Received Signal Strength Indication
*This word is used in the Programming Item List.

S

S-CO
Single-CO

SIP
Session Initiation Protocol

SLT
Single Line Telephone

T

TRS
Toll Restriction

V

VM

Voice Mail

Table of Contents

Section 1

For Your Safety

1.1 For Your Safety

1.1.1 For Your Safety

Description

To prevent personal injury and/or damage to property, be sure to observe the following safety precautions.
The following symbols classify and describe the level of hazard and injury caused when this unit is operated or handled improperly.

⚠ **CAUTION**	This notice means that misuse could result in injury or damage to property.

The following types of symbols are used to classify and describe the type of instructions to be observed.

❗	This symbol is used to alert users to a specific operating procedure that must be followed in order to operate the unit safely.

⚠ CAUTION

❗

- The software contained in the TRS and ARS features to allow user access to the network must be upgraded to recognize newly established network area codes and exchange codes as they are placed into service. Failure to upgrade the on-premise PBXs or peripheral equipment to recognize the new codes as they are established will restrict the customer and users of the PBX from gaining access to the network and to these codes.
  KEEP THE SOFTWARE UP TO DATE WITH THE LATEST DATA.
- There is a risk that fraudulent telephone calls will be made in the following cases:
  - A third party discovers a personal identification number (PIN) (verification code PIN or extension PIN) of the PBX.
  - Using the Trunk-to-Trunk Call feature of DISA.

  The cost of such calls will be billed to the owner/renter of the PBX. To protect the PBX from this kind of fraudulent use, we strongly recommend:
  a. Keeping PINs secret.
  b. Selecting complex, random PINs that cannot be easily guessed.
  c. Changing PINs regularly.
- To the Administrator or Installer regarding account passwords
  1. Please provide all system passwords to the customer.
  2. To avoid unauthorized access and possible abuse of the PBX, keep the passwords secret, and inform the customer of the importance of the passwords, and the possible dangers if they become known to others.

3. The PBX has default passwords preset. For security, change an installer password as soon as the PBX system is installed at the site.
4. Change the passwords periodically.
5. It is strongly recommended that passwords of 16 numbers or characters be used for maximum protection against unauthorized access.

Section 2

Feature Number Lists

2.1 Feature Number Lists

2.1.1 Feature Number Lists for User

Description

It is possible to change extension settings by dialing from an extension. In addition, dialing feature numbers allows you to use those features.

Operation

Dial the following feature numbers from an extension. Feature numbers are specified in 2.2.4 PBX Configuration—[1-4] System—Numbering Plan in the Programming Item List.

[Programming by dial]

Function Name	Feature Number (Default)	Parameter	References
Walking COS	*47	[Extension Number] [Extension PIN[*1]] [Idle Line Access Number] [Destination phone number]	3.4.1 Toll Restriction (TRS) 3.4.2 Walking COS (TRS level)
Account Code		[*] [Account Code] [Idle Line Access Number] [Destination phone number]	3.4.1 Toll Restriction (TRS) 3.4.3 Account Code Entry
Extension Dial Lock Set / Cancel	*77	[0: Cancel] [Extension PIN[*1]] [#]	3.4.5 Extension Dial Lock
		[1: Set]	
FWD/DND Set / Cancel	*710	[0: FWD/DND OFF]	3.1.9 Call Forwarding (FWD)
		[1: DND ON]	
		[2: FWD ALL ON] [Forwarding destination phone number]	
		[3: FWD BUSY ON] [Forwarding destination phone number]	
		[4: FWD NA ON] [Forwarding destination phone number]	
		[5: FWD Busy/NA ON] [Forwarding destination phone number]	
Extension PIN Set / Cancel	*799	[0: Cancel] [Extension PIN[*1]] [#]	3.4.2 Walking COS (TRS level) 3.4.3 Account Code Entry 3.4.5 Extension Dial Lock
		[1: Set] [New Extension PIN[*1]] [#] [New Extension PIN[*1]] [#]	

[*1] The extension PIN is specified in 2.3.2 PBX Configuration—[2-2] Extension—Phone—PIN in the Programming Item List.

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port
2.3.2 PBX Configuration—[2-2] Extension—Phone
2.2.4 PBX Configuration—[1-4] System—Numbering Plan

[Feature List]

Function Name	Feature Number (Default)	Parameter	References
Trunk Line / Trunk Group	8	[#] [Trunk Group Number] [Destination phone number]	3.2.3 Line Access, Trunk Group
	8	[0] [Single Trunk line Number] [Destination phone number]	3.2.4 Line Access, Trunk Line
Idle Line Access (Local Access) - 1	9	[Destination phone number]	3.2.2 Line Access, Automatic
Idle Line Access (Local Access) - 2	0		
Redial	#	None	3.2.7 Redial
System Speed Dialing	**	[0][0][0]–[1][9][9]	3.3.1 System Speed Dial
Paging	*33	[Group Number (2 digits)]	3.10.1 Paging
Group Call Pickup (Specified Extension Group)	*40	[Group Number (2 digits)]	3.1.8 Call Pickup
Group Call Pickup (Own Extension Group)	*40	[#]	3.1.8 Call Pickup
Directed Call Pickup	*41	[Extension number] [Extension number (2 digits)] [#]	3.1.8 Call Pickup
Call Park*1	7	[0][0]	3.7.2 Call Park
Call Park Retrieve	7	[0][1]-[2][4]: Park Area Number	3.7.2 Call Park
Door Open	5 (Fixed)	None * This feature can only be used when talking to the doorphone.	3.11.2 Door Open

*1 Before the operation, put the current call on Consultation Hold. (Refer to 3.9.1 Three-party Conference)

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port
2.3.2 PBX Configuration—[2-2] Extension—Phone
2.2.4 PBX Configuration—[1-4] System—Numbering Plan
2.3.5 PBX Configuration—[2-5] Extension—Doorphone

2.1.2 Programming by Dial for Manager

Description

It is possible to change the settings of the PBX system or view them from an extension assigned as a Manager. In addition, dialing feature numbers allows you to use those features.

Operation

Dial the following feature numbers from the Manager extension. Feature numbers are specified in 2.2.4 PBX Configuration—[1-4] System—Numbering Plan in the Programming Item List.

When dialing is complete and the settings are changed, the call will be disconnected after the confirmation tone is heard.

[Programming by dial]

Function Name	Feature Number (Default)	Parameter	References
OGM Record / Clear / Playback	*36	[0: Clear] [DISA floating extension number]	3.12.4 Outgoing Message (OGM)
		[1: Record] [DISA floating extension number]	
		[2: Playback] [DISA floating extension number]	
Time Service (Day / Lunch / Night) Switch	*780	[0: Day]	3.12.1 Time Service
		[1: Night]	
		[2: Lunch]	
		[#: Confirm the current Time Service]	
Door Open	*55	[Doorphone Port No.]	3.11.2 Door Open
System Setting	*#	Trunk Attribution [System PIN[*1]] [#] [400] [#] [Trunk line number] [#] [N] [#] **Note** [N] = [0]: No Connect [N] = [1]: Analog	Programming Item List 2.4.1 PBX Configuration—[3-1] Trunk—Port—Main—Attribution
		Trunk Dialing Mode [System PIN[*1]] [#] [410] [#] [Trunk line number] [#] [N] [#] **Note** [N] = [0]: DTMF [N] = [1]: Pulse	Programming Item List 2.4.1 PBX Configuration—[3-1] Trunk—Port-Analog - Basic—Dialing Mode
		HTTPS Port [System PIN[*1]] [#] [196] [#] [N] [#] **Note** [N] = [0]: Close [N] = [1]: Open	Programming Item List 4.2.1 Maintenance—[1-1] Management—Web Programming-Web Programming-HTTPs Enable

[*1] The System PIN is specified in 4.2.1 Maintenance—[1-1] Management—Web Programming—System PIN for Manager in the Programming Item List.

PIL Reference
2.2.4 PBX Configuration—[1-4] System—Numbering Plan
2.3.1 PBX Configuration—[2-1] Extension—Port
2.4 PBX Configuration—[3] Trunk

PIL Reference
3 Network Configuration
4.2 Maintenance—[1] Management

Section 3

Call Control Features

3.1 Incoming Call Features

3.1.1 Calling Line Identification (CLI) Distribution & Call Block

Description

Directs an incoming Trunk line call to a preprogrammed destination when the caller's identification number (e.g., Caller ID) matches the number in the System Speed Dialing Table that is used as the Caller ID Table. Each Caller ID number (telephone number for each System Speed Dialing number) can have its own destination.
If an incoming call does not notify a Caller ID, it is possible to refuse reception (Call Block).

Caller ID Modification
Received Caller ID is modified in the following steps:

1. **Caller ID Modification by Length of Digits**
   Pre-programmed numbers (max. 6 digits) can be added to the front of the Caller ID if the Call ID of digits received from the network is within the range below:
   - International: 12 digits or more (default)
   - National: 8 digits (default) to 11 digits (1 value removed from International)

   [Example]
   - Minimum Caller ID Digits (International): 12
   - Added Number (International): 001

   Before modification: 81-50-1234-5678
   ↓
   After modification: **001**81-50-1234-5678 (Add "001")
2. **Caller ID Modification by leading numbers**
   After the Caller ID is modified by the Length of Digits, the PBX checks the leading numbers of the modified number for an area code programmed in Caller ID Modify Table in Web Maintenance Console. If it finds such a code, it removes digits and adds a number to the modified number.

   [Example]
   - Area Code: 00181
   - Removed Number of Digits: 5
   - Added Number: 0

   Before modification: **00181**-50-1234-5678
   ↓
   After modification (1): 50-1234-5678 (remove "00181")
   ↓
   After modification (2): **0**50-1234-5678 (Add "0")
3. **Adding the Idle Line Access (Local Access) number**
   After the Caller ID is modified by the leading numbers, the Idle Line Access number (Idle Line Access (Local Access) - 1) is added to the modified number. (Refer to 2.1.1 Feature Number Lists for User)

   [Example]
   - Idle Line Access (Local Access) number: 9

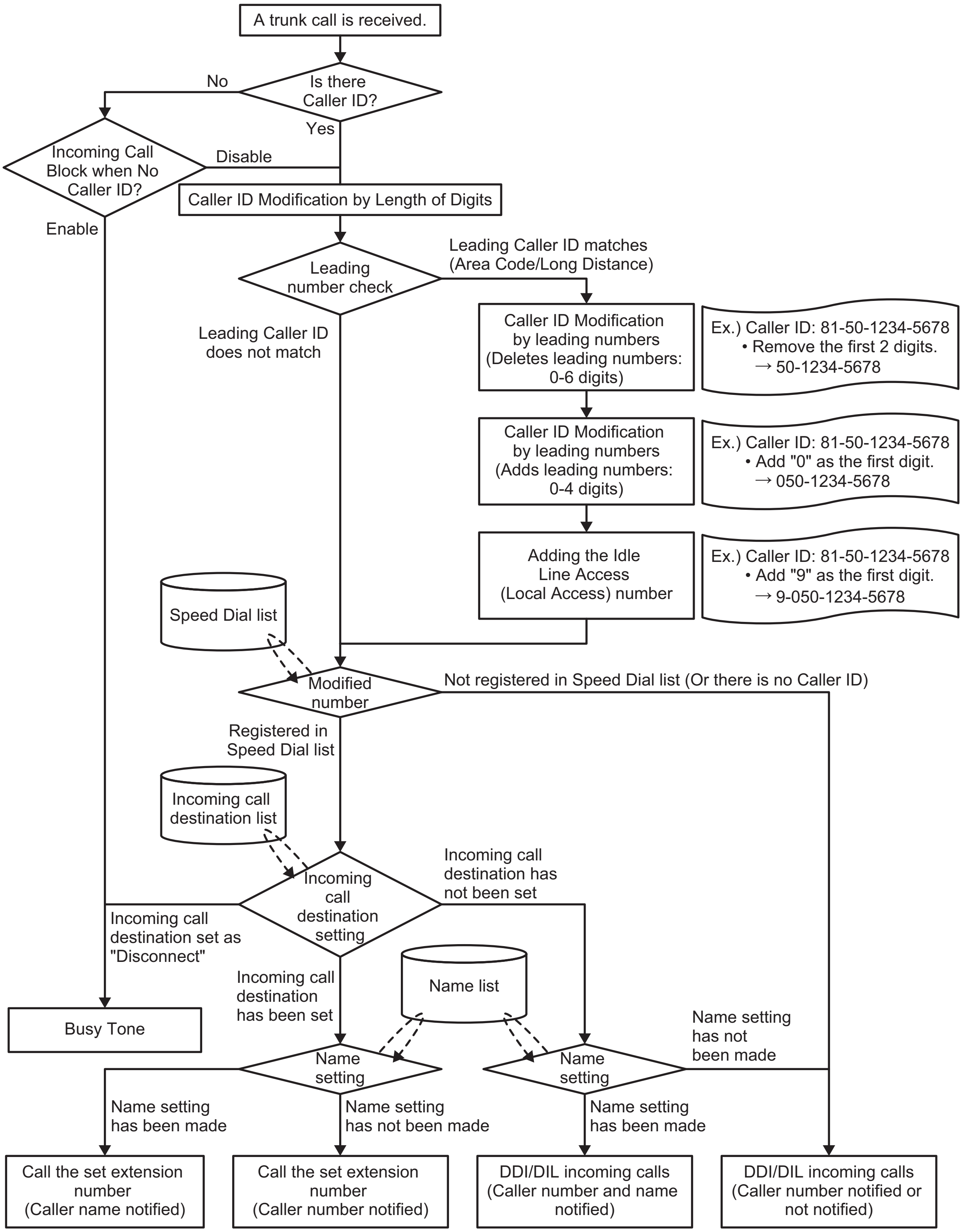
Before modification: 050-1234-5678
↓
After modification: **9**-050-1234-5678 (Add "9")

CLI Destination
Directs an incoming call to a pre-programmed destination when the modified number matches the number in the System Speed Dialing table.
For the incoming call destination setting of System Speed Dialing, the extension number (including floating extension number) and "Disconnect" can be set as CLI Destination. When "Disconnect" is set, incoming calls will be disconnected. (Refer to 3.3.1 System Speed Dial)

Searching for the caller name
If System Speed Dialing is searched for the modified number, and the caller name corresponding to the modified number is registered, the name information of the caller (Display Name) registered in System Speed Dialing will be sent to an extension when calling the extension. Whether Caller ID or Display Name is displayed depends on the extension.

For the modification flow of caller numbers, see below:

[Flow Chart]



Important

FWD No Answer Time is used as the timer for forced call disconnection. For more information, refer to 3.1.9 Call Forwarding (FWD).

Conditions	Note
[Caller ID Modification] If "–" is included in the received Call ID, it is ignored by the Call ID modification.	
[Caller ID Modification by Length of Digits] • The length of digits for National must be set smaller than International. • It is possible to specify whether to modify the caller ID received from an Analog trunk or SIP trunk.	
[Caller ID Modification by Leading numbers] The Modification table has 10 entries. It is possible to make an exception setting in case the caller's number does not match any of the 10 set entries.	
[DDI/DIL distribution] When the modified number is not registered in System Speed Dialing table, or the setting of the incoming call destination for each modified number does not exist, the call will be handled using the DDI/DIL distribution methods specified for each trunk. The call is disconnected when DDI/DIL settings are not programmed.	Refer to 3.1.2 Direct Inward Dialing (DID)/Direct Dialing In (DDI)

PIL Reference
2.4.4 PBX Configuration—[3-4] Trunk—Caller ID Modify & Block
2.6 PBX Configuration—[5] System Speed Dialing

3.1.2 Direct Inward Dialing (DID)/Direct Dialing In (DDI)

Description

Provides automatic direction of an incoming call with a DID/DDI number to a preprogrammed destination. Each DID/DDI number has a destination for each time mode (day/lunch/night).
Incoming calls with DID/DDI numbers that match extension numbers at this PBX will be sent to the corresponding extension.
For other incoming trunk call distribution methods, the priority of methods is as follows:

1. CLI Destination
2. DID/DDI
3. DIL

Important

FWD No Answer Time is used as the timer for forced call disconnection. For more information, refer to 3.1.9 Call Forwarding (FWD).

Conditions	Note
DID/DDI can be enabled or disabled for each SIP carrier in the system.	When DID/DDI is disabled, DIL will operate. Refer to 3.1.3 Direct In Line (DIL)

Conditions	Note
It is possible to modify (remove/add) the common incoming call destinations received by each SIP carrier.	Example: • Remove Digit: 6 • Additional Dial: 10 Received number: **876543**21 ↓ Modified number: **10**21 (Remove the first 6 digits, and add "10" as the first digits.)
A maximum of 200 DDI numbers can be programmed in the system.	
If a modified incoming call number does not match the DDI/DID table, it will be routed to the corresponding extension if it matches the extension number.	The floating extension number is also targeted.
If a modified incoming call number does not match the extension/floating extension number, and the DIL settings are programmed, DIL will operate. The call will be disconnected when DIL settings are not programmed.	Refer to 3.1.3 Direct In Line (DIL)

PIL Reference
2.4.3 PBX Configuration—[3-3] Trunk—DDI

3.1.3 Direct In Line (DIL)

Description

Provides automatic direction of an incoming trunk call to a preprogrammed destination without DID/DDI number. Each Trunk line has a destination for each time mode (day/lunch/night).
For other incoming trunk call distribution methods, the priority of methods is as follows:

1. CLI Destination
2. DID/DDI
3. DIL

Important

FWD No Answer Time is used as the timer for forced call disconnection. For more information, refer to 3.1.9 Call Forwarding (FWD).

Conditions	Note
The call will be disconnected when an invalid destination number is set.	

PIL Reference
2.4.2 PBX Configuration—[3-2] Trunk—DIL

3.1.4 DISA (Direct Inward System Access)

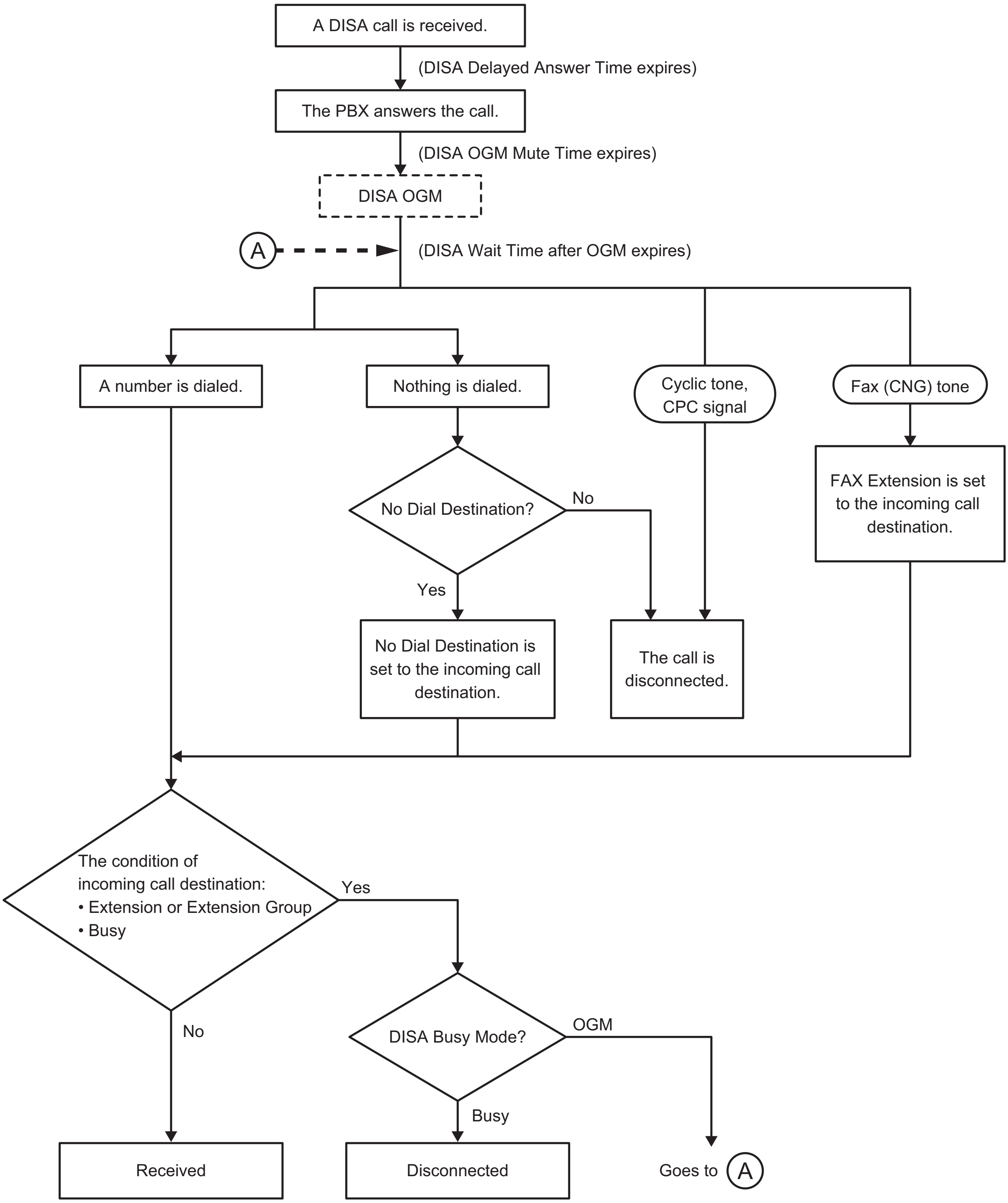
Description

A caller can access specific PBX features as if the caller is an SLT extension user in the PBX when the incoming call destination is a DISA floating extension number assigned to each DISA message. The caller can have direct access to features such as:

- Placing an intercom call to an extension or any floating extensions (e.g., Extension group)
- Calling an outside party via the PBX.
- Operating some PBX remote features (e.g., FWD)
- End of call detection

For the function flow regarding incoming calls to DISA, see [Flow Chart].

[Flow Chart]


A DISA call is received.
(DISA Delayed Answer Time expires)
The PBX answers the call.
(DISA OGM Mute Time expires)
DISA OGM
A
(DISA Wait Time after OGM expires)
A number is dialed.
Nothing is dialed.
Cyclic tone, CPC signal
Fax (CNG) tone
No Dial Destination?
No
Yes
FAX Extension is set to the incoming call destination.
No Dial Destination is set to the incoming call destination.
The call is disconnected.
The condition of incoming call destination:
• Extension or Extension Group
• Busy
Yes
No
DISA Busy Mode?
OGM
Busy
Received
Disconnected
Goes to A

Important

FWD No Answer Time is used as the timer for forced call disconnection. For more information, refer to 3.1.9 Call Forwarding (FWD).

Conditions	Note
Up to 10 messages can be recorded as guidance for automatic response.	Messages can be recorded for each DISA floating extension number.
Up to four simultaneous incoming calls are possible to a DISA floating extension number, and callers connected to DISA are able to hear guidance from the beginning independently.	
After DISA responds, the caller to DISA can call by dialing either during or after listening to the DISA guidance.	
After DISA responds, if a caller calls to the dialing destination, such as an extension or a trunk, the caller to DISA will hear Music on Hold or a Ring Back Tone according to the setting of Sound on DISA.	Refer to 3.7.3 Music on Hold
It is possible to make a call to a trunk after DISA responds. When doing so, it is necessary to enter the Walking COS feature number.	
If a call through DISA is routed to a trunk, DISA can be used to detect the end of the call.	The following three types of tone detection can be enabled for each trunk port to disconnect a trunk-to-trunk call via DISA. – **DISA Tone Detection Silence** – **DISA Tone Detection Continuous** – **DISA Tone Detection Cyclic**
If the caller does not dial any digits within a preprogrammed time period (**No Dial Intercept Timer (s)**) after DISA is received the call, the call is redirected to a preprogrammed destination (**No Dial Destination**). The following number cannot be set as **No Dial Destination**: • Trunk Line / Trunk Group, Idle Line Access (Local Access) - 1/2 (Refer to 2.1.1 Feature Number Lists for User) • Doorphone (Refer to 3.11.1 Doorphone Call) • Built-in VM (Refer to 5 Voice Mail Features)	• For **No Dial Destination**, each DISA can be set to virtual extensions. • **No Dial Destination** for each DISA floating extension number can be set.

PIL Reference
2.4.1 PBX Configuration—[3-1] Trunk—Port
2.4.5 PBX Configuration—[3-5] Trunk—DISA

3.1.5 DISA-AA (IVR)

Description

DISA-AA means DISA Automated Attendant.
After listening to the outgoing message (OGM), the caller may dial a single digit (DISA AA number). The destination for each DISA AA number can be assigned for each message.

Conditions	Note
The following number cannot be set as DISA-AA Destination: • Trunk Line / Trunk Group, Idle Line Access (Local Access) - 1/2 (Refer to 2.1.1 Feature Number Lists for User) • Doorphone (Refer to 3.11.1 Doorphone Call) • Built-in VM (Refer to 5 Voice Mail Features)	
It is possible to register destination numbers for 0 to 9 for each OGM.	
It is possible to make a call to an extension (including a floating extension number) by inputting a number set by **1 Digit AA Destination (Extension Number).**	In order to identify that it is **1 Digit AA Destination (Extension Number)**, do not dial any additional numbers until the 2nd digit input timer is timed-out which is set in **2nd Dial Timer for AA.** If the 2nd digit is input, the AA operation will be invalid.

PIL Reference
2.4.5 PBX Configuration—[3-5] Trunk—DISA

3.1.6 FAX Detection

Description

The PBX can distinguish between fax calls and other types of calls arriving on DISA lines, and automatically transfer fax calls to preprogrammed destinations. When a call arrives on a DISA line, an OGM is played.
At the same time, the PBX begins fax signal detection. If a fax signal is detected, the PBX recognizes that the call is a fax call, and transfers the call to the fax destination assigned to that OGM through system programming.
This allows a single trunk to be used seamlessly for both voice and fax calls, with only voice calls arriving at user extensions.

Conditions

Conditions	Note
The PBX can detect only a fax tone from an analog trunk.	
The following numbers cannot be set as FAX Destination: • Trunk Line / Trunk Group, Idle Line Access (Local Access) - 1/2 (Refer to 2.1.1 Feature Number Lists for User) • DISA (Refer to 3.1.4 DISA (Direct Inward System Access)) • Meet Me (Refer to 3.9.2 Meet-me conference) • Doorphone (Refer to 3.11.1 Doorphone Call) • Built-in VM (Refer to 5 Voice Mail Features)	
This feature is only effective for calls arriving on DISA lines.	
It is possible to register FAX destinations for each OGM. The destination settings are made the same way as for DISA-AA.	

PIL Reference
2.4.5 PBX Configuration—[3-5] Trunk—DISA

3.1.7 Caller ID

Description

The PBX receives caller information, such as the caller's name, telephone number, date and time, through the Trunk line. This information can then be shown on the phones with a display.

Conditions

[For SLT extensions]

Conditions	Note
• Signal type for SLT complies with ETSI (European Telecommunications Standards Institute)-type FSK and Bellcore-type FSK. • Visual Caller ID and DTMF Caller ID are not supported.	Caller ID notification to SLT is only conducted when receiving calls.
SLT displays the caller information sent from a trunk or the extension. Displayed information is depended on a phone. The caller information is sent to the extension as follows: • When the PBX receives the name (max. 16 characters), telephone number (max. 20 digits) or indication reason information from the carrier by an analog trunk, it sends that information to an extension as it is. • The date and time information of the PBX is sent as the caller information.	• The name in the caller information uses the character code IA5. • Through Web Maintenance Console, the Installer can set whether each SLT receives the caller information.

[For all extensions]

Conditions	Note
Analog trunks should be provided with a Caller ID notification service by a carrier.	Through Web Maintenance Console, the Installer can set whether the PBX receives the caller ID from a carrier.
When a Caller ID is received from a trunk, the PBX searches System Speed Dialing for the modified Caller ID. If a name corresponding to the modified Caller ID is registered, the name information of the caller (Display Name) registered in System Speed Dialing is sent to the called extension. Otherwise, the name in the caller information is sent.	Reference: • 3.1.1 Calling Line Identification (CLI) Distribution & Call Block • 3.3.1 System Speed Dial
When receiving an intercom call, the extension name and number of the caller are sent to the extension.	
When Transfer Recall is operated, “RCL:” is added to the head of a Caller ID. And the Caller ID is notified to the operated extension.	
Even if the caller's name is sent, the name may not be shown depending on the type of extension.	

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port—Analog Extension
2.4.1 PBX Configuration—[3-1] Trunk—Port—Analog - Basic
2.4.6 PBX Configuration—[3-6] Trunk—Analog CO Property

3.1.8 Call Pickup

Description

An extension user can answer a call ringing at any other extension.
There are two types of Call pickup (Directed/Group).

Operation

[Directed call pickup]

- An extension user can answer a call ringing at a specific other extension by entering the Directed Call Pickup feature number. (Refer to 2.1.1 Feature Number Lists for User)
- By pressing a DSS key indicating a call is being received, the user can pick up that call on the specified extension. (Refer to 4.1 DSS Key)
- By pressing a Single-CO key that indicates “Receiving a call”, the user can pick up that call at the Single-CO Key. (Refer to 4.2 Single-CO Key)

[Group call pickup]
An extension user can answer a call to an extension in the specific extension group ringing at another extension by entering the Group Call Pickup feature number. Also, an extension user can answer a call to an extension in the own extension group ringing by entering the Group Call Pickup feature number and [#]. (Refer to 2.1.1 Feature Number Lists for User)

Conditions

Conditions	Note
Call Pickup applies to: Intercom calls, trunk calls, and doorphone calls.	Reference: 4.1 DSS Key 3.11.1 Doorphone Call
Call Pickup Deny: An extension user can prevent other extensions from picking up calls ringing at his or her own extension. This feature can set through the extension port setting. If this feature is enabled, other users will hear a busy tone when trying to pick up calls.	
Calls from Hold Recall and Transfer Recall cannot be picked up with this feature.	

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port

3.1.9 Call Forwarding (FWD)

Description

Extensions and extension groups can forward their calls to preset destinations. There are several different types of forwarding, and the circumstances under which the calls are forwarded for each type differ as follows:
[FWD ALL] Any time
[FWD Busy] When extensions or extension groups user's line is busy.
[FWD NA (No Answer)] When extensions or extension groups user does not answer within a preprogrammed time.
[FWD Busy/NA] When extensions or extension groups user's line is busy or the user does not answer within a preprogrammed time.

Important

If FWD setting is OFF, and the extension user doesn't answer the incoming call within the pre-programmed time (FWD No Answer Time), the call will be disconnected forcefully.
If the incoming call is from an analog trunk, the caller is charged.

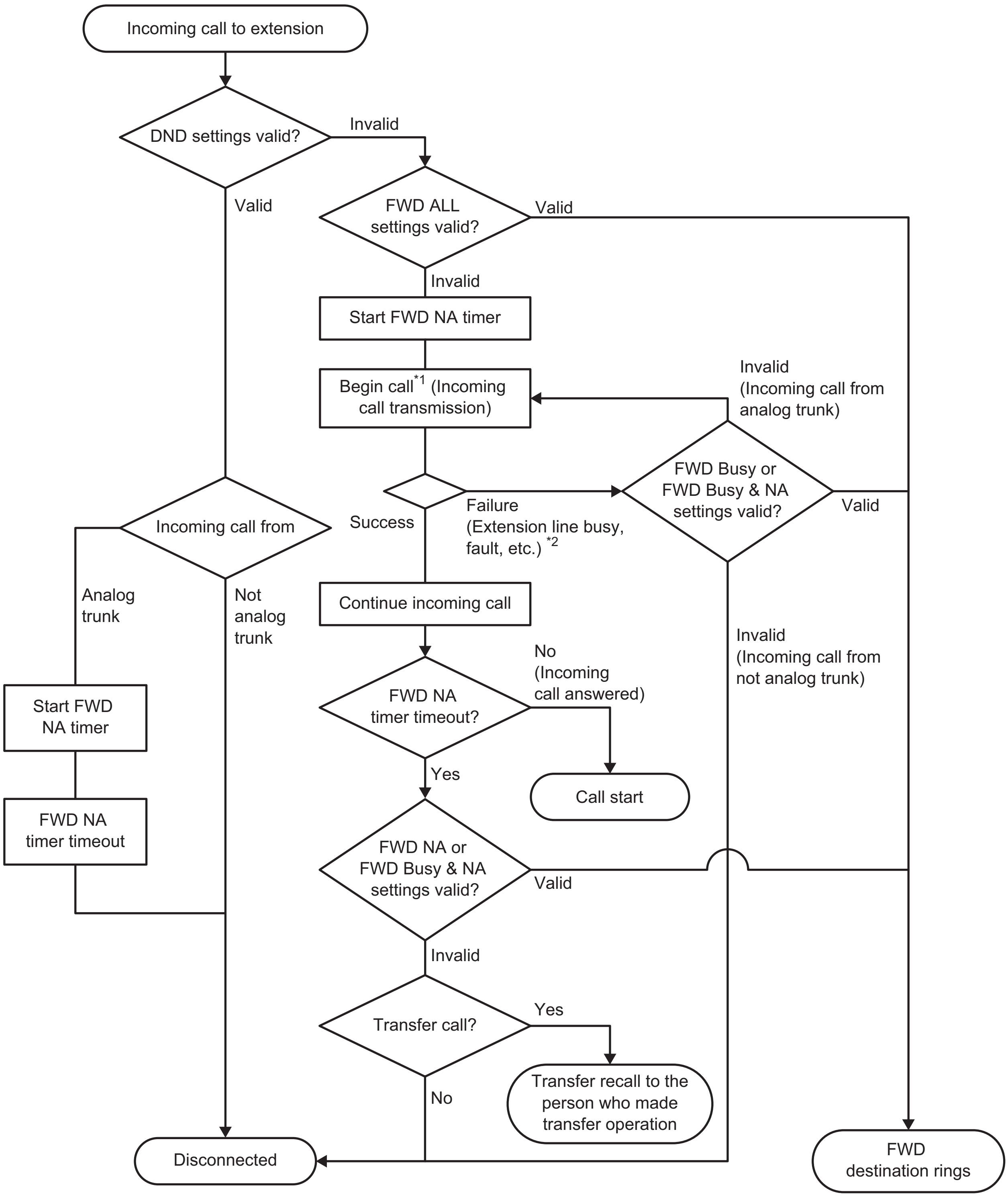
As explained above, it is recommended to set the item long enough to avoid as many charges to the caller as possible when the FWD setting is OFF and an incoming call is received from an analog trunk. For more information, refer to "2.3.2 PBX Configuration—[2-2] Extension—Phone-FWD/DND－FWD No Answer Time" in the Programming Item List.

You can also prevent the call from being forcefully disconnected by setting an FWD destination. To allow the caller to leave a message, it is recommended to set the FWD destination below to “Voice Mail”. For more information, refer to "2.3.2 PBX Configuration—[2-2] Extension—Phone-FWD/DND－Destination Number" in the Programming Item List.

If you set your e-mail address to the item below, you can also receive an e-mail with the recorded message attached. For more information, refer to "2.3.2 PBX Configuration—[2-2] Extension—Phone-Voice Mail－Send Email when Message is left" in the Programming Item List.

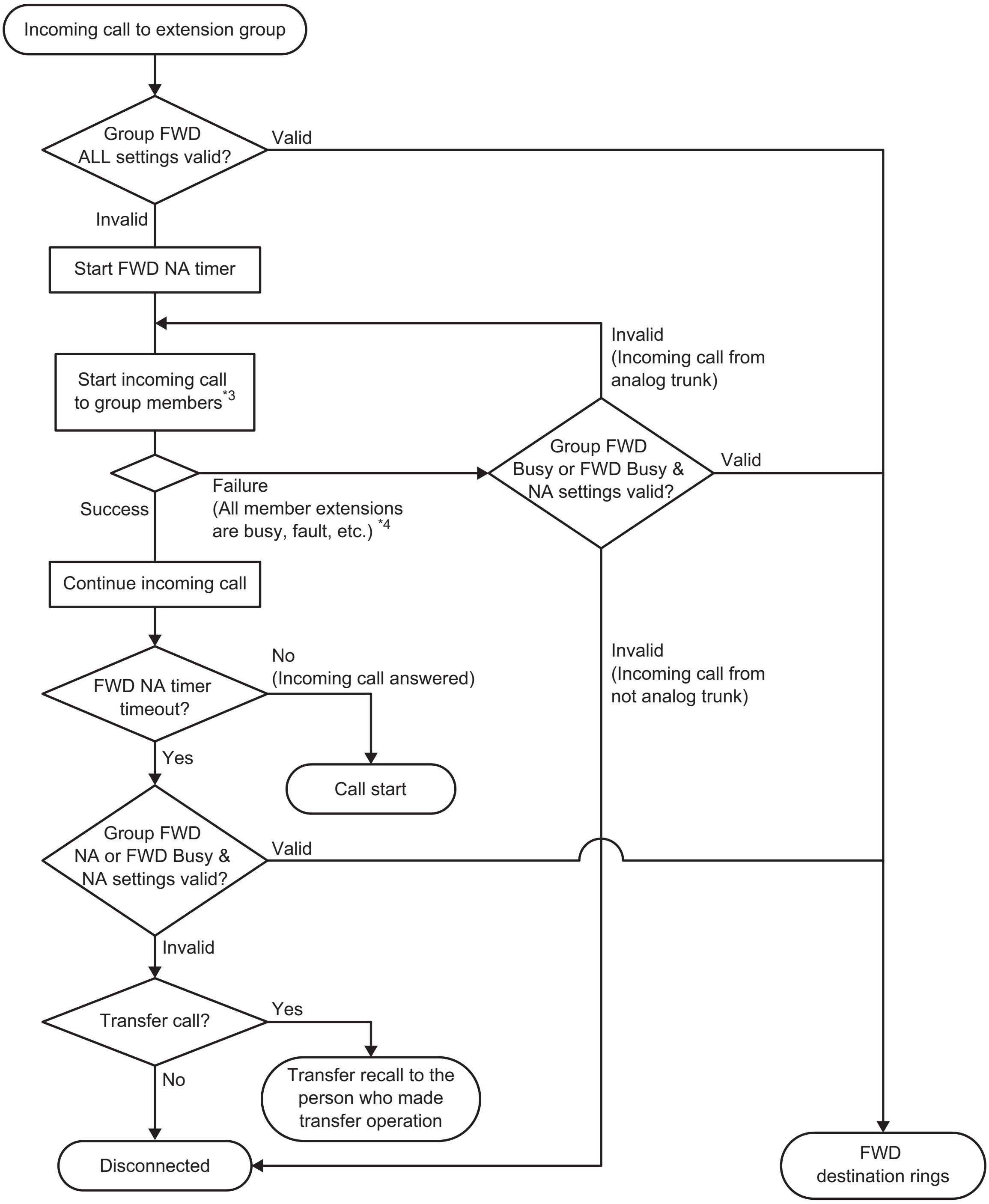
[Flow chart]

For incoming call to extension:



*1: Includes Call Waiting incoming calls.
*2: Goes into DISA Busy mode at time of incoming DISA call.

For incoming call to extension group:



*3: Includes incoming Call Waiting calls of extension members. However, extensions with DND settings will not receive the call.
All members will be rung and the NA timer will start at the time of incoming Hunting group calls.

*4: Call will fail and be disconnected when all members' incoming calls fail at the time of incoming Hunting group calls.
The operation at time of an incoming DISA call follows the settings of DISA Busy Mode (disconnect or transfer to other DISA incoming call).

Operation

[FWD Set/Cancel (For extensions only)]

An extension user can Set/Cancel call forwarding by a feature number.
(Refer to 2.1.1 Feature Number Lists for User)

When setting is complete, a confirmation tone is heard.

Conditions

Conditions	Note
A call can only be automatically forwarded one time. (Except FWD NA) **Example:** If Extension A's FWD settings are as follows, when Extension A receives a call, only Extension B will receive the call (it will not be forwarded to Extension C): • Extension A's FWD settings are FWD ALL, and the forwarding destination is set to Extension B. • Extension B's FWD settings are FWD ALL or FWD Busy, and the forwarding destination is set to Extension C.	
For extension groups, FWD settings can only be programmed in the Web Maintenance Console.	Programming Item List 2.3.4 PBX Configuration—[2-4] Extension—Extension Group—Group Setting
FWD destination can be selected in Web Maintenance Console (Include User level account). When an extension user dials the feature number and changes the FWD destination number, the FWD destination number set as **Other** is changed.	Programming Item List 2.2.1 PBX Configuration-[2-2]Extension-Phone-FWD/DND-Destination Number **Phone(Home)** **Phone(Mobile1)** **Phone(Mobile2)** **Voice Mail** **Other**
A call received at an extension via Extension Group cannot be forwarded.	
A call received by Hold Recall/Transfer Recall cannot be forwarded.	
A call received from a doorphone cannot be forwarded.	Refer to 3.11.1 Doorphone Call
A doorphone cannot be set as a forward destination.	Refer to 3.11.1 Doorphone Call
A Paging feature number cannot be set as a forward destination.	
An extension which has Extension Dial Lock set cannot change FWD settings from a telephone.	However, the FWD settings can be changed through Web Maintenance Console.
A different FWD No Answer time can be set for each Time Service (Day /Lunch/Night) of an Extension/Extension Group.	
If the transfer destination does not answer within the pre-programmed time (**FWD No Answer Time**), either FWD NA or Transfer Recall will operate. When the call is forwarded, you can select whether to prioritize going to FWD NA or Transfer Recall.	Refer to 3.8.2 Call Transfer-Unscreened
The FWD to trunk setting can be set independently for each extension port. Do not set an outside telephone number as the FWD destination, when the FWD to trunk setting is disabled.	Programming Item List 2.3.1 PBX Configuration—[2-1] Extension—Port—Main—Call Forward to CO

Conditions	Note
Call Waiting is preferred to FWD (Busy).	

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port
2.3.2 PBX Configuration—[2-2] Extension—Phone
2.3.4 PBX Configuration—[2-4] Extension—Extension Group

3.1.10 Do Not Disturb (DND)

Description

An extension user can make use of the DND feature. If this feature is set, calls will not arrive at the extension.

Operation

[DND Set/Cancel]
An extension user can Set/Cancel DND by a feature number.
(Refer to 2.1.1 Feature Number Lists for User)
When setting is complete, a confirmation tone is heard, and then a dial tone is heard.

Conditions

Conditions	Note
DND operations can only be used for incoming calls (including via DISA) from SIP trunks or extensions.	
DND settings are not applied to incoming calls by Hold Recall.	
DND settings are not applied to incoming calls from doorphones.	Refer to 3.11.1 Doorphone Call
If the call through an analog trunk is received, the pre-programmed timer (**FWD No Answer Time**) will start. If the timer expires, the call will be disconnected. **Important** As the call is disconnected immediately after it is answered by an extension, the caller may be charged. It is recommended to set the item below long enough to avoid as many charges to the caller as possible. For more information, refer to "2.3.2 PBX Configuration—[2-2] Extension—Phone-FWD/DND-FWD No Answer Time" in the Programming Item List.	
Either DND or FWD can be set.	Refer to 3.1.9 Call Forwarding (FWD)

PIL Reference
2.3.2 PBX Configuration—[2-2] Extension—Phone

3.1.11 Distinctive Ringtone

Description

It is possible to select the type of ring tone pattern that arrives at an extension for each type of incoming call, etc. This feature is only available for SLT and KX-HDV series SIP phones.

Conditions

Conditions	Note
Sets the ring tone patterns (Single/Double) for incoming trunk calls or incoming intercom calls (including doorphones) by System Options.	
If a trunk call is forwarded, the ring tone pattern for incoming intercom calls is heard at the forwarded extension.	

PIL Reference
2.2.6 PBX Configuration—[1-6] System—System Options

3.1.12 Extension Group Call

Description

An extension group is a group of extensions programmed through system programming.
An extension group receives calls directed to the group. Each extension group has a floating extension number (default: 6 + two-digit group number [up to group 16]).
Following distribution methods of incoming group calls can be selected in the Web Maintenance Console:

Distribution method	Description
Ring group	All extensions in the Extension Group ring simultaneously. **Note** If all extensions in the extension group are busy, the call will be disconnected.
Hunting group	One extension in Extension Group rings. If a call received at an extension is not answered within a pre-programmed time period (**FWD No Answer Time**), the call will be redirected to the next member extension. Extensions are hunted in a circular way in the preprogrammed order for the group, starting at the extension after the extension that received the last call. If none of the extensions in the group answer, the caller will hear a busy tone. (Refer to [Flow chart] in 3.1.9 Call Forwarding (FWD))

Operation

By calling to the floating extension number of Extension Group, it is possible to call extension members belonging to the Extension Group.

Conditions

Conditions	Note
It is possible to create a maximum of 16 Extension Groups. Up to 24 extensions can belong to one Extension Group.	One extension can belong to multiple Extension Groups.

Conditions	Note
The extension will be received an Extension Group call when the status of the extension is as follows: • Idle • Receiving a call (Only when Call Waiting is enabled) • FWD All is enabled (Only Hunting group is selected)	When the extension is received an Extension Group call, the FWD setting for the extension is ignored. Refer to 3.1.9 Call Forwarding (FWD)
When FWD is set in an Extension Group, calls follow the FWD setting.	If call distribution is selected for Hunting group, Call Forwarding is operated when all extensions in Extension Group cannot receive a call. (Only when FWD Busy/NA is enabled)
For the following features, the extension settings registered at the head of the Extension Group are applied to the Extension Group: • FWD to trunk settings: Permission to FWD to a trunk. • TRS settings: Trunk call restriction.	Reference: • 3.1.9 Call Forwarding (FWD) • 3.4 Toll Restriction (TRS)/Call Barring (Barring) Features
Extension Groups are used for Call Pickup and Paging feature.	Reference: • 3.1.8 Call Pickup • 3.10.1 Paging

PIL Reference
2.3.4 PBX Configuration—[2-4] Extension—Extension Group

3.1.13 Calling Party Control (CPC) Signal Detection

Description

The Calling Party Control (CPC) signal is an on-hook indication (disconnect signal) sent from the analog Trunk line when the other party hangs up. To maintain efficient utilization of Trunk lines, the PBX monitors their state and when CPC signal is detected from a line, it disconnects the line.

Conditions

Conditions	Note
CPC signal detection is programmable for incoming Trunk line calls, and for outgoing Trunk line calls.	
If a CPC signal is detected during a Conference call, that line is disconnected, but the remaining parties stay connected.	
If a CPC signal is detected during a call between a caller using the DISA feature and an extension or an outside party, the line is disconnected.	
For CPC signal detection, enabling of detection and detection time can be set for both the outgoing CPC signal and incoming CPC signal. • **CPC Signal Detection Time Outgoing** • **CPC Signal Detection Time Incoming**	

PIL Reference
2.4.1 PBX Configuration—[3-1] Trunk—Port—Analog - CPC Detection

3.2 Making Call Features

3.2.1 Emergency Call

Description

An extension user can dial preprogrammed emergency numbers after seizing a Trunk line regardless of the restrictions imposed on the extension.

The emergency number "911" is treated as an exception in the U.S.A. as follows:
[USA Only]

- Calls can be made to "911" without entering a Trunk Access number.
- "911" has already been registered in the Emergency Dial. (Not changeable)

Conditions

Conditions	Note
Emergency numbers may be called, regardless of Toll Restriction (TRS) level.	Refer to 3.4.1 Toll Restriction (TRS)

PIL Reference
2.5.4 PBX Configuration—[4-4] TRS/ARS—Emergency Dial—Emergency Dial

3.2.2 Line Access, Automatic

Description

An extension user can select an idle trunk automatically by dialing the Idle Line Access (Local Line Access) number.

Operation

Dial the Idle Line Access (Local Line Access) number, and then dial the number to be called.
(Refer to 2.1.1 Feature Number Lists for User)

Conditions

Conditions	Note
When multiple trunks are available, set the **Trunk Access number** priority in the Trunk Access Priority setting in system programming. Priority setting can be select as follows: • **CO-8 to CO-1** • **CO-1 to CO-8** • **Rotation**: Use the trunk with secondary priority following the trunk used in the previous call.	
Each trunk can be disabled by setting **Attribution** to **No Connect** in Web Maintenance Console.	

Conditions	Note
Extension users can make an outside call without entering the Idle Line Access (Local Line Access) number in the following case: The Idle Line Access (Local Line Access) number is not set in the system, and the outside telephone number does not match the registered extension number or feature number.	See 3.5.1 ARS
In the Web Maintenance Console setting, the installer can specify the Idle Line Access number to be used for each extension.	Refer to 2.3.1 PBX Configuration—[2-1] Extension—Port-1.Port—Main-Trunk Call Block in Programming Item List.

PIL Reference
2.2.6 PBX Configuration—[1-6] System—System Options
2.4.1 PBX Configuration—[3-1] Trunk—Port

3.2.3 Line Access, Trunk Group

Description

An extension user can select an idle trunk line from the corresponding Trunk Group by dialing the Trunk Group Access number and a Trunk Group number.

Operation

Dial the Trunk Group Access number, specify a trunk group number to seize a trunk, and then dial the number to be called.
(Refer to 2.1.1 Feature Number Lists for User)

Conditions

Conditions	Note
Each trunk can be disabled by setting **Attribution** to **No Connect** in Web Maintenance Console.	
In the Web Maintenance Console setting, the installer can specify the Idle Line Access number to be used for each extension.	Refer to 2.3.1 PBX Configuration—[2-1] Extension—Port-1.Port—Main-Trunk Call Block in Programming Item List

PIL Reference
2.4.1 PBX Configuration—[3-1] Trunk—Port

3.2.4 Line Access, Trunk Line

Description

An extension user can select a Trunk line directly by dialing the Trunk Line Access number and a Trunk line number.

Operation

Dial the Trunk Line Access number, specify a Trunk line number, and then dial the number to be called.
(Refer to 2.1.1 Feature Number Lists for User)

For KX-HDV230 SIP phone:
Press the S-CO key (that indicates "Idle" [LED is OFF]), and then dial the number to be called. (Refer to 4.2 Single-CO Key)

Conditions

Conditions	Note
Trunk line numbers can be referred on a trunk port basis.	
Each trunk can be disabled by setting **Attribution** to **No Connect** in Web Maintenance Console.	
In the Web Maintenance Console setting, the installer can specify the Idle Line Access number to be used for each extension.	Refer to 2.3.1 PBX Configuration—[2-1] Extension—Port-1.Port—Main-Trunk Call Block in Programming Item List.

PIL Reference
2.4.1 PBX Configuration—[3-1] Trunk—Port

3.2.5 Calling Line Identification Presentation (CLIP)

Description

When calling to a SIP Trunk line, notifies a CLIP number set for each extension as the caller's number.
It is possible to notify the name information set for each extension as well.

Conditions

Conditions	Note
When executing Walking COS, a CLIP set for the extension specified by Walking COS is used.	Refer to 3.4.2 Walking COS (TRS level)
When a CLIP number is not set for a calling extension, the Subscriber Number for each SIP Carrier will be notified.	
When an outgoing call to an extension is forwarded to a trunk, the CLIP and the name of the caller is notified to the trunk.	
[For Call Forwarding feature] • When incoming calls from a SIP carrier are forwarded to same SIP carrier, the PBX selects whether the Caller ID of the calling party, or the CLIP number and the name of the forwarding extension is sent to the forward destination. • When incoming calls from a SIP carrier are forwarded to the other SIP carrier, the CLIP number and the name of the forwarding extension is sent to the forward destination.	Refer to 2.4.7 PBX Configuration—[3-7] Trunk—SIP Trunk Property – Send CLIP of caller when call from SIP carrier is forwarded to same SIP carrier in Programming Item List.

Conditions	Note
[For Call Transfer-Blind feature] • When SIP trunk calls are transferred to same SIP carrier using the Call Transfer-Blind feature, the PBX selects whether the Caller ID of the calling party, or the CLIP number and the name of the transferring extension is sent to the transfer destination. • When SIP trunk calls are transferred to the other SIP carrier using the Call Transfer-Blind feature, the CLIP number and the name of the transferring extension is sent to the transfer destination.	Refer to 2.4.7 PBX Configuration—[3-7] Trunk—SIP Trunk Property – Send CLIP of caller when call from SIP carrier is forwarded to same SIP carrier in Programming Item List.

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port
2.4.7 PBX Configuration—[3-7] Trunk—SIP Trunk Property

3.2.6 Intercom Calling

Description

An extension user can call another extension user.

About the floating extension number
Virtual extension numbers can be assigned to resources to make them appear as extensions. This feature is also known as Floating Station. These numbers are defined as floating extension numbers and can be assigned as a group extension number etc. Floating extension numbers can be dialed by an extension user.

Important

FWD No Answer Time is used as the timer for forced call disconnection. For more information, refer to 3.1.9 Call Forwarding (FWD).

Conditions

Conditions	Note
Extension numbers and names are assigned to all extensions.	
After dialing an extension number, the following tone is heard on the caller side. • Ringback tone: Indicates a connection is being made to the called extension • Busy tone: Indicates the called extension is busy / Indicates the dialed extension number is invalid	
When a call is received at an extension, the extension number and name which made the call are indicated at the extension which receives the call.	
Through system settings, it is possible to make an intercom call by pressing the DSS key without going off-hook.	Refer to 4.1 DSS Key

PIL Reference
2.2.5 PBX Configuration—[1-5] System—Timers
2.3.1 PBX Configuration—[2-1] Extension—Port
2.3.3 PBX Configuration—[2-3] Extension—Flexible Buttons

3.2.7 Redial

Description

Every extension automatically saves recently dialed external telephone numbers to allow the same number to be dialed again easily.

Operation

For all Extensions:
Enter the redial feature number while a dial tone is heard.
(Refer to 2.1.1 Feature Number Lists for User)
For KX-HDV series SIP phones:
Press the [REDIAL] button while a dial tone is heard.

Conditions

Conditions	Note
It is possible to redial a maximum of 32 digit dial numbers.	
Numbers dialed after starting the conversation are not stored in the redial memory.	
When executing Walking COS, the extension being operated will store the redial memory.	

3.2.8 Video Communication for Intercom Call

Description

It is possible to make a video call between 2 SIP extensions.
Devices that support video communication are listed below:

- KX-NTV series
- KX-HDV430
- General SIP phones
- Mobile Softphone (KX-UCMA)

Operation

SIP extensions can make a video call only when a call is started with an extension using the following features:

- 3.2.6 Intercom Calling
- 3.2.7 Redial
- 3.1.9 Call Forwarding (FWD)
- 3.1.12 Extension Group Call
- 3.3.1 System Speed Dial
- 3.6.1 Call Waiting

Conditions

Conditions	Note
SIP extension users cannot make a video call if they make a call using the following features (audio call only): • 3.1.4 DISA (Direct Inward System Access) • 3.1.8 Call Pickup • 3.9 Conference Features • 3.10 Paging Features • 5 Voice Mail Features	

Conditions	Note
SIP extensions cannot use the following features during a video call: • Video to Audio/Audio to Video switching (Refer to the manual for your telephone.) • 3.7 Holding Features • 3.8 Transferring Features	

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port—SIP Extension

3.3 Memory Dialing Features

3.3.1 System Speed Dial

Description

An extension user can make calls using abbreviated dialing for frequently dialed numbers which are stored in the PBX system data.

Operation

Enter the System Speed Dialing feature number and the System Speed Dialing number while hearing a dial tone.
(Refer to 2.1.1 Feature Number Lists for User)

Conditions

Conditions	Note
A maximum of 200 System Speed Dialing numbers (e.g., telephone numbers, feature numbers) can be programmed in Web Maintenance Console.	
Making System Speed Dialing calls can be restricted by TRS settings.	Refer to 3.4.1 Toll Restriction (TRS)
It is not possible to register a Quick Dial number as the dialing destination of System Speed Dialing.	Refer to 3.3.2 Quick Dial
It is not possible to call a phone number which is set as "Disconnect" in CLI destination by using System Speed Dial (Possible using ordinary dialing).	Refer to 3.1.1 Calling Line Identification (CLI) Distribution & Call Block
Extension number and feature number can be registered in System Speed Dial.	
The System Speed Dial data can be transferred to the KX-HDV230 terminal and used as the phonebook of the terminal.	Refer to the KX-HDV series manual for operations of the terminal.

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port
2.6 PBX Configuration—[5] System Speed Dialing—System Speed Dialing

3.3.2 Quick Dial

Description

An extension user can access an extension or feature by simply dialing a 1-2 digit Quick Dialing number.

Operation

Enter a 1-2 digit Quick Dialing number, while hearing a dial tone.

Conditions

Conditions	Note
Up to 20 Quick Dialing numbers can be assigned in Web Maintenance Console.	
It is possible to register a System Speed Dial as a Quick Dialing destination (Phone Number).	
It is not possible to register a Quick Dial number as the Quick Dialing destination (Phone Number).	

PIL Reference
2.2.4 PBX Configuration—[1-4] System—Numbering Plan

3.4 Toll Restriction (TRS)/Call Barring (Barring) Features

3.4.1 Toll Restriction (TRS)

Description

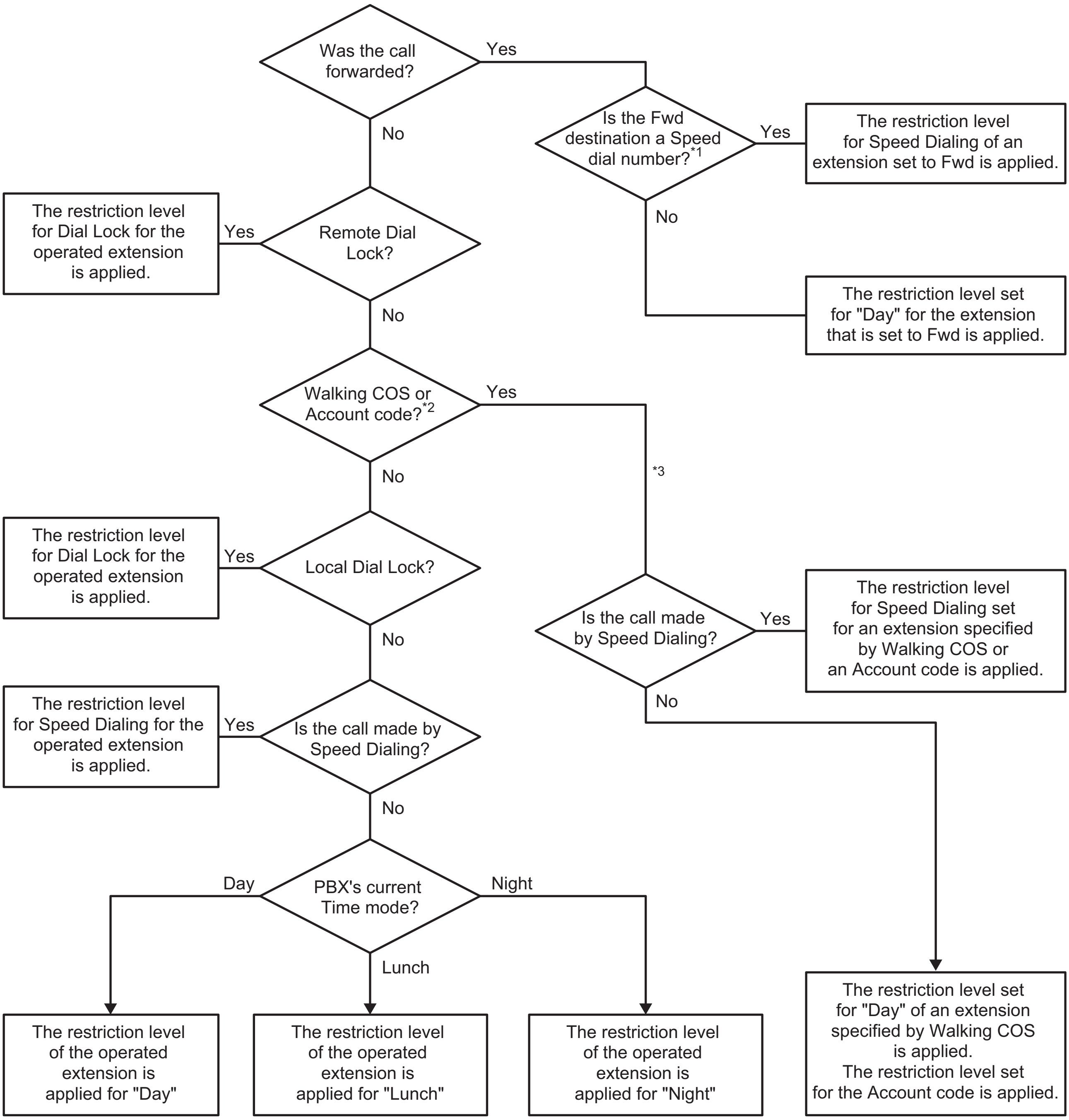
TRS can prohibit an extension user from making certain trunk calls by COS programming. It is applied when the user goes off-hook, a Trunk line is seized and then a dialed number is sent to the Trunk line.
Each TRS is programmed to have a level for each time mode (day/lunch/night).
There are five levels available. Each levels are used to restrict calls.

Leading Digits table
In the Leading Digits table, the installer can set the leading digits to be restricted for each TRS level.
An outgoing trunk call made by an extension with a TRS level is first checked against the Leading Digits table (longest match searching). If the leading digits of the dialed number (not including the Trunk Access number) are found in the table, the call is restricted (made or disconnected).

TRS Override by System Speed Dialing
If the call is made using System Speed Dialing, the call can override the TRS. Each extension is programmed to have a TRS level for System Speed Dialing. This feature permits all extension users to make System Speed Dialing calls with the level for System Speed Dialing.

Allows or denies outgoing trunk calls according to Toll Restriction Level as decided in the following Flow chart:

[Flow chart]



*1 Extension Dial Lock is not applied to call restriction level judgments during Fwd.

*2 It is also possible to specify an operated extension for operation where a local Dial Lock is constantly applied.

*3 The restriction level of the head member extension is applied during Fwd from the Extension Group.

Conditions

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port
2.3.2 PBX Configuration—[2-2] Extension—Phone

PIL Reference
2.5 PBX Configuration—[4] TRS/ARS—Leading digits
2.5 PBX Configuration—[4] TRS/ARS—Account code
2.5 PBX Configuration—[4] TRS/ARS—Options

3.4.2 Walking COS (TRS level)

Description

A user can enter his extension number and extension personal identification number (PIN) at another extension, to make the call using his TRS level. (Refer to 3.4.1 Toll Restriction (TRS))

Operation

Dial the Walking COS feature number, enter an extension number and PIN, and then seize a trunk to call. (Refer to 2.1.1 Feature Number Lists for User)

Conditions

Conditions	Note
There is a risk that fraudulent telephone calls will be made if a third party discovers a personal identification number (PIN) (account code or extension PIN) of the PBX. The cost of such calls will be billed to the owner/renter of the PBX. To protect the PBX from this kind of fraudulent use, we strongly recommend: **a.** Keeping PINs secret. **b.** Selecting complex, random PINs that cannot be easily guessed. **c.** Changing PINs regularly.	Refer to 3.4.5 Extension Dial Lock
It is possible to set the applicability of this feature for each extension. When set as not applicable, a call is disconnected even if using Walking COS.	
It is possible to set the applicability regarding the Walking COS operation through DISA for each extension.	
An extension user can use Quick Dial/System Speed Dial, even when making a trunk call using Walking COS.	Refer to 3.3 Memory Dialing Features
If an Extension PIN is not set, an extension user cannot use Walking COS.	

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port
2.3.2 PBX Configuration—[2-2] Extension—Phone

3.4.3 Account Code Entry

Description

An account code is used to identify outgoing trunk calls for accounting and billing purposes. The account code is appended to the CDR call record.

Operation

Enter the Account Code Entry feature number, enter the account code, and then seize a trunk to make a call.
(Refer to 2.1.1 Feature Number Lists for User)

Conditions

Conditions	Note
There is a risk that fraudulent telephone calls will be made if a third party discovers an account code of the PBX. The cost of such calls will be billed to the owner/renter of the PBX. To protect the PBX from this kind of fraudulent use, we strongly recommend: **a.** Keeping account codes secret. **b.** Selecting complex, random account codes that cannot be easily guessed. **c.** Changing account codes regularly.	
An extension user can use Quick Dial/System Speed Dial, even when making a trunk call using an account code.	
If the entered account code does not match an entry in account code list, an extension user cannot make a call.	

PIL Reference
2.5.3 PBX Configuration—[4-3] TRS/ARS—Account Code

3.4.4 Limited Call Duration

Description

During a conversation, when the system time limit expires the call will be disconnected automatically.
This feature applies to Trunk-to-Trunk calls. Also, this feature can applies to Extension-to-Extension/Trunk calls by the system settings.

Conditions

Conditions	Note
This feature is not applied, when the other party is below: • DISA (Refer to 3.1.4 DISA (Direct Inward System Access)) • Meet Me (Refer to 3.9.2 Meet-me conference) • Built-in VM (Refer to 5 Voice Mail Features) • Paging (Refer to 3.10 Paging Features)	

Conditions	Note
Call Duration timers can be programmed individually as follows: (1). **Extension-CO/Extension Duration Time (x60s)** (2). **CO-CO Duration Time (x60s)**	• While the timer in (1) is in operation, if the communication changes to a Trunk-to-Trunk call, the timer in (1) continues and the timer in (2) starts to operate simultaneously. • The timer in (1) can be set to none (no limit). • The timer in (2) does not start when the incoming or outgoing trunk is a SIP trunk.
Once a call duration timer starts, it continues even if the call is on hold, the held call is retrieved by another extension, or transferred to another extension.	If the call is parked, or the call is held by pressing the S-CO key, a call duration timer is stopped. After the call is retrieved, the timer is reset. (Refer to 3.7 Holding Features)

PIL Reference
2.2.5 PBX Configuration—[1-5] System—Timers
2.3.1 PBX Configuration—[2-1] Extension—Port

3.4.5 Extension Dial Lock

Description

[Local Dial Lock]
Executes Dial Lock (Local Dial Lock) for the extension using a PIN for each extension. It is possible to set individually for each user using the feature number.
By the user operates Dial Lock, the Operating extension can be set to the pre-programmed TRS level. (Refer to 3.4.1 Toll Restriction (TRS))
Using this method, an outgoing call cannot be made from that extension.
[Remote Dial Lock]
It is possible for Installer to execute Dial Lock (Remote Dial Lock) for other extensions through Web Maintenance Console.

Operation

[Extension PIN Set / Cancel]
Dial the Extension PIN Set / Cancel feature number and enter a predetermined parameter. The extension PIN can be set 4-10 digits. After hearing a confirmation tone, Set / Cancel the Extension PIN.
(Refer to 2.1.1 Feature Number Lists for User)

[Extension Dial Lock Set/Cancel]
Dial the Extension Dial Lock Set / Cancel feature number and enter a predetermined parameter. After hearing a confirmation tone, Set / Cancel Extension Dial Lock.
(Refer to 2.1.1 Feature Number Lists for User)

Conditions

Conditions	Note
Extension Dial Lock (Local Dial Lock) can be set for each phone through Web Maintenance Console. (Including User level account)	
Dial Lock (Remote Dial Lock) for other extensions by Installer is set through Web Maintenance Console for each extension. An extension user cannot unlock an extension that is set Remote Extension Lock.	Programming Item List 2.3.1 PBX Configuration—[2-1] Extension—Port—Main—Remote Dial Lock
As Extension PIN is not able to overwrite a setting, if an extension PIN has already been set by an Extension PIN Set operation, when trying to enter the extension PIN, the call will be disconnected.	To Change the Extension PIN, the extension user must cancel the Extension PIN beforehand.
When entering the PIN for confirmation in an Extension PIN Set operation, if the extension PIN is different from the previous one, the call will be disconnected after entering the extension PIN for confirmation has been completed.	
If the extension PIN is incorrectly entered for an Extension PIN Cancel operation or an Extension Dial Lock Cancel operation, the call will be disconnected.	

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port
2.3.2 PBX Configuration—[2-2] Extension—Phone

3.5 Automatic Route Selection (ARS) Features

3.5.1 ARS

Description

ARS automatically selects the carrier available at the time an outgoing Trunk line call is made according to preprogrammed settings. The dialed number will be checked and modified to connect the appropriate carrier.

Leading Digits table
In this table, an ARS Carrier to be used is assigned to each set of leading digits (e.g. area code).
If the dialed number matches the set of leading digits in the table, a call will be made to a trunk using the specified ARS Carrier.
The ARS Carrier is used according to the order of priority (Priority1-3). If the specified ARS Carrier is busy, the next highest priority ranked carrier will be used.
This table is also used as the TRS leading digits table (Refer to 3.4.1 Toll Restriction (TRS))

[Example]
Leading Digits table setting:

No.	Leading Digits	TRS Level (COS)			ARS Carrier		
		1	...	5	Priority-1	Priority-2	Priority-3
1	092	Allow	...	Allow	1: A telecom	2: B telecom	3: C telecom
2	001201NPX3333	Deny	...	Deny	:	:	:

No.	Leading Digits	TRS Level (COS)			ARS Carrier		
		1	...	5	Priority-1	Priority-2	Priority-3
:	:	:	...	:	:	:	:

If the dialed number is 0921234567, "A telecom" is selected as the ARS Carrier.

ARS Carrier Table
This table allows you to set the following settings for each ARS Carrier:

- Modification method for the dialed number.
- Assigning the Trunk Group to be used

[Example]
ARS Carrier table setting:

No.	Carrier Name	Dial Modification		Trunk Group
		Remove	Add	
1	A telecom	3 digits	0077	TRG1
2	B telecom	0 digit	0088	TRG2
3	C telecom	3 digits		TRG1
:		:	:	:

If the dialed number is 0921234567 and "A telecom" is selected as the "ARS Carrier", the dialed number will be modified to 00771234567 (Remove the first 3 digits and add "0077").
After the number is modified, the new number will be sent to the trunk using Trunk Group 1.

Condition

Conditions	Note
When an extension user specifies the Trunk Group number to make a call to a trunk, the specified Trunk Group number must match the Trunk group assigned to the selected ARS Carrier.	Refer to 3.2.3 Line Access, Trunk Group
When an extension user specifies the trunk line number to make a call to a trunk, the specified trunk line number must belong to the Trunk group assigned to the selected ARS Carrier.	Refer to 3.2.4 Line Access, Trunk Line

PIL Reference
2.5.1 PBX Configuration—[4-1] TRS/ARS—Leading Digits
2.5.2 PBX Configuration—[4-2] TRS/ARS—ARS Carrier

3.6 Busy Line/Busy Party Features

3.6.1 Call Waiting

Description

The busy extension user can answer the second call (including Hold Recall) by disconnecting the current call or placing it on hold. This feature is not available for SLT extensions.

Operation

[Disconnecting the current call]
The current call is disconnected by going on-hook during a conversation and if a call rings, go off-hook.

[Placing the current call on hold]
For All Extensions:
Execute the call holding operation (Refer to 3.7.1 Call Hold). Then go on-hook, and go off-hook again.
For KX-HDV series SIP phones:
Press the DN key or the S-CO key on an extension which displays incoming calls during a conversation. (Refer to 4.4 DN Key and 4.2 Single-CO Key)

Conditions

Conditions	Note
It is possible to select whether to enable the Call Waiting feature for each extension through Web Maintenance Console.	
The number of simultaneous operation depends on the pre-programmed setting (**Call Limit**). **Example:** If **Call Limit** is set to 8 (default), Up to 7 operations can perform simultaneously.	
A user cannot perform the **[Placing the current call on hold]** operation, during the following calls: • DISA (Refer to 3.1.4 DISA (Direct Inward System Access)) • Meet Me (Refer to 3.9.2 Meet-me conference) • Doorphone (Refer to 3.11.1 Doorphone Call) • Built-in VM (Refer to 5 Voice Mail Features) • Paging (Refer to 3.10 Paging Features)	

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port—SIP Extension
2.3.2 PBX Configuration—[2-2] Extension—Phone

3.7 Holding Features

3.7.1 Call Hold

Description

An extension user can put a call on hold. Only the extension user who held the call can retrieve it.
For KX-HDV230 SIP phone:
If a call is held using S-CO key, any KX-HDV230 SIP phone can retrieve a held call. Refer to 4.2 Single-CO Key.

Operation

For All Extensions:
[Call Hold]
To hold a call, press the [HOLD] button during a conversation.

[Call Hold Retrieve]
Typically press the [HOLD] button to retrieve. Operation for retrieving a held call depends on the type of telephone being used. For details, refer to manual for your telephone.

For KX-HDV230 SIP Phone:
[Call Hold]
To hold a call, press the [HOLD] button during a conversation.
[Call Hold Retrieve]
Press an S-CO key that indicates "Holding a call" (LED is flashing slowly). For more information about a status of the S-CO key, refer to 4.2 Single-CO Key.

Conditions

Conditions	Note
The following calls cannot be held: • DISA (Refer to 3.1.4 DISA (Direct Inward System Access)) • Meet Me (Refer to 3.9.2 Meet-me conference) • Doorphone (Refer to 3.11.1 Doorphone Call) • Built-in VM (Refer to 5 Voice Mail Features) • Paging (Refer to 3.10 Paging Features)	
For KX-HDV series only: If a call on hold is not retrieved within a preprogrammed time period (**Hold Recall (s)**), the operated extension rings. If **Hold Recall (s)** is set to 0, this feature is disabled.	
A person holding a line will hear Music on hold (MOH).	Refer to 3.7.3 Music on Hold.

PIL Reference
2.2.5 PBX Configuration—[1-5] System—Timers

3.7.2 Call Park

Description

An extension user can hold a call by placing it into a common parking zone of the PBX.
A parked call can be retrieved by any extension user. This feature is useful when an extension user wants to hold more than one intercom call or trunk call with an extension.

Operation

[Call Park]
Perform Call Park using the following procedure:

- Put the current call on Consultation Hold by executing the following operations:
  SLT: Hooking
  SIP: Press the [TRANSFER] button
- After a dial tone is heard, dial the Call Park feature number and [0] [0] (Refer to 2.1.1 Feature Number Lists for User).
- When a Call Park is successful, a Park Area Number will be announced. Record it as it is a necessary parameter when performing Call Park Retrieve. After Park Area Number announcement is complete, the operated extension will hear the Music on hold (MOH).

[Call Park Retrieve]
To retrieve a parked call, go off-hook and dial the Call Park feature number and a Park Area Number. (Refer to 2.1.1 Feature Number Lists for User)

Conditions

Conditions	Note
Call Park cannot be used for doorphone calls.	
When an extension user operates Call Park, A Park area number (01-24) will be specified automatically.	
If a parked call is not retrieved within a preprogrammed time period (**Hold Recall (s)**), the operated extension rings. If **Hold Recall (s)** is set to 0, this feature is disabled.	If the operated extension does not answer the call within a preprogrammed time period (**Disconnect after Recall (x60s)**), the call on hold will be disconnected.
A party that has been put on hold will hear the Music on hold (MOH).	Refer to 3.7.3 Music on Hold.

3.7.3 Music on Hold

Description

Music can be played to a party that has been put on hold.

Operation

[Playing of MOH]
When the following features are implemented, the called party will hear Music on Hold.

- Call Hold (Refer to 3.7.1 Call Hold)
- Call Park (Refer to 3.7.2 Call Park)

When the following features are implemented, the called party will hear Music on Hold depending on settings.

- Call Transfer-Unscreened (Refer to 3.8.2 Call Transfer-Unscreened)
- Call Transfer-Blind (Refer to 3.8.3 Call Transfer-Blind)
- Call Splitting (Refer to 3.8.1 Call Transfer-Screened)

When the following features are implemented, the executor will hear Music on Hold depending on settings.

- After DISA answers, calls to dialing destinations such as an extension/trunk (Refer to 3.1.4 DISA (Direct Inward System Access))

[MOH Registration]
Up to one audio file can be uploaded from the Web Maintenance Console. After uploading, the old audio file will be overwritten to the new audio file.

Conditions

Conditions	Note
The User-supplied audio files or original music (pre-installed) can be used. It is possible to select which will be used through system settings.	

Conditions	Note
While uploading an audio file, you cannot make or receive calls. It is recommended to upload audio files outside the operation hours.	
In the case of calling to an extension/trunk after DISA answers, it is possible to select whether Music on Hold or a ring back tone is heard for the caller to DISA through system settings.	

PIL Reference
2.2.2 PBX Configuration—[1-2] System—MOH

3.8 Transferring Features

3.8.1 Call Transfer-Screened

Description

It is possible to transfer a trunk or an extension which is in a conversation to another extension after confirming by speaking with the other party at the transfer destination.
Call Transfer—Screened is also known as Call Transfer with Announcement.

Operation

Perform Call Transfer-Screened using the following procedure:

1. Put the current call on Consultation Hold by executing the following operations:
   SLT: Hooking
   SIP: Press the [TRANSFER] button
2. After a dial tone is heard, make a call to the transfer destination. It is possible to call from an extension using the DSS button while a dial tone is being heard for entering the transfer destination.
3. After the transfer destination answers, go on-hook.

Perform Cancel Call Transfer-Screened using the following procedure:
SLT: Operation is as follows:

- Hooking. After that, the original two-party conversation will be restart.
- Go on-hook before dialing the telephone number. After that, the operated extension will ring.

SIP extension: Press the [CANCEL] button twice.

Conditions

Conditions	Note
The following cannot be set as transfer destinations: • DISA (Refer to 3.1.4 DISA (Direct Inward System Access)) • Meet Me (Refer to 3.9.2 Meet-me conference) • Doorphone (Refer to 3.11.1 Doorphone Call) • Built-in VM (Refer to 5 Voice Mail Features) • Paging (Refer to 3.10 Paging Features)	
Analog extension, SIP phone can operate the Call Transfer feature. General SIP extension depends on the terminal specifications.	

Conditions	Note
The FWD setting for the transfer destination extension is enabled. If the FWD destination of the Transfer destination is DISA or Meet-me, call transferring will be restricted.	Reference: 3.1.4 DISA (Direct Inward System Access) 3.9.2 Meet-me conference
A party who is put on Consultation Hold will hear Music on hold (MOH).	Refer to 3.7.3 Music on Hold.

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port

3.8.2 Call Transfer-Unscreened

Description

This feature transfers a trunk or an extension in a conversation to another extension without speaking to the other party at the transfer destination, with the caller being transferred hearing a ring back tone while the transfer destination is being called.
Call Transfer—Unscreened is also known as Call Transfer without Announcement.

Operation

Perform Call Transfer-Unscreened using the following procedure:

1. Put the current call on Consultation Hold by executing the following operations:
   SLT: Hooking
   SIP: Press the [TRANSFER] button
2. After a dial tone is heard, make a call to the transfer destination.
3. Before the transfer destination answers, go on-hook.

Perform Cancel Call Transfer-Unscreened using the following procedure:
SLT: Hooking.
SIP extension: Press the [CANCEL] button twice.
[Transfer Recall]

- If a transfer destination does not answer within a certain length of time, the call will ring at the extension of the person who made the transfer operation.

Conditions

Conditions	Note
The following cannot be set as transfer destinations: • DISA (Refer to 3.1.4 DISA (Direct Inward System Access)) • Meet Me (Refer to 3.9.2 Meet-me conference) • Doorphone (Refer to 3.11.1 Doorphone Call) • Built-in VM (Refer to 5 Voice Mail Features) • Paging (Refer to 3.10 Paging Features)	
The FWD setting for the transfer destination extension is enabled. If the FWD destination of the Transfer destination is DISA or Meet-me, call transferring will be restricted.	Reference: 3.1.4 DISA (Direct Inward System Access) 3.9.2 Meet-me conference

Conditions	Note
A party who is put on Consultation Hold will hear Music on hold (MOH).	Refer to 3.7.3 Music on Hold.
When the transfer destination is an extension: If the transfer destination does not answer within the pre-programmed time period (**FWD No Answer Time**), either FWD NA or Transfer Recall will operate. **When the transfer destination is a trunk:** If the transfer destination does not answer within 180 seconds, Transfer Recall will operate.	Refer to 3.1.9 Call Forwarding (FWD)
It is possible to select the operation when the call is transferred to extension or extension group as follows: • **FWD Priority** FWD NA will operate when FWD NA is set. Otherwise Transfer Recall will operate. • **Transfer Recall** Transfer Recall will always operate.	Programming Item List: • 2.3.2 PBX Configuration—[2-2] Extension—Phone—FWD/DND—Transferred call no answer option • 2.3.4 PBX Configuration—[2-4] Extension—Extension Group—Group Setting—Transferred call no answer option
Transfer Recall operation is different depending on the status of the forwarding extension: • **Idle** Transfer Recall will operate. • **During a conversation or receiving a call** If Call Waiting is enabled, Transfer Recall will operate. Otherwise, Transfer Recall will operate as soon as the forwarding extension goes becomes idle.	If there is no answer to the second call from the transfer destination after calling for a preprogrammed time period (**Disconnect after Recall (x60s)**), it will be disconnected.

PIL Reference
2.2.5 PBX Configuration—[1-5] System—Timers
2.3.1 PBX Configuration—[2-1] Extension—Port

3.8.3 Call Transfer-Blind

Description

The Function to Transfer an external line or an extension in communication to another extension without confirming the ring back tone of the outgoing call to the Transfer destination.
This feature is only available for SIP extensions. (For general SIP phones, the operation differs depending on the phone.)

Operation

Perform Call Transfer-Blind using the following procedure: (For KX-HDV series only)

1. Press the [Blind] Key
2. Dial the transfer destination number and go on-hook.

[Transfer Recall]

- If a transfer destination does not answer within a certain length of time, the call will ring at the extension of the person who made the transfer operation. For more information, refer to 3.8.2 Call Transfer-Unscreened.
- If a call cannot be transferred to the transfer destination (e.g. when the transfer destination is on another call), the Transfer Recall immediately operates.

Conditions

Conditions	Note
The following cannot be set as transfer destinations: • DISA (Refer to 3.1.4 DISA (Direct Inward System Access)) • Meet Me (Refer to 3.9.2 Meet-me conference) • Doorphone (Refer to 3.11.1 Doorphone Call) • Built-in VM (Refer to 5 Voice Mail Features) • Paging (Refer to 3.10 Paging Features)	
The FWD setting for the transfer destination extension is enabled. If the FWD destination of the Transfer destination is DISA or Meet-me, call transferring will be restricted.	Reference: 3.1.4 DISA (Direct Inward System Access) 3.9.2 Meet-me conference
A party who is put on Consultation Hold will hear Music on hold (MOH).	Refer to 3.7.3 Music on Hold.

PIL Reference
2.2.5 PBX Configuration—[1-5] System—Timers
2.3.1 PBX Configuration—[2-1] Extension—Port

3.8.4 Call Splitting

Description

A KX-HDV series SIP phone user can speak alternately with two parties. Placing the current call on hold allows the user to speak with the other party.

Operation

While having a call on hold (including Consultation Hold) and having another conversation, pressing the DN key or the S-CO key (that status is holding a call) allows the following operations:

- The current call → put on hold
- The call on hold → changes to conversation with the extension

Note

- For more information about Call Hold and Consultation Hold, refer to 3.7.1 Call Hold and 3.8.1 Call Transfer-Screened.
- For more information about the DN key and the S-CO key, refer to 4.4 DN Key and 4.2 Single-CO Key.

Conditions

Conditions	Note
A party who is put on hold will hear Music on hold (MOH).	Refer to 3.7.3 Music on Hold.

3.9 Conference Features

3.9.1 Three-party Conference

Description

During a two-party conversation, an extension user can add a third party to the conversation, thereby establishing a three-party conference call.

Operation

Perform Three-party Conference using the following procedure:
For SIP phone:

- Operation of this feature depends on the type of SIP phone being used. For details, refer to manual for your telephone.
- Operation for KX-HDV series is as follows:
  1. Press [CONF] button during a conversation.
  2. Dial the party you want to add to the conversation.
  3. Press [CONF] button.

For SLT extension:

1. Hooking.
   The current call is put on Consultation Hold. (To retrieve Consultation Hold, hooking again)
2. Dial the party you want to add to the conversation.
3. Hooking after the new party answers the call.

Three-party conference will be finished when an extension user operates as follows:
For SIP phone:
Going On-hook.
If the conference originator leaves the conference, remaining two-party conversation will be also finished.

For SLT extension:

- Going On-hook.
  Remaining two-party conversation will be continued.
- Hooking (For the conference originator only).
  The new party will be disconnected, and remaining two-party conversation will be continued.

Conditions

Conditions	Note
When a conference is started, all conference participants will hear a confirmation tone.	
If an SLT extension user originates the conference, only the conference originator can put the conference call on hold.	

3.9.2 Meet-me conference

Description

By calling a floating extension number set as a Conference Room, it is possible to perform a Meet-me conference.

[Make a conference room]
Perform the following settings through Web Maintenance Console. It is possible to specify a maximum of 3 conference rooms.

- The name of the Meet-me conference room
- The Floating extension number of the Meet-me conference room
- The Access code of the Meet-me conference room

Operation

[Attend a conference room]
Access the Meet-me conference room using the following procedure. The Meet-me conference can accommodate a maximum of 6 parties.

- Call the Floating extension number of the Meet-me conference room
- Enter the Access code of the Meet-me conference room after a confirmation tone is heard.

Conditions

Conditions	Note
Analog trunk and KX-NTV series cannot access a Meet-me conference room.	
When only one party is in the conference room, Music on hold will be heard	
If the wrong access code is entered, the extension user will hear the busy tone.	

PIL Reference
2.7 PBX Configuration—[6] Conference—Meet Me

3.10 Paging Features

3.10.1 Paging

Description

An extension user can make a voice announcement to many destinations simultaneously.
The message is announced over the built-in speakers of SIP phones which belong to the extension group.

Operation

[Paging Execution]
While a dial tone is being heard, enter a Paging feature number and a Group Number (2 digits) (Refer to 2.1.1 Feature Number Lists for User).
After a confirmation tone is heard on the Paging device, Paging will operate.

Conditions

Conditions	Note
• Only KX-HDV series SIP phones can make a paging call. • Paging targets are the SIP extensions only.	
Up to four extensions will be paged within the member extensions of the Extension Group. Paging will start from the oldest affiliated registered extension in the Extension Group.	Refer to 3.1.12 Extension Group Call
Paging type for each Extension Group can be selected as follows (default: One-way): • One-way: Extension group members can hear paging announcements, but they cannot answer paging announcements. • Two-way: Extension group members can hear paging announcements, and they can answer paging announcements.	
Paging call can be made while the extensions are waiting. If Paging call becomes impossible during a Paging call, Paging will be canceled only for that extension.	
When DND is set for extensions, Paging is not available to the extension and Paging will not work.	
When the Extension group receives the paging call, if all extension group member cannot answer the paging call, the Caller will hear the busy tone.	
Paging calls are targeted for limited simultaneous calls.	Refer to 2.3.3 System Capacity in Getting Started
If more than 2 Remote extensions are included in paging or paged member, paging doesn't work. Paging member hear busy tone.	

PIL Reference
2.3.4 PBX Configuration—[2-4] Extension—Extension Group

3.11 External Device Features

3.11.1 Doorphone Call

Description

It is possible to connect doorphones directly to the PBX. When a visitor presses the call button on a doorphone, the doorphone calls a preprogrammed destination (extension). In addition, extension users can dial the preset number of a doorphone to call that doorphone.

Operation

[Receive a doorphone call]
When pressing the call button on a doorphone, a ring back tone is heard and an extension will be called. Answer the call that rings at the incoming doorphone call destination set through Web Maintenance Console.

[Make a doorphone call]

While a dial tone is heard, call the floating extension number assigned to the doorphone. After a confirmation tone is heard, a conversation will start.

Conditions

Conditions	Note
Hardware Requirement: An optional doorphone, and doorphone card.	
Up to two doorphones can be installed.	
Set each incoming doorphone call destination for each Time mode (Day/Lunch/Night).	
It is possible to set a floating extension number and name for each doorphone port. When a call is received at an extension from the doorphone, the preset floating extension number and name will be notified to the extension.	
If an extension does not answer an incoming call from the doorphone within a preprogrammed time period, the call will be disconnected.	
FWD/DND settings of the extension will ignore incoming calls from Doorphone.	Reference: 3.1.9 Call Forwarding (FWD) 3.1.10 Do Not Disturb (DND)

PIL Reference
2.3.5 PBX Configuration—[2-5] Extension—Doorphone

3.11.2 Door Open

Description

An extension user can unlock the door for a visitor using his telephone.
The door can be unlocked by extension users who are allowed to unlock the door through Web Maintenance Console.
However, while engaged on a doorphone call, any extension user can unlock the door to let the visitor in.

Operation

Door Open is performed using one of the following three methods:

[Type A]
After answering an incoming call from a doorphone, dial the Door Open feature number during a conversation. (Refer to 2.1.1 Feature Number Lists for User)
After the door is opened, a confirmation tone will be heard. The conversation with the doorphone continues.

[Type B] Manager Extension Only
After calling a doorphone, dial the Door Open feature number during a conversation. (Refer to 2.1.1 Feature Number Lists for User)
After the door is opened, a confirmation tone will be heard. The conversation with the doorphone continues.

[Type C] Manager Extension Only
Go off-hook and dial the Door Open feature number and the door number. (Refer to 2.1.2 Programming by Dial for Manager)
After the door is opened, a confirmation tone will be heard and the call will be disconnected.

Conditions

Conditions	Note
Hardware Requirement: A user-supplied door opener on each door, and a doorphone card.	
The door can remain unlocked for a preprogrammed time period. In the case of Type A and B, when the door is open and if the Door Open feature number is dialed again during a conversation, the Door Open Duration will be extended for a preprogrammed time period.	When the Door Open Duration is extended, a confirmation tone will be heard.
The door opener will unlock the door even if a doorphone is not installed.	

PIL Reference
2.3.5 PBX Configuration—[2-5] Extension—Doorphone

3.12 Other Features

3.12.1 Time Service

Description

Time service modes are used by many PBX features to determine how they will function during different times of day. For example, incoming calls can be directed to sales staff during the day and to a Voice Mail at night, extension users can be prohibited from making long-distance calls during lunch time, etc.
There are 3 time service modes—day, night, and lunch.

Operation

Though it is possible to set/cancel Time mode through Web Maintenance Console as well as confirm the current Time mode, operations by a feature number are also possible using one of the following methods:

[Time mode set/cancel] Manager extension Only
After a dial tone is heard, enter the Time Service (Day / Lunch / Night) Switch feature number and dial the desired Time Mode (Refer to 2.1.2 Programming by Dial for Manager)
After the caller hears the confirmation tone, the present mode will be announced and the call will be disconnect.

[Confirm Time mode] Manager extension Only
After a dial tone is heard, enter the Time Service (Day / Lunch / Night) Switch feature number and dial #
(Refer to 2.1.2 Programming by Dial for Manager)
After a confirmation tone is heard, the call will be disconnected.

Conditions

Conditions	Note
The current time service mode can switch automatically to another time service mode at the time assigned in the Time Table. It is possible, however, to switch time service modes manually. Whether time service modes are normally switched manually or automatically is determined through system programming.	

Conditions	Note
In regards to automatic mode switching, the start time of Day Mode, Lunch Mode, and Night Mode can be set through Web Maintenance Console. For Day Mode, in order to divide the time zone, it is possible to set two start times (Day Mode 1 and Day Mode 2).	Each mode will operate from its start time until the next mode's start time.
Automatic mode will continue even if the time service mode is switched manually when the automatic mode is enabled.	

PIL Reference
2.2.3 PBX Configuration—[1-3] System—Week Table

3.12.2 CDR (Call Detail Record)

Description

This function records the information of outgoing and incoming call numbers, etc. for all calls to a csv file. System users can confirm the necessary information by utilizing Macros and tools with this csv file.

Note

This CDR data can be used just for reference. This is because an accurate duration of a call may not be counted by CDR data, when the call is transferred (Call Transfer) or parked (Call Park).

CDR file information recording is as follows:

Column	Item[*1]	Description
1	Account Code	Shows the account code appended to the call.
2	Caller	**Outgoing call:** Shows the caller's extension number. **Incoming trunk call:** Shows the Caller ID. **Blind transferring call:** Shows "*".
3	Callee	Shows the destination dial number.
4	Source Channel	Shows the channel that is used by the caller. The information recorded varies depending on the type of terminal or trunk. For more information, refer to [Channel Information].
5	Destination Channel	**Incoming call:** Shows the channel that is used by the called party. **Outgoing trunk call:** Shows the trunk channel that is used by the caller. The information recorded varies depending on the type of terminal or trunk. For more information, refer to [Channel Information].
6	Feature	If the called party is a floating extension (Voicemail, Paging or MeetMe), the following feature names are displayed: • Voicemail • Paging • MeetMe

Column	Item[*1]	Description
7	Option 1	Used as developer’s information.
8	Option 2	Used as developer’s information.
9	Start Time	Shows the date and the start time of the call.
10	Answer Time	Shows the date and time of answering the call.
11	End Time	Shows the date and the end time of a call.
12	Total Duration	Shows the duration of the call including the ring duration. (Seconds)
13	Talking Duration	Shows the duration of the conversation. (Seconds)
14	Status	The call making/receiving results are displayed as below: • ANSWERED • NO ANSWERED • BUSY • FAILED
15	Unique ID	Used as developer’s information.

*1 Item names are not recorded in the CDR file.

[Channel Information]

Terminal/ Extension type	Channel Information	Note
SLT Analog trunk Doorphone	DAHDI/n-m	n indicates the port number as follows: • n = 1—8: Trunk port (1—8) • n = 9—32: Extension port (1—24) • n = 33—34: Doorphone port (1—2) **m** is used as developer’s information.
SIP phone	SIP/n-yyyyyyyy	n shows SIP phone user name. Refer to **SIP User Name** in **2.3.1 PBX Configuration—[2-1] Extension—Port** in the Programming Item List. **yyyyyyyy** is used as developer’s information.
SIP trunk	SIP/n-qqqqqqqq	n indicates information as follows: • “**carrier1_**”: Carrier 1 • “**carrier2_**”: Carrier 2 • Authentication ID: SIP carrier’s Authentication ID. Refer to **2.4.7 PBX Configuration—[3-7] Trunk—SIP Trunk Property** in the Programming Item List. **qqqqqqqq** is used as developer’s information.

[Example]
When the following operations are performed, CDR data is recorded as follows:

Pattern A: Extension (102) calls to extension (201). Extension (201) answers the call, and starts the conversation.

Account Code	Caller	Callee	Source Channel	Destination Channel	Feature	Option 1	Option 2
	102	201	SIP/ name-00e e9a70	DAHDI/9-1		Dial	DAHDI/ 9,30,

Start Time	Answer Time	End Time	Total Duration	Talking Duration	Status	Unique ID
2016/6/9 16:38:49	2016/6/9 16:38:51	2016/6/9 16:38:54	5	3	ANSWERED	1465457929

Pattern B:
Extension (201) uses an Account code to call an analog trunk.
(Account code: 555555, destination number: 1234567890, Analog trunk port: 4)

Account Code	Caller	Callee	Source Channel	Destination Channel	Feature	Option 1	Option 2
555555	201	12345678 90	DAHDI/9- 1	DAHDI/4-1		Dial	DAHDI/ 4/9108,,

Start Time	Answer Time	End Time	Total Duration	Talking Duration	Status	Unique ID
2016/6/9 17:08:23	2016/6/9 17:08:39	2016/6/9 17:08:43	20	4	ANSWERED	1465459708

Pattern C: Extension (102) accesses Voicemail (500).

Account Code	Caller	Callee	Source Channel	Destination Channel	Feature	Option 1	Option 2
	102	500	SIP/ name-00b 27aa8		VoiceMail	VoiceMail Main	101

Start Time	Answer Time	End Time	Total Duration	Talking Duration	Status	Unique ID
2016/6/9 18:14:27	2016/6/9 18:14:27	2016/6/9 18:14:33	6	6	ANSWERED	1465463668

Operation

[CDR Record Enable/Disable]
CDR Recording can be set through Web Maintenance Console screen in the HTS system.

[CDR Recording Mode]
There are 2 modes for recording the CDR data, as follows:

CDR Recording Mode	Description
For Web Maintenance Console	A CDR data file can (Master.csv) be downloaded from the Web Maintenance Console. A maximum of 5 CDR data files, including the most recent file, can be retained in the PBX. **To download the CDR data files:** Click the **Save** button in Web Maintenance Console. Then, the PC can download all CDR data files as the compressed file (.zip). Refer to 2.2.7 PBX Configuration—[1-7] System—CDR in the Programming Item List.
For External Application	An external application can download a CDR data file without using Web maintenance console. One CDR data file can be retained. **To download the CDR data file**: An external application needs to send the following HTTP request to the PBX: **GET xxx.xxx.xxx.xxx/INSTALLER/yyyyy/Master.csv** – GET: HTTP request (Fixed) – xxx.xxx.xxx.xxx: IP address of the PBX – INSTALLER: The Installer level account name (Fixed) – yyyyy: The Installer level account password – Master.csv: The CDR data file name (Fixed) After that, the PBX will send a HTTP response with the CDR data file to the external application. **Note** • HTTPS request is not supported for downloading the CDR data file. • If downloading this file from the WAN port of the PBX, the firewall setting must be disabled through Web Maintenance Console. Before you disable the firewall setting, be sure to enable network security features (e.g. enable the firewall on the router). • Every time the CDR data file is downloaded, All CDR data files will be cleared.

Note

- These modes can be selected through Web Maintenance Console. If the CDR Recording Mode setting is changed, All CDR data files will be cleared.
- If the CDR Recording Mode setting is changed during a call, the CDR data for the call will not be recorded.

Conditions

Conditions	Note
The PBX records all communications regarding extensions and trunks if **CDR Recording** is Enabled.	
Each CDR data file has a maximum size of 2.5 MB. (Around 10000 call records)	

Conditions	Note
[CDR Recording Mode (For Web Maintenance Console)] • If the number of files is greater than 5, the oldest file will be deleted in sequence.	As old csv files have different extensions (.csv.X), renaming an extension is required when editing it by spreadsheet software, etc.
[CDR Recording Mode (For External Application)] • If the size of the CDR data file exceeds the maximum size, the data will be cleared.	
The setting of CDR Record Enable/Disable for **CDR Recording** is not immediately reflected after being changed through Web Maintenance Console. When all calls during a conversation are completed, the change of the CDR Record Enable/Disable setting for **CDR Recording** is reflected.	

PIL Reference
2.2.7 PBX Configuration—[1-7] System—CDR

3.12.3 E-mail Notification Features

Description

This function notifies each type of event information that has occurred within/outside the PBX by e-mail.
As the PBX actively sends "Failure information" and "VM absence record notice", etc. by e-mail, quick maintenance implementation and timely access to incoming call information becomes possible.
When the following events occur, event information is sent to pre-set e-mail addresses (The sending address is set using Web Maintenance Console):

- System Alarm (system settings)
- Send Test Email (system settings)
- VM recording notice (settings for each phone)

The e-mail sending addresses for VM recording notifications are as follows:

- E-mail addresses which are set for the extension number that has received a call.
- When a call arrives at an Extension Group, an e-mail address of the extension number set as the first member of the Extension Group becomes the sending address.

Conditions

Conditions	Note
It is possible to filter the e-mail sending suitability for System Alarm e-mail notifications by Error Level (Major/Minor).	
A VM recording notification is sent with the recorded Voice Mail attached.	Refer to 5.4 Message Notification by E-mail
An e-mail sent from the PBX is used exclusively for sending. Even if a user replies to an e-mail sent by the PBX, the PBX will discard the reply.	The length of the forwarding mail address can be set up to 80 characters.
An e-mail sent from the PBX uses the character code UTF-8.	

Conditions	Note
If an e-mail fails to be sent for temporary reasons, resends will be attempted periodically until it is sent successfully. If the e-mail fails to be sent even after 5 days have passed, the resend process will cease.	

PIL Reference
2.3.2 PBX Configuration—[2-2] Extension—Phone
4.4.3 Maintenance—[3-3] Utility—Email Notification

3.12.4 Outgoing Message (OGM)

Description

An extension assigned as the manager (manager extension) can record outgoing messages (OGMs) for the DISA features (Refer to 3.1.4 DISA (Direct Inward System Access)).

Operation

The operation to record/delete/playback for OGMs is as follows:
While a dial tone is heard, enter the OGM Record / Delete / Playback feature number and select an operation (Record/Delete/Playback) by the parameter, and a DISA floating extension number.
(Refer to 2.1.2 Programming by Dial for Manager)
Go on-hook to finish recording the OGM.
You can upload an audio file (WAV format) recorded with an external device to the OGM #1-#10 and use it as an OGM. For how to upload audio files, refer to 2.4.5 PBX Configuration—[3-5] Trunk—DISA—Message—WAV file upload in Programming Item List.

Conditions

Conditions	Note
The operation to record/delete/playback for OGMs is only available for an extension which is authorized to operate as a manager through Web Maintenance Console.	When operated from an unauthorized extension, a busy tone will be heard.
While uploading an audio file, you cannot make or receive calls. It is recommended to upload audio files outside the operation hours.	
When other manager extensions are recording or playing back OGMs, the OGM operation using the feature number will be disabled and a busy tone will be heard.	
The recording time for each message is up to 60 seconds. When the recording time has passed 60 seconds, recording will be finished automatically and the busy tone will be heard.	

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port
2.4.5 PBX Configuration—[3-5] Trunk—DISA

Section 4

Flexible Button Features

4.1 DSS Key

Description

It is possible to set DSS (Direct Station Selection) to the Flexible buttons of a KX-HDV series SIP phone.
Flexible buttons can be set for each user through Web Maintenance Console.

Note

- The DSS key will be available in 10 minutes after System Data is saved by Web Maintenance Console.
- The DSS key will be available in 20 seconds after DSS key setting is changed.

Operation

By pressing a DSS key indicating a call is being received, the user can pick up that call on the specified extension.

Conditions

Indicates the status of an extension specified as DSS using the button's LED.

Usage Status	LED Indication Pattern/LED Image
The specified extension is waiting	OFF
The specified extension is off-hook	SLT: Red ON Other: OFF
The Specified extension is making a call	Red ON
The Specified extension is receiving a call	Red flashing
The specified extension is in a conversation	Red ON

PIL Reference
2.3.3 PBX Configuration—[2-3] Extension—Flexible Buttons

4.2 Single-CO Key

Description

It is possible to set a key which corresponds to a Trunk line number to a Flexible button of a SIP phone. This feature is only available for KX-HDV230.
Flexible buttons can be set for each user through Web Maintenance Console.

Operation

- By pressing a key that indicates “Idle” before dialing an outside phone number, the user can make an outside call.
- By pressing a key that indicates “Receiving a call”, the user can answer that call at the Single-CO Key.
- By pressing a key that indicates “Holding a call”, the user can retrieve the call on hold on the Single-CO Key.

Conditions

Conditions	Note
Up to 8 Single-CO Key can assign to the flexible button.	
If a Single-CO key is added to flexible button, or if the phone is connected to the PBX, LED indication of the Single-CO key is not updated immediately. The LED indication is updated first when the status of the Single-CO key has been updated.	

The LED patterns and the corresponding status of Single-CO keys are as follows:

Usage Status	LED Indication Pattern/LED Image
Idle/No connect	OFF
Receiving a call (at own extension)	Flashing blue rapidly
Receiving a call (at another extension)	Flashing red rapidly
Dialing/During a conversation (at own extension)	Blue ON
Dialing/During a conversation (at another extension)	Red ON
Holding a call (at own extension)	Flashing blue slowly
Holding a call (at another extension) [*1]	Flashing red slowly
During a Consultation Hold (at own extension)	Flashing blue slowly
During a Consultation Hold (at another extension)	Red ON

[*1] If an extension with a call on hold has no S-CO key (or if no S-CO key is assigned to the extension with a call on hold), the LED indication pattern will be "Red On (During a conversation)".

PIL Reference
2.3.3 PBX Configuration—[2-3] Extension—Flexible Buttons

4.3 One touch dial key

Description

A KX-HDV series SIP phone user can access a person or feature by pressing a single button. This is activated by storing the number (e.g., extension number, telephone number, or feature number) in a One-touch Dialing button. For more information about this feature, refer to the manual for your telephone. Flexible buttons can be set for each user through Web Maintenance Console.

Operation

Pressing the One-touch dialing button will dial a preset telephone number with a maximum of 32 digits.

PIL Reference
2.3.3 PBX Configuration—[2-3] Extension—Flexible Buttons

4.4 DN Key

Description

It is possible to set a DN (Directory Number) key to a Flexible button of a KX-HDV series SIP phone (except KX-HDV100). For more information about this feature, refer to the manual for your telephone.

Flexible buttons can be set for each user through Web Maintenance Console.

PIL Reference
2.3.3 PBX Configuration—[2-3] Extension—Flexible Buttons

Section 5

Voice Mail Features

5.1 Mailbox Recording

Description

When an incoming call is received by an extension or an Extension Group, and if there is no answer or the line is busy for a preset time, this function allows conversations forwarded to Voice Mail to be recorded in the mailbox, if the mailbox is available.

Condition

Conditions	Note
The recording time for the entire System is as follows: Recording time: 120 minutes	The recording time include Personal Greetings, recorded messages and Outgoing Messages (OGM).
The maximum number of recordings and the recording time can be set for each extension through Web Maintenance Console: • **Recording Number**: 10 messages (default) • **Recording Time (s)**: 60 seconds (default)	If **Recording Number** is set to 0, a caller cannot record a message.
The maximum number of recordings and the recording time of each Extension Group is as follows: Number of recordings: Follows the extension setting registered at the head of the Extension Group. Recording time: Follows the extension setting registered at the head of the Extension Group.	
The Number of mailboxes is 24 (Maximum number of extensions)	A mailbox is attached to each extension (Except a floating extension). If the extension number is changed, the recorded message within the mailbox will be cleared.
The maximum length of Personal Greeting message is 30 seconds.	
Forwarded calls to the mailbox targeted for the message recording are as follows: • Call Forwarding (FWD All, FWD Busy, FWD NA) from Extension. • Call Forwarding (FWD ALL, FWD Busy, FWD NA) From Extension Group.	Refer to 3.1.9 Call Forwarding (FWD)
When the call is forwarded to Voice Mail, the mailboxes targeted for recording are as follows: • When the call is forwarded from an extension, the voice is recorded to the mailbox for the forwarded extension. • When the call is forwarded from an Extension Group, the voice is recorded to the mailbox for the first member extension in the forwarding Extension Group.	The call will be disconnected if already exceeding the number of messages.

Conditions	Note
When the total time of the recorded message has reached the entire recording time limit, the message is automatically deleted from the oldest. In this case, 5% of the messages of the total recording time limit is deleted. (Except Personal Greetings and Outgoing Messages (OGM))	
Once the total recording time has reached 80% of the system capacity, an error log is recorded in the system, and the email is sent to the administrator.	• Refer to 5.4 Message Notification by E-mail • For detail of error log, refer to 5.4 System Logging in the Getting Started.
New message is stored to “New” folder in the mailbox. At that time, MWI (Message Waiting Indication) is notified to the extension that assigned to the mailbox.	The message under 3 seconds cannot be stored to the mailbox. Email is not also sent.

PIL Reference
2.3.1 PBX Configuration—[2-1] Extension—Port－Main－Voice Mail

5.2 Voice Mail Menu

Description

When dialing from an extension to a VM floating extension, the mailbox of the extension will answer, and the extension user can operate the Voice Mail Menu below:

- Playback recorded message/Delete/Reply/Transfer to other mailbox
- Switch message folder (New/Old/Work/Family/Friend folder)
- Advanced options (record to designated mailbox)
- Mailbox options (record type of greeting message/record name/change password)

Operation

1. Call VM virtual extension.
2. If the mailbox password is available, input the password (Extension PIN). Call will be disconnected if password input fails 3 times. If the mailbox password is not available, advance to step 3.
3. Operate voice menu via the below dial inputs.
   - Dial [1]: Playback recorded message. After message playback, other operations (Deletion of message, replay, transfer to other mailbox, or playback next message) can be performed.
   - Dial [2]: Switch message folder.
   - Dial [3]: Record a message to the specified extension’s mailbox.
   - Dial [0]: Select mailbox option. (Changing greeting messages, the name, or the password)
   - Dial [#]: Close voice mail menu.
4. Operate in accordance with the guidance.

Condition

Conditions	Note
Extension PIN will also be changed when the mailbox password is changed by the Voice Mail menu operation.	Refer to 3.4.5 Extension Dial Lock
After main menu is played, if the extension user who listens to the menu does not enter the dial, the same menu will repeat twice. And then if the user does not enter the dial number, Voice Mail menu will be closed.	
The language for voice guidance can be set from the Web Maintenance Console.	
By downloading the voice guidance files of the other language, it is possible to add and set the language other than the built-in voice guidance. Up to one voice guidance file (one language) can be add to the system.	Also, the file will be overwritten when there is downloaded voice guidance.

PIL Reference
2.8 PBX Configuration—[7] Voice Mail

5.3 Personal Greeting (No Answer, Busy, Temporary Greeting)

Description

Are the greetings heard when a caller reaches a subscriber's mailbox. Subscribers can record a personal greeting.

Operation

Record Personal Greeting by the following procedures:

1. From an extension, log in to Voice Menu and select Mailbox Option. (Refer to 5.2 Voice Mail Menu)
2. Select the Greeting message to record in accordance with the guidance. Operate in accordance with the guidance after recording is complete.

Condition

Conditions	Note
The below greeting messages can be set for each Voice Mail. • **Unavailable Greeting**: This message will play when the extension that has set other than FWD (Busy) is received the call. • **Busy Greeting**: This message will play when the extension that has set FWD (Busy) is received the call. • **Name**: When this message is recorded, it will play back the subscriber's name. The recorded name will be played instead of the extension number. • **Temporary Greeting**: Use when wanting to change the message temporarily.	

Conditions	Note
Greeting messages cannot be reset to the default message, If Unavailable Greeting and Busy Greeting are recorded.	
When Name is recorded, directly before playing back the default message, the Name and message will be played.	Name cannot be played back when Unavailable Greeting and Busy Greeting are recorded.
When Temporary Greeting is recorded, this message will be played back as the greeting message. When Temporary Greeting is deleted at the mailbox options, the previous message will be set.	

5.4 Message Notification by E-mail

Description

Sends a voice message recorded to a mailbox as an e-mail with the voice message attached to an e-mail address set in advance.

Condition

Conditions	Note
When the message is recorded to the mailbox, E-mail with attached the message (WAV file format) is sent to the extension user.	
The destination e-mail address for each extension can be set.	
The pre-programmed title (**Voice Mail**—**Subject**) + "Caller name / Caller id" is set as the Subject of the e-mail. The e-mail body is blank.	
It is possible to specify for each extension whether to delete the message after the e-mail is sent.	

PIL Reference
2.3.2 PBX Configuration—[2-2] Extension—Phone
4.4.3 Maintenance—[3-3] Utility—Email Notification

Section 6

Networking Features

6.1 Networking Features

Description

The following table shows descriptions of the networking features and router features.

Functions		Description
WAN	Interface	10/100/1000Mbps Ethernet (PPPoE/DHCP/Static IP)
LAN (Wired)	Interface	10/100/1000Mbps Ethernet
Wireless LAN	Interface	IEEE 802.11b/g/n
	Security	WPA2(CCMP), WPA-WPA2(TKIP-CCMP), Basic(WEP64/128)
	Multiple SSID	No
	Automatic WLAN Channel Selection	Yes
	WLAN Associations Filtering(ACL)	Yes
	Automatic set up for Wireless LAN devices	Yes
	Throughput	Target throughout under ideal conditions – 802.11b: 5-6Mbps – 802.11g: 20Mbps – 802.11n: 70-80Mbps QoS feature (IEEE802.11e EDCA) is supported.
Network Layer Protocols	IPv4 Networking	Yes (IPv6 is not supported)
Network Core Functions	Static Routing	Yes
	Firewall	Packet Filtering (IPv4, MAC address, port numbers and protocols) Basic DoS Attack Prevention Stateful Packet Inspection (SPI) DMZ host
	NAT/NAPT	Yes
	Quality of Service	DiffServ for IP-QoS. Optimum default QoS settings that prioritize Voice and Voice calls.
Networking Applications	DHCP	DHCPv4
	NTP	NTP Client
	DNS	DNSv4
	DNS Proxy or Relay	Yes
	Dynamic DNS (DDNS)	Yes
	HTTP Server	HTTP-1.0, Secured-HTTP(HTTPs)

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