Panasonic®

Feature Guide

Hybrid IP-PBX

Model No. KX-NS300

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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 (Page 16)" before using this product.

KX-NS300: PFMPR Software File Version 001.00000 or later.

Introduction

About this Feature Guide

This Feature Guide is designed to serve as an overall feature reference for the Panasonic IP-PBX. It explains what this PBX can do, and how to obtain the most out of its many features and facilities. This manual contains the following sections:

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, Call Handling Features
Provides details about the call handling features.

Section 3, Unified Messaging System
Provides details about the features of the Unified Messaging system.

Section 4, Network Features
Provides details about public and private networks you can connect the PBX to.

Section 5, System Configuration and Administration Features
Provides details about the system configuration and administration features.

Section 6, Appendix
Provides tables listing capacity of system resources, exclusive features for each PBX model, tone and ring tone tables, and the revision history of this Feature Guide.

Index
Provides feature titles and important words to help you access the required information easily.

Functional Limitation

Depending on the PBX's software version, these features may not function. For details about which versions support these features, consult your dealer.

- Features are provided by ISDN services

[Example]

4.1.2 Integrated Services Digital Network (ISDN) Service Features
4.1.2.6 Call Transfer (CT)—by ISDN
4.1.2.7 Three-party Conference (3PTY)—by ISDN

References Found in the Feature Guide

Installation Manual References
The required installation instruction titles described in the *Installation Manual* are noted for your reference.

PC Programming Manual References
The PC Programming titles and parameters described in the *PC Programming Manual* are noted for your reference.

PT Programming Manual References
The PT Programming titles described in the *PT Programming Manual* are noted for your reference.

Feature Guide References
The related feature titles described in this *Feature Guide* are noted for your reference.

User Manual References
The operation required to implement the feature described in the *User Manual* is noted for your reference.

Abbreviations

There are many abbreviations used in this manual (e.g., "PT", for proprietary telephone). Please refer to the list in the next section for the meaning of each abbreviation.

About the other manuals

Along with this Feature Guide, the following manuals are available to help you install, and use this PBX:

Installation Manual

Provides instructions for installing the hardware and maintenance of the PBX.

PC Programming Manual

Provides step-by-step instructions for performing system programming using a PC.

PT Programming Manual

Provides step-by-step instructions for performing system programming using a PT.

User Manual

Provides operating instructions for end users using IP-PTs, DPTs, APTs, SIP phones, SLTs, PSs, or DSS Consoles.

Other Information

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Note

- The contents of this manual apply to PBXs with a certain software version, as indicated on the cover of this manual. To confirm the software version of your PBX, refer to How do I confirm the software version of the PBX or installed cards? in 2.3 Frequently Asked Questions (FAQ) of the PC Programming Manual, or [190] Main Processing (MPR) Software Version Reference in the PT Programming Manual.
- 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.
- Throughout this manual, PT 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-NS300**BX**) is omitted unless necessary.
- All system programming can be performed through PC programming (→ 5.5.2 PC Programming). However, only a subset can be performed through PT programming (→ 5.5.3 PT Programming). In Section 1 Call Handling Features and Section 2 System Configuration and Administration Features, programming references that include a three-digit number, such as "000" indicate that system programming can be performed through PT programming.

  PC Programming

  The number within the brackets indicates the system menu number for the Maintenance Console.

  → 14.1 PBX Configuration—[6-1] Feature—System Speed Dial—◆ CO Line Access Number + Telephone Number

  PT Programming

  The number within the brackets indicates the programming number that is entered when performing PT programming.

  → [001] System Speed Dialling Number

For further details, please refer to the PC Programming Manual and PT Programming Manual.

List of Abbreviations

A

AA
Automated Attendant

ACD
Automatic Call Distribution

ANI
Automatic Number Identification

AOC
Advice of Charge

APT
Analogue Proprietary Telephone

ARS
Automatic Route Selection

B

BGM
Background Music

C

CA
Communication Assistant

CCBS
Completion of Calls to Busy Subscriber

CDPG
Call Distribution Port Group

CF
Call Forwarding—by ISDN

CLI
Calling Line Identification

CLIP
Calling Line Identification Presentation

CLIR
Calling Line Identification Restriction

CNIP
Calling Name Identification Presentation

CNIR
Calling Name Identification Restriction

COLP
Connected Line Identification Presentation

COLR
Connected Line Identification Restriction

CONP
Connected Name Identification Presentation

CONR
Connected Name Identification Restriction

COS
Class of Service

CPC
Calling Party Control

CS
Cell Station

CT
Call Transfer—by ISDN

CTI
Computer Telephony Integration

D

DDI
Direct Dialling In

DHCP
Dynamic Host Configuration Protocol

DID
Direct Inward Dialling

DIL
Direct In Line

DISA
Direct Inward System Access

DND
Do Not Disturb

DPT
Digital Proprietary Telephone

DSS
Direct Station Selection

DTMF
Dual Tone Multi-Frequency

E

EFA
External Feature Access

F

FWD
Call Forwarding

G

G-CO
Group-CO

I

ICD
Incoming Call Distribution

ICMP
Internet Control Message Protocol

IP-PT
IP Proprietary Telephone

IRNA
Intercept Routing—No Answer

ISDN
Integrated Services Digital Network

L

L-CO
Loop-CO

LCS
Live Call Screening

LED
Light Emitting Diode

M

MCID
Malicious Call Identification

N

NDSS
Network Direct Station Selection

NTP
Network Time Protocol

O

OGM
Outgoing Message

OHCA
Off-hook Call Announcement

P

P2P
Peer-to-Peer

PDN
Primary Directory Number

PIN
Personal Identification Number

PING
Packet Internet Groper

P-P
Point-to-Point

PRI
Primary Rate Interface

PS
Portable Station

PT
Proprietary Telephone

S

S-CO
Single-CO

SDN
Secondary Directory Number

SIP
Session Initiation Protocol

SLT
Single Line Telephone

SMDR
Station Message Detail Recording

SNMP
Simple Network Management Protocol

S-PS
SIP-CS compatible Portable Station

T

TAFAS
Trunk Answer from Any Station

TEI
Terminal Endpoint Identifier

TRG
Trunk Group

TRS/Barring
Toll Restriction/Call Barring

U

UCD
Uniform Call Distribution

UM
Unified Messaging

UPS
Uninterruptible Power Supply

V

VM
Voice Mail

VoIP
Voice over Internet Protocol

VPN
Virtual Private Network

VPS
Voice Processing System

X

XDP
EXtra Device Port

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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/Barring and ARS features to allow user access to the network must be upgraded to recognise 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 recognise 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 unauthorised 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 no passwords set initially. For security, select 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 10 numbers or characters be used for maximum protection against unauthorised access.

Section 2

Call Handling Features

2.1 Incoming Call Features

2.1.1 Incoming Trunk Call Features

2.1.1.1 Incoming Trunk Call Features—SUMMARY

Description

Incoming calls via a trunk (public line) are distributed to their destinations according to one of several distribution methods.

1. **Available Networking Type for Each Card Type**
   Each trunk port of an optional trunk card or the mother board can be assigned a networking type: Public, Private, or VPN (Virtual Private Network).

Card Type	Channel/ Protocol Type	Networking Type		
		Public (DIL/DID/ DDI)	Private (TIE)*1	Virtual Private Network (VPN)
Mother Board (V-IPGW)	H.323	✓	✓*	✓
Mother Board (V-SIPGW)	SIP	✓*		
LCOT	—	✓*		
PRI	CO	✓*		✓*2
	Extension			
	QSIG-Master		✓*	
	QSIG-Slave		✓*	
E1	DR2	✓*	✓	

✓*: Enable (default); ✓: Enable

*1 → 4.2.1 TIE Line Service

*2 → 4.2.3 ISDN Virtual Private Network (ISDN-VPN)

2. **Distribution Method**
   One of the following methods can be assigned to each trunk port:

Method	Description & Reference
Direct In Line (DIL)	Directs a call to a preprogrammed single destination (e.g., the operator). → 2.1.1.2 Direct In Line (DIL)

Method	Description & Reference
Direct Inward Dialling (DID)	Directs a call with a DID number from a DID line to a preprogrammed destination. DID is also known as Direct Dialling In (DDI). → 2.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)

3. Destination Change with the Caller's Identification Number

The Calling Line Identification (CLI) Distribution feature works in conjunction with the DIL/DID/DDI features.

Feature	Description & Reference
Calling Line Identification (CLI) Distribution	Directs a call to a CLI destination if the caller's identification number has been assigned in the Caller ID Table. → 2.1.1.4 Calling Line Identification (CLI) Distribution

4. Available Distribution Feature for Each Optional Trunk Card Type

Trunk Card Type	Channel Type	Feature	
		DIL	DID/DDI
Mother Board (V-IPGW)	—	✓	✓*
Mother Board (V-SIPGW)	—	✓	✓*
LCOT	—	✓*	
PRI	CO	✓	✓*
E1	DR2	✓	✓*

✓*: Enable (default); ✓: Enable

5. Available Destinations

Destination	Availability
Wired Extension (PT/SLT/SIP Extension/ISDN Extension)	✓
PS	✓
Incoming Call Distribution Group	✓
PS Ring Group	✓
UM Group	✓
VM Group (DTMF/DPT)	✓
External Pager (TAFAS)	✓
DISA	✓
Analogue/ISDN Remote Maintenance	✓
Idle Line Access no. + Phone no.	

Destination	Availability
Trunk Group Access no. + Trunk Group no. + Phone no.	
Other PBX Extension (TIE with no PBX Code)	✓
Other PBX Extension (TIE with PBX Code)	

6. Intercept Routing

After setting distribution, it may also be necessary to set the following features.

Feature		Description & Reference
Intercept Routing	**No Answer (IRNA)**	If a called party does not answer a call within a preprogrammed time period (Intercept time), it is redirected to the preprogrammed destination. → 2.1.1.5 Intercept Routing
	Busy/DND	If a called party is busy or in DND mode, the call is redirected to the preprogrammed destination. → 2.1.1.5 Intercept Routing
	No Destination	If a destination is not assigned, the call is redirected to the operator. → 2.1.1.6 Intercept Routing—No Destination

PC Programming Manual References

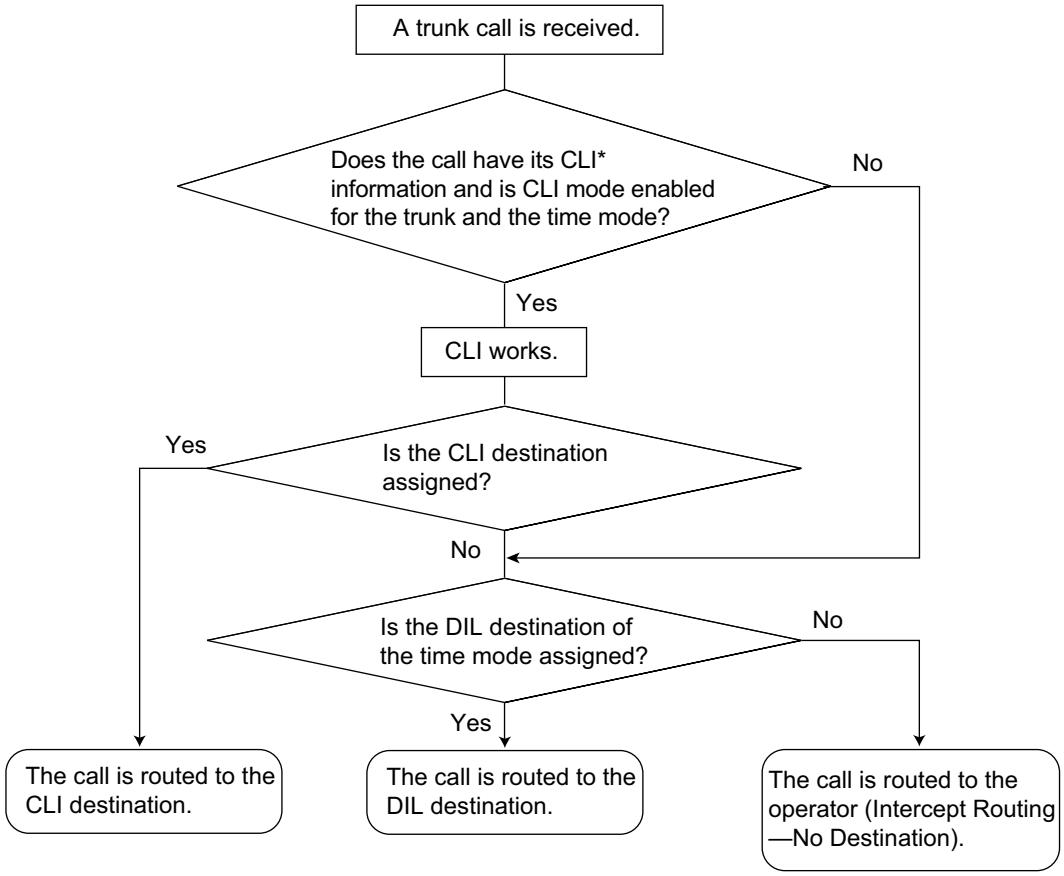
18.2 PBX Configuration—[10-2] CO & Incoming Call—DIL Table & Port Settings—DIL—◆ Trunk Property

2.1.1.2 Direct In Line (DIL)

Description

Provides automatic direction of an incoming trunk call to a preprogrammed destination. Each trunk has a destination for each time mode (day/lunch/break/night).

[Method Flowchart]



*: Calling Line Identification (CLI) Distribution:
If the CLI routing is enabled and the caller's identification number is assigned in the Caller ID Table, the call will not be routed to the DIL destination, but routed to the CLI destination.

[Programming Example of DIL Table]

The table can be programmed for each trunk.

Trunk No.	CLI			Destination*1		
	Day	Lunch	...	Day	Lunch	...
01	Enable	Disable	...	101	100	...
02	Enable	Disable	...	102	100	...
:	:	:	:	:	:	:

*1 → 18.2 PBX Configuration—[10-2] CO & Incoming Call—DIL Table & Port Settings—DIL—◆ DIL Destination—Day, Lunch, Break, Night

Note

The following settings can also be specified in the DIL table:

- Tenant number: determines the time mode (day/lunch/break/night) for the corresponding trunk.
- UM service group number: determines the service group to use when a call is handled by the Unified Messaging system.
- VM trunk group number: used in Voice Mail DPT (Digital) Integration with a VPS.

Explanation:
If a trunk call is received from trunk 01;
In Day mode: CLI is enabled. Route to CLI destination.
In Lunch mode: CLI is disabled. Route to DIL destination, extension 100.

PC Programming Manual References

18.2 PBX Configuration—[10-2] CO & Incoming Call—DIL Table & Port Settings—DIL
→◆ DIL Destination—Day, Lunch, Break, Night
→◆ Tenant Number
→◆ UM Service Group No.
→◆ VM Trunk Group No.

PT Programming Manual References

[450] DIL 1:1 Destination

Feature Guide References

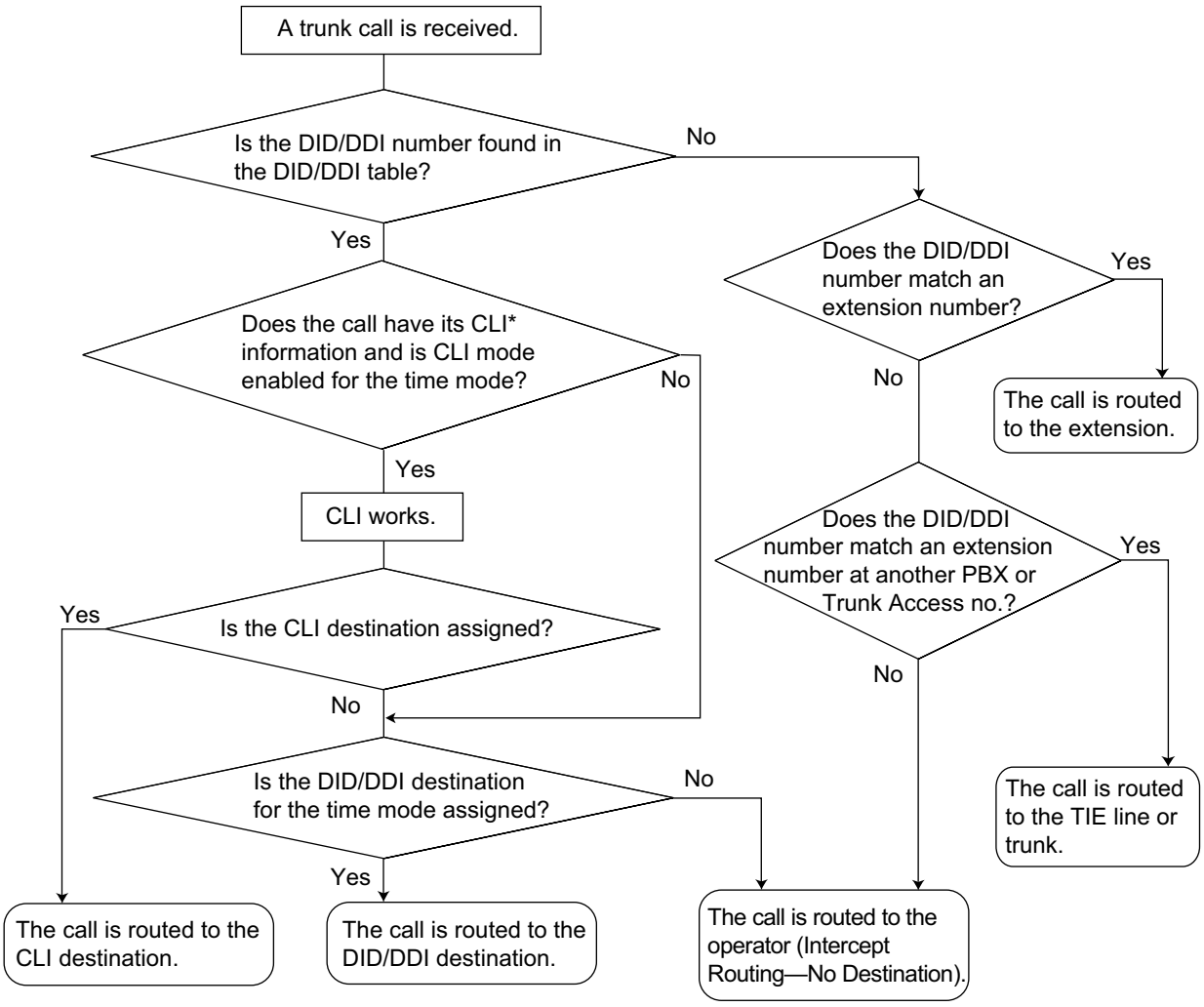
2.1.1.4 Calling Line Identification (CLI) Distribution
3.2.1.39 Service Group
5.1.3 Tenant Service
5.1.4 Time Service
6.1 Capacity of System Resources

2.1.1.3 Direct Inward Dialling (DID)/Direct Dialling 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/break/night).
Incoming calls with DID/DDI numbers that match extension numbers at this PBX will be sent to the corresponding extension. Incoming calls with DID/DDI numbers that match extensions at other PBXs or trunk access numbers will be sent to the corresponding TIE line or trunk.

[Method Flowchart]



*: Calling Line Identification (CLI) Distribution:
If the CLI routing is enabled and the caller's identification number is assigned in the Caller ID Table, the call will not be routed to the DID/DDI destination, but routed to the CLI destination.

[Programming Example of DID/DDI Table]

DDI can be programmed as DID.

Location	No.*1	Name*2	CLI*3			Destination*4		
			Day	Lunch	...	Day	Lunch	...
0001	123-4567	John White	Enable	Disable	...	105	100	...
0002	123-2468	Tom Smith	Enable	Disable	...	102	100	...
0003	123-456	A company	Enable	Disable	...	101	101	...
:	:	:	:	:	:	:	:	:

*1 → 18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table—◆ DDI / DID Number
*2 → 18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table—◆ DDI / DID Name
*3 → 18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table—◆ CLI Ring for DDI/DID—Day, Lunch, Break, Night
*4 → 18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table—◆ DDI / DID Destination—Day, Lunch, Break, Night

Note

The following settings can also be specified in the DID/DDI table:

- Tenant number: determines the time mode (day/lunch/break/night) for the corresponding trunk.
- UM service group number: determines the service group to use when a call is handled by the Unified Messaging system.
- VM trunk group number: used in Voice Mail DPT (Digital) Integration with a VPS (→ 2.28.3 Voice Mail DPT (Digital) Integration).

Explanation:

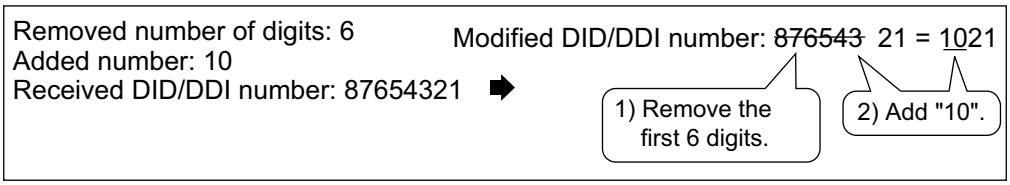
If the DID/DDI number is "123-4567":

1. Checks the number in the table.
   → Matches the number in location 0001.
2. Checks the time mode.
   In Day mode: CLI is enabled. Route to CLI destination.
   In Lunch mode: CLI is disabled. Route to DID/DDI destination, extension 100.

Conditions

- To use this feature, DID/DDI service must be assigned as the distribution method for a trunk port.
- **DID/DDI Number Modification**
  It is possible to modify a received DID/DDI number, which may be convenient when programming the DID/DDI table. The modification method (removed number of digits/added number) can be programmed on a trunk port basis.

  [Modification Example]



- **Inter-digit Time**
  When the Inter-digit time expires, the PBX stops receiving the DID/DDI number and starts to check the DID/DDI table. (Refer to the [Programming Example of DID/DDI Table] above).

Even if the Inter-digit time does not expire, the PBX stops receiving the DID/DDI number when the received number is found in the DID/DDI table. The PBX then routes the call to the corresponding destination. If the received number matches several entries in the table, the call is directed to the destination of the first matching entry.

[Example]

If a call is received in Lunch mode;

Received Number	Destination	Explanation
123-4567	Extn. 100	The PBX finds the match in location 0001 in the table after receiving "7". So the call is routed to extension 100.
123-456	Extn. 101	The Inter-digit time expired after receiving "6". The PBX finds the match in location 0003 in the table. So the call is routed to extension 101.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Incoming Call Inter-digit Timer—DDI / DID (s)

18.2 PBX Configuration—[10-2] CO & Incoming Call—DIL Table & Port Settings—DDI/DID/TIE

→◆ Distribution Method

→◆ DDI/DID/TIE—Remove Digit

→◆ DDI/DID/TIE—Additional Dial

18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table

PT Programming Manual References

[451] DID Number

[452] DID Name

[453] DID Destination

Feature Guide References

2.1.1.4 Calling Line Identification (CLI) Distribution

3.2.1.39 Service Group

5.1.3 Tenant Service

5.1.4 Time Service

6.1 Capacity of System Resources

2.1.1.4 Calling Line Identification (CLI) Distribution

Description

Directs an incoming trunk call to a preprogrammed destination when the caller's identification number (e.g., Caller ID) matches the number in the System Speed Dialling Table that is used as the Caller ID Table. Each Caller ID number (telephone number for each System Speed Dialling number) can have its own destination.

CLI Feature	Description & Reference
Caller ID	Caller's number is sent from an analogue trunk. → 2.19.1 Caller ID
Calling Line Identification Presentation (CLIP)	Caller's number is sent from an ISDN line. → 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP)
Automatic Number Identification (ANI)	Caller's number is sent from an E1 line. → 2.29.1 E1 Line Service

CLI always works in conjunction with the following call distribution methods:

a. DIL

b. DID/DDI

Each trunk (for DIL) and the DID/DDI number can enable or disable the CLI feature for each time mode (day/lunch/break/night) (→ 5.1.4 Time Service).

When the call has Caller ID information and the CLI is enabled for the time mode, the call will be handled by the CLI method.

[Programming Example of System Speed Dialling Table for CLI]

Location (System Speed Dialling No.)	System Speed Dialling Name*1	Telephone No.*2	CLI Destination*3
000	ABC Company	901234567890	200
001	:	:	:
:	:	:	:

*1 → 14.1 PBX Configuration—[6-1] Feature—System Speed Dial—◆ Name

*2 → 14.1 PBX Configuration—[6-1] Feature—System Speed Dial—◆ CO Line Access Number + Telephone Number

*3 → 14.1 PBX Configuration—[6-1] Feature—System Speed Dial—◆ CLI Destination

Explanation:

If the caller's number is "0123-456-7890" (The Trunk Access number is disregarded):

1. Checks the number in the table.
→ Matches the number in location 000.

2. The call is routed to the CLI destination, extension 200.

Conditions

- **Automatic Caller ID Number Modification**
  The Caller ID number is used after modification by the Automatic Caller ID Number Modification. (→ 2.19.1 Caller ID)

PC Programming Manual References

18.2 PBX Configuration—[10-2] CO & Incoming Call—DIL Table & Port Settings—CLI for DIL—◆ CLI Ring for DIL—Day, Lunch, Break, Night
18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table—◆ CLI Ring for DDI/DID—Day, Lunch, Break, Night
14.1 PBX Configuration—[6-1] Feature—System Speed Dial
→◆ Name
→◆ CO Line Access Number + Telephone Number
→◆ CLI Destination

PT Programming Manual References

[001] System Speed Dialling Number
[002] System Speed Dialling Name

Feature Guide References

2.1.1.2 Direct In Line (DIL)
2.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)
2.6.4 Speed Dialling—Personal/System

2.1.1.5 Intercept Routing

Description

Provides automatic redirection of incoming trunk and intercom calls. There are three types of Intercept Routing as follows:

Feature	Description
Intercept Routing—No Answer (IRNA)	If a called party does not answer a call within a preprogrammed time period (IRNA Timer), the call is redirected to the preprogrammed destination.
Intercept Routing—Busy	If a called party is already handling a call, new calls are handled as follows: – The call is redirected to the preprogrammed Intercept Routing—Busy destination. – If an Intercept Routing—Busy destination is not enabled, the caller will hear a busy tone. However, if the call is made through an LCOT card, the caller hears a ringback tone.
Intercept Routing—DND	If a called party is in DND mode, the call is redirected to the preprogrammed destination.

Intercept destinations can be assigned to extension ports.
→ 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Intercept Destination—◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night
→ 12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Intercept Destination—◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night

When the original destination is:	The Available Intercept Destination is:
• Wired Extension (PT/SLT/SIP Extension) • PS	The destination assigned to the original extension. 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Intercept Destination → ◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night → ◆ Intercept Destination—When Called Party is Busy 12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Intercept Destination → ◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night → ◆ Intercept Destination—When Called Party is Busy

When the original destination is:	The Available Intercept Destination is:
• ICD Group	The ICD Group Overflow destination assigned to the group. (→ 2.2.2.6 Overflow Feature) → 11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Overflow No Answer—◆ Time out & Manual Queue Redirection—Destination-Day, Lunch, Break, Night
• UM Group	The destination assigned to the first extension of the UM group.
• VM Group (DTMF/DPT)	The destination assigned to the first extension of the VM group.
• DISA[*1]	If all DISA ports are busy when a call is made using DISA, one of the following can be selected through system programming: • **Disable:** Busy tone is sent to the caller. When using an analogue trunk, a ringback tone is sent. • **Operator:** The call will be redirected to the operator. • **AA-0, AA-9:** The call will be redirected to the destination assigned to that AA number. → 13.3.1 PBX Configuration—[5-3-1] Optional Device—Voice Message—DISA System—Option 1—◆ DISA Intercept—Intercept when all DISA ports are busy
• PS Ring Group[*2] • External Pager (TAFAS)[*2] • Analogue/ISDN Remote Maintenance • Other PBX Extension (TIE with no PBX Code)	Not assignable (Intercept Routing is not available.)

[*1] Intercept Routing for DISA will redirect a call only if all of the Direct Inward System Access (DISA) ports are busy. Once the call reaches the destination extension by using the DISA feature, the Intercept Routing feature of the extension is used.

[*2] By assigning the forward destination of a Virtual PS to a PS Ring Group or external pager, and assigning Intercept Routing destinations to the Virtual PS, calls to these destinations will be redirected to the Intercept Routing destination of the Virtual PS.

Programming Example

<Forward Destination of Virtual PS>

Ext. No. of the Virtual PS	FWD Destination
2001	600
...	...

"600" is an example of a Floating Extension No. for the External Pager

<Intercept Destination of Virtual PS>

Ext. No. of the Virtual PS	Intercept Destination				...
	Day	Lunch	Break	Night	
2001	1001	1001	1001	1001	...
...	...	...	...	...	...

When extension 2001 is called from another extension or is the first destination of an incoming trunk call, etc., the call will ring at extension 600 (external pager) first, and then ring at the intercept destination (extension 1001) after the IRNA Timer expires.

Different intercept destinations can be programmed for each time mode (day/lunch/break/night).

[Available Intercept Destinations]

Intercept Destination	Availability
Wired Extension (PT/SLT/SIP Extension/ISDN Extension)	✓
PS	✓
Incoming Call Distribution Group	✓
PS Ring Group	✓
UM Group	✓
VM Group (DTMF/DPT)	✓
External Pager (TAFAS)	✓
DISA	✓
Analogue/ISDN Remote Maintenance	✓
Idle Line Access no. + Phone no.	✓
Trunk Group Access no. + Trunk Group no. + Phone no.	✓
Other PBX Extension (TIE with no PBX Code)	✓
Other PBX Extension (TIE with PBX Code)	✓

Intercept for calls to an outside destination

When an intercom, trunk, or DISA call is received by an extension and forwarded to an outside destination by FWD—All Calls or FWD—Busy, the Intercept Routing feature can be used, if the outside destination is busy or does not answer. For example, when a call is forwarded to a cellular phone by FWD—All Calls, and the cellular phone is busy, the call will be routed to Voice Mail in the PBX.

This feature is also available when a call is forwarded by FWD—No Answer under the following conditions:
– The forwarding destination is in a private network and is busy or does not answer.
– The forwarding destination is in a public network and does not answer.
However, this feature is not available when a call is forwarded by FWD—No Answer to a destination in a public network that is busy.

Conditions

- **Intercept Routing—DND on/off**
  Intercept Routing—DND can be enabled or disabled system programming.
  If disabled, one of the following is activated depending on the type of line that a call arrives through:
  a. **LCOT Card:** The incoming trunk call will ring at the original destination while the caller hears a ringback tone.
  b. **Other Trunk Cards:** A busy tone will be sent to the caller.
- **If the intercept destination cannot receive the call:**
  a. **Intercept Routing—No Answer:** Intercept timer will restart at the original destination, until the call is answered.
  b. **Intercept Routing—Busy/DND:** The call will be sent back to the original destination when the call arrives through the LCOT card. When the call arrives through other trunk cards the caller will hear a busy tone.
- **Idle Extension Hunting**
  If an extension is a member of an idle extension hunting group, calls to that extension will not be redirected by Intercept Routing—Busy/DND. If the extension is busy or in DND mode, calls to that extension will be redirected to the next extension in the idle extension hunting group.
- Intercept Routing for intercom calls can be enabled or disabled on a system basis.
  → 18.4 PBX Configuration—[10-5] CO & Incoming Call—Miscellaneous—◆ Intercept—Intercept Routing for Extension Call
- **IRNA Timer**
  The IRNA timer can be set on a system basis and an extension basis for each time mode (day, lunch, break, night).
  → 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Intercept No Answer Time—◆ Intercept No Answer Time—Day, Lunch, Break, Night
- The Intercept Routing destination for each time mode will not apply for Intercept Routing—Busy. When the original destination is busy, the call is redirected to the Intercept Routing—Busy destination assigned through system programming. If no destination is assigned the caller will hear a busy tone.
- The time modes that are selected for trunk calls arriving at extensions and UM groups are decided on a tenant basis.
- **Intercept for calls to an outside destination**
  – This feature for LCOT depends on the settings for reverse signal detection. (→2.5.4.5 Reverse Circuit)
  – This feature may not be available depending on the specifications of the telephone network.
  – This feature is not available when the original call was made from a SIP extension.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters
→Dial / IRNA / Recall / Tone—◆ Intercept Routing No Answer (IRNA)—Day (s), Lunch (s), Break (s), Night (s)
→DISA / Door / Reminder / U. Conf—◆ DISA—Intercept Timer—Day (s), Lunch (s), Break (s), Night (s)
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Overflow No Answer—◆ Time out & Manual Queue Redirection—Destination-Day, Lunch, Break, Night
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings

→Intercept Destination
→Intercept No Answer Time
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings
→Intercept Destination
→Intercept No Answer Time
13.3.1 PBX Configuration—[5-3-1] Optional Device—Voice Message—DISA System—Option 1—◆ DISA Intercept—Intercept when all DISA ports are busy
18.4 PBX Configuration—[10-5] CO & Incoming Call—Miscellaneous
→◆ Intercept—Intercept Routing - DND (Destination sets DND.)
→◆ Intercept—Routing to Operator - No Destination (Destination is not programmed.)
→◆ Intercept—Intercept Routing for Extension Call

PT Programming Manual References

[203] Intercept Time
[604] Extension Intercept Destination
[625] Destination for Overflow Time Expiration

Feature Guide References

2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features

2.1.1.6 Intercept Routing—No Destination

Description

Provides automatic redirection of incoming trunk calls that do not have a destination assigned. The intercept destination is an operator (tenant/PBX).

Conditions

- **Intercept Routing—No Destination on/off**
  The Intercept Routing—No Destination feature can be enabled or disabled through system programming. If disabled, a reorder tone will be sent to the caller. However, the Intercept Routing—No Destination feature always functions for calls through the LCOT card even when disabled.
- **If an operator (tenant/PBX) is not assigned:**
  The extension that is connected to the lowest-numbered port and is ready to receive calls will be the intercept destination.
- Intercept Routing—No Destination also applies to calls from doorphones.

PC Programming Manual References

10.2 PBX Configuration—[2-2] System—Operator & BGM—◆ PBX Operator—Day, Lunch, Break, Night
18.4 PBX Configuration—[10-5] CO & Incoming Call—Miscellaneous—◆ Intercept—Routing to Operator - No Destination (Destination is not programmed.)

PT Programming Manual References

[006] Operator Assignment

Feature Guide References

5.1.5 Operator Features

2.1.2 Internal Call Features

2.1.2.1 Internal Call Features—SUMMARY

Description

The following types of internal calls are available:

Feature	Description & Reference
Intercom Call	A call from one extension to another. → 2.5.3 Intercom Call
Doorphone Call	When a call from a doorphone reaches its destination, the recipient can talk to the visitor. → 2.18.1 Doorphone Call

[Available Destination]

The destinations of doorphone calls can be assigned for each time mode (day/lunch/break/night) (→ 5.1.4 Time Service) on a doorphone port basis.

Destination	Calling from	
	Extension	Doorphone
Wired Extension (PT/SLT/SIP Extension/ISDN Extension)	✓	✓
PS	✓	✓
Incoming Call Distribution Group	✓	✓
PS Ring Group	✓	✓
UM Group	✓	✓
VM Group (DTMF/DPT)	✓	✓
External Pager (TAFAS)	✓	✓
DISA		
Analogue/ISDN Remote Maintenance	✓	
Idle Line Access no. + Phone no.	✓	✓
Trunk Group Access no. + Trunk Group no. + Phone no.	✓	✓
Other PBX Extension (TIE with no PBX Code)	✓	✓
Other PBX Extension (TIE with PBX Code)	✓	✓

✓: Available

PC Programming Manual References

13.1 PBX Configuration—[5-1] Optional Device—Doorphone—◆ Destination—Day, Lunch, Break, Night

PT Programming Manual References

[720] Doorphone Call Destination

2.1.2.2 Internal Call Block

Description

Internal calls can be restricted on a COS basis. This is done by specifying which COS destinations are blocked for each COS.

[Programming Example]

Caller	Called Party			
	COS 1	COS 2	COS 3	...
COS 1				
COS 2		✓	✓	✓
COS 3	✓	✓		✓
:	:	:	:	:

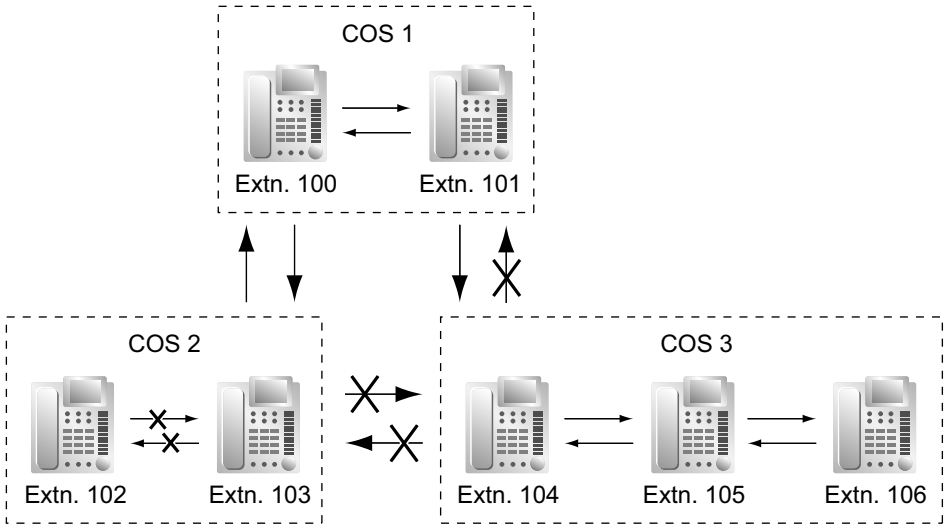
✓: Block

Explanation:

a. COS 1 extensions can make calls to all extensions.

b. COS 2 extensions can make calls to COS 1 destinations only. (COS 2 extensions cannot make calls to COS 2 destinations.)

c. COS 3 extensions can make calls to COS 3 destinations only.



Conditions

- Restricted extension numbers cannot be used as the parameter of a feature setting (e.g., FWD).
- All extensions can make an Operator Call (→ 5.1.5 Operator Features) regardless of Internal Call Block.
- This feature can also restrict calling a doorphone from an extension on the basis of the COSs assigned to the extension and doorphone port. (→ 2.18.1 Doorphone Call)

PC Programming Manual References

10.7.3 PBX Configuration—[2-7-3] System—Class of Service—Internal Call Block—◆ COS Number of the Extension Which Receive the Call from Other Extension 1–64
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Main—◆ COS
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ COS
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main—◆ COS
13.1 PBX Configuration—[5-1] Optional Device—Doorphone—◆ COS
13.4 PBX Configuration—[5-4] Optional Device—External Relay—◆ COS Number

Feature Guide References

5.1.1 Class of Service (COS)

2.1.3 Incoming Call Indication Features

2.1.3.1 Incoming Call Indication Features—SUMMARY

Description

Incoming calls are indicated by various methods as follows:

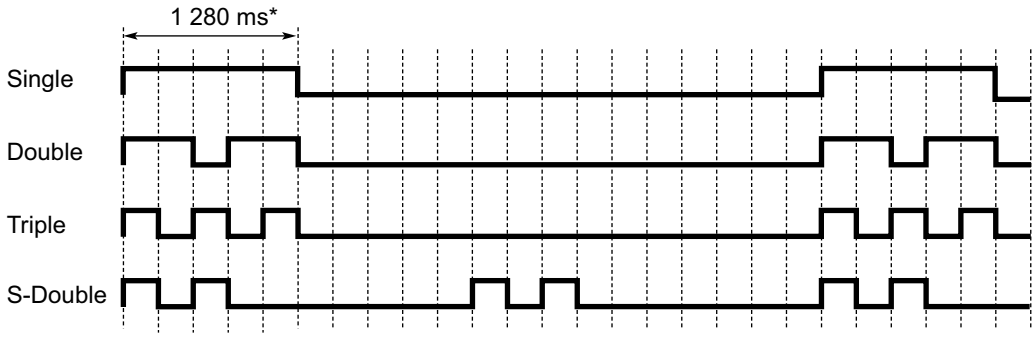
Type	Feature	Description & Reference
Ring Tone	**Ring Tone Pattern Selection**	A telephone rings when receiving a call. The ring tone patterns can be changed for each incoming call type. → 2.1.3.2 Ring Tone Pattern Selection
Voice-calling	**Alternate Receiving—Ring/ Voice**	A PT user can select to receive intercom calls by ring tone or by voice, through personal programming. → 2.5.3 Intercom Call
LED (Light Emitting Diode)	**LED Indication**	The light shows line conditions with a variety of light patterns. → 2.21.3 LED Indication
Display (Caller's Information)	**Display Information**	The display shows the caller's information. → 2.21.4 Display Information
External Pager	**Trunk Answer from Any Station (TAFAS)**	The external pager sends a ring tone when receiving a call. → 2.17.2 Trunk Answer From Any Station (TAFAS)
Tone/Voice during a Conversation	**Call Waiting**	A busy extension hears a tone, or voice from the handset/built-in speaker indicating that another incoming call is waiting. → 2.1.3.3 Call Waiting

2.1.3.2 Ring Tone Pattern Selection

Description

It is possible to select the type of ring tone pattern that arrives at an extension for each type of incoming call, etc.

[Ring Tone Patterns]



* The duration of a ring tone may vary by country/area.

[Ring Tone Pattern Table]

The ring tone pattern table is categorised into three parts, each containing a specified number of pattern plans. The ring tone pattern table is categorised as follows:

- Incoming Trunk Calls: each pattern plan can assign a ring tone pattern for each trunk group.
  → 10.8.1 PBX Configuration—[2-8-1] System—Ring Tone Patterns—Call from CO
- Incoming Doorphone Calls: each pattern plan can assign a ring tone pattern for each doorphone.
  → 10.8.2 PBX Configuration—[2-8-2] System—Ring Tone Patterns—Call from DOORPHONE
- Others: each pattern plan can assign a ring tone pattern for incoming intercom calls as well as ring tones assigned to certain features (e.g., timed reminder).
  → 10.8.3 PBX Configuration—[2-8-3] System—Ring Tone Patterns—Call from Others

The ring tone patterns that arrive at an extension are determined by the pattern plan that is assigned to that extension through system programming.

[Programming Example of Ring Tone Pattern Table]

Table No.	Intercom Call/Hold Recall	Trunk Call/ Hold Recall		Doorphone Call		Timed Reminder	Call Back	LCS	External Sensor
		TRG1	...	Port 1	...				
1	Double	Single		Single					
2	Single	Double		Double					
:	:	:	:	:	:	:	:	:	:

Conditions

- "**PT Ring Off Setting**" can be enabled or disabled through system programming. If disabled, PT users cannot turn incoming call ringing off for their extension.
- For the S-CO, G-CO, L-CO, ICD Group, INTERCOM, PDN and SDN buttons, one of 30 ring tones can be assigned through personal programming. (DPT/IP-PT only)
- On extensions using KX-UT series SIP phones, only the TRG1 setting for Trunk Call/Hold Recall is valid.

- For the One-numbered Extension feature, the ring tone pattern for the main extension and the sub extension can be set individually. (→ 2.11.11 One-numbered Extension)
- For KX-UT series SIP phones, changes to the ring tone pattern are applied after the telephone is restarted.

PC Programming Manual References

10.8.1 PBX Configuration—[2-8-1] System—Ring Tone Patterns—Call from CO—◆ Ring Tone Pattern Plan 1–8
10.8.2 PBX Configuration—[2-8-2] System—Ring Tone Patterns—Call from DOORPHONE—◆ Ring Tone Pattern Plan 1–8
10.8.3 PBX Configuration—[2-8-3] System—Ring Tone Patterns—Call from Others—◆ Extension—Ring Tone Pattern Plan 1–8
10.9 PBX Configuration—[2-9] System—System Options—Option 1—◆ PT Operation—PT Ring Off Setting
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Option 1—◆ Ring Pattern Table
→Option 6—◆ ICM Tone
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Optional Parameter (Ringing Tone Type Number) (for Loop CO, Single CO, Group CO, ICD Group, SDN)
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 1—◆ Ring Pattern Table

Feature Guide References

6.1 Capacity of System Resources

User Manual References

3.1.3 Customising the Buttons—To specify the ringing tones for each CO, ICD Group, PDN, SDN, or INTERCOM button (DPT/IP-PT only)

2.1.3.3 Call Waiting

Description

Used to inform a busy extension that another incoming call is waiting. The busy extension user can answer the second call by disconnecting the current call or placing it on hold. This feature is also known as Busy Station Signalling (BSS).
The following notification method can be assigned for each extension depending on the call waiting and the telephone type:

a. **Call Waiting Tone:** Tone from the handset or built-in speaker
b. **OHCA:** Voice from the built-in speaker
c. **Whisper OHCA:** Voice from the handset
d. **Off:** No notification.

Call Type	Notification Method		
	DPT	IP-PT	Other Telephone
Intercom Call	Call Waiting tone/OHCA/ Whisper OHCA/Off	Call Waiting tone/ Whisper OHCA/Off	Call Waiting tone/Off
Trunk Call[*1]	Call Waiting tone/Off		

[*1] Including a doorphone call, call via an incoming call distribution group, and a trunk call transferred from another extension.

Conditions

- **Automatic Call Waiting**
  Through system programming, it is possible to select whether a call waiting tone is automatically sent to the extension when receiving trunk calls, doorphone calls, external sensor calls and hold-recall calls. Through system programming, it is also possible to select whether extensions will receive Automatic Call Waiting from intercom calls.
- Call Waiting for an extension in a UM group or in a VM group (DTMF/DPT) is not available.
- **Data Line Security**
  Setting Data Line Security cancels the Call Waiting setting. (→ 2.11.5 Data Line Security)
- **Call Waiting Tone**
  A PT user can hear different Call Waiting tones for trunk call and intercom call if "**Tone 2**" has been selected through personal programming (Call Waiting Tone Type Selection). If "**Tone 1**" has been selected, the same Call Waiting tone will be heard for both trunk call and intercom call.
  All Call Waiting tone patterns have a default (→ 6.2.1 Tones/Ring Tones).
- **Caller Information**
  With the Call Waiting tone, the caller's information flashes on the display for five seconds, followed by a 10-second pause, then flashes again for five seconds.
- **Call Waiting from the Telephone Company**
  Besides the Call Waiting service within the PBX, the Call Waiting tone offered by an analogue line from the telephone company informs the extension user of another incoming trunk call that is waiting. He can answer the second call by disconnecting the current call or placing it on hold using EFA. (→2.11.7 External Feature Access (EFA)) For details, consult your telephone company.
  Call Waiting Caller ID (Visual Caller ID):
  When using the call waiting tone supplied by the telephone company over analogue lines, the waiting caller's telephone number can be received. The number will flash on the display for five seconds, followed by a 10-second pause, then flash again for five seconds.
  Note that the received caller information will not be displayed on telephones or wireless phones connected to SLT ports.

PC Programming Manual References

9.23 PBX Configuration—[1-1] Configuration—Slot—Card Property - LCO type—◆ Caller ID—Caller ID Signalling
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ Manual Call Waiting for Extension Call
→◆ Automatic Call Waiting
10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature
→◆ BSS / OHCA / Whisper OHCA / DND Override
→◆ BSS / OHCA / Whisper OHCA / DND Override-2
10.9 PBX Configuration—[2-9] System—System Options—Option 5—◆ Call Waiting—Automatic Call Waiting for Extension Call
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Option 2—◆ Manual C. Waiting for Extension Call
→Option 2—◆ Automatic C. Waiting
→Option 4—◆ Call Waiting Tone Type
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings
→Option 2—◆ Manual C. Waiting for Extension Call
→Option 2—◆ Automatic C. Waiting
→Option 4—◆ Call Waiting Tone Type

Feature Guide References

2.10.4 Second Call Notification to Busy Extension
2.19.2 Incoming Call Log
2.22.1.1 Station Message Detail Recording (SMDR)

User Manual References

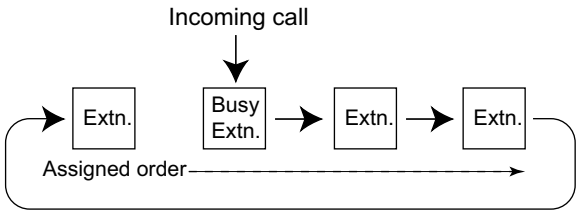
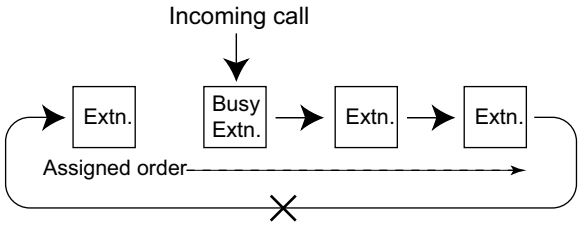
1.4.4 Answering Call Waiting
1.9.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)
3.1.2 Settings on the Programming Mode

2.2 Receiving Group Features

2.2.1 Idle Extension Hunting

Description

If a called extension is busy or in DND mode, Idle Extension Hunting redirects the incoming call to an idle member of the same idle extension hunting group, which can be programmed through system programming. Idle extensions are automatically searched according to a preprogrammed hunting type. This feature is also known as Station Hunting.

Type	Description
Circular Hunting	An idle extension is searched for in the order specified in the idle extension hunting group in a circular way. 
Terminated Hunting	An idle extension is searched for in the order specified in the idle extension hunting group until reaching the last assigned extension. 

Conditions

- **Idle Extension Hunting applies to:**
  Intercom, trunk, and doorphone calls to a single destination.
- An extension user can belong to only one idle extension hunting group.
- **If all the searched extensions are busy:**
  The PBX redirects the call to an overflow destination which can be assigned for each idle extension hunting group and each time mode (day/lunch/break/night) (→ 5.1.4 Time Service).

[Available Destination]

Destination	Availability
Wired Extension (PT/SLT/SIP Extension/ISDN Extension)	✓
PS	✓
Incoming Call Distribution Group	✓
PS Ring Group	✓

Destination	Availability
UM Group	✓
VM Group (DTMF/DPT)	✓
External Pager (TAFAS)	✓
DISA	✓
Analogue/ISDN Remote Maintenance	
Idle Line Access no. + Phone no.	✓
Trunk Group Access no. + Trunk Group no. + Phone no.	✓
Other PBX Extension (TIE with no PBX Code)	✓
Other PBX Extension (TIE with PBX Code)	✓

- **FWD/DND Mode**
  While searching for an idle extension within an idle extension hunting group, any extension that has set FWD—All Calls or DND feature will be skipped, and the call will go to the next extension in the group.

PC Programming Manual References

11.6 PBX Configuration—[3-6] Group—Extension Hunting Group
11.6.1 PBX Configuration—[3-6] Group—Extension Hunting Group—Member List

PT Programming Manual References

[680] Idle Extension Hunting Type
[681] Idle Extension Hunting Group Member

Feature Guide References

2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features
6.1 Capacity of System Resources

2.2.2 Incoming Call Distribution Group Features

2.2.2.1 Incoming Call Distribution Group Features—SUMMARY

Description

An incoming call distribution group is a group of extensions programmed through system programming.
→ 11.5.1.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Member List

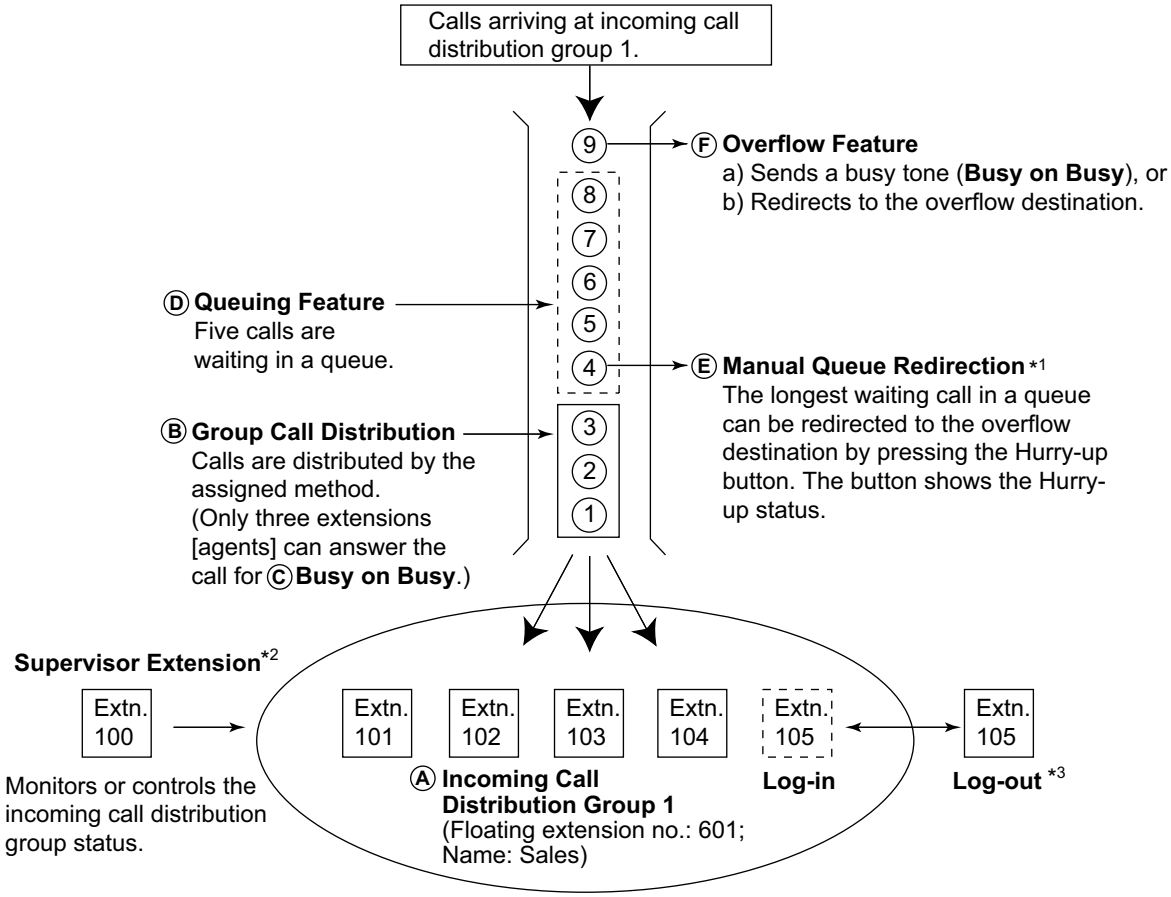
An incoming call distribution group receives calls directed to the group. Each incoming call distribution group has a floating extension number (default: 6 + two-digit group number [up to group 64]).
Incoming calls directed to an incoming call distribution group are distributed to the member extensions in the group using a distribution method. When a preprogrammed number of extensions in the group are busy, the incoming calls can wait in a queue.
Each incoming call distribution group and member extensions can be programmed as desired to handle incoming calls. Calls to the group can be monitored by an extension assigned as a supervisor.
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings
→ Main
→ Overflow Queuing Busy
→ Overflow No Answer
→ Miscellaneous

Programming Item Example for Incoming Call Distribution Group 1 with Diagram
A through F in the table are described in the following diagram.

A			B	C	D	E	F					
Group No.	Floating Extn. No.*1	Group Name	Distribution Method	Max. No. of Busy Extensions	Queuing Call Capacity	Hurry-up Level	Overflow Time	Overflow Destination				Tenant No.*2
								Day	···	···	Night	
1	601	Sales	Ring	3	5	3	60	100	···	···	100	1
2	602	Engineering	UCD	Max.	11	8	90	200	···	···	200	5
3												
:												

*1 The number of digits for **Floating Extn. No** depends on the value specified for **Numbering Plan** in Easy Setup.
→ 5.4.1 Easy Setup Wizard

*2 The tenant number is required to determine the time mode (day/lunch/break/night) (→ 5.1.4 Time Service) and the music source (for Music on Hold) for each group.



*1 → 2.2.2.4 Queuing Feature
*2 → 2.2.2.8 Supervisory Feature
*3 → 2.2.2.7 Log-in/Log-out

1. **Group Call Distribution** [→ 2.2.2.2 Group Call Distribution]
   Incoming calls are distributed using one of the following methods:

Distribution Method	Description
Uniform Call Distribution (UCD)	Calls are distributed evenly to a different extension each time a call is received.
Priority Hunting	An idle extension is searched for in the specified order, always starting from the same location.
Ring	All extensions in the incoming call distribution group ring simultaneously.

2. **Queuing Feature** [→ 2.2.2.4 Queuing Feature]
   If a preprogrammed numbers of extensions in an incoming call distribution group are busy, a preprogrammed number of additional calls can wait in a queue.
   While calls are waiting in the queue, an outgoing message (OGM) or Music on Hold can be sent to the waiting callers.
3. **VIP Call** [→ 2.2.2.5 VIP Call]
   It is possible to assign a priority to incoming call distribution groups so that incoming calls can be received in priority order.
4. **Overflow Feature** [→ 2.2.2.6 Overflow Feature]

A call is redirected to a preprogrammed destination when it cannot be answered or queued (**Intercept Routing—Overflow in an Incoming Call Distribution Group**). It is also possible to send a busy tone (**Busy on Busy**) or disconnect the line.

5. **Incoming Call Distribution Group Controlling Feature**

Feature		Description & Reference
Log-in/Log-out		Member extensions can join the group to handle calls (Log-in) or leave the group for a break (Log-out). They can leave the group temporarily when they are away from their desks, to prevent calls being sent to their extensions. → 2.2.2.7 Log-in/Log-out
Supervisory Feature	**Incoming Call Queue Monitor**	The supervisor extension can monitor various information about the incoming calls for each incoming call distribution group on his display. → 2.2.2.8 Supervisory Feature
	Log-in/Log-out Monitor and Remote Control	**Monitor:** The supervisor extension can monitor the log-in/log-out status of the group members. **Remote Control:** The supervisor extension can change the status of the members. → 2.2.2.8 Supervisory Feature

Conditions

- One extension can belong to multiple incoming call distribution groups.
- **ICD Group button**
  An Incoming Call Distribution (ICD) Group button can be assigned on a flexible button for each incoming call distribution group. It receives the incoming calls to the group.
  One extension can have more than one ICD Group button of the same or different incoming call distribution groups (**Multiple ICD Group**). If all ICD Group buttons in the same incoming call distribution group are occupied, the next incoming call will be held in a queue or will overflow. If the ICD Group button is not assigned, incoming calls will arrive at the INTERCOM, CO or PDN button.
  The mode of ICD Group buttons can be selected through system programming, as follows:
  – Standard Mode (Group DN Button Mode)
    An extension can have an ICD Group button for an incoming call distribution group that the extension does not belong to through system programming. However, the ICD Group button will not receive calls to that group.
  – Enhanced Phantom Button Mode
    An extension can join an ICD Group just by creating a button for that group, even if the extension was not previously registered as a member. When the button is created, the extension will be automatically registered in the lowest-numbered available member slot for the group. Calls to the group can be received at the extension with no further programming. If no member slots are available for that group, the button cannot be created and an alarm tone will be heard.
    When creating an ICD Group button in this mode, the user can also specify the delayed ringing settings.
    If an extension user deletes the last ICD Group button at his extension for a certain group, he will also be deregistered as a member from that group.
- **Group FWD**

The FWD feature can be assigned on an incoming call distribution group basis.

- **COS for Incoming Call Distribution Groups**
  Each incoming call distribution group is assigned a COS number. Group FWD to an outside party can be enabled or disabled for each COS. The COS for incoming call distribution groups is also used for the Internal Call Block feature; when an extension user calls an incoming call distribution group, the PBX checks the COS of the calling extension against the COS of the incoming call distribution group (→ 2.1.2.2 Internal Call Block).

Installation Manual References

5.4.1 Easy Setup Wizard

PC Programming Manual References

11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings
→Main
→Overflow Queuing Busy
→Overflow No Answer
→Miscellaneous
11.5.1.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Member List
11.5.3 PBX Configuration—[3-5-3] Group—Incoming Call Distribution Group—Miscellaneous—◆ Options—ICD Group Key Mode

PT Programming Manual References

[620] Incoming Call Distribution Group Member
[621] Incoming Call Distribution Group Delayed Ringing
[622] Incoming Call Distribution Group Floating Extension Number
[623] Incoming Call Distribution Group Name
[624] Incoming Call Distribution Group Distribution Method
[625] Destination for Overflow Time Expiration
[626] Overflow Time
[627] Destination When All Busy
[628] Queuing Call Capacity
[629] Queuing Hurry-up Level
[630] Queuing Time Table
[631] Sequences in Queuing Time Table
[632] Maximum Number of Agents

Feature Guide References

2.3.2 Call Forwarding (FWD)
2.21.2 Flexible Buttons
5.5.8 Floating Extension
6.1 Capacity of System Resources

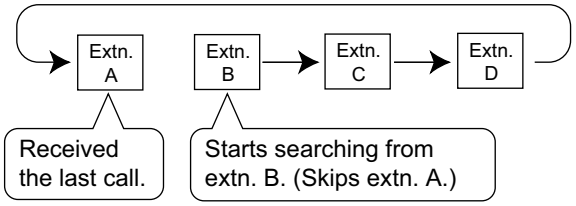
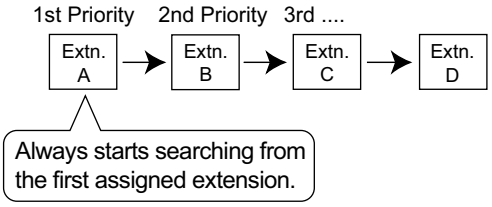
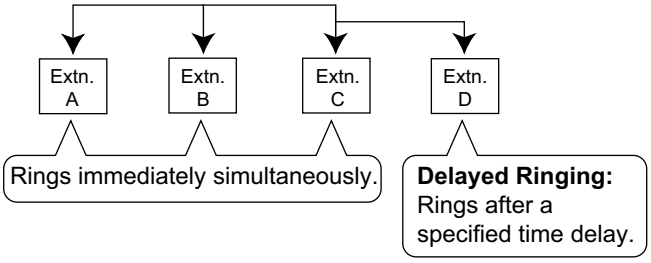
2.2.2.2 Group Call Distribution

Description

Incoming calls directed to an incoming call distribution group are distributed to the member extensions using the selected distribution method until a preprogrammed number of extensions (agents) are busy with calls. When incoming calls exceed the number of available extensions, calls enter a queue (→ 2.2.2.4 Queuing Feature).

1. Distribution Method

One of the three distribution methods below can be assigned to each incoming call distribution group.

Distribution Method	Description
Uniform Call Distribution (UCD)	Calls are distributed evenly to a different extension each time a call is received. 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. 
	Depending on system programming, calls can be directed to the longest idle extension. This is known as Automatic Call Distribution (ACD).
Priority Hunting	An idle extension is searched for using the preprogrammed order for the group. 
Ring	All extensions in the group ring simultaneously. **Delayed Ringing:** Delayed ringing or no ringing can be programmed for each extension in the group. The call can be answered by pressing the flashing button even if no ring or a delayed time is set. 

2. Call Waiting for Incoming Call Distribution Group (Group Call Waiting)

When there are no available extensions in an incoming call distribution group, the group members can receive the Call Waiting tone. To use this feature:

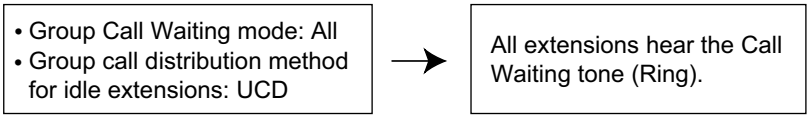
- Select the Group Call Waiting mode through system programming. This determines the distribution method for waiting calls.
- Member extensions must assign the Call Waiting mode individually, or they will not be notified. (→ 2.1.3.3 Call Waiting)

[How the Group Call Waiting Feature Activates]

Programming Conditions		Result	
Group Call Waiting Mode	**Group Call Distribution Method**	**Group Call Waiting Distribution Method**	**Capable Telephone**
Distribution	UCD	UCD	PT/PS with idle ICD Group button
	Priority Hunting	Priority Hunting	
	Ring	Not available*	Any telephone
All	UCD/Priority Hunting/ Ring	Ring	

* Incoming calls enter the queue immediately. Member extensions do not receive the Call Waiting tone.

[Example]



[ICD Group Button for Group Call Waiting]

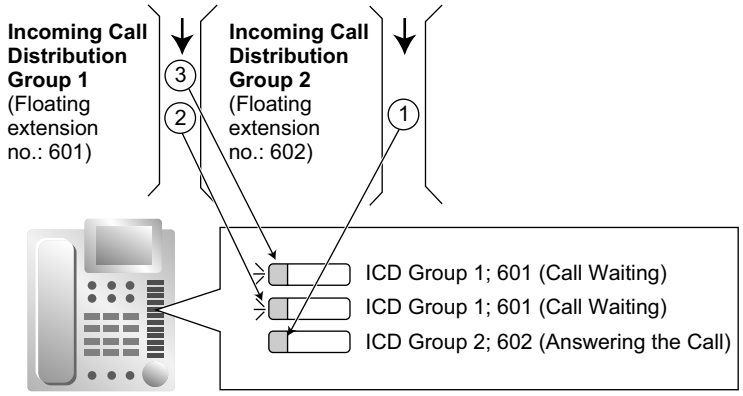
The way that the Group Call Waiting feature works depends on the Group Call Waiting Distribution method as follows:

a. Ring: The Group Call Waiting feature activates for all busy member extensions (even when the extensions do not have ICD Group buttons) simultaneously for only one incoming call—additional calls will wait in a queue.

b. UCD/Priority Hunting: The Group Call Waiting feature activates on an idle ICD Group button located on busy member extensions in a certain order. (This order depends on the type: UCD or Priority Hunting.) Calls will arrive at idle buttons until all ICD Group buttons are occupied—additional calls will wait in a queue.

Note

In method b), if an extension has one or more ICD Group buttons for an incoming call distribution group and all the ICD Group buttons on the extension are occupied, the Group Call Waiting feature for the group will not work at the extension.



3. **No Reply Redirection (UCD or Priority Hunting Method)**
   If a call received at a member extension is not answered within a preprogrammed time period (No Answer time), the call will be redirected to the next member extension. If there is no idle group member, the call queues at the target extension until a group member becomes available.

Conditions

- Automatic Call Distribution (ACD) does not work for ISDN extensions or PS Ring Groups.
- **FWD/DND Extension**
  System programming for each incoming call distribution group is required to skip or ring extensions which have the FWD or DND feature set. If set to ring, the FWD/DND settings are ignored. (→ 2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features)
- The Group Call Waiting feature cannot be used with the VIP Call feature (→ 2.2.2.5 VIP Call) and/or Wrap-up feature (→ 2.2.2.7 Log-in/Log-out). To use the VIP Call feature and/or Wrap-up feature, Call Waiting mode on each extension should be off.

PC Programming Manual References

11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Main—◆ Line Hunting Order
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings
→Main—◆ Distribution Method
→Main—◆ Call Waiting Distribution
→Miscellaneous—◆ Extension No Answer Redirection Time
→Miscellaneous—◆ Maximum No. of Busy Extension
11.5.1.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Member List—◆ Delayed Ring
11.5.3 PBX Configuration—[3-5-3] Group—Incoming Call Distribution Group—Miscellaneous
→◆ Options—Wrap-up Timer based on
→◆ Options—Longest Idle Distribution

PT Programming Manual References

[621] Incoming Call Distribution Group Delayed Ringing
[624] Incoming Call Distribution Group Distribution Method

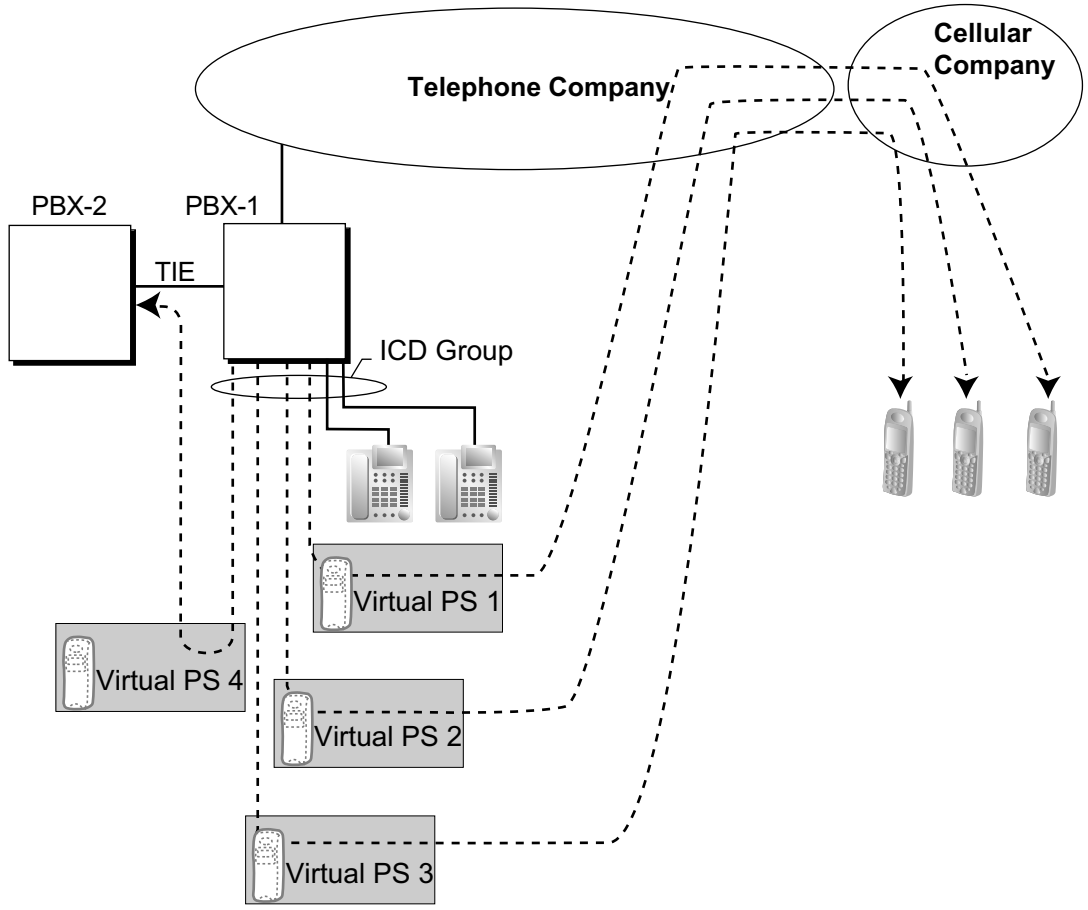
[632] Maximum Number of Agents

2.2.2.3 Outside Destinations in Incoming Call Distribution Group

Description

Up to 4 outside parties or destinations at another PBX can be assigned as members of an Incoming Call Distribution (ICD) Group, using the following method: A virtual PS is registered as a member of the ICD Group. Then, the telephone number of the outside destination is specified as the FWD—All Calls destination. Calls to the ICD Group will also ring at the outside destination as if that destination were an extension within the PBX. This is useful in situations such as the following:

- An extension user can have his PT and multiple cellular phones ring together for calls.
- An employee who is not in the office, but is still available to answer calls, can receive calls to an ICD Group.



Cellular Phone XDP Parallel

The extension registered first in an ICD Group can programme the Forward settings for trunk calls to up to 4 virtual PSs through PT programming. Using this feature, an extension user can assign his cellular phone to ring with his PT, so that he can easily receive trunk calls even when not at his desk.

Conditions

- KX-NSE101, KX-NSE105, KX-NSE110 or KX-NSE120 (Activation Key for Mobile Extension) is required to use this feature. One activation key is required for each extension (virtual PS) that will use this feature.
- For this feature to be activated, the following conditions must be met:
  - A virtual PS is assigned as a member of the ICD Group. (→ 5.2.4.6 Virtual PS)
  - The **Mobile Extension** setting for the virtual PS extension must be set to **Enable**.
  - The forwarding type of the virtual PS is set to All Calls. (→ 2.3.2 Call Forwarding (FWD))

  - The forwarding destination is an outside party, including an extension at another PBX in the network.
  - FWD to trunk is allowed through COS programming for the virtual PSs.
- Up to 4 virtual PSs can be assigned to a single ICD Group. If more than 4 are assigned, the 4 virtual PSs with the lowest member numbers are available.
- This feature for LCOT depends on the settings for reverse signal detection. (→2.5.4.5 Reverse Circuit)
- Calls to an ICD Group will ring at a virtual PS even if all of the other extensions assigned to the group are busy.
- If all members of an ICD Group are virtual PSs, and trunk lines are available but the called parties are all busy, neither the queuing or overflow features will operate. Therefore, it is recommended that at least one PT or SLT is also assigned to an ICD Group.
- To log in to or out of a group, a virtual PS user can access the PBX through DISA, enter the Walking COS feature number (if required), and access log-in/log-out settings.
- Delayed ringing can be assigned for virtual PSs in the same way as for other extensions.
- The Wrap-up time feature and Automatic Log-out feature are not available for virtual PSs.
- When forwarding calls to a public trunk, system programming selects whether the CLIP number of the calling party or of the virtual PS is sent to the forward destination.
  When calling using a private network, the CLIP number of the calling party will always be sent.
- **DSS button for Cellular Phone XDP Parallel**
  The DSS button light of the extension registered first in an ICD Group will turn red if the parallel cellular phone:
  - is on a trunk call that was received via the ICD Group.
  - is on a trunk call that was made using the Walking COS Through DISA feature. (→ 2.16.1 Direct Inward System Access (DISA))

PC Programming Manual References

10.9 PBX Configuration—[2-9] System—System Options—Option 4
→◆ Send CLIP of CO Caller—when call is forwarded to CO
→◆ Send CLIP of Extension Caller—when call is forwarded to CO
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 9—◆ Mobile Extension

Feature Guide References

5.2.4.6 Virtual PS
4.2.6 Network ICD Group

2.2.2.4 Queuing Feature

Description

When a preprogrammed number of extensions in an incoming call distribution group are busy, additional incoming calls can wait in a queue. The number of calls which can wait in the queue is programmable. While calls are waiting in the queue, the calls are handled by the Queuing Time Table, which can be assigned for each time mode (day/lunch/break/night) (→ 5.1.4 Time Service). This PBX supports a specified number of Queuing Time Tables which each have a specified number of sequences (specific commands which are performed when a caller enters a queue). The following commands can be assigned to each sequence when making a Queuing Time Table:

[Command Table]

Command	Description	Condition
OGM xx	An outgoing message is sent to the caller. "xx" applies to the OGM number.	After the OGM, Music on Hold will be sent and the next event in the sequence will be activated.
Wait xx s	Puts the caller in the waiting queue for b (01-16) × 5 seconds. "xx" applies to the number of seconds to wait (05-80).	If an OGM has not been sent to the caller, the caller hears a ringback tone. If an OGM has been sent to the caller, the caller hears Music on Hold.
Sequence xx	Redirects to sequence xx. "xx" applies to the sequence number.	None
Overflow	Redirects to the overflow destination.	None
Disconnect	Disconnects the line.	None
Queue No.	Announces the number of calls in the waiting queue ahead of the caller.	Activation key required: Call Centre Feature Enhancement (KX-NSF201)
Queue No. and Time	Announces the number of calls in the waiting queue ahead of the caller and the estimated waiting time.	Activation key required: Call Centre Feature Enhancement (KX-NSF201)
None (No command)	Redirects to the next sequence.	If assigned as sequence 01, the Queuing Time Table will not be activated.

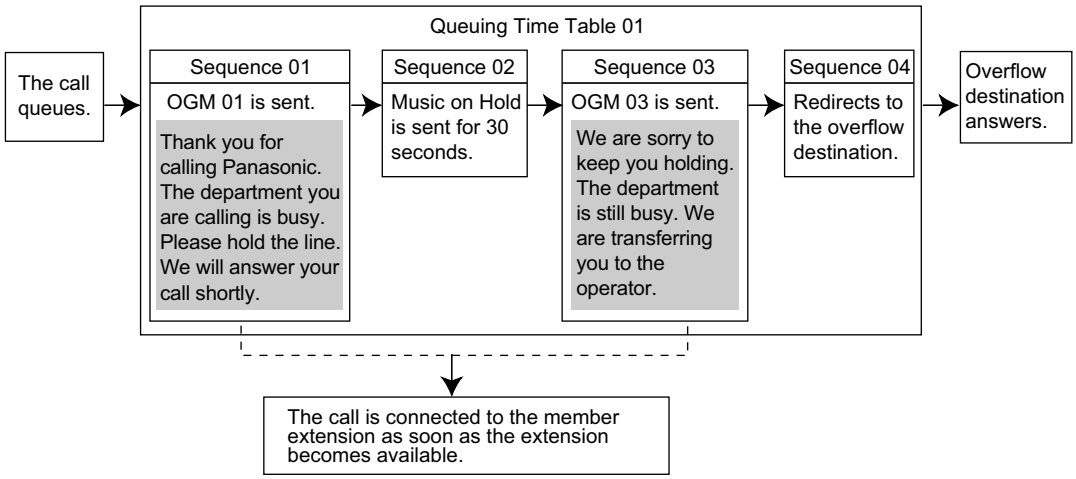
[Programming Example of Queuing Time Table]

Queuing Time Table No.	Sequence[*1]				
	Sequence 01	**Sequence 02**	**Sequence 03**	**Sequence 04**	**...**
01	OGM 01	Wait 30 s	OGM 03	Overflow	
02[*2]	OGM 02	Wait 30 s	OGM 04	None	
03					
:	:	:	:	:	:

[*1] → 11.5.2 PBX Configuration—[3-5-2] Group—Incoming Call Distribution Group—Queuing Time Table—◆ Queuing Sequence—Sequence 01–16

[*2] If a call has not reached a destination by the time the final sequence is completed, the call will be disconnected.

Explanation for Queuing Time Table 01:



Conditions

- **If the call is transferred to the incoming call distribution group and is handled by the Queuing Time Table:**
  Transfer Recall will not occur even if the Transfer Recall time expires.
- **Manual Queue Redirection**
  It is possible to redirect the longest waiting call in a queue to the overflow destination by pressing the Hurry-up button. (If the call is already ringing at an extension, it will not be redirected.) This feature is also known as Hurry-up Transfer.
- **Hurry-up Button**
  A flexible button can be customised as the Hurry-up button. The number of calls queuing before Manual Queue Redirection may be performed is programmable. The button shows the current status as follows:

Light Pattern	Calls in the Waiting Queue
Off	No queued call
Red on	At or under the assigned number for Hurry-up
Rapid red flashing	Over the assigned number for Hurry-up

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Incoming Call Queue Monitor
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings
→Overflow Queuing Busy
→Overflow No Answer
→Queuing Time Table
→Miscellaneous—◆ Extension No Answer Redirection Time
→Miscellaneous—◆ Maximum No. of Busy Extension
11.7.1 PBX Configuration—[3-7-1] Group—UM Group—System Settings—◆ Call Waiting on UM Group
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Extension Number (for Hurry-up)
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button

→◆ Type
→◆ Extension Number (for Hurry-up)

PT Programming Manual References

[628] Queuing Call Capacity
[629] Queuing Hurry-up Level
[630] Queuing Time Table
[631] Sequences in Queuing Time Table
[632] Maximum Number of Agents

Feature Guide References

2.2.2.6 Overflow Feature
2.30.2 Outgoing Message (OGM)
2.21.2 Flexible Buttons
6.1 Capacity of System Resources

User Manual References

1.10.3 Forwarding a Waiting Call (Manual Queue Redirection)

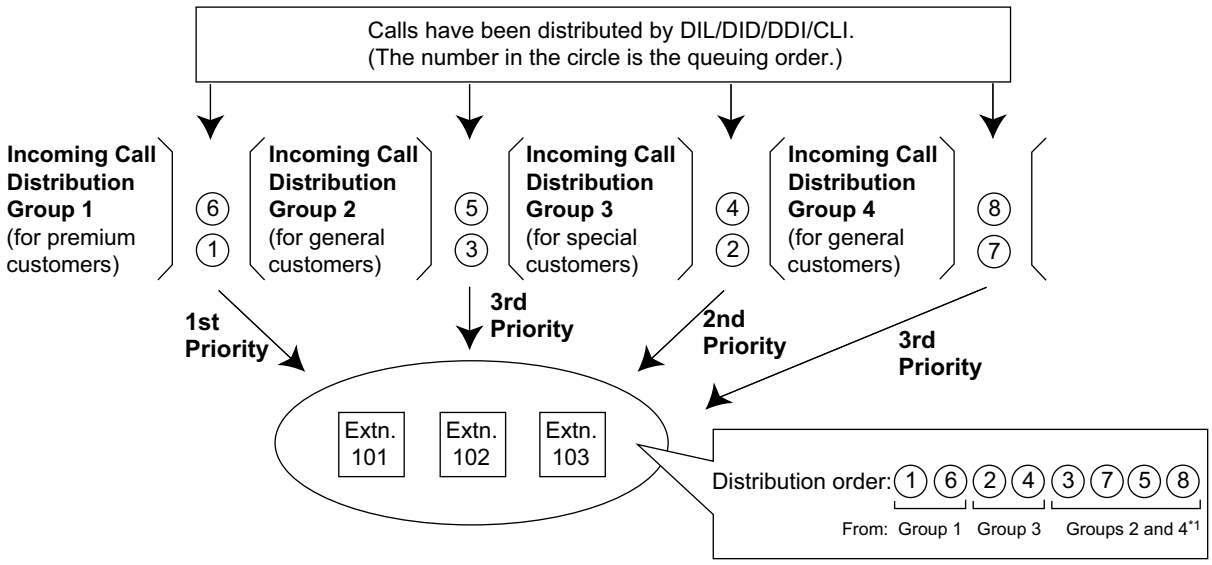
2.2.2.5 VIP Call

Description

It is possible to assign a priority to incoming call distribution groups. If an extension belongs to multiple groups and the extension becomes idle, queuing calls in the groups will be distributed to the extension in priority order. Each incoming call distribution group can enable or disable the VIP Call mode. When multiple groups enable the VIP Call mode, the incoming call distribution group with the lowest numbered group has the highest priority. When multiple groups disable the VIP Call mode, queuing calls are distributed, one from each ICD group in turn, irrespective of the order in which calls were received.

[Example]

In the call centre, incoming call distribution groups 1 and 3 enable the VIP Call mode, while incoming call distribution groups 2 and 4 disable the VIP Call mode.



*1 After call 3 is distributed from group 2, call 7 will be distributed from group 4 next, even though call 5 arrived earlier.

PC Programming Manual References

11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Miscellaneous—◆ VIP Call Mode

2.2.2.6 Overflow Feature

Description

When waiting calls exceed the waiting queue capacity, they may be redirected to a preprogrammed destination or a busy tone may be sent to the callers by the following features:

1. Intercept Routing—Overflow in an Incoming Call Distribution Group
2. Busy on Busy

1. **Intercept Routing—Overflow in an Incoming Call Distribution Group**
   Intercept Routing—Overflow in an Incoming Call Distribution Group works in one of following conditions:
   - **a.** There is no space in the waiting queue.
   - **b.** The Queuing Time Table is not assigned and there are no extensions logged in.
   - **c.** The Queuing Time Table is assigned, but there are no extensions logged in and the "**`Overflow immediately when All Logout`**" setting is enabled.
   - **d.** An Overflow command is assigned to the Queuing Time Table.
   - **e.** The Overflow time expires.
   - **f.** Manual Queue Redirection is performed.

[Available Destination]

The overflow destinations can be assigned for each incoming call distribution group and each time mode (day/lunch/break/night) (→ 5.1.4 Time Service). The destination can be assigned as follows, depending on the above conditions.

- For a), b), and c):
  → 11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Overflow Queuing Busy—◆ Queuing Busy—Destination-Day, Lunch, Break, Night
- For d), e), and f):
  → 11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Overflow No Answer—◆ Time out & Manual Queue Redirection—Destination-Day, Lunch, Break, Night

Destination	Availability
Wired Extension (PT/SLT/SIP Extension/ISDN Extension)	✓
PS	✓
Incoming Call Distribution Group	✓
PS Ring Group	✓
UM Group	✓
VM Group (DTMF/DPT)	✓
External Pager (TAFAS)	✓
DISA	✓
Analogue/ISDN Remote Maintenance	✓
Idle Line Access no. + Phone no.	✓
Trunk Group Access no. + Trunk Group no. + Phone no.	✓
Other PBX Extension (TIE with no PBX Code)	✓
Other PBX Extension (TIE with PBX Code)	✓

2. **Busy on Busy**
   The Busy on Busy feature works when the destination for the Intercept Routing—Overflow in an Incoming Call Distribution Group feature is not assigned in one of the following conditions:
   a. There is no space in the Waiting queue.
   b. The Queuing Time Table is not assigned and there are no extensions logged-in.

   [Example of a)]
   There are five assistants in a shop. When the answering agent number is "2", and the queuing call number is "0":
   If two of the assistants are talking on the phone, the next caller will hear a busy tone to prevent the caller from thinking that there is no one in the shop or that the shop is closed.

Conditions

[Intercept Routing—Overflow in an Incoming Call Distribution Group]

- **If the Overflow time expires, and the overflow destination is unavailable:**
  a. If the trunk call arrives through the LCOT card:
  (1) If the call was once in a queue and an outgoing message (OGM) was sent to it, or the call reached an incoming call distribution group by using the DISA feature (→ 2.16.1 Direct Inward System Access (DISA)): The line is disconnected.
  (2) In all other cases: Redirection is ignored and the Overflow timer activates again.
  b. If the call arrives through another card: Redirection is ignored and the Overflow timer activates again.

[Busy on Busy]

- If a trunk call arrives through the LCOT card, a busy tone will not be sent to the caller.

PC Programming Manual References

11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings
→Overflow Queuing Busy
→Overflow No Answer
11.5.2 PBX Configuration—[3-5-2] Group—Incoming Call Distribution Group—Queuing Time Table
11.5.3 PBX Configuration—[3-5-3] Group—Incoming Call Distribution Group—Miscellaneous—◆ Options—Overflow immediately when All Logout

PT Programming Manual References

[625] Destination for Overflow Time Expiration
[626] Overflow Time
[627] Destination When All Busy
[628] Queuing Call Capacity
[632] Maximum Number of Agents

Feature Guide References

2.2.2.4 Queuing Feature

2.2.2.7 Log-in/Log-out

Description

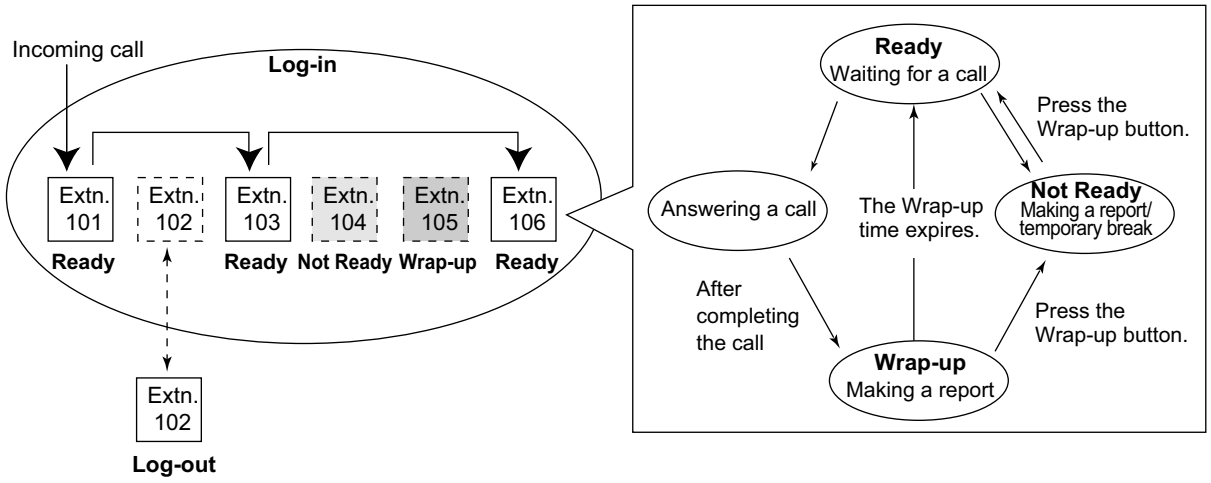
Members of an incoming call distribution group can join (Log-in) or leave (Log-out) the group manually. They can leave the group temporarily when they are away from their desks, to prevent calls being sent to their extensions. They can return to the group when they are ready to answer calls.

Wrap-up:

While logged in, a member extension can have a preprogrammed time period automatically for refusing calls after completing the previous call (Wrap-up time). While the Wrap-up timer is active, calls to all incoming call distribution groups to which the extension belongs will skip the extension so that the extension user can perform necessary tasks such as reporting on the previous call.
Wrap-up mode can also be activated manually (Not Ready) by pressing the Wrap-up button.

[Log-in/Log-out and Wrap-up Status Example]

<When the incoming call distribution group is in Priority Hunting distribution method>



Conditions

- It is programmable whether the last remaining logged-in extension can log out.
- **Log-in/Log-out Button**
  A flexible button can be customised as the Log-in/Log-out button with the following parameters:

Parameter	Usage	Light Pattern	
		Red on	Off
No parameter	Used with an ICD Group button, or with the floating extension number of an incoming call distribution group, or with ✳ (All).	—	—
Floating extension number of a specified incoming call distribution group	Used to log in to or out of the specified incoming call distribution group.	Log-out Status	Log-in Status

Parameter	Usage	Light Pattern	
		Red on	Off
⚹ (All)	Used to log in to or out of all incoming call distribution groups to which the extension user belongs.	After Log-out Operation	After Log-in Operation

- If an ICD Group button is assigned, it also shows the log-in/log-out status of the corresponding group. The light pattern is the same as the Log-in/Log-out button that includes the group number.
- **Wrap-up Timer**
  - Two wrap-up timers can be programmed, an ICD Group member wrap-up timer and an extension wrap-up timer. System programming selects which timer is used. When the ICD Group member wrap-up timer is selected, the timer is only activated after calls to the extension through an ICD Group. When the extension wrap-up timer is selected, the timer is activated after all calls to or from the extension, including a retrieved call on hold.
  - Only calls from ICD Groups cannot be received during the wrap-up time. Other calls are received as normal.
  - The wrap-up timer does not work for ISDN extensions or PS Ring Groups.
- **Wrap-up Button**
  A flexible button can be customised as the Wrap-up button. It shows the current status as follows:

Light pattern	Status
Slow red flashing	Wrap-up
Red on	Not Ready
Off	Ready (Wrap-up mode cancel)

- When a PS in Wireless XDP Parallel Mode completes a call, neither the PS nor its wired telephone can have Wrap-up time. (→ 5.2.4.5 Wireless XDP Parallel Mode)
- **Automatic Log-out**
  A member extension may be logged out automatically, if the Unanswered time expires a preprogrammed number of times consecutively. The number of consecutive unanswered calls can be assigned for each incoming call distribution group. If the extension is a member of more than one incoming call distribution group, the unanswered number is counted across all corresponding incoming call distribution groups. It is possible to return to log-in mode manually.
  The Automatic Log-out feature does not work for extensions in an incoming call distribution group using the Ring distribution method (→ 2.2.2.2 Group Call Distribution).
- **Log-in/Log-out Monitor**
  The supervisor extension can monitor and control the log-in/log-out status of the incoming call distribution group members. (→ 2.2.2.8 Supervisory Feature)
- **Log-in/Log-out Information on SMDR**
  Log-in/Log-out information can be printed out on SMDR. (→ 2.22.1.1 Station Message Detail Recording (SMDR))

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ Log-in / Log-out
→◆ Not Ready (Manual Wrap-up) Mode On / Off
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Miscellaneous

→◆ No. of Unanswered Calls for Automatic Log-out
→◆ Last Extension Log-out
11.5.1.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Member List—◆ Wrap-up Timer
11.5.3 PBX Configuration—[3-5-3] Group—Incoming Call Distribution Group—Miscellaneous—◆ Options—Wrap-up Timer based on
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 8—◆ Wrap-up Timer
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Parameter Selection (for Log-in/Log-out)
→◆ Extension Number (for Log-in/Log-out)
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 8—◆ Wrap-up Timer
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Parameter Selection (for Log-in/Log-out)
→◆ Extension Number (for Log-in/Log-out)
19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ Print Information—Log-in / Log-out

Feature Guide References

2.21.2 Flexible Buttons

User Manual References

1.10.1 Leaving an Incoming Call Distribution Group (Log-in/Log-out, Wrap-up)

2.2.2.8 Supervisory Feature

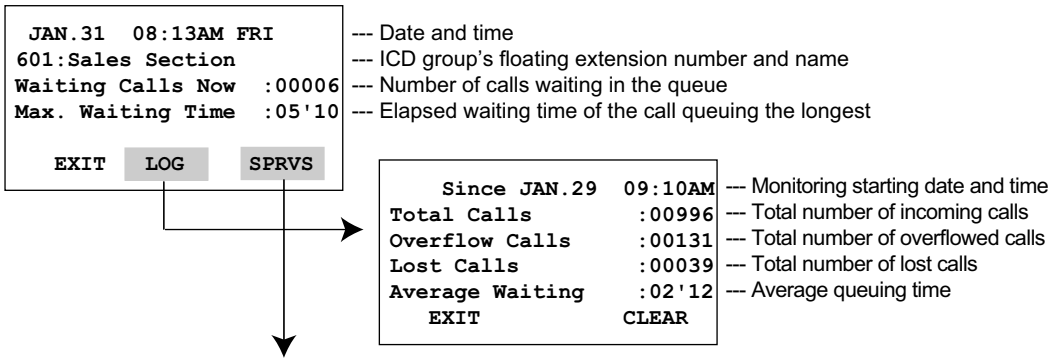
Description

An extension preprogrammed as a supervisor (supervisor extension) can monitor and control each member's status within the incoming call distribution group using a 6-line display PT.

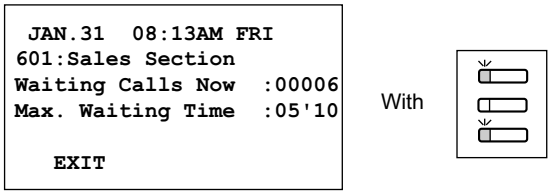
Feature	Description
Incoming Call Queue Monitor	The supervisor extension can monitor the status of an incoming call distribution group with the display.
Log-in/Log-out Monitor and Remote Control	**Monitor:** The supervisor extension can monitor the log-in/log-out status of the incoming call distribution group members through the corresponding DSS button light. **Remote Control:** The supervisor extension can change the status of the members by pressing the corresponding DSS button.

[Example]

<Incoming Call Queue Monitor Display>



<Log-in/Log-out Monitor/Remote Control Mode with DSS Button light>



Log-in/Log-out Monitor
DSS buttons show the status of the corresponding group members.

Light pattern	Status
Green on	Log-in (Ready)
Slow Green Flashing	Log-in (Not Ready)
Red on	Log-out
Off	Extension in another ICD group

Log-in/Log-out Remote Control
Pressing a DSS button changes the extension's status as follows:

Status	Light pattern
Log-out	Red on
Log-in (Ready)	Green on

Conditions

- **Available Extension as a Supervisor Extension**

a. One supervisor extension can be assigned for each incoming call distribution group, but it need not belong to the group.
b. One extension can be the supervisor extension of more than one incoming call distribution group.

- **Available Paired DSS Console**
  This feature is available for the KX-T7640, KX-DT390, KX-DT590, and KX-NT505.
- **Accumulation Value Clear**
  Accumulation value data (total incoming calls, total overflowed calls, lost calls, average queuing time) can be cleared manually. The date and time of clearing is saved and is shown on the display (monitoring starting date and time). When the value exceeds 99999 before clearing, "*****" will be shown.
- **If a call to an incoming call distribution group is overflowed:**
  If the display is in idle status, it will change to monitor mode for the corresponding incoming call distribution group automatically.
  If the display is monitoring another incoming call distribution group, it will not change.
- **Other Features while in Monitor Mode**
  The supervisor can use other features on the extension (making calls, pressing the MESSAGE button, etc.) even while in monitor mode. When each operation is finished, his telephone returns to the queue monitor display.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Incoming Call Queue Monitor
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Miscellaneous—◆ Supervisor Extension Number

User Manual References

1.10.2 Monitoring and Controlling the Call Status of an Incoming Call Distribution Group (Incoming Call Distribution Group Monitor)

2.2.2.9 Supervisory Feature (ACD)

Description

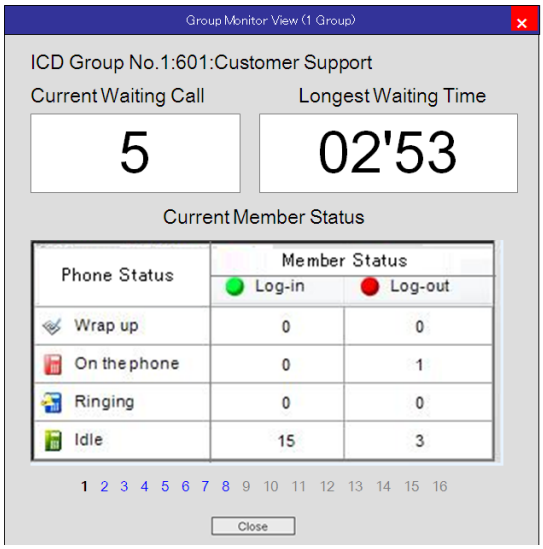
By specifying an ACD supervisor for extension users, the ACD supervisor can check and analyse the operating conditions of an ICD group. This is done by monitoring the current condition of the ICD group, and by collecting and analysing statistical ACD report information.

Feature	Description
ICDG Management–Group Monitor	The ACD supervisor can monitor the condition of up to 4 ICD groups on the same screen by specifying monitoring conditions. Furthermore, the ACD supervisor can manage up to 64 ICD groups.
ICDG Management–ACD Report	The ACD supervisor can analyse the monitoring result as follows: • Filter: Filters the monitoring result according to Group, Agent and Call. • View Report: Displays the filtered monitoring result in a format that is easy to analyse (graph, file export, or print).

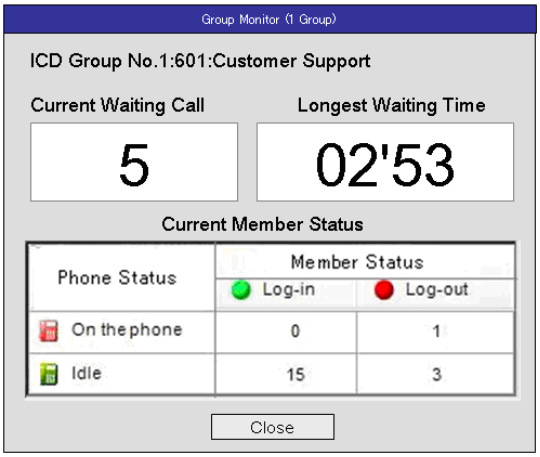
The screen of a user set as an ACD supervisor is displayed as follows. There are 2 modes – Simple Mode and Standard Mode – and the displayed information is different depending on the mode.

[Group Monitor Example]

Standard Mode



Simple Mode



[ACD Report]

An ACD report can be made with the following items included.

Group

Item	Description
Incoming Calls	
Total	The number of incoming calls received by the target ICD group.
Answered	The number of incoming calls answered by the target ICD group.
Lost	The number of incoming calls to the target ICD group cancelled by the caller.
Overflow	The number of overflowed incoming calls to the target ICD group.
Talk Time	
Total	The total talking time of answered calls for the target ICD group. (HH:MM:SS)
Average	The average talking time of answered calls for the target ICD group. (HH:MM:SS)
Max.	The longest talking time of answered calls for the target ICD group. (HH:MM:SS)
Wait Time	
Wait Time (Answered)	
Total	The total waiting time of answered calls for the target ICD group. (HH:MM:SS)
Average	The average waiting time of answered calls for the target ICD group. (HH:MM:SS)
Max.	The longest waiting time of answered calls for the target ICD group. (HH:MM:SS)
Wait Time (Lost)	

Item	Description
Total	The total waiting time of cancelled calls for the target ICD group. (HH:MM:SS)
Average	The average waiting time of cancelled calls for the target ICD group. (HH:MM:SS)
Max.	The longest waiting time of cancelled calls for the target ICD group. (HH:MM:SS)
Max. Waiting Calls	The maximum number of calls waiting in the queue of the target ICD group.

Agent

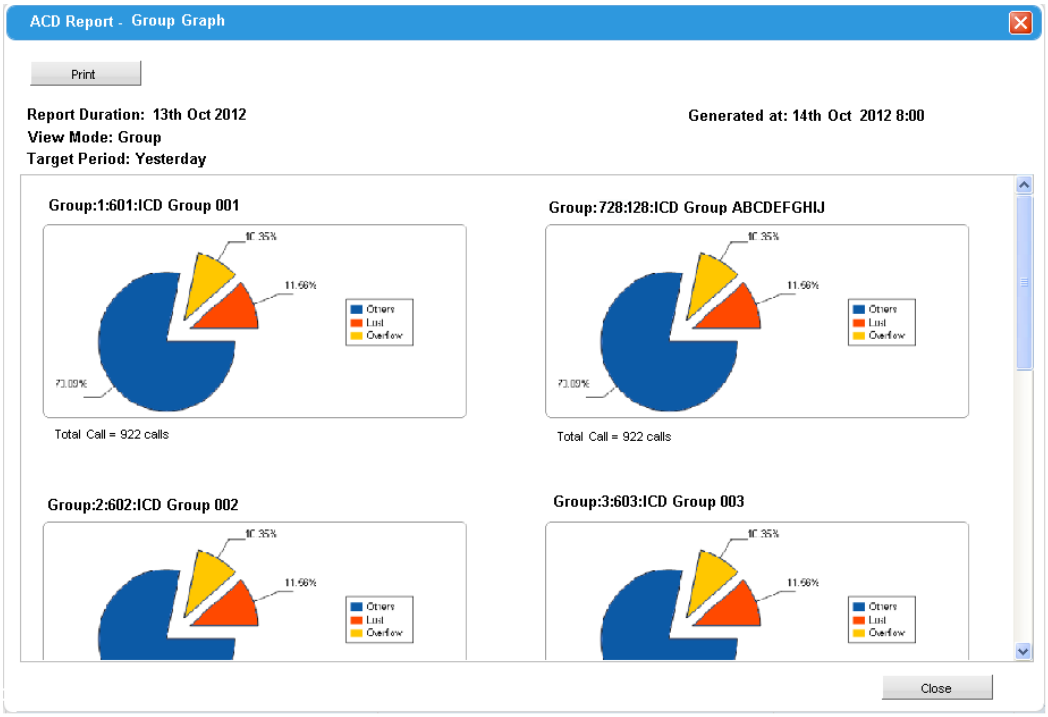
Item	Description
Total Answer	
Total Answer	The number of calls that the target agent answers.
Talk Time	
Total	The total talking time for the target agent. (HH:MM:SS)
Average	The average talking time for the target agent. (HH:MM:SS)
Max.	The longest talking time for the target agent. (HH:MM:SS)
Login Time	The total login time for the target agent. (HH:MM:SS)
Not-ready Time	The total not ready time for the target agent. (HH:MM:SS)
Wrap-up Time	The total wrap-up time for the target agent. (HH:MM:SS)

Call

Item	Description
ACD Report - Call Report	
Start Date	The start date of the call.
Start Time	The start time of the call. (HH:MM:SS)
End Date	The end date of the call.
End Time	The end time of the call. (HH:MM:SS)
Result	The processing result. (Answered/Abandoned/Overflowed)
ICDG	The incoming ICD Group number.
Answering Agent	The answering member. (Extension Name/Extension Number)
Talk Time	The talking time. (HH:MM:SS)
Wait Time	The waiting time. (HH:MM:SS)
Trunk	The incoming trunk group number.
Caller ID/CLIP	The caller’s number.

Reports can be output as a graph, exported as a file or printed out, after filtering results as necessary. For details about ACD report items, refer to the PC Programming Manual.

[Graph Example]



Conditions

- KX-NSF201 (Call Centre Feature Enhancement) is required to use this feature.
- Up to 16 users can be set as an ACD supervisor through system programming.
- An SD Memory Card (KX-NS3134/KX-NS3135/KX-NS3136) is required to save ACD reports.
  – With SD Memory Card: 300000 calls

PC Programming Manual References

8.4 Users—ICDG Management
11.5.3 PBX Configuration—[3-5-3] Group—Incoming Call Distribution Group—Miscellaneous
11.5.4 PBX Configuration–[3-5-4] Group—Incoming Call Distribution Group—ACD Supervisor

User Manual References

4.1.3 Supervisory Monitor (ACD) Control

2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features

2.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)—SUMMARY

Description

When an extension user cannot answer calls (e.g., he is busy, or not at his desk), it is possible to forward or refuse calls using the following features:

1. Call Forwarding (FWD)
2. Do Not Disturb (DND)

1. **FWD**
   Extensions and incoming call distribution groups can forward their incoming calls to preset destinations. (→ 2.3.2 Call Forwarding (FWD))
2. **DND**
   Callers to an extension will hear a tone to inform them that the extension user is not available. (→ 2.3.3 Do Not Disturb (DND))
3. **FWD/DND Button, Group FWD Button**
   The FWD/DND fixed button, or a customised flexible button, can display the FWD/DND setting status of the extension. (→ 2.3.4 FWD/DND Button, Group FWD Button)

Conditions

- FWD and DND are set for intercom calls (including doorphone calls), and trunk calls (including a call from an extension that placed a trunk call on a consultation hold) separately.

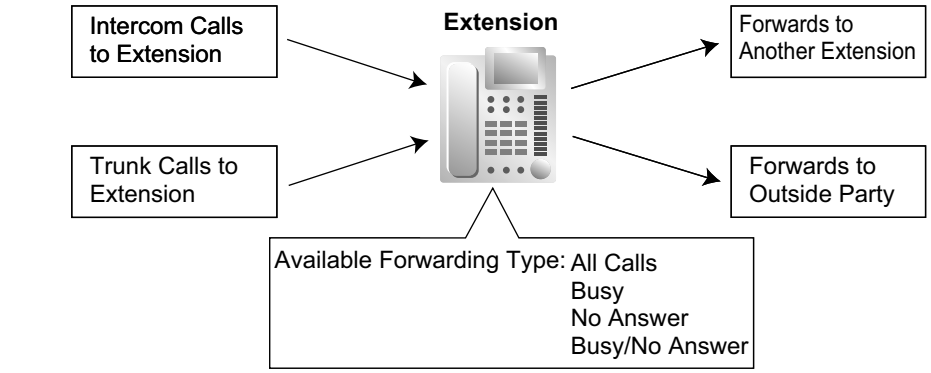
2.3.2 Call Forwarding (FWD)

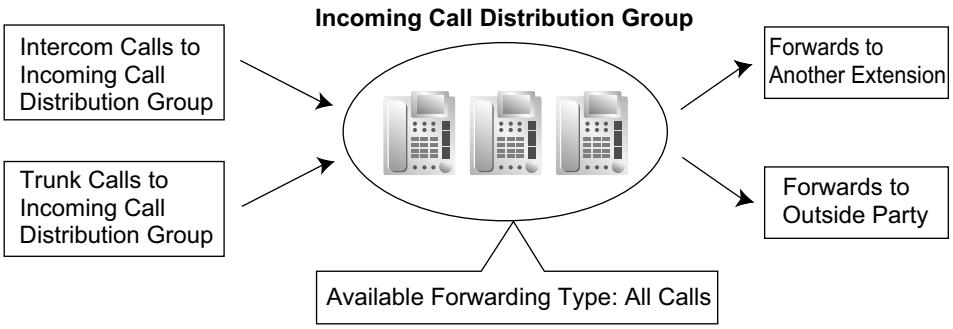
Description

Extensions and incoming call distribution 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:

Type	Circumstance
All Calls	Any time **Follow Me:** When an extension user fails to set this feature before leaving the desk, this feature can be set from the destination extension.
Busy	When the extension user's line is busy.
No Answer	When the extension user does not answer within a preprogrammed time.
Busy/No Answer	When the extension user's line is busy or the user does not answer within a preprogrammed time.

Depending on the type of incoming intercom or trunk calls, it is possible to set a different destination for each.





[Available Destinations]

Destination	Availability	Condition for Original Extension/ Incoming Call Distribution Group
Wired Extension (PT/SLT/SIP Extension/ISDN Extension)	✓	Only available when FWD to extension is allowed through COS programming.*1
PS	✓	
Incoming Call Distribution Group	✓	
PS Ring Group	✓	–
UM Group	✓	–
VM Group (DTMF/DPT)	✓	–
External Pager (TAFAS)	✓	–
DISA	✓	Only available for incoming trunk calls. Incoming intercom and doorphone calls cannot be forwarded to a DISA floating extension number.
Analogue/ISDN Remote Maintenance	✓	–
Idle Line Access no. + Phone no.	✓	Only available when FWD to trunk is allowed through COS programming.
Trunk Group Access no. + Trunk Group no. + Phone no.	✓	
Other PBX Extension (TIE with no PBX Code)	✓	–
Other PBX Extension (TIE with PBX Code)	✓	Only available when FWD to trunk is allowed through COS programming.

*1 If an extension user is not permitted by COS to call a certain extension (→ 2.1.2.2 Internal Call Block), the FWD feature will not function if that extension is set as the forwarding destination.

Parallel Ringing When Forwarding to Trunk

When an unanswered call is forwarded to an outside line, such as a cellular phone, the forwarding extension's phone will continue ringing until the forwarded call is answered at either phone. This feature can be enabled for each extension through system programming. Even when the outside destination answers the call, DSS buttons for the forwarding extension are displayed as busy. The Intercept Routing feature is available for when parallel ringing is not answered.

Conditions

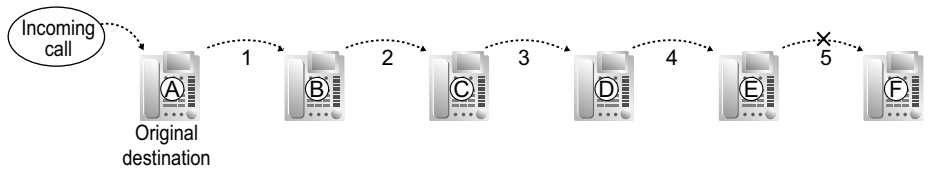
[General]

- **FWD for Trunk Calls/Intercom Calls**
  The extension user can set the FWD feature for trunk calls, for intercom calls, or for both.
- **FWD from Incoming Call Distribution Group (Group FWD)**
  COS programming determines the incoming call distribution groups that can use this feature.
- **FWD to Trunk**
  COS programming determines the extensions or incoming call distribution groups that can forward calls externally. The original extension's TRS/Barring and ARS still apply to the forwarded call.
- **Trunk Call Duration**

The duration of a trunk call can be restricted by a system timer. Trunk call duration is assigned separately for calls between an extension user and an outside party, and calls between two outside parties.
If the timer expires, the line will be disconnected. (→ 2.11.8 Trunk Call Limitation)

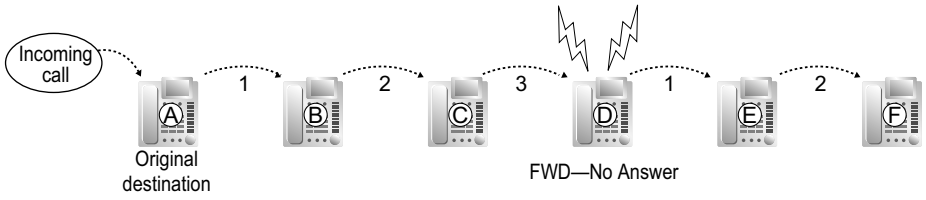
- **Multiple FWD**
  Calls can be forwarded up to four times. The following forwarding features are counted as Multiple FWD:
  – FWD—Busy or Busy/No Answer (when the destination extension is busy), or All Calls
  – Idle Extension Hunting—Overflow
  – Intercept Routing—Busy/DND/No answer (when the destination extension is busy or in DND or No answer mode)*1
  – Incoming Call Distribution Group—Overflow

  *1 Intercept Routing features can be applied to the original destination (refer to 2.1.1.5 Intercept Routing).

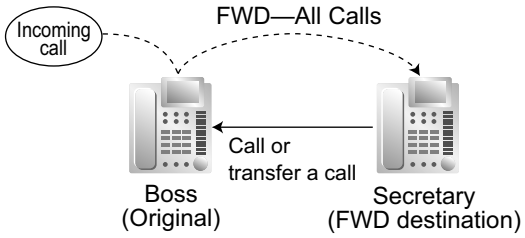


In the above illustration, forwarding stops at extension E. However, forwarding can go farther in the following cases:
– If a destination extension rings, and then the call is redirected to the forward destination by the FWD—No Answer or Busy/No Answer feature.
– If a call waits in a queue of an incoming call distribution group, and then the call is redirected to the overflow destination by the Queuing Time Table. (→ 2.2.2.4 Queuing Feature)

In the above cases, the forwarding counter resets to zero, and the call can be forwarded up to four times again from the destination extension described above.



- **Boss & Secretary feature**
  It is possible to call the original extension from the destination extension regardless of the forward setting.



- **Message Waiting**
  While calls are forwarded, Message Waiting information is not forwarded. The Message button light turns on at the originally called extension. (→ 2.20.1 Message Waiting)
- **Idle Extension Hunting**
  Idle Extension Hunting applies to calls forwarded to a busy extension in an idle extension hunting group.

[All Calls and Busy]

- If the forward destination is not available to answer a call, this feature is cancelled and the original destination will ring for the following type of call:
  - Doorphone call
  - Trunk calls via the LCOT card

[No Answer and Busy/No Answer]

- **No Answer Time**
  The number of rings before the call is forwarded is programmable for each extension.

[Follow Me]

- This feature is only available when the original extension has set "**Remote Operation by Other Extension**" to "**Allow**" through COS programming.

[Parallel Ringing When Forwarding to Trunk]

- KX-NSE101, KX-NSE105, KX-NSE110 or KX-NSE120 (Activation Key for Mobile Extension) is required to use this feature. One activation key is required for each extension that will use this feature.
- Even though DSS buttons for the forwarding extension indicate that the extension is busy, it can still receive calls. If another call is received, FWD—No Answer will operate as normal.
- **When the forwarding extension is a virtual PS** (→ 5.2.4.6 Virtual PS)
  - The No Answer Time setting is ignored: All calls are forwarded immediately, even if No Answer is specified as the forwarding method.
  - If the forward destination is unavailable, the forwarding extension is treated as busy. (In this case, FWD—Busy is ignored even if it is enabled.) However, if the original caller is on an analogue trunk, to which busy signals/tones cannot be sent, the PBX will continue to try to connect to the forward destination until a connection is established or the original caller hangs up.
- If an extension goes on-hook while transferring a call to an extension ringing in parallel with a trunk, the trunk will stop ringing for a moment, then begin ringing again.
- This feature for LCOT depends on the settings for reverse signal detection. (→2.5.4.5 Reverse Circuit)
- This feature may not be available depending on the specifications of the telephone network.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ FWD No Answer Timer Set
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings
→CO & SMDR—◆ Call Forward to CO
→Manager—◆ Group Forward Set
→Optional Device & Other Extensions—◆ Remote Operation by Other Extension
11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Main
→◆ CO-CO Duration Time (*60s)
→◆ Extension-CO Duration Time (*60s)
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Group Log / Group FWD
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—FWD / DND
12.1.2 PBX Configuration—[4-1-2] Extension—Wired Extension—FWD/DND
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—FWD / DND
12.2.2 PBX Configuration—[4-2-2] Extension—Portable Station—FWD/DND

PT Programming Manual References

[472] Extension-to-Trunk Call Duration

[473] Trunk-to-Trunk Call Duration
[504] Call Forwarding to Trunk
[605] Call Forwarding—No Answer Time

Feature Guide References

2.1.1.5 Intercept Routing
2.2.1 Idle Extension Hunting
2.2.2.6 Overflow Feature
5.1.1 Class of Service (COS)

User Manual References

1.6.1 Forwarding Calls

2.3.3 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, but will arrive at another extension using the Idle Extension Hunting feature (→ 2.2.1 Idle Extension Hunting) or the Intercept Routing—Busy/DND feature (→ 2.1.1.5 Intercept Routing). When a destination cannot be found, the calling extension will hear the DND tone, while the calling outside party will hear a busy tone.

Conditions

- **DND for Trunk Calls/Intercom Calls**
  The DND feature can be set for trunk calls, for intercom calls, or for both of them by the extension user.
- **DSS button in DND Mode**
  The DSS button light will turn red if the assigned extension has set DND.
- **DND Override**
  An extension in DND mode can be called by other extension users who are allowed to override DND in their COS.
- **Paging DND**
  It is programmable whether the PBX pages extensions in DND mode through system programming. (→ 2.17.1 Paging)
- **Intercept Routing—Busy/DND**
  If a call arrives at an extension in DND mode, the call can be redirected to a preprogrammed destination by the Intercept Routing—Busy/DND feature.
- **Idle Extension Hunting**
  While searching for an idle extension within an idle extension hunting group, any extension that has DND set will be skipped. The call will go to the next extension in the group, not the Intercept Routing—Busy/DND destination.
- If (1) a trunk call via the LCOT card arrives at an extension in DND mode and (2) the Intercept Routing—Busy/DND destination is not available and (3) there is no available extension in the idle extension hunting group, then the original extension in DND mode will ring.
- Calls from a doorphone arrive at the extension even when the extension is in DND mode.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Tone Length—Busy Tone / DND Tone (s)
10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature
→◆ BSS / OHCA / Whisper OHCA / DND Override
→◆ BSS / OHCA / Whisper OHCA / DND Override-2
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Assistant—◆ DND Override
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—FWD / DND
12.1.2 PBX Configuration—[4-1-2] Extension—Wired Extension—FWD/DND
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—FWD / DND
12.2.2 PBX Configuration—[4-2-2] Extension—Portable Station—FWD/DND

PT Programming Manual References

[507] DND Override

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer
1.9.2 Refusing Incoming Calls (Do Not Disturb [DND])

2.3.4 FWD/DND Button, Group FWD Button

Description

The FWD/DND fixed button, or a customised flexible button, can display the FWD/DND setting status of the extension. Using this button, the FWD status and DND status of the extension can be temporarily set or cancelled without clearing FWD destination settings.

FWD/DND Button Types

Multiple types of FWD/DND buttons can be customised on an extension.

Type		Description
FWD/DND for Extension	FWD/DND—Internal	Works for incoming intercom calls
	FWD/DND—External	Works for incoming trunk calls
	FWD/DND—Both	Works for all incoming calls

[Button Status]
The FWD/DND button shows the current status as follows:

Light Pattern	Status (default)
Red on	FWD on
Slow red flashing	DND on
Off	FWD/DND off

The functions assigned to the "on" and "flashing" patterns can be changed through system programming.

Group FWD Button Types

The FWD feature for the incoming call distribution group can be customised on a flexible button. Multiple types of Group FWD buttons can be customised on an extension.

Type		Description
FWD for Incoming Call Distribution Group	Group FWD—Internal	Works for incoming intercom calls
	Group FWD—External	Works for incoming trunk calls
	Group FWD—Both	Works for all incoming calls

[Button Status]
The Group FWD button shows the current status as follows:

Light Pattern	Status (default)
Red on	FWD on
Off	FWD off

FWD/DND Setting by Fixed FWD/DND button

Pressing the fixed FWD/DND button in idle status allows the extension user to set the following items for FWD/DND:

- FWD/DND for trunk calls

The FWD/DND status for trunk calls to that extension can be switched temporarily without clearing the FWD destination. During setting, the LED of the button shows the current trunk call FWD/DND status. The forwarding type and destination for trunk calls can also be set.

- FWD/DND for intercom calls
  The FWD/DND status for intercom calls to that extension can be switched temporarily without clearing the FWD destination. During setting, the LED of the button shows the current intercom call FWD/DND status. The forwarding type and destination for intercom calls can also be set.
- FWD—No Answer timer
  The length of time until unanswered calls are forwarded can be modified. This setting is applied to both intercom and trunk call forwarding.
- FWD for Virtual PS
  If the extension is the first registered extension in an Incoming Call Distribution Group, the extension user can set the FWD destination and forwarding status (on/off) for up to 4 virtual PSs preregistered to the group. (→ 5.2.4.6 Virtual PS)

These settings are only available when FWD/DND buttons are set through system programming to FWD/DND Setting mode.

Conditions

- When FWD/DND buttons are set to FWD/DND Cycle Switch mode, pressing the FWD/DND button cycles the FWD/DND setting.
  In this mode, when intercom calls are set to be handled differently from trunk calls (forwarding type, forward destination, DND on/off):
  a. in idle mode, the light patterns of the FWD/DND—Both button (including FWD/DND button [fixed button]) and the Group FWD—Both button will indicate the setting for either trunk calls or intercom calls, but not both.
  b. the FWD and DND icons on a PS display reflect the settings for trunk calls only.
  c. pressing the FWD/DND—Both button (including FWD/DND button [fixed button]) or the Group FWD—Both button will be ignored.
- When both the FWD and DND features are assigned simultaneously, pressing the button changes the settings as follows:

- A FWD/DND button customised on a flexible button is always in FWD/DND Cycle Switch mode, and the mode cannot be changed.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ FWD/DND Set / Cancel: Call from CO & Extension
→◆ FWD/DND Set / Cancel: Call from CO
→◆ FWD/DND Set / Cancel: Call from Extension
→◆ Group FWD Set / Cancel: Call from CO & Extension
→◆ Group FWD Set / Cancel: Call from CO
→◆ Group FWD Set / Cancel: Call from Extension
10.9 PBX Configuration—[2-9] System—System Options—Option 1
→◆ PT Fwd / DND—Fwd LED
→◆ PT Fwd / DND—DND LED
→◆ PT Fwd / DND—Fwd/DND key mode when Idle
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—FWD / DND
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type

→◆ Extension Number (for Group Fwd (Both))
→◆ Extension Number (for Group Fwd (External))
→◆ Extension Number (for Group Fwd (Internal))

12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—FWD / DND

12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button

→◆ Type
→◆ Extension Number (for Group Fwd (Both))
→◆ Extension Number (for Group Fwd (External))
→◆ Extension Number (for Group Fwd (Internal))

Feature Guide References

2.21.1 Fixed Buttons
2.21.2 Flexible Buttons

User Manual References

3.1.2 Settings on the Programming Mode

2.4 Answering Features

2.4.1 Answering Features—SUMMARY

Description

An extension user can answer incoming calls by the following methods:

Destination	Feature	Description & Reference
At the own extension (PT only)	**Line Preference —Incoming**	A user can select the line seized when going off-hook. → 2.4.2 Line Preference—Incoming
	Direct One-touch Answering	A user can answer an incoming call simply by pressing the flashing button.
	Hands-free Answerback	A user can receive a call automatically and establish a hands-free conversation. → 2.4.4 Hands-free Answerback
At another extension	**Call Pickup—Directed/ Group**	A user can pick up a specified extension's call or a call in a specified call pickup group. → 2.4.3 Call Pickup

2.4.2 Line Preference—Incoming

Description

A PT user can select the method used to answer incoming calls from the following three line preferences: Each of these line preferences can be assigned on each extension through personal programming (Preferred Line Assignment—Incoming).

Type	Description
No Line	Select a line by pressing the desired Line Access button to answer an incoming call after you go off-hook.
PDN	Answers a call arriving at a Primary Directory Number (PDN) button simply by going off-hook. This works even when multiple calls are received simultaneously. (→ 2.9.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension)
Prime Line	Answer a call arriving at a Flexible CO or ICD Group button (on which the "Prime Line" is assigned) simply by going off-hook. This works even when multiple calls are received simultaneously.
Ringing Line (default)	Answer the longest ringing call at one's telephone simply by going off-hook when multiple calls arrive.

Conditions

[Prime Line]

- The priority of the incoming call is as follows:
  1. The call arriving at a button on which the "Prime Line" is assigned.
  2. The call arriving at the INTERCOM button.

PC Programming Manual References

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 4—◆ Incoming Preferred Line
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 4—◆ Incoming Preferred Line

User Manual References

3.1.2 Settings on the Programming Mode

2.4.3 Call Pickup

Description

An extension user can answer a call ringing at any other extension.
The following types are available:

Type	Picking up Call Type
Directed	A specified extension's call.
Group	A call within a specified call pickup group.

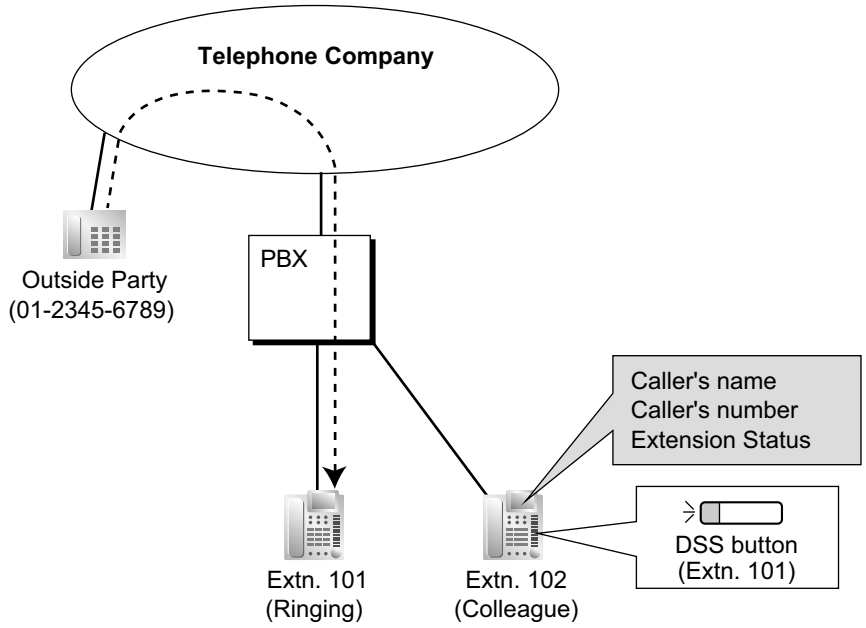
Call Pickup Deny:

Preventing other extensions from picking up calls ringing at your extension is also possible.

Caller Information Display before Call Pickup

PT/PS users such as colleagues can confirm the caller information of a call to another extension using the DSS button before picking up the call.
Pressing the corresponding DSS button will call the extension receiving the call rather than picking up the call. While hearing the busy tone, the user can check the caller's information on the LCD. The call can be picked up by pressing the same DSS button again.

[Example]



Conditions

- **Call Pickup applies to:**
  Intercom, trunk, and doorphone calls
- **Internal Call Block**
  An extension that is restricted by COS from calling certain extensions (→ 2.1.2.2 Internal Call Block) also cannot pick up any calls ringing at those extensions.

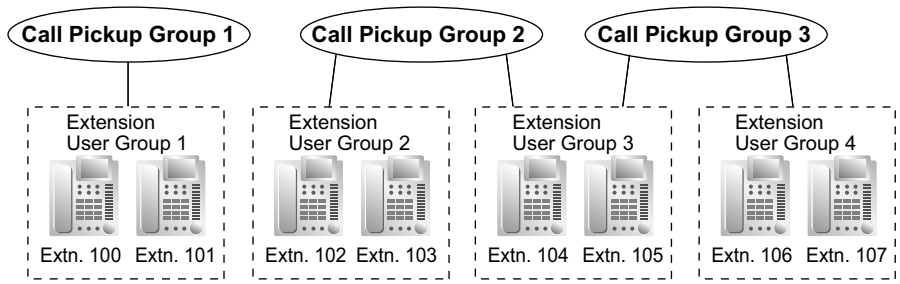
[Directed Call Pickup]

- A user can also pick up a call to a specified extension by pressing the corresponding DSS button. This feature is only available when (1) the user's extension is allowed to use this feature through COS programming, (2) DSS buttons for extensions or incoming call distribution (ICD) groups have this feature enabled through system programming, and (3) the light pattern of DSS buttons for incoming calls to extensions or ICD groups is set to "**On or Flash**" through system programming.
  The light pattern of a DSS button for an incoming call to an extension or incoming call distribution group can be programmed through system programming. Call Pickup is available only when the DSS button is flashing red.

[Group Call Pickup]

- A specified number of call pickup groups can be created, each of which consist of extension user groups. One extension user group can belong to several call pickup groups. (→ 5.1.2 Group)

[Example]



[Caller Information Display before Call Pickup]

- This feature is only available under the following conditions:
  - The "Caller Information Display before Call Pick-up" setting is enabled.
  - Directed Call Pickup is enabled for the extension picking up the call.
  - The assigned button is a DSS button and not an NDSS button.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ Group Call Pickup
→◆ Directed Call Pickup
→◆ Call Pickup Deny Set / Cancel
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Assistant—◆ Call Pickup by DSS
10.9 PBX Configuration—[2-9] System—System Options—Option 4
→◆ DSS Key—DSS key mode for Incoming Call
→◆ DSS Key—Call Pick-up by DSS key for Direct Incoming Call
→◆ DSS Key—Call Pick-up by DSS key for ICD Group Call
→◆ DSS Key—Caller Information Display before Call Pick-up
11.3 PBX Configuration—[3-3] Group—Call Pickup Group
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 3—◆ Call Pickup Deny
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 3—◆ Call Pickup Deny

Feature Guide References

5.1.1 Class of Service (COS)
6.1 Capacity of System Resources

PT Programming Manual References

[650] Extension User Groups of a Pickup Group

User Manual References

1.3.3 Answering a Call Ringing at Another Telephone (Call Pickup)

2.4.4 Hands-free Answerback

Description

A PT user with a speakerphone can talk to a caller without lifting the handset. If the user receives a call while in Hands-free Answerback mode, a hands-free conversation is established using one of the following methods:

Type	Answering Method
Intercom Call	Established immediately after a beep tone at the called extension and the caller hears a confirmation tone.
Trunk Call	Established after a specified number of rings, a called extension hears a beep tone.

Class of Service (COS) with Hands-free Answerback

Hands-free Answerback can be enabled only for specific callers based on the caller's COS setting. For example, a nurse could call a room and the called extension will answer automatically so that the nurse can check on the occupant of the room. However, other intercom calls would ring even if Hands-free Answerback was set. Walking COS can also be used with this feature.

Conditions

- **Hands-free Answerback applies to:**
  Intercom calls and trunk calls, including calls directed to an incoming call distribution group in UCD or Priority Hunting distribution method. (→ 2.2.2.2 Group Call Distribution)
- **Hands-free Answerback for Trunk Calls**
  System programming is required to use this feature.
- **Hands-free Answerback for Calls From an Extension That Placed a Trunk Call on Consultation Hold**
  Calls from an extension that placed a trunk call on Consultation Hold can be treated by this feature as either intercom calls or trunk calls, depending on system programming. If treated as intercom calls, the call will be established immediately.
  When transferring a call from an analogue trunk, users are strongly recommended to perform a screened transfer, so that the outside caller is not automatically connected to an extension using Hands-free Answerback when the extension user is absent.
  Extensions that perform unscreened transfers often, such as operators, should have the Class of Service (COS) with Hands-free Answerback feature disabled. Otherwise, transferred outside calls may be automatically connected by Hands-free Answerback, even when the transfer destination is absent.
- **Secret Monitor**
  The beep tone that the called party hears before answering can be eliminated through system programming.
- **Alternate Receiving/Calling Mode (Ring/Voice) Override**
  Hands-free Answerback overrides the Alternate Receiving mode preset on the telephone and the Alternate Calling mode from the caller.
- **Hands-free Answerback with Headset**
  The Hands-free Answerback feature can also be used with a headset.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Tone Length—Reorder Tone for PT Hands-free (s)
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Assistant—◆ Automatic Answer (Caller)
10.9 PBX Configuration—[2-9] System—System Options

→Option 1—◆ PT Operation—Automatic Answer for Call from CO after
→Option 3—◆ Confirmation Tone—Tone 2 : Paged / Automatic Answer
→Option 4—◆ Transfer—Automatic Answer for Transferred Call

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Option 5—◆ Automatic Answer for CO Call
→Option 6—◆ Forced Automatic Answer

Feature Guide References

2.5.3 Intercom Call
5.1.1 Class of Service (COS)

User Manual References

1.3.2 Answering Hands-free (Hands-free Answerback)

2.5 Making Call Features

2.5.1 Predialling

Description

A display PT user can check and correct the entered number before it is dialled, while on-hook. The call will be initiated after going off-hook.

Conditions

- **Storing the Predialled Number in the Personal Speed Dialling**
  The predialled number can be stored in the Personal Speed Dialling by pressing the AUTO DIAL/STORE button. (→ 2.6.4 Speed Dialling—Personal/System) In this case, the extension will enter into the personal programming mode automatically so that a name can be assigned for the stored number.

User Manual References

3.1.2 Settings on the Programming Mode

2.5.2 Automatic Extension Release

Description

After going off-hook, if an extension user fails to dial any digits within a preprogrammed time period, the user will hear a reorder tone. This operation applies to intercom calls only. This feature is also known as Automatic Station Release.

Conditions

- A PT/PS user hears a reorder tone for a preprogrammed time period, and then the PT/PS returns to idle status automatically. However, an SLT user hears a reorder tone until he goes on-hook.
- **This feature works in one of the following cases:**
  When making an intercom call
  a. If the first digit is not dialled within a preprogrammed time period.
  b. After a digit is dialled, if subsequent digits are not dialled within a preprogrammed time period.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone
→◆ Dial—Extension First Digit (s)
→◆ Dial—Extension Inter-digit (s)
→◆ Tone Length—Reorder Tone for PT Handset (s)
→◆ Tone Length—Reorder Tone for PT Hands-free (s)

2.5.3 Intercom Call

Description

An extension user can call another extension user.

Conditions

- **Extension Number/Name Assignment**
  Extension numbers and names are assigned to all extensions. The assigned number and name are shown on display PTs during intercom calls.
- **DSS Button**
  It is possible to access another extension with one touch by pressing the corresponding Direct Station Selection (DSS) button. A flexible button can be customised as a DSS button.
- **Call Directory—Extension Dialling**
  A display PT user can make a call by selecting one of the stored names on the display.
- **Limiting the display by tenant—Call Directory**
  For Call Directory, an extension can reference the data for all tenants or for each tenant the extension is member of, depending on system programming. In "**Each Tenant**" mode, Call Directory is displayed on display PTs as follows:
  Only information about extensions that belong to the tenant is displayed.
- **Alternate Receiving—Ring/Voice**
  A PT user can select to receive intercom calls by ring tone or by voice, through personal programming (Alternate Receiving—Ring/Voice). If a user selects voice-calling, the calling party talks to the user immediately after a confirmation tone. Denying voice-calling can also be selected.
- **Alternate Calling—Ring/Voice**
  A caller can change the called party's preset call receiving method (ring tone or voice) temporarily. By doing so, ring-calling is switched to voice-calling, or vice versa, at the called party. The called party may deny voice-calling.
- **PDN/SDN**
  It is not possible to temporarily change the called party's preset call receiving method when making a call using a Primary Directory Number (PDN) button or Secondary Directory Number (SDN) button (→ 2.9.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension).
- **Tone after Dialling**
  After dialling an extension number, a user will hear one of the following:

Type	Description
Ringback Tone	Indicates the called party is being called.
Confirmation Tone	Indicates the called party has set voice-calling.
Busy Tone	Indicates the called party is busy.
DND Tone	Indicates the called party has set DND.

PC Programming Manual References

10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature—◆ Alternate Calling - Ring / Voice
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Main—◆ Extension Number
→Main—◆ Extension Name
→Option 3—◆ Intercom Call by Voice

12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main
→◆ Extension Number
→◆ Extension Name

PT Programming Manual References

[003] Extension Number
[004] Extension Name

Feature Guide References

2.21.2 Flexible Buttons

User Manual References

1.2.1 Basic Calling
1.2.6 Alternating the Calling Method (Alternate Calling—Ring/Voice)
1.14.2 Using the Directories
3.1.2 Settings on the Programming Mode

2.5.4 Trunk Call Features

2.5.4.1 Trunk Call Features—SUMMARY

Description

An extension user can use the following features when making a trunk call:

Feature	Description & Reference
Emergency Call	A user can dial the preprogrammed emergency numbers regardless of the restrictions imposed on the extension. → 2.5.4.2 Emergency Call
Account Code Entry	A user can enter an account code to identify outgoing calls for accounting and billing purposes. → 2.5.4.3 Account Code Entry
Pulse to Tone Conversion	A user can temporarily switch from Pulse mode to DTMF mode to access special services. → 2.5.4.4 Dial Type Selection
Pause Insertion	A user can insert a preprogrammed Pause time into a dialling number by pressing the PAUSE button, or it is automatically inserted between the user-dialled code (e.g., Host PBX Access code or Special Carrier Access code) and the following digits. → 2.5.4.7 Pause Insertion → 2.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX) → 2.5.4.9 Special Carrier Access Code

2.5.4.2 Emergency Call

Description

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

Conditions

- A specified number of emergency numbers can be stored (some may have default values).
- Emergency numbers may be called even when:
  - in Account Code—Forced mode (→ 2.5.4.3 Account Code Entry)
  - in any TRS/Barring levels (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring))
  - after the preprogrammed call charge limit is reached (→ 2.7.2 Budget Management)
  - in Extension Dial Lock (→ 2.7.3 Extension Dial Lock)
- **CLIP Number Notification**
  When dialling an emergency number, the preassigned CLIP number for the extension will be sent as a location identification number. (→ 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP))

PC Programming Manual References

15.4 PBX Configuration—[7-4] TRS—Emergency Dial

PT Programming Manual References

[304] Emergency Number

Feature Guide References

6.1 Capacity of System Resources

2.5.4.3 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 SMDR call record. If, for example, a firm uses an account code for each client, the firm can determine what calls were made for the client, and can submit a bill to the client according to the client's account code on the SMDR call record.
There are two methods of entering account codes as follows:
One of the methods is selected for each extension on a COS basis.

Mode	Description
Option	A user can enter an account code if needed at any time desired.
Forced	A user must always enter an account code before seizing a trunk.

Conditions

- An account code can be stored into Memory Dialling (e.g., One-touch Dialling).
- **Account Button**
  A flexible button can be customised as the Account button. The Account button is used in place of the feature number for entering an account code. This button is useful because it can be used at any time, while feature number entry is allowed only when hearing a dial tone before seizing a trunk.
- Extension users can enter an account code at any time during a call, including after the call has been disconnected and a reorder tone is heard. However, if an account code is entered after there is no longer a reorder tone, the call will not be stored in the SMDR record.
- If more than one account code is entered, the code entered last is printed out on SMDR.
- Even in Forced mode, emergency numbers can be dialled out without an account code.
  (→ 2.5.4.2 Emergency Call)
- PT users can also enter an account code for incoming trunk calls during a conversation.
- **Verification Code Entry**
  To identify who made a trunk call for accounting and billing purposes, a verification code is used. This code can be used at any extension. (→ 2.7.6 Verification Code Entry)

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Account Code Entry
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—CO & SMDR—◆ Account Code Mode

PT Programming Manual References

[508] Account Code Mode

Feature Guide References

2.21.2 Flexible Buttons
2.22.1.1 Station Message Detail Recording (SMDR)
5.1.1 Class of Service (COS)

User Manual References

1.2.1 Basic Calling

2.5.4.4 Dial Type Selection

Description

The dialling mode (rotary or tone) can be selected for each analogue trunk through system programming regardless of the originating extension (under contract with the telephone company).
There are the following modes:

Mode	Description
DTMF (Dual Tone Multi-Frequency)	The dialling signal from an extension is converted to tone dialling. DTMF signals are transmitted to the trunk.
Pulse Dial (Rotary)	The dialling signal from an extension is converted to rotary dialling. Rotary pulses are transmitted to the trunk.

Conditions

- **Pulse to Tone Conversion**
  It is possible for an extension user to temporarily switch from Pulse mode to DTMF mode so that the user can access special services such as computer-accessed long distance calling or voice mail services. To switch to DTMF mode, wait for a preprogrammed time period (Default: five seconds) after the trunk is connected, or press ✳. This feature works only on trunks set to Pulse mode. DTMF mode cannot be changed to Pulse mode.
- It is possible to select the pulse rate for a trunk port that has been set to Pulse mode. There are two pulse rates: Low (10 pps) and High (20 pps).
- It is possible to assign the minimum duration of the DTMF signal sent to a trunk port that has been set to DTMF mode.

PC Programming Manual References

9.24 PBX Configuration—[1-1] Configuration—Slot—Port Property - LCO Port
→◆ Dialling Mode
→◆ DTMF Width
→◆ Pulse Speed

PT Programming Manual References

[410] LCOT Dialling Mode
[411] LCOT Pulse Rate
[412] LCOT DTMF Minimum Duration

2.5.4.5 Reverse Circuit

Description

A circuit in the PBX can detect the reverse signal from the telephone company when an extension user tries to make a trunk call. This detects the start (a called party goes off-hook) and end (the called party goes on-hook) of an outgoing trunk call. When a trunk call is received, the circuit can also detect the reverse signal after an outside caller goes on-hook.
If Reverse Circuit Detection is disabled, the total duration of the call is not accurately recognised by the PBX. The duration of a call can be verified on SMDR using this feature (→ 2.22.1.1 Station Message Detail Recording (SMDR)).
It is possible to select whether the PBX detects the reverse signal for outgoing trunk calls only, or for both outgoing and incoming trunk calls, or for no trunk calls (detection disabled) through system programming.

PC Programming Manual References

9.24 PBX Configuration—[1-1] Configuration—Slot—Port Property - LCO Port—◆ Reverse Detection

PT Programming Manual References

[415] LCOT Reverse Circuit

2.5.4.6 Trunk Busy Out

Description

The PBX can monitor the loop current sent through analogue trunks, preventing users from seizing trunks where a loop current is not detected. When loop currents are not detected, trunks are set to Busy Out status, and become unable to make or receive calls. A trunk in Busy Out status cannot be used for making calls as a TIE line, as part of a trunk group, or with the ARS feature, and cannot receive trunk calls. If a user tries to seize a trunk set to Busy Out status, the user will hear a reorder tone.
This is useful if some or all trunks are occasionally unavailable because of problems with the external telecommunications environment.

Conditions

- Loop current detection is performed on active trunks whenever the trunk is seized and/or at fixed intervals.
- When a trunk is in busy-out status, loop current detection is performed at fixed intervals, returning the trunk to in-service status once a loop current is detected. An extension assigned as the manager can manually change the trunk back to in-service status.
- Trunk status changes are recorded in the error log of the PBX.
- Busy Out status is maintained even when the PBX is reset.
- Busy Out status is cleared when:
  - a call is successfully received (i.e., a loop current is detected) on that trunk.
  - the S-CO button for that trunk is pressed and a loop current is detected.

PC Programming Manual References

9.24 PBX Configuration—[1-1] Configuration—Slot—Port Property - LCO Port—◆ Busy Out Status
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Busy Out Cancel
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Manager—◆ Manager
10.9 PBX Configuration—[2-9] System—System Options—Option 5—◆ Busy Out—Busy Out for Analogue CO

PT Programming Manual References

[511] Manager Assignment

Feature Guide References

5.1.6 Manager Features

User Manual References

2.1.6 Allowing Users to Seize an Unavailable Outside Line (Trunk Busy Out)

2.5.4.7 Pause Insertion

Description

Pressing a PAUSE button inserts a preprogrammed Pause time between digits of a user-dialled number before the number is dialled out, allowing certain numbers separated with a pause to be used to access certain features (e.g., access codes, seizing idles lines, etc.).
When a pause is needed, pauses must be inserted manually (pressing the PAUSE button) in all cases, except for the following access codes where a pause is automatically inserted between the user-dialled access code and the subsequent digits:

a. Host PBX Access code (→ 2.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX))
b. Special Carrier Access code (→ 2.5.4.9 Special Carrier Access Code)
c. Second Dial Tone Waiting code

Conditions

- The Pause time is programmable for each trunk.
- Pauses can be stored in Memory Dialling.
- When a Second Dial Tone Waiting code is dialled after seizing a trunk, a preprogrammed number of pauses are inserted after the code.
- **ARS**
  A pause is not automatically inserted between the user-dialled access code and the subsequent digits when the ARS mode is enabled. (2.8.1 Automatic Route Selection (ARS))

PC Programming Manual References

9.24 PBX Configuration—[1-1] Configuration—Slot—Port Property - LCO Port—◆ Pause Time
10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ During Conversation—Pause Signal Time (s)
14.4 PBX Configuration—[6-4] Feature—Second Dial Tone

PT Programming Manual References

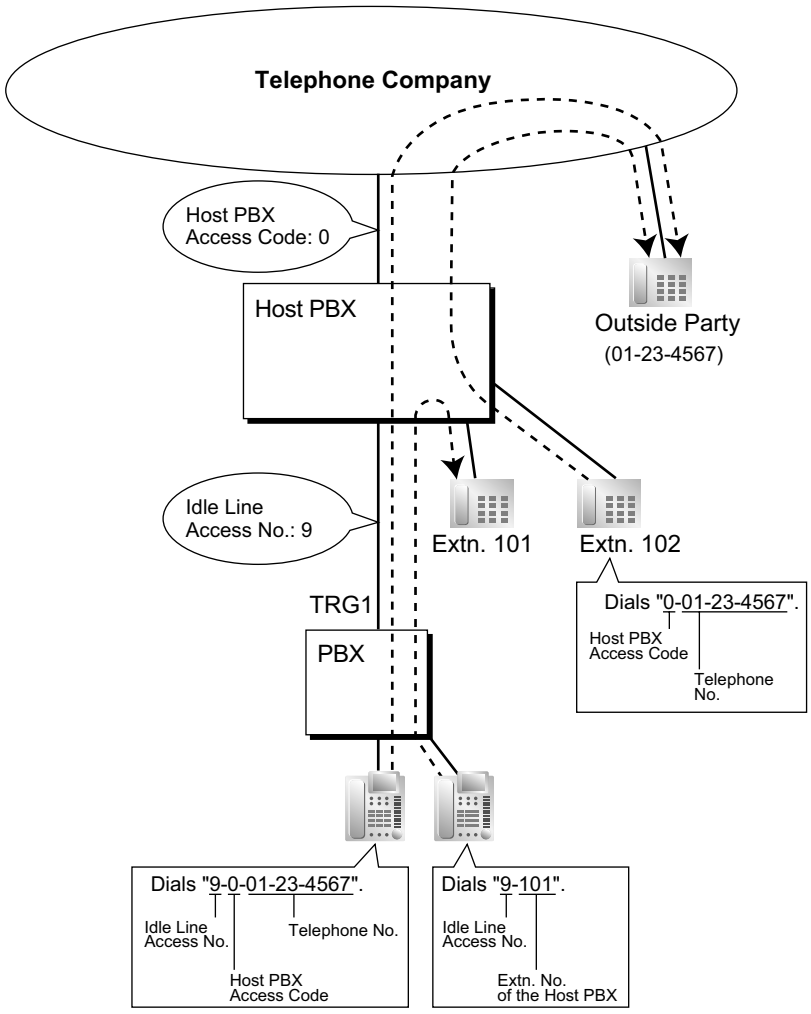
[416] LCOT Pause Time

2.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)

Description

This PBX can be installed behind an existing PBX (host PBX) by connecting the extension ports of the host PBX to the trunk ports of this PBX (behind PBX). A Host PBX Access code is required for the behind PBX to access the telephone company (e.g., to make outside calls) through the host PBX. The Trunk access number of the host PBX should be stored as a Host PBX Access code on a trunk group of the behind PBX.
A preprogrammed Pause time will be automatically inserted between the user dialled Host PBX Access code and the subsequent digits (2.5.4.7 Pause Insertion).

[Example]



Note

"0" should be assigned as a Host PBX Access code for trunk group (TRG) 1 of the behind PBX.

Conditions

- **TRS/Barring**

TRS/Barring checks only the dialled telephone number excluding the Host PBX Access code when accessing the telephone company through the host PBX. (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring))

- **ARS**
  A pause is not automatically inserted between the user-dialled access code and the subsequent digits when the ARS mode is enabled. (2.8.1 Automatic Route Selection (ARS))
- **SMDR**
  The dialled number including the Host PBX Access code will be recorded on SMDR only if the modified number setting is selected in the ARS setting for SMDR.
- When a Host PBX Access code is assigned to a trunk group, calls to extensions of the host PBX are not recorded on SMDR.
- A Host PBX Access Code can be used to record only long distance calls on SMDR when a trunk port is connected directly to the telephone company (not a host PBX). This is allowed when the long distance code (e.g., "0") is assigned as the Host PBX Access code. All local calls (e.g., calls that do not require a "0" to be dialled first) are treated as extensions of the telephone company and do not get recorded on SMDR, because in this case this PBX recognises the telephone company as the host PBX. Therefore, only long distance calls are recorded on SMDR.

PC Programming Manual References

11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Host PBX Access Code
19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—ARS Dial

PT Programming Manual References

[471] Host PBX Access Code

Feature Guide References

2.22.1.1 Station Message Detail Recording (SMDR)
6.1 Capacity of System Resources

2.5.4.9 Special Carrier Access Code

Description

If the PBX has access to multiple telephone companies, a Special Carrier Access code assigned through system programming is required every time a trunk call is made without using ARS.
A preprogrammed Pause time will be automatically inserted between the user-dialled Special Carrier Access code and the subsequent digits. (→ 2.5.4.7 Pause Insertion)

Conditions

- **TRS/Barring**
  TRS/Barring checks only the dialled telephone number excluding the Special Carrier Access code. (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring))
- **ARS**
  A pause is not automatically inserted between the user-dialled access code and the subsequent digits when the ARS mode is enabled. (2.8.1 Automatic Route Selection (ARS))
- **If this PBX is installed behind an existing host PBX:**
  A Special Carrier Access code and a Host PBX Access code should be assigned separately: these codes cannot be assigned together as one code. (→ 2.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX))

PC Programming Manual References

15.3 PBX Configuration—[7-3] TRS—Special Carrier

PT Programming Manual References

[303] Special Carrier Access Code

Feature Guide References

2.8.1 Automatic Route Selection (ARS)
6.1 Capacity of System Resources

2.5.5 Seizing a Line Features

2.5.5.1 Seizing a Line Features—SUMMARY

Description

An extension user can select the line seized for making calls by the following methods:

Feature	Description & Reference
Line Preference—Outgoing	A user can select the line to be seized when going off-hook. → 2.5.5.2 Line Preference—Outgoing
Trunk Access	A user can select the Trunk Access method every time he makes a trunk call. → 2.5.5.3 Trunk Access

2.5.5.2 Line Preference—Outgoing

Description

Through personal programming (Preferred Line Assignment-Outgoing), PT users can select the outgoing line they prefer to originate calls on when going off-hook, from the following line preferences:

Line Preference	Description
ICM/PDN	When an extension user goes off-hook, an extension line is selected automatically. If the extension is a PDN extension, the first available Primary Directory number (PDN) button is selected automatically. (→ 2.9.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension)
Idle Line	When an extension user goes off-hook, an idle trunk is selected automatically from the assigned trunk groups.
No Line	When an extension user goes off-hook, no line is selected. The extension user must select the desired line to make a call.
Prime Line	When an extension user goes off-hook, the preset line is selected automatically. A prime line can be selected from the Line Access buttons: S-CO, G-CO, L-CO, ICD Group.

Conditions

- **Line Preference Override**
  A user can override the preset Line Preference temporarily by pressing the desired Line Access button or Memory Dialling button (e.g., One-touch Dialling) before going off-hook.
- To select Idle Line Preference, the trunk groups available to the extension should be programmed on a COS basis. Also trunk groups available for Idle Line Access should be assigned.

PC Programming Manual References

10.7.2 PBX Configuration—[2-7-2] System—Class of Service—External Call Block
11.1.2 PBX Configuration—[3-1-2] Group—Trunk Group—Local Access Priority
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 4—◆ Outgoing Preferred Line
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 4—◆ Outgoing Preferred Line

PT Programming Manual References

[103] Idle Line Access (Local Access)
[500] Trunk Group Number

User Manual References

3.1.2 Settings on the Programming Mode

2.5.5.3 Trunk Access

Description

The following methods can be used to access a trunk:

Method	Description	Accessing method
Idle Line Access (Local Access)	Selects an idle trunk automatically from the assigned trunk groups.	Dial the Idle Line Access number, or press a L-CO button.
Trunk Group Access	Selects an idle trunk from the corresponding trunk group.	Dial the Trunk Group Access number and a trunk group number, or press a G-CO button.
S-CO Line Access	Selects the desired trunk directly.	Dial the S-CO Line Access number and the trunk number, or press the S-CO button.

Conditions

- COS programming determines the trunk groups available for making calls.
- Trunk numbers can be referred on a trunk port basis.
- **Button Assignment**
  A flexible button can be customised as a G-CO, L-CO, or S-CO button as follows:

Type	Parameter
Loop-CO (L-CO)	No parameter (all assigned trunk groups through system programming are applied.)
Group-CO (G-CO)	A specified trunk group.
Single-CO (S-CO)	A specified trunk.

  It is possible to assign:
  – the same trunk to the S-CO button and to a G-CO button.
  – the same trunk group to more than one G-CO button.
  – more than one L-CO button.

  Dialling the Trunk Access number selects a CO button in the following order: S-CO → G-CO → L-CO
- **Direct Trunk Access**
  – Pressing an idle CO button automatically switches on the hands-free operation mode and allows a user to use On-hook Dialling. The user need not press the SP-PHONE button, MONITOR button, or lift the handset.
  – When a user of a UT-series SIP extension uses an S-CO button to seize a trunk and then initiates a call, the outgoing call may be disrupted by an incoming call. In this case, the user hears a reorder tone.
- **Group Hunting Order for Idle Line Access**
  An idle trunk is selected from the trunk groups assigned for Idle Line Access. If multiple trunk groups are available, the trunk group hunting sequence can be determined through system programming.
- **Trunk Hunting Order for Idle Line Access and Trunk Group Access**
  The trunk hunting sequence in a trunk group (from lowest numbered trunk, from highest numbered trunk or rotation) can be determined through system programming.

- A company name or customer name can be assigned on a trunk port basis so that the operator or extension user can view the destination that the external caller is trying to reach before answering. This is useful, for example, when multiple companies share the same operator.
- It is possible to identify the trunk ports that have trunks connected. This prevents extension users from originating a call to a trunk that is not connected.

PC Programming Manual References

9.22 PBX Configuration—[1-1] Configuration—Slot—Extension—Port Property—◆ Connection
9.24 PBX Configuration—[1-1] Configuration—Slot—Port Property - LCO Port—◆ Connection
9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—◆ Connection
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ Idle Line Access (Local Access)
→◆ Trunk Group Access
→◆ Single CO Line Access
10.7.2 PBX Configuration—[2-7-2] System—Class of Service—External Call Block
11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Main—◆ Line Hunting Order
11.1.2 PBX Configuration—[3-1-2] Group—Trunk Group—Local Access Priority
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Parameter Selection (for Single CO)
→◆ Parameter Selection (for Group CO)
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Parameter Selection (for Single CO)
→◆ Parameter Selection (for Group CO)
16.1 PBX Configuration—[8-1] ARS—System Setting—◆ ARS Mode
18.1 PBX Configuration—[10-1] CO & Incoming Call—CO Line Settings—◆ CO Name

PT Programming Manual References

[400] Trunk Connection
[401] Trunk Name
[409] Trunk Number Reference
[500] Trunk Group Number

Feature Guide References

2.21.2 Flexible Buttons
5.1.1 Class of Service (COS)

User Manual References

1.2.1 Basic Calling

2.6 Memory Dialling Features

2.6.1 Memory Dialling Features—SUMMARY

Description

An extension user can store frequently dialled numbers in the PBX extension data and/or the PBX system data. A stored number is dialled automatically with a simple operation.

1. Features

Feature		Storing Method & Reference
One-touch Dialling		• Personal Programming • System Programming (PC Programming only) → 2.6.2 One-touch Dialling
Last Number Redial (Outgoing Call Log)		Recently dialled telephone numbers are automatically stored. → 2.6.3 Last Number Redial
Speed Dialling	**Personal**	• Personal Programming • Personal Operation with the Feature Number • System Programming (PC Programming only) → 2.6.4 Speed Dialling—Personal/System
	System	System Programming → 2.6.4 Speed Dialling—Personal/System
Quick Dialling		System Programming (PC Programming only) → 2.6.5 Quick Dialling
Hot Line		• Personal Programming • Personal Operation with the Feature Number • System Programming (PC Programming only) → 2.6.6 Hot Line
KX-T7710 One-touch Dialling		System Programming (PC Programming only) → 2.6.7 KX-T7710 One-touch Dialling
Incoming Call Log		Incoming call information is automatically stored. → 2.19.2 Incoming Call Log

2. Valid Input

Input	Display while Entering	Description
0–9/✳/#	0–9/✳/#	Store the digits ✳ and #.
PAUSE (Pause)	P	Store a pause by pressing the PAUSE button. (→ 2.5.4.7 Pause Insertion)
FLASH/RECALL (Hooking)*1	F	Store a flash/recall signal (EFA mode) by pressing the FLASH/RECALL button at the beginning of the number. (→ 2.11.7 External Feature Access (EFA))
INTERCOM (Secret)*1	[ ]	Conceal all or part of the number by pressing the INTERCOM button at the beginning and at the end of the number to be concealed. It is programmable whether the concealed part will appear on SMDR.
TRANSFER (Transfer)*1	T	Store a transfer command by pressing the TRANSFER button at the beginning of the number (used only for a One-touch Dialling). (→ 2.12.1 Call Transfer) **[Example]** Storing "T + 305"= Transferring a call to extension 305.

*1 Available only when in system/personal programming mode

[Secret Dialling Example]

When storing the number "9-123-456-7890" and concealing the telephone number "123-456-7890",

Enter [9] → [INTERCOM] → [1][2][3][4][5][6][7][8][9][0] → [INTERCOM].

Note

- It is possible to store a Memory Dialling feature number at the beginning of the Memory Dialling numbers.
- It is possible to store several feature numbers in one Memory Dialling location.

Conditions

- **Trunk Access by Memory Dialling**
  A specific Trunk Access number can be stored with the telephone number in Memory Dialling. However, if Memory Dialling is done after selecting a trunk, the stored Trunk Access number is ignored and the telephone number is sent using the selected trunk.

2.6.2 One-touch Dialling

Description

A PT 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 Dialling button.

Example: One-touch Voice Mail Feature Access

It is possible to assign a One-touch Dialling button for direct access to a Unified Messaging feature (→ Section 3 Unified Messaging System). For example, to record a message to mailbox number 123 directly and the UM group's floating extension number is 165, assign "**165**#6**123**" to a One-touch Dialling button. When pressing this button, the mailbox's greeting message will be heard.

Conditions

- **One-touch Dialling Button**
  A flexible button can be customised as a One-touch Dialling button.
- **Full One-touch Dialling**
  There is no need to go off-hook before pressing the One-touch Dialling button.

PC Programming Manual References

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 6—◆ Flexible Button Programming Mode
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Dial (for One-touch)
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 6—◆ Flexible Button Programming Mode
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Dial (for One-touch)

Feature Guide References

2.21.2 Flexible Buttons
6.1 Capacity of System Resources

User Manual References

1.2.2 Easy Dialling

2.6.3 Last Number Redial

Description

Every extension automatically saves recently dialled external telephone numbers and extension numbers to allow the same number to be dialled again easily. Through system programming, the outgoing call log can be set to log dialled extension numbers.

Automatic Redial:
If Last Number Redial is performed in hands-free mode and the called party is busy, redialling will be automatically retried a preprogrammed number of times at preprogrammed intervals. The Redial Call No-answer Ring Duration time is programmable.
This feature is available only on certain PT models which have the SP-PHONE button.
Outgoing Call Log:
Information on outgoing trunk calls and intercom (including TIE) calls is automatically logged at each extension. Users of display PTs can view details of a preset number of recently dialled telephone numbers, and easily call the same party again.

Conditions

- If a new number is dialled when the Outgoing Call Log is full and/or Automatic Redial contains a number, the data of the oldest stored call will be deleted, and the new number will be stored.
- If any dialling operations are performed or an incoming call is answered during Automatic Redial, Automatic Redial is cancelled.
- Automatic Redial may not be available depending on the busy tone pattern.
- Automatic Redial is not available on SIP extensions.
- **Interrupt Redial**
  When an outside party, seized trunk, or extension number (including TIE connections) is busy, a user can attempt to redial the number by pressing the REDIAL button without going on-hook. This can be performed several times without having to go on-hook.
- **Outgoing Call Log Display by REDIAL Button**
  Pressing the REDIAL button on a display PT while on-hook can display the Outgoing Call Log. System programming is required for this operation.
- If the Outgoing Call Log is used to redial an outside party or an extension number (including TIE connections) or if a number that is already stored in the Outgoing Call Log is manually redialled again, the number will be stored in the call log multiple times. However, calls made using the REDIAL button are not stored in the Outgoing Call Log again.
- It is possible to change the number of records that can be stored at each extension through system programming.
- To log intercom calls in the outgoing call log, refer to "10.9 PBX Configuration—[2-9] System—System Options—Option 7—◆ Outgoing Call Log—Extension Call" in the PC Programming Manual.
- Logs for multiple calls to the same destination are combined and displayed with the most recent call log.
- If an extension user makes a call over a TIE connection using the PBX Code method (Access with PBX Code), the outgoing call log does not display the Access Code on the PT's display.
- If an extension user uses a DSS key to make a call to another extension, the user can use the redial feature to call the same extension number.
- If an extension user uses an SDN key to make a call to the corresponding owner extension, the user cannot use the redial feature to call the owner extension again.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone
→◆ Automatic Redial—Repeat Counter

→◆ Automatic Redial—Repeat Interval (x10s)
→◆ Automatic Redial—Redial Call Ring Duration (x10s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Redial
10.9 PBX Configuration—[2-9] System—System Options—Option 2
→◆ Redial—Automatic Redial when No Answer (ISDN)
→◆ Redial—Save Dial After Connection to Redial Memory
→◆ Redial—Call Log by Redial key
10.9 PBX Configuration—[2-9] System—System Options—Option 7—◆ Outgoing Call Log—Extension Call
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 7—◆ Outgoing Call Log Memory
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 7—◆ Outgoing Call Log Memory

PT Programming Manual References

[205] Automatic Redial Repeat Times
[206] Automatic Redial Interval

Feature Guide References

5.2.1 IP Proprietary Telephone (IP-PT)
6.1 Capacity of System Resources

User Manual References

1.2.3 Redial
1.14.1 Using the Call Log

2.6.4 Speed Dialling—Personal/System

Description

An extension user can make calls using abbreviated dialling for frequently dialled numbers which are stored in the PBX extension data, or the PBX system data.
Personal Speed Dialling is also known as Station Speed Dialling.
Depending on system programming, the System Speed Dial items displayed on a display PT can be limited to items related to the extension's tenant.

Conditions

[General]

- Any number (e.g., telephone number, feature number) can be stored in a speed dialling number. A name can be assigned to each Personal Speed Dialling number through personal programming, and System Speed Dialling number.

[Personal Speed Dialling]

- **Display Lock**
  An extension user can lock the Personal Speed Dialling number display to prevent the numbers from being viewed at any extension through personal programming (Display Lock). In this case, the Incoming/Outgoing Call Log displays are also locked, and the voice messages in the user's mailbox cannot be played back. An extension personal identification number (PIN) is required to use this feature. (→ 2.24.1 Extension Personal Identification Number (PIN))

[System Speed Dialling]

- **TRS/Barring Override by System Speed Dialling**
  It is possible to override the TRS/Barring using the System Speed Dialling. (→ 2.7.1 Toll Restriction (TRS)/ Call Barring (Barring))
- **System Speed Dialling Display by AUTO DIAL/STORE Button**
  Pressing the AUTO DIAL/STORE button on a display PT while on-hook can display the System Speed Dialling Directory.

[Limiting the display by tenant—System Speed Dial]

For System Speed Dial, an extension can reference the data for System or for each tenant the extension is member of, depending on system programming. In "**Tenant Exclusive**" mode, System Speed Dial is displayed on display PTs as follows:

- Only information about extensions that belong to the tenant as is displayed.
- Changes to System Speed Dial settings only affect each tenant. They do not affect the whole system.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ System Speed Dialling / Personal Speed Dialling
→◆ Personal Speed Dialling - Programming
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—TRS—◆ TRS Level for System Speed Dialling
12.1.3 PBX Configuration—[4-1-3] Extension—Wired Extension—Speed Dial
14.1 PBX Configuration—[6-1] Feature—System Speed Dial
14.6 PBX Configuration—[6-6] Feature—Tenant—◆ System Speed Dial
14.6 PBX Configuration—[6-6] Feature—Tenant—◆ Extension Directory

PT Programming Manual References

[001] System Speed Dialling Number
[002] System Speed Dialling Name
[509] TRS/Barring Level for System Speed Dialling

Feature Guide References

6.1 Capacity of System Resources

User Manual References

1.2.2 Easy Dialling
1.14.1 Using the Call Log
1.14.2 Using the Directories
3.1.2 Settings on the Programming Mode

2.6.5 Quick Dialling

Description

An extension user can access an extension or feature by simply dialling a 1–8 digit Quick Dialling number.

Conditions

- Quick Dialling is convenient in cases such as the following:
  - Room service calls in a hotel
  - Calling another branch via the public network.
- Quick Dialling numbers follow the flexible numbering plan.
  (→ 5.5.7 Flexible Numbering/Fixed Numbering)
- The following example shows how Quick Dialling numbers can be stored and utilised:

Location No.	Quick Dialling No.	Desired Number
Quick Dialling 01	110	9110 (Trunk Call)
Quick Dialling 02	5	3016 (Room Service)
Quick Dialling 03	2011	90123456789 (Another Branch)
:	:	:

PC Programming Manual References

10.6.2 PBX Configuration—[2-6-2] System—Numbering Plan—Quick Dial

Feature Guide References

4.2.3 ISDN Virtual Private Network (ISDN-VPN)
6.1 Capacity of System Resources

User Manual References

1.2.2 Easy Dialling

2.6.6 Hot Line

Description

An extension user can set his extension to automatically dial a preprogrammed telephone or extension number whenever he goes off-hook. This feature is also known as Pickup Dialling.
If the Hot Line feature is set, a dial tone is generated for a specified Waiting time assigned through system programming when the user goes off-hook. During the Waiting time the user can dial another party, overriding the Hot Line feature. If no number is dialled, the preprogrammed number will automatically start being dialled.

Conditions

- **Capable Telephone**
  PT, SLT, and PS

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Dial—Hot Line (Pickup Dial) Start (s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Hot Line (Pickup Dial) Program Set / Cancel
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 2
→◆ Pickup Dial Set
→◆ Pickup Dial No.
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 2
→◆ Pickup Dial Set
→◆ Pickup Dial No.

PT Programming Manual References

[204] Hot Line Waiting Time

Feature Guide References

6.1 Capacity of System Resources

User Manual References

1.2.2 Easy Dialling

2.6.7 KX-T7710 One-touch Dialling

Description

The Message button and One-touch buttons on all KX-T7710 telephones connected to the PBX can be customised at once through system programming. The same extension number, telephone number, or feature number will be assigned to the same buttons on each KX-T7710, useful for hotel room extensions or similar applications.

[Programming Example]

Button	Desired Number
MESSAGE	⚹702 (Message Waiting [To Call Back])
One-touch Dial 01	100 (Hotel Operator)
One-touch Dial 02	⚹7601 (Wake-up Call)
One-touch Dial 03	102 (Restaurant)
:	:

The MESSAGE button is programmed by default to call back a caller who left a message waiting indication. However, the MESSAGE button can be programmed to perform other features. The eight One-touch buttons have no default setting.

Conditions

- The KX-T7710 has two modes, NORMAL mode and PBX mode, selected by a switch on the telephone. This feature is available only when the KX-T7710 is in the PBX mode.
- This feature is available while hearing a dial tone.
- Please refer to the Quick Reference Guide of the KX-T7710 for additional information.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main

Feature Guide References

2.21.2 Flexible Buttons

User Manual References

1.2.2 Easy Dialling

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

2.7.1 Toll Restriction (TRS)/Call Barring (Barring)

Description

TRS/Barring can prohibit an extension user from making certain trunk calls by COS programming. It is applied when the user goes off-hook, a trunk is seized and then a dialled number is sent to the trunk.
Each COS is programmed to have a TRS/Barring level for each time mode (day/lunch/break/night).
There are seven levels available. Level 1 is the highest level and level 7 is the lowest. That is, level 1 allows all trunk calls and level 7 prohibits all trunk calls. Levels 2 through 6 are used to restrict calls by combining preprogrammed Denied and Exception Code Tables.

Denied Code Tables

An outgoing trunk call made by an extension with a level between 2 and 6 is first checked against the applicable Denied Code Tables. If the leading digits of the dialled number (not including the Trunk Access number) are not found in the table, the call is made. There are five Denied Code Tables, one for each of Levels 2 through 6 respectively.
Complete every table by storing numbers that are to be prohibited. These numbers are defined as denied codes.

Exception Code Tables

These tables are used to override a programmed denied code. A call denied by the applicable Denied Code Tables is checked against the applicable Exception Code Tables, and if a match is found, the call is made. There are five Exception Code Tables, for Levels 2 through 6 respectively.
Complete every table by storing numbers that are exceptions to the denied codes. These numbers are defined as exception codes.

TRS/Barring Override by System Speed Dialling

If the call is made using System Speed Dialling, the call can override the TRS/Barring. Each COS is programmed to have a TRS/Barring level for System Speed Dialling.
Once this feature is set, it permits all extension users to make System Speed Dialling calls with the level for System Speed Dialling. Any extension which sets Extension Dial Lock can also make a call using System Speed Dialling.
→ 10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—TRS—◆ TRS Level for System Speed Dialling

TRS/Barring Level

The TRS/Barring level is determined by the telephone codes set in the Denied Code Tables and Exception Code Tables.
As shown in the table below, the Denied Code Tables for the higher levels are applied to all levels below it, and the Exception Code Tables for the lower levels are applied to all levels above it.

	Denied Code Tables*1	Exception Code Tables*2
Level 1	Not Programmable	Not Programmable
Level 2	Table for Level 2	Tables for Levels 2 through 6
Level 3	Tables for Levels 2 and 3	Tables for Levels 3 through 6
Level 4	Tables for Levels 2 through 4	Tables for Levels 4 through 6

	Denied Code Tables*1	Exception Code Tables*2
Level 5	Tables for Levels 2 through 5	Tables for Levels 5 through 6
Level 6	Tables for Levels 2 through 6	Table for Level 6
Level 7	Not Programmable	Not Programmable

*1 → 15.1 PBX Configuration—[7-1] TRS—Denied Code—◆ Level 2–Level 6
*2 → 15.2 PBX Configuration—[7-2] TRS—Exception Code—◆ Level 2–Level 6

[Usage Example] Using this method, certain outgoing trunk calls (e.g., international/cellular phone/long distance) can be restricted as in the example below:

	Restricted	Allowed
Level 1	No restriction	
Level 2 (Boss)	• International Calls	• Countries where Clients are Located • Cellular Phone Calls • Long Distance Calls • Local Calls
Level 3 (Secretary)	• International Calls • Cellular Phone Calls	• Boss's Cellular Phone • Long Distance Calls • Local Calls
Level 4 (Operator)	• International Calls • Cellular Phone Calls • Long Distance Calls	• Local Calls
:	:	:

In this example, a level 1 user can make any trunk calls. A level 2 user can make international calls to the countries where clients are located, and can also make cellular phone/long distance/local calls. A level 3 user cannot make international/cellular phone calls apart from to the boss's cellular phone, but can make long distance/local calls. A level 4 user cannot make any international/cellular phone/long distance calls, but can make local calls.
To set TRS/Barring as in the example above, it is necessary to programme the Denied Code and Exception Code Tables as follows:

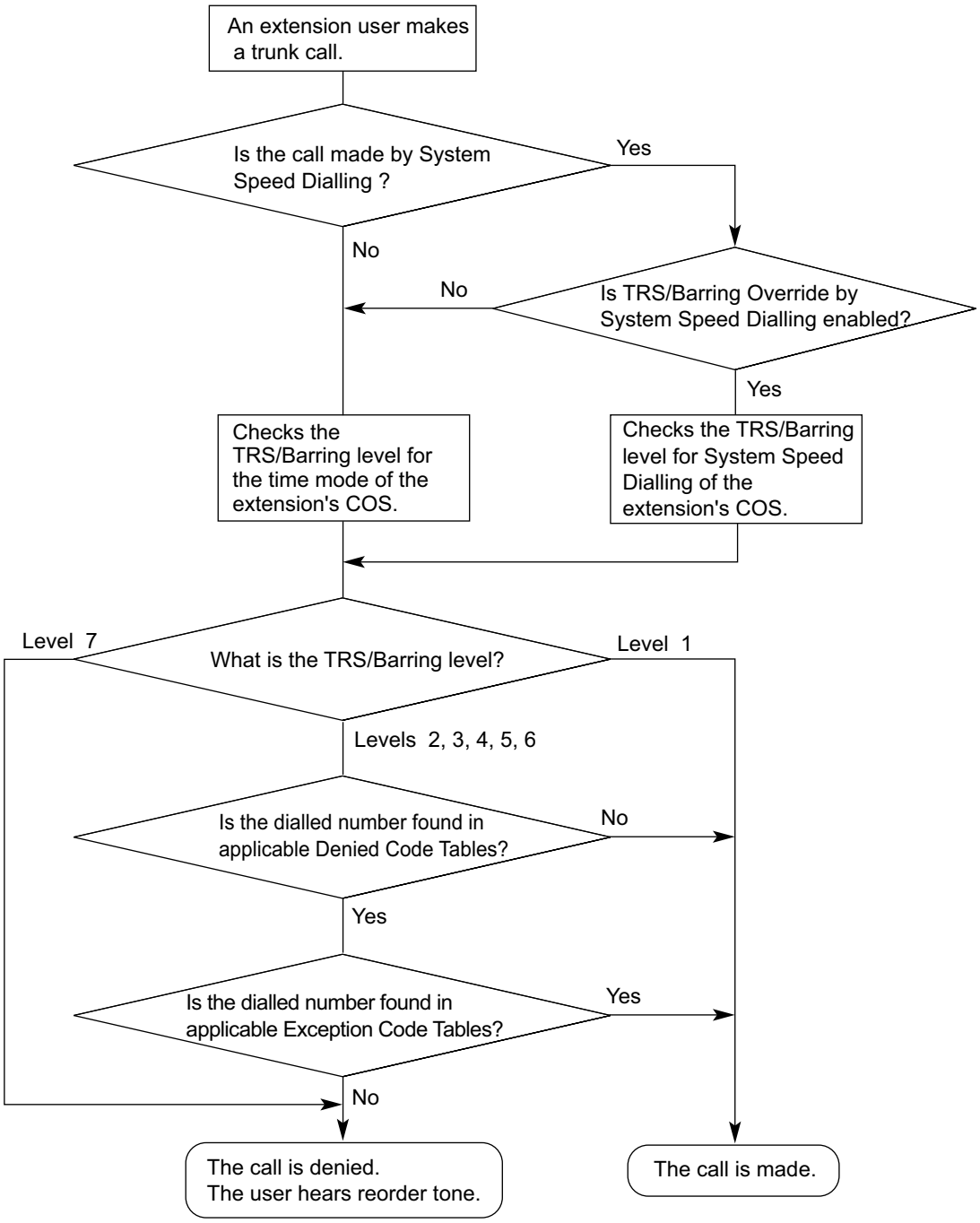
	Denied Code Tables		Exception Code Tables	
Level 1	Not Programmable		Not Programmable	
Level 2	00	Leading number to deny international calls	00xx	Leading number for countries to be allowed
Level 3	090	Leading number to deny cellular phone calls	090xxxxxxx	Boss's cellular phone number
Level 4	0	Leading number to deny long distance calls	–	Not required
:	:		:	

[Programming Example: COS Settings]

COS No.	Level for Time Mode[*1]				Level for System Speed Dialling[*2]
	Day	Lunch	Break	Night	
1	1	1	1	6	1
2	2	2	2	6	1
:	:	:	:	:	:

[*1] → 10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—TRS—◆ TRS Level—Day, Lunch, Break, Night
[*2] → 15.5 PBX Configuration—[7-5] TRS—Miscellaneous—◆ TRS Override by System Speed Dialling

[Flowchart]



TRS/Barring Settings for Each Level

Through system programming, it is possible to select a different method of TRS/Barring. With this method, each level has its own separate set of denied codes and exception codes, which are only applied to that level.
→ 15.5 PBX Configuration—[7-5] TRS—Miscellaneous—◆ TRS Table Mode for Level N (N=2_6)

	Denied Code Tables[*1]	**Exception Code Tables**[*2]
Level 1	Not Programmable	Not Programmable
Level 2	Table for Level 2	Table for Level 2

	Denied Code Tables*1	Exception Code Tables*2
Level 3	Table for Level 3	Table for Level 3
Level 4	Table for Level 4	Table for Level 4
Level 5	Table for Level 5	Table for Level 5
Level 6	Table for Level 6	Table for Level 6
Level 7	Not Programmable	Not Programmable

*1 → 15.1 PBX Configuration—[7-1] TRS—Denied Code—◆ Level 2–Level 6
*2 → 15.2 PBX Configuration—[7-2] TRS—Exception Code—◆ Level 2–Level 6

[Usage Example] Using this method, it is possible to restrict certain outgoing trunk calls (e.g., international/ cellular phone/long distance) on a department basis, as follows:

	Restricted	Allowed
Level 1	No restriction	
Level 2 (Engineering)	• International Calls	• Country where Factory is Located • Cellular Phone Calls • Long Distance Calls • Local Calls
Level 3 (Overseas Sales)	• Cellular Phone Calls	• Company Cellular Phone • International Calls • Long Distance Calls • Local Calls
Level 4 (Accounting)	• International Calls • Long Distance Calls	• Cities where Clients are Located • Cellular Phone Calls • Local Calls
:	:	:

In this example, a level 1 user can make any trunk calls. A level 2 user can only make international calls to the country where the factory is located, and can also make cellular phone/long distance/local calls. A level 3 user can only make cellular phone calls to the company cellular phone, and can also make any international/long distance/local calls. A level 4 user cannot make any international calls or most long distance calls, but can make long distance calls to cities where clients are located, cellular phone calls and local calls.
To set TRS/Barring as in the example above, it is necessary to programme the Denied Code and Exception Code Tables as follows:

	Denied Code Tables		Exception Code Tables	
Level 1	Not Programmable		Not Programmable	
Level 2	00	Leading number to deny international calls	00xx	Leading number for country to be allowed
Level 3	090	Leading number to deny cellular phone calls	090xxxx	Leading number for cellular phones to be allowed

	Denied Code Tables		Exception Code Tables	
Level 4	0	Leading number to deny both international and long distance calls	03 06 090	Long distance numbers for cities to be allowed, and leading number of cellular phones
:	:		:	

Conditions

CAUTION

The software contained in the TRS/Barring feature to allow user access to the network must be upgraded to recognise 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 recognise 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.

- A COS should be assigned for each extension.
  → 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ COS
  → 12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main—◆ COS
- TRS/Barring checks are applied to the following:
  – ARS
  – Trunk Access (Idle Line/Trunk Group/S-CO Line)
- It is programmable whether "⁎" or "#" is checked by the TRS/Barring. This is useful in preventing unauthorised calls which could be possible through certain telephone company exchanges.
  → 15.5 PBX Configuration—[7-5] TRS—Miscellaneous—◆ TRS Check for Dial " * # "
- It is programmable whether TRS/Barring checks the digits dialled after the External Feature Access during a trunk call. (→ 2.11.7 External Feature Access (EFA))
  → 15.5 PBX Configuration—[7-5] TRS—Miscellaneous—◆ TRS Check after EFA
- **Host PBX Access Code/Special Carrier Access Code**
  TRS/Barring checks for numbers dialled with a Host PBX Access code (→ 2.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)) or a Special Carrier Access code (→ 2.5.4.9 Special Carrier Access Code) in the following cases:

Type	Stored		Not stored
	Found	Not found	
Host PBX Access Code	Deletes the code. A TRS/Barring check is carried out on the following digits.	The call is made (excepted from TRS/Barring).	TRS/Barring checks the whole number.
Special Carrier Access Code	Deletes the code. A TRS/Barring check is carried out on the following digits.	TRS/Barring checks the whole number.	TRS/Barring checks the whole number.

- **ARS**
  If ARS is applied to a dialled number, TRS/Barring will check the user-dialled number (not the modified number by ARS). In this case, a Host PBX Access code and/or a Special Carrier Access code will not be checked.

- **Dialling Digit Restriction during Conversation**
  The dialling of digits can be restricted while engaged on a received trunk call. If the number of dialled digits exceeds the preprogrammed limitation, the line will be disconnected.
  → 15.5 PBX Configuration—[7-5] TRS—Miscellaneous—◆ Dial Digits Limitation After Answering—Dial Digits
- It is possible to select through system programming whether the trunk is disconnected when the Inter-digit time expires without the TRS/Barring check being completed.
  → 15.5 PBX Configuration—[7-5] TRS—Miscellaneous—◆ Mode when Dial Time-out before TRS Check
  – If no disconnection is chosen, the TRS/Barring check will also be performed after the Inter-digit time expires.
  – If disconnection is chosen, the line will be disconnected when the trunk Inter-digit time expires. This also prevents EFA from being used.

  This setting applies to all trunks.
- A TRS/Barring level can be changed by some features. The priority of features, when multiple features are used, is as follows:
  1. Dial Tone Transfer (→ 2.7.4 Dial Tone Transfer)
  2. Budget Management (→ 2.7.2 Budget Management)
  3. TRS/Barring Override by System Speed Dialling
  4. Walking COS/Verification Code Entry
     (→ 2.7.5 Walking COS, 2.7.6 Verification Code Entry)
  5. Extension Dial Lock

PC Programming Manual References

10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—TRS
→◆ TRS Level—Day, Lunch, Break, Night
→◆ TRS Level for System Speed Dialling
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ COS
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Parameter Selection (for TRS Level Change)
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main—◆ COS
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Parameter Selection (for TRS Level Change)
15.1 PBX Configuration—[7-1] TRS—Denied Code
15.2 PBX Configuration—[7-2] TRS—Exception Code
15.3 PBX Configuration—[7-3] TRS—Special Carrier
15.5 PBX Configuration—[7-5] TRS—Miscellaneous

PT Programming Manual References

[300] TRS/Barring Override by System Speed Dialling
[301] TRS/Barring Denied Code
[302] TRS/Barring Exception Code
[501] TRS/Barring Level
[509] TRS/Barring Level for System Speed Dialling
[602] Class of Service

Feature Guide References

2.5.5.3 Trunk Access

2.6.4 Speed Dialling—Personal/System
2.7.3 Extension Dial Lock
2.8.1 Automatic Route Selection (ARS)
5.1.1 Class of Service (COS)
5.1.4 Time Service
6.1 Capacity of System Resources

2.7.2 Budget Management

Description

Limits the telephone usage to a preprogrammed budget on an extension basis. If the amount of the call charge reaches the limit, an extension user cannot make further trunk calls. An extension assigned as the manager may increase the limit or clear the amount of the call charge.

Conditions

- If the limit is reached, TRS/Barring Level 7 is applied. (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring))
- **Budget Management for Verified Call**
  If an extension user makes a trunk call with a verification code, the call charge will be added to the total for the verification code (not the extension). (→ 2.7.6 Verification Code Entry) Each verification code can be assigned a call charge limit.
- **Budget Management for Walking COS**
  If an extension user makes a trunk call from an extension using the Walking COS feature, the call charge will be added to the extension of the extension user (not the extension that the call was made on). (2.7.5 Walking COS)
- Pay tone service or ISDN Advice of Charge (AOC) service is required for this feature.
- It is possible to select whether to disconnect the line (disconnect mode) after a warning tone or only to send a warning tone when the amount of the call charge reaches the preprogrammed limit during a conversation.
- When multiple extension users are using the same verification code or the same extension (through the use of Walking COS) simultaneously, each caller can have access to the total remaining budget of the extension or verification code.

PC Programming Manual References

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 3—◆ Charge Limit
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 3—◆ Charge Limit
14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Charge—◆ Charge Options—Action at Charge Limit
14.3 PBX Configuration—[6-3] Feature—Verification Code—◆ Budget Management

User Manual References

4.1.2 Manager Programming

2.7.3 Extension Dial Lock

Description

An extension user can change the TRS/Barring level of the telephone (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring)) so that other users cannot make inappropriate trunk calls. An extension personal identification number (PIN) is used to unlock the telephone (→ 2.24.1 Extension Personal Identification Number (PIN)). This feature is also known as Electronic Station Lockout.

Conditions

- This feature also restricts changing the FWD destination. (→ 2.3.2 Call Forwarding (FWD))
- **Remote Extension Dial Lock**
  Overrides Extension Dial Lock. If an extension assigned as the manager sets Remote Extension Dial Lock on an extension that has already been locked by the extension user, the user cannot unlock it. If a manager extension unlocks an extension that has been locked by the extension user, the extension will be unlocked. This feature is also known as Remote Station Lock Control.
- **TRS/Barring Level**
  COS programming determines the TRS/Barring level for Extension Dial Lock.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ Extension Dial Lock Set / Cancel
→◆ Remote Extension Dial Lock Off
→◆ Remote Extension Dial Lock On
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—TRS—◆ TRS Level on Extension Lock

PT Programming Manual References

[510] TRS/Barring Level for Extension Dial Lock

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

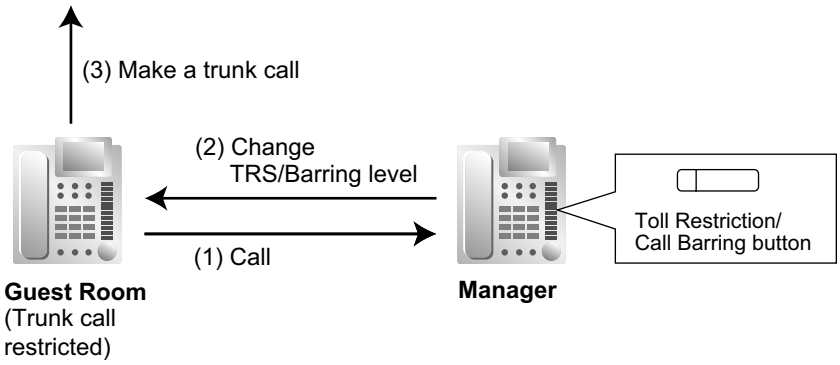
1.6.3 Preventing Other People from Using Your Telephone (Extension Dial Lock)
2.1.1 Extension Control

2.7.4 Dial Tone Transfer

Description

An extension assigned as the manager can change the TRS/Barring level (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring)) for an extension user temporarily. After that, the extension user can make his call.

[Example] An extension user can call a manager to release the restriction on outgoing calls (e.g., international calls).



Conditions

- The modified TRS/Barring level only applies to the next one call placed at the user's extension.
- **Toll Restriction/Call Barring Button**
  A manager extension must store the desired TRS/Barring level in the Toll Restriction/Call Barring button. A flexible button can be customised as the Toll Restriction/Call Barring button.

PC Programming Manual References

12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Parameter Selection (for TRS Level Change)
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Parameter Selection (for TRS Level Change)

Feature Guide References

2.21.2 Flexible Buttons

User Manual References

2.1.3 Restriction Level Control (Dial Tone Transfer)

2.7.5 Walking COS

Description

A user can enter his extension number and extension personal identification number (PIN) (→ 2.24.1 Extension Personal Identification Number (PIN)) at another extension, to make the following types of call using his Class of Service, including TRS level, overriding the other extension's Class of Service.

- Trunk call
- TIE line call
- Intercom call
- External Relay Control (→ 2.18.4 External Relay Control)

After performing Walking COS, the following features are also available for the specified extension:

- Call Forwarding (FWD)/Do Not Disturb (DND) setting (→ 2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features)
- Incoming Call Distribution Group Log-in/Log-out (→ 2.2.2.7 Log-in/Log-out)
- Absent Message setting (→ 2.20.2 Absent Message)
- Extension Dial Lock (→ 2.7.3 Extension Dial Lock)
- Time Service—Changing the Time Mode (day/lunch/break/night) (→ 5.1.4 Time Service)
- CLIP number setting (CLIP ID) (→ 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP))

Conditions

- When a trunk call is made using Walking COS:
  - the Class of Service of the specified extension is applied (→ 5.1.1 Class of Service (COS))
  - the budget of the specified extension is applied (→ 2.7.2 Budget Management)
  - the Itemised Billing code of the specified extension is applied (→ 2.8.1 Automatic Route Selection (ARS))
  - the specified extension number is recorded on SMDR as the call originator, instead of the extension number of the actual extension used (→ 2.22.1.1 Station Message Detail Recording (SMDR)).
- Walking COS is also available through DISA. (→ 2.16.1 Direct Inward System Access (DISA))
- **Extension PIN**
  An extension personal identification number (PIN) is required to use this feature. (→ 2.24.1 Extension Personal Identification Number (PIN)) If the wrong PIN is entered three times, the line will be disconnected.
- This feature cannot be used for extensions which the extension being operated is prevented from calling by Internal Call Block. (→ 2.1.2.2 Internal Call Block)

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ User Remote Operation / Walking COS / Verification Code
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Optional Device & Other Extensions—◆ Remote Operation by Other Extension
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ Extension PIN
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main—◆ Extension PIN

PT Programming Manual References

[005] Extension Personal Identification Number (PIN)

Feature Guide References

2.16.1 Direct Inward System Access (DISA)
5.1.1 Class of Service (COS)

User Manual References

1.2.7 Calling without Restrictions
1.2.9 Setting Your Telephone from Another Extension or through DISA (Remote Setting)

2.7.6 Verification Code Entry

Description

An extension user can enter a verification code when calling from his own or any other extension, to change the TRS/Barring level (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring)) or to identify the call for accounting and billing purposes. A verification code personal identification number (PIN) is required to use this feature.

Conditions

CAUTION

There is a risk that fraudulent telephone calls will be made if a third party discovers a personal identification number (PIN) (verification code PIN 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.

- When a trunk call is made using Verification Code Entry:
  – the Class of Service of the specified extension is applied (→ 5.1.1 Class of Service (COS))
  – the budget of the specified extension is applied (→ 2.7.2 Budget Management)
  – the Itemised Billing code of the specified extension is applied (→ 2.8.1 Automatic Route Selection (ARS))
  – ⚹ + verification code is recorded on SMDR as the call originator, instead of the extension number of the actual extension used (→ 2.22.1.1 Station Message Detail Recording (SMDR)).
- **Verification Code Entry through DISA**
  This feature is also available through DISA. (→ 2.16.1 Direct Inward System Access (DISA))
- **Verification Code PIN**
  A verification code PIN must be assigned for each verification code through system programming or through manager programming.
- **Verification Code PIN Lock**
  If the wrong PIN is entered three times, the line will be disconnected. If the wrong PIN is entered a preprogrammed number of times successively, the PIN for the verification code will be locked. Only an extension assigned as the manager can unlock it. In this case, the PIN will be unlocked and cleared.
- **Budget Management for Verified Calls**
  A limit can be assigned to the total of all call charges for each verification code.

[Example of Verification Codes and Their Programming]

Location	Code*1	Name*2	PIN*3	COS*4	Itemised Billing Code for ARS*5	Budget*6
0001	1111	Tom Smith	1234	1	2323	5000Euro
0002	2222	John White	9876543210	3	4545	3000Euro

Location	Code*1	Name*2	PIN*3	COS*4	Itemised Billing Code for ARS*5	Budget*6
:	:	:	:	:	:	:

*1 → 14.3 PBX Configuration—[6-3] Feature—Verification Code—◆ Verification Code
*2 → 14.3 PBX Configuration—[6-3] Feature—Verification Code—◆ User Name
*3 → 14.3 PBX Configuration—[6-3] Feature—Verification Code—◆ Verification Code PIN
*4 → 14.3 PBX Configuration—[6-3] Feature—Verification Code—◆ COS Number
*5 → 14.3 PBX Configuration—[6-3] Feature—Verification Code—◆ Itemised Billing Code for ARS
*6 → 14.3 PBX Configuration—[6-3] Feature—Verification Code—◆ Budget Management

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Extension PIN—Lock Counter
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ User Remote Operation / Walking COS / Verification Code
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Option 1—◆ ARS Itemised Code
→Option 3—◆ Charge Limit
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings
→Option 1—◆ ARS Itemised Code
→Option 3—◆ Charge Limit
14.3 PBX Configuration—[6-3] Feature—Verification Code

PT Programming Manual References

[120] Verification Code
[121] Verification Code Name
[122] Verification Code Personal Identification Number (PIN)
[123] Verification Code COS Number

Feature Guide References

2.7.2 Budget Management
2.8.1 Automatic Route Selection (ARS)
5.1.1 Class of Service (COS)
5.1.6 Manager Features
6.1 Capacity of System Resources

User Manual References

1.2.7 Calling without Restrictions
4.1.2 Manager Programming

2.8 Automatic Route Selection (ARS) Features

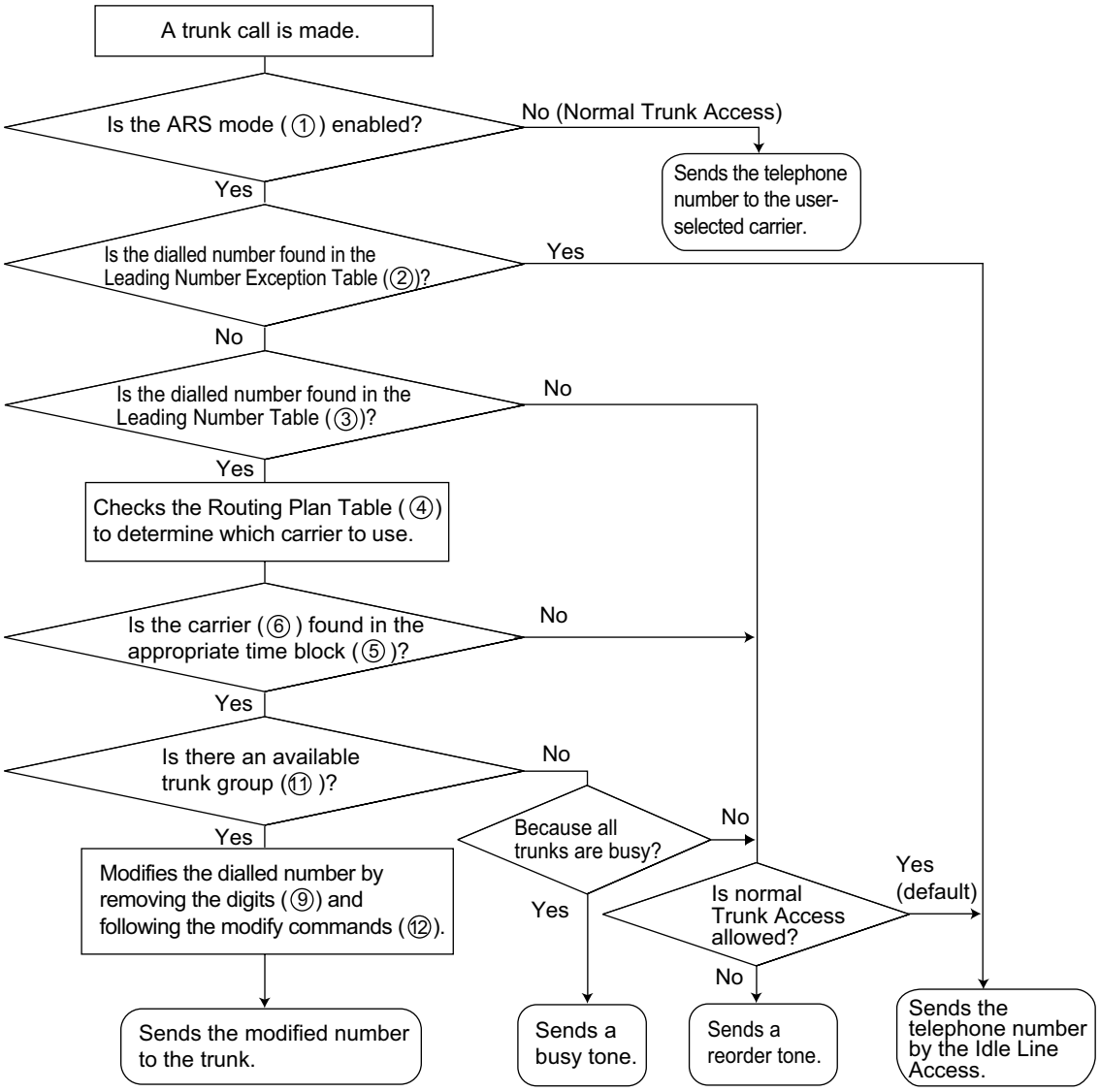
2.8.1 Automatic Route Selection (ARS)

Description

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

[Carrier Selection Procedure Flowchart]

The numbers ⓧ in the flowchart correspond to the [Programming Procedures] on the following pages.



[Programming Procedures]

1. ARS Mode ① Assignment

It is possible to select whether ARS operates when an extension user makes a call using any Idle Line Access method or when an extension user makes a call using any Trunk Access method. (→ 2.5.5.3 Trunk Access)
→ 16.1 PBX Configuration—[8-1] ARS—System Setting—◆ ARS Mode

2. **Leading Number Exception Table ② Assignment**
Store the telephone numbers that will avoid using the ARS feature.
→ 16.6 PBX Configuration—[8-6] ARS—Leading Number Exception

② ARS Leading Number Exception Table

Location No.	Leading No. Exception
001	033555
002	06456
:	:

3. **Leading Number Table ③ Assignment**
Store the area codes and/or telephone numbers as leading number that will be routed by the ARS feature. In this table, the Routing Plan (refer to "4. Routing Plan Table ④ Assignment") is selected for each number. The additional (remain) number of digits must be assigned only when "#", for example, is needed after a dialled number. The "#" is added after the assigned number of digits of dialled number (excluding a leading number).
→ 16.2 PBX Configuration—[8-2] ARS—Leading Number—◆ Leading Number
→ 16.2 PBX Configuration—[8-2] ARS—Leading Number—◆ Additional Number of Digits
→ 16.2 PBX Configuration—[8-2] ARS—Leading Number—◆ Routing Plan Number

③ ARS Leading Number Table

Location No.	Leading No.	Additional (Remain) No. of Digits	Routing Plan Table No.
0001	039	7	1
0002	03	0	4
0003	0444	5	5
:	:	:	:

If a dialled number matches a leading number, the number will be modified according to the corresponding Routing Plan Table and the modified number will be sent to the trunk when the assigned additional (remain) number of digits are dialled.
If a dialled number matches multiple leading number entries, the leading number entry with the lowest numbered location will have priority.

[Example]

Dialled Number	Corresponding Routing Plan Table No.	Description
039-123-4567	1	"039" is found in location 0001 and seven digits (assigned additional [remain] number of digits in location 0001) were dialled. The Routing Plan Table 1 is selected just after the seventh digit.

Dialled Number	Corresponding Routing Plan Table No.	Description
039-654-321	1	"039" is found in location 0001 and the Inter-digit time expired before the seventh digit is received. The Routing Plan Table 1 is selected after the Inter-digit time expired. → 10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Dial—Extension Inter-digit (s)
038	4	"03" is found in two locations (locations 0001 and 0002), so the PBX waits for the next digit "8". "038" is not found in any location, then "03" (location 0002) is selected. The Routing Plan Table 4 is selected.

4. Routing Plan Table ④ Assignment

Arrange the time schedule as desired and store the carrier priority.

Time Table ⑤

As the best carrier may vary with the day of the week and the time of day, four time blocks (Time-A through D) can be programmed for each day of the week.
→ 16.3.1 PBX Configuration—[8-3] ARS—Routing Plan Time—Time Setting

Carrier Priority ⑥

Assign the appropriate carrier (refer to "5. Carrier Table ⑦ Assignment") and their priority in each time block. The carrier is selected in the entry order (the order in which entries are listed).
→ 16.4 PBX Configuration—[8-4] ARS—Routing Plan Priority

③ ARS Leading Number Table

Location No.	Leading No.	Additional (Remain) No. of Digits	Routing Plan Table No.
0001	03	8	1
:	:	:	:

④ ARS Routing Plan Table

Routing Plan Table 1					
⑤ Time Table			⑥ Carrier		
			Priority 1	Priority 2	...
SUN	Time-A	9:00	1 (A telecom)	4 (D telecom)	···
	Time-B	12:00	1 (A telecom)	2 (B telecom)	···
	Time-C	15:00	1 (A telecom)	2 (B telecom)	···
	Time-D	21:00	3 (C telecom)	1 (A telecom)	···
:	:	:	:	:	···
SAT	Time-A	9:00	3 (C telecom)	2 (B telecom)	···
	Time-B	12:00	3 (C telecom)	1 (A telecom)	···
	Time-C	15:00	3 (C telecom)	1 (A telecom)	···
	Time-D	21:00	3 (C telecom)	2 (B telecom)	···

5. Carrier Table ⑦ Assignment

A specified number of carriers can be programmed. Assign the following items for each Carrier Table:
→ 16.5 PBX Configuration—[8-5] ARS—Carrier—Carrier
Carrier Name ⑧: Assign the carrier name.
→ 16.5 PBX Configuration—[8-5] ARS—Carrier—Carrier—◆ Carrier Name
Removed Number of Digits ⑨: Assign the number of digits to remove from the beginning of the user-dialled number.
→ 16.5 PBX Configuration—[8-5] ARS—Carrier—Carrier—◆ Removed Number of Digits

Carrier Access Code ⑩: Assign the code to access the carrier.
→ 16.5 PBX Configuration—[8-5] ARS—Carrier—Carrier—◆ Carrier Access Code
Trunk Group ⑪: Assign the trunk groups which connect to each carrier and the priority in which they are selected.
When using Web Maintenance Console, trunk groups can be assigned to a carrier using an on/off setting for each trunk group. They can also be given a priority setting (1–4) which decides the order they are searched when seizing a line. If there are no available lines in the trunk groups set to priority 1–4, the other trunk groups set to on are searched in the order of smallest number first.

[Example]

Carrier	Priority Setting				Trunk Groups Set to On	Searching Order
	1	2	3	4		
ABC	9	3	1	7	5, 7, 9, 11	9 → 3 → 1 → 7 → 5 → 11
XYZ	12	4	None	None	6, 10	12 → 4 → 6 → 10

→ 16.5 PBX Configuration—[8-5] ARS—Carrier—TRG Priority
→ 16.5 PBX Configuration—[8-5] ARS—Carrier—TRG 01–TRG 64
Modify Command ⑫: Assign the commands to modify the dialled number to access the carrier.
→ 16.5 PBX Configuration—[8-5] ARS—Carrier—Carrier—◆ Modify Command
CLIP Table No. ⑯: Assign the CLIP number for the carrier. CLIP numbers are assigned according to the CLIP Table No. assigned for the carrier.
→ 16.5 PBX Configuration—[8-5] ARS—Carrier—Carrier—◆ CLIP Table No.

[Command Explanation]

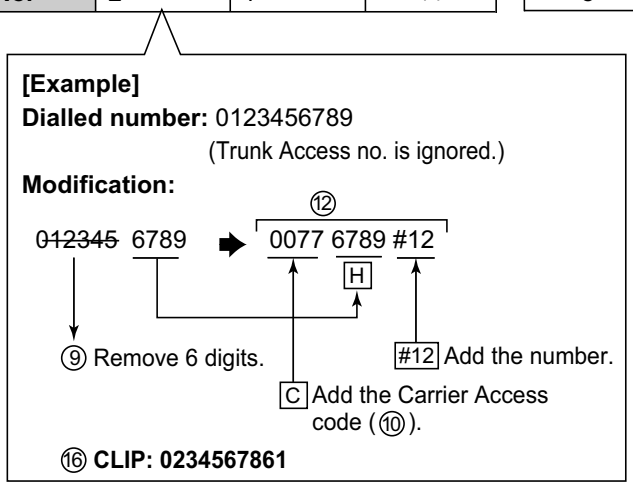
Command	Description
Number	Add the number.
C	Add the Carrier Access code.
P	**Analogue Line:** Insert a pause. **ISDN/E1 Line:** Insert a pause and change to tone (DTMF) signal.
A	Add the Authorisation code for a tenant (⑬).
G	Add the Authorisation code for a trunk group (⑭).
I	Add the Itemised Billing code (⑮).
H	Add the dialled number after the digits are removed (Home position).

[Programming Example]

⑦ Carrier Table	1	2	• •
⑧ Carrier Name	A telecom	B telecom	• •
⑨ Removed Number of Digits	6	0	• •
⑩ Carrier Access Code	0077	0088	• •
⑪ Trunk Group	1, 2, 3	1, 2	• •
⑫ Modify Command	CH#12	CH	• •
⑯ CLIP Table No.	2	1	• •

Ext. 1001

CLIP No.	CLIP
1	0123456789
2	0234567861
3	0356894526
⋮	⋮
8	0856325889



Note

- If the ARS Itemised Code is set to be sent as a CLIP with ARS, the following settings are prioritised and used as the CLIP.
  12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1—◆ ARS Itemised Code
  12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 1—◆ ARS Itemised Code
- CLIP Table No.1 is set automatically according to the following settings.
  12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—CLIP—◆ CLIP ID
  12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—CLIP—◆ CLIP ID

6. Optional Assignment

Authorisation Code for a Tenant ⑬

An Authorisation code can be assigned for each carrier and each tenant.
→ 16.5 PBX Configuration—[8-5] ARS—Carrier—Authorisation Code for Tenant

Authorisation Code for a Trunk Group ⑭

An Authorisation code can be assigned for each trunk group and each carrier.
→ 16.7 PBX Configuration—[8-7] ARS—Authorisation Code for TRG

Itemised Billing Code ⑮

An Itemised Billing code can be assigned for each extension and for each verification code.
If a call is not made from an extension (e.g., DISA or TIE) and no verification code is used, the Itemised Billing code assigned in the location 1 of the verification code will be used.
→ 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1—◆ ARS Itemised Code

→ 12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 1—◆ ARS Itemised Code
→ 14.3 PBX Configuration—[6-3] Feature—Verification Code—◆ Itemised Billing Code for ARS

Conditions

CAUTION

The software contained in the ARS feature to allow user access to the network must be upgraded to recognise 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 recognise 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.

- **Dialled Number on SMDR**
  It is possible to choose to print either the user-dialled number or the modified number on SMDR through system programming. (→ 2.22.1.1 Station Message Detail Recording (SMDR))
  → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—ARS Dial
- **ARS Data Download/Upload**
  It is possible to download or upload the following ARS data to the PBX using PC programming:
  – ② ARS Leading Number Exception Table
  – ③ ARS Leading Number Table
  – ④ ARS Routing Plan Table
    → 6.5 Tool—Import
    → 6.6 Tool—Export

  This is useful when a carrier has changed the call charge, and the updated data can be used for multiple customers.
- A TRS/Barring check is done before ARS is applied. (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring))

PC Programming Manual References

6.5 Tool—Import
→ARS - Leading Digit
→ARS - Except Code
→ARS - Routing Plan
6.6 Tool—Export
10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Dial—Extension Inter-digit (s)
10.9 PBX Configuration—[2-9] System—System Options—Option 3—◆ Dial Tone—Dial Tone for ARS
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1—◆ ARS Itemised Code
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 1—◆ ARS Itemised Code
14.3 PBX Configuration—[6-3] Feature—Verification Code—◆ Itemised Billing Code for ARS
Section 16 PBX Configuration—[8] ARS
19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—ARS Dial

PT Programming Manual References

[320] ARS Mode
[321] ARS Leading Number

[322] ARS Routing Plan Table Number
[325] ARS Exception Number
[330] ARS Routing Plan Time Table
[331–346] ARS Routing Plan Table (1–16)
[347] ARS Routing Plan Table (1–48)
[350] ARS Carrier Name
[351] ARS Trunk Group for Carrier Access
[352] ARS Removed Number of Digits for Carrier Access
[353] ARS Carrier Access Code

Feature Guide References

6.1 Capacity of System Resources

2.9 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Features

2.9.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension

Description

Primary Directory Number (PDN) buttons and Secondary Directory Number (SDN) buttons are ideal for use between bosses and secretaries. When a call (intercom or trunk) arrives at a PDN button on the boss's extension, the call will ring and the LED of the SDN button will flash at the secretary's extension as well, indicating that an incoming call is arriving at the boss's extension. In addition, caller information (e.g., Caller ID) of the incoming call will appear on the secretary's extension. The secretary can answer the call for the boss by simply pressing the SDN button. Delayed ringing can be set for a PDN or SDN button.
A secretary can hold a call answered on the SDN button, and the boss can retrieve the held call simply by pressing the PDN button, like when answering a call with an S-CO button. In addition, a secretary can transfer calls from an SDN button or other button (e.g., S-CO button) to the boss's extension with a simple operation, like when using a DSS button.
An extension can have several SDN buttons, each registered to a different boss's extension. However, only one SDN button can be registered for a single boss at each extension. An extension can have up to eight PDN buttons. PDN buttons can simplify the use of an extension because both intercom and trunk calls can be made and received at a PDN button.

Making Calls with an SDN Button

When Standard SDN Key mode is assigned to an SDN extension (secretary) through COS programming, SDN extensions (secretaries) can make calls for PDN extensions (bosses) on the SDN button. For example, a boss can ask a secretary to make a call and put the call on hold, after which, the boss can retrieve the held call. Through COS programming, it is possible to allow an SDN extension to make calls using the COS of the PDN extension. All other settings that are available when using the Walking COS feature are also applied (→ 2.7.5 Walking COS).

SDN Direct Dial

An SDN extension can call a PDN extension or transfer a call to a PDN extension using an SDN button. In this case:

– Only the PDN extension rings (i.e., other SDN extensions do not ring).
– The delayed ringing and DND settings of the PDN extension are ignored.

Depending on the mode selected through COS programming, SDN Direct Dial is performed in one of two ways, as follows:

– Enhanced DSS Key mode: pressing the SDN button once.
– Standard SDN Key mode: pressing the SDN button twice (a dial tone is heard the first time the SDN button is pressed).

Calls answered using the SDN button can be transferred to the PDN extension by simply pressing the SDN button once, regardless of the mode.

LED Indication

The LED patterns and the corresponding status of PDN and SDN buttons are as follows:

Light Pattern	PDN Button Status	SDN Button Status
Off	This extension is idle.	The corresponding PDN extension is idle.

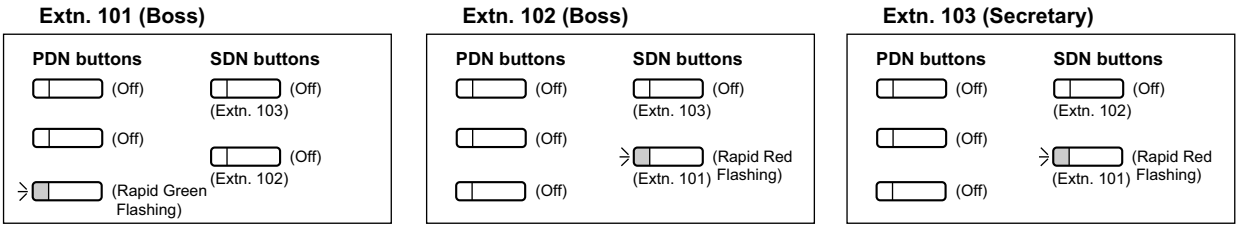
Light Pattern	PDN Button Status	SDN Button Status
Green on	The extension is on a call using the PDN button.	The extension is on a call using the SDN button.
Slow green flashing	A call is on hold using the PDN button.	A call is on hold using the SDN button.
Moderate green flashing	• A call on a PDN button is on Exclusive Call Hold or consultation hold. • The PDN extension is adding a member to a conference or using the line for an Unattended Conference, on a PDN button.	• A call answered using the SDN button is on Exclusive Call Hold or consultation hold. • The SDN extension is adding a member to a conference, or using the line for an Unattended Conference.
Rapid green flashing	An incoming call is arriving at this extension.	Receiving Hold Recall or automatic callback ringing from a call answered using the SDN button.
Red on	A corresponding SDN extension is: • on a call. • holding the line using Exclusive Call Hold or consultation hold. • adding a member to a conference. • using the line for an Unattended Conference. • receiving Hold Recall or automatic callback ringing.	The corresponding PDN extension or another corresponding SDN extension is: • on a call. • holding the line using Exclusive Call Hold or consultation hold. • adding a member to a conference. • using the line for an Unattended Conference. • receiving an incoming call directed only to the PDN extension (e.g., callback ringing).
Slow red flashing	A call is on hold by a corresponding SDN extension.	A call is on hold by the corresponding PDN extension or another corresponding SDN extension.
Rapid red flashing	A call is arriving at an Incoming Call Distribution (ICD) group in Ring Distribution method that this extension is a member of.	The corresponding PDN extension is receiving an incoming call.

When multiple calls are on a PDN extension, the LED pattern that appears on the corresponding SDN buttons is displayed according to the following priority:
Receiving an incoming call → holding a call → on a call → idle
For example, if a PDN extension receives an incoming call while on a call, the LEDs on the corresponding SDN extensions will show the incoming call.
However, if an SDN extension is handling a call using the SDN button (e.g., on a call, has a call on hold, etc.), the status of that call will be displayed on the SDN button, regardless of the call status of the PDN extension.

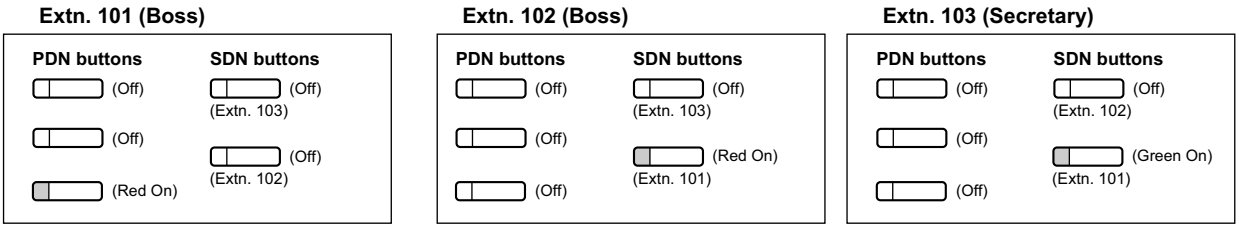
Example of a Secretary Handling Calls for Multiple Bosses

The following example shows the LED patterns of the PDN and SDN buttons of each extension and how calls can be handled.

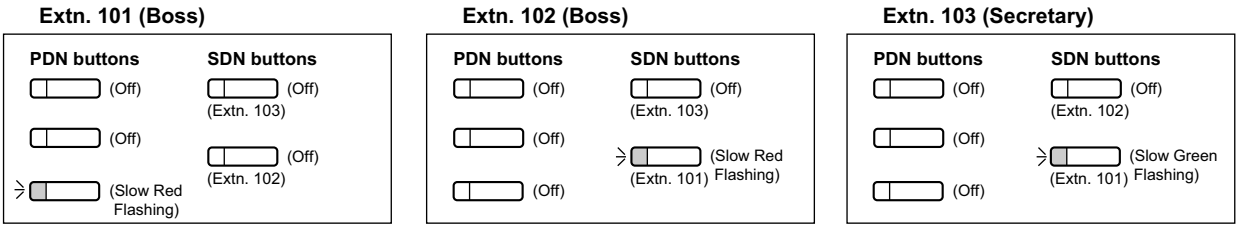
1. A Call From 111-1111 Arrives at Ext. 101



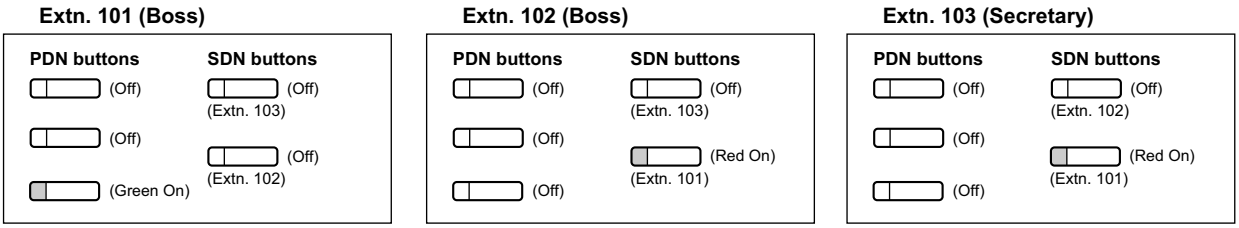
2. The Call From 111-1111 is Answered by Ext. 103



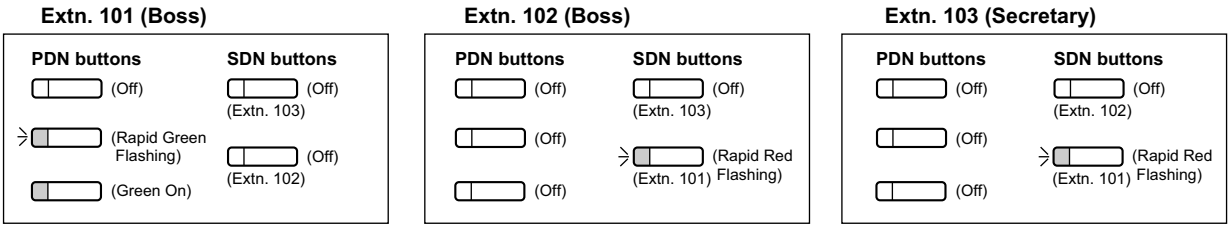
3. The Call From 111-1111 is On Hold by Ext. 103



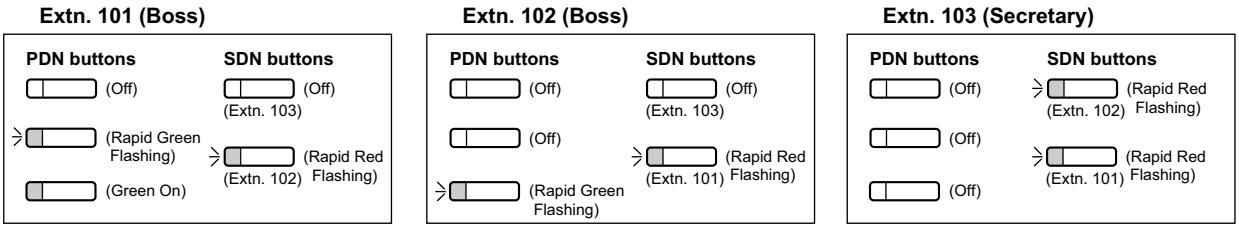
4. The Call Held by Ext. 103 is Answered by Ext. 101



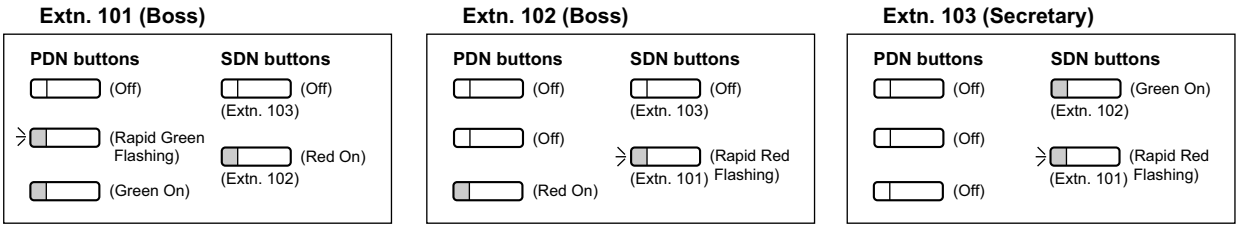
5. A Call From 222-2222 Arrives at Ext. 101



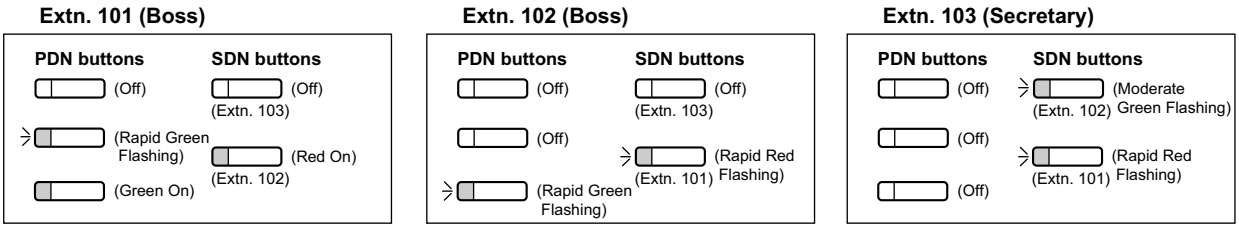
6. A Call From 333-3333 Arrives at Ext. 102



7. The Call From 333-3333 is Answered by Ext. 103



8. The Call From 333-3333 is Transferred by Ext. 103 to Extension 102



Conditions

[General]

- A flexible button of a PT and a PS can be customised as a PDN or SDN button. A flexible button on a DSS Console can be customised as an SDN button.
- An extension can have up to eight PDN buttons.
- If none of an extension's PDN buttons are idle, the extension will not receive incoming calls, including Call Waiting. Therefore, it is strongly recommended for PDN extensions to have at least three PDN buttons.
- Through COS programming, it is possible to select whether extensions can create SDN buttons on their own extensions using PT programming.
- Up to eight different extensions can assign SDN buttons corresponding to the same PDN extension.
- When a PDN extension has an idle CO button or ICD Group button, calls will arrive on the following buttons according to the following priority:
  - Incoming intercom calls to an ICD group: ICD Group button → PDN button
  - Incoming trunk calls: S-CO button → G-CO button → L-CO button → PDN button
  - Incoming trunk calls to an ICD group: ICD Group button → S-CO button → G-CO button → L-CO button → PDN button
- When multiple calls of the same status (e.g., on hold) are on a PDN extension, the status of the oldest call will be displayed on the corresponding SDN extensions. For example, if a PDN extension has two calls ringing, an SDN extension will answer the call that arrived at the PDN extension first, when pressing the SDN button.
- When a PDN extension is a member of an ICD group in Ring Distribution method, and an incoming call arrives at the ICD group, the incoming call status will not appear on the LEDs of the corresponding SDN extensions (→ 2.2.2.1 Incoming Call Distribution Group Features—SUMMARY).

- If none of an extension's PDN buttons are idle, DSS buttons of other extensions registered to the PDN extension will turn on red.
- **Ring Tone Pattern**
  Through system programming, each extension can set ring tone patterns for PDN buttons. Ring tone patterns can be assigned separately for each SDN button.
- **Outgoing Line Preference**
  When "PDN" is selected as the outgoing line preference, outgoing calls will originate on the first available PDN button (→ 2.5.5.2 Line Preference—Outgoing).
- **Incoming Line Preference**
  Through system programming, it is possible for only incoming calls arriving at PDN buttons to be answered simply by going off-hook, by selecting "PDN" as the incoming line preference (→ 2.4.2 Line Preference—Incoming). This prohibits calls that arrive on non-PDN buttons (e.g., an SDN button) to be answered when going off-hook.
- **Walking Extension**
  For PDN extensions, the Walking Extension feature can only be used when all PDN buttons are idle (→ 2.24.3 Walking Extension Features).
- **Wireless XDP Parallel Mode**
  If a PS has PDN or SDN buttons, Wireless XDP Parallel mode cannot be assigned to that PS (→ 5.2.4.5 Wireless XDP Parallel Mode).
- **One numbered extension**
  If an extension has PDN or SDN buttons, Sub extension of One numbered extension cannot be assigned to that extension (→ 2.11.11 One-numbered Extension).
- **OHCA/Whisper OHCA**
  A PDN extension cannot receive OHCA or Whisper OHCA unless the call is made using a corresponding SDN button (→ 2.10.4.3 Off-hook Call Announcement (OHCA), → 2.10.4.4 Whisper OHCA).
- **Alternate Calling—Ring/Voice**
  It is not possible to temporarily change the called party's preset call receiving method (ring tone or voice) when calling a PDN extension, unless the call is made using a corresponding SDN button (→ 2.5.3 Intercom Call).
- Through system programming, it is possible to force an extension to become idle (the SP-PHONE button light will turn off) when a speakerphone call using a PDN/SDN button is put on hold using CTI.

[Delayed Ringing]

- The same delayed ringing setting is applied to all PDN buttons on an extension. Delayed ringing can be assigned separately for each SDN button.
- Through system programming, it is possible to select whether caller information (such as Caller ID) is shown immediately on a PS when a call is received while delayed ringing is set.
- Caller information (such as Caller ID) is not shown immediately on a PT when a call is received while delayed ringing is set.
- SDN buttons can be set to not ring (only flash) for incoming calls. However, this setting is not available for PDN buttons.
- The forward no answer timer starts when a PDN extension starts ringing.

PC Programming Manual References

10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—PDN/SDN
10.8.3 PBX Configuration—[2-8-3] System—Ring Tone Patterns—Call from Others—◆ Extension—Ring Tone Pattern Plan 1–8
10.9 PBX Configuration—[2-9] System—System Options—Option 4—◆ System Wireless—SDN Delayed Ringing with LCD
10.9 PBX Configuration—[2-9] System—System Options—Option 6 (CTI)—◆ CTI Hold—Forced Idle when Hold by PDN/SDN Key

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1—◆ Wireless XDP / Shared Extension
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 9—◆ PDN Delayed Ringing
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Parameter Selection (for SDN)
→◆ Extension Number (for SDN)
→◆ Optional Parameter (Ringing Tone Type Number) (for Loop CO, Single CO, Group CO, ICD Group, SDN)
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 9—◆ PDN Delayed Ringing
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Parameter Selection (for SDN)
→◆ Extension Number (for SDN)
12.3 PBX Configuration—[4-3] Extension—DSS Console

Feature Guide References

2.2.2.2 Group Call Distribution
2.21.3 LED Indication
5.1.1 Class of Service (COS)
6.1 Capacity of System Resources

User Manual References

1.4.2 Holding a Call
1.5.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension
◆◆ Using Your Calling Privileges at Another Extension (Remote COS Access)

2.10 Busy Line/Busy Party Features

2.10.1 Automatic Callback Busy (Camp-on)

Description

If the destination or line is busy when a call is made, an extension user can set the Automatic Callback Busy feature. The PBX will monitor the status of the destination or trunk and, when it becomes available, will send a callback ringing to the calling extension to inform the user. After the extension answers the callback ringing, the previously dialled extension number is automatically redialled, or the trunk is automatically seized.

Conditions

- If the callback ringing is not answered within 10 seconds, the callback is cancelled.
- If the extension hears a busy tone before dialling the telephone number, only the trunk or trunk group is reserved. After answering the callback ringing, the extension should dial the telephone number.
- An extension can set only one Automatic Callback Busy. The last setting is effective.
- Multiple extension users can set this feature to one trunk simultaneously.
  However, a maximum of four extension users can set this feature to one extension.
  Callback ringing will be sent to extensions in the order that the feature was set. In other words, the extension that set the feature first will receive a callback ringing first.
- This feature cannot be used for calls to a VPS, the Unified Messaging system, or to an ISDN extension.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Automatic Callback Busy Cancel
10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature—◆ Automatic Callback Busy

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer

2.10.2 Executive Busy Override

Description

Allows an extension user to interrupt an existing call to establish a three-party conference call.

Executive Busy Override Deny:
It is possible for extension users to prevent their calls from being intercepted by another extension user.

One-touch Executive Busy Override:
Extension users can perform Executive Busy Override by simply pushing the S-CO button of a call in progress without entering a feature number. This feature can be enabled through system programming.

Conditions

[General]

- COS programming determines the extension users who can use Executive Busy Override and set Executive Busy Override Deny.
- This feature does not work when the busy extension is in one of the following conditions:
  - **a.** Executive Busy Override Deny or Data Line Security (→ 2.11.5 Data Line Security) has been set.
  - **b.** While being monitored by another extension (→ 2.10.3 Call Monitor).
  - **c.** While receiving OHCA (→ 2.10.4.3 Off-hook Call Announcement (OHCA), → 2.10.4.4 Whisper OHCA).
  - **d.** During a Conference call (→ 2.14 Conference Features).
  - **e.** During a doorphone call (→ 2.18.1 Doorphone Call).
  - **f.** While Live Call Screening (LCS) or Two-way Record is activated (→ 3.2.2.16 Live Call Screening (LCS) and 3.2.2.30 Two-way Record/Two-way Transfer).
  - **g.** During Consultation Hold.
- This feature is not available for a trunk-to-trunk call via DISA.

[One-touch Executive Busy Override]

- Automatic Callback Busy cannot be used on trunks that have this feature enabled (→ 2.10.1 Automatic Callback Busy (Camp-on)).

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Executive Override Deny Set / Cancel
10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature—◆ Executive Busy Override
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Executive
→◆ Executive Busy Override
→◆ Executive Busy Override Deny
10.9 PBX Configuration—[2-9] System—System Options—Option 1—◆ PT Operation—One-touch Busy Override by SCO key
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 3—◆ Executive Override Deny
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 3—◆ Executive Override Deny

PT Programming Manual References

[505] Executive Busy Override
[506] Executive Busy Override Deny

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer
1.9.7 Preventing Other People from Joining Your Conversation (Executive Busy Override Deny)

2.10.3 Call Monitor

Description

Allows an extension user to listen to a busy extension user's existing conversation. The user can hear the conversation, but the user's voice is not heard. If desired, interrupting the call to establish a three-party conference call is available.

Conditions

- COS programming determines extension users who can use this feature.
- This feature is available only when the busy extension is in a conversation with another extension or outside party.
- This feature does not work when the busy extension is in one of the following conditions:
  a. Executive Busy Override Deny (→ 2.10.2 Executive Busy Override) or Data Line Security (→ 2.11.5 Data Line Security) has been set.
  b. While receiving OHCA (→ 2.10.4.3 Off-hook Call Announcement (OHCA), → 2.10.4.4 Whisper OHCA).
  c. During a Conference call (→ 2.14 Conference Features).
  d. During a doorphone call (→ 2.18.1 Doorphone Call).
  e. While Live Call Screening (LCS) or Two-way Record is activated (→ 3.2.2.16 Live Call Screening (LCS) and 3.2.2.30 Two-way Record/Two-way Transfer).
  f. During Consultation Hold.
- This feature stops when the busy extension user presses the following buttons during a conversation (→ 2.21.1 Fixed Buttons and 2.21.2 Flexible Buttons):
  – FLASH/RECALL button
  – HOLD button
  – TRANSFER button
  – CONF (Conference) button
  – DSS button
  – EFA button
  – Two-way Record button
  – Two-way Transfer button
  – One-touch Two-way Transfer button
  – Voice Mail (VM) Transfer button

PC Programming Manual References

10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature—◆ Call Monitor
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Executive—◆ Call Monitor
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Option 2—◆ Data Mode
→Option 3—◆ Executive Override Deny
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 3—◆ Executive Override Deny

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer

2.10.4 Second Call Notification to Busy Extension

2.10.4.1 Second Call Notification to Busy Extension—SUMMARY

Description

When attempting to call a busy extension (ringing or having a conversation), an extension user can send a Call Waiting indication to the busy extension (**Call Waiting**). The notification receiving method depends on the called extension's personal setting and the telephone type:

Notification Receiving Method	Description & Reference
Call Waiting Tone	Sends the Call Waiting tone to the busy extension. → 2.10.4.2 Call Waiting Tone
Off-hook Call Announcement (OHCA)	Talk with the busy extension using the built-in speaker and microphone of the called extension while the existing call is made using the handset. → 2.10.4.3 Off-hook Call Announcement (OHCA)
Whisper OHCA	Send a spoken message to a busy extension that will be heard directly by only the called extension user, through the handset, without interrupting the ongoing conversation. → 2.10.4.4 Whisper OHCA

Conditions

- Each extension user can choose to receive Call Waiting tone, OHCA, Whisper OHCA, or none of these.
- OHCA and Whisper OHCA are enabled or disabled by the COS of the calling extension.
- OHCA and Whisper OHCA do not work for some telephone types. In such cases, the Call Waiting tone will be sent to the called extension.

Calling Extension's OHCA COS Mode	Called Extension's Call Waiting Mode			
	OFF	ON		
	Cancel	Call Waiting Tone	OHCA	Whisper OHCA
Disable	Call Waiting disabled	Call Waiting tone	Call Waiting tone	Call Waiting tone
Enable	Call Waiting disabled	Call Waiting tone	OHCA (or Call Waiting tone)	Whisper OHCA (or Call Waiting tone)

- The notification receiving methods (Call Waiting tone, OHCA, and Whisper OHCA) are available only when the called extension is having a conversation with another party. If the called party is not yet connected with the other party (e.g., still ringing, on hold, etc.), the calling extension will hear a ringback tone and will be kept waiting until the called extension becomes available to receive the call waiting notification.
- If none of these notification receiving methods (Call Waiting tone, OHCA, or Whisper OHCA) are set at the called party's extension, the caller will hear a reorder tone.

Feature Guide References

2.1.3.3 Call Waiting

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer
1.9.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)

2.10.4.2 Call Waiting Tone

Description

When an extension user attempts to call a busy extension (ringing or having a conversation), the Call Waiting tone can be sent to the called extension to let him know another call is waiting.

Conditions

- This feature only works if the called extension has activated Call Waiting. If it is activated, the calling extension will hear a ringback tone.
- Call Waiting tone can be selected (Tone 1 or Tone 2) through personal programming (Call Waiting Tone Type Selection).
- When the headset mode is on, you can choose whether the call waiting tone is heard form the speaker phone of the telephone or the earpiece of the headset. However, this setting is only available for terminals that support call waiting tone path switching (KX-DT521, KX-DT543, KX-DT546, KX-NT553, and KX-NT556).

PC Programming Manual References

9.14 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Port Property—Option—◆ C.Waiting with Headset
10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Caller ID—Visual Caller ID Display (s)
10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature
→◆ BSS / OHCA / Whisper OHCA / DND Override
→◆ BSS / OHCA / Whisper OHCA / DND Override-2
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Option 2—◆ Manual C. Waiting for Extension Call
→Option 2—◆ Automatic C. Waiting
→Option 4—◆ Call Waiting Tone Type
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings
→Option 2—◆ Manual C. Waiting for Extension Call
→Option 2—◆ Automatic C. Waiting
→Option 4—◆ Call Waiting Tone Type

User Manual References

1.9.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)
3.1.2 Settings on the Programming Mode

2.10.4.3 Off-hook Call Announcement (OHCA)

Description

An extension user can talk with a busy extension through the built-in speaker and microphone of the called party's PT. If the existing call is using a handset, a second conversation is made using the speakerphone and microphone so that the called extension can talk to both parties.

Conditions

- COS programming determines which extensions can use this feature.
- This feature is available when the *called extension* uses one of the following telephones:
  - KX-T7625, KX-T7630, KX-T7633, KX-T7636, KX-DT333, KX-DT343, KX-DT346, KX-DT543, KX-DT546
  - KX-T7536
  - KX-T7436
- The OHCA feature cannot be used in the following cases:
  a. COS or called extension's telephone type is not available for this feature.
  b. The called extension (DPT) is in the Digital XDP connection.
  The Call Waiting tone is sent to the called extension. (→ 2.10.4.2 Call Waiting Tone)
- While an extension is receiving OHCA, if the extension user places the current trunk call on hold or transfers the current intercom call or trunk call, OHCA will become disabled and the calling extension will start to hear a ringback tone.
- While an extension is receiving OHCA, if the extension user places the current intercom call on hold, the called extension can talk to the calling extension through the handset.

PC Programming Manual References

10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature
→◆ BSS / OHCA / Whisper OHCA / DND Override
→◆ BSS / OHCA / Whisper OHCA / DND Override-2
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Assistant—◆ OHCA / Whisper OHCA

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

1.9.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)

2.10.4.4 Whisper OHCA

Description

An extension user can send a spoken message to a busy extension that will be heard directly by only the called extension user, through the handset, without interrupting the ongoing conversation. The caller cannot hear the ongoing conversation or the called extension user's reply, unless the called extension user puts the current party on hold and switches to the waiting caller.

Conditions

- COS programming determines which extensions can use this feature.
- This feature is available when the calling and called extension use one of the following telephones:
  - KX-DT300 series
  - KX-DT500 series
  - KX-T7600 series
  - KX-T7500 series
  - KX-T7400 series (except KX-T7451)
  - IP-PT
- If the Whisper OHCA feature cannot be used due to COS or telephone type, the Call Waiting tone will be sent to the called extension. (→ 2.10.4.2 Call Waiting Tone)
- To receive Whisper OHCA on an IP-PT, the preferred codec must be either G.711 or G.729A. When an extension user is on a call using the G.722 codec and receives Whisper OHCA, he will hear the Call Waiting tone instead. (→ 2.10.4.2 Call Waiting Tone)
- If the called extension does not use a KX-DT300, KX-DT500, KX-T7600, KX-T7500, or KX-T7400 series telephone, or an IP-PT, but forces Whisper OHCA, the announcement may be heard by the other party.
- It is possible to enable Whisper OHCA on any telephone. However, it may not work properly. (e.g., The voice may be heard by the other party.)
- When a non-IP extension is connected to a non-IP trunk and the extension receives Whisper OHCA, Whisper OHCA will not function. The extension user will hear the Call Waiting tone instead. (→ 2.10.4.2 Call Waiting Tone)
- While an extension is receiving Whisper OHCA, if the extension user places the current trunk call on hold or transfers the current intercom call or trunk call, Whisper OHCA will become disabled and the calling extension will start to hear a ringback tone.
- While an extension is receiving Whisper OHCA, if the extension user places the current intercom call on hold, the called extension can talk to the calling extension through the handset.

PC Programming Manual References

9.14 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Port Property—Option—◆ IP Codec Priority
10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature
→◆ BSS / OHCA / Whisper OHCA / DND Override
→◆ BSS / OHCA / Whisper OHCA / DND Override-2
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Assistant—◆ OHCA / Whisper OHCA

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

1.9.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)

2.11 Conversation Features

2.11.1 Hands-free Operation

Description

A PT user can talk to another party without lifting the handset. Pressing specific buttons (e.g., REDIAL) automatically activates hands-free mode.

Conditions

- **PTs with the MONITOR Button**
  PTs with the MONITOR button can only dial in hands-free mode and cannot be used for hands-free conversations.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Tone Length—Reorder Tone for PT Hands-free (s)
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 4—◆ LCS Answer Mode

2.11.2 Off-hook Monitor

Description

A PT user can let others listen to the user's conversation through the built-in speaker, during a conversation using the handset.

Conditions

- **Capable Telephones**
  - KX-DT300 series
  - KX-DT500 series
  - KX-T7600 series
  - KX-T7500 series (display PTs only)
  - KX-T7400 series (display PTs only)
  - KX-NT series
- To enable this feature, system programming is required. If disabled, hands-free conversation is performed instead.

User Manual References

1.4.7 Letting Other People Listen to the Conversation (Off-hook Monitor)

2.11.3 Mute

Description

During a conversation, a PT user can disable the speaker microphone or the handset microphone to consult privately with others while listening to the other party on the phone through the built-in speaker or the handset receiver. The user can hear the other party's voice during Mute, but cannot be heard.

Conditions

- This feature is available with all PTs that have the AUTO ANS/MUTE button.

User Manual References

1.4.6 Mute

2.11.4 Headset Operation

Description

This PBX allows the use of headset-compatible PTs. A PT user can talk to another party without lifting the handset. This feature is also known as Handset/Headset Selection.
For connection and operation, refer to the Operating Instructions for the headset.

Conditions

- **Hardware Requirement:** An optional headset
- If headset mode is on, pressing the SP-PHONE button activates the headset, not the built-in speaker.
- To set headset mode on a DPT or IP-PT, use personal programming (Headset Operation) or press the Headset button. To set headset mode on an APT, use the handset/headset selector provided on the set and/or on the headset.
- **Headset Button**
  A flexible button on a DPT or IP-PT can be customised as a Headset button. It is possible to assign a Headset button to a flexible button on an APT, but the button will not function.
- **Answer/Release Button**
  A flexible button can be customised as an Answer button or a Release button. Such buttons are useful for headset operation. It is possible to answer an incoming call by pressing an Answer button. While hearing the Call Waiting tone during a conversation, pressing an Answer button enables one to answer the second call by placing the current call on hold. Pressing a Release button enables one to disconnect the line during or after conversation, or to complete a Call Transfer.
- It is possible to switch from headset mode to hands-free mode or vice versa during a conversation by pressing the Headset button.
- Headset users cannot use the following features:
  – Automatic Redial (→ 2.6.3 Last Number Redial)
  – Receiving OHCA
  – Receiving Whisper OHCA (→ 2.10.4.4 Whisper OHCA)

PC Programming Manual References

9.14 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Port Property—Option—◆ Headset OFF/ON
9.22 PBX Configuration—[1-1] Configuration—Slot—Extension—Port Property—◆ Headset OFF/ON
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.3 PBX Configuration—[4-3] Extension—DSS Console—◆ Type

Feature Guide References

2.21.2 Flexible Buttons

User Manual References

1.4.8 Using the Headset (Headset Operation)
3.1.2 Settings on the Programming Mode

2.11.5 Data Line Security

Description

Setting Data Line Security at an extension protects communications between the extension and the other party from being interrupted by signals such as Call Waiting, Hold Recall and Executive Busy Override. An extension that is using a connected data device (e.g., a fax machine) can set this feature to maintain secure data transmission by preventing tones or interruptions from other extensions during communication.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Data Line Security Set / Cancel
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 2—◆ Data Mode

User Manual References

1.9.9 Protecting Your Line against Notification Tones (Data Line Security)

2.11.6 Flash/Recall/Terminate

Description

The FLASH/RECALL button (Flash/Recall mode or Terminate mode) or Terminate button (Terminate mode) is used when a PT user disconnects the current call and originates another call without hanging up first. It performs the same function as going on-hook and then going off-hook.
[Explanation of Each Mode]
Flash/Recall Mode: Disconnects the line. The extension user hears the dial tone from the line used last. For example, if a trunk call is disconnected, the extension user will hear a new dial tone from the telephone company.
Terminate Mode: Disconnects the line. The extension user hears the dial tone determined by the Line Preference—Outgoing setting. (→ 2.5.5.2 Line Preference—Outgoing)

Conditions

- **FLASH/RECALL Button Mode**
  One of the following modes can be selected for each extension through system programming:
  – Flash/Recall mode
  – Terminate mode
  – External Feature Access (EFA) mode. (→ 2.11.7 External Feature Access (EFA))
- **Terminate Button**
  A flexible button can be customised as the Terminate button.
- **Disconnect Time (Only for Flash/Recall Mode)**
  The amount of time between successive accesses to the same trunk is programmable for each trunk port.
- This feature outputs an SMDR call record (→ 2.22.1.1 Station Message Detail Recording (SMDR)), restarts the call timer, inserts the automatic pause, and checks the TRS/Barring level (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring)) again.
- The Terminate feature will be performed when pressing the FLASH/RECALL button regardless of the mode that the FLASH/RECALL button has been set to, in the following situations:
  – When a call is made using ARS. (→ 2.8.1 Automatic Route Selection (ARS))
  – When a trunk call is made with the INTERCOM button.
  – When a trunk call is made with an ICD group button.
- For general SIP phones, the function of a FLASH button differs depending on the phone, and its functionality does not depend on the setting specified in the PBX.

PC Programming Manual References

9.24 PBX Configuration—[1-1] Configuration—Slot—Port Property - LCO Port—◆ Disconnect Time
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 7—◆ Flash Mode during CO Conversation
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 7—◆ Flash Mode during CO Conversation
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Type

PT Programming Manual References

[418] LCOT Disconnect Time

Feature Guide References

2.21.2 Flexible Buttons

2.11.7 External Feature Access (EFA)

Description

Normally, an extension user can only access features within the PBX. However, when performing External Feature Access (EFA) the extension user performs features outside of the PBX, such as using the transfer services of the telephone company or host PBX. When EFA is performed, the PBX sends a flash/recall signal to the telephone company or the host PBX (→ 2.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)).
This feature is only available on trunk calls.
This feature is performed by pressing the EFA button or the FLASH/RECALL button that is set to EFA mode (→ 2.11.6 Flash/Recall/Terminate).

Conditions

- **Flash/Recall Time**
  The Flash/Recall time can be assigned for each trunk port.
- **EFA Button**
  A flexible button can be customised as the EFA button.
- It is possible to perform this feature by entering the feature number while the current call is placed on Consultation Hold (e.g., is going to be transferred to an extension of the host PBX).
  Consultation Hold: a condition that a party is in, when an extension is calling other parties in order to perform Call Transfer, Conference, or Call Splitting.
  In Consultation Hold, the original call is treated as if it is on hold, allowing the extension to call a third party all on one line. In Call Hold, the party on hold and the third party are connected to the extension using separate lines.

PC Programming Manual References

9.24 PBX Configuration—[1-1] Configuration—Slot—Port Property - LCO Port—◆ Flash Time
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ External Feature Access
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 7—◆ Flash Mode during CO Conversation
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 7—◆ Flash Mode during CO Conversation
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Type
15.5 PBX Configuration—[7-5] TRS—Miscellaneous—◆ TRS Check after EFA

PT Programming Manual References

[417] LCOT Flash/Recall Time

Feature Guide References

2.21.2 Flexible Buttons

User Manual References

1.11.4 If a Host PBX is Connected

2.11.8 Trunk Call Limitation

Description

Trunk calls are limited by the following features:

Feature	Description
Extension-to-Trunk Call Duration	If a call between an extension user and an outside party is established, the call duration can be restricted by a system timer selected for each trunk group. Both parties will hear warning tones at five-second intervals starting 15 seconds before the time limit.*1 When the time limit expires, the line will be disconnected. COS programming determines whether this feature is enabled or disabled. Whether this feature applies to outgoing calls only, or to both outgoing and incoming calls is determined through system programming.
Trunk-to-Trunk Call (except Unattended Conference Call) Duration	If a call between two outside parties is established, the call duration can be restricted by a system timer selected for each trunk group. Both parties will hear warning tones at five-second intervals starting 15 seconds before the time limit.*1 When the time limit expires, the line will be disconnected. If both parties involved in the trunk-to trunk call were established by an extension (e.g., an extension makes a trunk call, then transfers the call to an outside party), the time limit applied to the trunk call that was made first will be used.
Budget Management	When the preprogrammed call charge limit has been reached, an extension user will hear 3 warning tones at five-second intervals. It is programmable whether the line is disconnected after the third tone. After the call has ended, the extension user cannot make further trunk calls until the charge limit has been increased or cleared by an extension assigned as a manager (→ 2.7.2 Budget Management).
Dialling Digit Restriction during Conversation	While engaged in an incoming trunk call, the dialling of digits can be restricted. If the number of dialled digits exceeds the limitation, the line will be disconnected.
Logical Partitioning Feature (For India only)	Some types of trunk calls can be restricted with the following Logical Partitioning settings: 9.2.2 PBX Configuration—[1-1] Configuration—Slot—System Property—Site - Main - ◆ Area ID for logical partition 10.9 PBX Configuration—[2-9] System—System Options - Option 2 -◆ Applying logical partitioning For details, refer to "10.9 PBX Configuration—[2-9] System—System Options - Option 2 -◆ Applying logical partitioning" in the PC Programming Manual. The following log data can be collected and displayed through PC Programming. Refer to "7.3.5 Utility—Log—Call Control Log" in the PC Programming Manual. **1.** Log data (condition) will be recorded when you change any of the Logical Partitioning settings. **2.** Log data (status) will be recorded whenever a call is restricted by the Logical Partitioning feature.

*1 A party connected via an IP trunk or SIP trunk will not hear the warning tone.

Conditions

- During an Unattended Conference Call, the Unattended Conference Recall time is applied. (→ 2.14.2 Conference)
- When using LCO trunks that do not support Calling Party Control (CPC) signal detection (→ 2.11.9 Calling Party Control (CPC) Signal Detection), the Trunk-to-Trunk Call Duration timer should not be disabled, as automatic end of call detection cannot be performed.
- For SIP Extension users, the line will be disconnected without hearing any warning tones when the trunk call limitation expires.

PC Programming Manual References

10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—CO & SMDR—◆ Extension-CO Line Call Duration Limit
10.9 PBX Configuration—[2-9] System—System Options—Option 2—◆ Extension - CO Call Limitation—For Incoming Call
11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Main
→◆ CO-CO Duration Time (*60s)
→◆ Extension-CO Duration Time (*60s)
15.5 PBX Configuration—[7-5] TRS—Miscellaneous—◆ Dial Digits Limitation After Answering—Dial Digits
7.3.5 Utility—Log—Call Control Log
9.2.2 PBX Configuration—[1-1] Configuration—Slot—System Property—Site—Main—◆ Area ID for logical partition
10.9 PBX Configuration—[2-9] System—System Options—Option 2—◆ Applying logical partitioning

PT Programming Manual References

[472] Extension-to-Trunk Call Duration
[473] Trunk-to-Trunk Call Duration
[502] Trunk Call Duration Limitation

Feature Guide References

5.1.1 Class of Service (COS)

2.11.9 Calling Party Control (CPC) Signal Detection

Description

The Calling Party Control (CPC) signal is an on-hook indication (disconnect signal) sent from the analogue trunk when the other party hangs up. To maintain efficient utilisation of trunks, the PBX monitors their state and when CPC signal is detected from a line, disconnects the line and alerts the extension with a reorder tone.

Conditions

- CPC signal detection is programmable for incoming trunk calls, and for outgoing trunk calls.
- If your telephone company sends other signals similar to CPC, it is recommended not to enable CPC signal detection on outgoing trunk calls.
- If a CPC signal is detected during a Conference call (→ 2.14.2 Conference), 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 (→ 2.16.1 Direct Inward System Access (DISA)) and an extension or an outside party, the line is disconnected.

PC Programming Manual References

9.24 PBX Configuration—[1-1] Configuration—Slot—Port Property - LCO Port—◆ CPC Signal Detection Time—Outgoing, Incoming

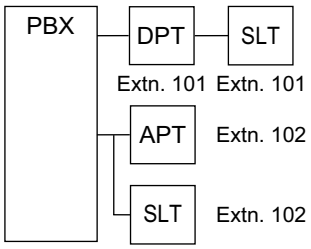
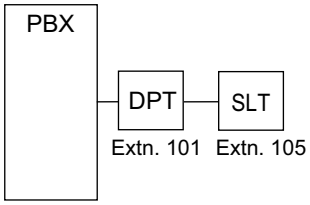
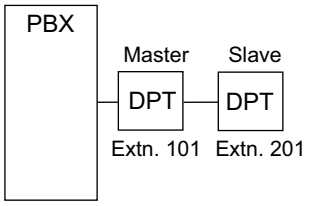
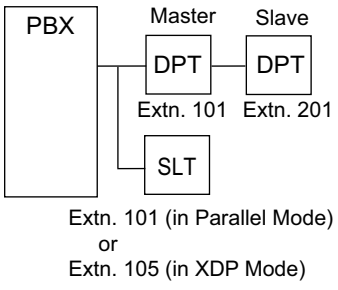
PT Programming Manual References

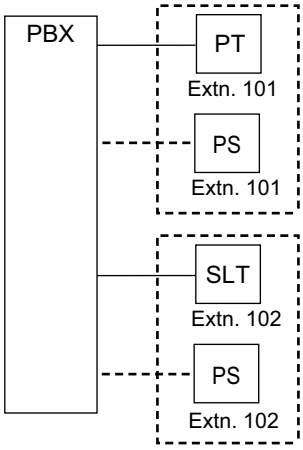
[413] LCOT CPC Signal Detection Time—Outgoing
[414] LCOT CPC Signal Detection Time—Incoming

2.11.10 Parallelled Telephone

Description

Multiple telephones can be connected to the same port. This is useful to increase the number of telephones without additional extension cards. The combinations and features of the parallelled telephones are described below.

Features	Descriptions	Connections
Parallel Mode	Parallel mode involves the connection of an SLT to an APT or a DPT that is connected to a Super Hybrid port. When parallel mode is enabled, the two telephones function as follows: • Both share the extension number of the telephone connected directly to the PBX (main telephone). • Either telephone can make or answer a call.	**APT/DPT + SLT** 
EXtra Device Port (XDP) Mode	XDP mode involves the connection of an SLT to a DPT that is connected to a Super Hybrid port. Unlike parallel mode, each telephone can act as a completely different extension with its own extension number. (→ 5.2.6 Extension Port Configuration)	**DPT + SLT** 
Digital XDP	Digital XDP involves the connection of a DPT to a DPT that is connected to a DPT port or Super Hybrid port. The DPT that is connected directly to the PBX is called the "master DPT", and the DPT connected to the master DPT is called the "slave DPT". Like XDP mode, each telephone can act as a completely different extension with its own extension number. If a master DPT is connected to the PBX by a Super Hybrid port (not a DPT port), a third telephone (SLT) can also be in parallel or XDP mode with the master DPT. Digital XDP connection allows the number of DPTs that the PBX supports to increase.	**DPT + DPT**  **DPT + DPT + SLT** 

Features	Descriptions	Connections
Wireless XDP Parallel Mode	For this connection, refer to 5.2.4.5 Wireless XDP Parallel Mode.	**APT/DPT/SLT + PS** 

Conditions

[APT + SLT]

- If one telephone goes off-hook while the other is on a call, a three-party call is established. If one user goes on-hook, the other user continues the call.
- An extension user cannot originate a call from the SLT if the APT is:
  – playing background music (BGM)
  – receiving a paging announcement over the built-in speaker.
- **For users in Germany and Austria only**
  Although the APT will ring for incoming calls, the SLT will not ring.
  For users in other countries/areas
  Both the APT and the SLT will ring for incoming calls, and the PBX cannot refuse calls arriving at the SLT.

[DPT + SLT]

- It is programmable whether to have the DPT and SLT in either parallel or XDP mode. Regardless of the mode, the SLT can be connected directly to the XDP port of the DPT or to a modular T-adapter with the DPT.
- When in parallel mode, it is programmable whether the SLT rings for incoming calls.
  Ring on: Both telephones ring except when the PT is in Hands-free Answerback mode (→ 2.4.4 Hands-free Answerback) or voice-calling mode (Alternate Receiving Ring/Voice) (→ 2.5.3 Intercom Call).
  Ring off: Only the PT rings. However, the SLT can answer the call.
- Both telephones cannot engage in calls simultaneously. If one telephone goes off-hook while the other is on a call, the call is switched to the former. The call is not switched in the following cases:
  a. While being monitored by another extension. (→ 2.10.3 Call Monitor)
  b. While receiving OHCA (→ 2.10.4.3 Off-hook Call Announcement (OHCA)) or Whisper OHCA. (→ 2.10.4.4 Whisper OHCA)
  c. During a Conference call (→ 2.14 Conference Features).
  d. While Live Call Screening (LCS) or Two-way Record is activated (→ 2.28.3 Voice Mail DPT (Digital) Integration).

[DPT + DPT]

- **Capable Telephones**

KX-DT300 series, KX-DT500 series, and KX-T7600 series, except KX-T7640. Note that the KX-T7667 can only be connected as a slave DPT.

- When using Digital XDP connection, the following features cannot be used with either the master or slave DPT:
  a. OHCA: A call waiting tone will be heard even if the OHCA feature is set.
  b. USB Module/Bluetooth Module: The DPTs will not work correctly if a USB Module or Bluetooth Module is connected. Do not connect USB Modules or Bluetooth Modules to the DPTs.

Note

Even if the slave DPT is disconnected, the OHCA feature and USB Module still cannot be used with the master DPT. To use them, the master DPT must be disconnected from the PBX, and then reconnected.

[DPT + DPT + SLT]

- When an SLT is connected to the slave DPT in parallel mode, the SLT works as the parallel extension of the master DPT.

Installation Manual References

2.3.3 System Capacity
4.8.2 Parallel Connection of the Extensions
4.8.3 Digital EXtra Device Port (Digital XDP) Connection

PC Programming Manual References

9.22 PBX Configuration—[1-1] Configuration—Slot—Extension—Port Property
→◆ XDP Mode
→◆ Parallel Telephone Ringing
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Parallel Telephone (Ring) Mode Set / Cancel

PT Programming Manual References

[600] EXtra Device Port (XDP) Mode

User Manual References

1.9.11 Setting the Parallelled Telephone to Ring (Parallelled Telephone)

2.11.11 One-numbered Extension

Description

It is possible to share one extension number between a main extension and a sub extension as a paired extension. The sub extension that is paired with the main extension can be called simultaneously by the extension number of the main extension (paired main extension number). When an incoming call is received at the main extension, it is treated as a call for the paired main extension number and the incoming call will arrive at the sub extension simultaneously.
A user can pick up the call for the paired extension by entering the corresponding feature number or pressing the corresponding flexible button. However, if one of the extensions is in a conference call, the other extension cannot pick up the call.

Conditions

- The following extensions can be assigned as a main/sub extension.
  PT, SLT, IP-PT(except S-PS), SIP extension (including KX-UT series SIP phones and general SIP phones)
- If an S-PS is paired as the sub extension, operation is the same as Wireless XDP.
- The paired main extension number is displayed on the sub extension when the telephone is idle. However, for KX-UT series SIP phones and general SIP phones, the original extension number is displayed on the sub extension when the telephone is idle.
- If a main extension is already configured with Wireless XDP or paired with one-numbered extension, the main extension cannot be paired with another sub extension.
- The sub extension operates according to the COS and extension settings of the main extension (except key settings, incoming line preference, and outgoing line preference).
- When calling from a sub extension, the caller information (extension number, extension Name, CLIP/CNIP) of the paired main extension number is used.
- When an incoming call is received at a one-numbered extension, the behaviour differs depending on the type of telephone, as follows:
  [KX-NT series/DPT]
  - If the main extension is busy and there is a flexible button available to receive the call, the call waiting feature can function.
  - If the main extension cannot receive the incoming call, the call waiting feature will not function.

  [SLT]
  - If the main extension is an SLT and is busy, the sub extension cannot receive the call.

  [KX-UT series]
  - If the main extension is a KX-UT series SIP phone and the sub extension is an SLT or a KX-UT series SIP phone, the sub extension cannot receive calls while the main extension is busy.
  - If the main extension is a KX-UT series SIP phone, neither the main extension nor the sub extension can receive calls. This condition applies regardless of the type of telephone used for the sub extension.

  [KX-UT series SIP phone and general SIP phones.]
  - The no-ring or delayed ringing features are not activated when a new call is received at a paired extension during a call. In this case, the telephone will ring normally.
- When making an extension call to a paired main extension number using voice-calling, a ring tone is heard at the sub extension.
- When a call is received to a paired main extension number in LCS with Hands-free mode, the call will not be received at a sub extension.
- The following features are available only at the main extension.
  – OHCA (→ 2.10.4.3 Off-hook Call Announcement (OHCA))
  – Whisper OHCA (→ 2.10.4.4 Whisper OHCA)
- When a paired main extension number is paged, the sub extension will not be paged.
- When a paired main extension number is called for a conference, the call will also arrive at the sub extension.

- When one of the main extension or sub extension is busy, the paired extension cannot make a call.
- While a sub extension is activated as a One-numbered extension, calls for the original extension number of the sub extension will not be received.
- The message waiting lamp can be controlled by both the main extension and sub extension simultaneously.
- The sub extension of a paired extension can be programmed by using the Wireless XDP feature number.
- When an extension that is registered as a member of an ICD group is paired as a sub extension, incoming calls to the sub extension (via the ICD group) will not ring. At the same time, the sub extension is forced to log out from the ICD group.
- The following settings for each telephone type are activated individually depending on the settings of the main/sub extension.
  SLT MW Mode
  Automatic Answer
  ICM Tone
  Ring Pattern Table
  ISDN Bearer
  BGM On/Off
  LCS On/Off
- For features such as Hold Recall and Timed Reminder that call the extension back and are set at the extension, the callback is received only at the main or sub extension that set the feature.
- When features such as Transfer Recall redirect a call back to the originating extension, the call is received at the extension number of the main extension (paired main extension number), and both the main and sub extensions will ring.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Parallel Telephone (Ring) Mode Set / Cancel
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Option 1—◆ Wireless XDP / Shared Extension
→Option 1—◆ Ring Pattern Table
→Option 5—◆ Automatic Answer for CO Call
→Option 6—◆ Forced Automatic Answer
→Option 7—◆ ISDN Bearer
→Option 8—◆ SLT MW Mode

User Manual References

1.9.13 Using Your phone in Parallel with a Wired Telephone (One-numbered extension)

Feature Guide References

2.9.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension
3.2.2.16 Live Call Screening (LCS)

2.12 Transferring Features

2.12.1 Call Transfer

Description

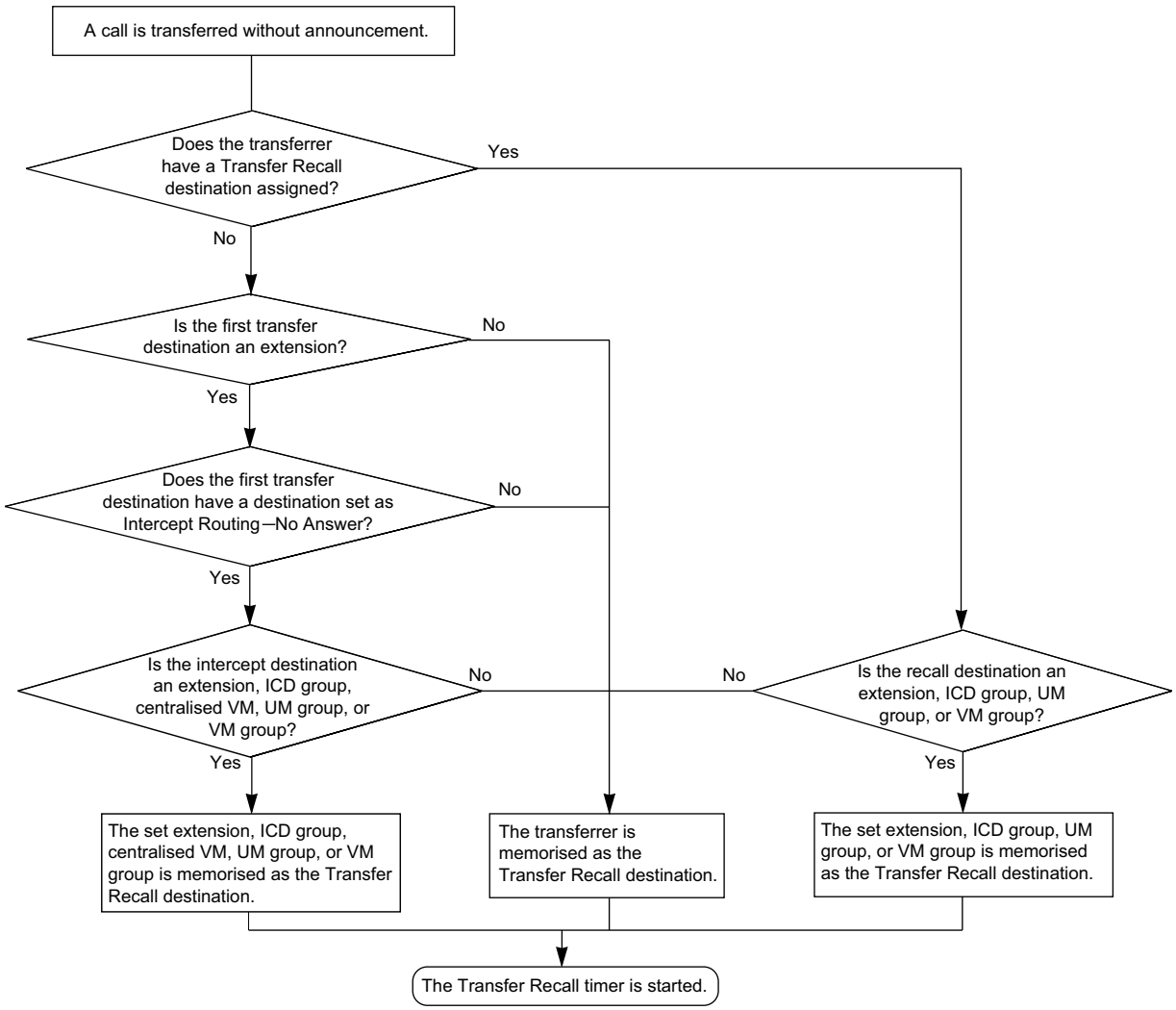
An extension user can transfer a call to another extension or an outside party. The following features are available:

Feature	Transferring method
With Announcement	Transfer is completed after announcing the destination party.
Without Announcement	Transfer is completed without an announcement. After dialling the destination, while hearing a ringback tone, the originator can replace the handset.

Call Transfer with Announcement is also known as Call Transfer—Screened.
Call Transfer without Announcement is also known as Call Transfer—Unscreened.

Transfer Recall for Call Transfer without Announcement

If the transfer destination does not answer within the preprogrammed Transfer Recall time, the call will be redirected to the Transfer Recall destination assigned to the extension which transferred the call.
If the transfer destination has a destination set as Intercept Routing—No Answer, the call will be routed to that destination.



[Available destination]

Destination	Availability
Wired Extension (PT/SLT/SIP Extension/ISDN Extension)	✓
PS	✓
Incoming Call Distribution Group	✓
PS Ring Group	
UM Group	✓*1
VM Group (DTMF/DPT)	✓ (DPT only)*1
External Pager (TAFAS)	

Destination	Availability
DISA	
Analogue/ISDN Remote Maintenance	
Idle Line Access no. + Phone no.	
Trunk Group Access no. + Trunk Group no. + Phone no.	
Other PBX Extension (TIE with no PBX Code)	
Other PBX Extension (TIE with PBX Code)	

*1 If the transfer destination does not answer, the call is sent to Voice Mail and a message can be recorded in the mailbox of the transfer destination.

Conditions

- When an extension is transferring a party to another destination, the party will be in consultation hold until they reach the transfer destination.
  Consultation Hold: a condition that a party is in, when an extension is calling other parties in order to perform Call Transfer, Conference, or Call Splitting.
  In Consultation Hold, the original call is treated as if it is on hold, allowing the extension to call a third party all on one line. In Call Hold, the party on hold and the third party are connected to the extension using separate lines.
- If Music on Hold is enabled, music can be sent to the held party while the call is transferred. (→ 2.13.4 Music on Hold) It is programmable whether a ringback tone or music is sent.
- If the transfer destination extension has set FWD to an outside party, the call will be transferred to the outside party. (→ 2.3.2 Call Forwarding (FWD))
- COS programming determines the extensions that are able to transfer a call to an outside party. COS can also prohibit transferring to an extension of another PBX via the TIE line service using the PBX Code method (Access with PBX Code) (→ 4.2.1 TIE Line Service).
- **One-touch Transfer**
  One-touch Transfer can be performed by pressing a One-touch Dialling button that has been assigned the TRANSFER command and the telephone number of the transfer destination. This is useful for transferring calls to an outside destination. (→ 2.6 Memory Dialling Features)
- **Automatic Transfer by SDN Button or DSS Button**
  Pressing an SDN button or DSS button during a conversation with an extension or outside party can automatically transfer the call to the specified destination (→ 2.9.1 Primary Directory Number (PDN)/ Secondary Directory Number (SDN) Extension). It is possible through system programming to prevent this feature from operating for extension to extension calls.
- **Transfer to Busy Extension using Queuing (Camp-on Transfer)**
  Through system programming, it is possible to enable the transferring of a call to a busy extension without needing to send a call waiting notification, based on the transferring party's COS setting. The transferred call will be placed in a queue.
  This feature is not available for SIP extensions.
- When transferring a call from an analogue trunk, users are strongly recommended to perform a screened transfer, so that the outside caller is not automatically connected to an extension using Hands-free Answerback when the extension user is absent.
- If a KX-UT series SIP phone user disconnects a call while the party to be transferred is still on consultation hold (i.e., has not been transferred), Hold Recall is heard at the extension immediately (→ 2.13.1 Call Hold). On other types of extensions, Hold Recall is heard after the Hold Recall timer expires.
- This PBX supports the Blind transfer feature found on some SIP phones. For details, refer to the phone's documentation.

PC Programming Manual References

10.2 PBX Configuration—[2-2] System—Operator & BGM—◆ BGM and Music on Hold—Sound on Transfer
10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Recall—Transfer Recall (s)
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—CO & SMDR—◆ Transfer to CO
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Assistant—◆ Transfer to busy Extension w/o BSS Operation
10.9 PBX Configuration—[2-9] System—System Options—Option 4—◆ DSS Key—Automatic Transfer for Extension Call
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1—◆ Transfer Recall Destination
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 1—◆ Transfer Recall Destination

PT Programming Manual References

[201] Transfer Recall Time
[503] Call Transfer to Trunk
[712] Music for Transfer

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

1.4.1 Transferring a Call (Call Transfer)

2.12.2 SIP Refer Transfer

Description

If enabled through system programming, calls transferred to a SIP trunk will be transferred using the SIP service provider's Transfer feature instead of the PBX's.
The following types of transfer are available:

Attended Transfer	Transfer is completed after announcing the transferred party. From the extension user's perspective, this is the same as Call Transfer with Announcement (→ 2.12.1 Call Transfer).
Blind Transfer	Transfer is completed immediately after dialling the transfer destination's number. This is similar to Call Transfer without Announcement (→ 2.12.1 Call Transfer), except that the transferrer does not hear even a ringback tone; the transferred call is connected directly to the destination.

Conditions

[General]

- The availability of this feature depends on the SIP service provider.
- Since the SIP service provider takes control of the transfer, the transferred call cannot be returned to the PBX for further handling even if the transfer fails.

[Blind Transfer]

- ISDN extensions and SIP extensions cannot use this feature.
- This feature cannot be used when calling through DISA (→ 2.16.1 Direct Inward System Access (DISA)).

Installation Manual References

4.4 Virtual Cards

User Manual References

1.4.1 Transferring a Call (Call Transfer)—◆◆ Transferring to an Outside Party Using the SIP Service

PC Programming Manual References

9.10 PBX Configuration—[1-1] Configuration—Slot—V-SIPGW—Port Property—Supplementary Service
→◆ Blind Transfer (REFER)
→◆ Attended Transfer (REFER)

2.13 Holding Features

2.13.1 Call Hold

Description

An extension user can put a call on hold. The following features are available depending on the result.

Feature	Description
Regular Hold	Any extension can retrieve a held call.
Exclusive Call Hold	Only the extension user who held the call can retrieve it.

The result of the holding operation can be determined through system programming. Pressing the HOLD button again just after the first time alternates the mode between Regular and Exclusive Call Hold.

Conditions

- **Call Hold Limitation**
  A PT user can hold one intercom call and/or multiple trunk calls at a time. An SLT user can hold either one intercom call or one trunk call at a time. By using the Call Park feature, PT and SLT users can hold multiple trunk calls and intercom calls simultaneously. (→ 2.13.2 Call Park)
- **Music on Hold**
  Music, if available, is sent to the held party. (→ 2.13.4 Music on Hold)
- **Hold Recall**
  If a call on hold is not retrieved within a preprogrammed time period, Hold Recall is heard at the extension which put the call on hold. If the extension is engaged in a call, the Hold Alarm will be heard.
- If an outside party is placed on hold and not retrieved within a preprogrammed time period, the call is automatically disconnected. This timer starts when Hold Recall activates.
- **Automatic Call Hold**
  A PT user can be programmed holding of the current call when pressing another CO/ICD Group/ INTERCOM/PDN button, through system programming. If this feature is not enabled, the current call will be disconnected.
  [Example]
  It is possible to receive a call by pressing the flashing ICD Group button, this puts the current intercom call (on the INTERCOM button) on hold. To return to the held call, press the INTERCOM button.
- **Call Hold Retrieve Deny**
  If an extension user cannot call certain extensions on a COS basis (→ 2.1.2.2 Internal Call Block), he cannot retrieve the held call which the extensions made.
- **SLT Hold Mode**
  It is possible to choose how to hold a line and transfer a call with an SLT in the following methods through system programming:

	Hold	**Hold (to be Retrieved from Another Extension)**[*1]	**Transfer to Trunk**	**Transfer to Extension**
Mode 1	Flashing the hookswitch + Going on-hook	Flashing the hookswitch + Hold Feature No. + Going on-hook	Flashing the hookswitch + Trunk Access No.	Flashing the hookswitch + Extension No.

	Hold	Hold (to be Retrieved from Another Extension)[*1]	Transfer to Trunk	Transfer to Extension
Mode 2 (Default)	Flashing the hookswitch + Hold Feature No. + Going on-hook	Flashing the hookswitch + Hold Feature No. + Going on-hook	Flashing the hookswitch + Trunk Access No.	Flashing the hookswitch + Extension No.
Mode 3	Flashing the hookswitch + Hold Feature No. + Going on-hook	Flashing the hookswitch + Hold Feature No. + Hold Feature No. + Going on-hook	Flashing the hookswitch + Hold Feature No. + Trunk Access No.	Flashing the hookswitch + Extension No.
Mode 4	Flashing the hookswitch + Hold Feature No. + Going on-hook	Flashing the hookswitch + Hold Feature No. + Hold Feature No. + Going on-hook	Flashing the hookswitch + Hold Feature No. + Trunk Access No.	Flashing the hookswitch + Hold Feature No. + Extension No.

[*1] These operations must be performed when the held call is intended to be retrieved from another extension using the holding extension number.

If the following occurs frequently with an SLT, choose "**Mode 2**", "**Mode 3**", or "**Mode 4**":

a. When an SLT user receives a call, reorder tone is heard or nobody answers the call.

b. When an SLT user goes off-hook, reorder tone is heard instead of a dial tone.

If a call is not terminated after going on-hook, the above cases occur. To avoid these problems, choose "**Mode 2**", "**Mode 3**", or "**Mode 4**". Every call will be terminated unless the Hold feature number is entered after flashing the hookswitch in Mode 2, Mode 3, and Mode 4.

- Hold Alarm tone pattern has a default. (→ 6.2.1 Tones/Ring Tones).

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone
→◆ Recall—Hold Recall (s)
→◆ Recall—Disconnect after Recall (x60s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ Call Hold / Call Hold Retrieve
→◆ Call Hold Retrieve : Specified with a Holding Extension Number
→◆ Hold Retrieve : Specified with a Held CO Line Number
10.9 PBX Configuration—[2-9] System—System Options
→Option 1—◆ PT Operation—Automatic Hold by ICM / CO / ICD Group Key
→Option 1—◆ PT Operation—Hold key mode
→Option 5—◆ SLT—SLT Hold Mode

PT Programming Manual References

[200] Hold Recall Time

User Manual References

1.4.2 Holding a Call

2.13.2 Call Park

Description

An extension user can place a call into a common parking zone of the PBX. The Call Park feature can be used as a transferring feature; this releases the user from the parked call to perform other operations. The parked call can be retrieved by any extension user.

Conditions

- **Automatic Call Park**
  It is possible to select an idle parking zone automatically.
- **Retry**
  If the specified parking zone is occupied or there is no vacant zone for Automatic Call Park, the originator will hear a busy tone. Retrying is possible while hearing the busy tone by selecting parking zone or a vacant zone.
- **Call Park Recall**
  If a parked call is not retrieved within a preprogrammed time period, Call Park Recall will be heard at the Transfer Recall destination assigned to the extension which parked the call. If the destination is engaged in a call, the Hold Alarm will be heard.
- If a parked trunk call is not retrieved within a preprogrammed time period (Default: 30 minutes), it is automatically disconnected.
- **Call Park Button**
  Pressing the Call Park button parks or retrieves a call in a preset parking zone.
  A flexible button can be customised as the Call Park button. It shows the current status of the preset parking zone as follows:

Light pattern	Status
Slow red flashing	Parked in the preset parking zone
Off	No parked call

- **Call Park (Automatic Park Zone) Button**
  Pressing the Call Park (Automatic Park Zone) button parks a call in an idle parking zone automatically. A flexible button can be customised as the Call Park (Automatic Park Zone) button.
  On a KX-UT series SIP phone, pressing a Call Park (Automatic Park Zone) button selects an idle parking zone from among the Call Park (preset parking zone) buttons configured on the phone.
- **Call Park Retrieve Deny**
  If an extension user cannot call certain extensions on a COS basis (→ 2.1.2.2 Internal Call Block), he cannot retrieve the parked call which the extensions made.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters
→◆ Recall—Call Park Recall (s)
→◆ Recall—Disconnect after Recall (x60s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Call Park / Call Park Retrieve
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Parameter Selection (for Call Park)
→◆ Optional Parameter (Ringing Tone Type Number) (for Call Park)
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button

→◆ Type
→◆ Parameter Selection (for Call Park)
→◆ Optional Parameter (or Ringing Tone Type Number) (for Call Park)

Feature Guide References

2.21.2 Flexible Buttons
6.1 Capacity of System Resources

User Manual References

1.4.2 Holding a Call

2.13.3 Call Splitting

Description

During a conversation, an extension user can call another extension while putting the original party on Consultation Hold. The extension user can then alternate between the two parties and/or connect the original party with the third party.

Conditions

- **Consultation Hold:** a condition that a party is in, when an extension is calling other parties in order to perform Call Transfer, Conference, or Call Splitting.
  In Consultation Hold, the original call is treated as if it is on hold, allowing the extension to call a third party all on one line. In Call Hold, the party on hold and the third party are connected to the extension using separate lines.
- When the extension user is having a conversation with one party, the other party is in consultation hold.

User Manual References

1.4.3 Talking to Two Parties Alternately (Call Splitting)

2.13.4 Music on Hold

Description

Music can be played to a party that has been put on hold. The following audio sources are available:

a. External music source
b. User-supplied audio file
c. Tone

The audio source for Music on Hold is selected from either a BGM number (1 to 8) or the built-in tone. BGM can be either an external music source or a user-supplied audio file. The following table shows which audio sources can be assigned to which BGM numbers:

[BGM Number and the Music Source]

BGM No.	Music Source
1	User audio data
2	User audio data
3	External music port 1
4	External music port 2
5	External music port 3
6	External music port 4
7	External music port 5
8	External music port 6

For tenant users, each tenant can select one of the BGMs or the tone to use for Music on Hold.

Conditions

[General]

- **Hardware Requirement:** User-supplied music source (when an external music source is assigned)
- **Volume Control**
  It is possible to change the volume of an internal and/or external music source.
- For tenants, the type of call determines which tenant's music source is used, as follows:

Type	Music Source
Incoming Intercom Calls/Outgoing Calls	Selected based on the tenant setting to which the extension user belongs.
Incoming Trunk Calls	Selected based on the tenant setting of the distribution method (DIL/DID/DDI).

- Even if an external music source or a user-supplied audio file is selected for Music on Hold, an IP-PT or SIP extension user who is put on hold by another extension will hear the telephone's hold tone and not the specified Music on Hold.

[User-supplied audio files]

- User-supplied audio files are uploaded via Web Maintenance Console. Audio files must meet the following specifications:

  - Format: WAV
  - Size: 40 MB or less
  - Length: 4 minutes or less
- Initially, a preinstalled audio file is set as the audio source for BGM 1. Through system programming, this file can be removed or replaced like any other BGM audio file. However, if the PBX is reinitialised, this preinstalled audio file is set to BGM 1 again.

Installation Manual References

4.10 Connection of Peripherals

PC Programming Manual References

5.2 System Control—MOH
10.2 PBX Configuration—[2-2] System—Operator & BGM
10.11.1 PBX Configuration—[2-11-1] System—Audio Gain—Paging/MOH—◆ Internal MOH—MOH1-2 (Music On Hold 1-2)

PT Programming Manual References

[711] Music on Hold

Feature Guide References

2.30.1 Background Music (BGM)
5.1.3 Tenant Service

2.14 Conference Features

2.14.1 Conference Features—SUMMARY

Description

A conference call allows a conversation between three or more parties simultaneously. The following features are available to establish a conference call:

Feature	Description & Reference
Conference	During a two-party conversation, an extension user can add other parties to establish a conference call with up to eight parties. → 2.14.2 Conference
Executive Busy Override	An extension user can interrupt an existing call to establish a three-party conference call. → 2.10.2 Executive Busy Override
Privacy Release	During a conversation with an outside party on the S-CO button, a PT/PS user can allow another extension to join the conversation. → 2.14.3 Privacy Release

Conditions

- One conference call supports a maximum of 8 parties.
- The maximum number of parties that can be engaged in conference calls simultaneously differs depending on the type of PBX:
  – KX-NS300: max. 32 parties
  Parties are counted at the PBX where the conference originated.
- It is possible to select which of the following devices to use when making a conference.
  – PBX MPR
  – Optional DSP card
  Better sound quality can be achieved by selecting the optional DSP rather than the PBX MPR. However, in this case DSP resources will be used.
- **DSP Resource Usage**
  A conference call requires a certain number of DSP resources. If all DSP resources are in use, this operation cannot be performed. To ensure a minimum level of performance, DSP resources can be reserved for conference calls. (→ 5.5.4 DSP Resource Usage)

PC Programming Manual References

9.37 PBX Configuration—[1-5] Configuration—DSP Resource

2.14.2 Conference

Description

An extension user can establish a conference call by adding additional parties to an already existing two-party conversation. This PBX supports three-party through eight-party conference calls. Conferences with more than four parties are only possible when a PT or PS user originates the conference.
Unattended Conference:
The conference originator can leave the conference and allow other parties to continue. Establishing an Unattended Conference allows the originator to return to the conference. Unattended Conferences can only be established by PT and PS users.

Conditions

- When an extension is establishing a conference call the original party is put on hold.
- **CONF (Conference) Button**
  For a PT/PS which does not have the CONF button, a flexible button can be customised as the Conference button.
- **Unattended Conference Call Duration**
  The length of time that a conference call can remain unattended is restricted by the following timers:
  – Callback Start Timer
  – Warning Tone Start Timer
  – Disconnect Timer
  These timers behave and operate according to the following chain of events:
  1. When the unattended conference is established, the Callback Start Timer will begin.
  2. When the Callback Start timer expires, the Unattended Conference originator's extension will start to receive a callback ringing from the PBX and the Warning Tone Start Timer begins.
  3. When the Warning Tone Start Timer expires, the remaining parties of the conference will start to hear a warning tone, the callback ringing will continue to be heard at the Unattended Conference originator's extension, and the Disconnect Timer begins.
  4. When the disconnect Timer expires, the conference is disconnected.

  If the Unattended Conference originator returns to the conference before the line is disconnected, all timers are cleared.
- If the originator of a conference with two trunks leaves the conference, the call can become a trunk-to-trunk call, if enabled through system programming.
  – When a trunk-to-trunk call is established, the call will end when the Trunk-to-Trunk Call Duration timer has elapsed (→ 2.11.8 Trunk Call Limitation). The timer applied is that of the trunk group containing the trunk being used for the call immediately preceding the conference. The timer cannot be extended.
  – If both trunks are analogue trunks, the end of the trunk-to-trunk call may not be detected. For this reason, if analogue trunks are used, it is not recommended to enable the establishment of trunk-to-trunk calls after a conference call through system programming.
- When a KX-UT series SIP phone is used to originate a conference and one of the other parties leaves the conference, the KX-UT series SIP phone user will still be able to talk to the remaining party but will not be able to use the standard Hold feature.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf
→◆ Unattended Conference—Recall Start Timer (x60s)
→◆ Unattended Conference—Warning Tone Start Timer (s)
→◆ Unattended Conference—Disconnect Timer (s)
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—CO & SMDR—◆ Transfer to CO

10.9 PBX Configuration—[2-9] System—System Options—Option 2—◆ CO - CO Call Limitation—After Conference
10.9 PBX Configuration—[2-9] System—System Options—Option 3
→◆ Confirmation Tone—Tone 4-1 : Start Conference
→◆ Confirmation Tone—Tone 4-2 : Finish Conference
→◆ Echo Cancel—Conference
10.9 PBX Configuration—[2-9] System—System Options—Option 8—◆ Conference Group—Maximum Number of Speakers During a Conference Group Call
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Type

Feature Guide References

2.21.2 Flexible Buttons
6.1 Capacity of System Resources

User Manual References

1.4.5 Multiple Party Conversation

2.14.3 Privacy Release

Description

By default, all conversations which take place on trunks, extension lines and doorphone lines are protected by privacy (**Automatic Privacy**).
Privacy Release allows a PT/PS user to suspend Automatic Privacy for an existing trunk call on the S-CO button in order to establish a three-party call.
System programming is required to enable or disable this feature.

Conditions

- **S-CO Button**
  A flexible button can be customised as the S-CO button.
- **Privacy Release Time**
  Privacy is released for five seconds to allow the conversation to be joined.
- This feature overrides Data Line Security (→ 2.11.5 Data Line Security) and Executive Busy Override Deny (→ 2.10.2 Executive Busy Override).

PC Programming Manual References

10.9 PBX Configuration—[2-9] System—System Options—Option 1—◆ PT Operation—Privacy Release by SCO key
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Type

Feature Guide References

2.21.2 Flexible Buttons

User Manual References

1.4.5 Multiple Party Conversation

2.15 Conference Group Call Features

2.15.1 Conference Group Call

Description

Conference group calling allows you to call a pre-determined group (conference group) of parties simultaneously. Each party that answers the call joins the conference.
An extension user can call a conference group of up to 31 parties to establish a conference call, for a maximum of 32 participants. During a conference group call, the caller can restrict the ability of other members to speak.
The following telephones will answer automatically and play the announcement through the telephone's speaker, even if Hands-free Answerback (→ 2.4.4 Hands-free Answerback) is not enabled for the extension:

- PTs
- KX-TCA175 (PS)
- KX-TCA275 (PS)
- KX-TCA185 (PS)
- KX-TCA285 (PS)
- KX-TCA385 (PS)

Broadcast Mode

When Broadcast Mode is enabled through system programming, an extension user can call a conference group of up to 31 call members to make a voice announcement. Members can listen to the announcement by answering the call.
During the announcement, the voices of members will not be heard. However, the caller can allow up to 31 specific members to speak, making a conference call. This conversation can be heard by the other members.
[Push-to-talk feature for PT/SLT/PS users]
PT/SLT/PS users that are members of a Broadcast Mode call can enable their own ability to speak by pressing any of their dial keys during the Broadcast Mode announcement. This feature can be disabled through system programming.
Broadcast Mode can be used to broadcast an announcement to multiple PS users. With Automatic Answer enabled, the PS users will all hear the announcement through their headset or the PS's hands-free speaker. Then, any PS user can respond by using push-to-talk to enable their ability to speak, and their reply is heard by all broadcast members.
For example, a central operator could make a Broadcast Mode call paging PS users working throughout a building for assistance, and an available PS user can reply. The reply is heard by the other PS users as well. This makes it easy to assign and coordinate tasks with multiple staff members that are frequently moving throughout a building.

Conference Group Call Control

During a conference group call, the caller can restrict or allow members' ability to speak, and can remove members from the call using the following buttons. These buttons will function irrelevant of the Conference Group Call mode. Pressing the other buttons during the conversation will be ignored.

Note

The operation of these buttons during a conference group call is different from the operations for the Conference feature (→ 2.14 Conference Features).

Button	Function
DSS	Disables or enables the corresponding member's ability to speak.

Button	Function
CONF (Conference)	Establishes a conversation with the current members in the order assigned in the conference group. Pressing this button again will add the next available member in the group to the conversation.
TRANSFER	Removes the member who joined the conversation last. The member can still listen to the announcement.
FLASH/RECALL (Flash/Recall mode)	Removes the member who joined the conversation last. The member will be disconnected from the conference group call and hear a reorder tone.
SP-PHONE	Enables a hands-free conversation.

A member extension can inform the caller that he wants to speak or join the conversation by sending a notification. The caller will hear a notification tone and the requesting extension's information will be shown on the display for five seconds.

Conference Groups

Eight conference groups can be programmed, and a maximum of 31 members can be assigned to each group. The available destinations as members of the conference group are as follows:

Destination	Availability
Wired Extension (PT/SLT/SIP Extension/ISDN Extension)	✓
PS	✓
Incoming Call Distribution Group	
PS Ring Group	
UM Group	
VM Group (DTMF/DPT)	
External Pager (TAFAS)	
DISA	
Analogue/ISDN Remote Maintenance	
Idle Line Access no. + Phone no.	✓
Trunk Group Access no. + Trunk Group no. + Phone no.	✓
Other PBX Extension (TIE with no PBX Code)	✓*1
Other PBX Extension (TIE with PBX Code)	✓*1

*1 Only available when the networking type of the trunk is assigned as private.

Join After Time Out

When conference group members do not answer a conference group call within the preprogrammed time limit, the member's telephones will stop ringing. However, even after the time limit has expired, members can join the conference.
Additionally, extension users not registered in the called conference group can join a conference after it has started. Outside callers using DISA and TIE line users can also join a conference after accessing their extension using Walking COS.

An entry code can also be specified to restrict access to the call. The entry code can be set when the caller initiates the conference group call. If an entry code is to be set, usually the caller will inform call participants in advance.

[Starting a Conference Group Call using Hands-free Answerback]

Hands-free Answerback can be enabled for conference group member extensions. By initiating a conference group call with a conference group that includes an extension that has Hands-free Answerback (→ 2.4.4 Hands-free Answerback) enabled, such as a softphone, the call can be automatically answered and the conference begins with only one participant (the initiator of the conference group call). Then, up to 6 participants can use Join After Time Out to join the conference. For example, a manager can organise a meeting where all the members are calling from cellular phones and use Join After Time Out to join the conference.

Conditions

- Only extensions that are permitted by COS programming can originate conference group calls.
- Conference Group Call Control features are unavailable if an SLT or SIP extension is used to initiate the call.
- Users of PSs other than the KX-TCA175/KX-TCA275/KX-TCA185/KX-TCA285/KX-TCA385 may be able to enable the automatic answering of calls for this feature by changing their PSs' settings. For details, refer to the operating instructions of the PS.
- After one conference group member answers the call, the conference or announcement is established.
- If no members answer the call within the preprogrammed time limit, the caller will hear a busy tone.
- The caller will hear a confirmation tone every time a member answers the call.
- When the originating caller of a conference group call goes on hook, the call ends and all participating members will be disconnected.
- The conference group call will reach a member extension regardless of settings such as Call Forwarding (except DND).
- If a member extension is busy and has Call Waiting for trunk calls activated when a conference group call is made, a call waiting tone will be sent to the extension.
- For members who use a KX-TCA175/KX-TCA275/KX-TCA185/KX-TCA285/KX-TCA385 PS, when automatic answer is enabled for the conference group and the extension is busy when a conference group call is made, the PS will automatically answer the call if the member goes on-hook while the conference call is still ringing. A PT will ring instead of answering the call.
- The call information of the caller (not members) will be recorded on SMDR.
- A caller cannot make a conference group call with a call on hold.
- Call Pickup is not available for a conference group call. (→ 2.4.3 Call Pickup)
- The conference group call will not reach members when:
  - the member extension has set DND for intercom calls.
  - the member extension is a PS in Wireless XDP Parallel Mode. (→ 5.2.4.5 Wireless XDP Parallel Mode)
- If a conversation has reached the maximum number of participants, the Join After Time Out feature cannot be used to join the conversation.
- If a member uses push-to-talk to enable the ability to speak during a Broadcast Mode call, the member cannot disable this ability. They can mute their microphone manually, or the originator of the call can use conference group call control to disable their ability to speak.
- Since each PS requires one wireless channel, note your PBX's wireless capacity when assigning multiple PSs to a conference group.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Conference Group Call—Ring Duration (s)

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Conference Group Call Operation
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Miscellaneous—◆ Conference Group Call Operation
10.9 PBX Configuration—[2-9] System—System Options—Option 8—◆ Conference Group—Maximum Number of Speakers During a Conference Group Call
11.9 PBX Configuration—[3-9] Group—Conference Group
→◆ Broadcast Mode
→◆ Ability to Talk
→◆ Automatic Answer w/o Extension Setting
11.9.1 PBX Configuration—[3-9] Group—Conference Group—Member List

Feature Guide References

2.4.4 Hands-free Answerback
6.1 Capacity of System Resources

User Manual References

1.7.3 Making a Conference Group Call

2.16 Direct Inward System Access (DISA) Features

2.16.1 Direct Inward System Access (DISA)

Description

An outside 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, operator or any floating extensions (e.g., an external pager for TAFAS).
- Calling an outside party via the PBX.
- Operating some PBX remote features (e.g., FWD)

DISA Intercept Routing—No Dial

If the caller fails to dial any digits within a preprogrammed time period (DISA 1st Dial Time for Intercept) after hearing the outgoing message (OGM), one of the following can be selected through system programming:

a. Disable: The call will be terminated.
b. Operator: The call will be redirected to the operator.
c. AA-0, AA-9: The call will be redirected to the destination assigned to that AA number.

→ 10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf—◆ DISA—No Dial Intercept Timer (s)

→ 13.3.1 PBX Configuration—[5-3-1] Optional Device—Voice Message—DISA System—Option 1—◆ DISA Intercept—Intercept when No Dial after DISA answers

DISA Built-in Automated Attendant Number (DISA AA Service)

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. It is also possible to assign other DISA floating extension numbers as the destination (**Multistep DISA AA Service**).
If the caller dials a second digit within a preprogrammed time period (DISA 2nd Dial Time for AA), the DISA AA service is not employed.

→ 13.3.2 PBX Configuration—[5-3-2] Optional Device—Voice Message—DISA Message—◆ 1 Digit AA Destination (Extension Number)—Dial 0–9

→ 10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf—◆ DISA—2nd Dial Timer for AA (s)

Outgoing Message (OGM)

When a call arrives on a DISA line, a prerecorded DISA message will greet and guide the caller.
Any extension assigned as the manager can record outgoing messages (OGMs). (→ 2.30.2 Outgoing Message (OGM))

[Programming Example]

Outgoing Message (OGM) No.	Floating Extn. No.*1	Automated Attendant No.*2										Busy/DND Message No.*3
		0	1	2	3	4	5	6	7	8	9	
01	501	100	301	200	103	202	101	102	400	104	205	04
02	502											05
:	:	:	:	:	:	:	:	:	:	:	:	:

*1 → 13.3.2 PBX Configuration—[5-3-2] Optional Device—Voice Message—DISA Message—◆ Floating Extension Number
The default floating extension number depends on the value specified for **Numbering Plan** in Easy Setup.

→ 5.4.1 Easy Setup Wizard

*2 → 13.3.2 PBX Configuration—[5-3-2] Optional Device—Voice Message—DISA Message—◆ 1 Digit AA Destination (Extension Number)—Dial 0–9

*3 → 13.3.2 PBX Configuration—[5-3-2] Optional Device—Voice Message—DISA Message—◆ Busy / DND Message No.

DISA Security Mode and Available Features

If the DISA AA service is not employed, the caller may access the PBX features by entering the feature numbers. To prevent others from accessing the PBX features, it is possible to assign DISA security.

→ 13.3.1 PBX Configuration—[5-3-1] Optional Device—Voice Message—DISA System—◆ DISA Security—DISA Security Mode

The features available depend on the preprogrammed DISA security mode as follows:

Security Mode	Intercom Call	TIE Line Call		Trunk Call
		Without PBX Code	With PBX Code	
All Security				
Trunk Security	✓	✓		
No Security	✓	✓	✓	✓*1

✓: Available

*1 If trunk call is available, Account Code Entry (→ 2.5.4.3 Account Code Entry) is also available.

Note

DISA AA service and Operator Call (→ 5.1.5 Operator Features) are available for any security mode.

Security Mode Override by Verification Code Entry

If the caller performs Verification Code Entry (→ 2.7.6 Verification Code Entry) while hearing a DISA message, the security mode can be temporarily changed to No Security mode.

Entry method:

Verification Code Entry feature number + ✳ + verification code + verification code PIN

After changing mode, the new mode remains in force for the duration of the call.

DISA Intercept Routing—Busy

If the first destination called by the outside party is busy, the call is redirected as follows:

a. The call is redirected to the Intercept Routing—Busy destination assigned to the first destination.

b. If an Intercept Routing—Busy destination is not assigned to the first destination and a prerecorded DISA Busy Message is assigned, the caller will hear the DISA Busy Message.

c. If neither an intercept destination nor a DISA Busy Message is assigned, the caller will hear a busy tone.

→ 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Intercept Destination—◆ Intercept Destination—When Called Party is Busy

DISA Intercept Routing—DND

If the destination called by the outside party is in DND mode and Idle Extension Hunting is not available, one of the following can be selected through system programming:

a. **Busy Tone:** The caller will hear a busy tone.

b. **Enable:** DND will redirect the call to the preprogrammed destination on an extension basis.

c. **OGM:** An outgoing message (OGM) will be sent to the caller. The message for DND mode can be assigned for each outgoing message (OGM) which has a DISA floating extension number.

→ 13.3.1 PBX Configuration—[5-3-1] Optional Device—Voice Message—DISA System—◆ DISA Intercept—Intercept when destination through DISA sets DND

DISA Intercept Routing—No Answer

If a destination is not available to answer a DISA call within a preprogrammed time period (DISA Intercept time) after the call is reached, the call will be redirected to the programmed destination by the Intercept feature. If the intercept destination is not available to answer the call within a preprogrammed time period (DISA Disconnect Time after Intercept) after the DISA Intercept time expires, the call will be disconnected.
10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf
→ ◆ DISA—Intercept Timer—Day (s), Lunch (s), Break (s), Night (s)
→ ◆ DISA—Disconnect Timer after Intercept (s)
→ 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Intercept Destination—◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night
→ 12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Intercept Destination—◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night

Walking COS Through DISA

If the caller performs Walking COS (extension number and PIN entry) while hearing a DISA message, the security mode can be temporarily changed to No Security mode (→ 2.7.5 Walking COS). After performing Walking COS, the following features are available, using the settings of the specified extension:

- Intercom call
- TIE line call
- Trunk call
- Call Forwarding (FWD)/Do Not Disturb (DND) setting (→ 2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features)
- Incoming Call Distribution Group Log-in/Log-out (→ 2.2.2.7 Log-in/Log-out)
- Absent Message setting (→ 2.20.2 Absent Message)
- Extension Dial Lock (→ 2.7.3 Extension Dial Lock)
- Time Service Switching Mode (→ 5.1.4 Time Service)

Note

When making a trunk call using Walking COS through DISA, the CLIP number for that call will be that of the extension seized by Walking COS. (→ 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP))

DISA Automatic Walking COS

Registered outside destinations such as cellular phones can be automatically recognised as PBX extensions when calling through DISA. When the Caller ID of a received trunk call matches an entry in the System Speed Dialling Table, the calling telephone is given Walking COS authorisation as assigned to the corresponding CLI destination extension. Therefore, the "CLI destination" setting in the System Speed Dialling Table is used here to specify the target extension that the calling telephone will be recognised as for Walking COS.
[Programming Example of DIL Table]

Trunk No.	CLI			Destination		
	Day	Lunch	...	Day	Lunch	...
01	Disable	Disable	...	5801 (DISA)	5801 (DISA)	...
:	:	:	:	:	:	:

"CLI" must be set to Disable, to allow incoming calls to be received by DISA.
[Programming Example of System Speed Dialling Table]

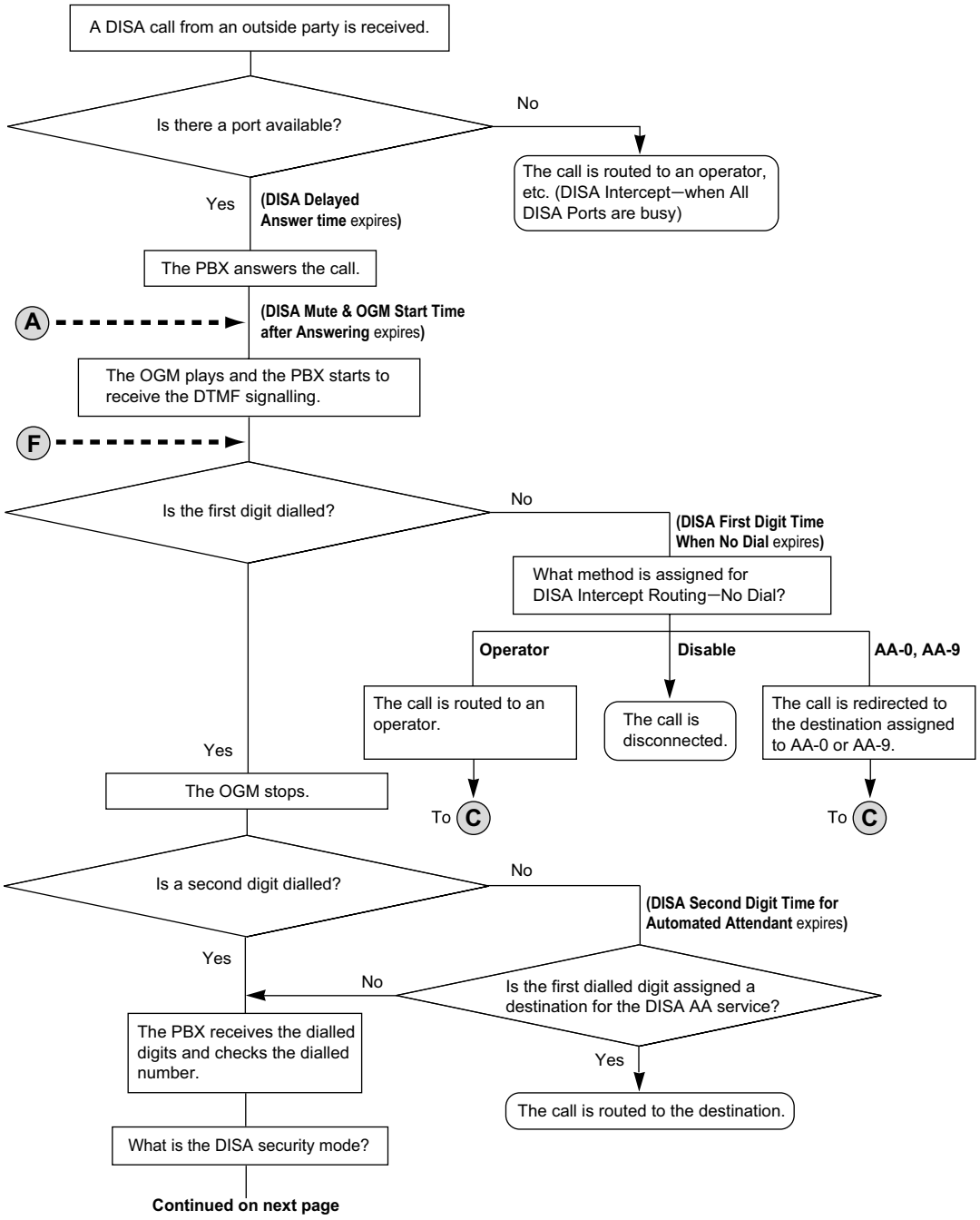
Location	Name	Trunk Access + Telephone Number	CLI Destination
000	J. Smith	912341115678	200
001	:	:	:
:	:	:	:

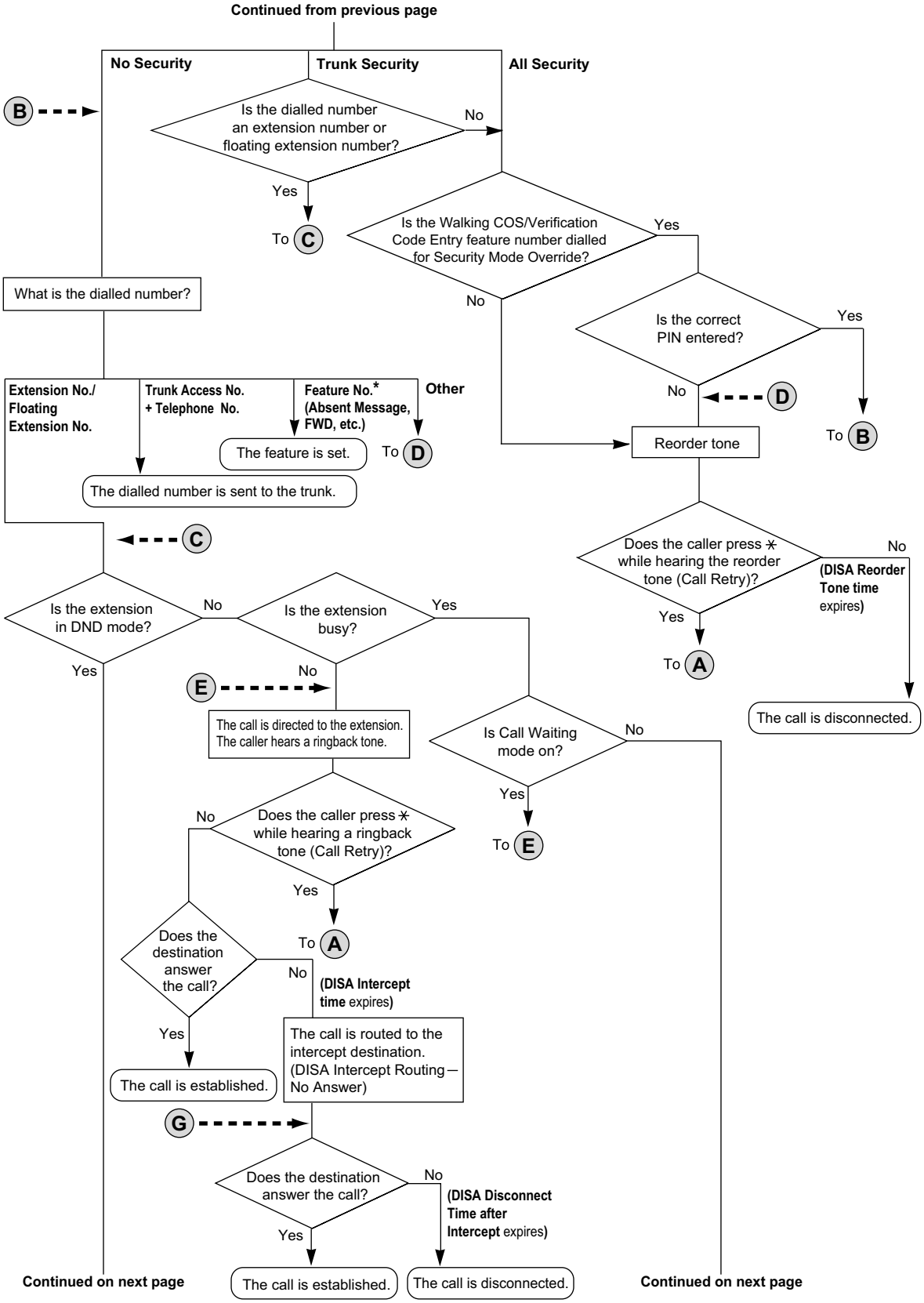
In this example, calls received on trunk 01 are routed to the DISA OGM with floating extension number 5801. If the number of the received call (after modification according to the Caller ID table) is "12341115678", the call originator is recognised as extension 200, and the Walking COS feature is automatically activated. System programming is required to enable this feature.

SMDR

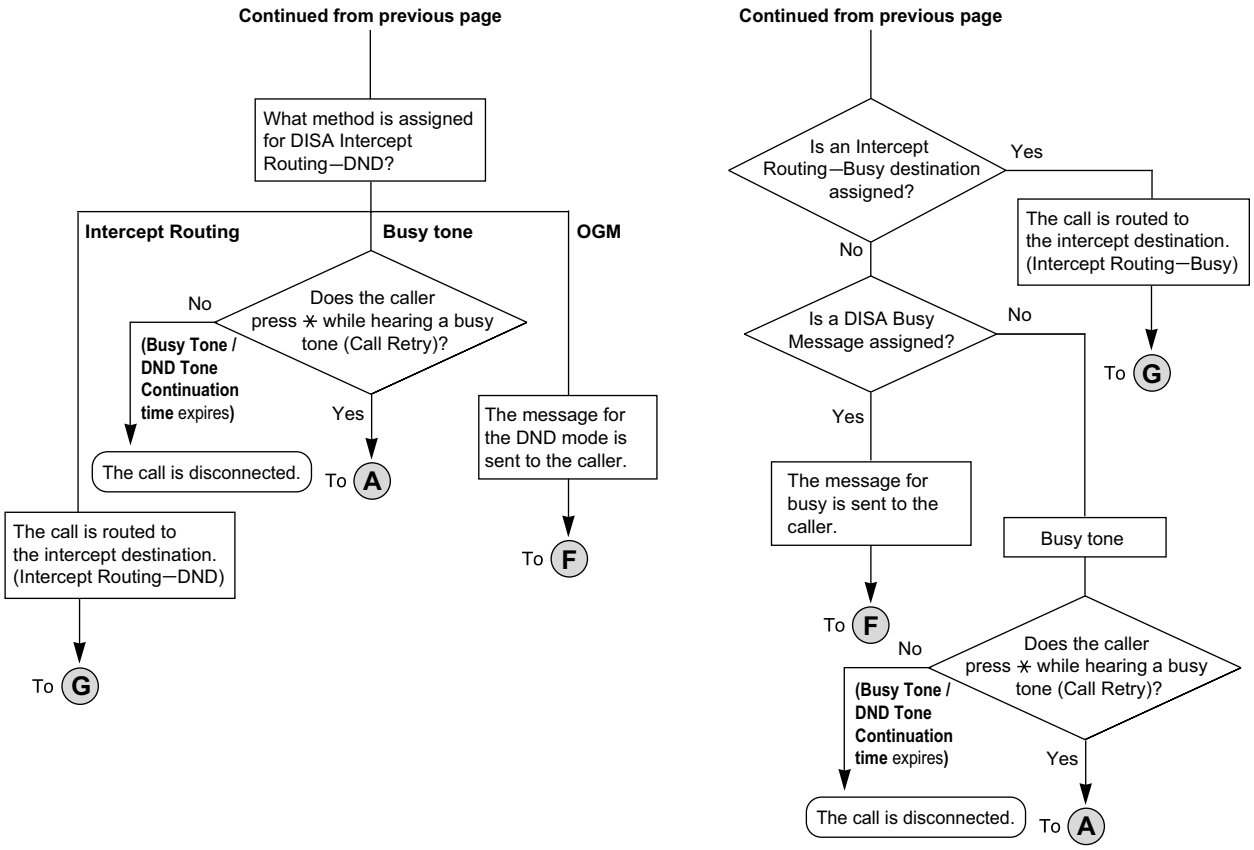
The call information for DISA is recorded as the one of the DISA floating extension numbers.
(→ 2.22.1.1 Station Message Detail Recording (SMDR))

[Flowchart]


A DISA call from an outside party is received.
Is there a port available?
No
The call is routed to an operator, etc. (DISA Intercept—when All DISA Ports are busy)
Yes
(DISA Delayed Answer time expires)
The PBX answers the call.
A
(DISA Mute & OGM Start Time after Answering expires)
The OGM plays and the PBX starts to receive the DTMF signalling.
F
Is the first digit dialled?
No
(DISA First Digit Time When No Dial expires)
What method is assigned for DISA Intercept Routing—No Dial?
Operator
Disable
AA-0, AA-9
The call is routed to an operator.
The call is disconnected.
The call is redirected to the destination assigned to AA-0 or AA-9.
To C
To C
Yes
The OGM stops.
Is a second digit dialled?
No
(DISA Second Digit Time for Automated Attendant expires)
Yes
No
Is the first dialled digit assigned a destination for the DISA AA service?
Yes
The PBX receives the dialled digits and checks the dialled number.
The call is routed to the destination.
What is the DISA security mode?
Continued on next page



* Feature numbers are available only when the Walking COS feature is used.



Conditions

CAUTION

There is a risk that fraudulent telephone calls will be made 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. Enabling DISA security (Trunk Security or All Security).
b. Keeping passwords (verification code PINs/extension PINs) secret.
c. Selecting complex, random PINs that cannot be easily guessed.
d. Changing PINs regularly.

- **Maximum simultaneous OGM Channel**
  If no DSP card is installed, the maximum number of simultaneous OGM channels is 2. If a DSP card is installed maximum simultaneous OGM channels is 64. However, in this case DSP resources will be used.
- **DSP Resource Usage**
  A DISA call requires a certain number of DSP resources. If all DSP resources are in use, this operation cannot be performed. To ensure a minimum level of performance, DSP resources can be reserved for conference calls. (→ 5.5.4 DSP Resource Usage)
- **DISA Delayed Answer Time**
  It is possible to set the Delayed Answer time so that the caller will hear a ringback tone within a preprogrammed time period first before hearing an outgoing message (OGM).
  → 10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf—◆DISA—Delayed Answer Timer (s)
- **Call Retry**

While hearing a ringback, reorder, or busy tone, retrying the call is possible by pressing "✳". System programming selects whether pressing "✳" during a trunk-to-trunk conversation returns to the DISA top menu or sends a DTMF tone.

- **DISA Mute Time**
  It is possible to set the Mute time until the outgoing message (OGM) plays and the PBX starts to receive the DTMF signalling after the caller reaches the DISA line.
  → 10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf—◆ DISA—Mute & OGM Start Timer after answering (s)
- **End of Call Detection**
  If a call through DISA is routed to a trunk, DISA can be used to detect the end of the call. This function can be disabled through system programming. If disabled, DISA is released when the trunk-to-trunk connection is made.
  The following three types of tone detection can be enabled for each trunk group to disconnect a trunk-to-trunk call via DISA.
  – Silence Detection
    → 11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Tone Detection—◆ DISA Tone Detection—Silence
  – Continuous Signal Detection
    → 11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Tone Detection—◆ DISA Tone Detection—Continuous
  – Cyclic Signal Detection
    → 11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Tone Detection—◆ DISA Tone Detection—Cyclic
- **Trunk-to-Trunk Call Duration Limitation**
  For a call between two outside parties, even if end of call detection cannot be performed, the call can be disconnected by a system timer. (→ 2.11.8 Trunk Call Limitation) If the timer expires, the line will be disconnected unless the originating caller extends the time by sending any DTMF signalling. The caller can prolong the call duration within the preprogrammed time period and preprogrammed number of times.
  10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf
  → ◆ DISA—CO-to-CO Call Prolong Counter
  → ◆ DISA—CO-to-CO Call Prolong Time (x60s)
- **Automatic DISA Activation**
  DISA can be set through system programming to automatically activate for the following types of trunk-to-trunk call, to enable detection of the end of the call.
  – When a trunk call is forwarded to another trunk
  – When a trunk call is transferred to another trunk
  – When a trunk call to an incoming call distribution group is answered by an outside destination member

  Before the call is made, the PBX confirms that a DISA port is available. If no DISA ports are available, the call is not routed to a trunk. For transferred calls or calls to an ICD Group, if the DISA port has become unavailable when the trunk-to-trunk conversation is actually established, the call is established without DISA.
  When using this feature, the Trunk-to-Trunk Call Limitation timer should be enabled. In addition, prolonging the call through DTMF signalling is not available.
- **DISA Call Transfer from Outside Destination**
  An outside party such as a cellular phone can transfer a trunk call to an extension (including TIE) or an outside party by pressing "#" + extension number (including TIE) or an outside party's number, if DISA is connected by the Automatic DISA Activation feature. This feature can be enabled or disabled through system programming.
  It is also possible to establish a Conference call (→ 2.14 Conference Features), perform Call Splitting (→ 2.13.3 Call Splitting), and page with a call on hold to transfer the call (→ 2.17.1 Paging).
  – DISA security mode should be set to No Security.
  – If the called extension does not answer, is busy, or is in DND mode, the DISA Intercept feature operates.
  – The party on hold can use the Call Retry feature.

– If the destination trunk supports End of Call Detection, a paging call can be made after dialling "#".
– If the transferred call is forwarded to another outside destination, COS settings are ignored.
  • If the cellular phone's number is registered in system speed dialling, its COS will be the COS of the extension specified as the CLI destination.
  • If the cellular phone's number is not registered in system speed dialling, its COS will be the COS of the trunk that the transferrer is using.
  • Operation is as follows, depending on the "10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—CO & SMDR—◆ Transfer to CO" setting and the availability of End of Call Detection on the destination trunk:
    – Case 1: **Transfer to CO** is enabled
      Transfer is available.
    – Case 2: **Transfer to CO** is disabled and the destination trunk supports End of Call Detection
      A reorder tone is heard. (Consultation hold can be released by pressing "#".)
    – Case 3: **Transfer to CO** is disabled and the destination trunk does not support End of Call Detection
      Transfer is cancelled, and the conversation returns to the trunk on hold.
– If the call is transferred to an ICD group, the call will wait in a queue until answered, but the Queuing Time Table will not function.
– When using this feature, do not use the Executive Busy Override feature to interrupt the established call.

Redial with DTMF "✳" when receiving a trunk call (before the transfer destination—cellular phone—answers).

- It is possible to redial by using the DTMF tone "✳" before the transfer destination trunk (cellular phone) answers.
- If the transfer destination (cellular phone) is a line that does not support answer notification, outgoing dial completion is considered as answering.
- If the destination trunk supports End of Call Detection, screened transfer is available. If the transferring party dials "#" while talking to the called party, the call can be put back on consultation hold.
- If the destination trunk does not support End of Call Detection, the call is transferred unscreened.

- **DISA Call Transfer to Outside User**

When a call is received through DISA and the receiving extension is set to forward the call to an extension (including over a TIE connection) or an outside destination, the call can be forwarded automatically to an outside telephone number if the following conditions are met:

- An outside telephone number (e.g., a cellular phone number) is registered as the forward destination of the receiving extension.
- The forward destination's telephone number is registered in the System Speed Dial of the receiving extension.
- The forward destination's telephone number is set in the CLI Destination of the receiving extension.
- Automatic Walking COS is enabled for the receiving extension.

After a conversation is established with the forward destination (e.g., a cellular phone number), it is possible to establish a Conference call, perform Call Splitting, and page with a call on hold to transfer the call. This is the same as DISA Call Transfer from Outside Destination. For details, see "DISA Call Transfer from Outside Destination".

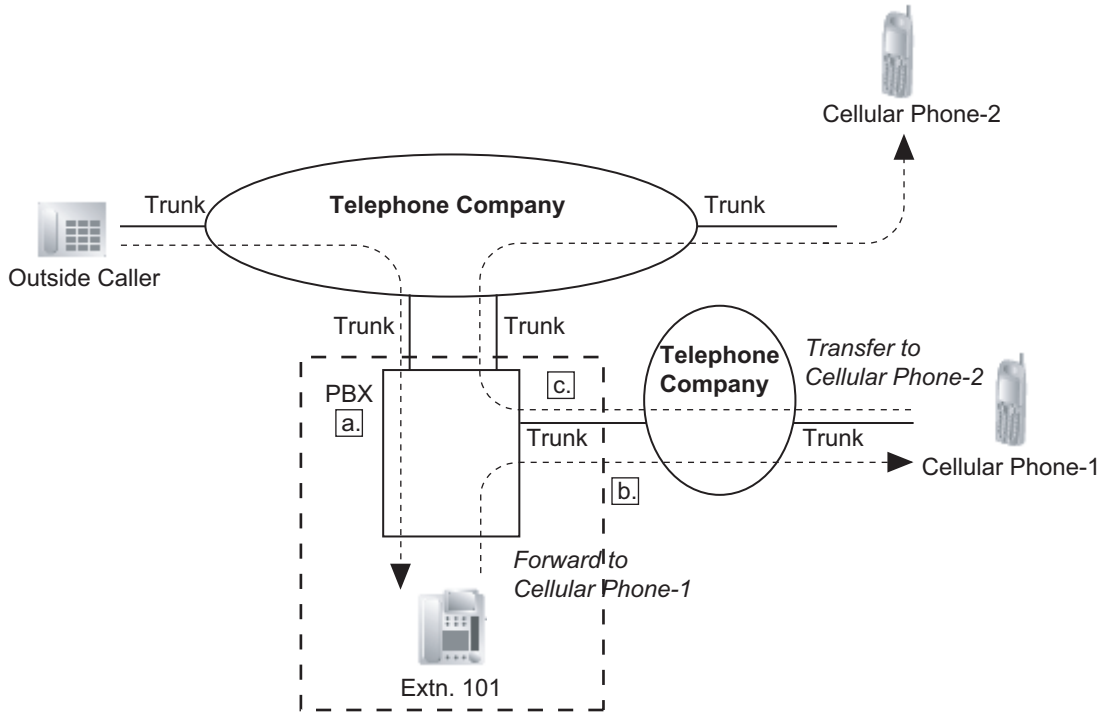
[Example]

a. Outside Caller calls Extn. 101 through DISA.
b. Extn. 101 forwards the call to Cellular Phone-1.
Outside Caller establishes a conversation with Cellular Phone-1.
c. Cellular Phone-1 presses "#" to put the conversation on hold, and then transfers the call to Cellular Phone-2.
At this point, the CLIP information shown on Cellular Phone-2 may be one of the following:

Case 1: When Extn. 101 forwarded the call, Automatic Walking COS was performed with Cellular Phone-1's telephone number.
– Displayed CLIP: Same as if Extn. 101 made a trunk call.

Case 2: When Extn. 101 forwarded the call, Automatic Walking COS was not performed, and ◆ Send CLIP of CO Caller—when call is transferred to CO (CLIP of Held Party) is set to **Enable**.
– Displayed CLIP: Outside Caller's telephone number and name.

Case 3: When Extn. 101 forwarded the call, Automatic Walking COS was not performed and ◆ Send CLIP of CO Caller—when call is transferred to CO (CLIP of Held Party) is set to **Disable**.
– Displayed CLIP: The CLIP set for the line that Cellular Phone-1 used to transfer the call.



- **DISA Reorder Tone Duration**
  It is possible to set the DISA Reorder Tone Duration time. This specifies the length of time that a reorder tone will be sent to the caller. When the timer expires, the call will be disconnected. Call Retry is possible during the DISA Reorder Tone Duration time.
  → 10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf—◆ DISA—Reorder Tone Duration (s)
- **Call Deny**
  Extensions can deny DISA calls on a COS basis.
  → 10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Optional Device & Other Extensions—◆ Accept the Call from DISA
- **Verification Code PIN Lock/Extension PIN Lock**
  If the wrong PIN is entered three times, the line will be disconnected. If the wrong PIN is entered a preprogrammed number of times successively, that extension or verification code will become locked, and even entering the correct PIN will not unlock it. Only an extension assigned as the manager can unlock it. In this case, the PIN will be unlocked and cleared.
  → 10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Extension PIN—Lock Counter
- **DISA Automatic Walking COS**

KX-NSE101, KX-NSE105, KX-NSE110 or KX-NSE120 (Activation Key for Mobile Extension) is required to use this feature. One activation key is required for each extension that will use this feature.

- Each outgoing message (OGM) can be assigned a name through system programming for programming reference.
  → 13.3.2 PBX Configuration—[5-3-2] Optional Device—Voice Message—DISA Message—◆ Name

Installation Manual References

5.4.1 Easy Setup Wizard

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf
→◆ DISA—Delayed Answer Timer (s)
→◆ DISA—Mute & OGM Start Timer after answering (s)
→◆ DISA—No Dial Intercept Timer (s)
→◆ DISA—2nd Dial Timer for AA (s)
→◆ DISA—Intercept Timer—Day (s), Lunch (s), Break (s), Night (s)
→◆ DISA—Disconnect Timer after Intercept (s)
→◆ DISA—CO-to-CO Call Prolong Counter
→◆ DISA—CO-to-CO Call Prolong Time (x60s)
→◆ DISA—Progress Tone Continuation Time before Recording Message (s)
→◆ DISA—Reorder Tone Duration (s)
10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Extension PIN—Lock Counter
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Optional Device & Other Extensions—◆ Accept the Call from DISA
11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings
→Tone Detection—◆ DISA Tone Detection—Silence
→Tone Detection—◆ DISA Tone Detection—Continuous
→Tone Detection—◆ DISA Tone Detection—Cyclic
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Intercept Destination—◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Intercept Destination—◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night
13.3.1 PBX Configuration—[5-3-1] Optional Device—Voice Message—DISA System
13.3.2 PBX Configuration—[5-3-2] Optional Device—Voice Message—DISA Message

PT Programming Manual References

[209] DISA Delayed Answer Time
[210] DISA Trunk-to-Trunk Call Prolong Time
[211] DISA Intercept Time
[475] DISA Silence Detection
[476] DISA Continuous Signal Detection
[477] DISA Cyclic Signal Detection
[604] Extension Intercept Destination
[730] Outgoing Message (OGM) Floating Extension Number
[731] Outgoing Message (OGM) Name
[732] DISA Security Mode

Feature Guide References

2.1.1.5 Intercept Routing
2.1.1.6 Intercept Routing—No Destination
2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features
5.1.1 Class of Service (COS)
5.5.8 Floating Extension

User Manual References

1.2.8 To Access Another Party Directly from Outside (Direct Inward System Access [DISA])
1.2.9 Setting Your Telephone from Another Extension or through DISA (Remote Setting)

2.16.2 Automatic Fax Transfer

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 (→ 2.30.2 Outgoing Message (OGM)). At the same time, the PBX begins fax signal detection. If a fax signal is detected, the PBX recognises 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.

[Available Automatic Fax Transfer Destinations]

Destination	Availability
Wired Extension (PT/SLT/ISDN Extension)	✓
PS	✓*1
Incoming Call Distribution Group	✓
SIP Extension	
PS Ring Group	
UM Group	✓
VM Group (DTMF/DPT)	
External Pager (TAFAS)	
DISA	
Analogue/ISDN Remote Maintenance	
Idle Line Access no. + Phone no.	
Trunk Group Access no. + Trunk Group no. + Phone no.	
Other PBX Extension (TIE with no PBX Code)	
Other PBX Extension (TIE with PBX Code)	

*1 A PS destination can be used to forward fax calls to a fax machine at another PBX connected by TIE line.
A virtual PS can be specified as the destination of fax calls. Then, the extension number of the fax machine at the other PBX can be specified as the FWD—ALL Calls destination for calls to that virtual PS. (→ 5.2.4.6 Virtual PS)

Conditions

- This feature is only effective for calls arriving on DISA lines.
- If a fax signal is not detected before the DISA Intercept Routing—No Dial timer expires, the call is redirected to the operator extension, and fax detection ends.
- If the fax tone (CNG signal) detection is delayed because of the fax machine type or the state of the line, the DISA intercept timer may time out and the fax cannot be received. In this case, increasing the DISA intercept timer by 5 to 10 seconds will help avoid this problem.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf—◆DISA—Intercept Timer—Day (s), Lunch (s), Break (s), Night (s)

13.3.2 PBX Configuration—[5-3-2] Optional Device—Voice Message—DISA Message—◆ Fax Extension

Feature Guide References

2.16.1 Direct Inward System Access (DISA)

2.16.3 Built-in Simplified Voice Message (SVM)

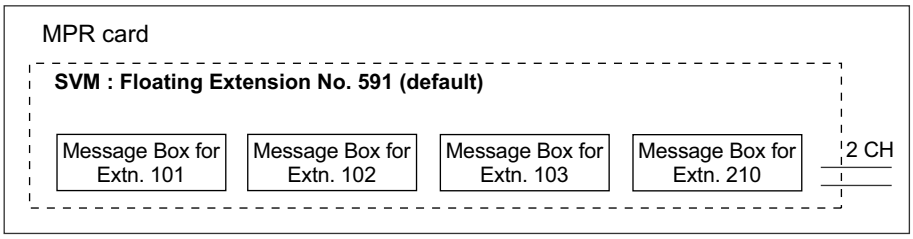
Description

This feature provides simple answering machine features for extensions.
Features for an extension user with a message box:
An extension user can record a personal greeting message to greet a caller and ask him to leave a voice message. The user can also play back and clear greeting messages and the voice messages left by callers.
Feature for a caller:
A caller can leave a voice message after hearing the greeting message.

SVM/OGM Block on the MPR Card

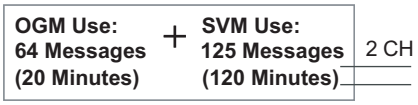
The MPR card has a built-in SVM/OGM Block. This block can be used both for the SVM feature, and for the OGM feature (→ 2.30.2 Outgoing Message (OGM)).
When used for the SVM feature, this block is assigned a floating extension number (default: 591). Two channels are available, allowing two users to access this feature at the same time.
When an extension is assigned to this feature through system programming, a message box is created for that extension. This message box is used to store greeting messages and voice messages for the extension. Each PT, SLT, and PS can be assigned its own message box.

[Example]



Up to 125 (MPR) messages (greeting messages and voice messages for extensions) with a maximum total recording time of 120 minutes can be recorded to the System Memory. This memory is shared between the message boxes of all extensions assigned to that memory.

You can use the SVM feature and OGM feature at the same time using the SVM/OGM Block on the MPR Card. There are 2 channels for both the SVM feature and OGM feature. The outgoing message (OGM) can be recorded only when both channels are vacant.125 messages (greeting messages and voice messages for extensions) (total 120 minutes) will be saved for SVM use, and 64 messages (total 20 minutes) are saved for OGM use as shown below.
In the System Memory of the Main unit



If the sum length of all recorded voice messages exceeds 90 % of the total recording space, the display informs all extension users that the total capacity has almost been reached. Users will hear dial tone 3 when going off-hook.

Floating Extension No. and Destination

An extension user can set incoming calls to be redirected to his message box when he cannot answer them. The user can set the floating extension number of this feature as the destination for redirected calls. Then, this feature answers redirected calls, plays back the relevant greeting message, and records a voice message. Incoming calls can be redirected to this feature by the following methods:
– Call Forwarding (FWD) (→ 2.3.2 Call Forwarding (FWD))

– Intercept Routing—No Answer/Busy/DND (→ 2.1.1.5 Intercept Routing)

[Programming Example for Intercept Routing]

Extension No.	Intercept Destination			
	Day	Lunch	Break	Night
101	102	**591**	**591**	**591**
102	103	**591**	**591**	**591**
301 (Operator)	–	**–**	**–**	**591**

In this example:
If a call is received at extension 101 and cannot be answered:
a. In day mode: the call will be redirected to extension 102.
b. In lunch/break/night mode: the call will be redirected to this feature, and a voice message will be recorded to the message box.
If a call is received at extension 301 (operator), the call will only be redirected to this feature in night mode.
If both FWD and Intercept Routing features are set for an extension, the FWD setting has priority. So, for example, it is possible for extension 101 to temporarily set FWD settings from his extension to forward calls to this feature even during day mode.

Greeting Message for Each Time Mode

When a call is redirected to this feature, the caller hears the designated greeting message. In addition to the normal greeting message, an extension user can record a different greeting message for each time mode (day/lunch/break/night) (→ 5.1.4 Time Service).
[Recording Example]
- Normal Greeting Message: "You have reached John. I am sorry I cannot take your call right now. Please leave a message."
- Greeting Message for lunch mode: "You have reached John. I am sorry I am out for lunch right now. Please leave a message."

If both the greeting message for a certain time mode and the normal greeting message have been recorded, callers will hear the greeting message for that time mode. However, if no greeting message has been recorded for a certain time mode, the normal greeting message will be played instead.
If neither the normal greeting message nor the greeting message for a certain time mode have been recorded, incoming calls will not be redirected to this feature in that time mode. For example, if a greeting message has only been recorded for night mode, and no normal greeting message has been recorded, incoming calls can only be redirected to this feature in night mode. No incoming calls will be redirected to this feature in day/lunch/break mode.

Direct Recording

An extension user can leave a voice message directly in the message box of an extension. In this case, the target extension will not ring. It is also possible to transfer a caller directly to the message box of an extension.

Message Notification

If a new voice message has been left in a message box, and distinctive dial tones are enabled, the user will hear dial tone 4 when going off-hook. If a message box contains only voice messages that have previously been listened to, dial tone 2 is heard instead. In addition, if the user's telephone has a Message button or Message/Ringer Lamp, the corresponding button or lamp will light when a voice message has been left. Pressing the lit button while on-hook shows the caller's information.

SVM Log

When a caller leaves a voice message, the following information is also recorded (as available):

a. Caller's Name
b. Caller's Telephone Number
c. Time recording started
d. Voice Message Status
  – "New" is displayed for voice messages which have not previously been listened to.
  – "Old" is displayed for voice messages which have previously been listened to.

This information can be viewed with the display of a PT or PS.
Please note that the information shown on the display may vary depending on the information that was received and the type of telephone used. Only users of a 6-line display PT can view all of the above information.

Checking Voice Messages Left by Callers

When an extension user accesses his message box, the most recent unplayed voice message is played first. When one voice message finishes playing, the next most recent voice message will begin playing automatically. After the last voice message in a message box has finished playing, the voice messages will begin playing from the most recent voice message again. When all voice messages have been played this time, the extension user will hear dial tone 4, and the line will be disconnected automatically.

SVM Remote Access from Trunk

An extension user can remotely access his message box through a trunk by calling his own extension and using the Walking COS feature. This allows the user to, for example, check the voice messages left in his message box when he is out of the office.

Accessing the Message Box of Another Extension

An extension user can access the message box of another extension by using the Walking COS feature to, for example, record a greeting message in a message box for an Incoming Call Distribution (ICD) Group as described below.
In addition, if the user's telephone has a Message button for another extension, the user can easily access the message box of that extension and listen to the voice messages left by callers.

Message Box for Incoming Call Distribution (ICD) Group

When a call is redirected to this feature from an ICD Group using Intercept Routing—Overflow (→ 2.2.2.6 Overflow Feature) or Call Forwarding (FWD), the greeting message for the first extension assigned to that ICD Group will be played, and the voice message from the caller will be recorded in that extension's message box.
A virtual PS can be assigned as the first extension of the ICD Group. This provides the ICD Group with a dedicated message box that is not shared with an actual extension. (→ 5.2.4.6 Virtual PS)

SLT Dial "*" Operation Mode

Depending on the environment, an SVM device might incorrectly detect DTMF signals from an SLT user. SLT Dial Mode can be set through system programming so that an SLT user can avoid having incorrect DTMF signals detected while listening to voice messages. When this setting is enabled, extension users must dial "*" prior to dialling other numbers.
Example:

SLT Dial "*" Operation Mode setting	To play back a voice message from the beginning
Disabled (Default)	Dial "1"
Enabled	Dial "*" and then "1"

Expanding the SVM feature

SVM features are limited (e.g., only 2 channels are available; total recording time is 120 minutes). To use richer voice messaging features (e.g., longer recording time, integration with Microsoft Outlook), use the Unified

Messaging system. (→ Section 3 Unified Messaging System) You can use SVM feature and the Unified Messaging system simultaneously.

Conditions

[General]

- This feature will function using the preinstalled MPR card.
- The maximum number of voice messages (not including greeting messages) that can be recorded for an extension can be set between 1 and 100 through system programming. (Default: 10)
- The Message button or Message/Ringer Lamp will light when an incoming call is answered by this feature. However, if a message is not left by the caller, the light will turn off when the call has ended.
- It is not possible to back up the SVM Log, recorded voice messages or greeting messages. If the PBX is initialised, all this information is cleared.
- When using the MPR card, this feature can be disabled for each extension through system programming. This can be useful for only allowing certain users (e.g., bosses, executives, etc.) access to the limited capacity of recorded voice messages for this feature. When this feature is disabled, all the recorded voice messages and greeting messages for the extension will be cleared.
- The voice messages for an extension will be cleared when the extension is checked out using the Hospitality feature (→ 2.23.1 Hospitality Features—SUMMARY). However, greeting messages will not be cleared.
- An extension is able to receive calls even while a voice message is being recorded in that extension's message box.
- If an extension user tries to access this feature when it is already being accessed by two other users, the user will hear a busy tone.
- If an extension user tries to clear a voice message shown on the display while the message box is already in use, the extension user will hear a notification tone.
- Voice message recording will stop when:
  a. a caller hangs up.
  b. the recording time for the voice message reaches the preprogrammed limit. (120 seconds)
  c. the recording space reaches the limit.
  In cases **b** and **c**, the caller will hear a notification tone, and the line will be disconnected.
- New voice messages cannot be recorded in a message box when:
  a. the total recording time or number of voice messages exceeds the limit.
  b. the number of voice messages for an extension exceeds the limit.
  c. the destination of a call has been changed more than once.
  d. the desired message box is already in use by another user.
  e. a call is received at an ICD Group, and the message box of the first extension assigned to that group is not available, or no extension is assigned as the first member of the group.
- **Recording Greeting Messages**
  If an extension user tries to record a new greeting message to his message box when the total recording time or number of messages has reached its limit, the user will hear a reorder tone.
- **Display Lock**
  A display PT user can lock the SVM Log display to prevent the user's information from being viewed and voice messages from being played back at any extension through personal programming (Display Lock). In this case, the Incoming/Outgoing Call Log displays and the Personal Speed Dialling number display are also locked. An extension personal identification number (PIN) (→ 2.24.1 Extension Personal Identification Number (PIN)) is required to lock/unlock the display.
- **Moving From SVM to DISA**
  When a call is answered by SVM but the caller wants to call another extension instead of leaving a message, it is possible to access DISA (DISA OGM 01 will be played) by dialling the Operator Call feature number. This allows the caller to call another extension via DISA or access the DISA AA service.

The OGM feature for the card/block to which the SVM mailbox belongs must be enabled via system programming.

[SVM Remote Access from Trunk]

- Whether Walking COS is available or not can be set through system programming.
  If the PBX uses analogue trunks, it is strongly recommended to prohibit Walking COS access. If an extension user who is accessing his message box through an analogue line goes on-hook while the voice messages are being played, the line will remain connected until all voice messages finish playing. This is because a reorder tone cannot be detected from an analogue line while playing voice messages.
- The first digit of the Walking COS feature number must be entered before the greeting message finishes playing.
- When accessing SVM through a trunk, it is possible to change message boxes (e.g., to leave a message in another user's message box after listening to one's own messages).
- **Accessing the Message Box of Another Extension**
  Listening to voice messages left by callers using a Message button for another extension is available only when voice messages have been left in the message box.
- It is not recommended to record music when creating a greeting message.
- This feature does not answer calls that are forwarded using the Call Forwarding by QSIG feature (4.2.4.4 Call Transfer (CT)—by QSIG).

PC Programming Manual References

6.15.1 Tool—SVM (Simplified Voice Message)—Delete All Recording
6.15.2 Tool—SVM (Simplified Voice Message)—Check Current Usage
10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous
→◆ SVM—Recording Time (s)
→◆ SVM—Dial Tone Continuous Time (s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Simplified Voice Message Access
11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings
→Tone Detection—◆ Simplified Voice Message Tone Detection—Silence
→Tone Detection—◆ Simplified Voice Message Tone Detection—Continuous
→Tone Detection—◆ Simplified Voice Message Tone Detection—Cyclic
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 6—◆ Display Lock/SVM Lock
12.1.8 PBX Configuration-[4-1-8] Extension-Wired Extension-Simplified Voice Message
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—◆ SVM Lock
12.2.6 PBX Configuration-[4-1-6]Extension—Portable Station—Simplified Voice Message
13.3.3 PBX Configuration-[5-3-3] Option Device - Voice Message - SVM

Feature Guide References

2.7.5 Walking COS
2.30.2 Outgoing Message (OGM)
2.25.1 Dial Tone
6.1 Capacity of System Resources

User Manual References

1.6.4 Using Voice Messaging (Built-in Simplified Voice Message [SVM])
3.1.2 Settings on the Programming Mode

2.17 Paging Features

2.17.1 Paging

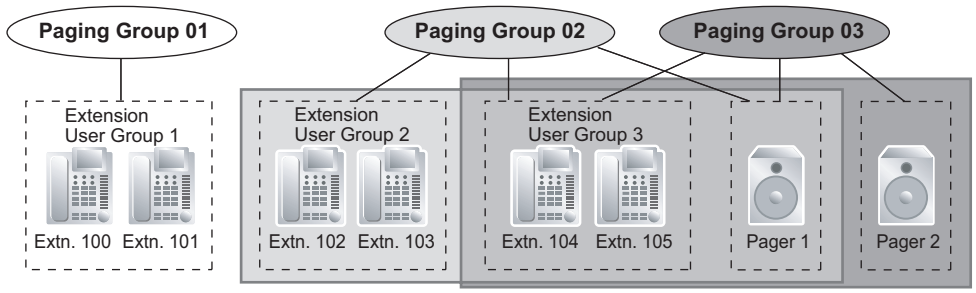
Description

An extension user can make a voice announcement to many destinations simultaneously.
The message is announced over the built-in speakers of PTs and/or external speakers (external pagers) which belong to the paging group. The PBX can connect to six external pagers.
The paged person can answer the page from a nearby telephone.
It is possible to page with a call on hold in order to transfer the call.
Paging Deny:
An extension user can choose not to receive paging announcements.

Paging Group

Each paging group consists of extension user groups and external pagers. One extension user group or external pager can belong to several paging groups.
(→ 5.1.2 Group)

[Example]



[Programming Example]

Paging Group No.	Extension User Group No.*1				External Pager*2			
	001	002	003	...	1	2	3	...
01	✓			...				...
02		✓	✓	...	✓			...
03			✓	...	✓	✓		...
04				...				...
05	✓	✓	✓	...	✓	✓	✓	...
:	:	:	:	...	:	:	:	...

✓: Constituent

*1 → 11.4 PBX Configuration—[3-4] Group—Paging Group
or
11.4.1 PBX Configuration—[3-4] Group—Paging Group—All Setting

*2 →11.4.2 PBX Configuration—[3-4] Group—Paging Group—External Pager

Conditions

- Paging announcements cannot be heard at the following types of extensions:
  – PSs
  – SLTs
  – Ringing or busy PTs
  – PTs in Paging Deny mode
  – PTs in Paging DND mode
  – IP-PTs assigned to extension user group 31 (default)[*1]
  – Non-KX-UT series SIP phones

  Although paging announcements cannot be heard at these types of extensions, they can answer paging announcements.

  [*1] Changing the extension user group of the extension, allows it to receive pages. However, doing so may affect the number of simultaneous IP extension and IP trunk calls available on the mother board.
- **External Pager Priority**
  External pagers can be used with the following priorities:
  TAFAS → Paging → BGM
  (→ 2.17.2 Trunk Answer From Any Station (TAFAS), 2.30.1 Background Music (BGM))
- **Volume Control**
  Paging volume from the PTs and the external pagers can be changed through system programming.
- **Paging DND**
  When DND (→ 2.3.3 Do Not Disturb (DND)) is set for incoming calls, it is programmable whether your extension receives paging through system programming.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ Group Paging
→◆ Group Paging Answer
→◆ Paging Deny Set / Cancel
10.9 PBX Configuration—[2-9] System—System Options
→Option 1—◆ PT Fwd / DND—Paging to DND Extension
→Option 3—◆ Confirmation Tone—Tone 2 : Paged / Automatic Answer
10.11.1 PBX Configuration—[2-11-1] System—Audio Gain—Paging/MOH
→◆ Paging—EPG 1-6 (External Pager 1-6)
→◆ Paging—Paging Level from PT Speaker
11.4 PBX Configuration—[3-4] Group—Paging Group
11.4.1 PBX Configuration—[3-4] Group—Paging Group—All Setting
11.4.2 PBX Configuration—[3-4] Group—Paging Group—External Pager
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 6—◆ Paging Deny

PT Programming Manual References

[640] Extension User Groups of a Paging Group
[641] External Pagers of a Paging Group

Feature Guide References

6.1 Capacity of System Resources

User Manual References

1.7.1 Paging
1.7.2 Answering/Denying a Paging Announcement

2.17.2 Trunk Answer From Any Station (TAFAS)

Description

When a call is received at the floating extension number assigned to the external pager, a ring tone is sent through the pager. Any extension user can then answer the call.

Conditions

- **Hardware Requirement:** A user-supplied external pager
- **Floating Extension Number**
  A floating extension number must be assigned for an external pager (default: 600 or 6000[*1]). It is possible to access an external pager by dialling its floating extension number.
- **Pager Volume**
  It is possible to change the volume of an external pager through system programming.

[*1] The default floating extension number depends on the value specified for **Numbering Plan** in Easy Setup.
→ 5.4.1 Easy Setup Wizard

Installation Manual References

4.10 Connection of Peripherals
5.4.1 Easy Setup Wizard

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ TAFAS Answer
10.11.1 PBX Configuration—[2-11-1] System—Audio Gain—Paging/MOH—◆ Paging—EPG 1-6 (External Pager 1-6)
13.2 PBX Configuration—[5-2] Optional Device—External Pager

PT Programming Manual References

[700] External Pager Floating Extension Number

Feature Guide References

5.5.8 Floating Extension

User Manual References

1.3.4 Answering a Call via an External Speaker (Trunk Answer From Any Station [TAFAS])

2.18 External Device Features

2.18.1 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 or outside party). In addition, extension users can dial the preset number of a doorphone to call that doorphone.

Conditions

- **Hardware Requirement:**
  An optional doorphone and a DOORPHONE card
- Each doorphone port can only be assigned to one tenant. The Time Table (day/lunch/break/night) of the tenant applies. (→ 5.1.4 Time Service)
- **Call Destination**
  The incoming doorphone call destination(s) can be assigned for each time mode (day/lunch/break/night) for each doorphone port. Destinations can be selected. (→ 2.1.2.1 Internal Call Features—SUMMARY)
- COS programming determines the doorphone ports that are able to make an outgoing trunk call.
- Internal Call Block determines which extensions can call a doorphone. (→ 2.1.2.2 Internal Call Block)
- **Ring Duration**
  If an incoming call is not answered within a preprogrammed time period, ringing stops and the call is cancelled.
- **Call Duration**
  The call duration can be restricted by a system timer. If the timer expires, the call will be disconnected.
- **Door Open**
  While engaged on a doorphone call, the extension user can unlock the door to let the visitor in. (→ 2.18.2 Door Open)
- A doorphone number can be referenced for each doorphone port.
- **For KX-UT670 users**
  The video feed from a network camera can be displayed on the telephone's display when a call from a doorphone is received if the following settings are configured on the telephone:
  – The doorphone's 2-digit number is registered to a contact, and **Doorphone** is selected as the number's label.
  – The network camera is registered to the contact.
  For details about the settings, refer to the telephone's documentation.

Installation Manual References

4.7.1 DPH2 Card (KX-NS5162)
4.9 Connecting to a Doorphone, Door Opener, and/or External Sensor

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf
→◆ Doorphone—Call Ring Duration (x10s)
→◆ Doorphone—Call Duration (x10s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ DOORPHONE Call
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—TRS—◆ TRS Level—Day, Lunch, Break, Night

10.8.2 PBX Configuration—[2-8-2] System—Ring Tone Patterns—Call from DOORPHONE
10.9 PBX Configuration—[2-9] System—System Options—Option 3—◆ Confirmation Tone—Tone 1 : Called by Voice
13.1 PBX Configuration—[5-1] Optional Device—Doorphone

PT Programming Manual References

[720] Doorphone Call Destination
[729] Doorphone Number Reference

User Manual References

1.11.1 If a Doorphone/Door Opener is Connected

2.18.2 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 COS programming. However, while engaged on a doorphone call, any extension user can unlock the door to let the visitor in (→ 2.18.1 Doorphone Call).

Conditions

- **Hardware Requirement:**
  A user-supplied door opener on each door, and a DOORPHONE card
- The door opener will unlock the door even if a doorphone is not installed.
- **Door Open Duration**
  The door can remain unlocked for a preprogrammed time period.
  If the door opener is a type that locks automatically when the door is closed, it is recommended that Door Open Duration be set to 2 seconds.

Installation Manual References

4.7.1 DPH2 Card (KX-NS5162)
4.9 Connecting to a Doorphone, Door Opener, and/or External Sensor

PC Programming Manual References

9.33 PBX Configuration—[1-1] Configuration—Slot—DOORPHONE Card—Card Property—◆ For Output - Device Type
10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf—◆ Doorphone—Open Duration (s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Door Open
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Optional Device & Other Extensions—◆ Door Unlock

PT Programming Manual References

[207] Door Unlock Time
[512] Permission for Door Open Access

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

1.11.1 If a Doorphone/Door Opener is Connected

2.18.3 External Sensor

Description

External sensing devices, such as security alarms or smoke detectors, can be connected to the PBX. When the PBX receives input from a sensor, a call is made to the preset destination, alerting the extension user. The available destinations of a sensor call are as follows:

[Available Destinations]

Destination	Availability
Wired Extension (PT/SLT/ISDN Extension)	✓
PS	✓
SIP Extension	✓
Incoming Call Distribution Group	
PS Ring Group	
UM Group	
VM Group (DTMF/DPT)	
External Pager (TAFAS)	
DISA	
Analogue/ISDN Remote Maintenance	
Idle Line Access no. + Phone no.	
Trunk Group Access no. + Trunk Group no. + Phone no.	
Other PBX Extension (TIE with no PBX Code)	
Other PBX Extension (TIE with PBX Code)	

When the call is answered, if distinctive dial tones are enabled, dial tone 3 is heard, and continues until the user goes on-hook. If the sensor call is not answered within a specified time, the call will be cancelled. It is possible to set a different ring tone pattern for calls received from each external sensor, to distinguish between them.
Also an e-mail can be sent to a specified e-mail address when the external sensor detects alarm.
For more detail information, refer to 5.4.3 E-mail Notification of Sensor Alarm.

Conditions

- **Hardware Requirement:**
  An external sensor and a DOORPHONE card
- Some devices may be unable to communicate correctly with the PBX. Confirm compatibility with the manufacturer of a device before installing it.
- After a sensor has been activated, the PBX will ignore any further alerts from the same sensor for the duration specified by a timer. This timer can be set separately for each sensor.
- As long as the previous sensor call is still being performed, any further alerts from the same sensor are ignored.
- The assigned sensor name and/or number are shown on the display of PTs and PSs when a sensor call is received.

- If the destination of a sensor call has set FWD, the sensor call will be redirected to the FWD destination. However, if the FWD destination is not supported as the destination of a sensor call (e.g., an outside party), the call will be received at the original destination. (→ 2.3.2 Call Forwarding (FWD))
- The following features cannot be used when a sensor call is received:
  - Alternate Receiving—Voice (→ 2.5.3 Intercom Call)
  - Hands-free Answerback (→ 2.4.4 Hands-free Answerback)
  - Consultation Hold/Call Hold (→ 2.13.1 Call Hold)
  - Call Transfer (→ 2.12.1 Call Transfer)
  - Executive Busy Override (→ 2.10.2 Executive Busy Override)
- Sensor call information is output on SMDR.

Installation Manual References

4.7.1 DPH2 Card (KX-NS5162)
4.9 Connecting to a Doorphone, Door Opener, and/or External Sensor

PC Programming Manual References

9.33 PBX Configuration—[1-1] Configuration—Slot—DOORPHONE Card—Card Property
→◆ For Sensor - Input Signal Decision Time
→◆ For Sensor - Input Signal Detection Reopening Time
10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ External Sensor—Ring Duration (s)
10.8.3 PBX Configuration—[2-8-3] System—Ring Tone Patterns—Call from Others—◆ External Sensor—Ring Tone Pattern Plan 1–8
13.5 PBX Configuration—[5-5] Optional Device—External Sensor

Feature Guide References

5.4.3 E-mail Notification of Sensor Alarm

2.18.4 External Relay Control

Description

By turning external device relays on and off, the PBX can control external devices such as alarms.
When an extension user enters the External Relay Control feature number, the specified relay turns on for a preprogrammed length of time. When this timer expires, the relay turns off automatically. This gives the PBX simple control over other equipment, allowing an extension user to, for example, activate an alarm from his extension.
If the relay cannot be accessed (for example, because use is not permitted by COS, or the port is not in service), a reorder tone will be heard at the extension.

Conditions

- **Hardware Requirement:**
  An external relay device and a DOORPHONE card
- The port of the DOORPHONE card to which the relay is connected must be assigned through system programming as a relay port (not a door opener port).
- Some devices may be unable to communicate correctly with the PBX. Confirm compatibility with the manufacturer of a device before installing it.
- Each external relay port has a COS assigned. This and the COS of an extension determine the extension users who can use External Relay Control.
- The length of time that a relay is turned on can be specified separately for each relay through system programming.
- If the same or another extension tries to access an external relay that has already been switched on, the timer for that relay is reset.

Installation Manual References

4.7.1 DPH2 Card (KX-NS5162)
4.9 Connecting to a Doorphone, Door Opener, and/or External Sensor

PC Programming Manual References

9.33 PBX Configuration—[1-1] Configuration—Slot—DOORPHONE Card—Card Property—◆ For Output - Device Type
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ External Relay Access
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Optional Device & Other Extensions—◆ External Relay Access
13.4 PBX Configuration—[5-4] Optional Device—External Relay

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

1.11.2 If an External Relay is Connected

2.19 Caller ID Features

2.19.1 Caller ID

Description

The PBX receives caller information, such as the caller's name and telephone number, through the trunk. This information can then be shown on the displays of PTs, PSs, or SLTs that support FSK-type Caller ID.
The PBX can modify a received number according to preprogrammed tables, so that an extension user can easily use the received number to call the caller back. For example, if an area code is not required to call outside destinations in a certain area, but received Caller ID numbers from that area contain an area code, it is possible to store that area code in a modification table so that it is deleted automatically from received numbers (Automatic Caller ID Number Modification).

1. **Features**
   Caller ID includes the following features:

Feature	Description & Reference
Caller ID	Caller's information which is sent from **an analogue trunk**. The following Caller ID signalling types are supported: FSK and DTMF.
Calling Line Identification Presentation (CLIP)	Caller's information which is sent from **an ISDN line**. → 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP)
Automatic Number Identification (ANI)	Caller's information which is sent from **an E1 line**. →2.29.1 E1 Line Service

2. **Service Features**

Features	Description & Reference
Calling Line Identification (CLI) Distribution	Directs a DIL/DID/DDI call to a CLI destination if the caller's identification (Caller ID/CLIP/ANI) has been assigned to the Caller ID Table. → 2.1.1.4 Calling Line Identification (CLI) Distribution
Incoming Call Log	Caller's information is automatically recorded in the call log of the called extension. This information is used for confirming the caller, calling the caller back, and/or storing the number and name in the Personal Speed Dialling. → 2.19.2 Incoming Call Log

3. **Number/Name Assignment**

Automatic Caller ID Number Modification

This PBX automatically modifies the incoming caller's number according to preprogrammed tables. The modified number will be recorded for calling back.

This PBX supports 4 modification tables, each of which can be used for any number of trunk groups. Each table has 10 locations for local/international calls and one for long distance calls. The PBX checks the local/international call data first. If a match is not found, the long distance call data is applied.
After the caller's number is modified by the Length of Digits Modification Tables or CLIP Modification Tables, the PBX checks the leading digits of the modified number for an area code programmed in the Caller ID Modification Table assigned to that trunk group. For more information, refer to 11.1.3 PBX Configuration—[3-1-3] Group—Trunk Group—Caller ID Modification—Leading Digits in PC Programming Manual.

[Example]

<Table Selection>

Trunk Group No.	Modification Table
1	1
2	3
:	:

<Modification Table>

Modification Table 1			
	Area Code	Removed No. of Digits	Added No.
Local/International Call Data 1	012	3	Blank
Local/International Call Data 2	00	2	001
:	:	:	:
Local/International Call Data 10			
Long Distance Call Data	Not programmable	0	0

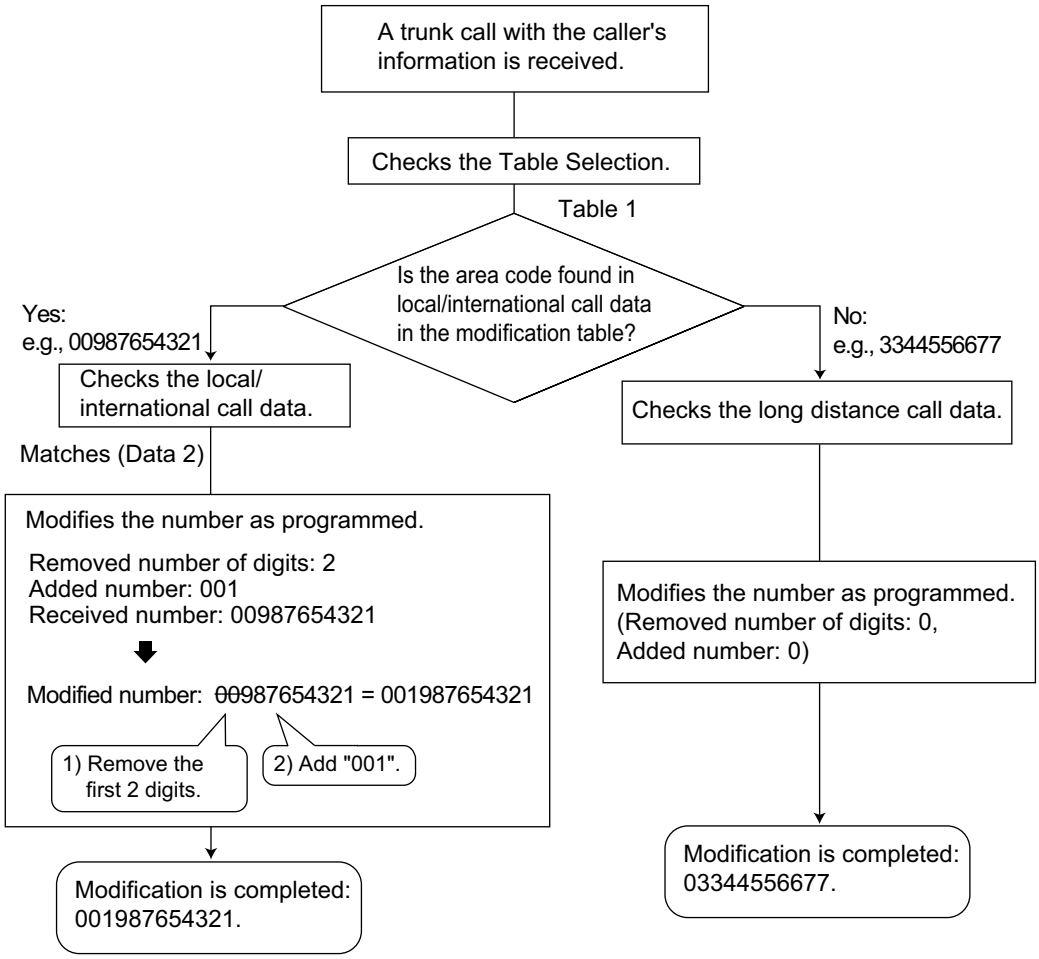
Note

When caller's information is sent through an ISDN line and the call type is Subscriber, National, or International, the following modification table is used instead of the above table:

<Modification Table>

	Removed No. of Digits	Added No.
Subscriber Call Data	0	Blank
National Call Data	0	0
International Call Data	0	00

<Modification Flowchart>



Caller ID Table Assignment

The System Speed Dialling Table is also used as the Caller ID Table.
In each location of the table, the following items can be assigned:

a. Telephone number (Trunk Access number + caller's telephone number)
b. System Speed Dialling name (caller's name)
(shown on the display or SMDR)
c. CLI destination
(used for CLI feature)

When a caller's modified telephone number matches a telephone number (the Trunk Access number is disregarded.) in the table, the call is sent to the assigned CLI destination.

[Example]

Location (System Speed Dialling No.)	Telephone No.*1	System Speed Dialling Name*2	CLI Destination*3
000	90123456789	ABC Company	200
001	:	:	:

Location (System Speed Dialling No.)	Telephone No.*1	System Speed Dialling Name*2	CLI Destination*3
:	:	:	:

*1 → 14.1 PBX Configuration—[6-1] Feature—System Speed Dial—◆ CO Line Access Number + Telephone Number
*2 → 14.1 PBX Configuration—[6-1] Feature—System Speed Dial—◆ Name
*3 → 14.1 PBX Configuration—[6-1] Feature—System Speed Dial—◆ CLI Destination

Caller's Name Reference

A name can also be shown on the display or SMDR. The PBX searches for the name in the following order:

1. Personal Speed Dialling data of the original called extension
2. System Speed Dialling (Caller ID) Table
3. Caller ID name received from the public line (Caller ID Name Reference)

If the name is not found, it will not be displayed.

Conditions

[General]

- Caller ID signalling type can be selected through system programming.
- The Caller ID Name Reference is only available for calls from the public network.

[Caller ID to SLT Port]

- **Hardware Requirement:**
  An MCSLC8 or MCSLC16 card
- This feature complies with ETSI (European Telecommunications Standards Institute)-type FSK and Bellcore-type FSK.
- When the caller's number is sent to an SLT, a Trunk Access number can be automatically added to the telephone number through system programming for calling back.
- When the caller's number exceeds 16 digits, the SLT receives only the first 16 digits, not counting the preceding Trunk Access number (if it is programmed to be added).
- If a call is transferred to an SLT, the transferring extension's information will be shown on the SLT. If the transferring extension goes on-hook before the call is answered, the original caller's information will be shown.
- When the Caller ID has information, such as private, out of area, or long distance, the information will be shown instead of the caller's number and name.
- Even if the caller's name is sent, the name may not be shown depending on the type of SLT.
- Incoming Call Log information is not shown on the SLT.
- Caller ID shows whether the call is an intercom or trunk call by default. This setting can be disabled through system programming.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous
→◆ Caller ID—Waiting to receive (s)
→◆ Caller ID—Visual Caller ID Display (s)
10.9 PBX Configuration—[2-9] System—System Options—Option 4—◆ Private Network—Public Call through Private Network—Minimum Public Caller ID Digits
10.10 PBX Configuration—[2-10] System—Extension CID Settings
11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Main—◆ Caller ID Modification Table
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 8
→◆ Extension Caller ID Sending

→◆ Incoming Call Wait Timer for Extension Caller ID
14.1 PBX Configuration—[6-1] Feature—System Speed Dial
11.1.3 PBX Configuration—[3-1-3] Group—Trunk Group—Caller ID Modification

PT Programming Manual References

[001] System Speed Dialling Number
[002] System Speed Dialling Name
[490] Caller ID Signal Type

Feature Guide References

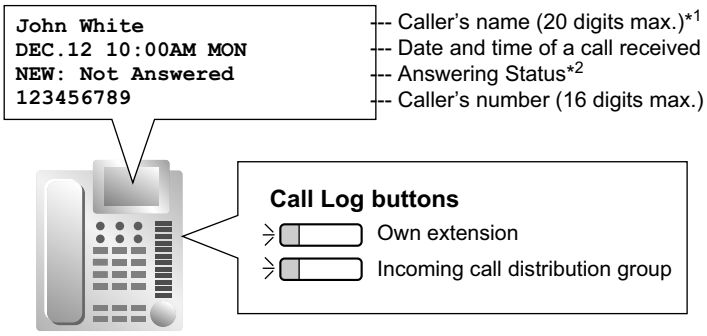
2.6.4 Speed Dialling—Personal/System
2.22.1.1 Station Message Detail Recording (SMDR)

2.19.2 Incoming Call Log

Description

When an incoming public trunk call with the caller's information (e.g., Caller ID) is directed to an extension, the information is automatically recorded in the call log of the called extension. Also, depending on system programming, incoming intercom calls (including over a TIE connection) are logged in the incoming call log. This information is shown on the telephone display and is used for confirming the caller, calling back, or storing the number and name in the Personal Speed Dialling.

[Example]



*1: If a call is received from an extension and no name is assigned to the extension, the incoming call log shows the extension number.
*2: "NEW" is displayed for call records which have not previously been viewed; "OLD" is displayed for call records which have previously been viewed.

Conditions

- **Call Log Button**
  A flexible button can be customised as the Call Log button for the extension or an incoming call distribution group. The button light shows the current status as follows:

Light pattern	Status
Red on	There is unchecked information.
Off	All information has been checked.

- **If the answering destination is not the original extension (FWD—No Answer, Intercept Routing—No Answer, Overflow, and Call Pickup):**
  If a call is forwarded because it is not answered or another extension picks up the call, the information is logged in the call logs of both the original destination and the answering destination. If a call is forwarded to several extensions before being answered, the information is logged in the call logs for all the extensions it was forwarded to. If a call is forwarded to an incoming call distribution group and is not answered, the information is not logged in the call log for the incoming call distribution group.
- The following types of calls will be recorded as "Not Answered" in the incoming call log of the original destination:
  – Calls received when the extension is in use (the caller hears a busy tone).
  – Calls rerouted using the Intercept Routing—Busy, FWD—All Calls, or FWD—Busy features.
  If disabled through system programming, these types of calls will not leave a record in the incoming call log.

It is also possible to specify through system programming if calls answered using Call Pickup are recorded as "Not Answered" or "Answered" in the incoming call log of the original destination.

- **Call Log for PS Calls**
  If a PS or a CS is in one of the following situations when a call arrives, the information is logged in the call log for the PS:
  a. When the PS is out of range.
  b. When the PS is turned off.
  c. When the CS is busy.
- **Display Lock**
  An extension user can lock the Incoming Call Log display to prevent the call information from being viewed at any extension through personal programming (Display Lock). In this case, the Outgoing Call Log display and the Personal Speed Dialling number display are also locked. An extension personal identification number (PIN) is required to use this feature. (→ 2.24.1 Extension Personal Identification Number (PIN))
- **Storing the Call Log Information in Personal Speed Dialling**
  When storing the number and name into Personal Speed Dialling from the call log information, the Idle Line Access number or the TIE Line Access number is automatically attached to the telephone number.
- **Storing the Call Log Information from an Extension**
  Depending on system programming, the information about an extension (including over a TIE connection) logged in the incoming call log can be stored in Personal Speed Dialling.
- **Incoming Call Log Memory**
  The total memory for the Incoming Call Log is determined in the PBX. The maximum number that can be logged for each extension and incoming call distribution group is also determined through system programming. If the memory becomes full, the new call record overwrites the oldest one.
- **Call Log for Incoming Call Distribution Group Calls**
  If the original destination of a call is an incoming call distribution group, and the call is not answered, the information is logged in the call log for the group. If it is answered, the information is logged in the call log for the answering extension.
- Through system programming, it is possible to select which Incoming Call Logs record call information when a call to an incoming call distribution group is answered by a member of the group:
  – Only the Incoming Call Log of the extension that answered the call.
  – Both the Incoming Call Log of the extension that answered the call and that of the incoming call distribution group.
- Through system programming, it is possible to select which Incoming Call Logs record call information when a call to an incoming call distribution group is answered by the overflow destination of the group:
  – Only the Incoming Call Log of the overflow destination.
  – Both the Incoming Call Log of the overflow destination and that of the incoming call distribution group.
- **E-mail Notification of Missed Calls**
  Extension users can receive an e-mail notification when they have a missed trunk call.
  → Contact—Email 1–3 in 8.2 Users—Add User
  → Email notification in 8.2 Users—Add User
- Through system programming, it is possible to select which Incoming Call Logs record call information when a member of an incoming call distribution group answers a call to the group:
  – Only the Incoming Call Log of the extension that answered the call.
  – Both the Incoming Call Log of the extension that answered the call and that of the incoming call distribution group.
- When the **Incoming Call Log**—**Extension / TIE Call** setting is enabled, the caller's information (e.g., extension number) is logged in the incoming call log of the extension that answered the call.

PC Programming Manual References

10.9 PBX Configuration—[2-9] System—System Options
→Option 7—◆ Incoming Call Log—Extension / TIE Call

→Option 7—◆ Outgoing Call Log—Extension Call
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Miscellaneous—◆ Supervisor Extension Number
11.5.3 PBX Configuration—[3-5-3] Group—Incoming Call Distribution Group—Miscellaneous—◆ Options—Call Log to ICD Group when ICD Member Answered
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Option 5—◆ Incoming Call Display
→Option 6—◆ Display Lock/SVM Lock
→Option 7—◆ Incoming Call Log Memory
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings
→Option 5—◆ Incoming Call Display
→Option 7—◆ Incoming Call Log Memory

Feature Guide References

2.6.4 Speed Dialling—Personal/System
2.21.2 Flexible Buttons
5.4.1 E-mail Notification for Extension Users
6.1 Capacity of System Resources

User Manual References

1.14.1 Using the Call Log
3.1.2 Settings on the Programming Mode

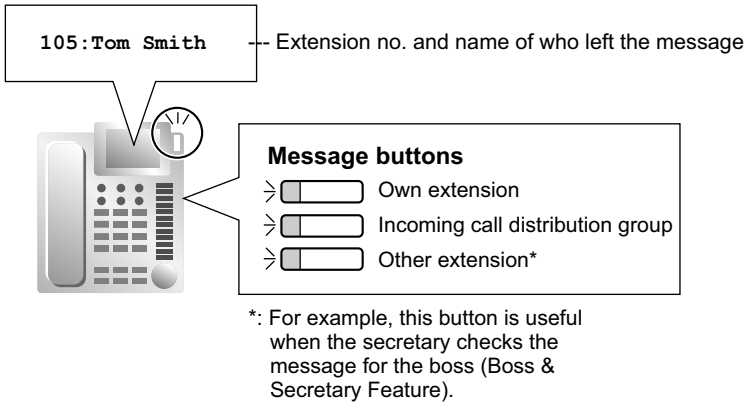
2.20 Message Features

2.20.1 Message Waiting

Description

An extension user can notify another extension user that he wishes to talk to the user. The notified extension user can return the call or listen to the messages recorded by the Unified Messaging system, a Voice Processing System (VPS) or the Built-in Simplified Voice Message feature.
When a message is left on a PT, the Message button lights or the Message/Ringer Lamp turns on red, and a message is shown on the display of a display PT. Pressing the lit Message button while on-hook shows the caller's information as shown below:

[Example]



Conditions

- **Message Button**
  A flexible button can be customised as the Message button for the extension, other extensions, or an incoming call distribution group.
- **Distinctive Dial Tone for Message Waiting**
  If the Distinctive Dial mode is enabled, dial tone 4 will be sent to an extension when a message has been left on the extension. (→ 2.25.1 Dial Tone)
- It is possible to set Message Waiting while hearing a ringback tone, busy tone, or DND tone.
- Messages are always left on the original destination extension, regardless of that extension's FWD settings.
- Both the extension that sent and received a message waiting notification can cancel the left message.
- If the extension that received a notification calls back the extension that sent the notification, and the call is answered, the notification will be cleared automatically. However, if a voice message has been left in a mailbox, whether the notification is cleared or not depends on the Unified Messaging settings or the VPS's settings.
- **SLT with a Message Waiting Lamp**
  The lamp activates in the same way as the MESSAGE button on a PT. The Message Waiting Lamp light pattern can be selected from one of 12 patterns. For details, refer to the documentation of the PBX to which the extensions are connected.
- Only Standard type or Unsolicited type general SIP extensions can control message waiting indication LEDs. For details, see 9.16 PBX Configuration—[1-1] Configuration—Slot—V-SIPEXT—Port Property—◆ MWI Method.

PC Programming Manual References

9.16 PBX Configuration—[1-1] Configuration—Slot—V-SIPEXT—Port Property
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Message Waiting Set / Cancel / Call Back
10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature—◆ Message Waiting Set
10.9 PBX Configuration—[2-9] System—System Options—Option 3—◆ Dial Tone—Distinctive Dial Tone
10.9 PBX Configuration—[2-9] System—System Options—Option 5—◆ SLT—Message Waiting Lamp Pattern
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Type

Feature Guide References

2.21.2 Flexible Buttons
3.2 System and Subscriber Features
6.1 Capacity of System Resources

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer
1.8 Using the Unified Messaging Features

2.20.2 Absent Message

Description

An extension user can set or select a message (e.g., the reason for absence) to be displayed on his telephone. When a display PT user calls the extension, the message is shown on the caller's telephone. The following messages can be programmed as desired:

Type	Message No.	Message (Example)	Description
System message	1	**Will Return Soon**	Messages may be edited through system programming. They are used for every extension user commonly.
	2	**Gone Home**	
	3	**At Ext** %%%% (Extension Number)	
	4	**Back at** %%:%% (Hour:Minute)	
	5	**Out until** %%/%% (Month/Day)	
	6	**In a Meeting**	
	7		
	8		
Personal message	9		A message is programmable at each extension through personal programming (Personal Absent Message), which can only be used by that extension user.

Note

The "%" means a parameter to be entered when assigning a message at an individual extension. Up to seven "%"s can be stored for each message.

Conditions

- An extension user can select only one message at a time. The selected message is displayed at the extension while on-hook.
- An extension user who has a Unified Messaging mailbox can also set his absent message from a remote location by following the voice guidance (→ 3.2.2.25 Remote Absent Message).

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Absent Message Set / Cancel
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 3—◆ Absent Message
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 3—◆ Absent Message
14.5 PBX Configuration—[6-5] Feature—Absent Message

PT Programming Manual References

[008] Absent Message

Feature Guide References

6.1 Capacity of System Resources

User Manual References

1.6.2 Showing a Message on the Caller's Telephone Display (Absent Message)
3.1.2 Settings on the Programming Mode

2.21 Proprietary Telephone (PT) Hardware Features

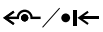
2.21.1 Fixed Buttons

Description

PTs, DSS Consoles, and Add-on Key Modules are provided with the following feature/Line Access buttons: Depending on your device type, some buttons may not be provided.
As for buttons on PSs, please refer to the Operating Instructions for each PS.

[PT and Add-on Key Module]

Button		Usage
Navigator Key, Jog Dial, Volume Key		Used to adjust the ringer, speaker, handset and headset volume and the display contrast. Navigator Key and Jog Dial can also be used to select data from the Call Directory and the System Feature Access Menu on the display.
ENTER		Used to confirm the selected item.
CANCEL		Used to cancel the selected item.
PROGRAM	*PROG.*	Used to enter and exit the programming mode.
FLASH/RECALL	R	Used to disconnect the current call and make another call without hanging up (Flash/Recall mode or Terminate mode) or used to send a flash/recall signal to the telephone company or host PBX to access their features (External Feature Access mode). This button can also be used as a CANCEL button while on-hook.
HOLD		Used to place a call on hold.
SP-PHONE (Speakerphone)		Used for hands-free operation. Also used to switch between handset and hands-free operation.

Button		Usage
MONITOR		Used for a hands-free dialling. Also used to monitor the party's voice in hands-free mode.
MESSAGE		Used to leave a message waiting indication or call back the party who left the message waiting indication.
REDIAL		Used to redial the last dialled number.
TRANSFER		Used to transfer a call to another party.
Flexible CO (Trunk)		Used to make or receive a trunk call or can be reassigned to a different Trunk Access button (Default: S-CO) or to another feature button.
INTERCOM	*INT'*	Used to make or receive intercom calls.
AUTO ANS (Auto Answer)/MUTE		Used to receive an incoming call in hands-free mode, or used for microphone or handset mute during a conversation. (Dual feature button)
VOICE CALL/MUTE		Used to monitor an intercom call automatically (a hands-free conversation is not possible). Also used for handset microphone mute during a conversation.
AUTO DIAL/STORE		Used for System/Personal Speed Dialling and storing programme changes.
CONF (Conference)		Used to establish a multiple-party conversation.
FWD/DND		Used to perform FWD or DND. (Dual feature button)
PAUSE		Used to insert a pause in a stored number. With an APT, it is used as the PROGRAM button.
Soft		Used to select the item displayed on the bottom line of the display.
SELECT		Used to select the displayed item or to call the displayed number.
SHIFT		Used to access the second level of Soft button items.
MODE		Used to shift the display to access various features.
NEXT PAGE		Used to switch the page for the Self Labelling feature (KX-NT366/KX-NT553/KX-NT556/KX-NT560 only).

[DSS Console]

Button		Usage
ANSWER		Used to answer an incoming call or place the current call on hold and answer another call with one touch.
RELEASE		Used to disconnect the line during or after a conversation or to complete a Call Transfer.
Flexible CO (Trunk)		Used to make or receive a trunk call or can be reassigned to a different trunk or to another feature button.

Button	Usage
Flexible DSS (Direct Station Selection)	Used to access an extension with one touch. Every button is programmed to correspond to an extension. DSS buttons can also be reassigned to other features.
PF (Programmable Feature)	Used to access a preprogrammed feature with one touch. (no default)

Conditions

- Certain buttons are equipped with a light to show line or feature status.

2.21.2 Flexible Buttons

Description

You can customise the flexible buttons and/or programmable feature (PF) buttons on PTs, Add-on Key Modules, and PSs through either system or personal programming. They can then be used to make or receive intercom or trunk calls or be used as feature buttons, as follows:

[Button Usage]

Button	Usage
Single-CO (S-CO)	Used to access a specified trunk for making or receiving calls.
Group-CO (G-CO)	Used to access an idle trunk in a specified trunk group for making calls. Incoming calls from trunks in the assigned trunk group arrive at this button.
Loop-CO (L-CO)	Used to access an idle trunk for making calls. Incoming calls from any trunk arrive at this button.
Direct Station Selection (DSS)	Used to access an extension with one touch.
Network Direct Station Selection (NDSS)	Used to access an extension at another PBX within the same network.
One-touch Dialling	Used to access a preprogrammed party or feature with one touch.
Incoming Call Distribution (ICD) Group	Used to access a specified incoming call distribution group for making or receiving calls.
Message	Used to leave a message waiting indication or call back the party who left the message waiting indication.
FWD/DND (External/Internal/ Both)[*1]	Used to perform the FWD or DND feature for the extension. The feature is applied to trunk calls, intercom calls, or both.
Group FWD (External/Internal/ Both)	Used to perform the FWD feature for a specified incoming call distribution group. The feature is applied to trunk calls, intercom calls, or both.
Account Code Entry (Account)	Used to enter an account code.
Conference	Used to establish a multiparty conversation.
Terminate	Used to disconnect the current call and make another call without hanging up.
External Feature Access (EFA)	Used to send a flash/recall signal to the telephone company or host PBX to access their features.
Charge Reference	Used to check the total call charge for your own extension.
Call Park	Used to park or retrieve a call in a preset PBX parking zone.
Call Park (Automatic Park Zone)	Used to park a call in an idle PBX parking zone automatically.
Call Log	Used to show the incoming call information.
Log-in/Log-out[*1]	Used to switch between log-in and log-out mode.
Hurry-up	Used to redirect the longest waiting call in the queue of an incoming call distribution group to the overflow destination.

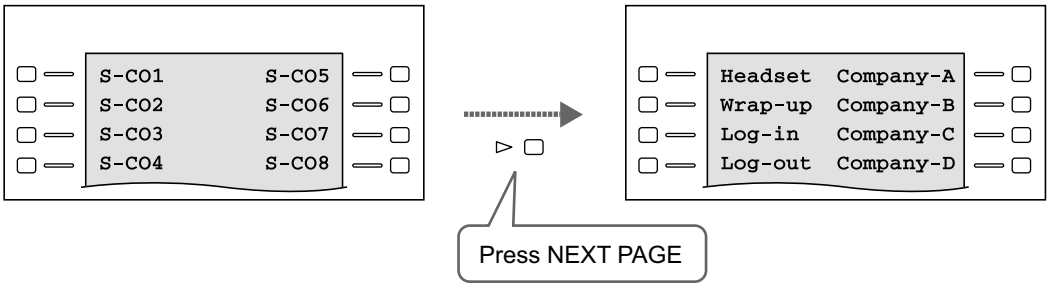
Button	Usage
Wrap-up[*1]	Used to switch the Wrap-up/Not Ready and Ready modes.
System Alarm	Used to confirm a PBX error.
Time Service[*1]	Used to switch the assigned time modes: day, lunch, break or night. Also used to check the current time mode status.
Answer	Used to answer an incoming call.
Release	Used to disconnect the line during or after a conversation, or to complete a Call Transfer.
Toll Restriction/Call Barring	Used to change the TRS/Barring level of other extension users temporarily.
ISDN Service	Used to access an ISDN service.
Calling Line Identification Restriction (CLIR)[*1]	Used to switch between the CLIP and CLIR service.
Connected Line Identification Restriction (COLR)[*1]	Used to switch between the COLP and COLR service.
ISDN Hold	Used to transfer a call using the telephone company.
Headset	Used to turn on/off the headset mode while idle. Used to switch between hands-free mode and headset modes during a conversation.
Time Service Switching Mode (Automatic/Manual)[*1]	Used to switch between the Automatic Switching and Manual Switching mode.
Two-way Record	Used to record a conversation into your own mailbox.
Two-way Transfer	Used to record a conversation into the mailbox of a specified extension.
One-touch Two-way Transfer	Used to record a conversation into the mailbox of a specified extension with one touch.
Live Call Screening (LCS)	Used to monitor your own voice mailbox while an incoming caller is leaving a message and, if desired, intercept the call.
Voice Mail (VM) Transfer	Used to transfer a call to the mailbox of a specified extension. Also used to access the Unified Messaging system (→3.2 System and Subscriber Features) or the voice mail features of a VPS (→2.28.3 Voice Mail DPT (Digital) Integration).
Check-in	Used to switch the status of extensions from Check-out to Check-in.
Check-out	Used to switch the status of extensions from Check-in to Check-out.
Cleaned-up	Used to switch the room status of extensions between Ready and Not Ready.
CTI	Used to access CTI features.
Primary Directory Number (PDN)	Used to make and receive both outside and intercom calls. (→ 2.9.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension)

Button	Usage
Secondary Directory Number (SDN)	Used to show the current status of another extension, call the extension, and pick up or transfer calls to it. (→ 2.9.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension)

*1 **One-touch Feature Setting Buttons:** Pressing these buttons while on-hook changes the feature settings. The new mode will be displayed for a preprogrammed time period.

Self Labelling (KX-NT366/KX-NT553/KX-NT556/KX-NT560 only)

The KX-NT366 IP-PT and KX-NT553/KX-NT556/KX-NT560 IP-PT have an LCD screen next to their flexible buttons. The label for each button can be set through personal/system programming to reflect the button's function. Additionally, the flexible buttons can be organised into multiple "pages". You can toggle between pages by pressing the NEXT PAGE key, as follows:



Note

The appearance of the NEXT PAGE button differs depending on the telephone model.

Conditions

[General]

- Not all buttons are available for KX-UT-series SIP phones. For details about available flexible button types, see 5.2.2.1 KX-UT Series SIP Phones.

[Self Labelling]

- Up to 12 characters can be assigned to the LCD of each flexible button through personal/system programming.
  → 12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Label Name
- When an incoming trunk call is answered or a trunk is seized, the corresponding CO button will turn Green and the LCD display will switch to the page that the corresponding CO button is registered in.
- It is not recommended to assign the System Alarm button when using this feature, because if an alarm occurs when the System Alarm button is not on the visible page, the alarm will not be noticed.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ PT Display—PT Last Display Duration in Idle Mode (s)
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
12.1.4.1 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—Flexible button data copy
12.1.5 PBX Configuration—[4-1-5] Extension—Wired Extension—PF Button
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button

User Manual References

1.14.4 Self Labelling (KX-NT366/KX-NT553/KX-NT556/KX-NT560/KX-UT248/KX-UT670 only)
3.1.3 Customising the Buttons

2.21.3 LED Indication

Description

The LED (Light Emitting Diode) of the Message/Ringer Lamp and following buttons (Line Status Buttons and Corresponding Extension Status Button) show line conditions with a variety of light patterns.
Line Status Buttons: S-CO, G-CO, L-CO, INTERCOM, ICD Group, PDN
Corresponding Extension Status Button: DSS, SDN

1. **Light Pattern of the Message/Ringer Lamp**
   [IP-PT and DPT]
   - Incoming call from a trunk: Red flashing
   - Incoming call from another extension: Green flashing
   - Message(s) present (no incoming call): Red on
   - No message(s) present (no incoming call): Off

   [APT]
   - Incoming call: Red flashing
   - Message(s) present (no incoming call): Red on
   - No message(s) present (no incoming call): Off
2. **Light Pattern of the Line Status Buttons**

<table>
<tr><th rowspan="3">Light Pattern</th><th colspan="5">Line Status Button</th></tr>
<tr><th colspan="3">Trunk Status</th><th>Intercom Line Status</th><th>Incoming Call Distribution Group Line Status</th></tr>
<tr><th>S-CO</th><th>G-CO</th><th>L-CO</th><th>INTERCOM</th><th>ICD Group</th></tr>
<tr><td>Off</td><td colspan="5">Idle</td></tr>
<tr><td>Green on</td><td colspan="5">This extension is using the line.</td></tr>
<tr><td>Slow green flashing</td><td colspan="5">This extension is holding the line.</td></tr>
<tr><td>Moderate green flashing</td><td colspan="5">This extension is holding the line using Exclusive Hold or using the line for an Unattended Conference.</td></tr>
<tr><td>Rapid green flashing</td><td>Incoming call/ Privacy Release</td><td colspan="4">Incoming call</td></tr>
<tr><td>Red on</td><td>Incoming call for another extension/ Another extension is using the line/ Another extension has the line on Exclusive Hold.</td><td>Other extensions are using all trunks in the trunk group.</td><td colspan="2">—</td><td>This extension is logged out of the incoming call distribution group.</td></tr>
</table>

Light Pattern	Line Status Button				
	Trunk Status			Intercom Line Status	Incoming Call Distribution Group Line Status
	S-CO	G-CO	L-CO	INTERCOM	ICD Group
Slow red flashing	Another extension is holding the line.	—			
Rapid red flashing	Incoming call to the incoming call distribution group in Ring distribution method			—	

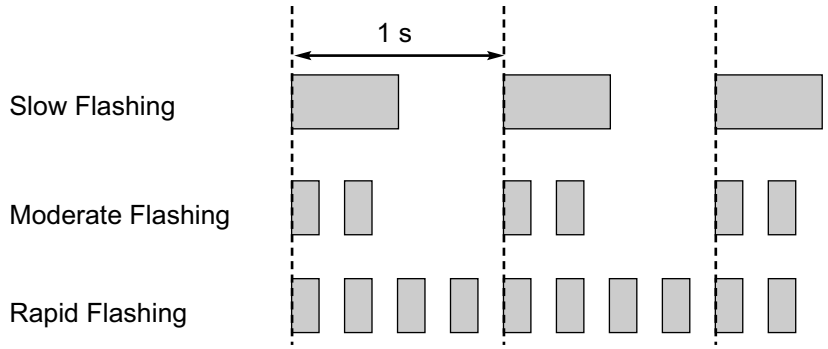
For information on the light patterns of PDN and SDN buttons, refer to 2.9.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension.

3. Light Pattern of the Corresponding Extension Status Button

Light Pattern	Corresponding Extension Status Button (DSS)
Off	Idle
Red on	Busy/Incoming call*1/DND for trunk calls
Rapid red flashing	Incoming call*2

*1 Only when Call Pickup by a DSS button is disabled.
*2 Only when Call Pickup by a DSS button is enabled.

4. Flashing Light Patterns



Conditions

- The incoming call shows on the buttons in the following priority:
  ICD Group→S-CO→G-CO→L-CO→PDN→INTERCOM
- The light pattern of a DSS button for incoming call can be set to "Off" through system programming. In this case, the DSS button light will not indicate the status of the corresponding extension.

PC Programming Manual References

10.9 PBX Configuration—[2-9] System—System Options
→Option 1—◆ PT Fwd / DND—Fwd LED
→Option 1—◆ PT Fwd / DND—DND LED
→Option 4—◆ DSS Key—DSS key mode for Incoming Call

Feature Guide References

2.2.2.2 Group Call Distribution
2.2.2.7 Log-in/Log-out
2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features
2.13.1 Call Hold
2.14.2 Conference
2.14.3 Privacy Release

2.21.4 Display Information

Description

A display PT shows the user the following information while making or receiving calls if they are available:

Display Item	Display Example	Condition
The extension number and name of the calling or called extension, or incoming call distribution group	`123: Tom Smith`	–
Status of the called extension	`123: Busy`	–
The number and name of the optional device	`D02: 1st Door`	–
The dialled telephone number	`1234567890`	–

The received call information
a. Caller's name
b. Caller's number
c. Trunk number/name
d. Original Destination, if the call is forwarded
e. DDI/DID name
	`ABC Company`
	`12345678`
	`Line 001: Sales`
	`→102:Mike`
	`Panasonic`
		The first line message can be either (a), (c), or (e) at each extension through system programming. These can be displayed in turn by pressing the TRANSFER button or DISP Soft button during a call.

Call charge fee during a trunk call.	`12.35€`	The currency, position of the currency symbol, and the decimal point are programmable.
Call duration during a trunk call.	`Line 001 11:02'28`	–

Conditions

- **Multilingual Display**
  Each extension can select its display language through personal programming (Display Language Selection).
- **Display Contrast**
  It is possible to adjust the display contrast through personal programming (Display Contrast Selection). This is available only for DPTs and IP-PTs.
- **Display Backlight**
  Some extensions can select whether to turn the display backlight on or off through personal programming (Display Backlight Selection). For details, refer to the manual for your telephone.
- Characters (name) or digits (number) exceeding the display's size limitation are not displayed. In this case, information which have been programmed is hidden, but not altered.

PC Programming Manual References

10.9 PBX Configuration—[2-9] System—System Options—Option 5—◆ PT Feature Access—No. 1–8
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Option 5—◆ Display Language
→Option 5—◆ Incoming Call Display
→Option 5—◆ Automatic LCD Switch when Start Talking

12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings
→Option 5—◆ Display Language
→Option 5—◆ Incoming Call Display
→Option 5—◆ Automatic LCD Switch when Start Talking
14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Charge
→◆ Charge Options—Digits After Decimal Point
→◆ Charge Options—Currency
→◆ Charge Options—Currency Display Position
18.1 PBX Configuration—[10-1] CO & Incoming Call—CO Line Settings—◆ CO Name

PT Programming Manual References

[130] Decimal Point Position for Currency
[131] Currency

User Manual References

3.1.2 Settings on the Programming Mode

2.22 Administrative Information Features

2.22.1 Record Log Features

2.22.1.1 Station Message Detail Recording (SMDR)

Description

Automatically records detailed information for each extension.

1. **SMDR Output Port**
   The following output methods can be selected through system programming:
   → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ SMDR Format—Port

Output Method	Description
Telnet compatible terminal emulator	SMDR information is sent to a Telnet compatible terminal emulator via LAN.

2. **SMDR Output Data**
   The following data will be recorded and sent to the SMDR output port:
   a. Trunk call information (incoming/outgoing)
   b. Intercom call information (outgoing)
   c. Log-in/Log-out information
   d. PBX error log (→ 5.6.4 Local Alarm Information)
   e. Hospitality feature information (→ 2.23.1 Hospitality Features—SUMMARY)
   f. Printing Message information (→ 2.22.2 Printing Message)
   Memory for SMDR: A specified number of call records can be stored in the PBX. If more calls are originated or received, the oldest record is overwritten by the newest one.
3. **SMDR Format Type and Contents**
   The following three types of output format can be selected through system programming:

Pattern A: 80 digits without call charge information

```
Date       Time     Ext   CO  Dial Number                  Ring  Duration  ACC Code    CD
(8 digits) (7)      (5)   (2) (25)                         (4)   (8)       (10)        (3)
-------------------------------------------------------------------------------------------
01/02/02 10:03AM  1200  01  <I>12345678901234567890      5'15  00:00'00                NA
01/02/02 10:07AM  1200  01  <I>                          0'05  00:01'05  9876543210
01/02/02 10:15AM  1200  01  1234567890123456                 00:01'05  9876543210  TR
01/02/02 10:30AM  *123  01  1234567890123456                 00:01'05  9876543210


01/02/02 01:07PM  1234  01  <I>ABC COMPANY12345678   0'05  00:01'05  9876543210
01/02/02 01:07PM  1234  01  <D>CDE9876<I>Q COMPANY  0'05  00:01'05  9876543210
01/02/02 01:07PM  1234  01  ABC COMPANY12345678          00:01'05
01/02/02 01:07PM  1234  01  123.....                     00:01'05
01/02/02 01:07PM  1234  01  123456XX                     00:12'05  98765

01/02/02 08:33AM  1234      In the office
01/02/02 01:07PM  1234      LOG IN
01/02/02 03:35PM  1234      LOG OUT
01/02/02 03:45PM  1234      EXT1235
01/02/02 03:50PM  1234      Check in
01/02/02 03:55PM  1234      Check out
01/02/02 04:00PM  1234      Timed Reminder/Start
01/02/02 04:01PM  1234      Timed Reminder/No Answer
01/02/02 04:01PM  1234      Timed Reminder/Answer
01/02/02 04:05PM  1234      <I>S003                                                  RC
   :         :       :   :       :                :       :         :          :
  (1)       (2)     (3) (4)     (5)              (6)     (7)       (8)        (9)
```

Pattern B: 80 digits with call charge information

```
Date       Time     Ext   CO  Dial Number                  Duration  Cost        ACC Code    CD
(8 digits) (7)      (5)   (2) (20)                         (8)       (8+2)       (10)        (3)
-------------------------------------------------------------------------------------------
01/02/02 10:03AM  1210  01  <I>                                                             NA
01/02/02 10:07AM  2005  01  12345678901234567890     00:00'05  00560.00EU  9876543210
   :         :       :   :       :                      :         :           :          :
  (1)       (2)     (3) (4)     (5)                    (7)       (10)        (8)        (9)
```

Pattern C: 120 digits

```
Date       Time     Ext   CO    Dial Number                                  Ring  Duration  Cost        ACC Code    CD
(8 digits) (7)      (5)   (4)   (50)                                         (4)   (8)       (8+3)       (10)        (3)
-------------------------------------------------------------------------------------------------------------------
01/02/02 10:03AM  1230  0001  12345678901234567890123456789012345678901234          00:00'05  00560.00EUR 9876543210  TR
01/02/02 10:07AM  1230  0001  <I>ABC COMPANY123456789012345                 0'05  00:00'05                9876543210  TR
   :         :       :   :       :                                            :     :         :           :           :
  (1)       (2)     (3) (4)     (5)                                          (6)   (7)       (10)        (8)         (9)
```

[Explanation]

The following table explains the SMDR contents which are based on the numbers in the previous pattern examples. For the programmable items, refer to the following [Programmable Items].

Number in the Pattern	Item	Description
(1)	**Date**	Shows the date of the call.
(2)	**Time**	Shows the end time of a call as Hour/Minute/AM or PM.
(3)	**Ext (Extension)**	Shows the extension number, floating extension number, etc., which was engaged in the call. Also shows the following codes: **Dxxx:** Outgoing trunk call from a doorphone (xxx=doorphone number) (→ 2.18.1 Doorphone Call) **Txxx:** Outgoing trunk call by TIE line service (xxx=trunk group number) ***xxx:** Verified call (xxx=verification code) (→ 2.7.6 Verification Code Entry)
(4)	**CO (Trunk)**	Shows the trunk number used for the call. For patterns A and B, "00" will be shown for trunk numbers over hundred.

Number in the Pattern	Item	Description
(5)	**Dial Number**	**[Trunk Call]** **Outgoing Trunk Call** Shows the dialled telephone number. Valid digits are as follows: **0 through 9, ✳, #** **P:** Pause **F:** EFA signal **=:** A Host PBX Access code (→ 2.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)) **. (dot):** Secret dialling **X:** Privacy dial **–:** Transferred call If the transfer destination extension enters some digits, the entered digits will be added after "–". **Incoming Trunk Call** Shows <I> + the caller's identification name/number. It is also possible to show the DDI/DID call information. In this case, <D> + DDI/DID name/number is added before <I>.
		[Outgoing Intercom Call] Shows the dialled extension number followed by "EXT".
		[Log-in/Log-out] Shows the log-in or log-out status.
		[Check-in/Check-out] Shows the check-in or check-out status. (→ 2.23.2 Room Status Control)
		[Timed Reminder] Shows the status of a timed reminder, either "Start", "No Answer", or "Answer". (→ 2.24.4 Timed Reminder)
		[Printing Message] Shows the selected message. (→ 2.22.2 Printing Message)
		[Sensor Call] Shows calls from an external sensor as follows: <I> S + sensor number. (→ 2.18.3 External Sensor)
(6)	**Ring**	Shows the ring duration before answering a call in Minutes/Seconds.
(7)	**Duration**	Shows the duration of the trunk call in Hours/Minutes/Seconds.
(8)	**Acc Code (Account Code)**	Shows the account code appended to the call. (→ 2.5.4.3 Account Code Entry)

Number in the Pattern	Item	Description
(9)	**CD (Condition Code)**	Shows other call information with the following codes: **CL:** Collect call **TR:** Transfer **FW:** FWD to trunk **D0:** Call using DISA or TIE line service **RM:** Remote maintenance (modem) (→ 5.5.2 PC Programming) **NA:** Not answered call **RC:** Received call **AN:** Answered call **VR:** Received call with Call Waiting Caller ID (Visual Caller ID) **VA:** Answered call with Call Waiting Caller ID (Visual Caller ID)
(10)	**Cost**	Shows the call charge.

[Programmable Items]

Item	Description
Outgoing trunk call	Controls whether the outgoing trunk calls are shown. This setting is common throughout the PBX. COS programming is also required. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ Print Information—Outgoing Call
Incoming trunk call	Controls whether the incoming trunk calls are shown. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ Print Information—Incoming Call
Outgoing intercom call	Controls whether the outgoing intercom calls are recorded. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ Print Information—Intercom Call
Log-in/Log-out status	Controls whether the log-in/log-out status is recorded. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ Print Information—Log-in / Log-out
ARS dial	Controls whether the user-dialled number or the modified number is shown. The Host PBX Access code ("=" followed by the access code) can be shown (as supplementary information) only when the modified number is selected in this setting. (→ 2.8.1 Automatic Route Selection (ARS)) → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—ARS Dial

Item	Description
Caller's identification	Controls whether the caller's identification number, name, number and name, or nothing is shown. If "**none**" is selected, <I> will not be shown. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—Caller ID Number & Name
DID/DDI number	Controls whether the DID/DDI number, name, number and name, or nothing is shown. If "**none**" is selected, <D> will not be shown. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—DDI/DID Number & Name
Secret dialling	Controls secret dialling. If enabled, the dialled number will be shown as dots. This setting is effective only when the modified number is selected in ARS dial setting above. If the user-dialled number is selected in ARS dial setting, the dialled number will be shown as dots regardless of this setting. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—Secret Dial
Privacy dial	Enables or disables privacy dial. If enabled, the last four digits of the dialled telephone number and any additional digits after connection will be shown as "X". (e.g., 123-456-XXXX) → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—Privacy Mode
Date order	The date order is changeable: month/day/year, day/month/year, year/month/day, year/day/month. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ SMDR Format—Date Format
Received call	Controls whether the time of receiving an incoming trunk call is shown. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—Condition Code "RC"
Answered call	Controls whether the time of answering an incoming trunk call is shown. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Option—Condition Code "AN"
Room status	Controls whether room status changes are shown. → 14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Main—◆ SMDR for External Hotel Application 1—Room Status Control
Timed Reminder call	Controls whether Timed Reminder calls are shown (→ 2.24.4 Timed Reminder). → 14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Main—◆ SMDR for External Hotel Application 1—Timed Reminder (Wake-up Call)

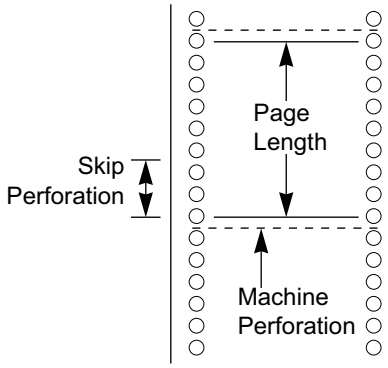
Item	Description
Printing Message	Specifies the messages that can be selected from an extension (→ 2.22.2 Printing Message). → 14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Main—◆ SMDR for External Hotel Application 2—Printing Message 1–8
Time format	Controls whether time is displayed in 12-hour or 24-hour format. → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ SMDR Format—Time Format (12H / 24H)

Conditions

[General]

- **SMDR Format**
  The following SMDR format can be set through system programming in order to match the paper size being used in the printer:
  a. Page Length: determines the number of lines per page.
  → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ SMDR Format—Page Length (Number of Lines)
  b. Skip Perforation: determines the number of lines to be skipped at the end of every page.
  → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ SMDR Format—Blank Footer Length (Number of Lines)
  The page length should be at least four lines longer than the skip perforation length.

Explanation:



- SMDR data is not deleted even if the PBX is reset.
- If the PBX is reset during a conversation, the call will not be recorded on SMDR.
- When a call is made from an extension to a number in the Emergency Dial Table (→ 2.5.4.2 Emergency Call), the PBX can be programmed to record the call information on SMDR both immediately after the number is dialled and after the call ends. (Normally, the PBX records the call information only after the call ends.)
  → 19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR Options—◆ Emergency Call Notification
- The following calls are regarded as two separate calls for SMDR:
  – Calls before and after the flash/recall/EFA signal is manually sent during a conversation
  – Trunk-to-trunk calls by Call Transfer, FWD or DISA (recording each as "incoming call" and "outgoing call")

  – Incoming calls to a PDN or SDN extension.
- The PBX waits for a preprogrammed time period between the end of dialling and start of the SMDR timer for outgoing trunk calls. When the PBX has sent out all dialled digits to the telephone company and this timer expires, the PBX starts counting the call. A display PT shows the elapsed time of the call. The starting time and the total duration of the call are recorded on SMDR.
  → 10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Dial—Analogue CO Call Duration Start (s)
  If the reverse signal detection has been set (→ 2.5.4.5 Reverse Circuit), the PBX will start counting the call after detecting the reverse signal from the telephone company regardless of the above timer.
- If a call is transferred to an ICD group using Automatic Transfer, the condition code "TR" will not be recorded on SMDR (→ 2.12.1 Call Transfer).

[Host PBX Access Code]

- The dialled number including the Host PBX Access code will be recorded on SMDR only if the modified number setting is selected in the ARS setting for SMDR.
- When a Host PBX Access code is assigned to a trunk group, calls to extensions of the host PBX are not recorded on SMDR.
- A Host PBX Access Code can be used to record only long distance calls on SMDR when a trunk port is connected directly to the telephone company (not a host PBX). This is allowed when the long distance code (e.g., "0") is assigned as the Host PBX Access code. All local calls (e.g., calls that do not require a "0" to be dialled first) are treated as extensions of the telephone company and do not get recorded on SMDR, because in this case this PBX recognises the telephone company as the host PBX. Therefore, only long distance calls are recorded on SMDR.

[Output to a Telnet compatible Terminal Emulator]

- In order to activate a connection to a terminal emulator, the IP address of the mother board, port number, user ID ("SMDR"), and password must be entered.
- If a terminal emulator user incorrectly enters the user ID or password 3 times consecutively, an alarm will be sent and connection will not be possible for 10 minutes.
- Through system programming, it is possible to assign the PBX port number and password.
- The terminal emulator application must be running constantly. If the application is terminated, call records that occur after the termination will be recorded in the PBX's memory. However, if the number of call records exceeds the PBX's capacity, older records will be deleted. Also, when the application restarts or is reconnected, duplicated call records may be output.

[Using SMDR with applications]

SMDR data can also be monitored by applications such as Panasonic CA Call Accounting. For more information, see your application's documentation.

Installation Manual References

4.10 Connection of Peripherals

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Dial—Analogue CO Call Duration Start (s)
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—CO & SMDR—◆ Outgoing CO Call Printout (SMDR)
14.2 PBX Configuration—[6-2] Feature—Hotel & Charge
→Main—◆ SMDR for External Hotel Application 2—Printing Message 1–8
→Charge—◆ Charge Options—Currency
19.1 PBX Configuration—[11-1] Maintenance—Main

→SMDR
→SMDR Options

PT Programming Manual References

[802] SMDR Page Length
[803] SMDR Skip Perforation
[804] SMDR Outgoing Call Printing
[805] SMDR Incoming Call Printing

Feature Guide References

2.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)
2.2.2.7 Log-in/Log-out
2.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)
2.12.1 Call Transfer
2.16.1 Direct Inward System Access (DISA)
4.2.1 TIE Line Service
5.1.1 Class of Service (COS)
6.1 Capacity of System Resources

2.22.1.2 Syslog Record Management

Description

By connecting this PBX to a Syslog server over a LAN, it is possible to output local alarm information (major alarms/minor alarms) to a external PC.

Conditions

- To be able to use this feature, through system programming, it is required to enable this feature and register the IP address of the Syslog server.

PC Programming Manual References

7.3.2 Utility—Log—Syslog
27.3.2 Network Service—[3-2] Client Feature—Syslog

Feature Guide References

5.6.4 Local Alarm Information

2.22.2 Printing Message

Description

An extension user can select a message to be output on SMDR. Up to eight messages can be preprogrammed in the Printing Message table, and are available to all extensions connected to the PBX. A message can contain the "%" symbol, which requires a number to be entered in its place when the message is selected at an extension.
Depending on the content of the preprogrammed messages, this feature can be used to record a variety of information, which can be output on SMDR to, for example, a connected PC.
[Example]
If message 1 is preprogrammed as "Started work", and message 2 as "Finished work", employees can sign in by selecting message 1 when starting work, and sign out by selecting message 2 when finishing. A connected PC can then be used to generate employee work records.

Conditions

- Up to seven "%"s can be stored for each message.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Printing Message
14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Main—◆ SMDR for External Hotel Application 2—Printing Message 1–8

Feature Guide References

2.22.1.1 Station Message Detail Recording (SMDR)
6.1 Capacity of System Resources

User Manual References

1.12.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)

2.22.3 Call Charge Services

Description

The PBX receives a call charge signal during or after a conversation with an outside party. The call charge information is shown on the telephone display and SMDR.

1. **Call Charge Signal Services**
   The type of call charge service that is used by the PBX is decided by the type of signal received from the telephone company. The type of call charge signal received from the telephone company depends on the trunk of the outgoing call. The services for each available trunk type are as follows:

Trunk	Service
ISDN line	Advice of Charge (AOC) (→ 4.1.2.3 Advice of Charge (AOC))
E1 line	Meter Pulse

2. **Call Charge Display**
   – Up to eight digits including a decimal (e.g., 12345.78)
   – The decimal point position (the number of significant decimal digits) for currency is programmable.
   – Up to three currency characters are programmable. (e.g., EUR or € for Euro).
   – Through PC programming, you can select whether the currency characters or symbol are placed in front of or behind the call charge. (e.g., € 45.12 or 45.12 €)
3. **Margin/Tax Rate Assignment**
   It is possible to add a margin and a tax to the call charges. The call charge rate per meter indication is programmable on a trunk group basis.
   [Calculation Method]
   The margin or tax rate must consist of four digits, two digits before and after the decimal (xx.xx%). The calculation method used by the PBX varies, depending on whether the telephone company sends the meter indication or the actual call charge.
   a. Call charge with tax and margin in meter indication:

$$\frac{[\text{Meter indication received from the telephone company}] \times [\text{Call Charge Rate}] \times [1 + \text{Tax Rate}]}{[1 - \text{Margin Rate}]}$$

   b. Call charge with tax and margin in charge:

$$\frac{[\text{Charge received from the telephone company}] \times [1 + \text{Tax Rate}]}{[1 - \text{Margin Rate}]}$$

   The calculation result is rounded up to the least significant decimal digit.
4. **Total Call Charge**
   – A PT user can show the total call charges on the display.
   – The call charge is totalled on an extension, trunk, or verification code basis.
   – When a verification code is used, the call is charged on the verification code and not the extension that the call was made on.
5. **Budget Management**
   It is possible to limit telephone usage to a preprogrammed budget on each extension or verification code. For example, an extension in a rented office has a prepaid limit for telephone usage. If the amount of the call charge reaches the limit, the extension user cannot make further trunk calls. An extension assigned as the manager may increase the limit or clear the previous call charge (→ 2.7.2 Budget Management).
6. **Call Charge Management**
   An extension assigned as a manager can perform the following:
   a. Clear the call charges for each extension and verification code.

b. Clear the call charges of all extensions and verification codes.
c. View the call charges (Call Charge Reference) for each trunk, extension, or verification code.
d. Set the call charge rate for each trunk group.
e. Print out the total call charges for all extensions and verification codes.
f. Set a budget for each extension and verification code.

[Examples of Call Charge Reference]

```
*****************************************************
*    Charge Meter Print Out - Total & All CO      *
*****************************************************
Total Charge:   €00175.95

CO Line
   001:  €00194.00      002:  €00073.00      003:  €00161.00      004:  €00033.00
```

```
*****************************************************
*    Charge Meter Print Out - All Extensions      *
*****************************************************

 *775:  €00194.00      *102:  €00073.00      *776:  €00161.00

  104:  €00194.00       105:  €00073.00       106:  €00161.00       107:  €00033.00
```

Note

*: extension or verification code number

Conditions

[General]

- **Call Charge Reference by Call Charge Reference Button**
  A display telephone user can check the total call charge for his own extension using the Call Charge Reference button. A flexible button can be customised as the Call Charge Reference button.

[Pay Tone Service]

- It is possible to select whether the PBX starts counting the call charge from when the PBX detects the answer signal from the telephone company.

PC Programming Manual References

11.1.5 PBX Configuration—[3-1-5] Group—Trunk Group—Charge Rate
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Type
14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Charge
→◆ Margin & Tax—Margin Rate for "Telephone" (%)
→◆ Margin & Tax—Tax Rate for "Telephone" (%)
→◆ Charge Options—Digits After Decimal Point
→◆ Charge Options—Currency
→◆ Charge Options—Currency Display Position
→◆ Charge Options—Action at Charge Limit
→◆ Charge Options—Meter Start on Answer Detection

PT Programming Manual References

[010] Charge Margin
[011] Charge Tax
[012] Charge Rate per Unit
[130] Decimal Point Position for Currency
[131] Currency

Feature Guide References

2.7.6 Verification Code Entry
2.21.2 Flexible Buttons
6.1 Capacity of System Resources

User Manual References

1.12.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)
4.1.2 Manager Programming

2.23 Hospitality Features

2.23.1 Hospitality Features—SUMMARY

Description

This PBX has several features that support its use in a hotel-type environment, where extensions correspond to guest rooms.

Feature	Description & Reference
Room Status Control	An extension designated as the hotel operator can set the check-in status of rooms remotely. → 2.23.2 Room Status Control
Call Billing for Guest Room	Charges for calls from guest rooms can be logged and output as a guest bill. → 2.23.3 Call Billing for Guest Room
Remote Wake-up Call	An extension designated as the hotel operator can set a timed reminder for a room remotely. → 2.24.4 Timed Reminder
SMDR for External Hotel Application	Hospitality feature data, including check-in, check-out, and timed reminder times, can be output to SMDR for use in a PC-based hotel application. → 2.22.1.1 Station Message Detail Recording (SMDR)
Hospitality Mode for Unified Messaging	Extensions whose Unified Messaging mailboxes are set to hospitality mode can be restricted to using certain features, such as listening to messages and changing the mailbox owner name. → 3.2.1.21 Hospitality Mode

2.23.2 Room Status Control

Description

A PT with a 6-line display designated as a hotel operator extension can be used to view and set the Check-in/Check-out/Cleaned-up (Ready or Not Ready) status of guest rooms associated with extensions.
Any wired extension can be used as a room extension without special programming.
Flexible buttons on the hotel operator's extension can be set as Room Status Control buttons. The 3 types of Room Status Control buttons are as follows:

- **Check-in**
  Switches the status of selected room extensions from Check-out to Check-in.

  Telephone charges are cleared and Remote Extension Dial Lock is turned off, allowing calls to be made from the extension.

- **Check-out**
  Switches the status of selected room extensions from Check-in to Check-out.

  Room extension data, such as Timed Reminder or Last Number Redial data, is cleared, and Remote Extension Dial Lock is turned on, restricting some calls. This can be useful to prevent the room extension from being used when no guest is checked in.
  When checking a room extension out, the operator can enter customer charges such as minibar charges. A guest bill showing these charges, as well as call charges, can be printed. If necessary, the guest charge data entered can be edited later, and the bill reprinted.

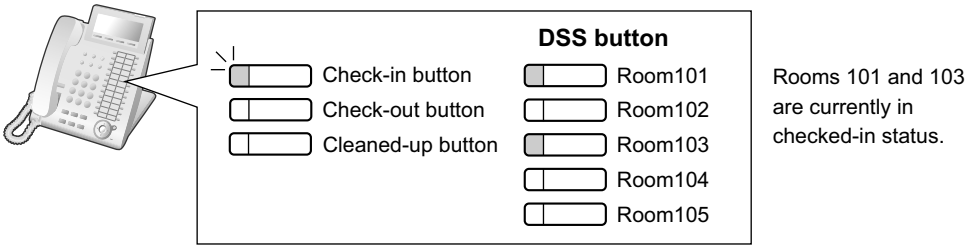
- **Cleaned-up**
  Switches the status of selected room extensions between Ready and Not Ready.

  When a guest checks out of a room, the room status becomes Checked-out and Not Ready. After the room has been cleaned, the status can be changed to Checked-out and Ready using this button. It is also possible to change the status back to Checked-out and Not Ready if necessary.

Room Status Control Mode
Pressing a Room Status Control button when the PT is idle allows the hotel operator extension to enter Room Status Control mode. When in Room Status Control mode, the corresponding Room Status Control button's light flashes red. The Room Status Control button that was pressed determines which room status each room extension can be switched to. For example, if the Check-in button was pressed, the Check-in button's light flashes red and the hotel operator can select which room extensions to check-in.
In addition, DSS buttons on the hotel operator's extension or a paired DSS Console show the room status of each extension as follows:

Light Pattern	Status
Off	Checked-out and Ready
Flashing Red	Checked-out and Not Ready
Red on	Checked-in

[Example Use: Checked-in Mode]



When in Room Status Control mode, the hotel operator's extension is treated as a busy extension, similar to when performing PT programming. Callers to that extension will hear a busy tone.
All other operations, including pressing other Room Status Control buttons, will be ignored. In addition, the lights of fixed and flexible buttons do not show their normal display pattern. In order to perform other operations, the hotel operator must exit Room Status Control mode.

Conditions

- SVM voice messages and messages left on the extension's Voice Mail (VM) will be cleared at Check-out.
- A maximum of four hotel operators can be assigned.
- Only one of each type of Room Status Control button can be assigned.
- Extensions associated with rooms must be one of the following types:
  PT, KX-UT SIP, general SIP, SLT, ISDN Extension
  PSs cannot be checked in and out as room extensions.
- It is recommended that the extension number of a room extension is the same or similar to the room number, for convenience.
- The previous guest's billing data is only cleared when an extension is set back to Check-in status. Thus, it is possible to edit guest charge data and reprint the bill at any time until another guest is checked in to the same room.
- If enabled through system programming, the check-in and check-out information is recorded on SMDR.

PC Programming Manual References

12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
14.2 PBX Configuration—[6-2] Feature—Hotel & Charge
→Main—◆ Hotel Operator—Extension 1–4
→Bill—◆ Checkout Billing—Billing for Guest

Feature Guide References

2.6.3 Last Number Redial
2.7.3 Extension Dial Lock
2.22.1.1 Station Message Detail Recording (SMDR)
2.24.4 Timed Reminder

User Manual References

1.12.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)

2.23.3 Call Billing for Guest Room

Description

Separately from SMDR, it is possible to output a record of calls along with charges (e.g., telephone charges, minibar, etc.), which can be used in billing a guest.

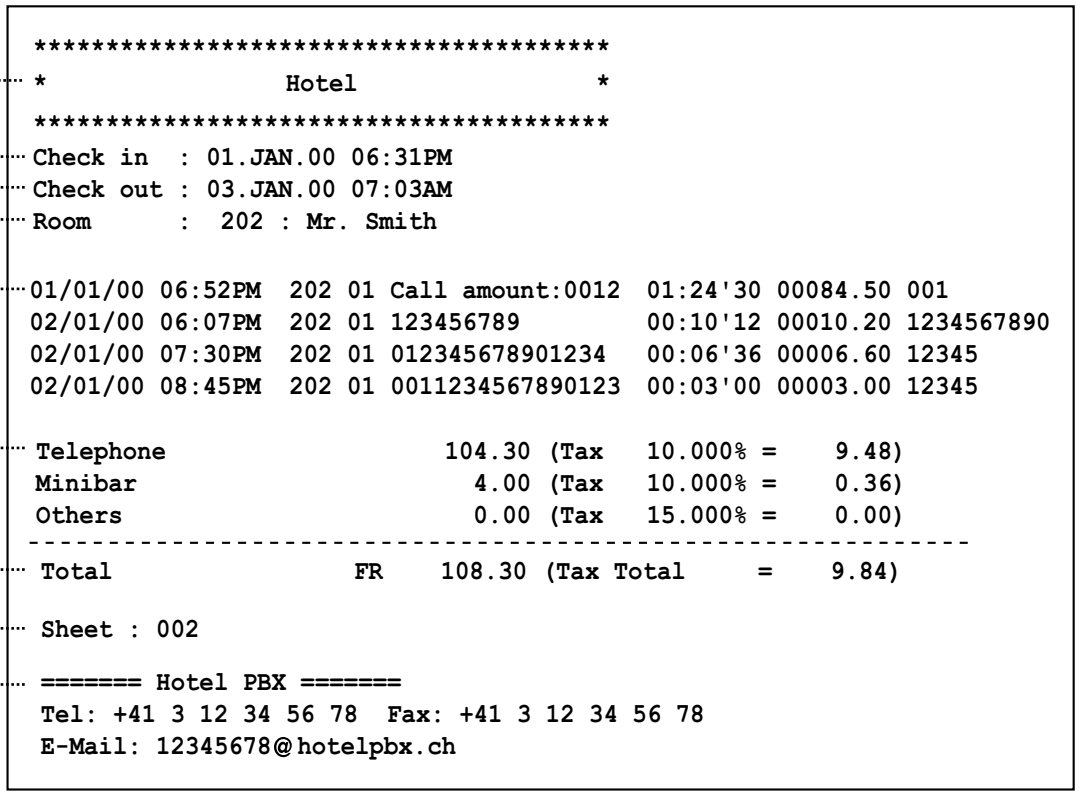
Charge Items

This feature provides three types of programmable charge items (Charge Item 1, Charge Item 2, and Charge Item 3) which can be used for billing guests for various services (e.g., telephone charges). Each charge item can be customised in the following ways:

– A name that appears on the call billing print out.
  14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Bill
  → ◆ Checkout Billing—Bill (SMDR) for "Telephone"
  → ◆ Checkout Billing—Bill (SMDR) for "Minibar"
  → ◆ Checkout Billing—Bill (SMDR) for "Others"
– A name that appears on the display telephone of the hotel operator.
  14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Bill
  → ◆ Checkout Billing—LCD for "Telephone"
  → ◆ Checkout Billing—LCD for "Minibar"
  → ◆ Checkout Billing—LCD for "Others"
– A tax rate.
  14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Charge
  → ◆ Margin & Tax—Tax Rate for "Telephone" (%)
  → ◆ Margin & Tax—Tax Rate for "Minibar" (%)
  → ◆ Margin & Tax—Tax Rate for "Others" (%)

Charge Item 1 can also be assigned a margin rate, which is useful for charging guests an additional rate for using the telephone services.
→ 14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Charge—◆ Margin & Tax—Margin Rate for "Telephone" (%)
It is possible to print out a bill for a guest. The bill will show the following items:

[Example of Call Billing Sheet]



```
                                        ****************************************
(1)                                     *                Hotel                 *
                                        ****************************************
(2)                                     Check in  : 01.JAN.00 06:31PM
(3)                                     Check out : 03.JAN.00 07:03AM
(4)                                     Room      :  202 : Mr. Smith

(5)                                     01/01/00 06:52PM  202 01 Call amount:0012  01:24'30 00084.50 001
                                        02/01/00 06:07PM  202 01 123456789         00:10'12 00010.20 1234567890
                                        02/01/00 07:30PM  202 01 012345678901234   00:06'36 00006.60 12345
                                        02/01/00 08:45PM  202 01 0011234567890123  00:03'00 00003.00 12345

(6)                                     Telephone                   104.30 (Tax   10.000% =    9.48)
                                        Minibar                       4.00 (Tax   10.000% =    0.36)
                                        Others                        0.00 (Tax   15.000% =    0.00)
                                        ------------------------------------------------------------
(7)                                     Total                 FR    108.30 (Tax Total     =    9.84)

(8)                                     Sheet : 002

(9)                                     ======= Hotel PBX =======
                                        Tel: +41 3 12 34 56 78  Fax: +41 3 12 34 56 78
                                        E-Mail: 12345678@hotelpbx.ch
```

1. A programmable title (e.g., hotel name).
   → 14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Bill—◆ SMDR for External Hotel Application—Header 1–3
2. The check-in time.
3. The check-out time.
   If the guest has already been checked out, the check-out time will be shown. If not, the time that the bill was printed will be shown.
4. The extension number and name.
5. A list of all calls made and call charges (using the same format as SMDR output Pattern B [→ 2.22.1.1 Station Message Detail Recording (SMDR)]).
6. The total charge for each charge item and tax, including the preprogrammed tax rate.
7. The combined charges of all three charge items, currency of the charge, and tax.
8. The sheet number (the number of times that the current guest's charge data has been printed out and then cleared).
9. A programmable footer (e.g., the contact information of the hotel).
   → 14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Bill—◆ SMDR for External Hotel Application—Footer 1–3

It is possible to select the language used on the guest bill.
→ 14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Bill—◆ SMDR for External Hotel Application—Language for Bill (SMDR)

Walking COS

If guests are given extension personal identification numbers (PINs), it is possible for calls made from other extensions (e.g., an extension in a hotel restaurant) to be charged to the guest's room extension by using the Walking COS feature (→ 2.7.5 Walking COS).

→ 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ Extension PIN

Conditions

- If the total number of call records exceeds 90 % of available memory, call records from the extension with the largest number of records will be automatically printed out, and the records printed out will be combined in memory into one aggregate record to save space.

PC Programming Manual References

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ Extension PIN
14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Bill—◆ Checkout Billing—LCD for "Telephone"
14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Charge
→◆ Margin & Tax—Margin Rate for "Telephone" (%)
→◆ Margin & Tax—Tax Rate for "Telephone" (%)
→◆ Margin & Tax—Tax Rate for "Minibar" (%)
→◆ Margin & Tax—Tax Rate for "Others" (%)

Feature Guide References

2.6.3 Last Number Redial
2.7.3 Extension Dial Lock
2.22.1.1 Station Message Detail Recording (SMDR)
2.24.4 Timed Reminder
6.1 Capacity of System Resources

User Manual References

1.12.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)

2.24 Extension Controlling Features

2.24.1 Extension Personal Identification Number (PIN)

Description

Each extension user can have his own PIN through system programming or personal programming (Extension PIN [Personal Identification Number]) to set features or access his own telephone remotely.
The following features cannot be used without the PIN:

a. Live Call Screening (LCS)*1 (→ 3.2.2.16 Live Call Screening (LCS))
b. Display Lock (→ 2.6.4 Speed Dialling—Personal/System, 2.16.3 Built-in Simplified Voice Message (SVM), 2.19.2 Incoming Call Log)
c. Walking Extension (→ 2.24.3 Walking Extension Features)
d. Extension Dial Lock (→ 2.7.3 Extension Dial Lock)
e. Walking COS (→ 2.7.5 Walking COS)
f. Walking COS through DISA (→ 2.16.1 Direct Inward System Access (DISA))

*1 If an extension user has assigned an extension PIN, this feature cannot be used without the PIN.

Conditions

CAUTION

There is a risk that fraudulent telephone calls will be made if a third party discovers a personal identification number (PIN) (verification code PIN 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.

- **Extension PIN Lock**
  If the wrong PIN is entered three times, the line will be disconnected. If the wrong PIN is entered a preprogrammed number of times successively, that extension will become locked, and even entering the correct PIN will not unlock it. Only an extension assigned as the manager can unlock it. In this case, the PIN will be unlocked and cleared. This feature is also known as Station Password Lock.
- **Remote Extension PIN Clear**
  If an extension user forgets his PIN, a manager can clear the PIN. Then the extension user can assign a new PIN.
- **Extension PIN Display**
  It is possible to select whether to show the extension PIN on the display through system programming. By default, it is shown as dots.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Extension PIN—Lock Counter
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Extension PIN Set / Cancel
10.9 PBX Configuration—[2-9] System—System Options—Option 1—◆ PT LCD—Password / PIN Display
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ Extension PIN

12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main—◆ Extension PIN

PT Programming Manual References

[005] Extension Personal Identification Number (PIN)

User Manual References

3.1.2 Settings on the Programming Mode
4.1.2 Manager Programming

2.24.2 Extension Feature Clear

Description

Extension users can clear all the following features set on their own telephone at once. This feature is also known as Station Programme Clear.

Features	After Setting
Absent Message	**Off**
BGM	**Off**
FWD*/DND*	**Off**
Call Pickup Deny	**Allow**
Call Waiting*	**Disable** (In Canada, the default setting is "Enable" [Call Waiting tone].)
Data Line Security	**Off**
Executive Busy Override Deny	**Allow**
Log-in/Log-out	**Log-in**
Message Waiting	**All messages left by other extensions will be cleared.**
Paging Deny	**Allow**
Parallelled Telephone	**Paired SLT will ring**
Hot Line*	**Off**
Timed Reminder	**Cleared**

Note

The features with "*" can be programmed not to be cancelled by this feature.

Conditions

- Extension Dial Lock (→ 2.7.3 Extension Dial Lock) and the extension personal identification number (PIN) (→ 2.24.1 Extension Personal Identification Number (PIN)) will not be cleared by this feature.
- **For Users in Canada only**
  If dial tone 2 is heard after Extension Feature Clear:
  After performing Extension Feature Clear, Call Waiting will be enabled if "**Extension Clear: Call Waiting**" is set to "**Clear**" through system programming. In this case, dial tone 2 will be heard when going off-hook. (→ 2.25.1 Dial Tone)

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Extension Feature Clear
10.9 PBX Configuration—[2-9] System—System Options—Option 2
→◆ Extension Clear—Call Waiting
→◆ Extension Clear—Fwd/DND
→◆ Extension Clear—Hot Line (Pick-up Dial)

User Manual References

1.9.14 Clearing Features Set at Your Extension (Extension Feature Clear)

2.24.3 Walking Extension Features

2.24.3.1 Walking Extension

Description

It is possible to use any extension and have your extension settings available to you. Settings such as extension number, one-touch dialling memory, and COS are all available to you at the new location. This feature is also known as Walking Station.
[Example] This feature is useful when:

- Moving location
- There is no specific desk for your use.

Conditions

- This feature allows extension settings to be switched between PTs and SLTs. Moving between tenants is also possible.
- Incoming calls to your extension will also reach you at your new location.
- An extension personal identification number (PIN) is required to use this feature. (→ 2.24.1 Extension Personal Identification Number (PIN))
- If a DSS Console is connected to a PT and the DSS Console is continuously used with the PT after Walking Extension has been activated, the new extension number of the PT must be assigned as the paired extension through system programming.
- If PC programming is being performed for extensions whose extension settings are being transferred by the Walking Extension feature, the Walking Extension feature may not work properly (→ 5.5.2 PC Programming).
- If this feature is performed using an extension with a Bluetooth® wireless headset connected, the Bluetooth device cannot be used with the new extension. To use the Bluetooth wireless headset, register it to the new extension.
- This feature is not available for SIP extensions.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Walking Extension
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ Extension PIN

PT Programming Manual References

[007] DSS Console Paired Telephone

User Manual References

1.13.1 Walking Extension

2.24.3.2 Enhanced Walking Extension

Description

It is possible to use the Walking Extension feature with extensions in Service-in (functions normally) and Service-out (cannot make trunk calls or receive calls) modes, allowing extensions and locations (i.e., telephones) to be utilised more efficiently. Extensions can be set to Service-out mode when the extension user is not using the extension or to prevent calls from being made or received when no one is using the location. The extension can then be changed to Service-in mode when the extension user wishes to use the extension. Enhanced Walking Extension can be utilised as follows:

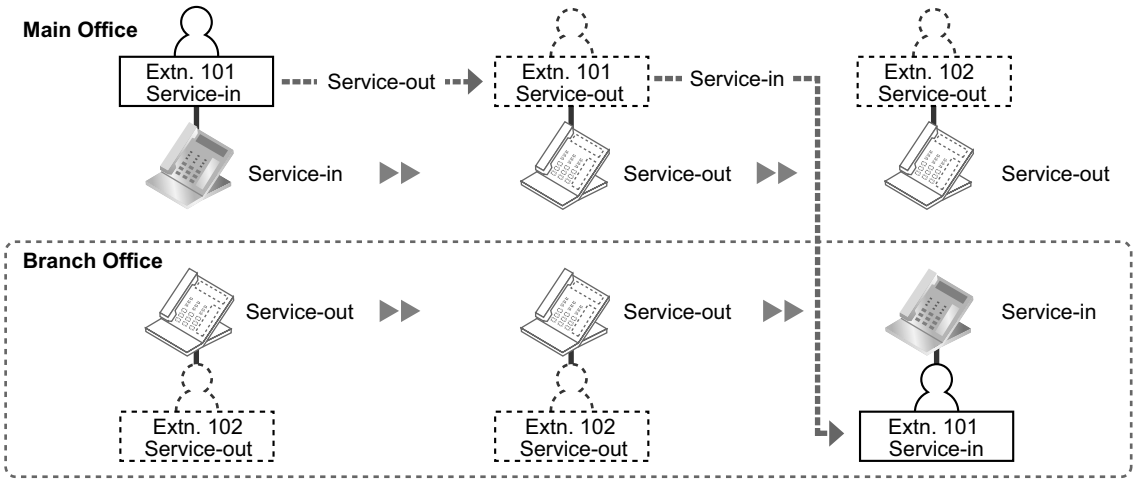
– **When a single extension user uses multiple telephones**
Extension users can switch locations with a Service-out extension. This allows extension users to use their settings at another location while the previous location is in service-out mode. This is ideal for when extension users need to work at multiple locations, such as another department, branch office, or at home.
– **When multiple extension users use the same telephone**
Extension users can change the service status of their Service-out extension to Service-in mode and switch locations with another extension with a simple operation. This is ideal for when the same telephone is used by multiple extension users who work in shifts.

Service-out mode

When an extension is in Service-out mode, the DND and Extension Lock features are set on the extension, preventing the extension from making trunk calls and receiving calls.

[Example]

Extension settings can be used at other locations as follows:

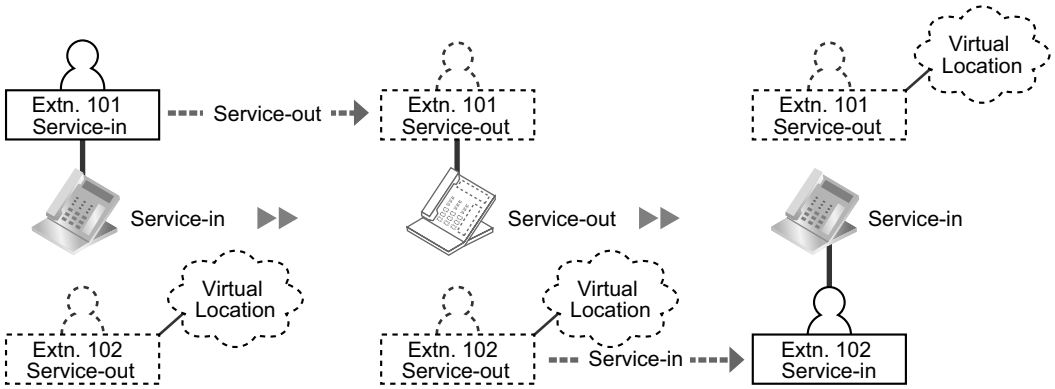


Explanation:
The extension user of extension 101 changes to Service-out mode at the main office. He then changes his extension to Service-in mode and switches extension settings at the branch office.

Virtual Locations

Instead of assigning all extensions to telephones, it is possible to store unused extensions (i.e., Service-out mode) on a preinstalled extension card (i.e., settings are made but no physical location is utilised). When the extension needs to be used, the extension can switch locations and service status with an extension on a physical location (i.e., a location with a telephone).
[Example]

Extension settings can be switched allowing multiple extension users to use the same telephone as follows:



Explanation:
The extension user of extension 101 changes to service out mode. The extension user of extension 102 switches extension settings and changes his extension to Service-in mode.

Conditions

- This feature is not available for PSs, ISDN extensions, or SIP extensions.
- An extension personal identification number (PIN) is required to use this feature. (→ 2.24.1 Extension Personal Identification Number (PIN))

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Walking Extension

Feature Guide References

2.24.3.1 Walking Extension

User Manual References

1.13.2 Enhanced Walking Extension

2.24.4 Timed Reminder

Description

An extension can be preset to ring at a certain time, to act as a wake-up call or reminder. This feature can be programmed to activate only once, or daily. If the user answers the alarm call, a prerecorded voice message will be heard. If a message is not assigned, a special dial tone (dial tone 3) will be heard.
Timed reminders can be set in one of two ways:

- By the extension user, from his own extension.
- Remotely, by the hotel operator (Remote Wake-up Call)

Conditions

- Be sure that the PBX clock works.
- Only one timed reminder can be set for an extension at a time. Setting a new reminder clears the previous reminder. If both the extension user and the hotel operator set a timed reminder for the same extension, the timed reminder that was set most recently is effective.
- **Programmable Time**
  The Alarm Ringing Duration time, the number of alarm repeat times, and intervals are programmable through system programming.
- **To use the voice message feature:**
  An extension assigned as the manager can record messages (→ 2.30.2 Outgoing Message (OGM)). A different message can be assigned for each time mode (day/lunch/break/night) (→ 5.1.4 Time Service).

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf
→◆ Timed Reminder—Repeat Counter
→◆ Timed Reminder—Interval Time (x10s)
→◆ Timed Reminder—Alarm Ringing Duration (x10s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ Remote Timed Reminder (Remote Wakeup Call)
→◆ Timed Reminder Set / Cancel
10.8.3 PBX Configuration—[2-8-3] System—Ring Tone Patterns—Call from Others—◆ Timed Reminder—Ring Tone Pattern Plan 1–8
10.9 PBX Configuration—[2-9] System—System Options—Option 1—◆ PT LCD—Time Display
13.3.1 PBX Configuration—[5-3-1] Optional Device—Voice Message—DISA System—Option 2—◆ Timed Reminder Message—Day, Lunch, Break, Night
14.2 PBX Configuration—[6-2] Feature—Hotel & Charge—Main—◆ SMDR for External Hotel Application 1—Timed Reminder (Wake-up Call)
19.1 PBX Configuration—[11-1] Maintenance—Main—SMDR—◆ Print Information—Timed Reminder (Wake-up Call)

Feature Guide References

2.23.2 Room Status Control

User Manual References

1.9.1 Setting the Alarm (Timed Reminder)
1.12.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)

2.25 Audible Tone Features

2.25.1 Dial Tone

Description

The following distinctive dial tones inform extensions about features activated on their extensions. Each dial tone type has two frequencies (e.g., dial tone 1A and dial tone 1B).

Type	Description
Tone 1A/1B	A normal dial tone is heard when: **a.** No features listed for dial tones 2 through 4 has been set, or **b.** ARS is used.

Tone 2A/2B	This tone is heard when:
	• There are messages that have previously been listened to and no new messages for the Built-in Simplified VoiceMessage (SVM) feature.*1
	• Any of the features below are set.
	  • Absent Message
	  • BGM
	  • FWD
	  • Call Pickup Deny
	  • Call Waiting
	  • DND
	  • Extension Dial Lock
	  • Executive Busy Override Deny
	  • Hot Line
	  • Timed Reminder

Tone 3A/3B	This tone is heard when:
	• A called PS is being searched for.
	• The recording time used by the Built-in Simplified Voice Message (SVM) feature reaches the limit. *2
	• Any of the features below are performed.
	  • Account Code Entry
	  • Consultation Hold
	  • Answering a Timed Reminder call with no message
	  • Answering a sensor call

Tone 4A/4B	This tone is heard when new messages have been recorded for the extension.

*1 Active when distinctive dial tones are enabled. TONE 1 is heard when distinctive dial tones are disabled.
*2 Active even when distinctive dial tones are disabled.

Conditions

- **Dial Tone Type A/B**
  It is possible to select dial tone type A or B for dial tones 1 through 4. If "**Type A**" is selected, all dial tones 1 through 4 will become dial tone type A.

The dial tone type for the ARS feature can be selected separately. If "**Type A**" is selected for the ARS, dial tone 1A will be heard. If "**Type B**" is selected, dial tone 1B will be heard.

- **Dial Tone Patterns**
  All dial tone patterns have a default (→ 6.2.1 Tones/Ring Tones).
- Only dial tone 1 is sent to the extensions in a VM (DPT/DTMF) group (2.28.1 Voice Mail (VM) Group).

PC Programming Manual References

10.9 PBX Configuration—[2-9] System—System Options—Option 3
→◆ Dial Tone—Distinctive Dial Tone
→◆ Dial Tone—Dial Tone for Extension
→◆ Dial Tone—Dial Tone for ARS

2.25.2 Confirmation Tone

Description

At the end of feature operations, the PBX confirms the success of the operation by sending a confirmation tone to extension users.

Type	Description
Tone 1	**a.** Sent when the setting is accepted. **b.** Sent when a call is received in voice-calling mode (Alternate Receiving—Ring/Voice). The caller's voice will be heard after the tone.
Tone 2	**a.** Sent from an external paging device or an extension before being paged. **b.** Sent when a call is received in Hands-free Answerback mode.
Tone 3-1	**a.** Sent before a conversation is established when using the Paging feature. **b.** Sent when a conversation is established with the extension in the following modes after the call making operation: • Hands-free Answerback mode • Voice-calling mode (Alternate Receiving—Ring/Voice) **c.** Sent when making a call to or from a doorphone.
Tone 3-2	Sent just before a conversation is established when accessing the following features by the feature numbers: • Call Park Retrieve • Call Pickup • Hold Retrieve • Paging Answer • TAFAS
Tone 4-1	Sent when moving from a two-party call to a three-party call. (e.g., Executive Busy Override, Conference, Privacy Release, Two-way Record.)
Tone 4-2	Sent when moving from a three-party call to a two-party call. (e.g., Executive Busy Override, Conference, Privacy Release, Two-way Record.)
Tone 5	Sent when a call is placed on hold (including Consultation Hold).

Conditions

- **Confirmation Tone Patterns**
  All confirmation tone patterns have a default (→ 6.2.1 Tones/Ring Tones).
- It is possible to eliminate each tone.

PC Programming Manual References

10.9 PBX Configuration—[2-9] System—System Options—Option 3
- →◆ Confirmation Tone—Tone 1 : Called by Voice
- →◆ Confirmation Tone—Tone 2 : Paged / Automatic Answer
- →◆ Confirmation Tone—Tone 3-1 : Start Talking after Making Call / Call from DOORPHONE
- →◆ Confirmation Tone—Tone 3-2 : Start Talking after Answering Call
- →◆ Confirmation Tone—Tone 4-1 : Start Conference
- →◆ Confirmation Tone—Tone 4-2 : Finish Conference
- →◆ Confirmation Tone—Tone 5 : Hold

2.26 Computer Telephony Integration (CTI) Features

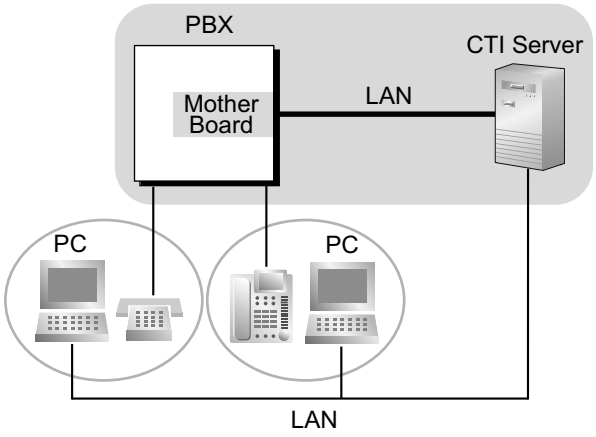
2.26.1 Computer Telephony Integration (CTI)

Description

The PBX supports a CTI interface using the LAN port of the mother board. The CTI interface allows extension users to make or receive calls with advanced features:

– Extension users can make calls easily from a phone book in their PC.
– When an extension user receives an incoming call, detailed caller information can be displayed on the PC automatically.

A PC and CTI server application software, such as Panasonic Communication Assistant (CA), are required to use CTI features. The PC running the application monitors the status of the PBX and controls the PBX via the CTI Server.



Conditions

- CTI application software must be installed on the connected PC. In addition, KX-NSF101 (Activation Key for CTI interface) is required to use CTI applications other than CA.
- CTI support for SIP extensions is available only for KX-UT series SIP phones.
- **Application Programming Interface (API)/Protocol**

Type	API/Protocol
Third Party Call Control	• ECMA CSTA Phase 3 • TAPI 2.1

- Only one CTI server can connect to the PBX at a time.
- For details about specific CTI features, refer to the manual for your CTI application software.
- When using a Panasonic TSP, refer to "Before Installing" in the KX-series TSP Installation Manual.

Installation Manual References

4.10 Connection of Peripherals

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Dial Information (CTI)
10.9 PBX Configuration—[2-9] System—System Options—Option 6 (CTI)

2.26.2 CA (Communication Assistant)

Description

Panasonic Communication Assistant (CA) is a CTI application usable with any telephone. A CTI server is not required to use CA. CA Client has 4 operating modes: Basic-Express, Pro, Supervisor, and Operator Console.

- **Basic-Express Mode:**
  Only basic features are available, such as call control.
- **Pro Mode:**
  A pro user can see the presence (phone status and absent message) of other extensions.
- **ICD Group Supervisor Mode:**
  A supervisor can use this feature to monitor users within an ICD group from a PC.
- **Operator Console Mode:**
  An operator or secretary can manage and redirect multiple calls simultaneously with a graphical interface.

Class of Service (COS) Settings

The following CA features can be disabled on a COS basis via system programming:

– Chat
– ICD Group Log Out

Microsoft® Outlook® Integration

CA users who have Microsoft Outlook installed can use CA features, such as making and answering calls, directly from Outlook.
Users who have Unified Messaging mailboxes can also use Microsoft Outlook to access their messages through the e-mail interface (→ 3.3.1 Integration with Microsoft Outlook).

Conditions

- Activation keys are required to enable the application, some of which are preinstalled on the mother board.
- CTI support for SIP extensions is available only for KX-UT series SIP phones.
- For information about the required CA version, refer to the corresponding documentation for CA.
- For details, refer to the documentation for CA.

PC Programming Manual References

9.2.2 PBX Configuration—[1-1] Configuration—Slot—System Property—Site—Port Number—◆ Built-in Communication Assistant Server
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—CA
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 9—◆ Built-in Communication Assistant
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 9—◆ Built-in Communication Assistant

Feature Guide References

5.1.1 Class of Service (COS)

2.27 Cellular Phone Features

2.27.1 Cellular Phone Features—SUMMARY

Description

This PBX provides features to support the use of cellular phones and other outside destinations with the PBX. Calls can be forwarded from virtual PSs to outside destinations such as cellular phones, and then answered as if the user was at an extension within the PBX. Also, when the receiving extension directly forwards the call to a cellular phone, the cellular phone can use PBX extension features in the same way as if it called the PBX directly.
The following features can be used with cellular phones and other outside extensions:

Feature	Description & Reference
Outside Destinations in Incoming Call Distribution Group	Up to 4 cellular phones can be assigned as members of an Incoming Call Distribution (ICD) Group, and receive calls to the group. → 2.2.2.3 Outside Destinations in Incoming Call Distribution Group
Cellular Phone XDP Parallel Mode	A PT user can set up to 4 cellular phones to ring in parallel for incoming calls. → 2.2.2.3 Outside Destinations in Incoming Call Distribution Group
Parallel Ringing When Forwarding to Trunk	When an unanswered call is forwarded to an outside line, such as a cellular phone, the forwarding extension's phone will continue ringing until the forwarded call is answered at either phone. → 2.3.2 Call Forwarding (FWD)
DISA Automatic Walking COS	Registered cellular phones are automatically recognised as PBX extensions when calling through DISA. → 2.16.1 Direct Inward System Access (DISA)
DISA Call Transfer From Outside Destination	A cellular phone user who answers a trunk call forwarded from the PBX using DISA can transfer that call to an extension (including over a TIE connection) or to an outside party. It is also possible to establish a Conference call, perform Call Splitting, and page with a call on hold to transfer the call. → 2.16.1 Direct Inward System Access (DISA)
DISA Call Transfer to outside user	From the transfer destination (including the outside party), it is possible to establish a Conference call, perform Call Splitting, and page with a call on hold to transfer the call. → 2.16.1 Direct Inward System Access (DISA)

Conditions

- KX-NSE101, KX-NSE105, KX-NSE110, or KX-NSE120 (Activation Key for Mobile Extension) is required to use these features. One activation key is required for each extension that will use these features. Also, the **Mobile Extension** setting for each extension must be set to **Enable**.
- Call disconnection detection
  When LCOT is used for the trunk, the system cannot detect call disconnection. Therefore, the system disconnects the trunk side after transferring the call (unscreened transfer).

PC Programming Manual References

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 9—◆ Mobile Extension
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 9—◆ Mobile Extension

Feature Guide References

5.2.4.6 Virtual PS

2.28 Voice Mail Features

2.28.1 Voice Mail (VM) Group

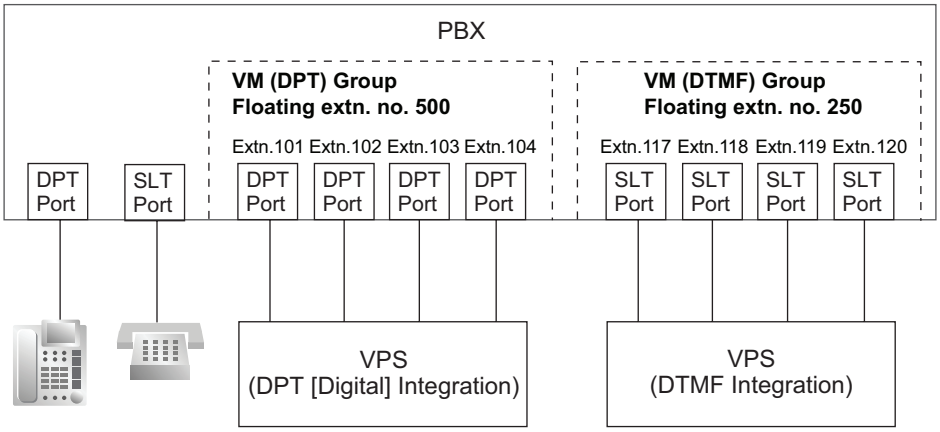
Description

A VPS can be connected to extension ports of the PBX. The extension ports make a group, called a VM group. This group has a floating extension number. The VM group can be the destination for redirected calls and incoming calls. When receiving a call, the VPS can greet the caller offering them the option to leave a message or dial a number to reach the desired party. The VPS can record the message for each extension and leave notification on the corresponding extension, if the called extension is not able to answer calls.

1. VM Group Type

Type	Description
VM (DTMF) Group	A group of SLT ports which use the Voice Mail DTMF Integration features. A maximum of 2 groups can be assigned.
VM (DPT) Group	A group of DPT ports which use the Voice Mail DPT (Digital) Integration features. • A maximum of 2 KX-TVM systems can be connected to the PBX. • A maximum of one VM (DPT) group per KX-TVM can be assigned. • A maximum of 12 ports (24 channels) of the VPS can form each group.

[Example]



VM (DTMF) Group Assignment:

The VPS is connected to the SLT ports of the PBX. These SLT ports, as well as VM (DTMF) Group settings, must be configured to allow DTMF Integration, as shown in the following [Programming Example of Extension Port] and [Programming Example of VM (DTMF) Group].

VM (DPT) Group Assignment:

The VPS is connected to the DPT ports of the PBX. These DPT ports, as well as VM (DPT) group settings, must be configured to allow DPT (Digital) Integration, as shown in the following [Programming Example of Extension Port] and [Programming Example of VM (DPT) Group].

[Programming Example of Extension Port]

Slot	Port	Port Type	Extn. No.	DPT Property			
				Type*1	Unit No. of VPS*2	Port No. of VPS*3	
3	1	DPT	101	VM (DPT)	1	1	VM (DPT) Group 1
3	2	DPT	102	VM (DPT)	1	2	
:	:	:	:	:	:	:	
4	1	S-Hybrid	201	VM (DPT)	2	1	VM (DPT) Group 2
4	2	S-Hybrid	202	VM (DPT)	2	2	
:	:	:	:	:	:	:	
5	1	SLT	301	Not assignable	Not assignable	Not assignable	VM (DTMF) Group 1
5	2	SLT	302	Not assignable	Not assignable	Not assignable	
:	:	:	:	:	:	:	
6	1	SLT	401	Not assignable	Not assignable	Not assignable	VM (DTMF) Group 2
6	2	SLT	402	Not assignable	Not assignable	Not assignable	

*1 →9.22 PBX Configuration—[1-1] Configuration—Slot—Extension—Port Property—◆ DPT Type—Type →[601] Terminal Device Assignment

*2 →9.22 PBX Configuration—[1-1] Configuration—Slot—Extension—Port Property—◆ DPT Type—VM Unit No.

*3 →9.22 PBX Configuration—[1-1] Configuration—Slot—Extension—Port Property—◆ DPT Type—VM Port No.

[Programming Example of VM (DTMF) Group]

VM (DTMF) Group No.	Floating Extension No.	Group Name	Service Mode	Extension No. of Port Connected to VPS Port		
				1	2	…
1	300	Company C	AA	301	302	…
2	400	Company D	VM	401	402	…

→ 11.12.2 PBX Configuration—[3-12-2] Group—VM (DTMF) Group—Group Settings

[Programming Example of VM (DPT) Group]

VM (DPT) Group No.	Floating Extension No.*1	Group Name*2
1	500	Company A
2	200	Company B

*1 →11.12.2 PBX Configuration—[3-12-2] Group—VM (DTMF) Group—Group Settings—◆ Floating Ext. No. →[660] UM Group Floating Extension Number

*2 →11.11.2 PBX Configuration—[3-11-2] Group—VM (DPT) Group—Unit Settings—◆ Group Name

2. Incoming Calls to VM Group

When incoming calls are received at the floating extension number of the VM group, calls will hunt starting at the lowest VM port number. In this case, the FWD and DND settings (→ 2.3.1 Call Forwarding (FWD)/ Do Not Disturb (DND)—SUMMARY) for each extension port are disregarded.

It is programmable whether the calls queue when all extension ports in the group are busy through system programming. If the queuing is disabled through system programming, the call will be redirected to the intercept destination of the head member extension of the VM (DPT) group.

Conditions

- It is possible to call an extension (extension port) in a VM group directly. If the calls are routed directly to the extension in the group, it is possible to enable some features (e.g., FWD, Idle Extension Hunting) on the extension in the group.
  The Voice Mail DTMF/DPT (Digital) Integration (e.g., command transmit) is also available on the extension.
- **One-touch Voice Mail Feature Access**
  It is possible to assign a One-touch Dialling button for direct access to a Voice Mail feature. (→ 2.6.2 One-touch Dialling) For example, to access a mailbox (mailbox number 123) of the VPS (extension number 165) directly, assign "**165**#6**123**" to a One-touch Dialling button. When pressing this button, the outgoing message (OGM) of the mailbox will be heard.
- All ports in a VM (DPT) group must be connected to a single DHLC or DLC card.

PC Programming Manual References

9.22 PBX Configuration—[1-1] Configuration—Slot—Extension—Port Property
→◆ DPT Type—Type
→◆ DPT Type—VM Unit No.
→◆ DPT Type—VM Port No.
11.11.1 PBX Configuration—[3-11-1] Group—VM (DPT) Group—System Settings
11.11.2 PBX Configuration—[3-11-2] Group—VM (DPT) Group—Unit Settings
11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings
11.12.2 PBX Configuration—[3-12-2] Group—VM (DTMF) Group—Group Settings

PT Programming Manual References

[601] Terminal Device Assignment
[660] UM Group Floating Extension Number

Feature Guide References

2.28.2 Voice Mail DTMF Integration
2.28.3 Voice Mail DPT (Digital) Integration
6.1 Capacity of System Resources

2.28.2 Voice Mail DTMF Integration

Description

The PBX and the VPS connected to the PBX can transmit commands using DTMF signals to each other. The PBX sends preprogrammed commands using DTMF to the VPS automatically to change the answering service between Voice Mail (VM) service mode and Automated Attendant (AA) service mode or to inform the extension status (e.g., busy). The VPS sends the commands to the PBX like an SLT.
The following answering services and features are available:

1. **Voice Mail (VM) Service Mode**
   When a caller reaches the VPS, the VPS greets and guides the caller to leave a voice message for a specified mailbox.

   [Example]

   The VPS sends the message to the caller, "Thank you for calling Panasonic. Please enter the mailbox number of the person you wish to leave your message for."

   ↓

   The caller dials the mailbox number. Then, the dialled number is sent to the VPS via the PBX.

   ↓

   The VPS sends the personal greeting to the caller, "You have reached Mike's voice mail. I am sorry I cannot take your call right now. Please leave a message and I will call you back."

   ↓

   The caller leaves a message.

   If the call reroutes to the floating extension number of the VM (DTMF) Group or the extension in the VM (DTMF) group by such as the FWD feature, when the VPS answers the call, the PBX will dial the mailbox number of the corresponding group or extension and any other digits required to the VPS automatically using the caller-dialled number (Follow on ID). In this case, the caller can reach a mailbox without knowing the mailbox number.

   [Available Features for Follow on ID]

   a. FWD to a Mailbox
   b. Intercept Routing to a Mailbox
   c. Call Transfer to a Mailbox
   d. Listening to a Message in a Mailbox
2. **Automated Attendant (AA) Service Mode**
   The VPS greets and guides the caller to the desired extension directly without operator assistance.
3. **VM → AA Service, AA → VM Service**
   It is possible to switch the service mode assigned on the VPS port, from the VM service to AA service or vice versa.

System Explanation

1. **Service Mode Assignment**
   Assign the service mode, VM service or AA service, to the VM (DTMF) group to correspond with the assignment of the VPS.
   → 11.12.2 PBX Configuration—[3-12-2] Group—VM (DTMF) Group—Group Settings—◆ Type
2. **DTMF Command Assignment**
   Assign the DTMF command to suit the VPS settings.

11.12.2 PBX Configuration—[3-12-2] Group—VM (DTMF) Group—Group Settings
→ ◆ VM DTMF Command—Recording Message
→ ◆ VM DTMF Command—Listening Message
→ ◆ VM DTMF Command—Switching to AA
→ ◆ VM DTMF Command—Switching to VM

	Command (Default)
Switching to VM	#6
Switching to AA	#8
Recording message	H
Listening message	✳H

Note

H = Mailbox Number

3. VM Service

a. FWD to a Mailbox of the VPS

The PBX sends a mailbox number of the corresponding extension to the VPS when a call is forwarded from an extension to the VPS. Therefore the caller can leave a message for the called extension without knowing the mailbox number.

[FWD to the VPS Sequence Selection]

If an extension user sets FWD to the VPS, any incoming call will be forwarded to the VPS. It is also possible to send the AA command, even in the VM service mode, when calls are forwarded so that the caller can be directed to an extension rather than a mailbox. This can be performed by selecting "**AA**" through system programming.

→ 11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings—◆ Others—FWD to the VPS Sequence

Parameter	Transmitted Command	
	In AA Service Mode	**In VM Service Mode**
Answer by Mailbox (Default)	Switching to VM command + Recording message command (#6 + H [H = Mailbox No.])	Recording message command (H [H = Mailbox No.])
AA	Switching to AA command (#8)	Switching to AA command (#8)
None	(DTMF commands are not sent. Work with default of the VPS.)	

b. Intercept Routing to a Mailbox of the VPS

The PBX sends a mailbox number of the corresponding extension to the VPS when a trunk call is intercepted from an extension to the VPS. Therefore the caller can leave a message for the called extension without knowing the mailbox number.

[Intercept Routing to the VPS Sequence Selection]

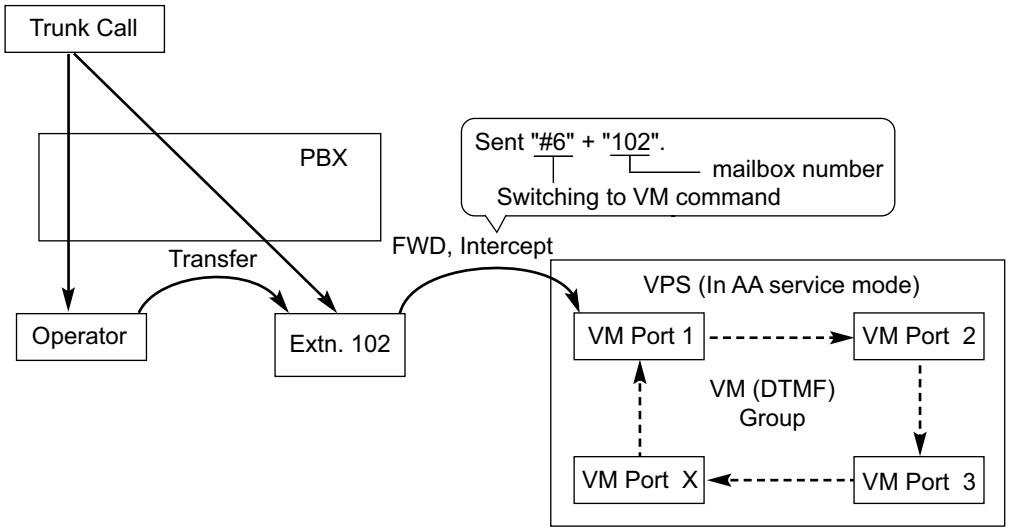
If an extension user sets Intercept Routing to the VPS, the intercepted trunk calls will be redirected to the VPS.

It is also possible to send the AA command, even in the VM service mode, when calls are intercepted so that the caller can be directed to an extension rather than a mailbox. This can be performed by selecting "**AA**" through system programming.

→ 11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings—◆ Others—Intercept Routing to the VPS Sequence

Parameter	Transmitted Command	
	In AA Service Mode	In VM Service Mode
Answer by Mailbox	Switching to VM command + Recording message command (#6 + H [H = Mailbox No.])	Recording message command (H [H = Mailbox No.])
AA	Switching to AA command (#8)	Switching to AA command (#8)
None (Default)	(DTMF commands are not sent. Work with default of the VPS.)	

[Example of a) & b)]



c. Voice Mail (VM) Transfer Button

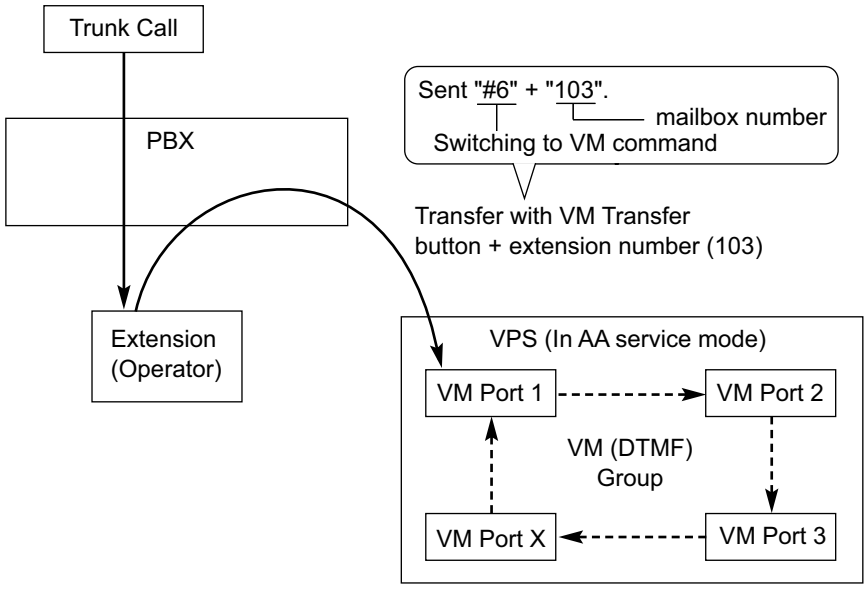
By pressing the Voice Mail (VM) Transfer button during a call, an extension user can transfer a call to a mailbox of the VPS so that the caller can leave a message in the mailbox of the desired extension. When the extension user presses this button and enters the desired extension number, the PBX will transfer the call to the VM group and dial the mailbox number of the desired extension with the required command (after the VPS answers the call). Therefore the caller can leave a message without knowing the mailbox number.

12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→ ◆ Type
→ ◆ Extension Number (for Voice Mail Transfer)
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→ ◆ Type
→ ◆ Extension Number (for Voice Mail Transfer)
12.3 PBX Configuration—[4-3] Extension—DSS Console
→ ◆ Type
→ ◆ Extension Number (for Voice Mail Transfer)

[Performance of Pressing the VM Transfer Button and Entering an Extension Number]

Transmitted Command	
In AA Service Mode	**In VM Service Mode**
Switching to VM command + Recording message command (#6 + H [H = Mailbox No.])	Recording message command (H [H = Mailbox No.])

[Example]



d. Listening to a Recorded Message

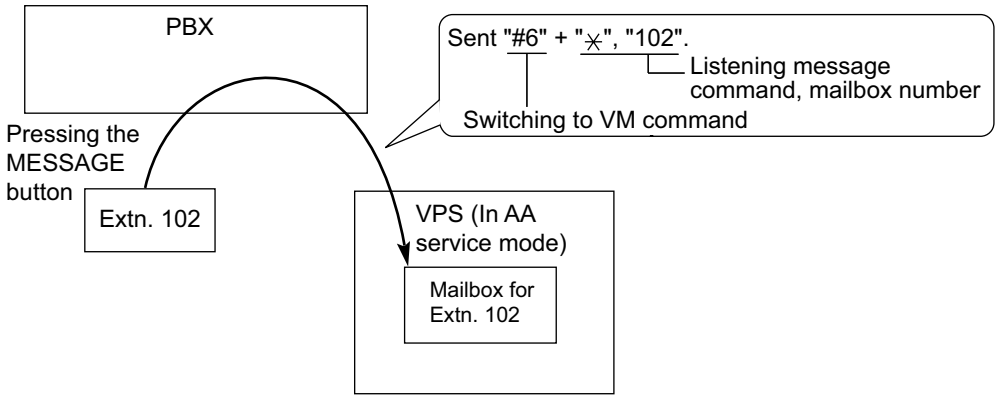
If the VPS receives a message, the VPS will set the Message Waiting feature on the corresponding telephone to notify the extension user that there is a message waiting in his mailbox. The Message button light of the extension will turn on (→ 2.20.1 Message Waiting), thereby notifying the extension user that there is a message waiting in his mailbox. When the Message button light turns on, pressing the button allows the extension user to play back the messages stored in his mailbox without dialling such as a mailbox number. It is programmable whether the PBX or the VPS cancels the Message Waiting feature (e.g., turning off the Message button light).
→ 11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings—◆ Others—Message Waiting Lamp Control

When the PBX is selected, the Message Waiting feature is cancelled after the extension user has accessed the VPS. When the VPS is selected, the Message Waiting feature is cancelled after the extension user has listened to messages stored in his mailbox.

[Performance of Pressing the MESSAGE Button]

Transmitted Command	
In AA Service Mode	**In VM Service Mode**
Switching to VM command + Listening message command (#6 + ⁎H [H = Mailbox No.])	Listening message command (⁎H [H = Mailbox No.])

[Example]



4. AA Service

If the VPS transfers the call using the AA service, the PBX will inform the VPS of the status of the called destination with the preprogrammed DTMF status signal so that the VPS can confirm the status of the extension without listening to the system tones (e.g., ringback tone).
Assign the DTMF status signal to suit the VPS settings.

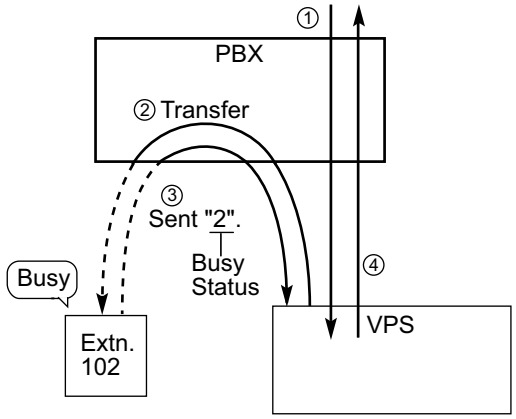
[DTMF Status Signals and Conditions]

Status	Condition	Default Command
RBT (ringback tone)	The PBX is ringing the corresponding extension.	1
BT (busy tone)	The called extension is busy.	2
ROT (reorder tone)	The dialled number is invalid.	3
DND (DND tone)	The called extension has set DND. (→ 2.3.3 Do Not Disturb (DND))	4
Answer	The called extension has answered the call.	5
FWD VM RBT (FWD to Voice Mail ringback tone)	The called extension has set FWD to VPS and the PBX is calling another port of the VPS.	6
FWD VM BT (FWD to Voice Mail busy tone)	The called extension has set FWD to VPS and all ports of the VPS are busy.	7
FWD EXT RBT (FWD to extension ringback tone)	The PBX is calling an extension other than the one dialled. FWD or Idle Extension Hunting (→ 2.2.1 Idle Extension Hunting) may be assigned by the called extension.	8
Confirm (confirmation tone)	The PBX receives confirmation that the feature has been successfully set or cancelled (e.g., Message Waiting) on the extension.	9
Disconnect	The caller has hung up.	#9

11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings
→ ◆ VM DTMF Status Signal—Ringback Tone
→ ◆ VM DTMF Status Signal—Busy Tone
→ ◆ VM DTMF Status Signal—Reorder Tone
→ ◆ VM DTMF Status Signal—DND Tone

→ ◆ VM DTMF Status Signal—Answer
→ ◆ VM DTMF Status Signal—Confirm
→ ◆ VM DTMF Status Signal—Disconnect
→ ◆ VM DTMF Status Signal—FWD to VM Ringback Tone
→ ◆ VM DTMF Status Signal—FWD to VM Busy Tone
→ ◆ VM DTMF Status Signal—FWD to Extension Ringback Tone

[Example]



① An incoming call reaches the VPS. The VPS greets the caller: "Thank you for calling Panasonic. If you know the extension number of the person you wish… ".

② The caller dials the extension number (extn. 102).
The VPS will transfer the call to the extension via the PBX.

③ If the extension is not available, the PBX sends DTMF status signal of the extension (busy status) to the VPS.

④ The VPS receives the DTMF status signal and send the appropriate message to the caller: "Sorry. The extension is busy. Would you like to leave a message… ".

Conditions

- **Voice Mail (VM) Transfer Button**
  A flexible button can be customised as the Voice Mail (VM) Transfer button with the floating extension number of the VM group as the parameter.
- It is possible to assign the time period between when the VPS answers the call and the PBX sends the Follow on ID to the VPS.
  → 11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings—◆ Timing—Waiting Time before Sending Follow on ID
- It is possible to assign the time period between when the VPS transfers the call using the AA service to the PBX and the PBX sends the DTMF status signal to the VPS.
  → 11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings—◆ Timing—DTMF Length for VM
- It is possible to select whether the mailbox number is the same as the extension number, or the mailbox number is programmable for each extension number and incoming call distribution group (Mailbox Access ID).
  → 11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Miscellaneous—◆ Programmed Mailbox No. (16 Digits)
  → 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1—◆ Programmed Mailbox No.
  → 12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 1—◆ Programmed Mailbox No.
- The Inter-digit time for the DTMF command and for DTMF status signal is programmable.
  11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings
  → ◆ Timing—Inter-digit Time
  → ◆ Timing—Waiting Time before Sending VM DTMF Status Signal
- Data Line Security is set automatically on the extensions in the VM (DTMF) group to achieve proper recording. (→ 2.11.5 Data Line Security)

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Voice Mail (Caller from VM to CO)—On-hook Wait Time (s)
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—◆ Programmed Mailbox No. (16 Digits)
11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings
11.12.2 PBX Configuration—[3-12-2] Group—VM (DTMF) Group—Group Settings
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1—◆ Programmed Mailbox No.
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Extension Number (for Voice Mail Transfer)
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 1—◆ Programmed Mailbox No.
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Extension Number (for Voice Mail Transfer)
12.3 PBX Configuration—[4-3] Extension—DSS Console
→◆ Type
→◆ Extension Number (for Two-way Record)

Feature Guide References

2.1.1.5 Intercept Routing
2.3.2 Call Forwarding (FWD)
2.12.1 Call Transfer
2.21.2 Flexible Buttons
2.28.1 Voice Mail (VM) Group

User Manual References

1.11.5 If a Voice Processing System is Connected

2.28.3 Voice Mail DPT (Digital) Integration

Description

A Panasonic VPS that supports DPT (Digital) Integration (e.g., the KX-TVM200) can be connected to this PBX in a tightly integrated fashion.
DPT (Digital) Integration features can be used when the VPS is connected through DPT ports of the PBX.

Feature Explanation

1. **Automatic Configuration—Quick Setup**
   The PBX informs the VPS of its extension numbers and the floating extension numbers of the incoming call distribution groups so that the VPS can create mailboxes with this data automatically.
2. **FWD to a Mailbox of the VPS**
   If an extension user sets FWD to the VPS, incoming calls are forwarded to the VPS. (→ 2.3.2 Call Forwarding (FWD)) The PBX sends a mailbox number of the forwarding extension to the VPS. Therefore the caller can leave a message in the mailbox of the extension without knowing the mailbox number.
3. **Intercept Routing to a Mailbox of the VPS**
   If an extension user sets Intercept Routing to the VPS, the intercepted trunk calls will be redirected to the VPS. (→ 2.1.1.5 Intercept Routing) If the "**Intercept to Mailbox for Call to Extension**" setting is enabled through system programming, the PBX sends the mailbox number of the intercepted extension to the VPS.
   If the VPS is set as the overflow destination of an Incoming Call Distribution (ICD) Group and the "**Overflow to Mailbox for Call to ICD Group**" setting is enabled through system programming, the PBX sends the mailbox number of the ICD group to the VPS.
   Therefore the caller can leave a message in the appropriate mailbox without knowing the mailbox number. If either of these settings is disabled, calls of the corresponding type are handled by the Trunk Service (e.g., Automated Attendant) in the VPS.
   → 11.11.1 PBX Configuration—[3-11-1] Group—VM (DPT) Group—System Settings—◆ Intercept to Mailbox for Call to Extension
   → 11.11.1 PBX Configuration—[3-11-1] Group—VM (DPT) Group—System Settings—◆ Overflow to Mailbox for Call to ICD Group
4. **Voice Mail (VM) Transfer Button**
   An extension user can transfer a call to a mailbox of an extension by pressing the VM Transfer button and entering the number of the extension. The transferred caller can then leave a message in the mailbox. The VM Transfer button can also be used by extension users at other times, as follows:
   a. By pressing the VM Transfer button when the extension is idle, the extension's mailbox is called and Voice Mail messages can be listened to. This feature can be used even if the Message Waiting lamp is not on.
   b. Pressing the VM Transfer button while a call is incoming will redirect the call to the called extension's mailbox. This is useful when the called extension user does not want to answer the call.
   c. If you call an extension, and the other party does not answer, pressing the VM Transfer button will transfer your call to the called extension's mailbox so that you can leave a message. This feature can also be used when the called extension is busy or set to DND.
5. **Transfer Recall to a Mailbox of the VPS**
   If a call is transferred to an extension via the Automated Attendant (AA) service of the VPS and the call is not answered within a preprogrammed Transfer Recall time, the PBX sends the mailbox number of the transfer destination extension to the VPS. Therefore the caller can leave a message in the mailbox of the extension without knowing the mailbox number. The "**Transfer Recall to Mailbox**" setting should be enabled through system programming to use this feature.
   → 10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Recall—Transfer Recall (s)
   → 11.11.1 PBX Configuration—[3-11-1] Group—VM (DPT) Group—System Settings—◆ Transfer Recall to Mailbox

→ [201] Transfer Recall Time

6. **Listening to a Recorded Message (Direct Mailbox Access)**
   If the VPS receives a message, the VPS will set the Message Waiting feature (e.g., turning on the Message button light, and showing the number of messages waiting on the display of a 6-line display PT) on the corresponding telephone as notification. (→ 2.20.1 Message Waiting) Thereby, the VPS notifies the extension user that there is a message waiting in his mailbox. When the Message button light turns on, pressing the button allows the extension user to play back the messages stored in his mailbox without dialling such as a mailbox.
   When the extension user dials an extension number of the VM (DPT) extension port or the floating extension number of the VM (DPT) group from his extension, he can listen to the messages stored in his mailbox without dialling his mailbox number (Direct Mailbox Access). It is possible to disable this feature by mailbox setting on the VPS.
7. **VPS Trunk Service & Automatic Time Mode Notification for Incoming Call**[*1]
   Multiple tenants can share a single VPS; each tenant does not require a dedicated VPS port. If the destination of the incoming trunk call is a VM (DPT) group, the PBX sends the VM trunk group number and time mode (day/lunch/break/night) of the tenant (→ 5.1.4 Time Service) assigned for the call to the VPS. Therefore the VPS can send the assigned message (company greeting) to the caller.
   Corresponding VM trunk group number and tenant number are determined by the setting of the incoming trunk call as follows:
   a. **DIL/TIE:** the setting of each trunk port (→ 2.1.1.2 Direct In Line (DIL), 4.2.1 TIE Line Service
      18.2 PBX Configuration—[10-2] CO & Incoming Call—DIL Table & Port Settings
      → ◆ Tenant Number
      → ◆ VM Trunk Group No.
   b. **DID/DDI:** the setting of each location number for DID/DDI (→ 2.1.1.3 Direct Inward Dialling (DID)/ Direct Dialling In (DDI))
      18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table
      → ◆ Tenant Number
      → ◆ VM Trunk Group No.

[Example]

For DID Calls:

Each location number can have its VM trunk group number (message number) and tenant number.

[DID Programming Example]

Location No.	DID No.	DID Destination				DID Name	VM Trunk Group No.	Tenant No.
		Day	Lunch	Break	Night			
0001	123-4567	105	100	105	100	John White	1	1
0002	123-2468	102	100	102	100	Tom Smith	2	3
:	:	:	:	:	:	:	:	:

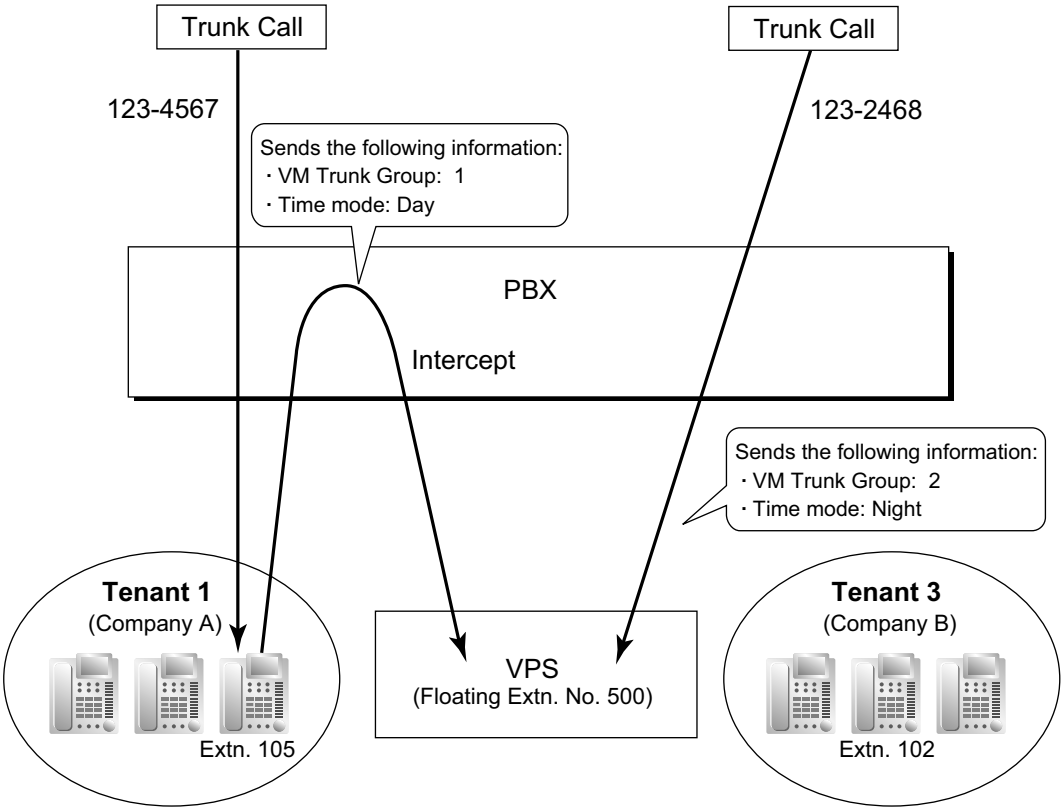
[VPS Programming—Programming Example of Trunk Group Assignment]

Trunk Group No.		Company Greeting No.	Incoming Call Service	. .
1	Day	1	Custom Service 11	. .
	Lunch	2	Custom Service 29	. .
	Break	3	Custom Service 31	. .
	Night	4	Custom Service 12	. .
2	Day	5	Custom Service 21	. .
	Lunch	6	Custom Service 15	. .
	Break	7	Custom Service 42	. .
	Night	8	Custom Service 30	. .
:	:	:	:	. .

Explanation:

A DID call reaches a VM (DPT) group directly or by the Intercept Routing feature. According to the [DID Programming Example] and [VPS Programming—Programming Example of Trunk Group Assignment], a caller will hear a corresponding company greeting of the VPS.

Time mode (day/lunch/break/night) of the preprogrammed tenant is applied to the DID destination and company greeting number.



8. **Caller's Identification Notification to the VPS**
   When receiving a trunk call, the PBX sends the caller's identification number/name to the VPS.
9. **DID Number Notification to the VPS**
   When receiving a trunk call with a DDI/DID number, the PBX sends the DDI/DID number to the VPS. The number will be sent to the VPS even if the call reaches the VPS after redirection by, for example, the Intercept Routing feature.
10. **Status Notification to the VPS**
    After the call is redirected by the VPS, the PBX sends the status of the redirected extension (e.g., busy) to the VPS.
11. **Paging by the VPS**
    The VPS can perform the Paging feature using the recorded message. (→ 2.17.1 Paging)
12. **Live Call Screening (LCS)**
    A PT or PS user can monitor his own mailbox while a caller is leaving a message and, if desired, answer the call by pressing the LCS button. When the caller is leaving a message in the mailbox, monitoring can be carried out in two ways: each PT user can choose which through personal programming (Live Call Screening Mode Set). PS users cannot choose the way: only Private mode is available for them.
    Hands-free mode: The user can monitor the call automatically through the built-in speaker.
    Private mode: The user will hear a warning tone. To monitor the call, the user goes off-hook with the handset, MONITOR button, or SP-PHONE button. However, PS users cannot monitor the call with the speakerphone.
    → 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 4—◆ LCS Answer Mode
13. **Two-way Recording into the VPS**
    A PT user can record a conversation into his own mailbox or another mailbox, while talking on the phone.

The Two-way Record button is used to record into one's own mailbox. The Two-way Transfer button is used to record into someone else's mailbox.

Note

Before recording a Two-way telephone conversation, you should inform the other party that the conversation will be recorded.

14. **VPS Data Control by the PBX**[*1]
    The date and time settings of the VPS are controlled by the PBX.
15. **Remote FWD Setting by the VM**[*1]
    Extension FWD settings can be programmed using the VPS.

[*1] This feature may not be supported depending on the software version of the VPS.

Conditions

[Live Call Screening (LCS)]

- This feature is not available for ISDN extensions and SIP extensions.
- If an SLT is connected in parallel to a PT, and if LCS is activated for the PT in Private mode, both the PT and SLT can be used to monitor calls while in idle status. The SLT will ring to indicate a message is being recorded. The call can be monitored with the SLT by going off-hook. To intercept the call, press Flash/Recall button or flash the hookswitch. (→ 2.11.10 Parallelled Telephone)
- **LCS Button**
  A flexible button can be customised as the LCS button.
- **Extension Personal Identification Number (PIN)**
  To prevent unauthorised monitoring, it is recommended the LCS user assign an extension PIN. This PIN will be required when setting LCS. (→ 2.24.1 Extension Personal Identification Number (PIN)) If the user forgets the PIN, it can be cleared by an extension assigned as the manager.
- Each extension can be programmed to either end recording or continue recording the conversation after the call is intercepted, through personal programming (LCS Mode Set [After Answering]).
  → 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 4—◆ LCS Recording Mode
  → 12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 4—◆ LCS Recording Mode
- To use the LCS feature on a PS in Wireless XDP Parallel Mode, LCS can only be turned on or off from the wired telephone. In Wireless XDP Parallel Mode, setting LCS on/off from the PS has no effect. (→ 5.2.4.5 Wireless XDP Parallel Mode)

[Two-way Recording into the VPS]

- **Two-way Record/Two-way Transfer Button**
  A flexible button can be customised as the Two-way Record or the Two-way Transfer button. An extension number can be assigned to the Two-way Transfer button so that it can be used as a one-touch record button for the mailbox of the specified extension. (**One-touch Two-way Transfer Button**).
- When all of the VPS ports are busy:
  a. Pressing the Two-way Record button sends a warning tone
  b. Pressing the Two-way Transfer button followed by an extension number sends a warning tone.

[VM Transfer Button]

- A flexible button can be customised as the VM Transfer button with the floating extension number of the VM group as the parameter.
- If two or more VPSs are connected to a PBX, the VM Transfer button will access the VPS of the VM group assigned as the parameter for the VM Transfer button.
- If a PS is paired with a PT or SLT (in Wireless XDP Parallel mode), the PS's VM Transfer button cannot be used to redirect an incoming call to the called extension's mailbox.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Recall—Hold Recall (s)
11.11.1 PBX Configuration—[3-11-1] Group—VM (DPT) Group—System Settings
11.11.2 PBX Configuration—[3-11-2] Group—VM (DPT) Group—Unit Settings
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1
→◆ Programmed Mailbox No.
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 4
→◆ LCS Recording Mode
→◆ LCS Answer Mode
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Extension Number (for Two-way Record)
→◆ Extension Number (for Two-way Transfer)
→◆ Extension Number (for Voice Mail Transfer)
→◆ Ext No. of Mailbox (for Two-way Transfer)
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 1
→◆ Programmed Mailbox No.
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 4—◆ LCS Recording Mode
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Extension Number (for Two-way Record)
→◆ Extension Number (for Two-way Transfer)
→◆ Extension Number (for Voice Mail Transfer)
→◆ Ext No. of Mailbox (for Two-way Transfer)
12.3 PBX Configuration—[4-3] Extension—DSS Console
→◆ Type
→◆ Extension Number (for Two-way Record)
→◆ Extension Number (for Two-way Transfer)
→◆ Extension Number (for Voice Mail Transfer)
→◆ Ext No. of Mailbox (for Two-way Transfer)
13.1 PBX Configuration—[5-1] Optional Device—Doorphone—◆ VM Trunk Group No.
18.2 PBX Configuration—[10-2] CO & Incoming Call—DIL Table & Port Settings—DIL
→◆ Tenant Number
→◆ VM Trunk Group No.
18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table
→◆ Tenant Number
→◆ VM Trunk Group No.

PT Programming Manual References

[201] Transfer Recall Time

Feature Guide References

2.21.2 Flexible Buttons
2.28.1 Voice Mail (VM) Group
5.5.7 Flexible Numbering/Fixed Numbering

User Manual References

1.11.5 If a Voice Processing System is Connected
3.1.2 Settings on the Programming Mode
4.1.2 Manager Programming

2.29 E1 Line Service Features

2.29.1 E1 Line Service

Description

The E1 line carries thirty 64 kbps-voice channels at 2.048 Mbps transmission speed as a trunk or private line. Voice is digitised by Pulse Code Modulation (PCM).

1. **Channel Type**
   The E1 card supports only DR2 (Digital System R2). Only DR2 can be assigned to the 30 channels of the E1 card.
2. **E1 Features**
   The following table shows the features available for each channel type:

[Feature Table]

Channel Type	(1)	(2)	(3)	(4)	(5)
	TIE	DID	DIL	ANI	Call Charge Information
DR2		✓	✓	✓*1	✓

✓: Available

*1 Receiving Dial Mode should be MFC-R2.

[Explanation]

Number in the Table	Feature	Description
(1)	**Direct Inward Dialling (DID)**	Refer to the DID feature. (→ Page 25)
(2)	**Direct In Line (DIL)**	Refer to the DIL feature. (→ Page 23)
(3)	**Automatic Number Identification (ANI)**	**Outgoing ANI:** Sends the caller's number to the E1 line. The sending method is the same as ISDN CLIP service. (→ Page 364) **Incoming ANI:** Receives the caller's number from the E1 line.When the ANI number is received, it can be treated the same as a Caller ID number. (→ Page 224)
(4)	**Call Charge Information**	The call charge meter pulses can be received during a conversation. (→ Page 259)

Conditions

- If an E1 line is used as a trunk, the channel type depends on the contract with the telephone company.
- If "MFC-R2" is selected as the Dial Mode, the PBX always sends a dial tone, instead of the telephone company, when making a trunk call using E1 line.

PC Programming Manual References

9.31 PBX Configuration—[1-1] Configuration—Slot—Port Property—E1 Port

2.30 Miscellaneous Features

2.30.1 Background Music (BGM)

Description

A PT user can listen to BGM through the built-in speaker while on-hook and idle. The following audio sources are available for BGM:

- External Music Source
- Internal Music Source

BGM—External:
BGM can also be broadcast in the office through the external pagers, this can be turned on and off by an extension assigned as the manager.

Conditions

[BGM]

- **Hardware requirement:** User-supplied music source (when an external music source is assigned)
- The music through the PT is interrupted when going off-hook.
- Each user can set/cancel BGM, and also select the music source.
- Through system programming, it is possible to specify the maximum number of IP-PTs that can simultaneously perform the BGM feature. Changing this setting may affect the number of simultaneous IP extension and IP trunk calls available on the mother board.

[BGM—External]

- **Hardware requirement:** A user-supplied external pager
- External pagers can be used with the following priorities:
  TAFAS → Paging → BGM
  (→ 2.17.1 Paging, 2.17.2 Trunk Answer From Any Station (TAFAS))

Installation Manual References

4.10 Connection of Peripherals

PC Programming Manual References

9.2.2 PBX Configuration—[1-1] Configuration—Slot—System Property—Site—VoIP-DSP Options—◆ IP Extension Count of BGM
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ External BGM On / Off
→◆ BGM Set / Cancel
10.11.1 PBX Configuration—[2-11-1] System—Audio Gain—Paging/MOH
→◆ Internal MOH—MOH1-2 (Music On Hold 1-2)
13.2 PBX Configuration—[5-2] Optional Device—External Pager

User Manual References

1.9.8 Turning on the Background Music (BGM)
2.1.4 Turning on the External Background Music (BGM)

2.30.2 Outgoing Message (OGM)

Description

An extension assigned as the manager (manager extension) can record outgoing messages (OGMs) for the following features:

Feature	Usage & Reference
Direct Inward System Access (DISA)	When a call arrives on a DISA line, the caller will hear a message. → 2.16.1 Direct Inward System Access (DISA)
Queuing Feature	If assigned in the Queuing Time Table of the incoming call distribution group, any caller who is waiting in a queue will hear a message. → 2.2.2.4 Queuing Feature
Timed Reminder	When answering the Timed Reminder Alarm, the user will hear a message. → 2.24.4 Timed Reminder

Conditions

- **Hardware Requirement**
  - If no DSP card is installed:
    There are 2 channels for both the OGM feature and the SVM feature. If 2 channels are in use by the OGM feature or SVM feature, an OGM cannot be recorded or played back. A total of 64 messages (total approximately 20 minutes) can be saved for OGM.
  - If an optional DSP card is installed:
    There are 64 channels reserved for the OGM feature. A total of 64 messages (total approximately 20 minutes) can be saved for OGM.

VoIP-DSP Card	Available Channels for OGM	Max. Number of OGM Message	Total recording time
Not Installed	2	64	approximately 20 minutes
Installed	64	64	approximately 20 minutes

  - Even if you recorded OGMs when no DSP card was installed, recorded OGMs can still be used after installing a DSP card in the PBX.
- **DSP Resource Usage**
  Playing back an OGM requires a certain number of DSP resources. If all DSP resources are in use, this operation cannot be performed. To ensure a minimum level of performance, DSP resources can be reserved for OGM playback. (→ 5.5.4 DSP Resource Usage)
- There is no limit to the length of an individual message, but the maximum recording time (all messages combined) is approximately 20 minutes.
- The same message can also be played simultaneously to multiple callers.
- **Recording Methods**
  a. Record voice messages through the extension telephone
  b. Transfer prerecorded voice messages from external sound source into the PBX via an external music port.
- After recording messages, a manager extension can also play them back for confirmation.

- Progress tone is sent to a manager extension before recording messages during a preprogrammed time period, or during clearing the prerecorded message stored at the floating extension number of desired message. The longer one is applied.
- When the manager tries to record a message, he will hear ringback tone if a message channel is in use. When all message ports become idle, he will hear the progress tone for a preprogrammed time period. After that, the PBX will automatically proceed into the recording mode.
- **Copying messages to and from a PC**
  Via Web Maintenance Console, messages can be copied to and from a PC.
  – Messages are downloaded to a PC in WAV (G.711a/μ) format.
  – Messages uploaded from a PC must be in one of the following formats: WAV (linear PCM) or WAV (G.711a/μ).

Installation Manual References

4.3.3 DSP S Card (KX-NS5110)
5.4.1 Easy Setup Wizard

PC Programming Manual References

7.2.5 Utility—File—Message File Transfer PC to PBX
7.2.6 Utility—File—Message File Transfer PBX to PC
10.3 PBX Configuration—[2-3] System—Timers & Counters—DISA / Door / Reminder / U. Conf—◆ DISA—Progress Tone Continuation Time before Recording Message (s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ OGM Record / Clear / Playback
11.5.2 PBX Configuration—[3-5-2] Group—Incoming Call Distribution Group—Queuing Time Table—◆ Queuing Sequence—Sequence 01–16
13.3.2 PBX Configuration—[5-3-2] Optional Device—Voice Message—DISA Message—◆ Floating Extension Number

PT Programming Manual References

[631] Sequences in Queuing Time Table
[730] Outgoing Message (OGM) Floating Extension Number

Feature Guide References

2.16.3 Built-in Simplified Voice Message (SVM)
5.5.8 Floating Extension
6.1 Capacity of System Resources

User Manual References

2.1.5 Recording Outgoing Messages (OGM)

Section 3

Unified Messaging System

3.1 Unified Messaging System Administration

3.1.1 Unified Messaging System Overview

Description

The KX-NS300 PBX has a built-in messaging system that provides voice mail to its subscribers. The Unified Messaging system can also provide voice guidance to outside callers, either directing them to their desired destination or to the mailbox of a subscriber, where they can leave a voice message.

Users

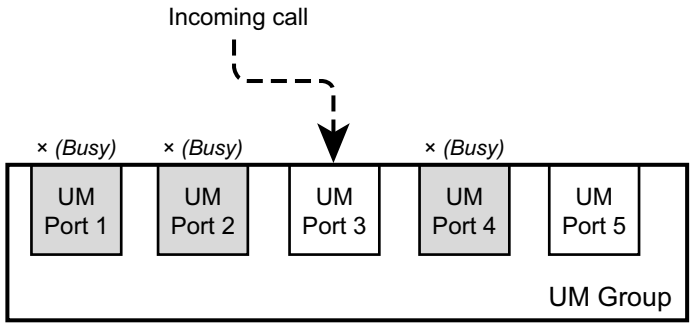
The following three types of users exist in the Unified Messaging system:

- Subscriber (maximum: 500)
  A subscriber is an extension user who has a mailbox assigned to his extension. Subscribers can play back messages saved in their mailboxes, leave messages for other subscribers, record a message that is then sent to multiple parties (including outside parties), record greeting messages, and more.
- Message Manager (maximum: 1)
  The message manager is in charge of the general delivery mailbox, as well as performing some setup, such as recording prompts and changing notification settings. Through system programming, subscribers can also be assigned Message Manager privileges.
- System Manager (maximum: 1)
  The system manager is in charge of the Unified Messaging system as a whole, and can access many settings. The system manager's duties include setting up mailboxes, assigning COS settings, and changing the service mode.

UM Ports and the UM Group

The PBX initially[*1] provides two ports (called UM ports) for use by the Unified Messaging system. A port acts as a pathway into the Unified Messaging system, so when a call is directed to the Unified Messaging system, it requires one available UM port. The number of ports for one PBX can be expanded to a maximum of 24. The UM ports of a PBX belong to the PBX's UM group. This group has a floating extension number, which can be the destination for incoming calls, redirected calls, transferred calls, etc. When incoming calls are received at the floating extension number of the UM group, calls will hunt starting at the lowest UM port number. Once an available port is found, the service assigned to the port, trunk, etc., determines how the call is handled (→ 3.2.1.39 Service Group). Services include features such as Voice Mail service (→ 3.2.1.45 Voice Mail Service), which allows a caller to leave a voice message at a subscriber's mailbox, and Automated Attendant (→ 3.2.1.3 Automated Attendant (AA)), which directs the caller to a subscriber's extension.

[*1] When installing the optional equipment such as VoIP DSP Card, even if the number of ports are not expanded, only 2 channels are available.

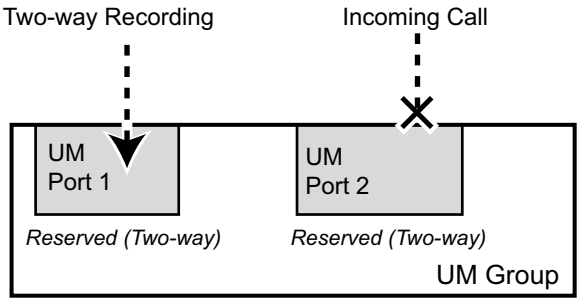


Conditions

- As the Unified Messaging system is part of the same system as the PBX, the Unified Messaging system's data coordinates with PBX settings. For details, refer to "5.9 Configuration of Users" in the Installation Manual.
- To use the Unified Messaging system, the following optional equipment is necessary:
  - Activation key: KX-NSF990
  - VoIP DSP Card: KX-NS5110 (S)
  - SD Memory Card: Depending on the message recording time.
    KX-NS3134 (XS), KX-NS3135 (S), KX-NS3136 (M)
- To expand the number of ports or the UM features, activation keys are required for each feature and for mail boxes (in accordance with the number of mailboxes). For details, refer to "3.1.1 Type and Maximum Number of Activation Keys" in the Installation Manual.
- Each port is assigned an extension number.
  → 9.6 PBX Configuration—[1-1] Configuration—Slot—UM Port Property—◆ Extension Number
- When hunting for an available UM port, the PBX ignores any FWD or DND settings (→ 2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features) applied to the ports.
  It is possible to call a port in a UM group directly. If a call is routed directly to a port in the group, it is possible to apply certain features (e.g., FWD) to that port.
- It is programmable whether the calls queue when all ports in the group are busy through system programming. If the queuing is disabled through system programming, the call will be redirected to the intercept destination of the head member extension of the UM group.
- **DSP Resource Usage**
  Connecting to the Unified Messaging system (including using features such as Two-way Recording) requires a certain number of DSP resources. If all DSP resources are in use, this operation cannot be performed. To ensure a minimum level of performance, DSP resources can be reserved for Unified Messaging operations. (→ 5.5.4 DSP Resource Usage)

  Notice

  Reserving resources for Two-way Recording (→ 3.2.1.4 Automatic Two-way Recording for Manager, → 3.2.2.30 Two-way Record/Two-way Transfer) reserves the necessary number of UM ports exclusively for Two-way Recording. For example, if 2 UM ports (the default) are available and you reserve resources for 2 Two-way Recording sessions, both UM ports will be reserved for Two-way Recording, and the Unified Messaging system will not be available for other uses.



  To provide access to the Unified Messaging system in this case, either the number of UM ports must be increased or the number of resources reserved for Two-way Recording must be lowered.
- Installing KX-NSU102 or KX-NSU104 (Unified Messaging Activation Key) increases the number of available UM ports at a PBX (maximum: 24).

Installation Manual References

4.3.2 SD Memory XS Card (KX-NS3134), SD Memory S Card (KX-NS3135), SD Memory M Card (KX-NS3136)

5.9 Configuration of Users

PC Programming Manual References

9.6 PBX Configuration—[1-1] Configuration—Slot—UM Port Property
10.5 PBX Configuration—[2-5] System—Holiday Table
10.9 PBX Configuration—[2-9] System—System Options—Option 9
11.7 PBX Configuration—[3-7] Group—UM Group
23.4 UM Configuration—[4-4] Service Settings—Holiday Table

User Manual References

1.8 Using the Unified Messaging Features

3.1.2 System Administration

Description

System administration (programming, diagnosis, system prompt administration, etc.) can be performed by the System Administrator using Web Maintenance Console. For more information, refer to the PC Programming Manual.

3.1.2.1 Automatic Configuration of Mailboxes

Description

Automatically associates extension numbers, mailboxes, and user profiles in bulk. If a mailbox with the same number as an extension number does not exist, one in created automatically. There are 2 modes available for the automatic creation of mailboxes.

PC Programming Manual References

20.1.3 UM Configuration—[1-3] Mailbox Settings—Auto Configuration

Installation Manual References

5.11 Automatic Configuration of Mailboxes

3.1.2.2 Custom Service Builder

Description

Allows the System Administrator to create Custom Services visually, using Web Maintenance Console. Each Custom Service and its functions can be edited and arranged using this feature.

PC Programming Manual References

23.3 UM Configuration—[4-3] Service Settings—Custom Service

Feature Guide References

3.2.1.15 Custom Service

3.1.2.3 Default Mailbox Template

Description

Is used as a template when the System Administrator creates consecutive mailboxes. It enables the System Administrator to apply basic settings (Mailbox Parameters, Message Waiting Notification, External Message Delivery, Auto Forwarding) to multiple mailboxes simultaneously.

PC Programming Manual References

20.1 UM Configuration—[1] Mailbox Settings

3.1.2.4 Password Administration

Description

Allows the System Administrator or System Manager to clear a subscriber password (so that a new one can be assigned).

User Manual References

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Up Mailboxes

3.1.2.5 System Backup/Restore

Description

Allows the System Administrator to back up or restore the following data as individual files: System Prompts, Mailbox Prompts, and Mailbox Messages. Data can be backed up in 2 ways:

- **Manual Backup:** The specified data is backed up manually.
  Backup data can be saved to the following destinations:
  – An external USB memory device
  – The local PC where you are running Web Maintenance Console
- **Scheduled Backup:** The specified data is backed up automatically at the specified times. Backup data can be saved to an external USB memory device. When scheduled backup is enabled, the following parameters can be set:
  – Set whether to back up messages received only within a specified period.
  – Set whether to back up messages only when the remaining capacity of the mailbox reaches a certain level.
  – Set whether to delete messages from the mailbox after backup.
  – Set whether all messages or just old messages are backed up.

A record of backup information (Description, Completion Status, Date & Time, and Total Elapsed Time) can be viewed in the backup history.

Conditions

- KX-NSU003 (Activation Key for Message Backup) is required to use the scheduled backup feature.
- If data is being backed up to the local PC, individual messages that are larger than 100 MB (about 3.5 hours long) cannot be backed up. However, even if some messages cannot be backed up, all other data will be backed up normally.
- If data is saved to a USB memory device, be sure to unmount the device before removing it from the PBX to avoid data loss or corruption.
  → 4.1.5 Status—Equipment Status—USB
- **KX-TVM Data Restore**
  This feature can be used to import KX-TVM VPS voice data to the KX-NS300's Unified Messaging system voice data.

Installation Manual References

5.3 Starting Web Maintenance Console—Converting KX-TDA100/KX-TDA200 or KX-TE series System Data for Use with the KX-NS300

PC Programming Manual References

6.8 Tool—UM Data Backup
6.9 Tool—UM Data Restore

3.1.2.6 System Reports

Description

There are several System Reports available to the System Administrator to monitor operating status. The System Administrator can print or export the System Reports. Certain reports can be printed in tabular form or graph form.
The following reports can be generated:

- Mailbox Information Report
- Call Account Report (Group calls by UM Extn)*1*2
- Call Account Report (Group calls by Mailbox)*1*2
- UM Extn Usage Report*1*2
- Memory Usage Report*1*2
- Mailbox Usage Report*1*2
- Fax Transfer Report*2
- Call Handling Statistics Report*2
- Custom Service Report*2
- Message Status Report
- Subscriber Setup Report
- Security Information Report
- Hourly Statistics Report*2

*1 This report can be printed in tabular form or graph form.
*2 This report can be set to be deleted automatically.

PC Programming Manual References

7.5 Utility—Report

3.1.2.7 System Security

Description

Protects the Unified Messaging system from unauthorised programming and/or use. By default, the System Manager and Message Manager cannot access the system from their telephones until the System Administrator enables the relevant settings and sets a password. The System Administrator can also choose to set a default password for subscribers' mailboxes. If this setting is enabled, a default password is automatically assigned when mailboxes are created.

PC Programming Manual References

26.1 UM Configuration—[7] System Security

3.2 System and Subscriber Features

3.2.1 System Features

Description

System management can be performed by the system manager and the message manager using an extension telephone, or by the System Administrator using Web Maintenance Console.
An example of items which are programmed by the system manager include:

- Creating, editing, deleting, and resetting mailboxes
- Class of Service (COS) settings
- Changing the company greeting

An example of items which are programmed by the message manager include:

- General Delivery Mailbox maintenance
- Message notification settings
- Recording/deleting messages (system prompts, company greetings, custom service menus, system caller names, etc.)

3.2.1.1 Alternate Extension Group

Description

A group of extensions that require a different call transfer sequence than other extensions and are therefore placed into a separate group.

PC Programming Manual References

23.1 UM Configuration—[4-1] Service Settings—Caller ID / PIN Call Routing—23.2 UM Configuration—[4-2] Service Settings—Parameters—◆ Alternate Extension

3.2.1.2 Auto Forwarding

Description

Moves or copies unplayed messages from one mailbox to another after a specified period of time has passed. A message can be forwarded up to 9 times, and forwarding stops at the 9th designated mailbox. Note that you cannot auto forward messages to the Mailbox Group, and messages marked as "private" cannot be forwarded. Also, a message is never forwarded to the original sender of the message.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—External MSG Delivery/Auto FWD/Personal Custom Serv

3.2.1.3 Automated Attendant (AA)

Description

Allows you to direct incoming calls without the use of an operator. Callers can be redirected to the desired extension in one of the following ways:

a. Dial extension numbers directly
b. Spell the name of the desired party using the dial keys on their telephones (→ 3.2.1.16 Dialling by Name)
c. Listen to all subscriber names and select the desired extension (→ 3.2.1.25 List All Names)

Automated Attendant (AA) answers incoming calls and redirects them to the desired extension based on numbers dialled by callers.
When calls from extensions are transferred to other subscribers, "Transferring you to (name)." can be heard by callers before the calls are transferred. This feature is not available when the name of called party is not recorded.
The service can be programmed for day, night, lunch, and break time modes, and is available for both Port and Trunk Services.
Callers will reach Automated Attendant service when:

a. The Incoming Call Service of a trunk or port is set to "Automated Attendant Service" (→ 22.1 UM Configuration—[3-1] UM Extension / Trunk Service—Service Group—◆ Day, Night, Lunch, and Break Mode - Incoming Call Service).
b. The call service of a Holiday is set to "Automated Attendant Service"
→ 23.4 UM Configuration—[4-4] Service Settings—Holiday Table—◆ Service
c. They press [#8] (Automated Attendant Service Access Command) during a call.
d. A Custom Service or Personal Custom Service option is set to transfer callers to Automated Attendant service.
→ 20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—External MSG Delivery/Auto FWD/ Personal Custom Serv
→ 23.3 UM Configuration—[4-3] Service Settings—Custom Service
e. A subscriber transfers the caller to Automated Attendant service.

PC Programming Manual References

23.2 UM Configuration—[4-2] Service Settings—Parameters

3.2.1.4 Automatic Two-way Recording for Manager

Description

Calls to and from specified extensions can be automatically recorded to a mailbox. Extensions are assigned to a supervisor, who can listen to the recorded messages through Web Maintenance Console. For each targeted extension, the following types of calls can be recorded:

- Intercom (internal) calls
- Trunk (external) calls
- Incoming ICD group calls only (i.e., when recording trunk calls, limit recording to incoming ICD group calls only)

Recorded conversations are treated as new messages in the dedicated mailbox.
The period within which calls are recorded can be set through system programming.

Note

You should inform the other party that the conversation will be recorded.

Conditions

- KX-NSU002 (Activation Key for Two-way Recording Control) is required to use this feature.
- Mailboxes specified as the recording destination become dedicated Automatic Two-way Recording mailboxes. It is not possible to record messages to these mailboxes through other means, and subscribers cannot log in to them.

- Forwarded calls and calls retrieved from being on hold will also be automatically recorded. However, conference calls will not be recorded.
- A call between 2 extensions connected via QSIG (TIE) is treated as an external call (regardless of whether KX-NSN002 [Activation Key for QSIG Network] is installed). To record these types of calls, trunk (external) calls must be selected as one of the types of calls to record.
- Recording will automatically stop when the mailbox reaches its capacity. Delete older messages in order to use this feature again.
- If the enabled extension has a Two-way Record button assigned to it, the button will flash during recording. However, that button cannot be used to cancel this feature. The extension cannot cancel this feature through Communication Assistant (CA) either.
- An extension whose conversation is being recorded cannot be the target of the following features:
  - Executive Busy Override (→ 2.10.2 Executive Busy Override)
  - Call Monitor (→ 2.10.3 Call Monitor)
  - Whisper OHCA (→ 2.10.4.4 Whisper OHCA)
- Automatic recording will not be performed when both the target extension and the other party are ISDN extensions.
- Conversations recorded with this feature are backed up via System Backup/Restore (→ 3.1.2.5 System Backup/Restore).
- In this feature, these functions are NOT supported:
  - Sending an e-mail which notifies the recording to the mailbox.
  - Sending an e-mail which attaches the recorded data to the mailbox.

PC Programming Manual References

7.10 Utility—Automatic Two-way Recording
8.3 Users—Automatic Two-way Recording
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—UM—◆ Two-way Recording
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—UM—◆ Two-way Recording

User Manual References

Manager Operation

3.2.1 User Programming—◆◆ Automatic Two-way Recording for Manager

3.2.1.5 Broadcasting Messages

Description

Allows the System Manager to deliver the same message to the mailboxes of all subscribers simultaneously. Broadcast Messages have priority over other regular or urgent messages during playback, but otherwise are treated like regular messages. They are not treated as "Urgent" messages; if "Only Urgent Messages" is selected as the Notification Type in the Notification Schedule, the Message Waiting Notification is not activated when a Broadcast Message is received.

Conditions

- This feature is only available for the System Manager.

User Manual References

Manager Operation

2.2.1 System Manager Features—◆◆ Broadcasting Messages

3.2.1.6 Call Services

Description

Include a series of both incoming and outgoing call services.
Incoming Call Services: Automated Attendant service, Voice Mail service, Interview service, Custom Service.
Outgoing Call Services: Message Waiting Notification and External Message Delivery.

PC Programming Manual References

22.1 UM Configuration—[3-1] UM Extension / Trunk Service—Service Group
→◆ Day, Night, Lunch, and Break Mode - Incoming Call Service
→◆ Day, Night, Lunch, and Break Mode - Incoming Call Service Parameter
→◆ Day, Night, Lunch, and Break Mode - Incoming Call Service Prompt

3.2.1.7 Call Transfer to Outside

Description

Enables the Unified Messaging system to transfer a call to a trunk from the following services:

- Custom Service
- Call Transfer Service
- Personal Custom Service
- Caller ID Callback
- Call-through Service

Conditions

- This feature cannot be used when COS programming does not allow incoming trunk calls to be transferred to a trunk.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Call Transfer—◆ Call Transfer to Outside
24.4 UM Configuration—[5-4] System Parameters—Parameters—Transfer to Outside

User Manual References

Subscriber Operation

1.8.5 Transferring Calls—◆◆ Assigning and Cancelling Telephone Numbers for Call Transfer to Outside Line

3.2.1.8 Caller ID Call Routing

Description

Allows the System Administrator to store a maximum of 200 telephone numbers and assign a specific destination (extension, mailbox, Mailbox Group, or Custom Service) to each telephone number for each time mode (day, night, lunch, and break). When Caller ID information is received that matches one of the stored telephone numbers, the call is automatically directed to its destination. Calls that are "Private" (when the caller's number is not received), "Out of Area" (when the caller is calling from an area that does not support Caller ID), and "Long Distance" (when the caller made a long distance call) can also be directed to a specific extension, mailbox or Custom Service.

Conditions

- The Company Greetings will not be played for callers when calls are routed by this feature.

PC Programming Manual References

23.1 UM Configuration—[4-1] Service Settings—Caller ID / PIN Call Routing

3.2.1.9 Caller ID Screening

Description

Allows the Unified Messaging system to announce the name of the caller when transferring a call to an extension from a preprogrammed caller (e.g., "You have a call from [name of caller]."). Caller names must be recorded beforehand. Caller names can be recorded by each subscriber (Caller Name Announcement—Personal) and for the entire system (Caller Name Announcement—System).

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Call Transfer—◆ Caller ID Screen

Feature Guide References

3.2.1.10 Caller Name Announcement

User Manual References

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Class of Service (COS) Parameters

3.2.1.10 Caller Name Announcement

Description

Allows you to store telephone numbers and record a caller name for each telephone number. The caller name is announced when playing a message in their mailbox from one of the preprogrammed callers, when the system directs a call to the subscriber from one of the preprogrammed callers (Caller ID Screening), and when the system pages the subscriber by intercom (Intercom Paging).

There are 2 types of Caller Name Announcement.

- **Caller Name Announcement—Personal**
  Allows subscribers to store a maximum of 30 telephone numbers using a telephone.
- **Caller Name Announcement—System**
  Allows the System Administrator to store a maximum of 200 telephone numbers using a PC.

Conditions

- If the same telephone number is programmed for both the system and personal caller name announcement, the personal caller name will be announced.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Number of CIDs for Caller Name Announcement (Selection)
24.3 UM Configuration—[5-3] System Parameters—System Caller Name Announcement

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Personal Caller Name Announcement

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Class of Service (COS) Parameters
2.2.2 Message Manager Features—◆◆ Recording Messages—To record system caller names

3.2.1.11 Class of Service (COS)

Description

Each mailbox is assigned a Class of Service (COS) that determines the set of services that are available to its subscriber.
Mailboxes can be assigned to their own or to the same COS as needed. COS No. 65 and 66 are assigned by default to the Message Manager and to the System Manager, respectively. No other mailboxes can be assigned to COS No. 65 and 66.

Conditions

- The System Administrator (using a PC) and the System Manager (using a telephone) can change COS assignments.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—◆ Class of Service (Mailbox)
Section 21 UM Configuration—[2] Class of Service

3.2.1.12 Company Greeting

Description

Is a prerecorded message designed to greet all incoming callers and provide relevant information. A maximum of 32 Company Greetings can be recorded, and a Company Greeting can be assigned for each time mode (day, night, lunch, and break) and holiday for each Service Group. The start time of morning greeting, afternoon greeting, and evening greeting can be set.

Conditions

- The System Manager can change the Company Greeting setting remotely by simply calling the Unified Messaging system.
- The System Administrator can assign specific greetings for holidays.

PC Programming Manual References

22.1 UM Configuration—[3-1] UM Extension / Trunk Service—Service Group—◆ Day, Night, Lunch, and Break Mode - Company Greeting No. (Selection)
23.4 UM Configuration—[4-4] Service Settings—Holiday Table—◆ Company Greeting No.
24.4 UM Configuration—[5-4] System Parameters—Parameters—Daily Hours Setting

User Manual References

Subscriber Operation

1.8.4 Sending Messages—◆◆ Receiving External Delivery Messages

Manager Operation

2.2.1 System Manager Features—◆◆ Changing the Company Greeting and Incoming Call Service Setting
2.2.2 Message Manager Features—◆◆ Recording Messages

3.2.1.13 Company Name

Description

Is used by External Message Delivery Service when the intended receiver enters the password incorrectly 3 times. The Unified Messaging system announces the Company Name so that the receiver realises what company placed the call to him or her.

User Manual References

Manager Operation

2.2.2 Message Manager Features—◆◆ Recording Messages

3.2.1.14 Covering Extension

Description

Forwards calls to a second extension when the first extension's subscriber is not available to take the call. The caller can also access the Covering Extension by pressing [0] while a Personal Greeting is being played, or while leaving a message.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—◆ Covering Extension

User Manual References

Subscriber Operation

1.8.5 Transferring Calls—◆◆ Assigning Your Covering Extension

3.2.1.15 Custom Service

Description

The following Custom Service types are available:

- Menu & Transfer
  Allow callers to perform specific functions by pressing dial buttons on their telephones while listening to voice guidance (Custom Service Messages). Callers can be guided to an extension, mailbox, outside destinations (including mobile phones), operator, fax machine, other Custom Services etc., without the assistance of an operator.
- Date Control
  Allow a different operation to be assigned for up to 5 time periods defined by date. The caller makes no selection and no menu is announced.
- Time Control
  Allow a different operation to be assigned for up to 5 time periods defined by time of day. The caller makes no selection and no menu is announced.
- Day Control
  Allow a different operation to be assigned for up to 5 time periods defined by day of the week. The caller makes no selection and no menu is announced.
- Password
  Require that callers enter a password. Each password is assigned an operation. If a password is entered correctly, the caller is handled by the password's preprogrammed operation.

Custom Service Messages ("Press 1 for Sales, press 2 for Service…", etc.) can be recorded by the System Administrator or the Message Manager, and can be recorded in multiple languages if needed. In total, a maximum of 200 Custom Services can be created.

Calls can be handled by Custom Service by:

- Setting the Incoming Call Service of a trunk or port to "Custom Service Menu" and specifying a Custom Service number.
  → 22.1 UM Configuration—[3-1] UM Extension / Trunk Service—Service Group—◆ Day, Night, Lunch, and Break Mode - Incoming Call Service
- Setting the call service of a Holiday to "Custom Service Menu" and specifying a Custom Service number.
  → 23.4 UM Configuration—[4-4] Service Settings—Holiday Table—◆ Service
- Using Custom Service or Personal Custom Service. A Custom Service can be assigned to one of the available options provided by another Custom Service or Personal Custom Service. After pressing the appropriate dial key, the caller is sent to assign Custom Service.

Custom Services can be created and edited using the Custom Service Builder utility of Web Maintenance Console.

PC Programming Manual References

7.5.3 Utility—Report—UM View Reports
23.3 UM Configuration—[4-3] Service Settings—Custom Service

User Manual References

Manager Operation

2.2.2 Message Manager Features—◆◆ Recording Messages

3.2.1.16 Dialling by Name

Description

Allows the caller to be connected to the desired subscriber's mailbox or extension by searching for the subscriber by name. Using the dialling keys, the caller can enter the first few letters of the subscriber's first and/or last name. The Unified Messaging system searches for possible matches and offers to connect the caller with the subscriber. Subscriber names are included or excluded from the Directory Listing according to their Class of Service (COS).

Conditions

- The System Administrator can change the Name Entry Mode (first, last, or full name entry) in Service Settings.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—◆ First Name
20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—◆ Last Name
21.1 UM Configuration—[2] Class of Service—◆ Directory Listing

3.2.1.17 Emergency Greeting

Description

An emergency greeting can be recorded as one of company greetings. The emergency greeting can be used in times such as when the company must be closed due to bad weather.

User Manual References

Manager Operation

2.2.2 Message Manager Features—◆◆ Recording an Emergency Greeting

3.2.1.18 Extension Group

Description

An Extension Group is a group of extensions that share a common mailbox. Extension Groups are created by the System Administrator. Each group has an Extension Group number. There are 20 group lists available, and each group list can have a maximum of 100 members.

PC Programming Manual References

24.2 UM Configuration—[5-2] System Parameters—Extension Group

3.2.1.19 Hold

Description

Provides the caller with the option of temporarily going on hold when the called extension is busy. The Unified Messaging system automatically recalls the extension after a specified period of time. When several callers are holding for the same extension, callers are connected to the extension in the order in which they originally called.

PC Programming Manual References

23.1 UM Configuration—[4-1] Service Settings—Caller ID / PIN Call Routing—23.2 UM Configuration—[4-2] Service Settings—Parameters—◆ Call Hold Mode

3.2.1.20 Holiday Service

Description

Allows the system to override the normal settings assigned for the appropriate trunk service or port service by playing a special holiday greeting. (e.g., "Happy New Year! Our office is closed today. If you wish to record a message, please press 1 now".) A maximum of 24 holidays can be programmed. A holiday can be a specific day or range of days.

Conditions

- Holidays cannot overlap with each other.
- The holidays stored in the Unified Messaging holiday table are managed separately from those in the holiday table used for Time Service (→ 5.1.4 Time Service). However, holidays specified in the Time Service holiday table can be copied to the Unified Messaging holiday table.

PC Programming Manual References

23.4 UM Configuration—[4-4] Service Settings—Holiday Table

3.2.1.21 Hospitality Mode

Description

Allows a subscriber to access certain subscriber services in a dedicated, 'hospitality' mode. Subscribers in hospitality mode can listen to messages, change the password, change personal greetings, and change the owner's name, if enabled to do so through system programming. Also, it is possible to set whether the password, personal greeting and owner name settings are deleted upon check-out.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Hospitality Mode

Feature Guide References

2.23 Hospitality Features

3.2.1.22 Intercept Routing to a Mailbox

Description

If the Intercept Routing destination is set to the floating extension number of the UM group, and the "**Intercept to Mailbox for Call to Extension**" setting is enabled through system programming, intercepted trunk calls will be redirected to the extension's mailbox. Therefore the caller can leave a message in the mailbox of the extension without knowing the mailbox number.

PC Programming Manual References

11.7.1 PBX Configuration—[3-7-1] Group—UM Group—System Settings—◆ Intercept to Mailbox for Call to Extension

Feature Guide References

2.1.1.5 Intercept Routing

3.2.1.23 Intercom Paging

Description

Allows callers to page subscribers when the subscribers have set Intercom Paging for Call Transfer or when the subscribers have set Intercom Paging for Incomplete Call Handling. The caller is briefly placed on hold while the system announces the page and until the subscriber answers the page. The subscriber can answer the page from any extension using the paging answer feature number. If Caller ID information is received and the caller's name has been recorded for the Caller Name Announcement feature, the name will be announced at the end of the page.
To utilise this feature, the following settings are necessary:

1. Assign an Intercom Paging Group to the desired Class of Service.
   → 21.1 UM Configuration—[2] Class of Service—Call Transfer—◆ Intercom Paging Group
2. Set "No Answer Time for Intercom Paging" to the desired setting.
   → 24.4 UM Configuration—[5-4] System Parameters—Parameters—◆ No Answer Time for Intercom Paging (1-30 s)

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Call Transfer—◆ Intercom Paging Group
24.4 UM Configuration—[5-4] System Parameters—Parameters—Intercom Paging Parameters

Feature Guide References

2.17.1 Paging

3.2.1.24 Interview Service

Description

Allows the Unified Messaging system to "interview" a caller by playing a series of prerecorded questions and recording the caller's responses.
As the callers answer questions, they are recorded into an interview mailbox.

- After a caller records the answers, the interview mailbox lights a message waiting lamp on the interview mailbox owner's extension.
- When retrieving the messages, the subscriber only hears the answers to the questions.

In order to use Interview service, at least one interview mailbox must be created in the system.
→ 20.1 UM Configuration—[1] Mailbox Settings
Callers will reach an interview mailbox when:

a. The Incoming Call Service of a trunk or port is set to "Interview Mailbox".
→ 22.1 UM Configuration—[3-1] UM Extension / Trunk Service—Service Group—◆ Day, Night, Lunch, and Break Mode - Incoming Call Service

b. The call service of a Holiday is set to "Interview Mailbox".
→ 23.4 UM Configuration—[4-4] Service Settings—Holiday Table—◆ Service

c. A Custom Service or Personal Custom Service option is set to transfer callers to an interview mailbox.
→ 23.3 UM Configuration—[4-3] Service Settings—Custom Service
→ 20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—External MSG Delivery/Auto FWD/ Personal Custom Serv—◆ Personal Custom Service

d. A subscriber has configured his or her mailbox so that unanswered calls are directed to the mailbox's interview mailbox via Incomplete Call Handling, and has set Message Reception Mode to "Interview Mode".
→ 1.8.6 Other Features—◆◆ Interview Mailbox—Message Reception Mode

e. A subscriber with a logical extension has set Message Reception Mode to "Interview Mode".
→ 1.8.6 Other Features—◆◆ Interview Mailbox—Message Reception Mode

f. A subscriber transfers the caller to an interview mailbox.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—◆ Interview Mailbox

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Interview Mailbox

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Up Mailboxes

3.2.1.25 List All Names

Description

Allows callers to listen to a list of all subscribers' names and extension numbers in Automated Attendant service or Custom Service.

Conditions

- This feature is only available when:
  a. The subscriber's name has been recorded.
  b. The subscriber's extension number is set.
  c. The "Directory Listing" parameter for the subscriber's Class of Service is set to "Yes".
- To return to the previous menu in Automated Attendant service or Custom Service, press ⚹.

Feature Guide References

3.2.1.3 Automated Attendant (AA)
3.2.1.15 Custom Service

3.2.1.26 Logical Extension (All Calls Transfer to Mailbox)

Description

Is an extension that always receives calls directly into its mailbox. This feature is used by subscribers that are often unavailable or that do not have a telephone.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—◆ All Calls Transfer to Mailbox

User Manual References

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Up Mailboxes

3.2.1.27 Message Reception Mode

Description

Determines whether incoming calls are directed to the subscriber's regular mailbox or interview mailbox. This mode is effective for Incomplete Call Handling Service (when the subscriber's line is busy, when the subscriber cannot take the call, or when the subscriber has enabled Call Blocking), and when the transfer destination of calls is set to a Logical Extension.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—◆ All Calls Transfer to Mailbox

Feature Guide References

3.2.1.24 Interview Service
3.2.1.26 Logical Extension (All Calls Transfer to Mailbox)

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Interview Mailbox—Message Reception Mode

3.2.1.28 Message Waiting Notification—E-mail Device

Description

Enables subscribers (including the message manager) to be notified by e-mail when they have new messages (voice). The notification will contain the message sender's information, the length of the message, the number of messages (new/old), and a callback number (if programmed). Subscribers can choose to have the voice message data attached to the notification and can also choose to have the message deleted after it has been sent. Up to 3 devices can be configured for receiving notifications.

Note

To receive notifications about missed calls, extension users should specify e-mail addresses in their user settings. (→ 5.4.1 E-mail Notification for Extension Users)

Conditions

- KX-NSU201, KX-NSU205, KX-NSU210, KX-NSU220, or KX-NSU299 (Activation Key for Unified Messaging E-mail Notification) is required to use this feature. One activation key is required for each mailbox that will be used with this feature.
- The System Administrator can enable or disable this feature for each mailbox, and can customise the time frame during which notifications are sent. For example, if the System Administrator sets the time frame for Monday to Friday between 9 AM and 5 PM, notifications will be sent only during those hours.
- When sending long voice message notifications, note the following, and confirm the settings.
  - If the voice message is longer than the time specified for "Maximum Message Length", the surplus parts of the message may be discarded when sending the notification.
    → 24.4 UM Configuration—[5-4] System Parameters—Parameters—E-mail Option—◆ Maximum Message Length (Selection)
  - Depending on the settings of the sender and recipient, voice messages may not be sent or received properly.
  - When the Unified Messaging system is programmed to delete messages after they are sent, the message will be deleted even if there is an error and the notification cannot be sent.
- If there are any errors when the system tries to send notifications, error messages will be sent to the preset address.
  → 24.4 UM Configuration—[5-4] System Parameters—Parameters—E-mail Option—◆ Mail Address (Up to 128 ASCII characters)

PC Programming Manual References

7.5.3 Utility—Report—UM View Reports
20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Notification Parameters
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—User Name
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—E-mail Address
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—Only Urgent Messages
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—Title Order
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—Title String
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—Callback Number
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—Send Wait Time [0-120 min]
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—Attach Voice File
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—Use Mode
21.1 UM Configuration—[2] Class of Service—General
→◆ E-mail Option
24.4 UM Configuration—[5-4] System Parameters—Parameters—E-mail Option

Feature Guide References

5.4.1 E-mail Notification for Extension Users

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Message Waiting Notification
→Device Notification:
→To turn Device Notification On/Off

3.2.1.29 Message Waiting Notification—Lamp

Description

Automatically lights the message waiting lamp on the subscriber's telephone when subscribers have new messages. When a proprietary telephone with display is used, the number of unplayed messages will be displayed when the message waiting lamp is lit. When the Message button light turns on, pressing the button allows the subscriber to play back the messages stored in his mailbox without dialling a mailbox number.

Conditions

- In order to display the number of unplayed messages on the display:
  - A compatible Panasonic Proprietary Telephone with 6-line display must be used.
  - The subscriber must have his or her own extension.
  - The Message Manager's extension number must be assigned as Operator 1 in Day Mode.
- Extensions assigned as operators can be called by dialling [0], however, when setting this feature the extension number (not "0") must be specified.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Notification Parameters—◆ Message Waiting Lamp

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Message Waiting Notification

Manager Operation

2.2.2 Message Manager Features—◆◆ Setting Up Message Waiting Notification

3.2.1.30 Message Waiting Notification—Telephone Device

Description

Automatically calls a preprogrammed telephone number when a subscriber has new messages (voice). New messages will be automatically played back when the subscriber answers the call.
Notifications can be scheduled. The System Administrator can program 2 different time frames (for example, 9 AM to 12 PM and 8 PM to 10 PM) independently for each day of the week. Message waiting notifications will

be sent on the selected days during the programmed time frames. Notifications can also be sent "continuously", meaning a notification will be sent whenever a new message is received, 24 hours a day.

Conditions

- Notifications can be sent for either all messages or for urgent messages only, depending on whether the "Only Urgent Messages" setting is enabled. When it is enabled, notifications are sent (according to the notification schedule, if programmed) only when urgent messages have been left in the subscriber's mailbox.
- A maximum of 3 devices can be programmed by the System Administrator or subscriber for use with this feature. When the Unified Messaging system calls a device and the call is not answered, the system will try to call the device again. The number of retries and the delay time between retries can be programmed by the System Administrator. The lowest numbered device is called first. If the first device cannot be called, the second (then third) device will be called.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Notification Parameters
→◆ Telephone Device
→◆ Telephone Device—Device No. 1, 2, 3—Only Urgent Messages
→◆ Telephone Device—Device No. 1, 2, 3—Dial Number [0-9 * # T X , ;]
→◆ Telephone Device—Device No. 1, 2, 3—No. of Retries
→◆ Telephone Device—Device No. 1, 2, 3—Busy Delay Time (min)
→◆ Telephone Device—Device No. 1, 2, 3—No Answer Delay Time (min)
→◆ Telephone Device—Device No. 1, 2, 3—Use Mode
→◆ Telephone Device—Device Notification Timer—Device Start Delay Time (0-120 min)
→◆ Telephone Device—Device Notification Timer—Device Interval Time between Device 1, 2, 3 and Next Device
24.4 UM Configuration—[5-4] System Parameters—Parameters—Dialling Parameters/MSW Notification

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Message Waiting Notification

Manager Operation

2.2.2 Message Manager Features—◆◆ Setting Up Message Waiting Notification

3.2.1.31 Multilingual Service

Description

Allows a maximum of 8 languages to be used for system prompts. The System Administrator can select 5 languages for the multilingual selection menu, and assign a key for each language in the menu. Callers can select the desired language by pressing the assigned key. The language used for system prompts heard by subscribers is determined by Class of Service (COS).

PC Programming Manual References

24.4 UM Configuration—[5-4] System Parameters—Parameters—Prompt Setting

User Manual References

Manager Operation

2.2.2 Message Manager Features—◆◆ Recording Messages

3.2.1.32 No DTMF Input Operation

Description

Provides guidance to callers when several seconds pass without anything being entered by the caller.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—External MSG Delivery/Auto FWD/Personal Custom Serv—◆ No DTMF Input Operation

3.2.1.33 On Hold Announcement Menu

Description

Allows callers (other than the first caller) in a queue to listen to prerecorded announcements or music. The On Hold Announcement Menu can be recorded by the System Administrator with a telephone while using Web Maintenance Console or by importing WAV files, and also by the Message Manager using a telephone. When the On Hold Announcement Menu is recorded, this feature is automatically enabled.

PC Programming Manual References

7.9 Utility—UM – System Prompts Customisation

User Manual References

Manager Operation

2.2.2 Message Manager Features—◆◆ Recording Messages

3.2.1.34 Operator Service

Description

Allows callers to be connected to a live operator. The operator can then transfer the caller to the appropriate party or service as needed. A maximum of 3 operators can be specified to receive calls in the day, night, lunch, and break modes, and each operator can be assigned a mailbox for message-taking.

Note

- Operator 1 in day mode is automatically designated as the Message Manager and is assigned the extension number "0" or "9". This operator's mailbox is the General Delivery Mailbox.
- Operators 2 and 3 can be assigned to a floating number that is assigned to a ring group (Incoming Call Distribution Group).

Calls can be directed to an operator when:

a. A caller does not or cannot send any DTMF signals to the Unified Messaging system (i.e., the caller does not dial any digits).

b. A caller dials "0" when the call is being handled by Automated Attendant service.

c. A Custom Service or Personal Custom Service option is set to transfer callers to Automated Attendant service. After pressing the appropriate dial key, the caller is directed to an operator.
→ 23.3 UM Configuration—[4-3] Service Settings—Custom Service
→ 20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—External MSG Delivery/Auto FWD/ Personal Custom Serv—◆ Personal Custom Service

Operator Service can be structured as a cascade so that if Operator 1 cannot take the call, it goes to Operator 2. If that fails, the call goes to Operator 3. If that fails, the caller can record a message. At each stage, there are other options for busy cases and no-answer cases.

- **Busy Coverage Mode**
  Determines how calls to an operator will be handled when the line is busy. The Busy Coverage options are: Hold, No Answer Coverage, Call Waiting, and Disconnect Message.
- **No Answer Coverage Mode**
  Determines how calls will be handled when an operator does not answer within the time specified for "Operator No Answer Time". The No Answer Coverage options are: Caller Select, Leave Message, Disconnect Message, and Next Operator.

PC Programming Manual References

23.1 UM Configuration—[4-1] Service Settings—Caller ID / PIN Call Routing—23.2 UM Configuration—[4-2] Service Settings—Parameters
→◆ Operator Service—Operator's Extension
→◆ Operator Service—Busy Coverage Mode
→◆ Operator Service—No Answer Coverage Mode

User Manual References

Manager Operation

2.2.2 Message Manager Features—◆◆ Customising the Message Manager's Mailbox

3.2.1.35 PIN Call Routing

Description

Allows the System Administrator to store a maximum of 200 PIN (Personal Identification Number) (max. 20 digits) numbers and assign a destination (extension, mailbox, mailbox group, or Custom Service) to each PIN for each time mode (day, night, lunch, and break). In a Custom Service, callers can be required to enter a PIN number in order to be directed to a destination.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Mailbox Parameters—◆ Call Transfer Anytime in Incomplete Handling Menu
23.1 UM Configuration—[4-1] Service Settings—Caller ID / PIN Call Routing

3.2.1.36 Play System Prompt After Personal Greeting

Description

Allows the Guidance for Recording message to be played for the caller after the Personal Greeting. The Guidance for Recording message instructs the caller how to terminate the call, access more features, and rerecord the message.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Play System Prompt after Personal Greeting

User Manual References

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Class of Service (COS) Parameters

3.2.1.37 Port Service

Description

Allows assignment of call services to each port. The incoming call service determines which service is used when answering incoming calls. These services include: Voice Mail, Automated Attendant, Interview, Custom Service, and Transfer to Mailbox. Custom Service is the most flexible of all the services because it allows access to the other services by pressing one key.

Note

For caller convenience, we recommend programming all ports to use Custom Service as the Incoming Call Service (→ 3.2.1.15 Custom Service).

PC Programming Manual References

Section 22 UM Configuration—[3] UM Extension / Trunk Service

Feature Guide References

3.2.1.3 Automated Attendant (AA)
3.2.1.15 Custom Service
3.2.1.24 Interview Service
3.2.1.43 Transfer to Mailbox
3.2.1.45 Voice Mail Service

3.2.1.38 Remote Time Service Set

Description

Allows the System Manager to programme his or her extension from a remote location in order to change the PBX's time service mode (day/night/lunch/break).

User Manual References

Manager Operation

2.2.1 System Manager Features—◆◆ Remote Time Service Mode Setting

3.2.1.39 Service Group

Description

Is a group of parameters that determine how incoming calls will be handled. 64 different Service Groups can be configured; one Service Group can be assigned to each port and each trunk group.

PC Programming Manual References

10.4 PBX Configuration—[2-4] System—Week Table
22.1 UM Configuration—[3-1] UM Extension / Trunk Service—Service Group

Feature Guide References

5.1.4 Time Service

User Manual References

Manager Operation

2.1.2 Time Service Mode Control

3.2.1.40 Simplified Tutorial

Description

If enabled in a subscriber's COS settings, the tutorial that guides the subscriber through setting up his or her mailbox is simpler and takes less time to complete than the standard tutorial.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—◆ Tutorial

3.2.1.41 System Prompts

Description

Are announcements that instruct a caller. A maximum of 8 languages can be stored as system prompts. The name of each language can be stored as well. When the System Administrator selects a language for the system settings, the desired language can be selected using this name label data. "Primary" can be specified by selecting one of the stored languages, and become the default setting unless another language is selected in the system setting.

Note

- System prompts can be changed or turned on/off, system prompts for each language can be imported as WAV files, or re-recorded using a telephone.

- In order to leave more time for recording, the System Administrator is also able to delete specific system prompts or one of the installed languages used for system prompts.

PC Programming Manual References

7.9 Utility—UM – System Prompts Customisation
21.1 UM Configuration—[2] Class of Service—◆ Prompt Mode
24.4 UM Configuration—[5-4] System Parameters—Parameters—Prompt Setting

User Manual References

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Class of Service (COS) Parameters
2.2.2 Message Manager Features—◆◆ Recording Messages

3.2.1.42 Transfer Recall to a Mailbox

Description

If a call is transferred to an extension via the Automated Attendant (AA) service and the call is not answered within a preprogrammed Transfer Recall time, the call is redirected to the mailbox of the transfer destination extension. Therefore the caller can leave a message in the mailbox of the extension without knowing the mailbox number. The "**Transfer Recall to Mailbox**" setting must be enabled through system programming to use this feature.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Recall—Transfer Recall (s)
11.7.1 PBX Configuration—[3-7-1] Group—UM Group—System Settings—◆ Transfer Recall to Mailbox

Feature Guide References

3.2.1.3 Automated Attendant (AA)

3.2.1.43 Transfer to Mailbox

Description

Forwards the call to a specified mailbox.
Calls reach the Transfer to Mailbox Service when the Incoming Call Service of a trunk group or port is set to "Transfer to Mailbox".

PC Programming Manual References

22.1 UM Configuration—[3-1] UM Extension / Trunk Service—Service Group—◆ Day, Night, Lunch, and Break Mode - Incoming Call Service

3.2.1.44 Trunk Service (Universal Port)

Description

Allows call handling features to behave differently depending on the trunk group that the calls are received on. A service can be assigned to each trunk group, including: Voice Mail, Automated Attendant, Interview, Custom Service, and Transfer to Mailbox. Custom Service is the most flexible of all the services because it allows access to the other services by pressing one key.

Note

For caller convenience, we recommend programming all trunk groups to use Custom Service as the Incoming Call Service (→ 3.2.1.15 Custom Service).

PC Programming Manual References

Section 22 UM Configuration—[3] UM Extension / Trunk Service

Feature Guide References

3.2.1.3 Automated Attendant (AA)
3.2.1.15 Custom Service
3.2.1.24 Interview Service
3.2.1.43 Transfer to Mailbox
3.2.1.45 Voice Mail Service

3.2.1.45 Voice Mail Service

Description

Is a message recording service that allows callers to leave messages for subscribers. They can dial mailbox numbers directly or "spell" the name of the desired party using the dial keys on their telephones (→ 3.2.1.16 Dialling by Name). Once the caller has dialled an appropriate number, the Unified Messaging system transfers the caller to the party.
Callers will reach Voice Mail service when:

a. The Incoming Call Service of a trunk group or port is set to "Voice Mail Service".
→ 22.1 UM Configuration—[3-1] UM Extension / Trunk Service—Service Group—◆ Day, Night, Lunch, and Break Mode - Incoming Call Service

b. The call service of a Holiday is set to "Voice Mail Service".
→ 23.4 UM Configuration—[4-4] Service Settings—Holiday Table—◆ Service

c. They press [#6] (Voice Mail Service Access Command) during a call.

d. A Custom Service or Personal Custom Service option is set to transfer callers to Voice Mail service.
→ 20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—External MSG Delivery/Auto FWD/ Personal Custom Serv—◆ Personal Custom Service
→ 23.3 UM Configuration—[4-3] Service Settings—Custom Service

e. A subscriber transfers the caller to Voice Mail service.

PC Programming Manual References

Section 22 UM Configuration—[3] UM Extension / Trunk Service

3.2.2 Subscriber Features

Description

PBX users who are assigned a mailbox in the Unified Messaging system are called subscribers. Subscribers can customise their mailboxes in the following ways:

- Set a password
- Record names
- Record Personal Greetings
- Set Personal Group Distribution Lists
- Set a Personal Custom Service
- Set a Call Transfer Status
- Set Covering Extensions
- Set a Message Reception Mode
- Set an Incomplete Call Handling Status
- Set a Message Waiting Notification
- Record questions for an interview mailbox

3.2.2.1 Auto Receipt

Description

Allows a subscriber or the System Manager to receive a message to confirm the reception of sent messages. When the sent messages are received by a subscriber, the sender will receive a message announcing "(mailbox name/number) has received your message". Subscribers need to request the Auto Receipt when they send messages.

Conditions

- This feature is not available when the System Manager sends a Broadcasting Message.
- The reception of External Delivery Messages can be confirmed using this feature as well. Subscribers can request an Auto Receipt when sending a message to a single recipient or to all members of an External Delivery Message List, and receive an Auto Receipt for each member in the list.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Auto Receipt

3.2.2.2 Automatic Login

Description

Allows subscribers and managers to log in to their mailbox directly without entering the mailbox number. A subscriber/manager can log in to his or her mailbox directly by:

- dialling a Unified Messaging extension number directly from his or her extension.
- calling the Unified Messaging system from a telephone number that is assigned to log him or her into the mailbox.
- dialling (from an outside telephone) the DID number assigned to log him or her into the mailbox.
- calling the Unified Messaging system so that the call is received on the trunk group assigned to log him or her into the mailbox.

It is also possible for subscribers and managers to log in to their mailboxes without entering a password.

Notice

When disabling the password requirement, ensure that an unauthorised third-party is not allowed access to your extension.

Conditions

- Access from outside telephones will be enabled automatically after a Caller ID number, DID number, or trunk group number is assigned. A trunk group number or DID number can be assigned by the System Administrator only.
- When this feature is activated, "Toll Saver" is also available.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Remote Call/Automatic Login/Direct Service
→◆ Auto Login Extension
→◆ Auto Login Extension Password Entry Requirement
→◆ Auto Login Caller ID 1, 2
→◆ Auto Login Caller ID Password Entry Requirement
→◆ Auto Login DDI/DID
→◆ Auto Login TRG No.
→◆ Auto Login DDI/DID, TRG No. Password Entry Requirement
→◆ Auto Login Toll Saver

Feature Guide References

3.2.2.29 Toll Saver

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Assigning Your Telephone Numbers for Remote Automatic Log-in and Toll Saver

3.2.2.3 Autoplay New Message

Description

It is possible to play new messages automatically when a subscriber or the System Manager/Message Manager logs into his or her mailbox. There is no need to press [1] to receive the new messages.

Conditions

- If there is more than one new message in the mailbox, it can be set whether or not messages will be played continuously without system prompts.
  → 21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Play New Messages Sequentially

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Autoplay New Message

3.2.2.4 Bookmark

Description

Enables a subscriber to set one bookmark per message while pausing a message. After setting a bookmark, a subscriber can listen to the message from that bookmark by pressing the specified key while pausing the message or after the message was played.

User Manual References

Subscriber Operation

1.8.3 Message Playback and Related Features—◆◆ Bookmarks (Voice Messages only)

3.2.2.5 Call-through Service

Description

Allows subscribers to make outside calls by accessing the subscriber service menu (from an outside telephone) and dialling an outside destination. This creates a trunk-to-trunk call.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—◆ Call-through Service
24.4 UM Configuration—[5-4] System Parameters—Parameters—Transfer to Outside—◆ Outside Transfer Sequence—Call Transfer to Outside Sequence (Up to 16 digits / [0-9 * # D F R T , ; N])

Feature Guide References

2.16.1 Direct Inward System Access (DISA)

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Call-through Service

3.2.2.6 Call Transfer Scenario

Description

By combining the settings for Call Transfer and Incomplete Call Handling, you can create a call handling "scenario" for your extension. Up to 20 scenarios can be created, and a scenario can be assigned to each absent message.

PC Programming Manual References

8.1.1 Users—User Profiles—Advanced setting—Advanced Call Transfer Setting
8.1.1 Users—User Profiles—Advanced setting—Scenario Setting

Feature Guide References

2.20.2 Absent Message
3.2.2.7 Call Transfer Status
3.2.2.15 Incomplete Call Handling Service

3.2.2.7 Call Transfer Status

Description

Allows subscribers to specify how the Unified Messaging system will handle calls to their individual extensions. Call Transfer Status options include: Call Blocking, Call Screening, Intercom Paging, Transfer to Mailbox, Transfer to Specified Telephone Number, and Custom Service.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—◆ Call Transfer Status

Feature Guide References

3.2.1.23 Intercom Paging

3.2.2.8 Callback Number Entry

Description

Enables the caller to leave a callback number in several different ways depending upon which option is programmed into the Unified Messaging system.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Notification Parameters
24.4 UM Configuration—[5-4] System Parameters—Parameters—Dialling Parameters/MSW Notification

User Manual References

Manager Operation

2.2.2 Message Manager Features—◆◆ Setting Up Message Waiting Notification

3.2.2.9 Caller ID Callback

Description

Enables a subscriber to call back a message sender with the Caller ID number attached to the message.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Caller ID Callback
24.4 UM Configuration—[5-4] System Parameters—Parameters—Transfer to Outside—◆ Trunk Group (1–64)—Caller ID Callback

Feature Guide References

3.2.1.7 Call Transfer to Outside

3.2.2.10 Delete Message Confirmation

Description

It is possible to request confirmation from the subscriber before erasing a message left in the mailbox. When messages are deleted, they will be retained until the time specified in **System Maintenance Start Time** on the following day.

PC Programming Manual References

7.11 Utility—UM - System Maintenance—System Maintenance Start Time
21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Delete Message Confirmation

3.2.2.11 Direct Service Access

Description

Allows a subscriber to access a feature directly by dialling a Unified Messaging extension number. The following features can be accessed directly:

- Record No Answer Greeting
- Record Busy Greeting
- Record After Hours Greeting
- Record Temporary Greeting
- Change Day Main Menu*1
- Change Night Main Menu*1
- Change Emergency Greeting*1

*1 Access to this feature is available only to the Message Manager and to subscribers with Message Manager privileges.

Conditions

- This feature is only available when the Automatic Login feature is enabled.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Remote Call/Automatic Login/Direct Service

3.2.2.12 External Message Delivery Service

Description

Allows a subscriber to send a message to several subscribers and non-subscribers (including outside parties) at a specified time (or immediately). This feature also allows the receiver to reply to the message without having to specify the mailbox number.
Further, an External Message Delivery List allows a subscriber to send a message to multiple parties with a single operation. One subscriber can maintain up to 2 lists with a maximum of 8 entries in each.
Also, it is possible to require the receiver to enter a 4-digit password to receive the sender's message. If the receiver enters the password incorrectly 3 times, the Unified Messaging system plays: the Company Name (if

it has been recorded), the Company's Telephone Number (if registered), and the sender's extension (if both the Company's Telephone Number and the Extension of the Owner have been registered). With this information, the receiver can track down the message even if he or she does not remember the password.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—◆ Extension
20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—External MSG Delivery/Auto FWD/Personal Custom Serv
→◆ External Message Delivery Active
→◆ External Message Delivery Prompt Mode
24.4 UM Configuration—[5-4] System Parameters—Parameters—External Message Delivery

User Manual References

Subscriber Operation

1.8.4 Sending Messages
→◆◆ External Message Delivery
→◆◆ External Message Delivery Lists
→◆◆ Receiving External Delivery Messages

3.2.2.13 Forwarding to a Mailbox

Description

A subscriber can set calls to be forwarded to the floating extension number of the UM group. In this case, the call is forwarded directly to the extension's mailbox. Therefore the caller can leave a message without knowing the mailbox number.

Feature Guide References

2.3.2 Call Forwarding (FWD)

3.2.2.14 Group Distribution Lists

Description

There are 2 types of Group Distribution Lists:

- **Group Distribution List—Personal**
  Allows a subscriber to simultaneously send a message to several mailboxes. These lists can be created or edited by the System Administrator and by the subscriber. Each subscriber can maintain a maximum of 4 lists with up to 40 entries in each list.
- **Group Distribution List—System**
  Also called "Mailbox Groups".
  Allows subscribers and callers to send messages to all mailboxes that belong to the list at once. These lists are created by the System Administrator. There are 20 group lists available, and each group list can have a maximum of 200 members.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Personal Distribution List
24.1 UM Configuration—[5-1] System Parameters—Mailbox Group

User Manual References

Subscriber Operation

1.8.4 Sending Messages—◆◆ Personal Group Distribution Lists

Manager Operation

2.2.1 System Manager Features—◆◆ Broadcasting Messages

3.2.2.15 Incomplete Call Handling Service

Description

Allows the subscriber to offer callers several service options when the extension is busy or there is no answer. Options available for Incomplete Call Handling for Busy or No Answer are:

- Leaving a Message
- Transfer to Covering Extension
- Returning to the Automated Attendant Top Menu
- Intercom Paging
- Custom Service
- Calling Operators

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting
→◆ Incomplete Call Handling for No Answer
→◆ Incomplete Call Handling for Busy

Feature Guide References

3.2.1.23 Intercom Paging

User Manual References

Subscriber Operation

1.8.5 Transferring Calls—◆◆ Incomplete Call Handling

3.2.2.16 Live Call Screening (LCS)

Description

Allows the subscriber to monitor incoming calls as messages are being recorded. The subscriber has the option of answering calls while monitoring or allowing the message to be recorded without interruption. There are 2 modes in this feature: Hands-free and Private. Hands-free mode allows the subscriber to hear the caller through the telephone's speaker and answer the call by lifting the handset. Private mode alerts with a tone and requires

the telephone handset to be lifted before the message can be monitored. The subscriber can talk to the caller by pressing the LCS button on his or her proprietary telephone (if programmed).

Conditions

- This feature is not available for ISDN extensions and SIP extensions.
- **LCS Button**
  A flexible button can be customised as the LCS button.
- **Extension Personal Identification Number (PIN)**
  To prevent unauthorised monitoring, it is recommended the LCS user assign an extension PIN. This PIN will be required when setting LCS (→ 2.24.1 Extension Personal Identification Number (PIN)). If the user forgets the PIN, it can be cleared by an extension assigned as the manager.
- Each extension can be programmed to either end recording or continue recording the conversation after the call is intercepted, through personal programming (LCS Mode Set [After Answering]).
- To use the LCS feature on a PS in Wireless XDP Parallel Mode, LCS can only be turned on or off from the wired telephone. In Wireless XDP Parallel Mode, setting LCS on/off from the PS has no effect.
  (→ 5.2.4.5 Wireless XDP Parallel Mode)
- A sub extension of a one-numbered extension cannot perform LCS. (→ 2.11.11 One-numbered Extension).

PC Programming Manual References

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 4
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 4
21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Message Cancel for Live Call Screening

Feature Guide References

2.21.2 Flexible Buttons

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Live Call Screening (LCS)

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Class of Service (COS) Parameters

3.2.2.17 Mailbox

Description

Is a place where all messages left for a subscriber are stored. Several mailbox options exist: Subscriber mailbox, Interview Mailbox, System Manager's mailbox, and Message Manager's mailbox (General Delivery Mailbox).

Installation Manual References

5.11 Automatic Configuration of Mailboxes

PC Programming Manual References

7.5.3 Utility—Report—UM View Reports
Section 20 UM Configuration—[1] Mailbox Settings

Feature Guide References

3.1.2.1 Automatic Configuration of Mailboxes

User Manual References

Subscriber Operation

1.8.2 Logging in to and Configuring Your Mailbox

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Up Mailboxes

3.2.2.18 Mailbox Capacity Warning

Description

Allows the Unified Messaging system to alert subscribers when recording time for their mailboxes is running low. The warning announcement will be heard at the beginning of Subscriber's Service.

Conditions

- If the setting is longer than the Mailbox Capacity Maximum Message Time, this feature is not available.

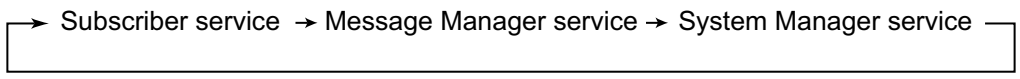
PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—Mailbox
→◆ Mailbox Capacity Warning (Selection)
→◆ Mailbox Capacity Maximum Message Time (Limited) (min)

3.2.2.19 Manager Service Switching

Description

Allows a subscriber to access the System Manager's mailbox and Message Manager's mailbox. Subscribers that have this feature enabled through Class of Service (COS) programming can switch mailboxes simply by pressing [#] twice from the subscriber service menu, in the following order:



Conditions

- If another user is already using the manager service, the subscriber will hear an error tone and cannot access that service.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—◆ System Manager Authority
21.1 UM Configuration—[2] Class of Service—◆ Message Manager Authority

3.2.2.20 Message Transfer

Description

Allows the subscriber to transfer messages to other mailboxes after listening to them. The subscriber can also add a personal comment at the beginning of the message before transferring it. One or more individual mailboxes can be specified for message transfer. Messages can also be transferred using either the System or Personal Group Distribution Lists.

User Manual References

Subscriber Operation

1.8.3 Message Playback and Related Features—◆◆ Transferring Messages

Manager Operation

2.2.2 Message Manager Features—◆◆ Managing the General Delivery Mailbox

3.2.2.21 Personal Custom Service

Description

Allows a subscriber to use the following Custom Services in his or her Personal Greeting (Personal Custom Service):

- Transfer to Mailbox
- Transfer to Extension
- Transfer to Voice Mail Service
- Transfer to Automated Attendant Service
- Custom Service
- Transfer to Operator
- Transfer to Outside
- Page the Party
- Repeat Greeting

A caller can select the desired service from those above while listening to the Personal Greeting or recording a message.
You can also choose what happens if a caller does not dial anything after the Personal Greeting (No DTMF Input Operation) as follows:

- Message Recording: The caller will be guided to leave a message.
- Disconnect (All Day): The line will be disconnected all day.
- Disconnect (Only After Hours): The line will be disconnected only after hours.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—External MSG Delivery/Auto FWD/Personal Custom Serv—◆ Personal Custom Service

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Personal Custom Service

3.2.2.22 Personal Greetings

Description

Are the greetings heard when a caller reaches a subscriber's mailbox. Subscribers can record 6 types of personal greetings. The maximum recording time for each personal greeting is determined by the subscriber's COS. Personal greetings are used in the following priority:

1. **Temporary Personal Greeting**
   Used to inform callers of the subscriber's absence (e.g., "I'm out of the office today…"). The Temporary Personal Greeting overrides other Personal Greetings; it is always played regardless of the time mode, whether or not the line is busy, etc. This feature is automatically activated when the subscriber records a Temporary Personal Greeting. He will then be asked whether it should be turned off the next time he logs in to his mailbox. When the Temporary Personal Greeting is turned off, it will be deleted automatically and other Personal Greetings will be played as normal.
2. **Busy Signal Greeting**
   Played whenever the subscriber's extension is busy.
3. **Personal Greeting for Caller ID**
   Allows subscribers to record a maximum of 4 personal greetings that are played for calls received from preprogrammed telephone numbers. Each greeting can be assigned to a maximum of 8 telephone numbers.
4. **Absent Message Greeting**
   Allows subscribers to assign a personal greeting to each of the PBX's absent messages.
5. **After Hours Greeting**
   Played when the system is in night mode.
6. **No Answer Greeting**
   Played during business hours (day, lunch, and break modes) when:
   – the caller is connected to the subscriber's extension but the call is not answered.
   – the caller logs in to the mailbox directly.
   – the Busy Signal greeting or the After Hours greeting has not been recorded.

PC Programming Manual References

Section 21 UM Configuration—[2] Class of Service
21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Personal Greeting for Caller ID

Feature Guide References

2.20.2 Absent Message

User Manual References

Subscriber Operation

1.8.2 Logging in to and Configuring Your Mailbox
→◆◆ Changing or Deleting Your Personal Greeting Messages
→◆◆ Personal Greetings for Caller ID

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Class of Service (COS) Parameters

3.2.2.23 Private Message

Description

Allows a subscriber and the System Manager to mark a message as "Private" when sending it to other subscribers, the Message Manager and/or the System Manager. A private message cannot be transferred. It is also possible to mark messages recorded in the Voice Mail service as "Private".

User Manual References

Subscriber Operation

1.8.4 Sending Messages—◆◆ Message Delivery

Manager Operation

2.2.1 System Manager Features—◆◆ Broadcasting Messages

3.2.2.24 Recover Message

Description

Enables a subscriber to recover deleted messages. After deleting messages, they will remain in the Deleted Message Box until the time specified in **System Maintenance Start Time** on the following day. Until then, the subscriber can retrieve the deleted messages from the Deleted Message Box and move them to the Old Message Box. Message retention time is reset when a message is recovered.

PC Programming Manual References

7.11 Utility—UM - System Maintenance—System Maintenance Start Time

User Manual References

Subscriber Operation

1.8.3 Message Playback and Related Features—◆◆ Recovering Deleted Messages

3.2.2.25 Remote Absent Message

Description

Allows a subscriber to change her absent message from a remote location by following the voice guidance.

Feature Guide References

2.20.2 Absent Message

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Setting Absent Message Remotely

3.2.2.26 Remote Call Forwarding Set

Description

Allows subscribers and the Message Manager to program their extensions from a remote location in order to forward calls to another extension or to an outside telephone.
There are 6 forwarding settings available:

- **FWD All**: Forwards all incoming calls to the desired extension number.
- **FWD Busy**: Forwards all incoming calls to the desired extension number when the line is busy.
- **FWD No Answer**: Forwards all incoming calls to the desired extension number when there is no answer.
- **FWD Busy or No Answer**: Forwards all incoming calls to the desired extension number when the line is busy or there is no answer.
- **FWD to CO**: Forwards all incoming calls to Telephone number 1 or 2 (programmed in the Mailbox Setting), or to any other number.
- **FWD Cancel**: Cancels the forwarding setting.

Conditions

- In order to use the FWD to CO option, the ability to forward calls to trunks must be enabled through system programming.
- Extensions assigned as operators can be called by dialling [0], however, when setting this feature the extension number (not "0") must be specified.

PC Programming Manual References

10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings
20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Remote Call/Automatic Login/Direct Service
21.1 UM Configuration—[2] Class of Service

User Manual References

Subscriber Operation

1.8.5 Transferring Calls—◆◆ Remote Call Forwarding

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Class of Service (COS) Parameters
2.2.2 Message Manager Features—◆◆ Remote Call Forwarding Set

3.2.2.27 Subscriber Tutorial

Description

Provides voice guidance to subscribers when they log in to their mailboxes for the first time. By following the guidance, subscribers can configure the following items:

- Password

- Owner's Name
- Personal Greetings (No Answer Greeting, Busy Signal Greeting, After Hours Greeting)

Note

If the simplified tutorial was selected through system programming, Busy Signal Greeting and After Hours Greeting cannot be recorded.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—◆ Tutorial

User Manual References

Subscriber Operation

1.8.2 Logging in to and Configuring Your Mailbox—◆◆ Subscriber Tutorial (Easy Mailbox Configuration)

Manager Operation

2.2.1 System Manager Features—◆◆ Setting Class of Service (COS) Parameters

3.2.2.28 Timed Reminder Setting

Description

Allows a subscriber to set a Timed Reminder by following system prompts provided by the Unified Messaging system. Timed Reminder is a feature that is similar to an alarm clock; the telephone will ring at the set time (once or daily) as set by the subscriber or message manager.
A special dial tone or prerecorded message will be heard by the subscriber when going off-hook to answer the Timed Reminder.

Conditions

- The time format (12 or 24 hour) used when setting the Timed Reminder is determined by the setting of "Position of "AM/PM" in Time Stamp".
  → 24.4 UM Configuration—[5-4] System Parameters—Parameters—Prompt Setting—◆ System Guidance—Select Language—Position of "AM/PM" in Time Stamp
- A subscriber must have his or her own extension in order to use this feature.
- Extensions assigned as operators can be called by dialling [0], however, when setting this feature the extension number (not "0") must be specified.

PC Programming Manual References

24.4 UM Configuration—[5-4] System Parameters—Parameters

Feature Guide References

2.24.4 Timed Reminder

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Setting the Alarm (Timed Reminder)

Manager Operation

2.2.2 Message Manager Features—◆◆ Setting the Timed Reminder

3.2.2.29 Toll Saver

Description

Allows a subscriber to check his or her mailbox from preprogrammed Caller ID numbers, the DID number or the telephone number specified to the preprogrammed trunk group number without incurring telephone charges.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Remote Call/Automatic Login/Direct Service
- →◆ Auto Login Extension
- →◆ Auto Login Extension Password Entry Requirement
- →◆ Auto Login Caller ID 1, 2
- →◆ Auto Login Caller ID Password Entry Requirement
- →◆ Auto Login DDI/DID
- →◆ Auto Login TRG No.
- →◆ Auto Login DDI/DID, TRG No. Password Entry Requirement
- →◆ Auto Login Toll Saver

23.1 UM Configuration—[4-1] Service Settings—Caller ID / PIN Call Routing—23.2 UM Configuration—[4-2] Service Settings—Parameters
- →◆ Delayed Answer Time for New Message (5-60 s)
- →◆ Delayed Answer Time for No New Message (5-60 s)

Feature Guide References

3.2.2.2 Automatic Login

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Assigning Your Telephone Numbers for Remote Automatic Log-in and Toll Saver

3.2.2.30 Two-way Record/Two-way Transfer

Description

The following manual recording features are available:

- **Two-way Record**
  Allows a subscriber to record the conversation that he or she is having with a caller. The conversation is saved in the subscriber's mailbox as either an old or new message, depending on the subscriber's COS.
- **Two-way Transfer**

Allows a subscriber to record the conversation that he or she is having with a caller. The conversation is saved in another subscriber's mailbox as a new message.

- **One-touch Two-way Transfer**
  Allows a subscriber to record the conversation that he or she is having with a caller, with a one-touch operation. The conversation is saved in another subscriber's mailbox as a new message.

Unlimited Message Length

Allows subscribers to record for an unlimited length of time when recording two-way conversations into their own or another subscriber's mailbox (Two-way Record or Two-way Transfer). The maximum recording time for other messages will automatically be set to 60 minutes.

Note

You should inform the other party that the conversation will be recorded before beginning to record any telephone conversation.

Conditions

- KX-NSU301, KX-NSU305, KX-NSU310, KX-NSU320, or KX-NSU399 (Activation Key for Two-way Recording) is required to use this feature. One activation key is required for each extension that will use this feature.
  Also, the **Two-way Recording** setting for each extension must be set to **Enable**.
- **Two-way Record/Two-way Transfer Button**
  A flexible button can be customised as the Two-way Record or the Two-way Transfer button. An extension number can be assigned to the Two-way Transfer button so that it can be used as a one-touch record button for the mailbox of the specified extension (**One-touch Two-way Transfer Button**).
- When all of the Unified Messaging ports are busy:
  a. Pressing the Two-way Record button sends a warning tone.
  b. Pressing the Two-way Transfer button followed by an extension number sends a warning tone.
- To allow unlimited recording time, the "Message Length" parameter of the subscriber's Class of Service (COS) must be set to "Unlimited".
  → 21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Message Length (Selection)

PC Programming Manual References

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—UM—◆ Two-way Recording
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Extension Number (for Two-way Record)
→◆ Extension Number (for Two-way Transfer)
→◆ Extension Number (for Voice Mail Transfer)
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—UM—◆ Two-way Recording
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Extension Number (for Two-way Record)
→◆ Extension Number (for Two-way Transfer)
→◆ Extension Number (for Voice Mail Transfer)
21.1 UM Configuration—[2] Class of Service—Mailbox—◆ Two-way Recorded Message Save Mode

Feature Guide References

2.21.2 Flexible Buttons

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ Recording Your Conversation
3.1.3 Customising the Buttons

3.2.2.31 Urgent Message

Description

Allows subscribers, callers, and the System Manager to specify a message as "Urgent". When a subscriber listens to messages, urgent messages will be given priority over other messages in a mailbox (i.e., they will be played first) if the "First Playback Urgent Messages" parameter of the subscriber's Class of Service (COS) is set to "Yes". If "Only Urgent Messages" is set as the Notification Type for the subscriber's mailbox, Message Waiting Notification will be activated only when an urgent message is received.

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Notification Parameters
→◆ Telephone Device—Device No. 1, 2, 3—Only Urgent Messages
→◆ E-mail/Text Message Device—Device No. 1, 2, 3—Only Urgent Messages
21.1 UM Configuration—[2] Class of Service—Mailbox—◆ First Playback Urgent Message

3.2.2.32 Voice Mail (VM) Transfer Button

Description

A subscriber can transfer a call to a mailbox of an extension by pressing the VM Transfer button and entering the number of the extension. The transferred caller can then leave a message in the mailbox. The VM Transfer button can also be used by extension users at other times, as follows:

a. By pressing the VM Transfer button when the extension is idle, the extension's mailbox is called and messages can be listened to. This feature can be used even if the Message Waiting lamp is not on.
b. Pressing the VM Transfer button while a call is incoming will redirect the call to the called extension's mailbox. This is useful when the called extension user does not want to answer the call.
c. If you call an extension, and the other party does not answer, pressing the VM Transfer button will transfer your call to the called extension's mailbox so that you can leave a message. This feature can also be used when the called extension is busy or set to DND.

Conditions

- A flexible button can be customised as the VM Transfer button with the floating extension number of the UM group as the parameter.
- If a PS is paired with a PT or SLT (in Wireless XDP Parallel mode), the PS's VM Transfer button cannot be used to redirect an incoming call to the called extension's mailbox. (→ 5.2.4.5 Wireless XDP Parallel Mode)

PC Programming Manual References

12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Extension Number (for Voice Mail Transfer)

12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Extension Number (for Voice Mail Transfer)

Feature Guide References

2.21.2 Flexible Buttons

3.2.2.33 Web Programming

Description

Subscribers can access and change various settings via Web Maintenance Console.

Users

Users logged in to a "User (User)" account can access the following items:

- **Mailbox settings**
  Subscribers can configure settings such as the name associated with their mailbox, the mailbox password, and notification parameters.
- **Voice prompts**
  Subscribers can play, record, and delete the following types of voice prompts for their mailbox:
  – Mailbox owner name
  – Personal greetings
  – Personal caller ID name
  – Personal distribution list voice label
  – Interview mailbox questions

Administrators

Users logged in to a "User (Administrator)" account can access the following items in addition to the settings available to "User (User)" accounts:

- **Voice prompts**
  – Custom service
  – Mailbox group name
  – System caller name
- **Reports**
  Administrators can view and clear the various types of reports.
  → 7.5.3 Utility—Report—UM View Reports

Conditions

- Subscribers must have a user ID and password to log in to Web Maintenance Console.
- The System Manager and the Message Manager do not necessarily have administrator privileges.

PC Programming Manual References

Section 20 UM Configuration—[1] Mailbox Settings
22.1 UM Configuration—[3-1] UM Extension / Trunk Service—Service Group—◆ Day, Night, Lunch, and Break Mode - Incoming Call Service Prompt
24.3 UM Configuration—[5-3] System Parameters—System Caller Name Announcement
24.4 UM Configuration—[5-4] System Parameters—Parameters—Prompt Setting

Feature Guide References

5.5.2 PC Programming

User Manual References

3.2 System Programming Using Web Maintenance Console

3.3 E-mail Client Integration Features

3.3.1 Integration with Microsoft Outlook

Description

Unified Messaging system integration with Microsoft Outlook allows subscribers to access the contents of their mailboxes through Microsoft Outlook in the same way they do e-mail. Voice messages appear in subscribers' inboxes as e-mail messages.
With the Outlook plug-in, subscribers can do the following:

- Play back voice messages directly from Outlook
- Record and send voice messages
- Forward and reply to messages
- Call back the sender of a message
- Export voice message data
- Attach voice messages data to other e-mail messages

Conditions

- Microsoft Outlook integration requires CA to be installed on the subscriber's computer (→ 2.26.2 CA (Communication Assistant)).
  - CA Pro, CA Operator Console, or CA Supervisor is required to use all the integration features.
  - CA Basic-Express users cannot access the call history or view the presence of extensions from Outlook.
- PBX resources are required to access the Unified Messaging system. To reduce the possibility that PBX performance will be affected, the account in Microsoft Outlook that is set up to access the subscriber's mailbox should be set to off-line mode. For details about off-line mode, refer to the documentation for Microsoft Outlook.
- Integration is available with Microsoft Outlook 2003 or later.
- For Outlook 2003 and Outlook 2007: While a subscriber is connected to his mailbox through Outlook, he will not be able to access the mailbox from a telephone. The subscriber must exit Outlook before he can access the mailbox's subscriber services from a telephone.
  This condition does not apply to users of Outlook 2007 SP2 or later.
- KX-NSU201, KX-NSU205, KX-NSU210, KX-NSU220 or KX-NSU299 (Activation Key for Unified Messaging E-mail Notification) is required to use this feature. One activation key is required for each mailbox that will be used with this feature.
- For details about installing and using Outlook integration, refer to the documentation for CA.

PC Programming Manual References

21.1 UM Configuration—[2] Class of Service—General—◆ Desktop Messaging

User Manual References

Subscriber Operation

1.8.6 Other Features

3.3.2 IMAP Integration

Description

By configuring an IMAP account, subscribers can access the contents of their mailboxes through an e-mail client. All that is necessary is an e-mail client that supports IMAP4.
With IMAP integration, users can do the following:

- Play back voice messages
- Save voice message data to their PCs
- Delete voice messages

Conditions

- KX-NSU201, KX-NSU205, KX-NSU210, KX-NSU220, or KX-NSU299 (Activation Key for Unified Messaging E-mail Notification) is required to use this feature. One activation key is required for each mailbox that will be used with this feature.
- PBX resources are required to access the Unified Messaging system. To reduce the possibility that PBX performance will be affected, the account in the e-mail client that is set up to access the subscriber's mailbox should be set to off-line mode. For details about off-line mode, refer to the e-mail client's documentation.
- New voice messages can be sent as attachments to e-mails to a separate e-mail account (→ 3.2.1.28 Message Waiting Notification—E-mail Device).
- A tool for users to help manage IMAP connections is available. A link to where users can download the tool is provided in Web Maintenance Console.
  → "Unified Message"—"Unified Messaging Plug in" in 8.2.1 Users—Add User—Single User

PC Programming Manual References

20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Mailbox Parameters—◆ Mailbox Password (Message Client)
21.1 UM Configuration—[2] Class of Service—General
→◆ Desktop Messaging

User Manual References

Subscriber Operation

1.8.6 Other Features—◆◆ IMAP

Section 4

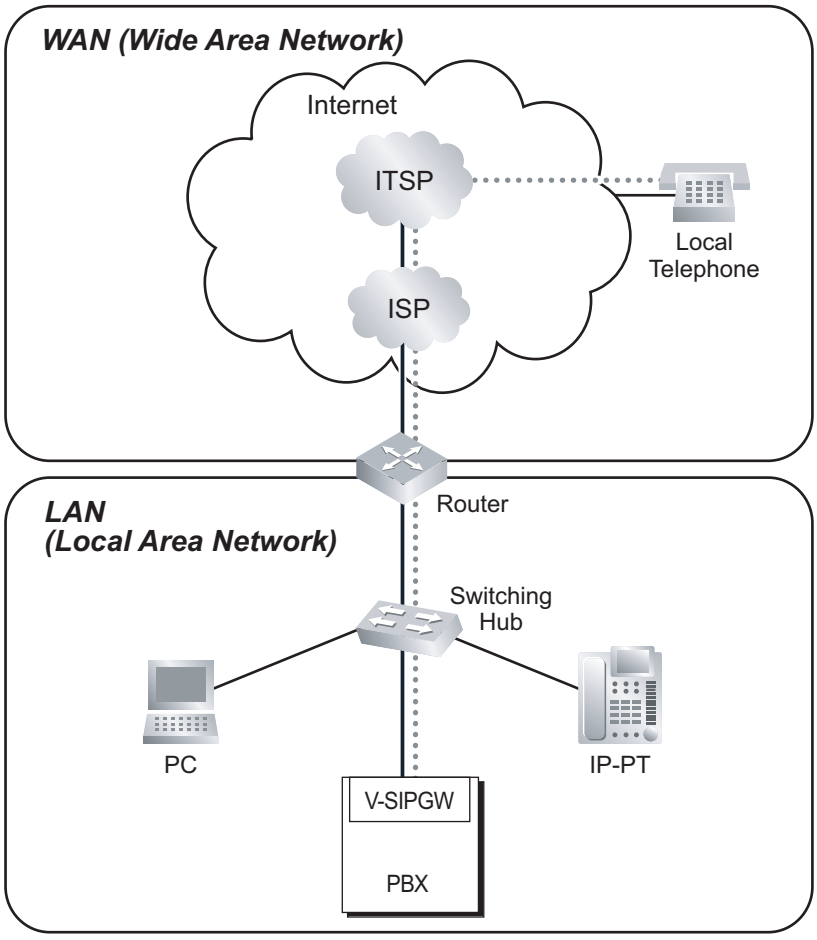
Network Features

4.1 Public Network Features

4.1.1 SIP (Session Initiation Protocol) Trunk

Description

Through a V-SIPGW card, the PBX can connect to an Internet telephony service provided by an ITSP (Internet Telephony Service Provider).
An ITSP provides its telephony service partly through the conventional telephone network (e.g., ISDN and Mobile), which is fee-based. An ISP (Internet Service Provider), which can also act as a SIP provider, does not provide the telephone connection itself. However, providing its users with Internet access, an ISP provides voice communication on the Internet for free. In this way, with VoIP technology based on the SIP protocol, the cost of voice communication can be much cheaper than conventional telephone networks.



Conditions

- A subscription with an ISP is required for an Internet connection.
- A subscription with an ITSP is required for a telephone connection. The ISP and ITSP may be part of the same company.

Installation Manual References

4.4 Virtual Cards
8.3 SIP Trunks

PC Programming Manual References

9.8 PBX Configuration—[1-1] Configuration—Slot—V-SIPGW—Shelf Property
9.10 PBX Configuration—[1-1] Configuration—Slot—V-SIPGW—Port Property

4.1.2 Integrated Services Digital Network (ISDN) Service Features

4.1.2.1 Integrated Services Digital Network (ISDN)—SUMMARY

Description

ISDN is a digital switching and transmission network. ISDN transmits voice, data, and image in digital format. ISDN lines, if available, can be connected to public line (trunk), private line (QSIG), or ISDN terminal devices (extension).

1. ISDN Interface and Configuration

Interface Type	Description	Port Mode	Configuration Type
Primary Rate Interface (PRI)	Provides thirty or twenty-three 64 kbps B channels for communication and one 64 kbps D channel for signalling (30B + D/23B + D).	Trunk, QSIG (Master, Slave), Extension*1	Point-to-Point

*1 → 9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Main—◆ Port Type

Note

Point-to-Point (P-P):
One ISDN terminal device can be connected to one ISDN port.

2. ISDN Supplementary Service Table for Public Network

Service	Description & Reference
Direct Dialling In (DDI)	Directs a call with a DDI number to a preprogrammed destination. (P-P only) → 2.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)
Calling Line Identification Presentation (CLIP)	Sends the caller's telephone number to the network when making a call. The called party can see the number on his telephone display before answering the call. → 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP)
Connected Line Identification Presentation (COLP)	Sends the telephone number of the answered party to the network when answering a call. The caller can see the number on his telephone display when the line is connected. → 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP)
Calling Line Identification Restriction (CLIR)	Prevents the caller's CLI being presented to the called party by the caller. → 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP)

Service	Description & Reference
Connected Line Identification Restriction (COLR)	Prevents COLP being sent by the answered party. → 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP)
Subaddressing (SUB)	You may add digits after the telephone number. These digits will be passed to ISDN terminal device.
Advice of Charge (AOC)	The PBX can receive the call charge information on ISDN lines from the telephone company. → 4.1.2.3 Advice of Charge (AOC)
Call Forwarding (CF)—by ISDN	Forwards an incoming call to another outside party using the ISDN service of the telephone company. → 4.1.2.4 Call Forwarding (CF)—by ISDN (P-P)
Call Hold (HOLD)—by ISDN	Puts one ISDN call on hold. → 4.1.2.5 Call Hold (HOLD)—by ISDN
Call Transfer (CT)—by ISDN	Transfers an ISDN call to an outside party. Call Transfer with Announcement and Call Transfer without Announcement are available. → 4.1.2.6 Call Transfer (CT)—by ISDN
Three-party Conference (3PTY)—by ISDN	Establishes a three-party conference call using the ISDN service of the telephone company. → 4.1.2.7 Three-party Conference (3PTY)—by ISDN
Malicious Call Identification (MCID)	An extension user can ask the telephone company to trace a malicious caller. Information on the malicious call will be received later on. → 4.1.2.8 Malicious Call Identification (MCID)
Completion of Calls to Busy Subscriber (CCBS)	If a call is made to an outside party and the party is busy, an extension can receive callback ringing when the called party becomes free. → 4.1.2.9 Completion of Calls to Busy Subscriber (CCBS)

3. **ISDN Centrex Service**
   An extension user can have access to the features of the ISDN Centrex Service of the telephone company (e.g., Call Transfer). This is used by putting the current ISDN call on hold by sending a flash/recall signal. This feature is enabled or disabled for each ISDN port.
   → 9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—CO Setting—◆ ISDN Centrex
4. **Private Networking Connection (QSIG)**
   Private networking with QSIG is possible using an ISDN line. The QSIG mode, Master or Slave, can be enabled on an ISDN port basis.
5. **ISDN Extension** (→ 5.2.5.1 ISDN Extension)

An ISDN (PRI) port can be used for extension connection. While the extension connection is enabled, ISDN terminal devices (e.g., ISDN telephone, G4 fax machine, PC) or a behind PBX can be connected to the port.
When the ISDN port is in P-P configuration, one ISDN terminal device can be connected to the port.

Conditions

- **Overlap/En bloc**
  For each ISDN port, either Overlap or En bloc can be selected as the dialling method for which the PBX sends telephone numbers to the telephone company. The selected dialling method must be offered by the telephone company. When "**Overlap**" is selected, the PBX sends each dialled digit individually.
  When "**En bloc**" is selected, the PBX sends all of the dialled digits at once.
  → 9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—CO Setting—◆ ISDN Outgoing Call Type

  In En bloc mode, the PBX recognises that the user is finished dialling when:
  – the # key is pressed (programmable).
    → 10.9 PBX Configuration—[2-9] System—System Options—Option 2—◆ ISDN en Bloc Dial—[#] as End of Dial for en Bloc mode
  – the dialled number is a preprogrammed telephone number.
    → 11.1.4 PBX Configuration—[3-1-4] Group—Trunk Group—Dialling Plan
  – the inter-digit timer expires.
    → 10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Dial—Extension Inter-digit (s)
- Some supplementary services are provided by the key protocol (→ 4.1.2.10 ISDN Service Access by Keypad Protocol).
- **Extension Number**
  An extension number can be assigned to each ISDN port.
  → 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ Extension Number
- **Bearer Mode**
  The bearer mode can be assigned on an extension basis.
  → 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 7—◆ ISDN Bearer
  → 12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 7—◆ ISDN Bearer

PC Programming Manual References

9.25 PBX Configuration—[1-1] Configuration—Slot—Card Property - PRI type
9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port
10.3 PBX Configuration—[2-3] System—Timers & Counters—Dial / IRNA / Recall / Tone—◆ Dial—Extension Inter-digit (s)
10.9 PBX Configuration—[2-9] System—System Options—Option 2—◆ ISDN en Bloc Dial—[#] as End of Dial for en Bloc mode
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Main—◆ Extension Number
→Option 7—◆ ISDN Bearer
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 7—◆ ISDN Bearer
11.1.4.1 PBX Configuration—[3-1-4] Group—Trunk Group—Dialling Plan—Auto Assign

Feature Guide References

4.2.4 QSIG Standard Features

4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP)

Description

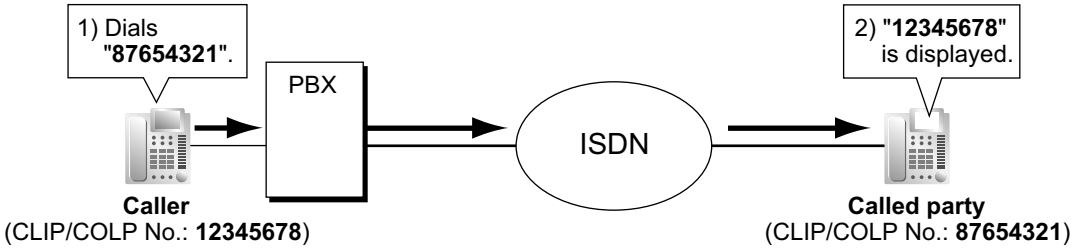
Calling Line Identification Presentation (CLIP):

The PBX can send a preprogrammed telephone number to the network when an extension user makes a call. The called party can see the number on his telephone display before answering the call.

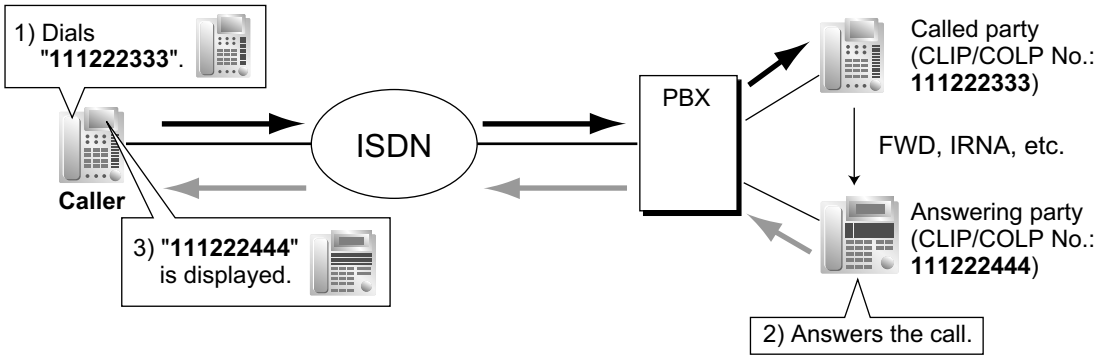
Connected Line Identification Presentation (COLP):

The PBX sends a preprogrammed telephone number to the network when the extension user answers an incoming call. The caller can see the number of the answering party on his telephone display when the call is answered.

[CLIP Example]



[COLP Example]



CLIP/COLP Number:

The telephone numbers sent to the network for CLIP/COLP can be assigned as follows:

- CLIP/COLP number for each ISDN port (subscriber's number).
- CLIP/COLP number for each extension.
- CLIP/COLP number for each incoming call distribution group.

Each extension can select either the CLIP/COLP number for the ISDN port or the extension to be used. The CLIP/COLP number for the incoming call distribution group is used when making a call by pressing the ICD Group button or receiving a call which arrives at the ICD Group button.

Calling/Connected Line Identification Restriction (CLIR/COLR):

It is possible for each extension to restrict the sending of its telephone number to the network by pressing the CLIR button, COLR button, or entering the feature number.

Conditions

- The availability of this feature is dependent on the contract with the telephone company.
- CLIP/COLP features comply with the following European Telecommunication Standard (ETS) specifications:
  – ETS 300 092 Calling Line Identification Presentation (CLIP) supplementary service.
  – ETS 300 097 Connected Line Identification Presentation (COLP) supplementary service.
- CLIR/COLR features comply with the following European Telecommunication Standard (ETS) specifications:
  – ETS 300 093 Calling Line Identification Restriction (CLIR) supplementary service.
  – ETS 300 098 Connected Line Identification Restriction (COLR) supplementary service.
- The CLIP/COLP number for the connected ISDN port can be used for the ISDN terminal devices which cannot be assigned their own CLIP/COLP number, such as a doorphone.
- **COLP/CLIR/COLR Assignment for Each Port**
  Each service can be enabled or disabled on each ISDN port of the PBX.
- **CLIR Button and COLR Button**
  It is possible to switch between CLIP and CLIR by pressing the CLIR button, and COLP and COLR by pressing the COLR button. A flexible button can be customised as the CLIR or COLR button.
- The CLIP/COLP number must match the telephone number provided by the telephone company. Otherwise it will be ignored or replaced by another number.
- When using a private network, the extension number assigned for each extension through system programming is sent for CLIP/COLP. (→ 4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG)
- When forwarding calls to a public trunk, system programming selects whether the CLIP number of the calling party or of the forwarding extension is sent to the forward destination.
  However, if the call is transferred to another PBX via a private network from a VPS or a UM, the CLIP number of the calling party is always sent, regardless of system programming.

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port
→CO Setting—◆ Subscriber Number
→Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ COLR Set / Cancel
→◆ CLIR Set / Cancel
→◆ Switch CLIP of CO Line / Extension
10.9 PBX Configuration—[2-9] System—System Options—Option 4
→◆ Send CLIP of CO Caller—when call is transferred to CO (CLIP of Held Party)
→◆ Send CLIP of CO Caller—when call is forwarded to CO
→◆ Send CLIP of Extension Caller—when call is forwarded to CO
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Main—◆ CLIP on ICD Group Button
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings
→Main—◆ Extension Number
→CLIP—◆ CLIP ID
→CLIP—◆ CLIP on Extension/CO
→CLIP—◆ CLIR
→CLIP—◆ COLR
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings
→CLIP—◆ CLIP ID

→CLIP—◆ CLIP on Extension/CO
→CLIP—◆ CLIR
→CLIP—◆ COLR
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Type

Feature Guide References

2.21.2 Flexible Buttons

PT Programming Manual References

[003] Extension Number
[606] CLIP/COLP Number

User Manual References

1.9.4 Displaying Your Telephone Number on the Called Party and Caller's Telephone (Calling/Connected Line Identification Presentation [CLIP/COLP])
1.9.5 Preventing Your Telephone Number Being Displayed on the Caller's Telephone (Connected Line Identification Restriction [COLR])
1.9.6 Preventing Your Number Being Displayed on the Called Party's Telephone (Calling Line Identification Restriction [CLIR])

4.1.2.3 Advice of Charge (AOC)

Description

The PBX can receive the call charge information on ISDN lines from the telephone company.
There are the following types:

Type	Description
Advice of Charge During Call (AOC-D)	AOC is received during the call and when the call is completed.
Advice of Charge At End of Call (AOC-E)	AOC is received when the call is completed.

Conditions

- This feature complies with the following European Telecommunication Standard (ETS) specification:
  – ETS 300 182 Advice of Charge (AOC) supplementary service Digital Subscriber Signalling System No. One (DSS1) protocol.
- A DPT user can see the call charge information on the display during the call.
- **Budget Management**
  If the amount of call charge reaches the preprogrammed limit, an extension user cannot make further calls. (→ 2.7.2 Budget Management)
- **AOC for ISDN extension**
  An ISDN extension also receives AOC.

PC Programming Manual References

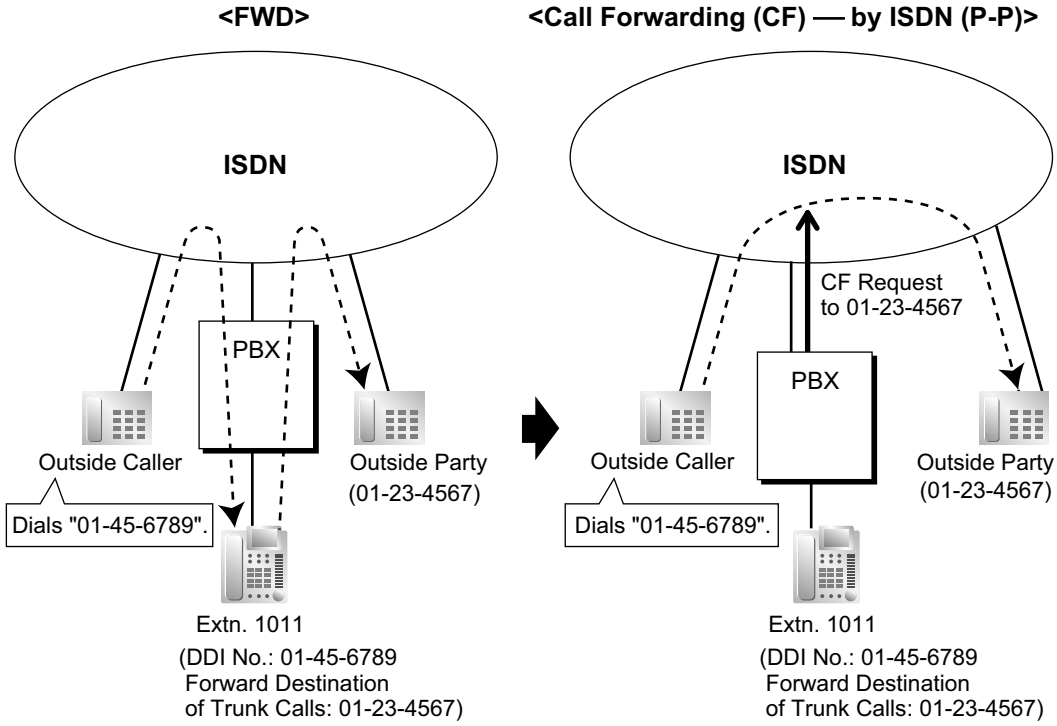
9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Supplementary Service—◆COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

4.1.2.4 Call Forwarding (CF)—by ISDN (P-P)

Description

An extension user can forward the incoming ISDN call to another outside party using the ISDN service of the telephone company, instead of the PBX feature, when the call is received through an ISDN line.
The network directly forwards the call to the destination which the extension user has set in the PBX as the forward destination of trunk calls; the network is instructed by the PBX. This feature is available only when the call is received through an ISDN port which supports this feature.
Call Forwarding—Unconditional (CFU), Call Forwarding—Busy (CFB), and Call Forwarding—No Reply (CFNR) are applied to this feature.

[Example]



Conditions

- This feature complies with the following European Telecommunication Standard (ETS) specification:
  – ETS 300 207 Diversion supplementary service.
- The availability of this feature is dependent on the contract with the telephone company.
- This feature can be enabled or disabled on each ISDN port of the PBX.
- This feature is available when the same trunk group is used for the incoming call and the forwarded call.

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Supplementary Service—◆COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

Feature Guide References

2.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)—SUMMARY
2.3.2 Call Forwarding (FWD)

User Manual References

1.6.1 Forwarding Calls

4.1.2.5 Call Hold (HOLD)—by ISDN

Description

An ISDN call can be put on hold using the ISDN service of the telephone company, instead of the PBX feature. This can be a part of a Call Transfer (CT)—by ISDN (→ 4.1.2.6 Call Transfer (CT)—by ISDN) and Three-party Conference (3PTY)—by ISDN (→ 4.1.2.7 Three-party Conference (3PTY)—by ISDN). This feature allows an ISDN call to be held, and a call to be made to another outside party using only one communication channel of ISDN. A PT user can easily use this feature by pressing the ISDN Hold button.

Conditions

- This feature complies with the following European Telecommunication Standard (ETS) specification:
  – ETS 300 141 Call Hold (HOLD) supplementary service.
- **ISDN Hold Button**
  A flexible button can be customised as the ISDN Hold button.
- The availability of this feature is dependent on the contract with the telephone company.
- The TRS/Barring feature is applied when making a call after activating this feature. (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring))
- ARS cannot be applied to the call dialled after activating this feature. (→ 2.8.1 Automatic Route Selection (ARS))
- It is impossible to seize any other trunk during this feature.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ ISDN Hold
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Type

Feature Guide References

2.21.2 Flexible Buttons

4.1.2.6 Call Transfer (CT)—by ISDN

Description

An ISDN call can be transferred to an outside party using the ISDN service of the telephone company, instead of the PBX feature, without occupying a second ISDN line.

Conditions

- This feature complies with the following European Telecommunication Standard (ETS) specification:
  – ETS 300 369 Explicit Call Transfer (ECT) supplementary service.
- The availability of this feature is dependent on the contract with the telephone company.
- This feature can be enabled or disabled on an ISDN port basis.
- If an ISDN port is in P-P configuration, this feature can be used only when the network supports the "explicit linkage" option.
- Call Transfer with Announcement and Call Transfer without Announcement is possible. (→ 2.12.1 Call Transfer)
- The call charges after completing this feature will not be recorded by the PBX.

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Supplementary Service—◆COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

User Manual References

1.4.1 Transferring a Call (Call Transfer)

4.1.2.7 Three-party Conference (3PTY)—by ISDN

Description

During a conversation using an ISDN line, an extension user can add another party and establish a three-party conference call using the ISDN service of the telephone company, instead of the PBX feature.

Conditions

- This feature complies with the following European Telecommunication Standard (ETS) specification:
  – ETS 300 188 Three-Party (3PTY) supplementary service.
- The availability of this feature depends on the contract with the telephone company.
- This feature can be enabled or disabled on an ISDN port basis.

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

User Manual References

1.4.5 Multiple Party Conversation

4.1.2.8 Malicious Call Identification (MCID)

Description

An extension user can ask the telephone company to trace a malicious caller during a call or while hearing reorder tone after the caller hangs up. Information on the malicious call will be received later on.

Conditions

- This feature complies with the following European Telecommunication Standard (ETS) specification:
  – ETS 300 130 Malicious Call Identification (MCID) supplementary service.

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ MCID

User Manual References

1.3.6 Identifying Malicious Calling Parties (Malicious Call Identification [MCID])

4.1.2.9 Completion of Calls to Busy Subscriber (CCBS)

Description

If the called party is busy and the call has been made using an ISDN line, an extension user can set to receive callback ringing when the called party becomes free. When the user answers the callback ringing, that party's number is automatically dialled.

Conditions

- This feature complies with the following European Telecommunication Standard (ETS) specification:
  – ETS 300 359 Completion of Calls to Busy Subscriber (CCBS) supplementary service.
- This feature is available under the following conditions:
  a. The caller's PBX is capable of using CCBS and the service is provided by the network.
  b. The called party's PBX is capable of accepting CCBS.
- To receive and send CCBS, receiving and sending CCBS must be enabled individually on an ISDN port basis through system programming.
- An extension user can set only one CCBS. The last setting is effective.
- The CCBS setting is cancelled if there is no callback ringing within 60 minutes or callback ringing is not answered within 10 seconds.
- After using the CCBS feature, using Last Number Redial will not retrieve the number dialled by CCBS. (→ 2.6.3 Last Number Redial)
- An extension user that has set the CCBS feature cannot receive callback ringing while the extension is holding a call.

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Supplementary Service
→◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY
→◆ CCBS Type
→◆ CCBS Delete Digits

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer

4.1.2.10 ISDN Service Access by Keypad Protocol

Description

ISDN provides some supplementary services by key protocol, and they may require a service access code to be dialled.

Conditions

- This feature complies with the following European Telecommunication Standard (ETS) specification:
  – ETS 300 122 Generic keypad protocol for the support of supplementary service (ISDN Service Access).
- **ISDN Service Button**
  A flexible button can be customised as an ISDN Service button. A service access code can also be assigned on this button for a quick operation.
- This feature is not available to an SLT.

PC Programming Manual References

12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Dial (for ISDN Service)
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Dial (for ISDN Service)

Feature Guide References

2.21.2 Flexible Buttons
6.1 Capacity of System Resources

User Manual References

1.2.5 Accessing the ISDN Service (ISDN Service Access)

4.2 Private Network Features

4.2.1 TIE Line Service

Description

A TIE line is a privately leased communication line between two or more PBXs, which provides cost effective communications between company members at different locations.

Interface

The following interfaces can be used to establish a private network:

Interface	Network Type
VoIP (H.323)	Internet Protocol (IP)
PRI (QSIG)	Digital (ISDN/30B+D)

→ 9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Network Numbering Plan—◆ Trunk Property

Conditions

A TIE line connection can be established through a Trunk Adaptor using a PRI line (QSIG).

Feature Guide References

2.1.1.5 Intercept Routing
2.3.2 Call Forwarding (FWD)
2.7.1 Toll Restriction (TRS)/Call Barring (Barring)
2.8.1 Automatic Route Selection (ARS)
2.12.1 Call Transfer
2.16.1 Direct Inward System Access (DISA)
5.5.7 Flexible Numbering/Fixed Numbering
6.1 Capacity of System Resources

User Manual References

1.2.1 Basic Calling

4.2.1.1 Making a TIE Line Call

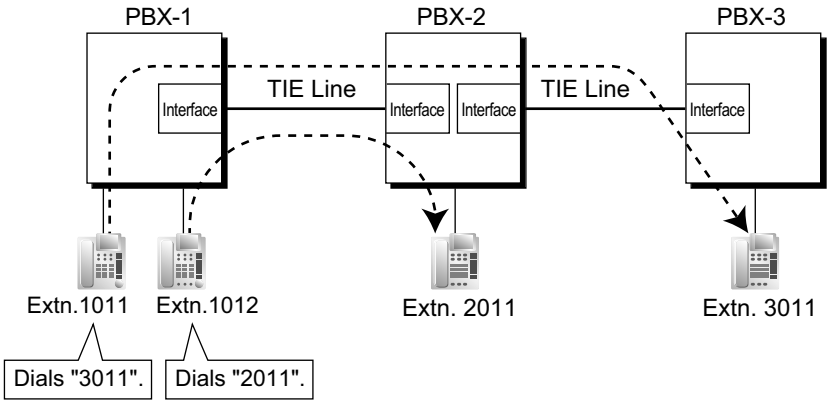
Description

One of the following two methods can be used to make a TIE line call.

<Extension Number Method (Access without PBX Code)>

Dial the [Extension Number] only.

[Example]



Explanation:

To use this method, it is necessary to change the first one or two digits of extension numbers of either PBX (e.g., 10XX for PBX-1, 20XX for PBX-2) to allow calls to be routed properly.

Case 1:

Extension 1012 of PBX-1 dials extension number "2011".

→ Extension 1012 of PBX-1 is connected to extension 2011 of PBX-2.

Case 2:

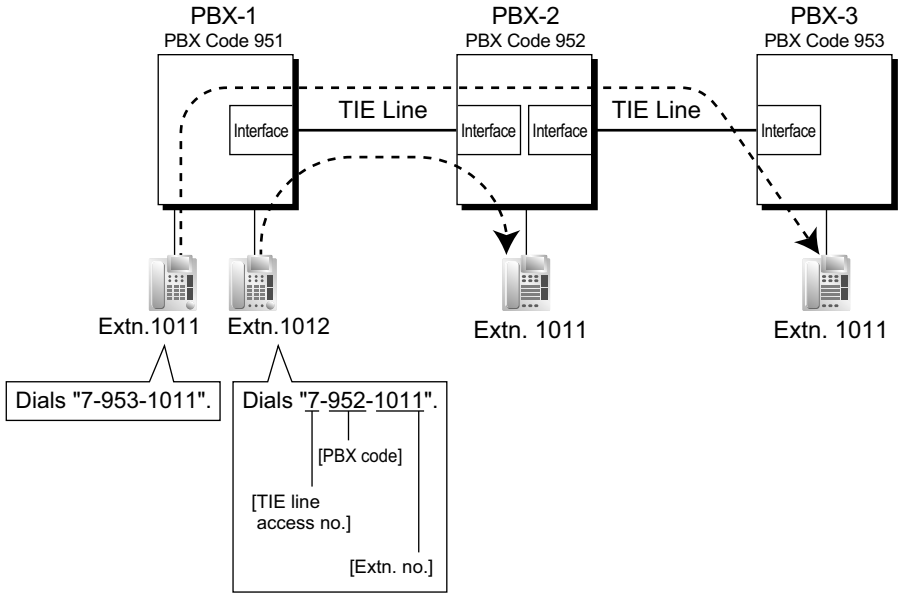
Extension 1011 of PBX-1 dials extension number "3011".

→ Extension 1011 of PBX-1 is connected to extension 3011 of PBX-3.

<PBX Code Method (Access with PBX Code)>

Dial the [TIE Line Access Number] + [PBX Code] + [Extension Number].
→ 10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ TIE Line Access
→ 17.1 PBX Configuration—[9-1] Private Network—TIE Table—◆ Own PBX Code

[Example]



Explanation:
To use this method, it is necessary to know each PBX code in order to identify the location of an extension.
Case 1:
Extension 1012 of PBX-1 dials TIE line access number "7", PBX code "952", and extension number "1011".
→ Extension 1012 of PBX-1 is connected to extension 1011 of PBX-2.
Case 2:
Extension 1011 of PBX-1 dials TIE line access number "7", PBX code "953", and extension number "1011".
→ Extension 1011 of PBX-1 is connected to extension 1011 of PBX-3.

4.2.1.2 TIE Line and Trunk Connection

Description

To connect the TIE line with the trunk, the following patterns are available:

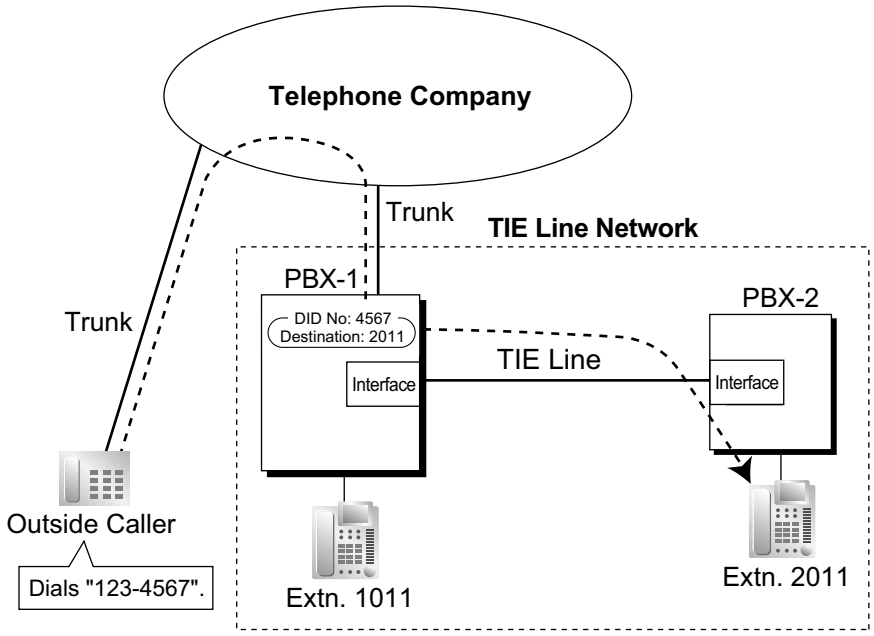
1. Trunk-to-TIE Access
2. TIE-to-Trunk Access
3. Trunk-to-TIE-to-Trunk Access

Trunk-to-TIE Access

It is possible to assign an extension of another PBX as the destination of incoming trunk calls to the own PBX. It is also possible to forward calls using a virtual PS. Using this method, trunk calls received at PBX-1 are forwarded directly to the extension at PBX-2, even when using the PBX Code method.

a. Incoming Trunk Call Destination Assignment

[Example]



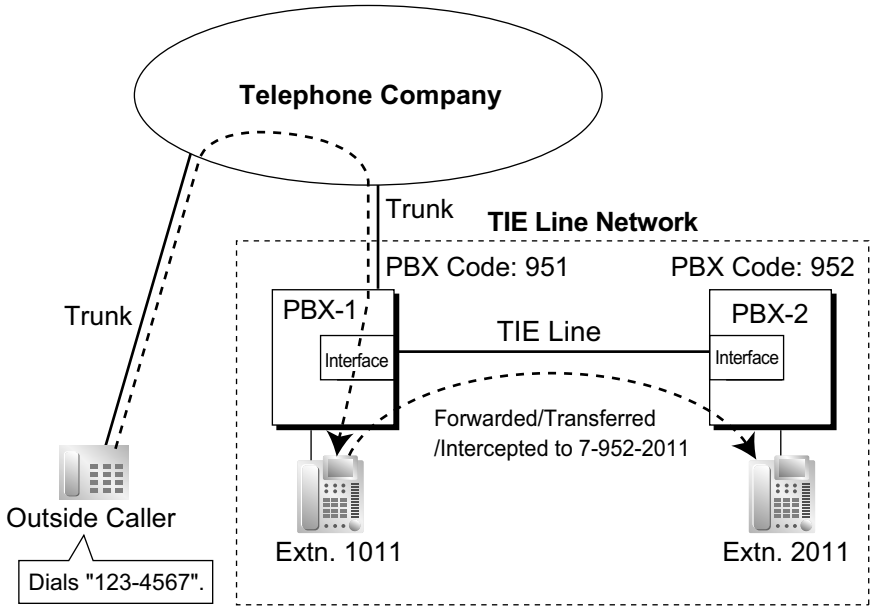
Explanation:

An outside caller dials "123-4567". The call is sent to extension "2011" of PBX-2 through the TIE line according to the assignment of the DID call destination of PBX-1. (→ 2.1.1.3 Direct Inward Dialling (DID)/ Direct Dialling In (DDI))

→ 18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table—◆ DDI / DID Destination—Day, Lunch, Break, Night

b. FWD/Call Transfer/Intercept Routing to the TIE Line

[Example]



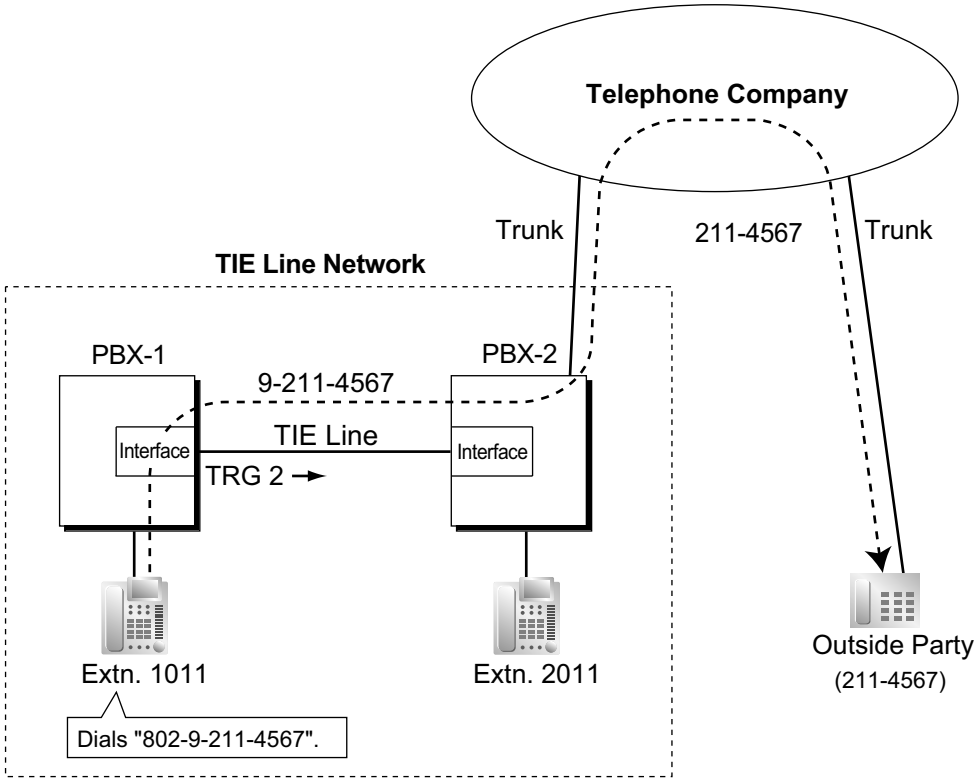
Explanation:
An outside caller dials "123-4567". The call reaches the destination (extension 1011 of PBX-1), and the call is forwarded, transferred, or intercepted to extension "2011" of PBX-2 through the TIE line.

TIE-to-Trunk Access

The PBX sends TIE line calls to the trunks of another PBX through the TIE lines.

a. Trunk Call through Other PBXs

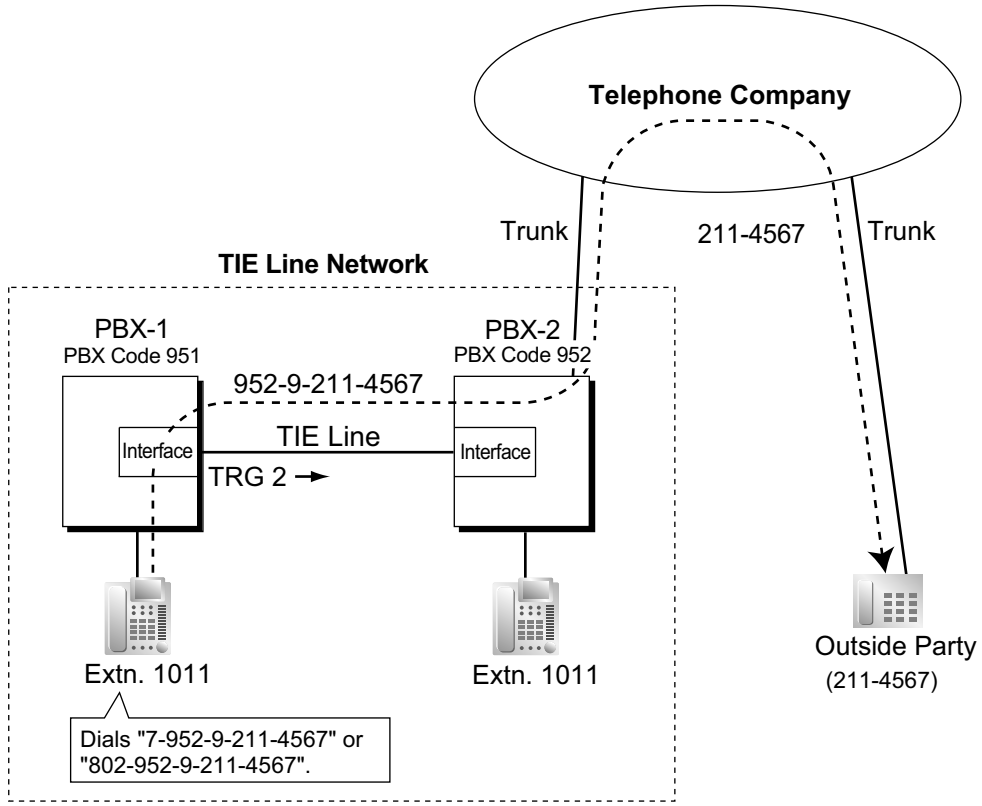
[Example]
<Extension Number Method (Access without PBX Code)>



Explanation:

1. Extension 1011 of PBX-1 dials the Trunk Group Access number of PBX-1 "8", trunk group number "02" (TRG2), Idle Line Access number of PBX-2 "9", and telephone number "211-4567".
   → 10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Trunk Group Access
2. PBX-1 sends the call to PBX-2 through the trunk group (TRG) 2 (TIE line).
3. PBX-2 sends the call to the outside party "211-4567".

<PBX Code Method (Access with PBX Code)>

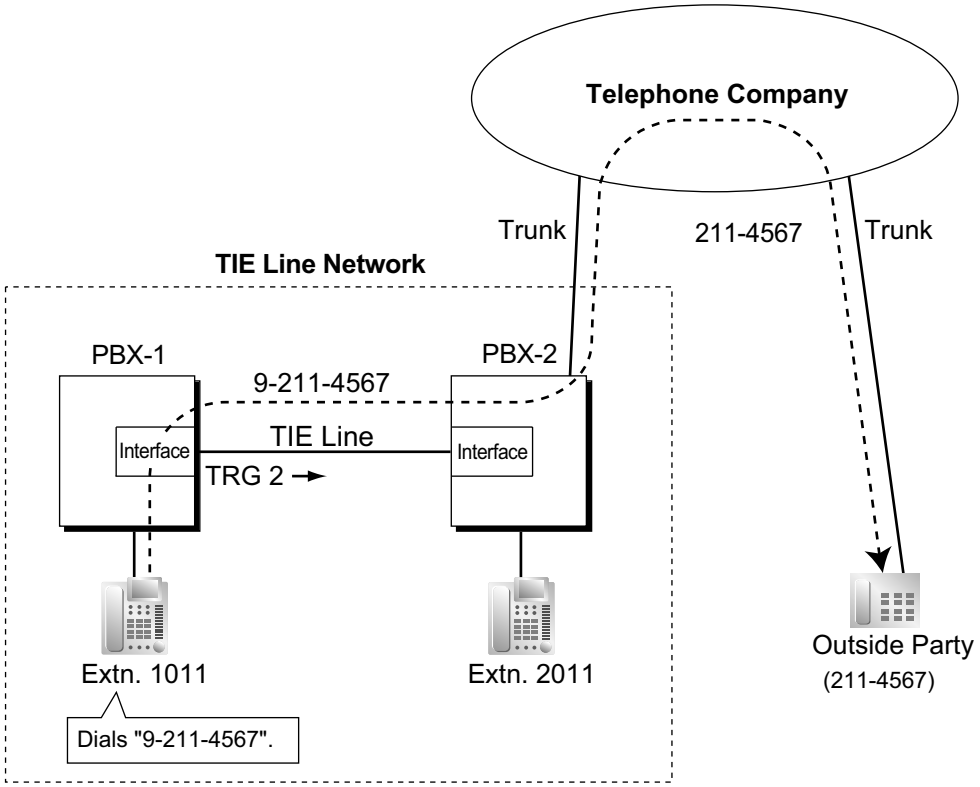


Explanation:

1. Extension 1011 of PBX-1 dials the TIE line access number "7", PBX code "952", Idle Line Access number of PBX-2 "9", and telephone number "211-4567"; or dials the Trunk Group Access number of PBX-1 "8", trunk group number "02" (TRG2), PBX code "952", Idle Line Access number of PBX-2 "9", and telephone number "211-4567".
2. The call is connected to the outside party "211-4567" through PBX-2 which has PBX code "952".

Trunk Call through Other PBXs—by the ARS feature

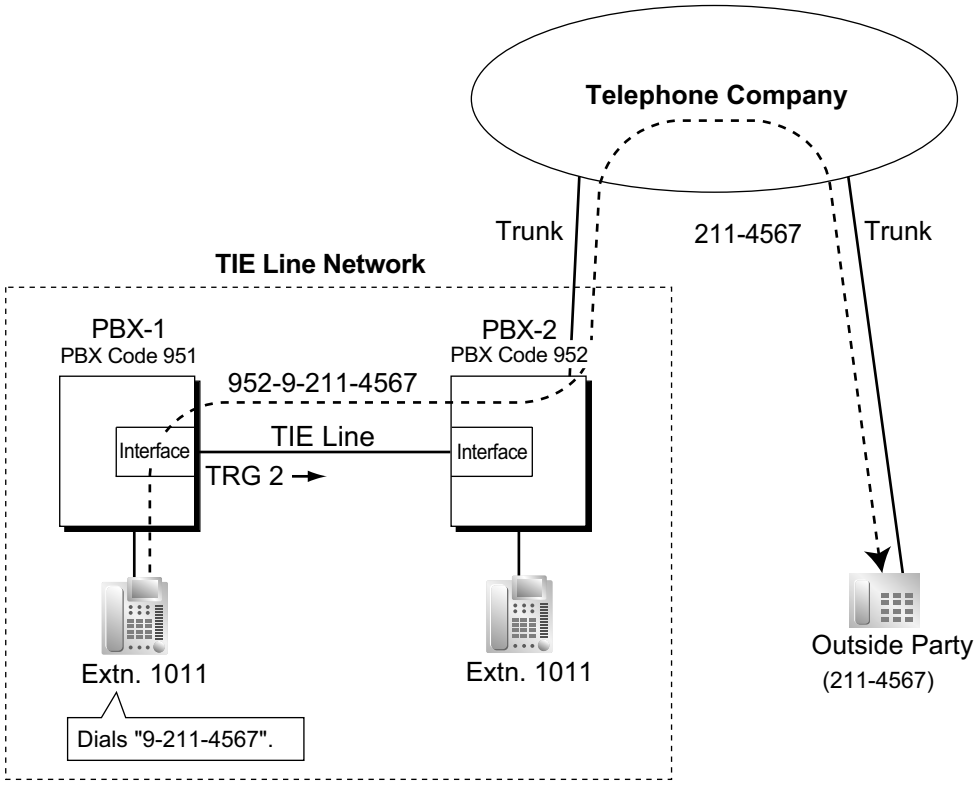
[Example]
<Extension Number Method (Access without PBX Code) using ARS>



Explanation:

1. Extension 1011 of PBX-1 dials the Idle Line Access number of PBX-1 "9" and telephone number "211-4567".
2. PBX-1 modifies the call (adds the Idle Line Access number of PBX-2 "9") and sends the call to PBX-2 through the TIE line (trunk group [TRG] 2) according to the ARS programming of PBX-1.
3. PBX-2 sends the call to the outside party "211-4567".

<PBX Code Method (Access with PBX Code) using ARS>



Explanation:

1. Extension 1011 of PBX-1 dials the Idle Line Access number of PBX-1 "9" and telephone number "211-4567".
2. PBX-1 modifies the call (adds "952" and the Idle Line Access number of PBX-2 "9") and sends the call to PBX-2 which has PBX code "952" through the TIE line (trunk group [TRG] 2) according to the ARS programming of PBX-1.
3. PBX-2 sends the call to the outside party "211-4567".

b. Blocking trunk calls made through another PBX and how to override it:

Whether an incoming TIE line call can make a trunk call through this PBX (i.e., PBX-2), depends on the COS that is assigned to the trunk group of this PBX, that the incoming TIE line is connected to. If the COS of the trunk group is unable to make outgoing calls by the Toll Restriction/Barring feature or External Call Block feature, trunk calls made through this PBX will be prohibited.

To override this prohibition, an extension of PBX-1 must enter a verification code assigned to PBX-2 to change the COS temporarily. It is also possible to override the prohibition by specifying an extension at PBX-2 with the Walking COS feature, to temporarily switch to that extension's COS.

→ 10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—TRS—◆ TRS Level—Day, Lunch, Break, Night

→ 10.7.2 PBX Configuration—[2-7-2] System—Class of Service—External Call Block

→ 11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Main—◆ COS

→ 14.3 PBX Configuration—[6-3] Feature—Verification Code

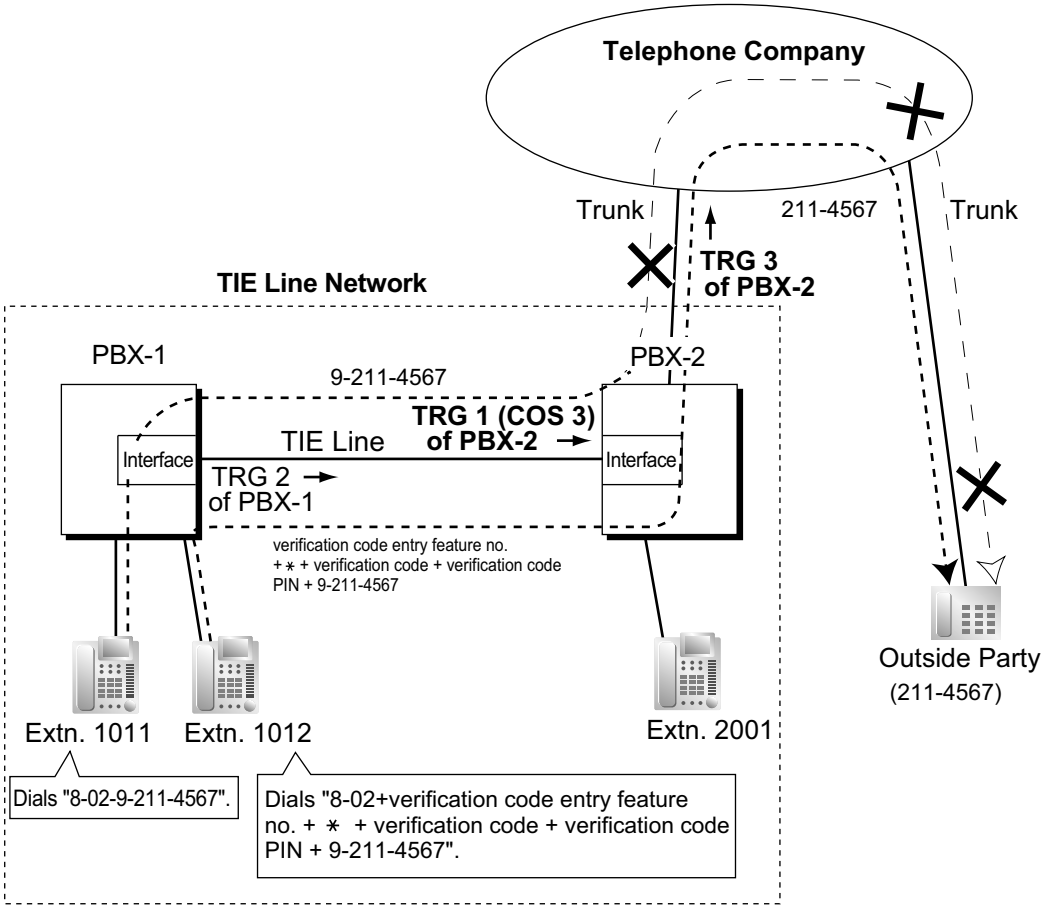
[Programming Example of PBX-2]

Trunk Group No.	COS No.
1	3
2	2
3	2
:	:

TRG of Incoming Call	Outgoing Call			
	TRG 1	TRG 2	TRG 3	…
COS 1		✔	✔	
COS 2	✔	✔		✔
COS 3		✔	✔	✔
:	:	:	:	:

✔ : Block

[Example]
<Extension Number Method (Access Without PBX Code)>



Explanation:

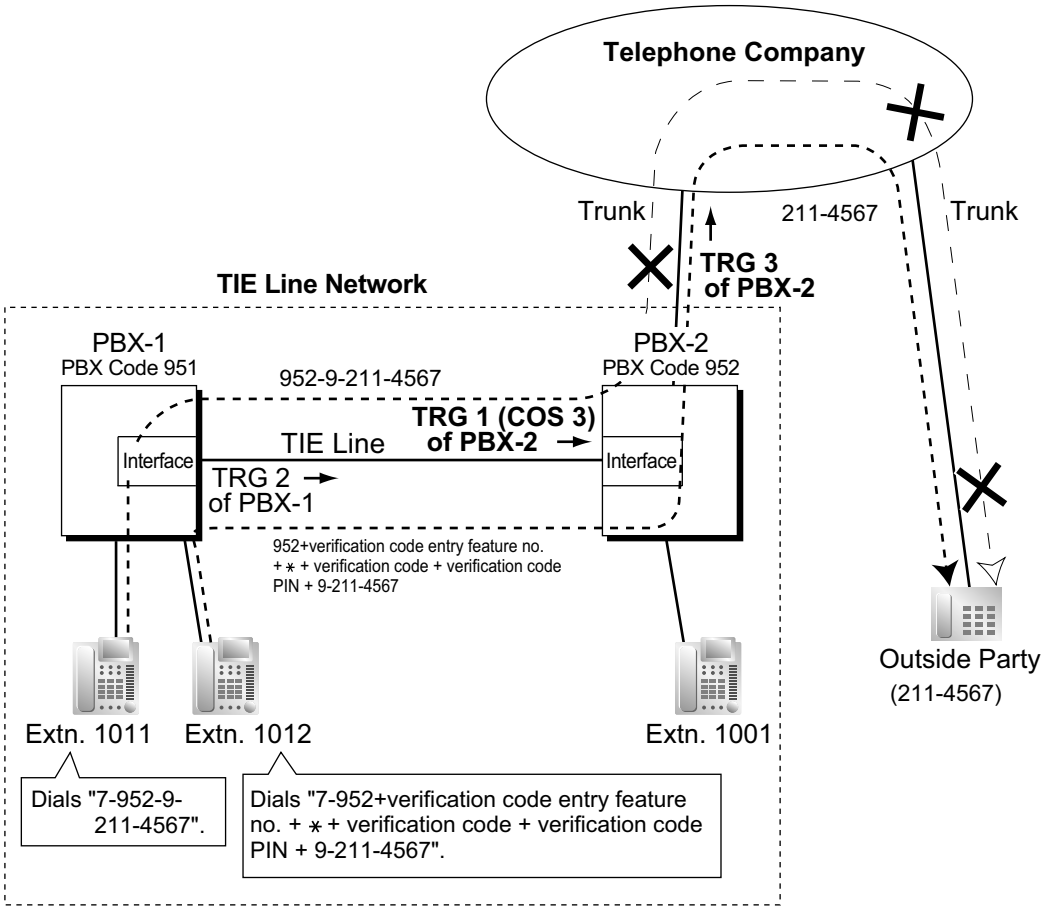
Case 1:

1. Extension 1011 of PBX-1 dials the Trunk Group Access number of PBX-1 "8", TIE line trunk group number (TRG 2), Idle Line Access number of PBX-2 "9", and the telephone number "211-4567".
2. The call is not connected to the outside party through PBX-2 because the COS of TRG 1 (COS 3) is blocked from accessing TRG 3 of PBX-2.

Case 2:

1. Extension 1012 of PBX-1 dials the Trunk Group Access number of PBX-1 "8", TIE line trunk group (TRG2), verification code entry feature number, ✳, verification code, verification code personal identification number (PIN), Idle Line Access number of PBX-2 "9", and the telephone number "211-4567".
2. If the specified verification code applies COS 2 of PBX-2, the call is connected to the outside party through PBX-2, because COS 2 is not blocked from accessing TRG 3 of PBX-2.

<PBX Code Method (Access with PBX Code)>



Explanation:

Case 1:

1. Extension 1011 of PBX-1 dials the TIE line access number "7", PBX code "952", Idle Line Access number of PBX-2 "9", and the telephone number "211-4567".
2. The call is not connected to the outside party through PBX-2 because the COS of TRG 1 (COS 3) is blocked from accessing TRG 3 of PBX-2.

Case 2:

1. Extension 1012 of PBX-1 dials the TIE line access number "7", PBX code "952", verification code entry feature number, ✳, verification code, verification code personal identification number (PIN), Idle Line Access number of PBX-2 "9", and the telephone number "211-4567".

2. If the specified verification code applies COS 2 of PBX-2, the call is connected to the outside party through PBX-2, because COS 2 is not blocked from accessing TRG 3 of PBX-2.

c. Override using an Itemised Billing Code for ARS

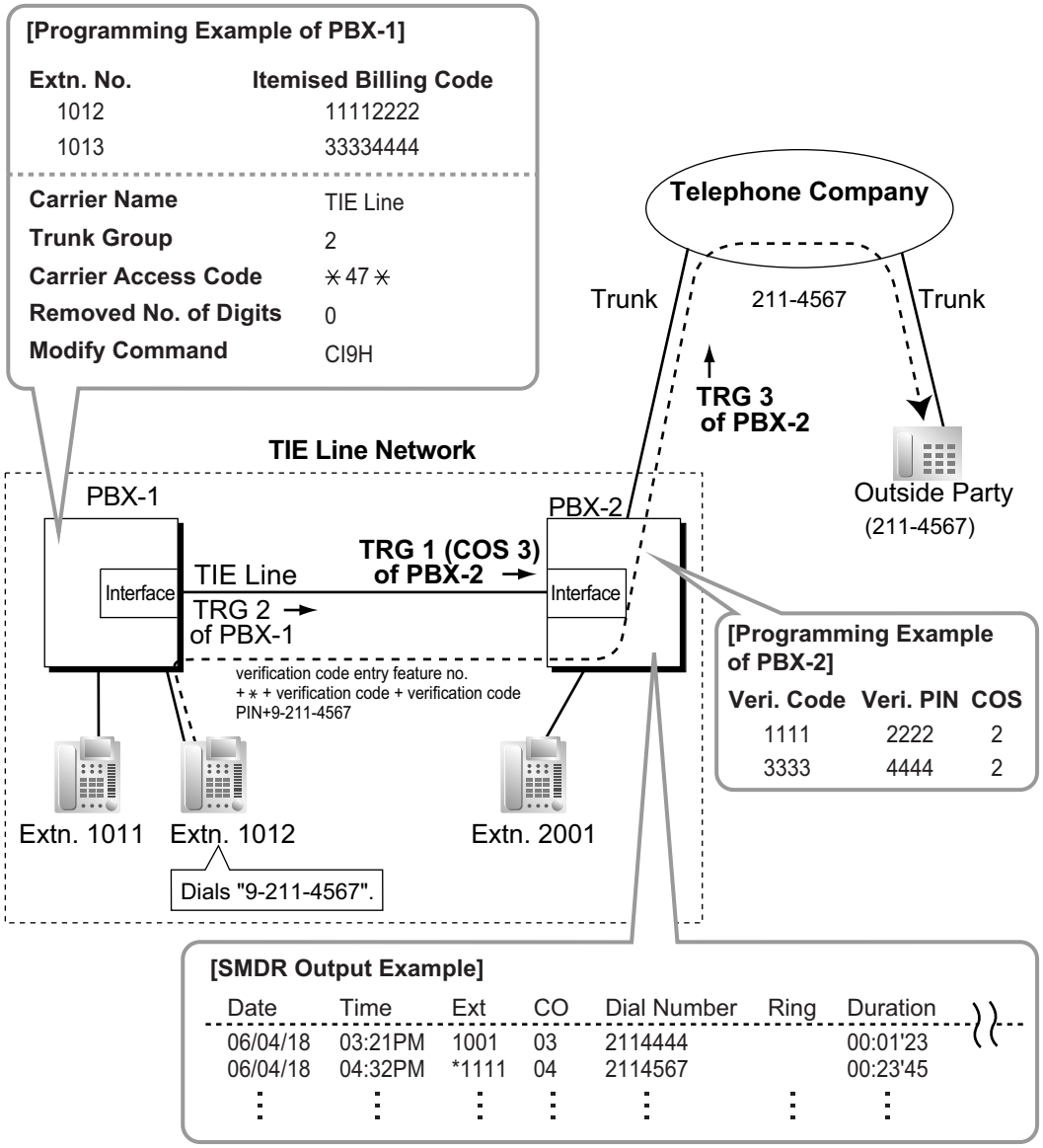
By assigning an Itemised Billing Code for ARS to PBX-1, an extension's verification code can be sent to PBX-2 automatically, without the extension having to dial the verification code.

→ 12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1—◆ ARS Itemised Code

→ 12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 1—◆ ARS Itemised Code

→ 16.5 PBX Configuration—[8-5] ARS—Carrier

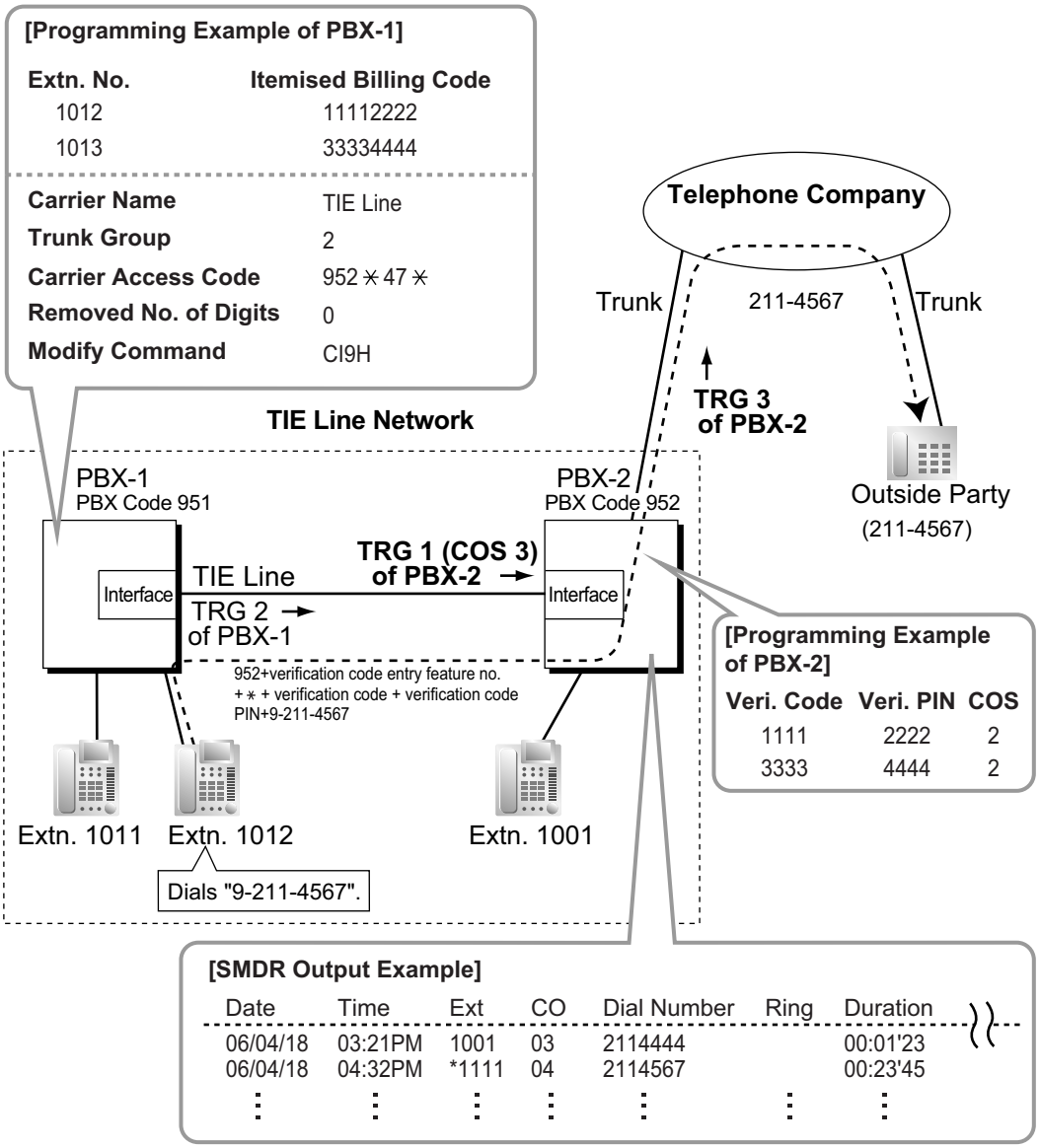
[Example]
<Extension Number Method (Access without PBX Code)>



Explanation:

1. Extension 1012 of PBX-1 dials the Idle Line Access number of PBX-1 "9", and the telephone number "211-4567".
2. PBX-1 modifies the call (adds the verification code entry feature number, verification code and verification code PIN, and the Idle Line Access number of PBX-2 "9") and sends the call to PBX-2 through the TIE line (trunk group [TRG] 2) according to the ARS programming of PBX-1.

<PBX Code Method (Access with PBX Code)>

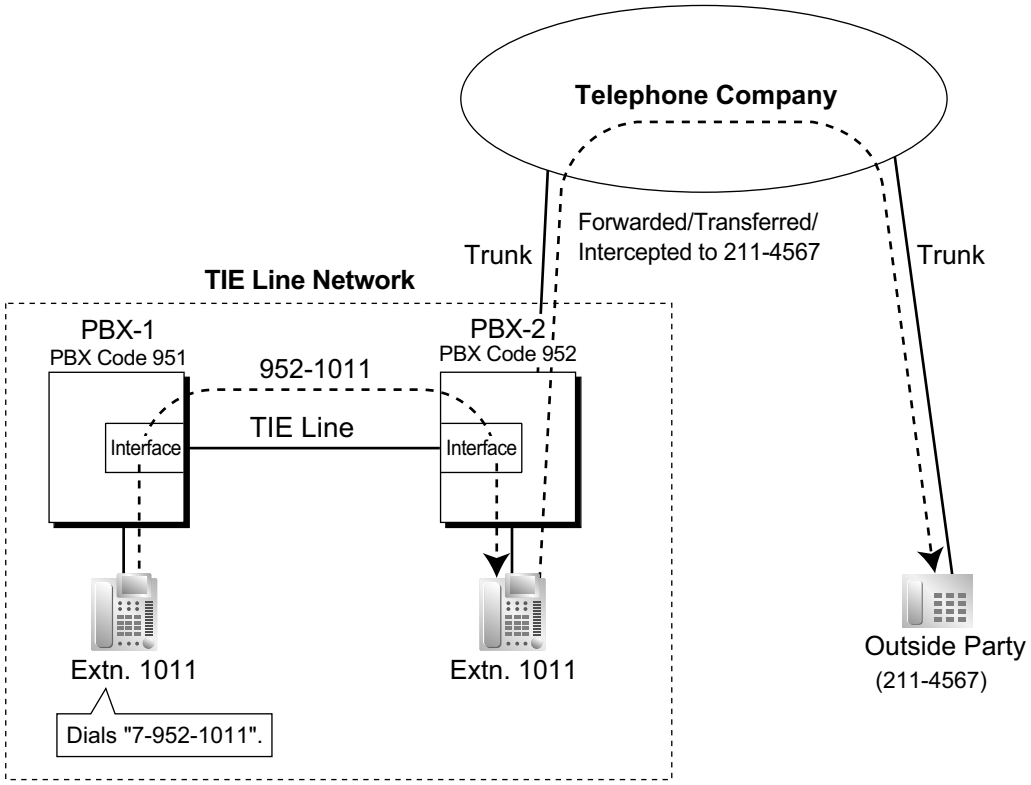


Explanation:

1. Extension 1012 of PBX-1 dials the Idle Line Access number of PBX-1 "9", and telephone number "211-4567".
2. PBX-1 modifies the call (adds "952", the verification code entry feature number, verification code and verification code PIN, and the Idle Line Access number of PBX-2 "9") and sends the call to PBX-2 which has PBX code "952" through the TIE line (trunk group [TRG] 2) according to the ARS programming of PBX-1.

d. FWD/Call Transfer/Intercept Routing to the Trunk

[Example]



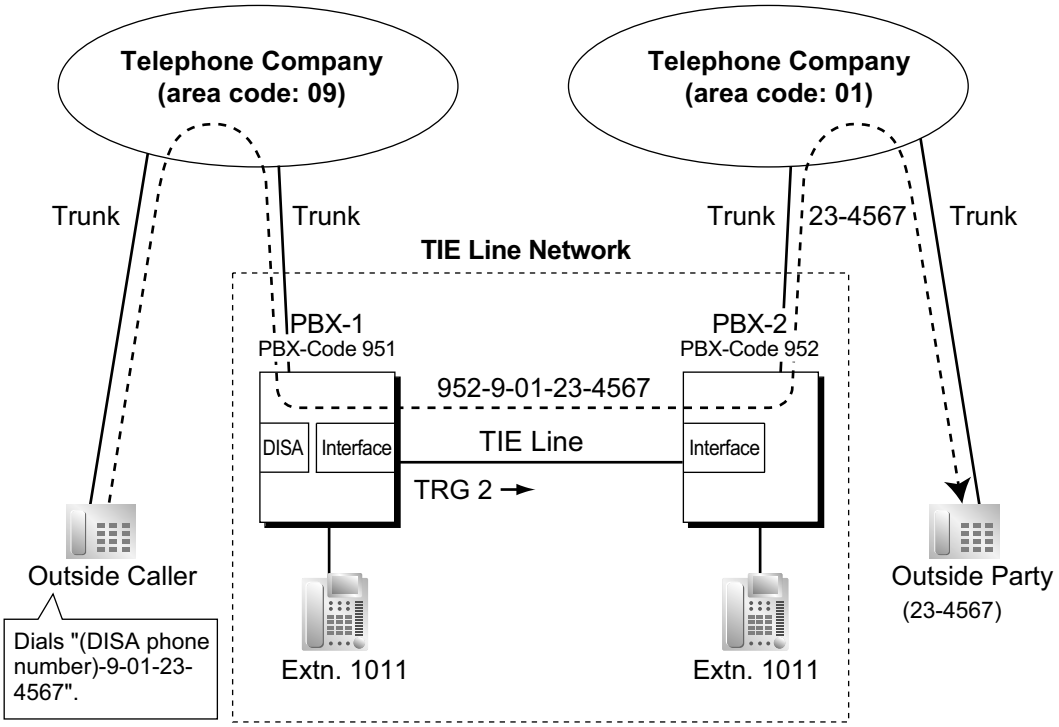
Explanation:

1. Extension 1011 of PBX-1 dials the TIE line access number "7", PBX code "952", and extension number "1011".
2. The call reaches the destination (extension 1011 of PBX-2) through the TIE line, and the call is forwarded, transferred or intercepted to the outside party "211-4567" through the trunk.

Trunk-to-TIE-to-Trunk Access

An outside caller can be connected to an outside party through the TIE line by using the DISA feature.

[Example]



Explanation:

1. The outside caller dials the "DISA phone number of PBX-1", Idle Line Access number of PBX-1 "9", and telephone number "01-23-4567".
2. PBX-1 modifies the call (adds "952" and the Idle Line Access number of PBX-2 "9") and sends the call to PBX-2 which has PBX code "952" through the TIE line (trunk group [TRG] 2) according to the ARS programming of PBX-1.
3. PBX-2 sends the modified call to the outside party "23-4567" according to its ARS programming.

4.2.1.3 TIE Line Programming

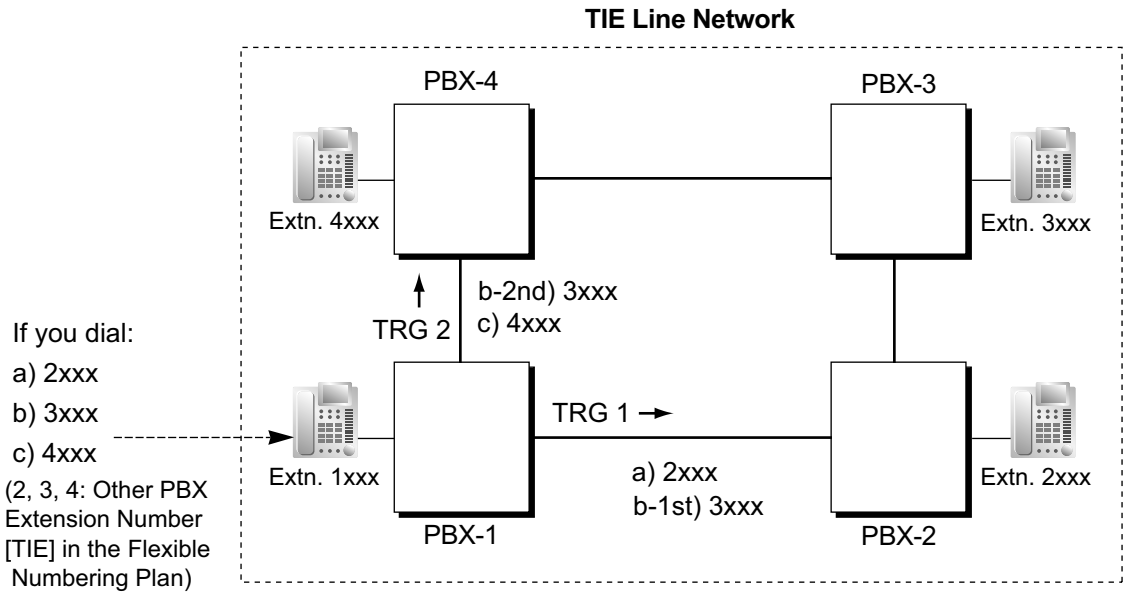
Description

To Make a TIE Line Call

The TIE Line Routing and Modification Table is referenced by the PBX to identify the trunk route when an extension user makes a TIE line call.
It is necessary to make unified tables with all PBXs in the TIE line network.
The routing pattern appropriate for each call is decided by the dialled number.
There are two system programmes for the tables:
TIE Line Routing Table: used to assign the leading numbers (PBX code or extension number) and trunk group hunt sequence.
→ 17.1 PBX Configuration—[9-1] Private Network—TIE Table—◆ Leading Number
TIE Modify Removed Number of Digits/Added Number: used to remove digits from and add a number to the dialled number of the TIE line call. This modification may be needed depending on the TIE line network configuration.
17.1 PBX Configuration—[9-1] Private Network—TIE Table
→ ◆ Removed Number of Digits
→ ◆ Added Number
→ ◆ Trunk Group
[Programming Examples]
Your PBX is PBX-1 and there are four PBXs in your TIE line network. To identify the trunk route as illustrated, you should make the following tables.

a. Extension Number Method (Access without PBX Code)

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Other PBX Extension—◆ Dial



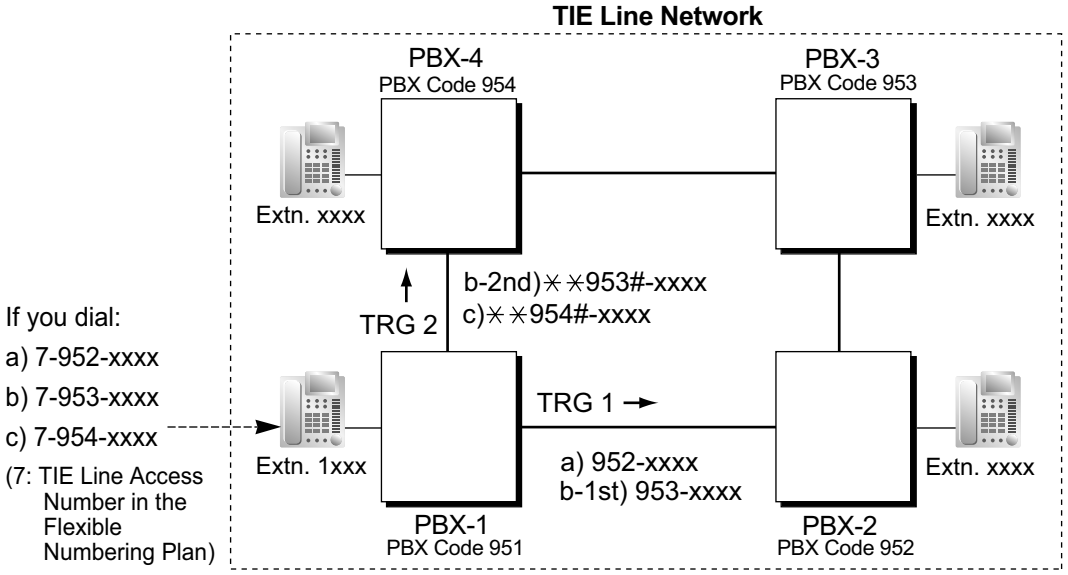
[TIE Line Routing and Modification Table of PBX-1]

Location No.	Leading No.	Priority 1			Priority 2			. .
		TRG	Dial Modification		TRG	Dial Modification		. .
			Removed No. of Digits	Added No.		Removed No. of Digits	Added No.	. .
01	2	1	0					. .
02	3	1	0		2	0		. .
03	4	2	0					. .
:	:	:	:	:	:	:	:	:

Explanation:
Location 01:
The hunt sequence by dialling [2XXX]:
The 1st route—trunk group (TRG) 1 ➡ Sending no. to PBX-2: 2XXX
Location 02:
The hunt sequence by dialling [3XXX]:
The 1st route—trunk group (TRG) 1 ➡ Sending no. to PBX-2: 3XXX
The 2nd route—trunk group (TRG) 2 ➡ Sending no. to PBX-4: 3XXX
Location 03:
The hunt sequence by dialling [4XXX]:
The 1st route—trunk group (TRG) 2 ➡ Sending no. to PBX-4: 4XXX

b. PBX Code Method (Access with PBX Code)

→ 10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ TIE Line Access
→ 17.1 PBX Configuration—[9-1] Private Network—TIE Table—◆ Own PBX Code



[TIE Line Routing and Modification Table of PBX-1]

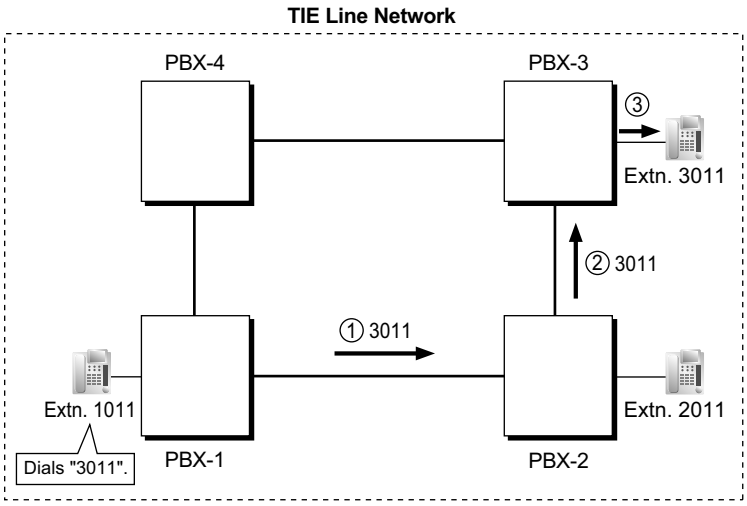
Location No.	Leading No.	Priority 1			Priority 2			. .
		TRG	Dial Modification		TRG	Dial Modification		. .
			Removed No. of Digits	Added No.		Removed No. of Digits	Added No.	. .
01	952	1	0					. .
02	953	1	0		2	3	✳✳953#	. .
03	954	2	3	✳✳954#				. .
:	:	:	:	:	:	:	:	:

Explanation:
Location 01:
The hunt sequence by dialling [7+PBX Code 952+XXXX]:
The 1st route — trunk group (TRG) 1
➡ Sending no. to PBX-2: 952–XXXX
Location 02:
The hunt sequence by dialling [7+PBX Code 953+XXXX]:
The 1st route — trunk group (TRG) 1
➡ Sending no. to PBX-2: 953–XXXX
The 2nd route — trunk group (TRG) 2
➡ Sending no. to PBX-4: ✳✳953#–XXXX
Location 03:
The hunt sequence by dialling [7+PBX Code 954+XXXX]:
The 1st route — trunk group (TRG) 2
➡ Sending no. to PBX-4: ✳✳954#–XXXX

To Receive a TIE Line Call

a. Extension Number Method (Access without PBX Code)

[Example]



① A TIE line call is sent to PBX-2 from PBX-1. If the number sent from PBX-1 is an extension number of PBX-2 (e.g., 2011), the call will be received at extension "2011". If not, PBX-2 checks the number in the TIE Line Routing and Modified Table of PBX-2.

② If the match is found in the table, the call will be modified according to the table and send to the corresponding PBX (PBX-3).

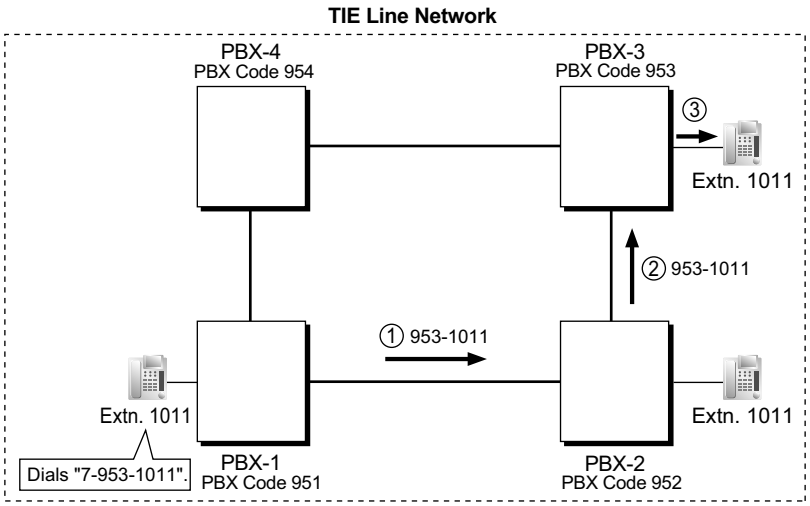
③ The number sent from PBX-2 "3011" is an extension number of PBX-3. The call is received at extension "3011".

Note

When a TIE line call is sent from one PBX to another, the receiving PBX first modifies the received number according to the assignment for the trunk port: the number of digits removed, and the number added, are determined by this assignment. Then the PBX checks whether the completed number is an existing extension number at that PBX.

b. PBX Code Method (Access with PBX Code)

[Example]



① A TIE line call is sent to PBX-2 from PBX-1. If the number sent from PBX-1 has the PBX code of PBX-2 "952", the call will be received at the corresponding extension of PBX-2 (e.g., 1011 of PBX-2). If not, PBX-2 checks the number in the TIE Line Routing and Modified Table of PBX-2.

② If the match is found in the table, the call will be modified according to the table and send to the corresponding PBX (PBX-3).

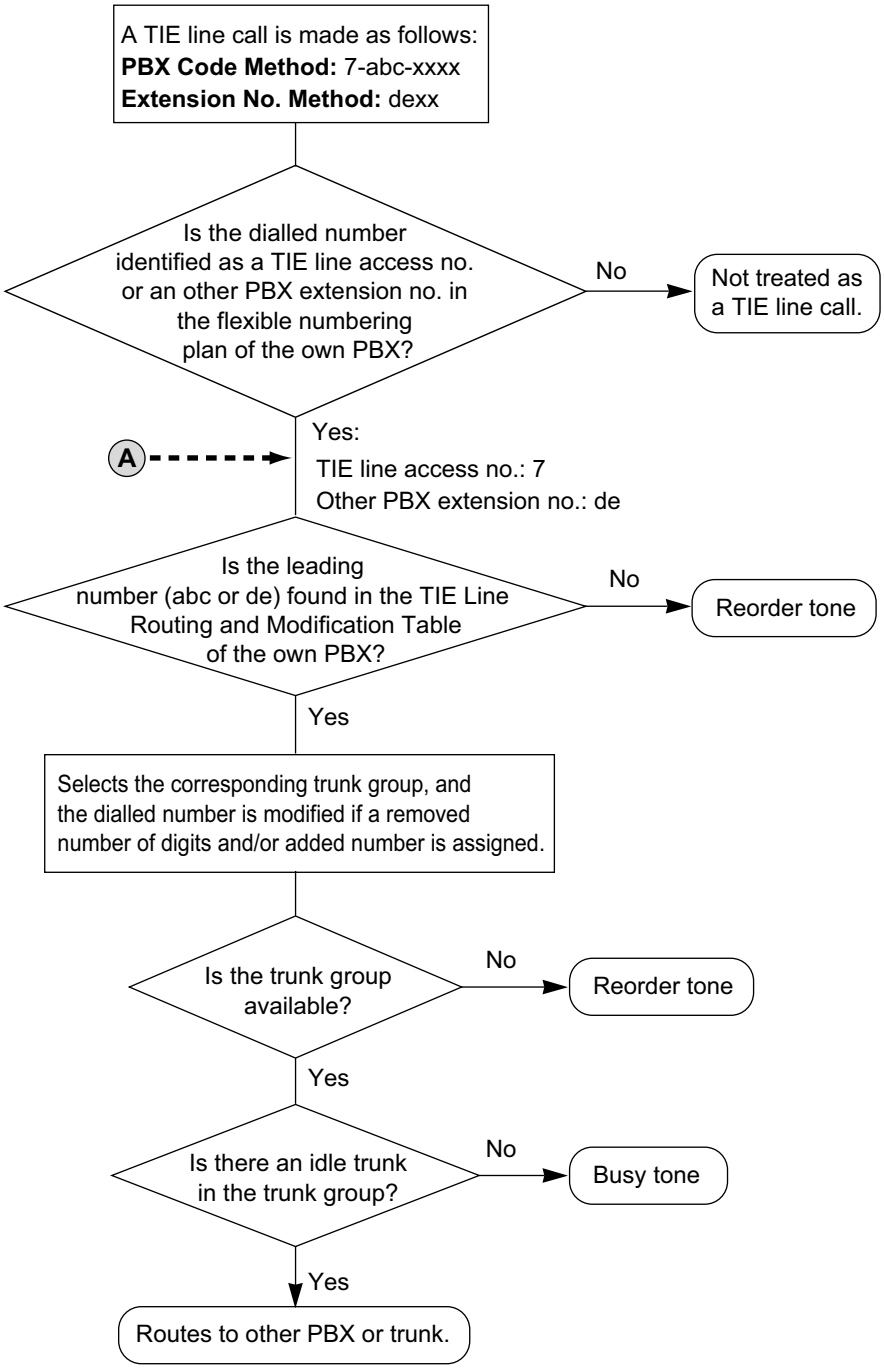
③ The number sent from PBX-2 "953-1011" has the PBX code of PBX-3 "953". The call is received at extension "1011" of PBX-3.

Note

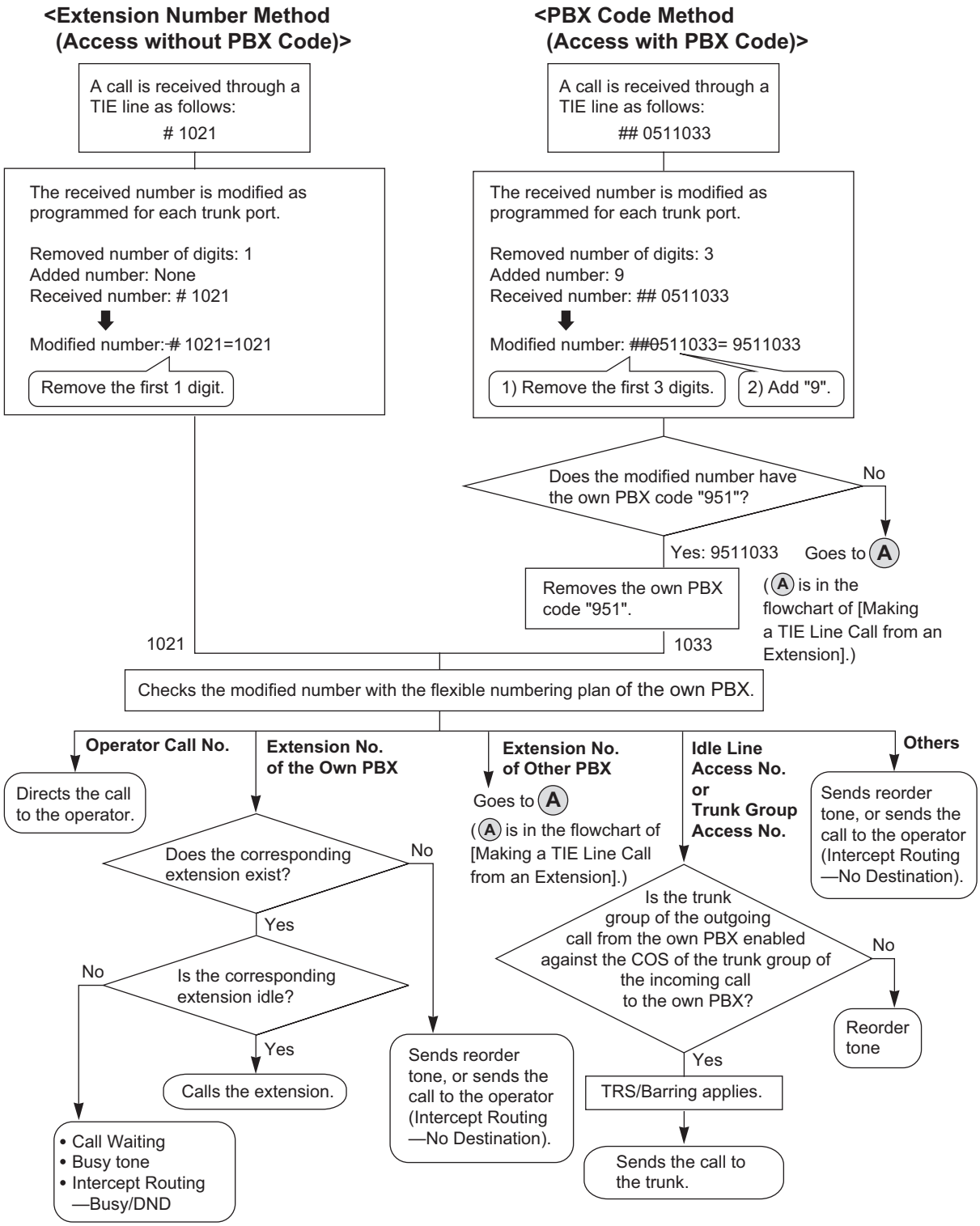
When a TIE line call is sent to a PBX from another PBX, first the PBX modifies the number sent to the PBX according to the assignment for each trunk port of the PBX: the removed number of digits from and/or added number to the number sent to the PBX is determined by the assignment. Then the PBX starts to check the number whether the number has the PBX code of the PBX.

TIE Line Routing Flowchart

[Making a TIE Line Call from an Extension]



[Receiving a Call through a TIE Line]

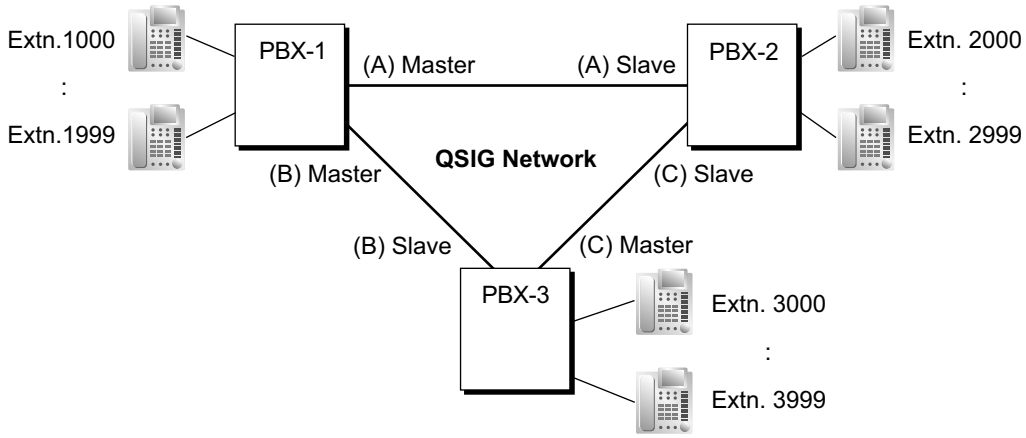


Conditions

- A trunk which is used for a private network should be assigned "**Private**" as the networking type. (→ 2.1.1.1 Incoming Trunk Call Features—SUMMARY)

- To establish a QSIG network (→ 4.2.4 QSIG Standard Features), each ISDN (QSIG) connection in a TIE line network must have the port on one PBX assigned as a master port, and the port on the other PBX assigned as a slave port. PBXs that support this feature are KX-NS series, and KX-NCP series, KX-TDE series, and KX-TDA series PBXs.

  [TIE Line Network Connection Example]



- When a TIE line call arrives at a busy extension which has disabled Call Waiting, the caller will hear a busy tone. If required, Intercept Routing—Busy/DND can be activated.
- The Inter-digit time can be assigned for TIE line calls.
  → 10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Incoming Call Inter-digit Timer—TIE (s)

PC Programming Manual References

9.2 PBX Configuration—[1-1] Configuration—Slot—System Property
→V-IPGW–GW Settings–Main
→V-IPGW–DN2IP
9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Network Numbering Plan—◆ Trunk Property
10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ Incoming Call Inter-digit Timer—TIE (s)
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ Idle Line Access (Local Access)
→◆ Trunk Group Access
→◆ TIE Line Access
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Other PBX Extension
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—TRS—◆ TRS Level—Day, Lunch, Break, Night
10.9 PBX Configuration—[2-9] System—System Options—Option 4—◆ Private Network—TIE Call by Extension Numbering
11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings—Main—◆ COS
17.1 PBX Configuration—[9-1] Private Network—TIE Table
18.2 PBX Configuration—[10-2] CO & Incoming Call—DIL Table & Port Settings
→DIL—◆ Trunk Property
→DIL—◆ DIL Destination—Day, Lunch, Break, Night
→DDI/DID/TIE
18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table—◆ DDI / DID Destination—Day, Lunch, Break, Night

18.4 PBX Configuration—[10-5] CO & Incoming Call—Miscellaneous—◆ Intercept—Routing to Operator - No Destination (Destination is not programmed.)

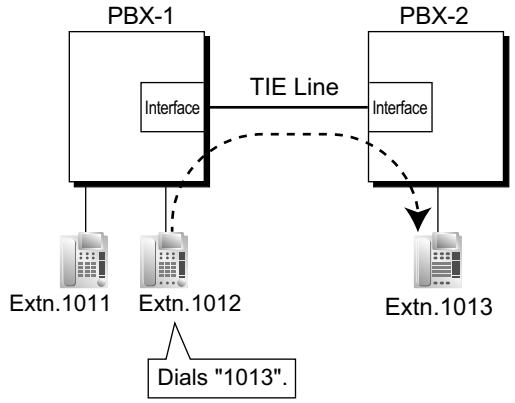
PT Programming Manual References

[453] DID Destination
[500] Trunk Group Number

4.2.1.4 Common Extension Numbering for 2 PBXs

Description

Two PBXs can have extensions with the same leading number. TIE Line calling is available using extension numbering.



Explanation:
If a dialled number is not found at the local PBX, the call can be sent to the remote PBX.
When an extension number is dialled, the PBX first searches local extensions for a matching number. If there is no match, the PBX then checks the TIE Line Routing Table for a corresponding entry. If an entry is found, the call is sent to the connected PBX.
Case 1:
Extension 1012 of PBX-1 dials extension number "1011".
→ The dialled number is found at the local PBX, so extension 1012 of PBX-1 is connected to extension 1011 of PBX-1.
Case 2:
Extension 1012 of PBX-1 dials extension number "1013".
→ The dialled number is not found at the local PBX, so the call is redirected to the specified TIE Line, and extension 1012 of PBX-1 is connected to extension 1013 of PBX-2.

Conditions

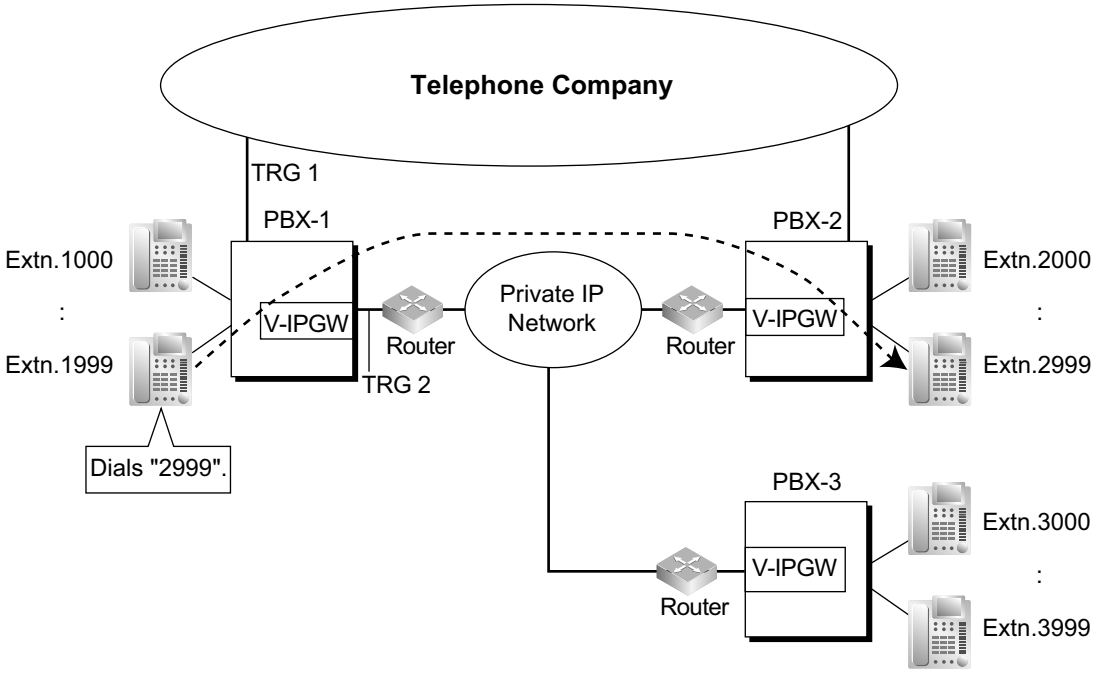
- KX-NSN002 (Activation Key for QSIG Network) is required to use this feature.
- System programming is required to enable this feature.

4.2.2 Voice over Internet Protocol (VoIP) Network

Description

When a PBX is connected to another PBX via a private IP network, voice signals are converted into IP packets and sent over the network. This is known as Voice over IP (VoIP). This PBX uses the H.323 standard for VoIP communication.
VoIP networks support private network communications using TIE line service.

[Example]



Required Programming

- **PBX**
  For making a call:
  ARS programming (→ 2.8.1 Automatic Route Selection (ARS)) or TIE line service programming
  For receiving a call:
  TIE line service programming

  [TIE Line Routing and Modification Table]

Location No.	Leading No.	Priority 1			Priority 2			. .
		TRG	Dial Modification		TRG	Dial Modification		. .
			Removed No. of Digits	Added No.		Removed No. of Digits	Added No.	. .
01	2	2 (VoIP port)	0					. .
02	3	2	0					. .
:	:	:	:	:	:	:	:	:

  ### [Explanation]

  Calls to destinations with leading number "2" or "3" are automatically routed through the VoIP ports, designated as trunk group 2.

- **IP Gateway**
  IP address assignment for the local PBX and other PBXs.
  [Programming Example]

Destination	
Leading No.	IP Address
2	200.45.11.35
3	199.176.64.1
:	:

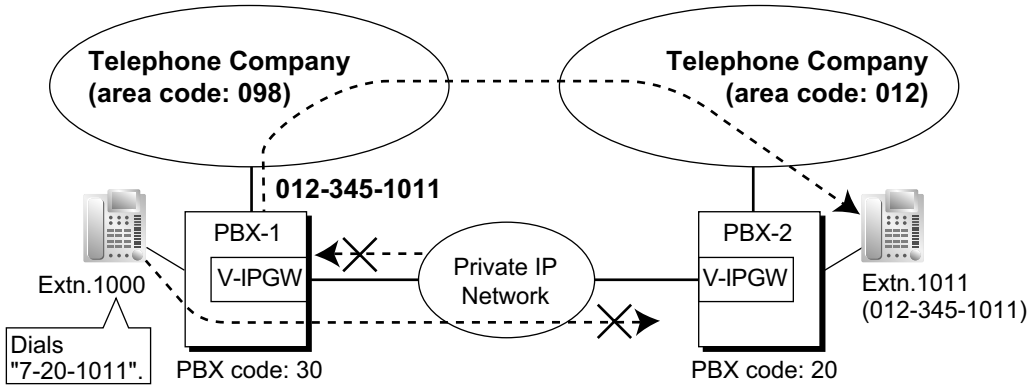
  ### [Explanation]

  Calls are routed to the IP address of each V-IPGW card based on the leading number dialled.

Automatic Rerouting of VoIP Calls to Public Trunks

When a VoIP call cannot be completed successfully, the PBX can automatically attempt to make the call using a public trunk instead. This provides a backup method of making calls in cases when IP network transmission cannot be completed successfully.

[Example]



The leading numbers of extensions accessed through the VoIP network are added as entries to the Quick Dialling table, in addition to being registered as Other PBX Extension Numbers, as shown below:

[Programming Example]

Number to dial to call an extension at another PBX using VoIP network:
7 (TIE line access number) + 20 (PBX Code) + 1011 (extension number)
Number to dial to call that extension using a public trunk:
9 (trunk access number) + 012-345-1011

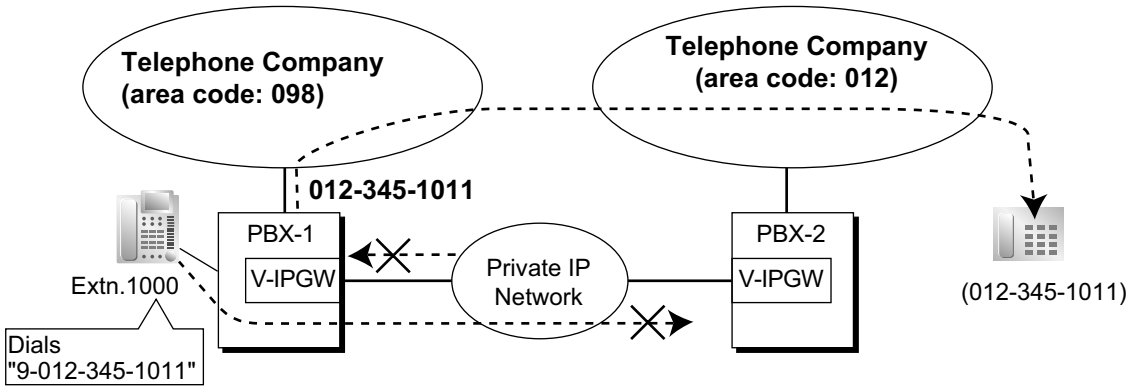
[Quick Dialling Table]

Quick Dialling No.	Destination No.
720	9012345

If the call cannot be completed using the VoIP network, and the dialled leading number is found in the Quick Dialling Table, the call will be automatically rerouted to a trunk as specified by the corresponding destination number.
When a call is made using the VoIP network, if the PBX does not receive a reply from the other PBX within about 4 seconds of making the call, or an error is returned, the call is rerouted to a public trunk as specified.

Automatic Rerouting of VoIP Calls to Public Trunks using ARS

When dialling an outside party using ARS, the call can be rerouted to a public trunk if the call cannot be completed successfully.



Explanation:

1. An extension of PBX-1 dials the Idle Line Access number of PBX-1 "9" and telephone number "012-345-1011".
2. PBX-1 modifies and routes the call to PBX-2 through a private IP network according to the ARS programming of PBX-1.

3. The IP network transmission cannot be completed successfully and the call is rerouted via a public trunk to the outside party "012-345-1011".

[Quick Dialling Table]

The leading number (in this case, "9") of the dialled number is found in the Quick Dialling Table, and the call is automatically rerouted to the specified trunk group. It is necessary to specify a trunk group to make this type of call. If the idle line access number is used in a destination number, the call will be rerouted through the same private IP network according to the ARS programming, and the call will not be completed.

Quick Dialling No.	Destination No.
9	802

Sending Faxes Through the Network

Through system programming, it is possible to select whether faxes are sent using analogue or T.38 protocol signals.

Conditions

- Some QSIG services are available. (→ 4.2.4 QSIG Standard Features)
- TRS/Call Barring settings apply to calls rerouted to public trunks. When making a call using System Speed Dialling, regular TRS/Call Barring settings are applied, even if the TRS/Barring Override by System Speed Dialling feature is enabled. (→ 2.7.1 Toll Restriction (TRS)/Call Barring (Barring))

Installation Manual References

4.4 Virtual Cards

PC Programming Manual References

9.11.2 PBX Configuration—[1-1] Configuration—Slot—V-IPGW—Shelf Property—Hunt Pattern
9.12 PBX Configuration—[1-1] Configuration—Slot—V-IPGW—Port Property
17.1 PBX Configuration—[9-1] Private Network—TIE Table

Feature Guide References

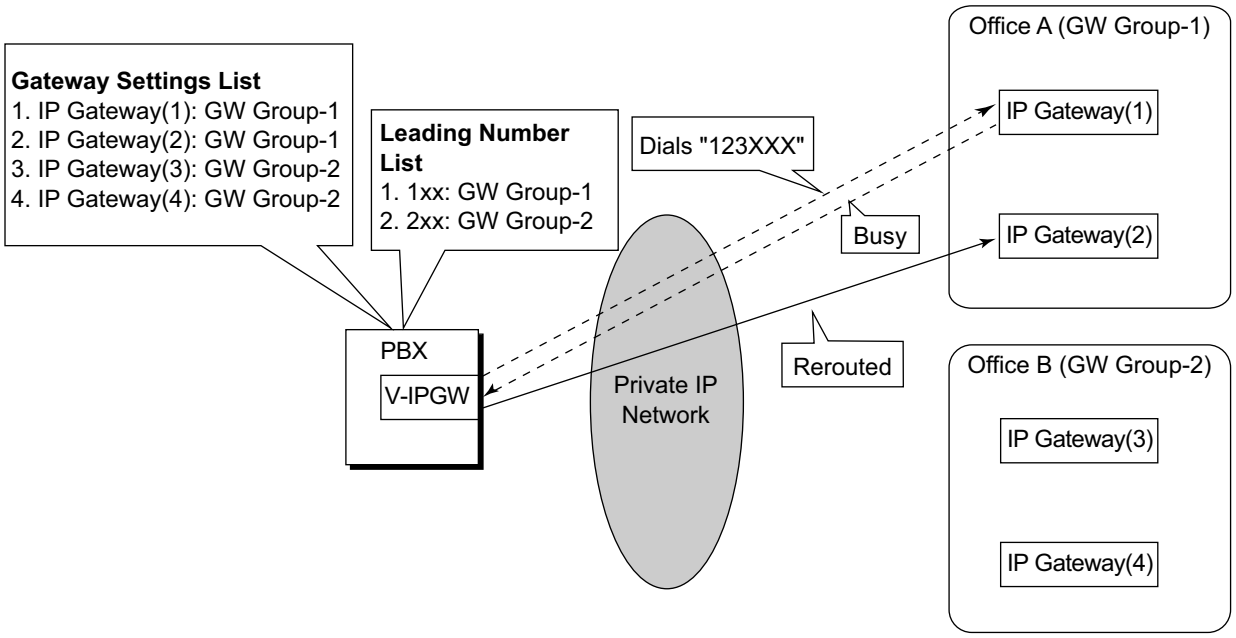
2.8.1 Automatic Route Selection (ARS)
4.2.1 TIE Line Service

4.2.2.1 Gateway Groups

Description

It is possible to automatically reroute outgoing VoIP calls according to preassigned gateway groups.
A gateway device can be assigned a gateway group number based on its IP Address. When a call is made using a leading number allocated to a gateway group, it is routed to the lowest-numbered available device within that group. In other words, if the lowest-numbered device is busy or not available, the call slides to the next available device.

[Example]

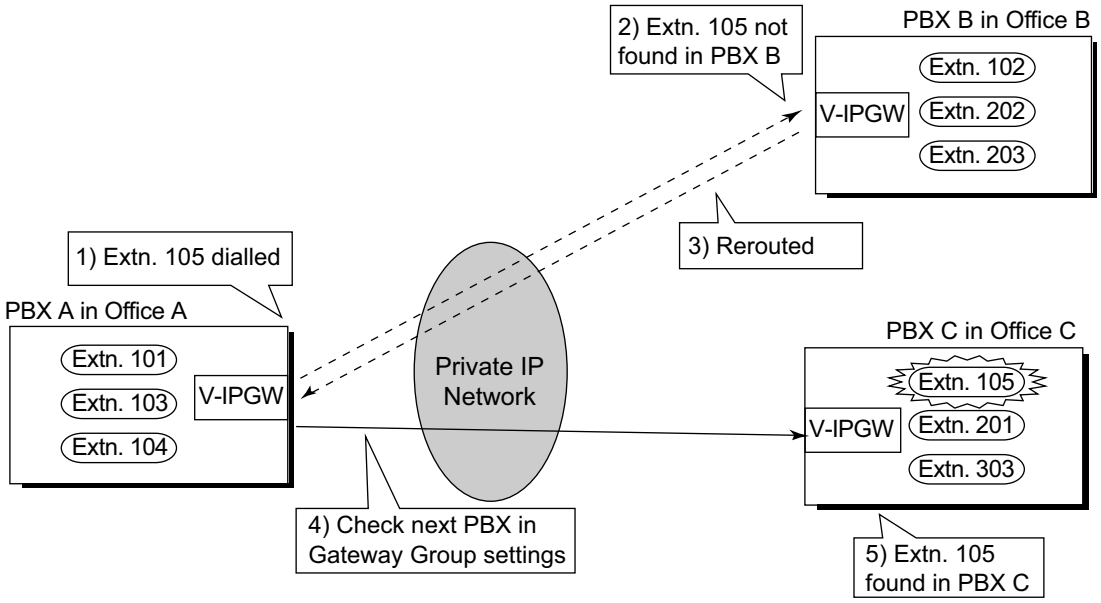


When "123XXX" is dialled, the call is routed to GW Group-1. However the lowest-numbered device (IP Gateway(1)) is busy or not available, so the call is rerouted to IP Gateway(2).

4.2.2.2 Common Extension Numbering for Multiple PBXs

Description

Multiple PBXs in separate locations, connected in an IP network, can share a common block of extensions designated in a gateway group.



Explanation:
In the same way as when connected by a TIE Line, if a dialled number is not found at the local PBX, the call can be sent to other PBXs connected via an IP network. When an extension number is dialled, the PBX first searches local extensions for a matching number. If there is no match, the PBX then checks the TIE Line Routing Table for the Gateway Group for a corresponding entry. If an entry is found, the call is sent to the connected PBX.

Conditions

- System programming is required to enable this feature.
- If the called extension does not exist at the called PBX, the next PBX in the same gateway group is called automatically.
- The Routing to Operator setting in system programming must be disabled to use this feature.
- To use this feature, all PBXs in the IP network must be KX-NS series PBXs, or KX-NCP/KX-TDE series PBXs with MPR Software Version 3.0000 or later.

PC Programming Manual References

18.4 PBX Configuration—[10-5] CO & Incoming Call—Miscellaneous—◆ Intercept—Routing to Operator - No Destination (Destination is not programmed.)

4.2.2.3 Call Distribution Port Group

Description

It is possible to set which virtual port receives each call depending on the telephone number of the called party. By assigning each port to a Call Distribution Port Group (CDPG), it is possible to select which group receives each call. CDPG settings cover all V-IPGW cards in the PBX, which allows for ports on different cards to be assigned to the same CDPG. In other words, two 8-port cards can be used as one 16-port card.

Programming Example:

To enable this feature, it is necessary to programme the following 2 tables through system programming:

- CDPG Table
  - in order to assign ports to Call Distribution Port Groups.
- Hunt Pattern Table
  - in order to programme a priority list of CDPG destinations for each leading number.

[Programming Example of the CDPG Table]

V-IPGW Card Number	Port Number	Call Distribution Port Group*1
1	**1**	**CDPG 1**
1	2	CDPG 2
1	:	:
1	8	CDPG 2
2	**1**	**CDPG 1**
2	:	:
2	8	CDPG 3

*1 → 9.12 PBX Configuration—[1-1] Configuration—Slot—V-IPGW—Port Property—◆ Call Distribution Port Group

[Programming Example of the Hunt Pattern Table]

No.	Leading Number*1	Call Distribution Port Group (1st)*2	Call Distribution Port Group (2nd)*2	…	Call Distribution Port Group (16th)*2
1	**10**	**CDPG 1**	**CDPG 4**	**…**	**-**
2	20	CDPG 1	CDPG 2	…	CDPG 3
:	:	:	:	…	:
32	300	CDPG 8	CDPG 11	…	-

*1 → 9.11.2 PBX Configuration—[1-1] Configuration—Slot—V-IPGW—Shelf Property—Hunt Pattern—Hunt Pattern 1–16—◆ Leading Number

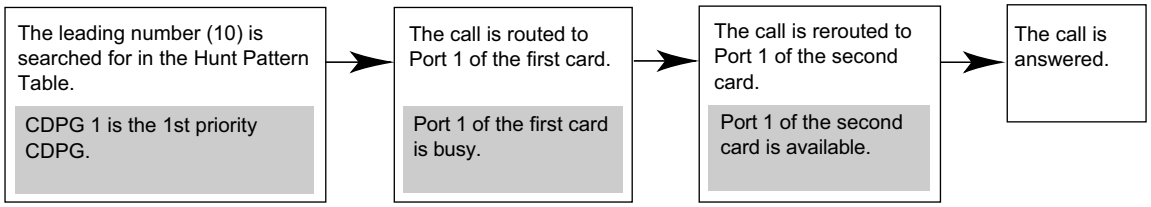
*2 → 9.11.2 PBX Configuration—[1-1] Configuration—Slot—V-IPGW—Shelf Property—Hunt Pattern—Hunt Pattern 1–16—◆ Call Distribution Port Group—1st–16th

Note

The same CDPG can be set for several leading numbers.

Explanation:

As shown in the CDPG table above, CDPG 1 consists of Port 1 of the first card and Port 1 of the second card. When "1023-456-7890" (leading number: 10) is dialled:



If all the ports belonging to CDPG 1 are busy, the call is rerouted to the lowest-numbered available port belonging to CDPG 4, which is set as the second priority for this leading number.

4.2.3 ISDN Virtual Private Network (ISDN-VPN)

Description

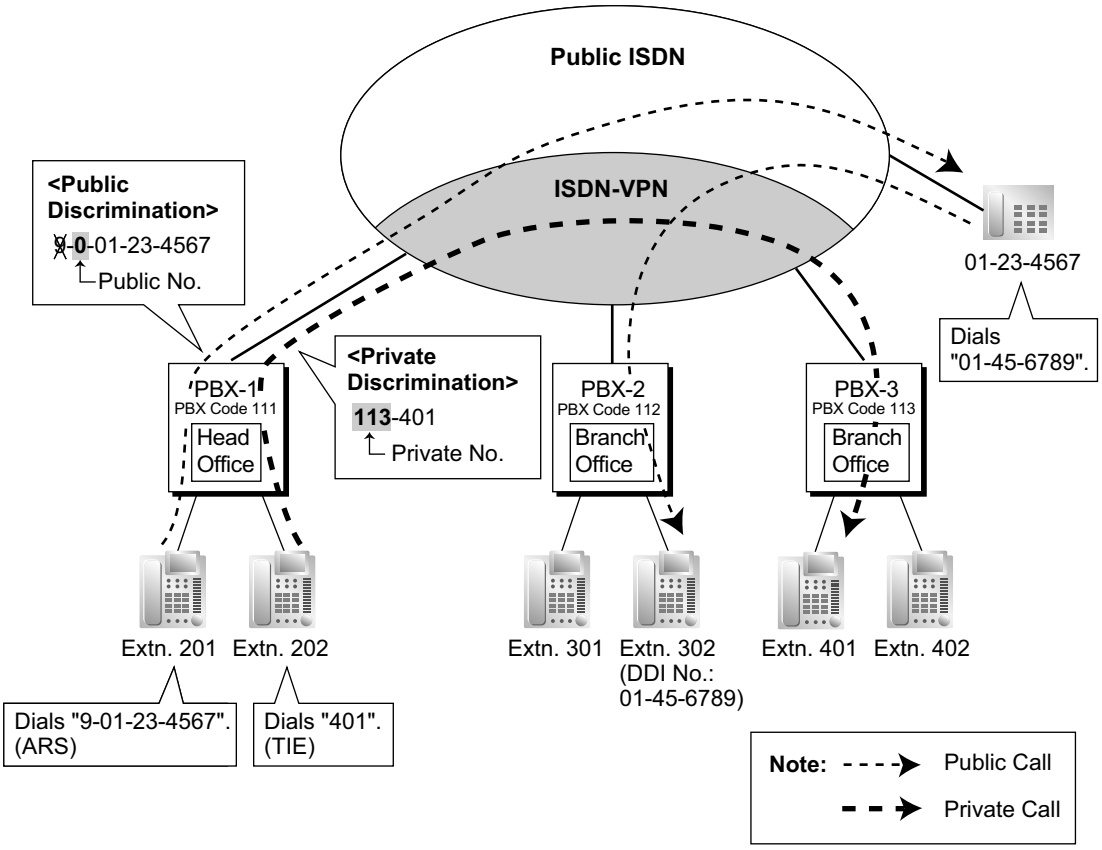
ISDN Virtual Private Network (ISDN-VPN) is a service provided by the telephone company. It uses an existing line as if it were a private line. There is no need to set up a private line or to lease a line from the telephone company. Making and receiving both public and private calls is possible using the same line.

Public/Private Discrimination:

a. **When making a call:** The public/private discrimination number is required before sending the dialled number to the telephone company. The public/private discrimination number can be dialled manually, or automatically by ARS programming (→ 2.8.1 Automatic Route Selection (ARS)) and/or TIE line service programming.

b. **When receiving a call:** The telephone company distinguishes the call type. If it is a private call, the call is received by the TIE line service method. If it is a public call, the call is received by the Incoming Trunk Call Distribution method (DIL/DDI) which is assigned on the trunk.

[Example]



Conditions

- Each PRI port can be set to public or VPN through system programming. To use this service, select VPN.
- Even if the telephone company does not support the ISDN-VPN service, it is possible to use the same kind of service when making a call by TIE line service programming, and/or Quick Dialling programming (→ 2.6.5 Quick Dialling).

[Quick Dialling Programming Example]

Location No.	Quick Dialling No.	Desired No.
Quick Dialling 01	2345 (extension no. of other PBX)	9-123-4321 (Public no. of extension 2345)
:	:	:

Explanation:
When an extension user dials "2345", he is connected to extension "2345" of other PBX whose public number is "123-4321".

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Network Numbering Plan—◆ Trunk Property
16.2 PBX Configuration—[8-2] ARS—Leading Number—◆ Leading Number
17.1 PBX Configuration—[9-1] Private Network—TIE Table
→◆ Leading Number
→◆ Removed Number of Digits
→◆ Added Number

Feature Guide References

4.2.1 TIE Line Service

4.2.4 QSIG Standard Features

4.2.4.1 QSIG Standard Features—SUMMARY

Description

QSIG is a protocol which is based on ISDN (Q.931) and offers enhanced PBX features in a private network. The QSIG network supports private communications by the TIE line service method.
The following features are available for an ISDN-QSIG or VoIP private network. For ISDN, system programming is required to specify whether each feature (excluding Calling Line Identification Presentation [CLIP]) is available for each port of the private network.

[Service Table]

Service	Description & Reference
Calling Line Identification Presentation (CLIP)	Sends the caller's number to the QSIG network when making a call. → 4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG
Connected Line Identification Presentation (COLP)	Sends the number of the answered party to the QSIG network when answering a call. → 4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG
Calling Line Identification Restriction (CLIR)	Prevents the caller's CLI being presented to the called party by the caller. → 4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG
Connected Line Identification Restriction (COLR)	Prevents COLP being sent by the answered party. → 4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG
Calling Name Identification Presentation (CNIP)	Sends the caller's name to the QSIG network when making a call. → 4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG
Connected Name Identification Presentation (CONP)	Sends the name of the answered party to the QSIG network when answering a call. → 4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG

Service	Description & Reference
Calling Name Identification Restriction (CNIR)	Prevents the caller's name being presented to the called party by the caller. → 4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG
Connected Name Identification Restriction (CONR)	Prevents CONP being sent by the answered party. → 4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG
Call Forwarding (CF)—by QSIG	Forwards a call to the QSIG network. → 4.2.4.3 Call Forwarding (CF)—by QSIG
Call Transfer (CT)—by QSIG	Transfers a call to the QSIG network. → 4.2.4.4 Call Transfer (CT)—by QSIG
Completion of Calls to Busy Subscriber (CCBS)—by QSIG	Receives callback ringing when a busy called party on the QSIG network becomes free. → 4.2.4.5 Completion of Calls to Busy Subscriber (CCBS)—by QSIG

Feature Guide References

4.1.2 Integrated Services Digital Network (ISDN) Service Features
4.2.1 TIE Line Service

4.2.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG

Description

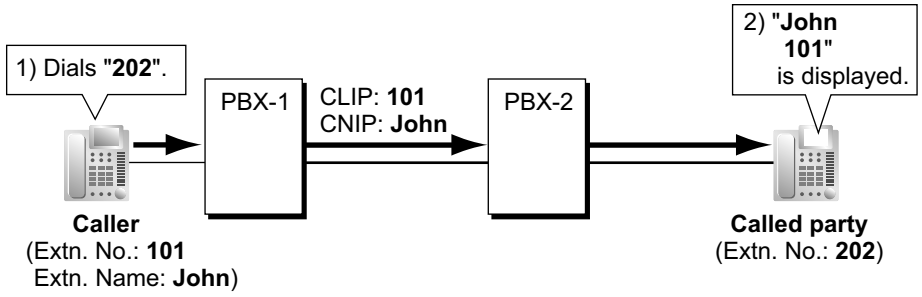
Calling Line/Name Identification Presentation (CLIP/CNIP):

The PBX can send a preprogrammed extension number and/or name to the QSIG network when an extension user makes a call. The called party can see the number and/or name on his telephone display before answering the call.

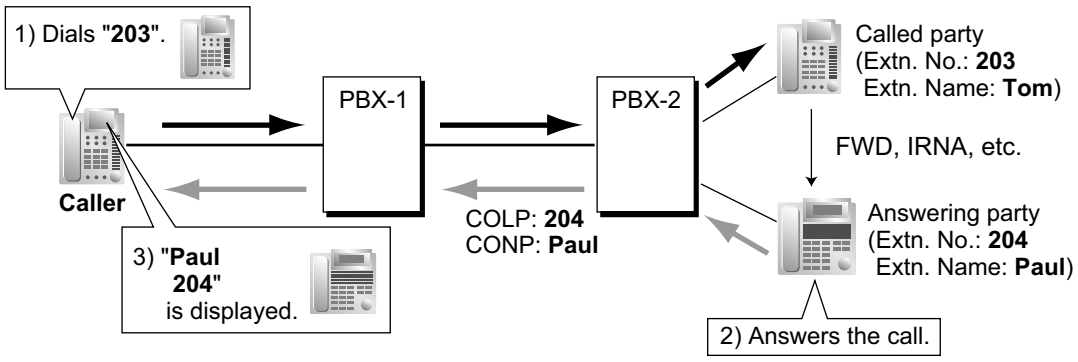
Connected Line/Name Identification Presentation (COLP/CONP):

The PBX sends a preprogrammed extension number and/or name to the QSIG network when the extension user answers an incoming call. The caller can see the number and/or name of the answering party on his telephone display when the call is answered.

[CLIP/CNIP Example]



[COLP/CONP Example]



CLIP/COLP Number:

The extension number sent to the QSIG network for CLIP/COLP can be assigned for each extension through system programming.

CNIP/CONP Name:

The extension name sent to the QSIG network for CNIP/CONP can be assigned for each extension through system programming.

Calling/Connected Line Identification Restriction (CLIR/COLR):

It is possible for each extension to restrict the sending of its extension number to the QSIG network by pressing the CLIR button, COLR button, or entering the feature number.

Calling/Connected Name Identification Restriction (CNIR/CONR):

It is possible for each extension to restrict the sending of its extension name to the QSIG network. When CLIR is activated, CNIR becomes active automatically. When COLR is activated, CONR becomes active automatically.

Conditions

- These features comply with the following European Telecommunication Standard (ETS) specifications:
  – CLIP/COLP: ETS 300 172 Circuit mode basis services.
  – CNIP/CONP: ETS 300 238 Name identification supplementary services.
- **COLP/CLIR/COLR/CNIP/CONP/CNIR/CONR Assignment for Each Port**
  Each service can be enabled or disabled on each ISDN (QSIG) port of the PBX.
- **CLIR Button and COLR Button**
  It is possible to switch between CLIP and CLIR by pressing the CLIR button, and COLP and COLR by pressing the COLR button. A flexible button can be customised as the CLIR or COLR button.

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features
→◆ COLR Set / Cancel
→◆ CLIR Set / Cancel
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main
→◆ Extension Number
→◆ Extension Name
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main
→◆ Extension Number
→◆ Extension Name

Feature Guide References

2.21.2 Flexible Buttons

PT Programming Manual References

[003] Extension Number
[004] Extension Name

User Manual References

1.9.4 Displaying Your Telephone Number on the Called Party and Caller's Telephone (Calling/Connected Line Identification Presentation [CLIP/COLP])
1.9.5 Preventing Your Telephone Number Being Displayed on the Caller's Telephone (Connected Line Identification Restriction [COLR])
1.9.6 Preventing Your Number Being Displayed on the Called Party's Telephone (Calling Line Identification Restriction [CLIR])

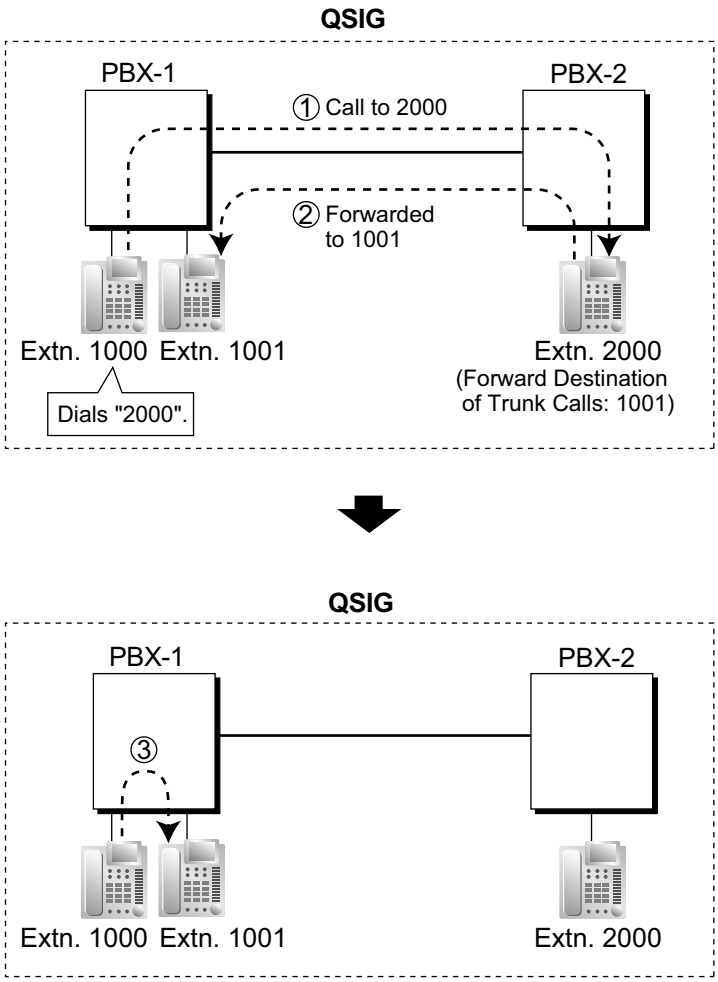
4.2.4.3 Call Forwarding (CF)—by QSIG

Description

The PBX forwards the call to a destination extension in another PBX in QSIG network. The destination can be set on your own PBX on an extension basis as the forward destination of trunk calls (→ 2.3.2 Call Forwarding (FWD)).

If the same trunk group is used for the incoming call and the forwarded call, the following situation will be possible.

[Example]



① Extension 1000 of PBX-1 dials extension number "2000", and the call is sent to extension "2000" of PBX-2 through QSIG network.

② The call is forwarded to the forward destination of trunk calls of extension 2000, which is extension "1001" of PBX-1.

③ The call between PBX-1 and PBX-2 is released, and the call is connected directly to the forward destination of extension 2000.

Conditions

- This feature complies with European Telecommunication Standard (ETS) specification ETS 300 257, Diversion supplementary services.
- This feature can be enabled or disabled on each ISDN (QSIG) port of the PBX.

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

User Manual References

1.6.1 Forwarding Calls

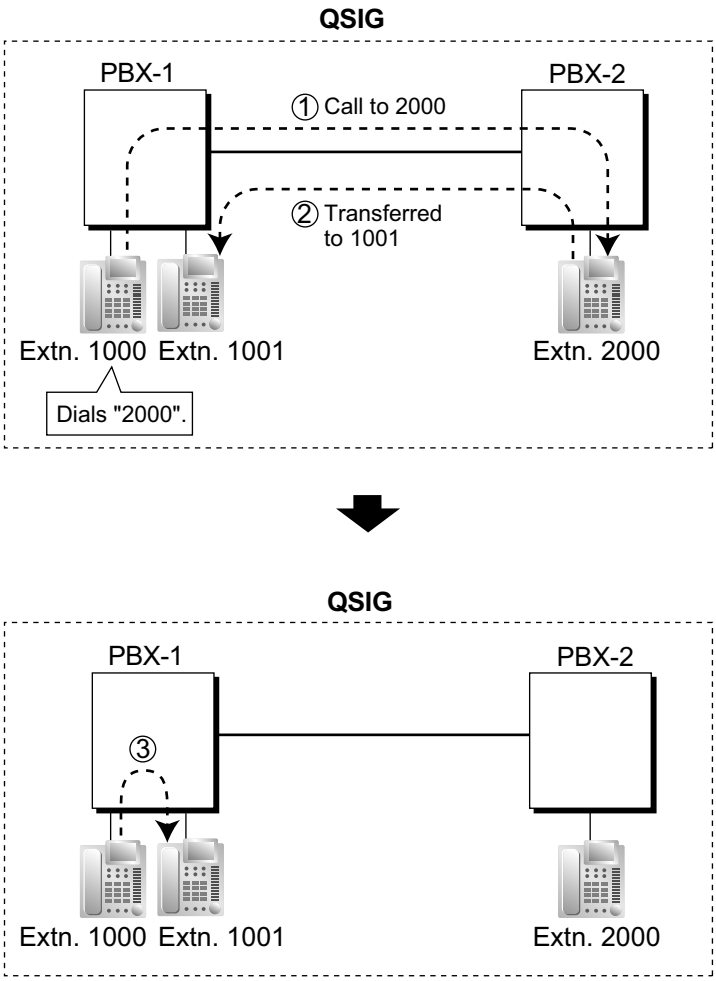
4.2.4.4 Call Transfer (CT)—by QSIG

Description

The PBX transfers the call to a destination extension in another PBX in QSIG network.

If the same trunk group is used for the incoming call and the transferred call, the following situation will be possible.

[Example]



① Extension 1000 of PBX-1 dials extension number "2000", and the call is sent to extension "2000" of PBX-2 through QSIG network.

② The call is transferred from extension 2000 to extension "1001" of PBX-1.

③ The call between PBX-1 and PBX-2 is released, and the call is connected directly to the transfer destination of extension 2000.

Conditions

- This feature complies with European Telecommunication Standard (ETS) specification ETS 300 261, Call transfer supplementary service.
- This feature can be enabled or disabled on an ISDN (QSIG) port basis.
- Call Transfer with Announcement and Call Transfer without Announcement is possible (→ 2.12.1 Call Transfer).

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Supplementary Service—◆COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

User Manual References

1.4.1 Transferring a Call (Call Transfer)

4.2.4.5 Completion of Calls to Busy Subscriber (CCBS)—by QSIG

Description

If the call has been made to an extension in another PBX in QSIG network and the called party is busy, an extension user can set to receive callback ringing when the called party becomes free. When the user answers the callback ringing, that party's number is automatically dialled.

Conditions

- This feature complies with European Telecommunication Standard (ETS) specification ETS 300 366, Call completion supplementary services.
- This feature is available under the following conditions:
  a. The caller's PBX is capable of using CCBS.
  b. The called party's PBX is capable of accepting CCBS.
- To receive and send CCBS, receiving and sending CCBS must be enabled individually on an ISDN (QSIG) port basis through system programming.
- An extension user can set only one CCBS. The last setting is effective.
- The CCBS setting is cancelled if there is no callback ringing within 60 minutes or callback ringing is not answered within 10 seconds.

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer

4.2.5 QSIG Enhanced Features

Description

When PBXs are networked using ISDN or V-IPGW cards, the following enhanced features are available.

When Calling an Extension in Another PBX and the Called Extension is Ringing

Feature	Description and Reference
Leave Message Waiting	→ 2.20.1 Message Waiting
Absent Message Display	→ 2.20.2 Absent Message
Ringing extension name display before answer	The ringing extension's name is displayed to the caller before the call is answered.

When the Called Extension of Another PBX is Busy

Feature	Description and Reference
Call Waiting	→ 2.1.3.3 Call Waiting **Note** When this feature is used over a network, a call waiting tone will be sent, even if OHCA or Whisper OHCA is enabled.
Executive Busy Override	→ 2.10.2 Executive Busy Override
Call Monitor	→ 2.10.3 Call Monitor
Leave Message Waiting	→ 2.20.1 Message Waiting

- When a called extension on another PBX is busy, Automatic Callback Busy can be used as a Standard QSIG feature (→ 4.2.4.5 Completion of Calls to Busy Subscriber (CCBS)—by QSIG).
- The caller will be informed an extension is busy with a busy tone and a display indication.
- Soft button operation is available for Call Waiting, Automatic Callback Busy and Executive Busy Override.

When the Called Extension of Another PBX has Do Not Disturb (DND) Set

Feature	Description and Reference
DND Override	→ 2.3.3 Do Not Disturb (DND)

- The caller will be informed an extension is set to DND by a DND tone and a display indication.
- Soft button operation is available for DND Override.

When Receiving a Call from an Extension of Another PBX

Feature	Description and Reference
Ring tone	Incoming calls from extensions in other PBXs are handled as intercom calls in regards to ring tone settings. → 2.1.3.2 Ring Tone Pattern Selection
Call Waiting	Incoming calls from extensions in other PBXs are handled as intercom calls in regards to Call Waiting settings. → 2.1.3.3 Call Waiting

Feature	Description and Reference
Call Forwarding (FWD)	Incoming calls from extensions in other PBXs are handled as intercom calls in regards to FWD settings. The Boss and Secretary feature will also function over a network. → 2.3.2 Call Forwarding (FWD)
Do Not Disturb (DND)	Incoming calls from extensions in other PBXs are handled as intercom calls in regards to DND settings. The Boss and Secretary feature will also function over a network. → 2.3.3 Do Not Disturb (DND)
Hands-free Answerback	Incoming calls from extensions in other PBXs are handled as intercom calls in regards to Hands-free Answerback settings. The Class of Service with Hands-free Answerback feature will also function over a network. → 2.4.4 Hands-free Answerback
Internal Call Block	Incoming calls from extensions in other PBXs are handled as intercom calls in regards to the Internal Call Block feature. For example, only branch office managers can be allowed to call the president at the main office, but other extensions in the branch office cannot call the president. → 2.1.2.2 Internal Call Block → 5.1.1 Class of Service (COS)

During a Call with Another Party

Feature	Description and Reference
Transfer to Busy Extension using Queuing (Camp-on Transfer)	You can transfer a call over the network to a busy extension in another PBX without having to use a Call Waiting operation. → 2.12.1 Call Transfer **Note** Call Transfer over a network is also supported as a standard QSIG feature. (→ 4.2.4.4 Call Transfer (CT)—by QSIG)

For Incoming Calls to an Extension on Another Networked PBX

Feature	Description and Reference
Directed Call Pickup	An extension user can answer a call ringing at another networked PBX extension. → 2.4.3 Call Pickup

Conditions

- KX-NSN002 (Activation Key for QSIG Network) is required for each Master unit to use these features. Furthermore, all PBXs in the network must be KX-NS series PBXs, or KX-NCP/KX-TDE series with MPR Software Version 4.1000 or later.
- A separate activation key is required for non-KX-NS series PBXs. For details, refer to the corresponding Feature Guide.
- Enhanced QSIG features can be disabled at the TIE Table level through system programming.

- Calls made by accessing a trunk via an S-CO button or Trunk Group Access feature number, etc. cannot use enhanced QSIG features because they do not refer to the TIE Table. This includes calls made from the call history of Communication Assistant (CA) because such calls specify a trunk group directly.
- Calls forwarded using Call Forwarding (CF)—by QSIG cannot use enhanced QSIG features.
- **Leave Message Waiting**
  Over a network, the Message Waiting set/cancel/callback feature number cannot be used to set or cancel Message Waiting.
- **Executive Busy Override**
  When executive busy override is used to interrupt a call between an extension in another PBX and an outside caller to make a 3-party conference call, and then the extension in the other PBX leaves the conversation, the call will be considered a trunk-to-trunk call. If enabled through system programming, the call will be terminated at this time.
- **Directed Call Pickup**
  - **When dialling using the PBX Code Method (Access with PBX Code)**
    Dial as follows: [TIE Line Access Number] + [PBX Code of ringing extension] + [Directed Call Pickup feature number] + [Ringing extension number].
    For example, to pick up the ringing extension 101 that is in another PBX with the PBX Code 123, you would dial "7-123-*41-101".
  - **When dialling using the Extension Number Method (Access without PBX Code)**
    Dial as follows: [All but last digit of the ringing extension number + *] + [Directed Call Pickup feature number] + [Ringing extension number].
    For example, to pick up the ringing extension 321 over a network, dial "32*-*41-321".
  - To use Directed Call Pickup over a VoIP network using Common Extension Numbering for Multiple PBXs, the operation must be predialled, or executed using CTI.

PC Programming Manual References

10.9 PBX Configuration—[2-9] System—System Options—Option 2—◆ CO - CO Call Limitation—After Conference
17.1 PBX Configuration—[9-1] Private Network—TIE Table—Enhanced QSIG

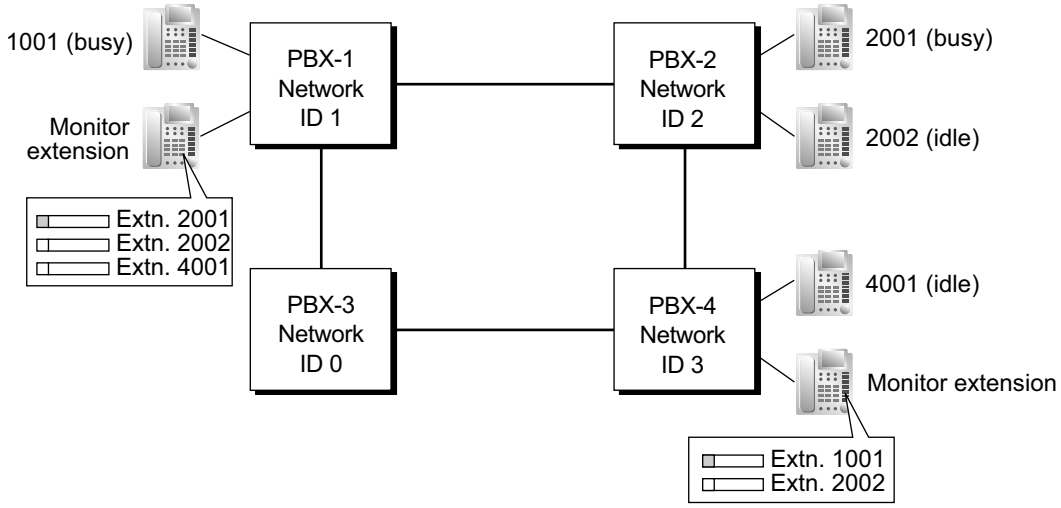
4.2.5.1 Network Direct Station Selection (NDSS)

Description

When PBXs are networked using ISDN or V-IPGW cards, it is possible to assign flexible buttons as Network Direct Station Selection (NDSS) buttons. These buttons are used to monitor the status of extensions connected to up to seven other PBXs in the network, and to make or transfer calls to those extensions with one touch, like normal DSS buttons. This allows operator functions to be centralised even when there are offices in remote locations.
NDSS buttons show the status of the monitored extension as follows:

Light pattern	Status
Off	The monitored extension is idle.
Red on	The monitored extension is busy or has set DND for trunk calls.

[Example Network]



[Programming Procedure]

1. **Routing Table Assignment**
   TIE Line Routing and Modification tables must have been programmed in advance for all PBXs in the network, to allow calls to be made and transferred between PBXs (→ 4.2.1 TIE Line Service).
   In the programming example that follows, the PBXs in the network use the Extension Number method, as shown in the illustration above. However, the PBX Access Code method can also be used.
2. **Network PBX ID Assignment**
   Assign a Network PBX ID to each PBX in the network.
   → 17.2 PBX Configuration—[9-2] Private Network—Network Data Transmission—◆ Network Data Transmission for Centralised Operator Feature—Network PBX ID
   - IDs 1-8: Can monitor extensions at other PBXs, and transmit monitor data about local extensions. Each ID number can be assigned to one PBX within the network.
   - ID 0: Retransmits monitor data throughout the network. This ID number can be assigned to multiple PBXs.
3. **QSIG Port Setting [Monitored PBX]**

ISDN-QSIG (PBX Direct Connection)

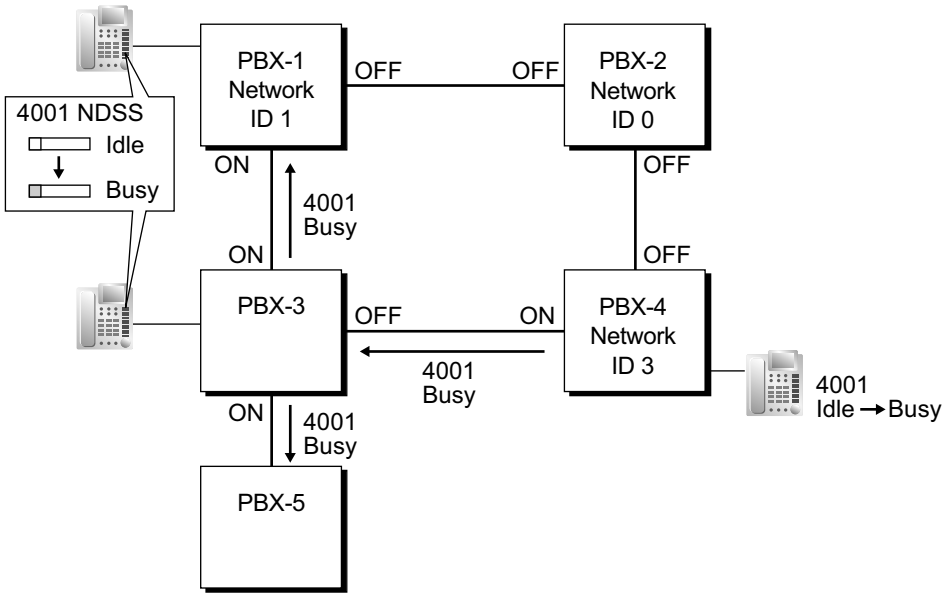
Each QSIG port of a PRI card that will be used to transmit extension status information must be set to do so through system programming. Even if a port is set to not transmit information, it will still receive information from other PBXs.

→ 9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—CO Setting—◆ Networking Data Transfer

Transmit	Extension Status Information Functions
Enabled	• Transmit information on own PBX extensions • Receive information from other PBXs; forward it to other QSIG ports of the PBX • Re-transmit information received by other QSIG ports of the PBX
Disabled	• Receive information from other PBXs; forward it to other QSIG ports of the PBX

Set the transmission setting for each port according to the layout of your network, so that extension status information can travel between NDSS monitor PBXs.

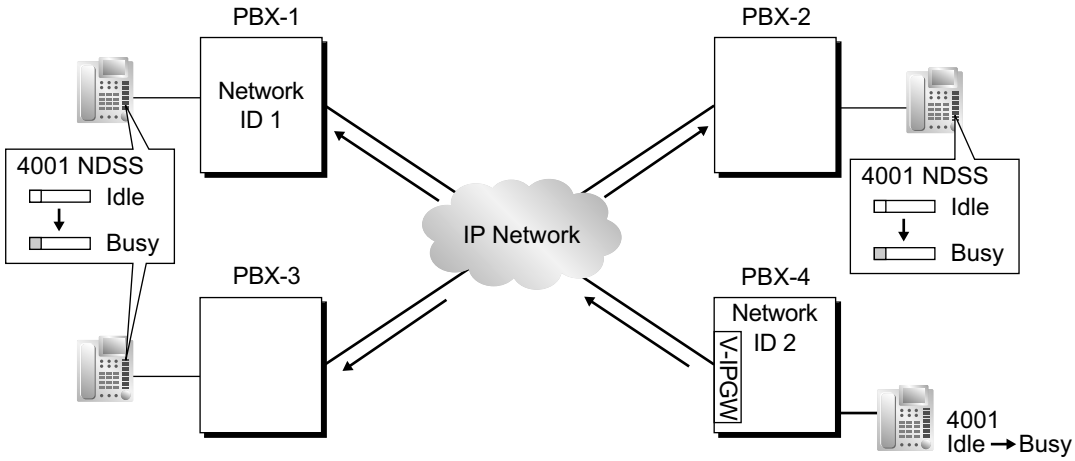
[Example ISDN Network Port Setting]



IP-Gateway-QSIG

IP data is routed through the network according to the routing tables of the V-IPGW cards. When using VoIP, it is necessary to identify the monitor PBXs, to which to transmit extension status information, for each card that will be used. This is done by specifying the extension number of any extension (for example, the PBX operator) at the monitor PBX as a Network Operator extension for that V-IPGW card.

[Example VoIP Network]



Mixed Network

When using the NDSS feature over a mixed network containing both VoIP portions and ISDN line portions, it is possible to set whether status information will be transferred between VoIP and ISDN cards within each PBX. For example, if the monitor PBX is on a VoIP network, and monitored PBXs are on an ISDN network, the PBX that acts as a gateway between the VoIP and ISDN networks must have this setting enabled for ISDN to VoIP.

4. **Network Monitor Extension Registration [Monitor PBX]**

Register the extensions (attached to other PBXs) that will be monitored. A maximum of 250 extensions can be registered. Only extensions that have been registered here can be assigned to NDSS buttons.
→ 17.4 PBX Configuration—[9-4] Private Network—NDSS Key Table—◆ Network Extension No.

Network Monitor Extensions Table

Index No.	Network Extn. No.	Network Extn. Name
001	2001	Branch 1: T. Durden
002	2002	Branch 1: M. Singer
003	4001	Branch 2: R. Paulson
:	:	:
250		

When using the PBX Access Code numbering method, the relevant access code must be added before the extension number registered here.

5. **NDSS Button Customisation [Monitor PBX]**

At any extension attached to a monitor PBX, customise a flexible button as an NDSS button for an extension registered above. Then go off-hook, press this button once, and go on-hook again. This activates the monitoring function. The monitored PBX will begin to transmit information about the status of that extension, and the monitor PBX will begin to receive the information. The button light will display the status of the extension connected to another PBX.

Removing or Editing a Registered Extension [Monitor PBX]

To monitor a new extension when 250 extensions are already being monitored, it is necessary to remove the registration of an existing extension.
The NDSS Monitor Release feature is used to stop monitoring a certain extension. When this feature is performed at a monitor PBX:

- If no other PBXs are monitoring the selected extension, the monitored PBX stops transmitting status information for that extension.
- The monitor PBX stops receiving status information for the selected extension. All NDSS buttons for that extension stop displaying status information.

However, the related information is not deleted from the Network Monitor Extensions Table. Therefore, if an extension user at a monitor PBX subsequently goes off-hook and presses the NDSS button for that extension, monitoring will be reactivated. To completely remove monitoring of an extension, the registration data must also be deleted from the Network Monitor Extensions Table at each monitor PBX.
The monitor destination of each NDSS button is determined by the registration information for a particular index number in this table. Therefore, if the registration information for an index number is changed (for example, the Network Extension Number assigned to Index No. 001 in the example above is changed from "2001" to "4002"), any NDSS buttons that have been set for that extension will automatically point to the new monitor destination.

Conditions

- KX-NSN002 (Activation Key for QSIG Network) is required for each Master unit that will monitor extensions or have extensions monitored.
  A separate activation key is required for non-KX-NS series PBXs. For details, refer to the corresponding Feature Guide.
- NDSS buttons will not function on networks using Common Extension Numbering for 2 PBXs, or Common Extension Numbering for Multiple PBXs.
- All PBXs in the network must be KX-NS series PBXs, KX-NCP series, KX-TDE series, or KX-TDA series PBXs. For information on the hardware requirements for non-KX-NS300 PBXs that will monitor extensions or have extensions monitored, refer to the corresponding Feature Guide.
- Each of Network PBX IDs 1-8 can only be assigned to one PBX within a network. Assigning the same Network PBX ID to two PBXs will cause network data transmission problems.
- It is only possible to assign NDSS buttons for extensions that have been previously registered in the Network Monitor Extensions Table.
  12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
  → ◆ Type
  → ◆ Dial (for NDSS)
  12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
  → ◆ Type
  → ◆ Dial (for NDSS)
- Extensions connected to one PBX can monitor a maximum of 250 extensions at other PBXs. One extension can be monitored by multiple extensions at multiple PBXs.
- To reduce NDSS data traffic, it is recommended that 8 or less extensions in a single Incoming Call Distribution Group be monitored.
- Activation of an NDSS button is only required the first time that a button for a newly registered extension is created. Once an NDSS button has been activated by being pressed the first time, any further NDSS buttons for the same monitored extension will automatically display the extension status without needing to be activated.
- To use the NDSS Monitor Release feature, an extension must be assigned as a manager.
  → 10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Manager—◆ Manager
- Through system programming, it is possible to assign a maximum number of "hops" (transfers between PBXs) that extension status information can travel. Each time a piece of extension status information is forwarded to another PBX, its counter is increased by one. When this counter reaches the assigned maximum, the data is discarded. This is used to prevent data from circling unnecessarily around the network.
  → 17.2 PBX Configuration—[9-2] Private Network—Network Data Transmission—◆ Network Data Transmission for Centralised Operator Feature—Data Transmission Counter

- If it is not possible to remotely turn off transmission of extension status information using the NDSS Monitor Release feature because of network conditions, it is possible to perform the same operation directly through system programming at the monitored PBX.
  → 12.1.6 PBX Configuration—[4-1-6] Extension—Wired Extension—NDSS Link Data - Send
  → 12.2.4 PBX Configuration—[4-2-4] Extension—Portable Station—NDSS Link Data - Send
- When using a VoIP network, if extension status information is lost by the network, in some cases an NDSS button may not be able to display the status of the relevant extension.
- NDSS cannot be used when two PBXs are networked using the **[TIE Call by Own PBX Extension Number]** feature (→ 4.2.1 TIE Line Service).

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—CO Setting—◆ Networking Data Transfer
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Manager—◆ Manager
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Dial (for NDSS)
12.1.6 PBX Configuration—[4-1-6] Extension—Wired Extension—NDSS Link Data - Send
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type
→◆ Dial (for NDSS)
12.2.4 PBX Configuration—[4-2-4] Extension—Portable Station—NDSS Link Data - Send
17.2 PBX Configuration—[9-2] Private Network—Network Data Transmission
17.3 PBX Configuration—[9-3] Private Network—Network Operator (VoIP)
17.4 PBX Configuration—[9-4] Private Network—NDSS Key Table

PT Programming Manual References

[511] Manager Assignment

Feature Guide References

4.2.1 TIE Line Service
4.2.2 Voice over Internet Protocol (VoIP) Network
4.2.4 QSIG Standard Features

User Manual References

1.2.1 Basic Calling
2.1.7 Releasing Network Direct Station Selection (NDSS) Monitor

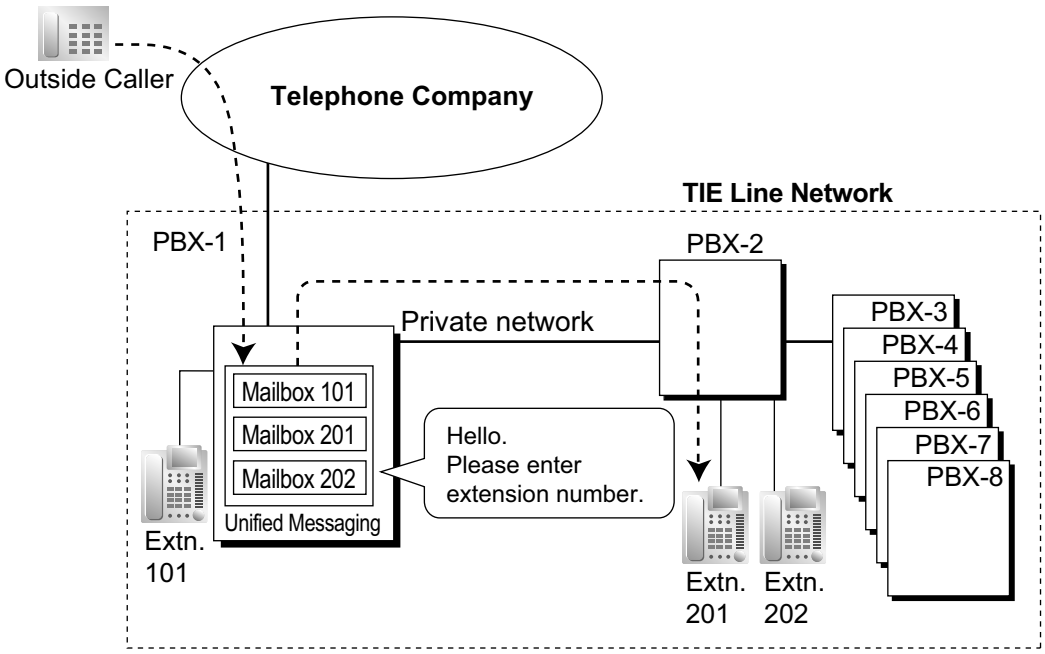
4.2.5.2 Centralised Voice Mail

Description

Up to 7 PBXs connected with this PBX in a TIE line network over VoIP or ISDN can share the services of the Unified Messaging system of this PBX. The Unified Messaging system can provide voice mail for extensions attached to any of the PBXs in the network. In addition, the Unified Messaging system can send Message Waiting notifications to extensions at any PBX, and users (subscribers) can access their mailboxes directly using the Message Waiting Button.
The Unified Messaging features available at extensions in a network are as follows:

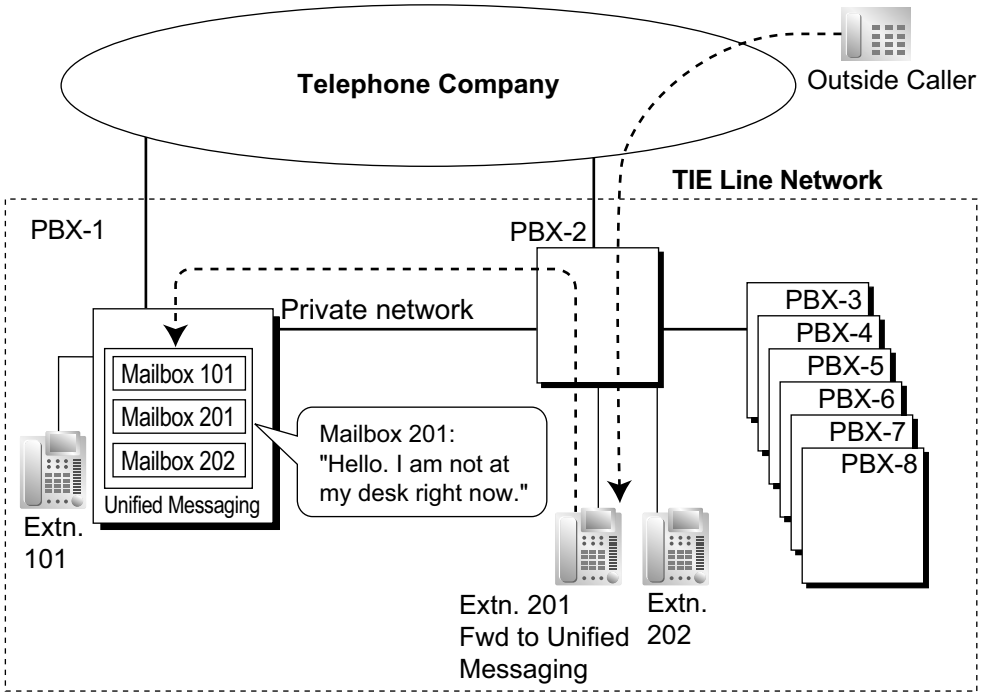
- FWD to a Mailbox
- Intercept Routing to a Mailbox
- Voice Mail (VM) Transfer Button
- Listening to a Recorded Message (Direct Mailbox Access)
- Trunk Service & Automatic Time Mode Notification for Incoming Call
- Caller's Identification Notification
- Status Notification
- Voice message number display

[Trunk Call Answered by the Unified Messaging System, Transferred to Extension at Other PBX (Extension Number Method)]



[Explanation]
A trunk call is answered by the AA service of the Unified Messaging system. The caller enters extension number 201, so the call is transferred over the private network to extension 201.
If extension 201 does not answer, mailbox 201 will answer it and play the appropriate message.

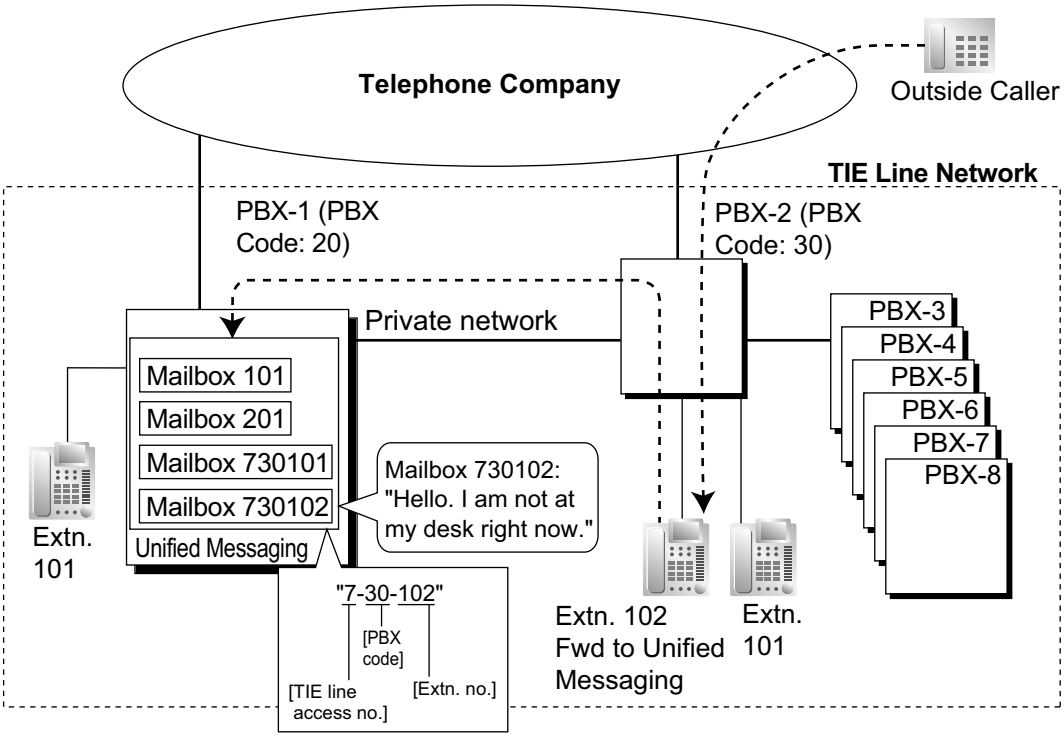
[Trunk Call to an Extension Not Answered, Forwarded to Mailbox (Extension Number Method)]



[Explanation]

Extension 201 does not answer the trunk call, so the call is forwarded to the Unified Messaging system and answered by mailbox 201. If the outside caller leaves a message, the Unified Messaging system sends a Message Waiting notification to the extension using Enhanced QSIG information over the private network. When forwarding the call, PBX-2 sends any received call information (Caller ID number/name, DDI number) along with the trunk group number to use, applicable time mode, and extension number and forwarding reason of the original destination extension to PBX-1 as Enhanced QSIG information.

PBX Code Method



[Explanation]
Extension 102 does not answer the trunk call, so the call is forwarded to the Unified Messaging system, using a mailbox number containing the **TIE line access number**, the **PBX code** of the PBX that received the call, and the **extension number** that received the call.
This mailbox number is the same as if an extension connected to PBX-1 called extension 102 of PBX-2. This mailbox number should be programmed as the mailbox number and owner extension number on the Unified Messaging system.

Multiple Voice Mail Services

More than one PBX in a network can provide voice mail services to extensions connected to other PBXs.

Conditions

[General]

- This section explains Centralised Voice Mail assuming that the Unified Messaging system is being shared. If you want to use the Centralised Voice Mail feature with a VPS over a stacking connection, refer to the documentation of the PBX from which you will share the VPS.
- KX-NSN002 (Activation Key for QSIG Network) is required for all KX-NS series PBXs that will use this feature, whether they are hosting the voice mail service or just using it.

  Note

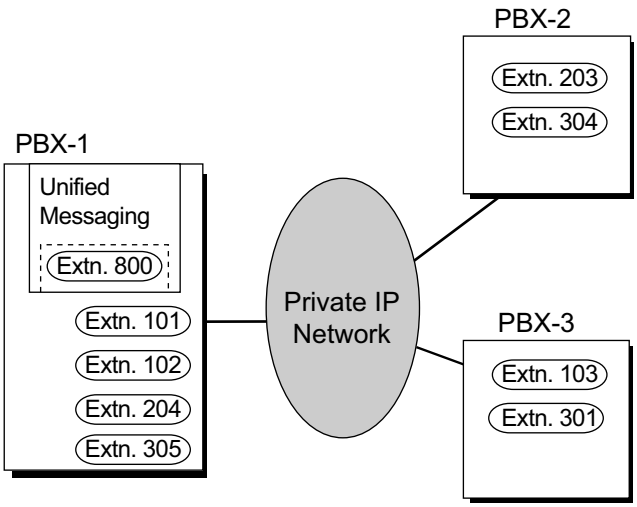
  Centralised voice mail is used only among PBXs connected over a TIE line.
- A separate activation key is required for non-KX-NS series PBXs. For details, refer to the corresponding Feature Guide.
- All PBXs in the network must be KX-NS series PBXs, KX-NCP series, KX-TDE series, or KX-TDA series PBXs. For information on the hardware requirements for non-KX-NS300 PBXs that share the same voice mail service, refer to the corresponding Feature Guide.

- An extension can receive Message Waiting notifications from multiple voice mail services connected to PBXs in the network. When multiple notifications of the number of unheard messages in an extension user's message box are sent from different voice mail services, the most recent notification will be displayed.
- A flexible button cannot be customised as a Message Waiting button for another extension at a different PBX.
- Whether or not Enhanced QSIG information is transmitted can be set in the TIE table.
- **PBX Code Method**
  The number used in this method must be no more than 8 digits.

[Voice Mail (VM) Transfer Button]

- A flexible button can be customised as the VM Transfer button with the floating extension number of the UM group (KX-NS series PBXs) or VM group (KX-TDA/KX-TDE/KX-NCP/KX-NS series PBX) of a remote voice mail system as the parameter.
- A call in progress can be transferred to a Unified Messaging mailbox by pressing the VM Transfer button and then (1) pressing a DSS button or NDSS button, or (2) dialling the desired extension number directly. When the desired extension number (or TIE Line Access number + PBX code + desired extension number) is dialled directly, it must be followed by "#" in these cases:
  – The extension user performing the transfer and the destination extension belong to different PBXs.
  – The PBX of the extension user pressing the VM Transfer button does not have its own voice mail service.
- VM Transfer button functions will not operate on networks using Common Extension Numbering for 2 PBXs, or Common Extension Numbering for Multiple PBXs. It is necessary to create a network where the Voice Mail can be accessed by the TIE Line Access number or an Other PBX Extension number.

Example: Using Centralised Voice Mail with Common Extension Numbering



[Explanation]

In this example, only PBX-1 has an extension beginning with "8" assigned (for the Unified Messaging system). In the Flexible Numbering Plan for PBX-2 and PBX-3, "1", "2", and "3" must be set as "Extension Numbers", and "8" must be set as an "Other PBX Extension Number (TIE)".

PC Programming Manual References

17.1 PBX Configuration—[9-1] Private Network—TIE Table—Enhanced QSIG

17.2 PBX Configuration—[9-2] Private Network—Network Data Transmission
17.5 PBX Configuration—[9-5] Private Network—Centralised UM/VM Unit
19.1 PBX Configuration—[11-1] Maintenance—Main—Maintenance
→◆ Error Log for Centralised VM—Network MSW Transmission (Counter)
→◆ Error Log for Centralised VM—Network MSW Transmission (Buffer)

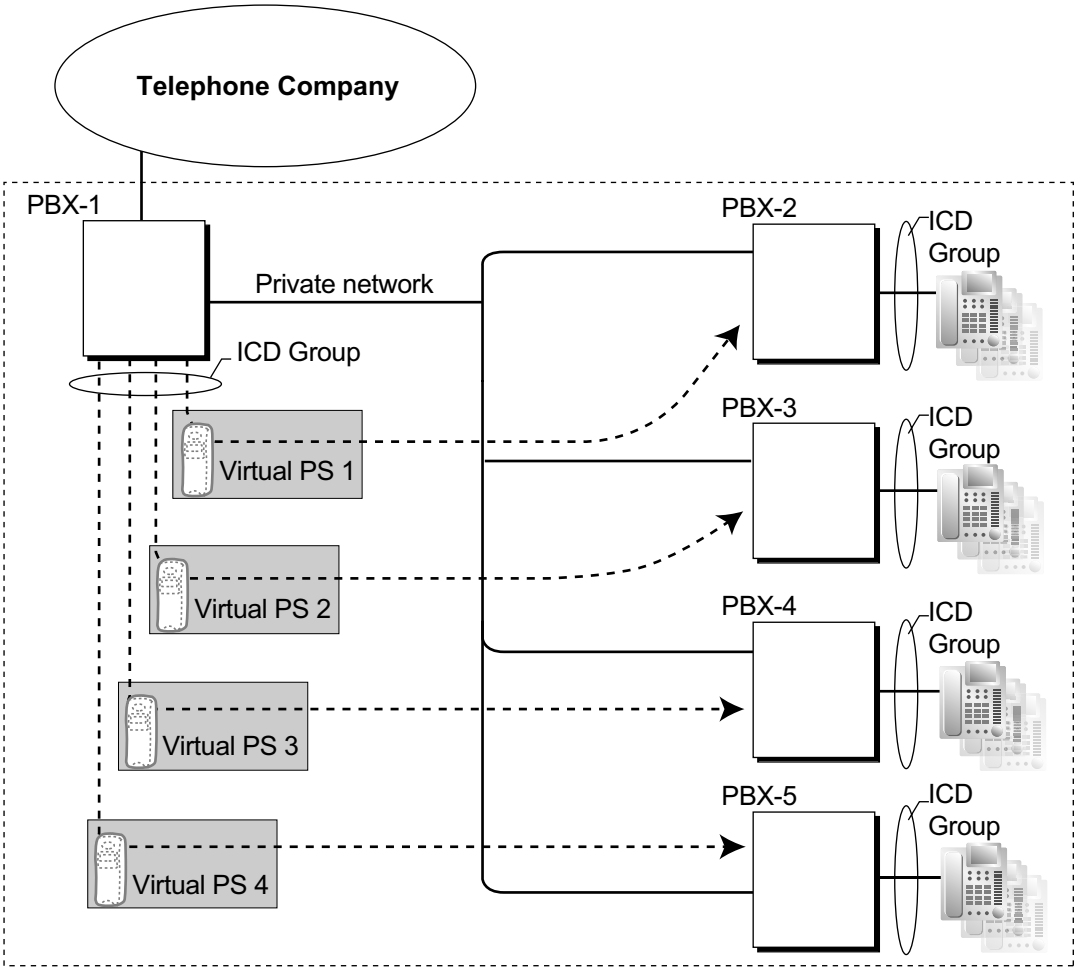
Feature Guide References

Section 3 Unified Messaging System
2.28.3 Voice Mail DPT (Digital) Integration
4.2.1 TIE Line Service

4.2.6 Network ICD Group

Description

An Incoming Call Distribution (ICD) Group can include up to 4 destinations at other PBXs in a private network, including the floating extension number of another ICD Group. This is done by assigning a virtual PS as a member of the ICD Group, and then setting the number of a destination at another PBX as the forwarding destination for that virtual PS. This allows multiple ICD groups at remote locations to receive calls together. In addition to the Ring distribution method, Uniform Call Distribution (UCD) and Priority Hunting can also be selected. (→ 2.2.2.2 Group Call Distribution)



Conditions

- KX-NSE101, KX-NSE105, KX-NSE110, or KX-NSE120 (Activation Key for Mobile Extension) is required to use this feature. One activation key is required for each extension (virtual PS) that will use this feature. Also, the **Mobile Extension** setting for each virtual PS must be set to **Enable**.
- The conditions for **2.2.2.3 Outside Destinations in Incoming Call Distribution Group** also apply to this feature.
- Call distribution to the longest idle extension (Automatic Call Distribution) cannot be used with Virtual PS ICD Group members.

PC Programming Manual References

10.9 PBX Configuration—[2-9] System—System Options—Option 4
→◆ Send CLIP of CO Caller—when call is forwarded to CO
→◆ Send CLIP of Extension Caller—when call is forwarded to CO
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 9—◆ Mobile Extension
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 9—◆ Mobile Extension

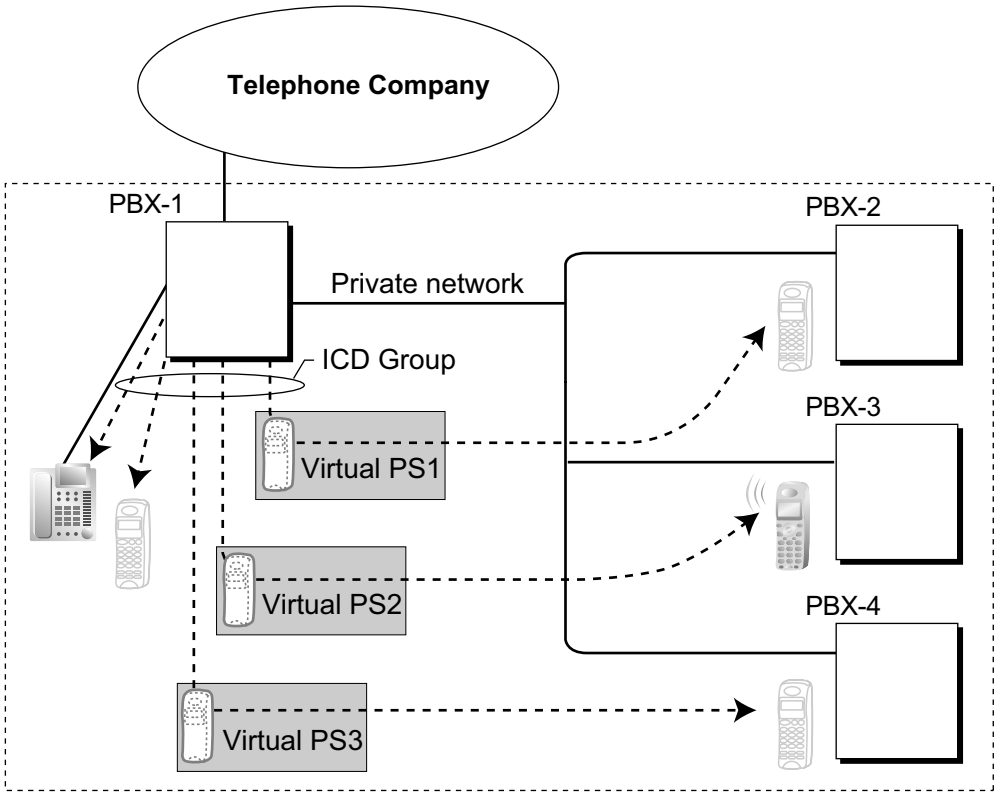
Feature Guide References

2.2.2.3 Outside Destinations in Incoming Call Distribution Group
5.2.4.6 Virtual PS

4.2.6.1 PS Roaming by Network ICD Group

Description

One PS can be registered to up to 4 PBXs in a private network, and a Network ICD group created for the PS at each PBX, with virtual PSs set to forward to the other PBXs in the network. When a call to the PS is received at one of the PBXs, the call rings simultaneously at all PBXs in the network to which that PS is registered.



Each virtual PS is set to forward calls to the extension number of the actual PS as registered at one of the other PBXs.
Then, an Incoming Call Distribution (ICD) Group is created containing the registered PS and the virtual PSs. When a call is received at one of the PBXs, it is forwarded to all of the other PBXs. One private network channel is used to forward an incoming call to one other PBX. Therefore, if a PS is registered at 3 other PBXs, 3 private network channels are needed to forward a single call to all of the PBXs.
Each PBX can store the current communication status of each PS (In Range or Out of Range). If the status of the PS is set to Out of Range when a call is received, the call will be refused and the private network channel will be released immediately. Since the PS can only be set to In Range at a single PBX at one time, any other PBXs to which the call is transferred will refuse the call, releasing the VoIP or ISDN channels.

Conditions

- KX-NSE101, KX-NSE105, KX-NSE110 or KX-NSE120 (Activation Key for Mobile Extension) is required to use this feature. One activation key is required for each extension (virtual PS) that will use this feature. Also, the **Mobile Extension** setting for each virtual PS must be set to **Enable**.
- If no signal is received from a PS for the preprogrammed length of time when an incoming call is received, the communication status of the PS is set to Out of Range, if enabled through system programming.
- When a PS comes within range of a certain PBX, Out of Range status is automatically released.

However, in some negative wireless network conditions, Out of Range status may not be released automatically. In this case, the PS user can manually release Out of Range status by pressing the TALK button and confirming that a dial tone can be heard from the PBX.

- If the status of a PS is set to Out of Range at all PBXs that it is registered to, the call will be redirected to the overflow destination of the ICD Group.
- Handover from one PBX to another PBX during a conversation is not possible.

PC Programming Manual References

10.3 PBX Configuration—[2-3] System—Timers & Counters—Miscellaneous—◆ System Wireless—PS Out of Range Timer (s)
10.9 PBX Configuration—[2-9] System—System Options—Option 4—◆ System Wireless—Out of Range Registration
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 9—◆ Mobile Extension
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Option 9—◆ Mobile Extension

Feature Guide References

2.2.2.3 Outside Destinations in Incoming Call Distribution Group

4.2.6 Network ICD Group

Section 5

System Configuration and Administration Features

5.1 System Configuration—System

5.1.1 Class of Service (COS)

Description

Each extension must belong to a Class of Service (COS). By assigning certain extensions to a COS, it is possible to control the behaviour and privileges of extension users (allowing or denying certain extensions access to various features, extensions, and trunks) depending on the duties appointed to them.
Many extensions can belong to the same COS by assigning each extension the same COS number, allowing the same restrictions and privileges to apply to a group of extensions.
The following features are controlled on a COS basis:

a. → 2.1.2.2 Internal Call Block
b. → 2.3.2 Call Forwarding (FWD)
c. → 2.3.3 Do Not Disturb (DND)—DND Override
d. → 2.4.3 Call Pickup
e. → 2.5.4.3 Account Code Entry
f. → 2.5.5.3 Trunk Access
g. → 2.10.2 Executive Busy Override
h. → 2.10.3 Call Monitor
i. → 2.10.4.3 Off-hook Call Announcement (OHCA)
j. → 2.10.4.4 Whisper OHCA
k. → 2.7.1 Toll Restriction (TRS)/Call Barring (Barring)
l. → 2.7.3 Extension Dial Lock
m. → 2.7.5 Walking COS
n. → 2.11.8 Trunk Call Limitation
o. → 2.12.1 Call Transfer
p. → 2.18.2 Door Open
q. → 2.16.1 Direct Inward System Access (DISA)
r. → 5.2.4.5 Wireless XDP Parallel Mode
s. → 2.22.1.1 Station Message Detail Recording (SMDR)—SMDR for Outgoing Trunk Calls
t. → 5.1.4 Time Service—Time Service Switching
u. → 5.1.6 Manager Features
v. → 5.5.3 PT Programming
w. → 2.9.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension—SDN Key mode, SDN Walking COS, and assigning SDN buttons through PT programming

COS for Unified Messaging

The Unified Messaging system has its own COS settings for controlling access to various functions.
(→ 3.2.1.11 Class of Service (COS))

Conditions

- **Walking COS**
  Extension users can temporarily use their own COS at another extension with a less-privileged COS to access features, extensions, or trunks that are normally inaccessible due to that extension's COS.

PC Programming Manual References

10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings

12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ COS
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main—◆ COS

PT Programming Manual References

[602] Class of Service

Feature Guide References

3.2.1 System Features
6.1 Capacity of System Resources

User Manual References

1.2.7 Calling without Restrictions

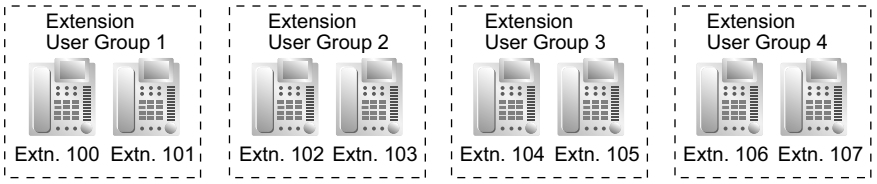
5.1.2 Group

Description

This PBX supports various types of groups.

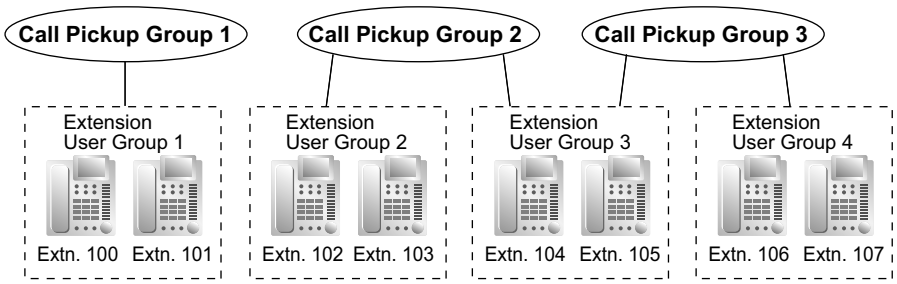
1. **Trunk Group**
   Trunks can be grouped into a specified number of trunk groups (e.g., for each carrier, trunk type, etc.). Several settings can be assigned on a trunk group basis. All trunks belonging to a trunk group follow the assignment determined for that trunk group.
   → 11.1.1 PBX Configuration—[3-1-1] Group—Trunk Group—TRG Settings
   → [402] Trunk Group Number
   One trunk can belong to only one trunk group on a port basis.
   Port basis: LCOT/ISDN-PRI30/SIPGW
   Channel basis: E1
2. **Extension User Group**
   The PBX supports extension user groups, each of which is used to compose the following groups:
   a. Tenant (→ 5.1.3 Tenant Service)
   b. Call Pickup Group (See below.)
   c. Paging Group (See below.)
   Every extension must belong to one extension user group, but cannot belong to more than one extension user group.
   → 11.2 PBX Configuration—[3-2] Group—User Group
   → [603] Extension User Group
   Assignable Extensions: PT/SLT/PS/SIP Extension/ISDN Extension

[Example]



Call Pickup Group

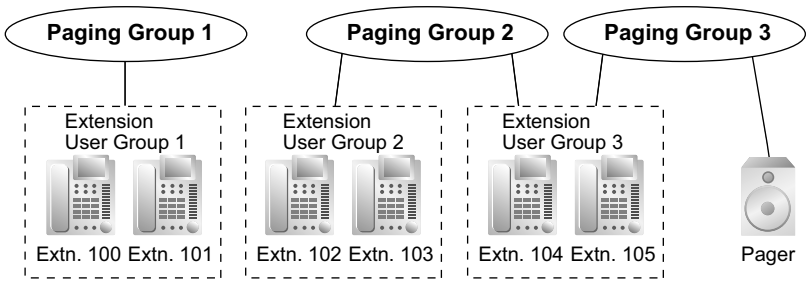
Using the Group Call Pickup feature, extensions can answer any calls within a specified group.
One extension user group can belong to several call pickup groups. (→ 2.4.3 Call Pickup)
→ 11.3 PBX Configuration—[3-3] Group—Call Pickup Group
→ 11.3.1 PBX Configuration—[3-3] Group—Call Pickup Group—All Setting
→ [650] Extension User Groups of a Pickup Group
[Example]



Paging Group

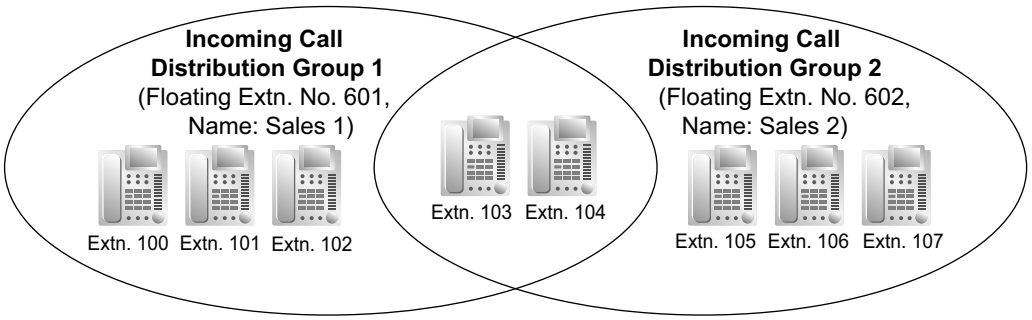
Using the Paging feature, extensions can make a page to any paging groups or answer a page to their own groups. One extension user group or external pager can belong to several paging groups.

(→ 2.17.1 Paging)
→ 11.4 PBX Configuration—[3-4] Group—Paging Group
→ 11.4.1 PBX Configuration—[3-4] Group—Paging Group—All Setting
→ 11.4.2 PBX Configuration—[3-4] Group—Paging Group—External Pager
→ [640] Extension User Groups of a Paging Group
[Example]



3. **Idle Extension Hunting Group**
If a called extension is busy or in DND mode, Idle Extension Hunting redirects the incoming call to an idle member of the same idle extension hunting group, which can be programmed through system programming. Idle extensions are automatically searched according to a preprogrammed hunting type: Circular Hunting or Terminated Hunting (→ 2.2.1 Idle Extension Hunting).
→ 11.6 PBX Configuration—[3-6] Group—Extension Hunting Group
→ 11.6.1 PBX Configuration—[3-6] Group—Extension Hunting Group—Member List
→ [680] Idle Extension Hunting Type
→ [681] Idle Extension Hunting Group Member
4. **Incoming Call Distribution Group**
An incoming call distribution group is a group of extensions which receives incoming calls directed to the group. Each incoming call distribution group has a floating extension number (default: 6 + two-digit group number[*1]) and name. One extension can belong to multiple groups.
→ 11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings
→ 11.5.1.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Member List
→ [623] Incoming Call Distribution Group Name
Assignable Extensions: PT/SLT/PS/SIP Extension/ISDN Extension/PS Ring Group
(→ 2.2.2 Incoming Call Distribution Group Features)

[Example]



[*1] The number of digits for **Floating Extn. No** depends on the value specified for **Numbering Plan** in Easy Setup.
→ 5.4.1 Easy Setup Wizard

5. **UM Group**
A UM group is the collection of all Unified Messaging ports of one PBX. A UM group is assigned a floating extension number.

(→ 3.1.1 Unified Messaging System Overview)
→ 11.7.1 PBX Configuration—[3-7-1] Group—UM Group—System Settings
→ 11.7.2 PBX Configuration—[3-7-2] Group—UM Group—Unit Settings
→ [660] UM Group Floating Extension Number

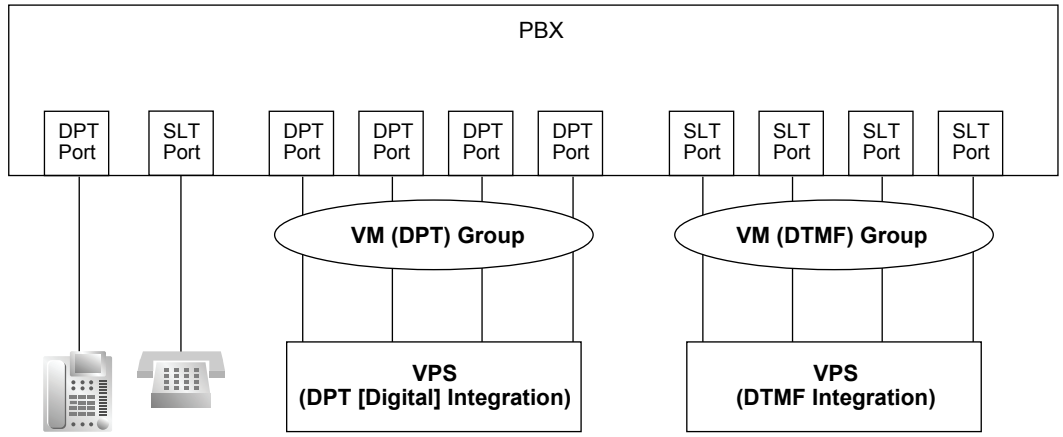
6. VM Group

There are two types of VM groups as follows:

Type	Description
VM (DTMF) Group	A group of SLT ports which use the Voice Mail DTMF Integration features. One SLT port can belong to only one group.
VM (DPT) Group	A group of DPT ports which use the Voice Mail DPT (Digital) Integration features. One DPT port can belong to only one group.

→ 2.28.1 Voice Mail (VM) Group
→ 11.11.1 PBX Configuration—[3-11-1] Group—VM (DPT) Group—System Settings
→ 11.11.2 PBX Configuration—[3-11-2] Group—VM (DPT) Group—Unit Settings
→ 11.12.1 PBX Configuration—[3-12-1] Group—VM (DTMF) Group—System Settings
→ 11.12.2 PBX Configuration—[3-12-2] Group—VM (DTMF) Group—Group Settings
→ 11.12.2.1 PBX Configuration—[3-12-2] Group—VM (DTMF) Group—Group Settings—Member List
→ [661] VM Group Floating Extension Number

[Example]

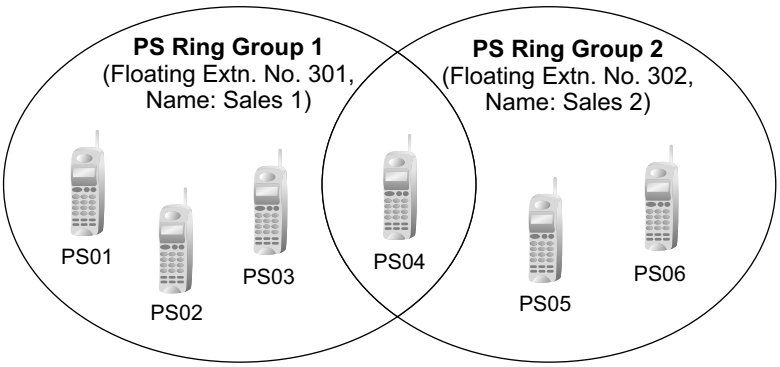


7. PS Ring Group

A PS ring group is a group of PS extensions that receives incoming calls directed to the group. Each group has a floating extension number and name through system programming. One PS can belong to multiple groups.

(→ 5.2.4.2 PS Ring Group)
→ 11.8 PBX Configuration—[3-8] Group—PS Ring Group

→ 11.8.1 PBX Configuration—[3-8] Group—PS Ring Group—Member List



8. Conference Group

A conference group is a group of parties that are called when an extension user uses the Conference Group Call feature (→ 2.15.1 Conference Group Call). When Broadcast Mode is enabled through system programming, a maximum of 31 parties can be assigned to a group. When Broadcast Mode is disabled, a maximum of 7 parties can be assigned to a group. A maximum of 8 conference groups can be programmed.

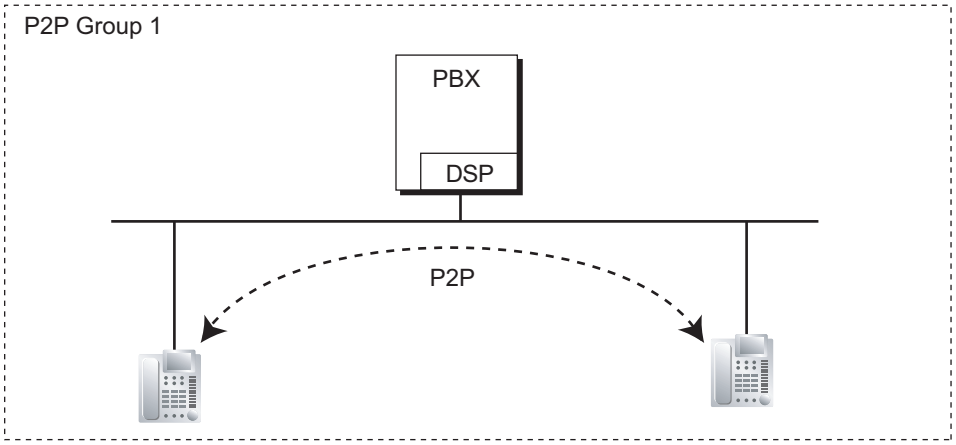
→ 11.9 PBX Configuration—[3-9] Group—Conference Group

→ 11.9.1 PBX Configuration—[3-9] Group—Conference Group—Member List

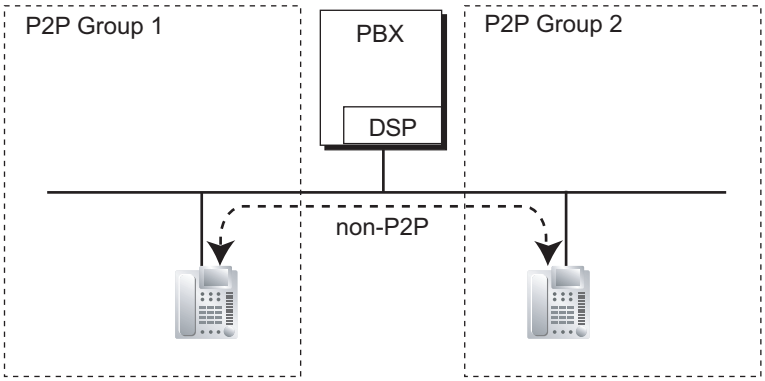
9. P2P Group

Devices in the same P2P group can establish peer-to-peer (P2P) connections and communicate (make calls) without using PBX resources. IP-PTs, SIP extensions, SIP Trunks, and PBXs are all assigned to P2P groups.

Connection between devices in the same P2P group



Connection between devices in different P2P groups



→ 11.10 PBX Configuration—[3-10] Group—P2P Group

Installation Manual References

5.4.1 Easy Setup Wizard

PC Programming Manual References

Section 11 PBX Configuration—[3] Group

PT Programming Manual References

[402] Trunk Group Number
[603] Extension User Group
[620] Incoming Call Distribution Group Member
[622] Incoming Call Distribution Group Floating Extension Number
[623] Incoming Call Distribution Group Name
[640] Extension User Groups of a Paging Group
[650] Extension User Groups of a Pickup Group
[660] UM Group Floating Extension Number
[661] VM Group Floating Extension Number
[680] Idle Extension Hunting Type
[681] Idle Extension Hunting Group Member

Feature Guide References

5.5.8 Floating Extension
6.1 Capacity of System Resources

5.1.3 Tenant Service

Description

This PBX can be shared with a certain number of tenants.

1. Tenant Configuration

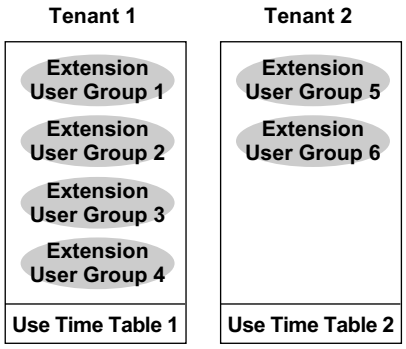
Tenant Member

The tenant members consist of extension user groups. One extension user group can belong to only one tenant. Therefore, one extension can belong to only one tenant.
(→ 5.1.2 Group)

Time Service

Each tenant has a Time Table. The Start and/or End time of each time mode (day/lunch/break/night) can be set for each day of the week. The Time Table numbers correspond to the tenant numbers respectively.
(→ 5.1.4 Time Service)

[Example]



2. System Management

Each of the following system management items can be assigned to each tenant.

a. Tenant Operator (extension number/floating extension number of incoming call distribution group/ none) (→ 5.1.5 Operator Features)
→ 14.6 PBX Configuration—[6-6] Feature—Tenant—◆ Operator (Extension Number)

b. ARS Mode (Off/Local Access/All Access/System) (→ 2.8.1 Automatic Route Selection (ARS))
→ 14.6 PBX Configuration—[6-6] Feature—Tenant—◆ ARS Mode

c. Music Source for Music on Hold (System/BGM Number/Tone)
(→ 2.13.4 Music on Hold)
→ 14.6 PBX Configuration—[6-6] Feature—Tenant—◆ Music On Hold

d. System Speed Dialling (System/Tenant Exclusive)
(→ 2.6.4 Speed Dialling—Personal/System)
→ 14.6 PBX Configuration—[6-6] Feature—Tenant—◆ System Speed Dial

[Programming Example]

Tenant No.	Operator	ARS Mode	Music Source	System Speed Dialling
1	Extn.101	Local Access	System*3	System*4
2	None*1	System*2	Tone	Extended/ Tenant Exclusive
3	Floating extn. no. 200	Off	BGM1	Extended/ Tenant Exclusive

Tenant No.	Operator	ARS Mode	Music Source	System Speed Dialling
:	:	:	:	:

*1 Follows the system assignment of a PBX operator.
→ 10.2 PBX Configuration—[2-2] System—Operator & BGM—◆ PBX Operator—Day, Lunch, Break, Night

*2 Follows the system assignment of the ARS mode.
→ Section 16 PBX Configuration—[8] ARS

*3 Follows the system assignment of the music source for the Music on Hold.
→ 10.2 PBX Configuration—[2-2] System—Operator & BGM—◆ BGM and Music on Hold—Music on Hold

*4 Follows the system assignment for System Speed Dialling.
→ 14.1 PBX Configuration—[6-1] Feature—System Speed Dial

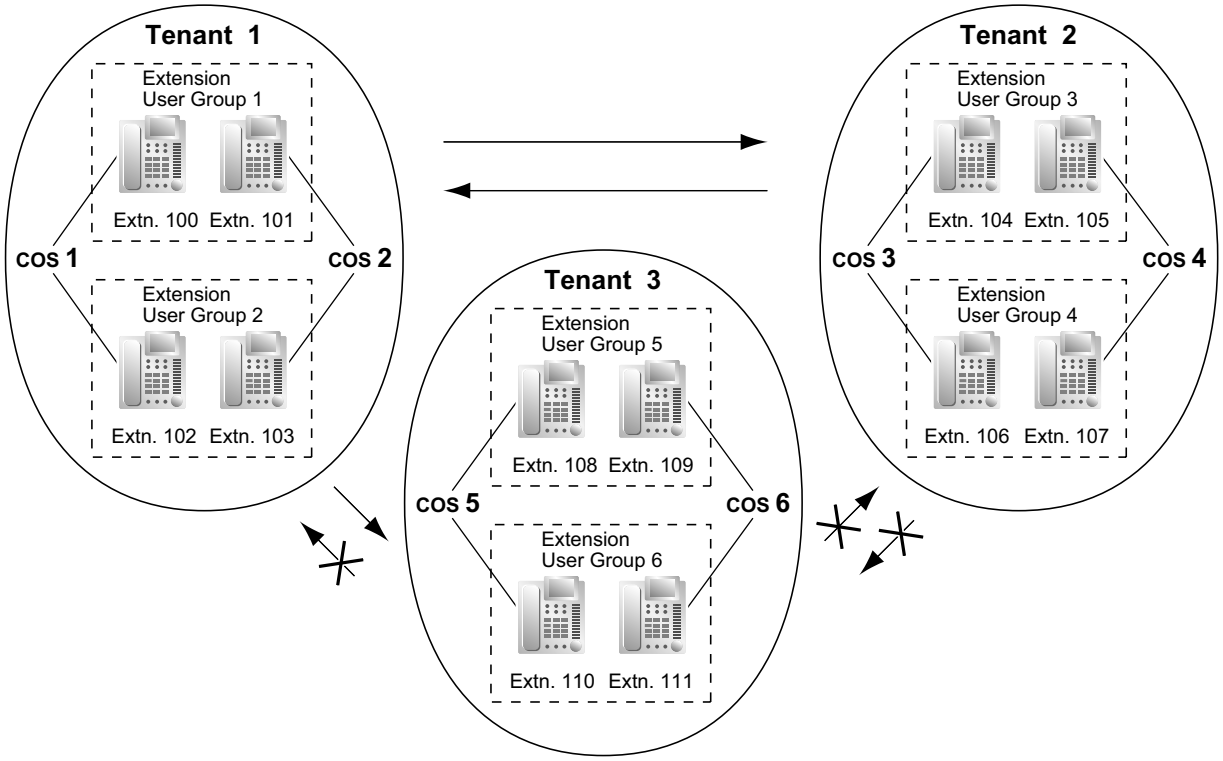
Conditions

- **ARS Assignment**
  When "**`On for Local Access Operation`**" or "**`On for Any CO Access Operation`**" is assigned as the ARS Mode for a tenant, only a subset of the ARS Leading Number Table is applied to that tenant's outgoing calls. Tenants 1 to 8 are assigned a range of 50 of the entries in the Leading Number Table as follows:
  – Tenant 1: Entries 1 to 50
  – Tenant 2: Entries 51 to 100
  – Tenant 3: Entries 101 to 150
  :
  – Tenant 8: Entries 351 to 400
  If "**`Same as System Setting`**" is selected, then all 1000 entries in the table will be applied to that tenant's outgoing calls. All 1000 entries in the table are applied when ARS is enabled, regardless of the tenant's ARS Mode.
  By dividing tenants, specific ARS settings can be applied to specific tenants according to the requirements of each tenant.
  The following example illustrates how the ARS Leading Number Table is applied to tenants:

Tenant No.	ARS Mode	Applied ARS Entries
1	On for Local Access Operation	Entries 1 to 50
2	Off	Not applied
3	Same as System Setting (System Setting: On)	Entries 1 to 1000
4	On for Local Access Operation	Entries 151 to 200
5	Off	Not applied

- **Tenant-to-Tenant Call Block**
  The following features can be restricted based on the COS for each extension (not based on the tenant) by the Internal Call Block feature (→ 2.1.2.2 Internal Call Block):
  – Calling extensions or doorphone(s) in the restricted tenant(s)
  – Picking up calls ringing in the restricted tenant(s)
  – Retrieving a call held within the restricted tenant(s)

[Example]



[Programming Example]

Caller	Called Party						
	COS 1	COS 2	COS 3	COS 4	COS 5	COS 6	...
COS 1							...
COS 2							...
COS 3					✓	✓	...
COS 4					✓	✓	...
COS 5	✓	✓	✓	✓			...
COS 6	✓	✓	✓	✓			...
:	:	:	:	:	:	:	:

✓: Block

Explanation:

1. Assign each extension in a tenant to a certain COS number. Each tenant must have unique COS numbers.
   Tenant 1: COS 1 and COS 2
   Tenant 2: COS 3 and COS 4
   Tenant 3: COS 5 and COS 6
2. Tenant-to-Tenant Call Block enables by the Internal Call Block feature.

a. Tenant 1 (COS 1 and COS 2) can make calls to both Tenant 2 (COS 3 and COS 4) and Tenant 3 (COS 5 and COS 6) as well as Tenant 1.
b. Tenant 2 (COS 3 and COS 4) can make calls to Tenant 1 (COS 1 and COS 2) and Tenant 2.
c. Tenant 3 (COS 5 and COS 6) can make calls to Tenant 3 itself only.

- An incoming call distribution group must belong to one tenant because the following features are determined on a tenant basis (→ 2.2.2.1 Incoming Call Distribution Group Features—SUMMARY):
  – Music on Hold while a call is waiting in the queue
  – The Time Table which determines the overflow destination

PC Programming Manual References

10.2 PBX Configuration—[2-2] System—Operator & BGM
→◆ PBX Operator—Day, Lunch, Break, Night
→◆ BGM and Music on Hold—Music on Hold
10.4 PBX Configuration—[2-4] System—Week Table
10.5 PBX Configuration—[2-5] System—Holiday Table
10.7.3 PBX Configuration—[2-7-3] System—Class of Service—Internal Call Block
11.2 PBX Configuration—[3-2] Group—User Group
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Main—◆ Tenant Number
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ User Group
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main—◆ User Group
13.1 PBX Configuration—[5-1] Optional Device—Doorphone—◆ Tenant Number
13.5 PBX Configuration—[5-5] Optional Device—External Sensor—◆ Tenant No.
14.1 PBX Configuration—[6-1] Feature—System Speed Dial
14.6 PBX Configuration—[6-6] Feature—Tenant
Section 16 PBX Configuration—[8] ARS
16.5 PBX Configuration—[8-5] ARS—Carrier—Authorisation Code for Tenant
18.2 PBX Configuration—[10-2] CO & Incoming Call—DIL Table & Port Settings—DIL—◆ Tenant Number
18.3 PBX Configuration—[10-3] CO & Incoming Call—DDI / DID Table—◆ Tenant Number

PT Programming Manual References

[001] System Speed Dialling Number
[006] Operator Assignment
[320] ARS Mode
[711] Music on Hold

Feature Guide References

6.1 Capacity of System Resources

5.1.4 Time Service

Description

This PBX supports day, night, lunch, and break modes of operation. TRS/Barring can be arranged separately. The destination of incoming calls can be set differently for each mode.

1. **Time Service Switching Mode**
   Day/lunch/break/night mode can be switched either automatically or manually. The switching mode can be assigned for each tenant.
   The switching mode can also be changed by pressing the Time Service Switching Mode (Automatic/Manual) button. This can be performed by only an extension assigned as the manager, or preprogrammed extension on a COS basis.

Type	Description
Automatic	The PBX will switch mode according to the preprogrammed Time Table.
Manual	A manager, or preprogrammed extension on a COS basis can switch mode by dialling the feature number or pressing the Time Service button.

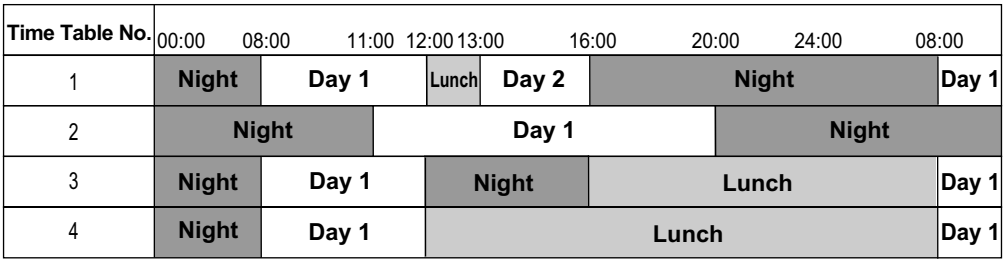
   The Unified Messaging System Manager can set the time service mode from an outside telephone. Even while in the Automatic Switching mode, day/lunch/break/night mode can be changed manually.

2. **Time Table**
   Each tenant has a Time Table used for the Automatic Switching mode. The Start and/or End time of each mode can be set for each day of the week. The Time Table numbers correspond to the tenant numbers respectively.

[Time Table Example]

Time Schedule		Time Table No. (Tenant No.)				
		1	2	3	4	…
MON	Day 1 start	08:00	11:00	08:00	08:00	…
	Lunch start	12:00	NONE	16:00	12:00	…
	Day 2 start	13:00	NONE	NONE	NONE	…
	Break 1 start	NONE	NONE	NONE	NONE	…
	Break 1 end (Day restart)	NONE	NONE	NONE	NONE	…
	Night start	16:00	20:00	12:00	NONE	…
TUE	Day 1 start	08:00	11:00	08:00	08:00	…
	Lunch start	12:05	NONE	13:00	13:00	…
	Day 2 start	13:00	NONE	NONE	NONE	…
	Break 1 start	NONE	NONE	NONE	NONE	…
	Break 1 end (Day restart)	NONE	NONE	NONE	NONE	…
	Night start	16:31	20:00	17:00	NONE	…
:	:	:	:	:	:	…

<Time Service Image of Monday>



3. **Features Using Time Service**
   The following features can be set in each time mode (day/lunch/break/night):
   a. Destination of incoming trunk calls (DIL/DID/DDI) (→ 2.1.1 Incoming Trunk Call Features)
   b. Destination of the Intercept Routing (→ 2.1.1.5 Intercept Routing)
   c. Queuing Time Table for incoming call distribution groups (→ 2.2.2.4 Queuing Feature)
   d. Overflow destination for incoming call distribution groups (→ 2.2.2.6 Overflow Feature)
   e. Destination of incoming doorphone calls (→ 2.18.1 Doorphone Call)
   f. PBX operator (→ 5.1.5 Operator Features)
   g. COS for TRS/Barring and for Trunk Access
   h. Outgoing Message (OGM) for Timed Reminder (→ 2.24.4 Timed Reminder)
   i. Intercept time for Intercept Routing—No Answer (→ 2.1.1.5 Intercept Routing) and for DISA Intercept Routing—No Answer (→ 2.16.1 Direct Inward System Access (DISA))
   j. Service group settings for voice mail (→ 3.2.1.39 Service Group)

[Programming Examples of DID/DDI Table and DIL Table]

DID/DDI table can be programmed for each DID/DDI number, and a tenant (Time Table) number is assigned to each DID/DDI number. DIL table can be programmed for each trunk, and a tenant (Time Table) number is assigned to each trunk.

<DID/DDI Table>

Location	DID/DDI No.	Tenant (Time Table) No.	DID/DDI Destination			
			Day	Lunch	Break	Night
001	123-4567	1	105	100 (UM)	105	100 (UM)
002	123-2468	1	102	100 (UM)	102	100 (UM)
:	:	:	:	:	:	:

<DIL Table>

Trunk No.	Tenant (Time Table) No.	DIL Destination			
		Day	Lunch	Break	Night
01	1	101	100 (UM)	101	100 (UM)
02	2	102	100 (UM)	102	100 (UM)
:	:	:	:	:	:

Explanation:

If a trunk call with a DID number (123-4567) is received at 20:00;
1. Tenant (Time Table) number 1 will be used.
2. The call is received during night mode in Time Table 1.
3. The call will be routed to extension 100 (UM Group).

4. **Holiday Mode**
   The holiday mode activates automatically using the Automatic Switching mode. Up to 24 holidays (start and end dates) can be stored, and one time mode can be selected for all holidays.
5. **Time Service Button**
   A flexible button can be customised as the following buttons:
   a. Day/Night button

b. Day/Night/Lunch button
c. Day/Night/Break button
d. Day/Night/Lunch/Break button
Each of these buttons is used for switching between modes. For example, pressing the Day/Night button switches between day and night modes. All of these buttons show the current status as follows:

Light Pattern	Status
Off	Day mode
Red on	Night mode
Green on	Lunch mode
Slow green flashing	Break mode
Slow red flashing	Holiday mode

Note

Any extension user (except extension users allowed to change the mode) can only check the current status on the display by pressing the Time Service button.

Conditions

- System programming can set the following time periods:
  – Day-1 (Day Start time)
  – Lunch (Lunch Start time)
  – Day-2 (Lunch End time)
  – Night (Night Start time)
  – Break-1 Start
  – Break-1 End (Day restart)
  – Break-2 Start
  – Break-2 End (Day restart)
  – Break-3 Start
  – Break-3 End (Day restart)
- **Time Service Switching Mode (Automatic/Manual) Button**
  A flexible button can be customised as the Time Service Switching Mode (Automatic/Manual) button.

PC Programming Manual References

10.4 PBX Configuration—[2-4] System—Week Table
10.5 PBX Configuration—[2-5] System—Holiday Table
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Time Service (Day / Lunch / Break / Night) Switch
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Manager—◆ Time Service Switch
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button
→◆ Type
→◆ Parameter Selection (for Time Service)
→◆ Parameter Selection (for Time Service - Automatic/Manual)
→◆ Optional Parameter (Ringing Tone Type Number) (for Time Service)
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button
→◆ Type

→◆ Parameter Selection (for Time Service)
→◆ Parameter Selection (for Time Service - Automatic/Manual)
→◆ Optional Parameter (or Ringing Tone Type Number) (for Time Service)

Feature Guide References

2.7.1 Toll Restriction (TRS)/Call Barring (Barring)
2.21.2 Flexible Buttons
3.2.1 System Features
5.1.1 Class of Service (COS)
5.1.3 Tenant Service
6.1 Capacity of System Resources

PT Programming Manual References

[101] Time Service Switching Mode
[102] Time Service Starting Time
[514] Time Service Manual Switching

User Manual References

1.9.10 Checking the Time Service Status
2.1.2 Time Service Mode Control
2.2.1 System Manager Features—◆◆ Remote Time Service Mode Setting

5.1.5 Operator Features

Description

Any extension or Incoming Call Distribution (ICD) group can be designated as an operator.
This PBX supports the following types of operators:

Type	Description
PBX operator	An extension or incoming call distribution group can be assigned as a PBX operator for each time mode (day/lunch/break/night).
Tenant operator	An extension or incoming call distribution group can be assigned as a tenant operator. The tenant operator may be the extension or incoming call distribution group of another tenant. **[Example]** Extension 110 in tenant 1 is the tenant operator of tenant 3.

Operator Call:

An extension user can call an operator by dialling the preprogrammed Operator Call feature number. The destination of the Operator Call depends on the following:

– If the Tenant Service is not in use:
The call is directed to the PBX operator according to the corresponding time mode.
– If the Tenant Service is in use:
The call is directed to the extension's tenant operator. If a tenant operator is not assigned, the call is directed to the PBX operator. In this case, the current time mode of the extension's tenant is used to determine the PBX operator that the call is directed to.

If neither a tenant operator nor a PBX operator is assigned, the caller will hear a reorder tone.

Conditions

- A single extension or incoming call distribution group can be assigned as both a tenant operator and the PBX operator.
- Tenant operators can be assigned individually for multiple tenants.

PC Programming Manual References

10.2 PBX Configuration—[2-2] System—Operator & BGM—◆ PBX Operator—Day, Lunch, Break, Night
14.6 PBX Configuration—[6-6] Feature—Tenant—◆ Operator (Extension Number)

PT Programming Manual References

[006] Operator Assignment

Feature Guide References

5.1.3 Tenant Service
6.1 Capacity of System Resources

User Manual References

1.2.1 Basic Calling
1.12.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)

5.1.6 Manager Features

Description

An extension assigned as the manager (manager extension) is allowed to use the specified features. COS programming determines the extensions which can use the following manager features:

Feature		Description & Reference	Manager Password
Manager Programming	**Manager Password Change**	Changes the manager password.	Required
	Call Charge Management	Sets, displays, clears, and prints the call charge data. → 2.22.3 Call Charge Services	Required
	Verification Code Personal Identification Number (PIN) Set	Sets a verification code PIN for each verification code. → 2.7.6 Verification Code Entry	Required
	Remote PIN Clear	Clears the extension PIN of an extension remotely, and a verification code PIN. PIN Lock is also unlocked. → 2.24.1 Extension Personal Identification Number (PIN) → 2.7.6 Verification Code Entry	Required
	Remote Extension Dial Lock	Sets or cancels the Extension Dial Lock on an extension remotely. → 2.7.3 Extension Dial Lock	Required
Dial Tone Transfer		Changes the TRS/Barring level of the extension temporarily. **[Example]** An extension user can call a manager to release the restricted outgoing call (e.g., international call). → 2.7.4 Dial Tone Transfer	Not required
Outgoing Message (OGM)		Records and plays back outgoing messages (OGMs). → 2.30.2 Outgoing Message (OGM)	Not required
Time Service		Switches the time mode (day/lunch/break/night) manually. → 5.1.4 Time Service	Not required
BGM—External		Sets the External BGM on and off. → 2.30.1 Background Music (BGM)	Not required

Feature	Description & Reference	Manager Password
Trunk Busy Out Clear	Clears the Busy Out status of a trunk. → 2.5.4.6 Trunk Busy Out	Not required
NDSS Monitor Release	Removes the monitor function from an NDSS button. → 4.2.5.1 Network Direct Station Selection (NDSS)	Not required

Conditions

CAUTION

There is a risk that fraudulent telephone calls will be made if a third party discovers a personal identification number (PIN) (verification code PIN 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.

- **Manager Password**
  One manager password can be assigned per PBX.

PC Programming Manual References

10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Manager—◆ Manager
19.1 PBX Configuration—[11-1] Maintenance—Main—Password—◆ Manager Password - PT Programming—Prog *1

PT Programming Manual References

[112] Manager Password
[511] Manager Assignment

Feature Guide References

5.1.1 Class of Service (COS)

User Manual References

2.1 Control Features
4.1.2 Manager Programming

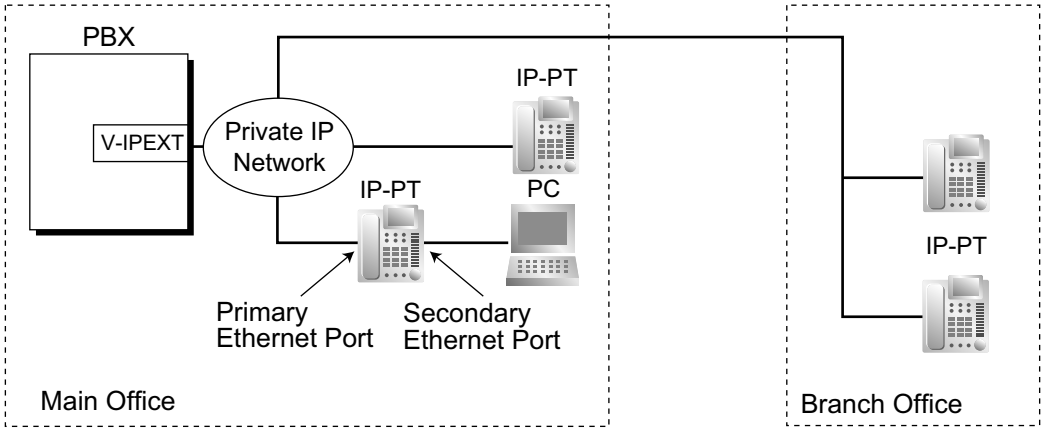
5.2 System Configuration—Extensions

5.2.1 IP Proprietary Telephone (IP-PT)

Description

IP Proprietary Telephones (IP-PTs) are telephones that make and receive calls using IP and that connect to the PBX over a LAN. Except for their use of IP, they function almost identically to traditional proprietary telephones.
IP-PTs have two Ethernet ports for connection, primary and secondary. They are connected to the PBX through a network hub or other splitting device, and can have a PC connected to the secondary Ethernet port.

[Connection Example]



Conditions

- IP-PT registration is required through system programming before an IP-PT can be used with the PBX. An IP-PT cannot be used unless an extension number is assigned. However, depending on system programming, registration may occur completely automatically, or may require only inputting the desired extension number.
  For details on how to register IP-PTs, refer to the Installation Manual.
- The KX-NT265 does not have a secondary Ethernet port.
- The following optional devices are available for the KX-NT300 series (except KX-NT321) IP-PT:
  – KX-NT307(PSLP1528) Bluetooth Module
  – KX-NT303 Add-on 12 Key Module (not available for KX-NT366)
  – KX-NT305 Add-on 60 Key Module (not available for KX-NT366)
- The following optional devices are available for the KX-NT553/KX-NT556 IP-PT:
  – KX-NT505 Add-on 48 Key Module (maximum four units connectable).
- **DSP Resource Usage**
  Making a call from an IP-PT requires a certain number of DSP resources, depending on the codec used. If all DSP resources are in use, this operation cannot be performed. To ensure a minimum level of performance, DSP resources can be reserved for VoIP communication. (→ 5.5.4 DSP Resource Usage) However, DSP resources are not required for P2P calls. (→ 5.2.3 Peer-to-Peer (P2P) Connection)
- **KX-NT307(PSLP1528) Bluetooth Module**
  A Bluetooth wireless headset can be registered to a KX-NT300 series (except KX-NT321) IP-PT containing the KX-NT307(PSLP1528) Bluetooth Module through personal programming. When Headset Mode is off, the Bluetooth headset can be used to answer calls or redial. In this case, Headset Mode will turn on automatically, and will turn off after you hang up.

This Bluetooth Module is also compatible with KX-DT343/KX-DT346 DPTs.

- Calls made using a Bluetooth wireless headset will not be disconnected immediately when the user wanders out of range. However, if the Bluetooth wireless headset user remains out of range for a specified time period, the call will be disconnected.
- The following features cannot be used with an IP-PT:
  – XDP
  – Digital XDP
  – OHCA
- **Automatic Rerouting to Secondary PBX**
  A KX-NT300 series or KX-NT500 series IP-PT can automatically connect to a secondary PBX, when the primary PBX becomes disconnected. When the primary PBX is connected again, the IP-PT reconnects to it.

Installation Manual References

4.4 Virtual Cards

PC Programming Manual References

9.2.2 PBX Configuration—[1-1] Configuration—Slot—System Property—Site—Main—◆ IP Terminal Registration Mode
9.13 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Card Property
9.14 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Port Property

5.2.2 SIP (Session Initiation Protocol) Extension

Description

This PBX supports the connection of SIP-compatible IP telephones (hardphones and softphones). SIP extensions make and receive calls using Internet Protocol (IP).
For information about SIP extension compatibility with feature numbers, see 5.5.7 Flexible Numbering/Fixed Numbering.
For information about Panasonic KX-UT series SIP phones, see 5.2.2.1 KX-UT Series SIP Phones.
Below are the features supported by SIP extensions:

- Absent Message (→ 2.20.2 Absent Message)
- Account Code Entry (→ 2.5.4.3 Account Code Entry)
- Automatic Route Selection (ARS) (→ 2.8 Automatic Route Selection (ARS) Features)
- Call Forwarding (FWD) (→ 2.3.2 Call Forwarding (FWD))
- Call Hold (→ 2.13.1 Call Hold)
- Call Park[*1] (→ 2.13.2 Call Park)
- Call Pickup (→ 2.4.3 Call Pickup)
- Call Transfer with Announcement (→ 2.12.1 Call Transfer)
- Call Waiting Tone (→ 2.1.3.3 Call Waiting)
- COLR/CLIR/CLIP/COLP (→ 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP))
- Conference[*2] (→ 2.14 Conference Features)
- Conference Group Call (→ 2.15.1 Conference Group Call)
- Data Line Security (→ 2.11.5 Data Line Security)
- Dial Information (CTI) (→ 2.26.1 Computer Telephony Integration (CTI))
- Direct Inward System Access (DISA) (→ 2.16.1 Direct Inward System Access (DISA))
- Door Open (→ 2.18.2 Door Open)
- Doorphone Call (→ 2.18.1 Doorphone Call)
- Emergency Call (→ 2.5.4.2 Emergency Call)
- Executive Busy Override Deny (→ 2.10.2 Executive Busy Override)
- Extension Dial Lock (→ 2.7.3 Extension Dial Lock)
- Extension Feature Clear (→ 2.24.2 Extension Feature Clear)
- Extension PIN (→ 2.24.1 Extension Personal Identification Number (PIN))
- External BGM On/Off (→ 2.30.1 Background Music (BGM))
- FWD/DND (→ 2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features)
- Group FWD (→ 2.3.2 Call Forwarding (FWD))
- Idle Line Access (→ 2.5.5.3 Trunk Access)
- Log-in/Log-out (→ 2.2.2.7 Log-in/Log-out)
- Message Waiting (→ 2.20.1 Message Waiting)
- Not Ready (→ 2.2.2.7 Log-in/Log-out)
- Operator Call (→ 5.1.5 Operator Features)
- Paging (→ 2.17.1 Paging)
- Personal Speed Dialling (→ 2.6.4 Speed Dialling—Personal/System)
- Quick Dialling (→ 2.6.5 Quick Dialling)
- Redial (→ 2.6.3 Last Number Redial)
- Remote Station Lock (→ 2.7.3 Extension Dial Lock)
- S-CO Line Access (→ 2.5.5.3 Trunk Access)
- System Speed Dialling (→ 2.6.4 Speed Dialling—Personal/System)
- TIE Line Call (→ 4.2.1 TIE Line Service)
- Time Service (→ 5.1.4 Time Service)
- Timed Reminder (→ 2.24.4 Timed Reminder)
- Trunk Group Access (→ 2.5.5.3 Trunk Access)

- Verification Code (→ 2.7.6 Verification Code Entry)

*1 SIP extensions can retrieve parked calls but cannot park calls.
*2 As a member only (not as an originator).

SIP Video Phone

Video phone calls can be established between SIP extensions with video phone capabilities.

Conditions

[General]

- This PBX supports SIP devices that use RFC 3261, 3264, 3310, 2327, or 4028.
- Some SIP phones may not be compatible with this PBX.
- Before a SIP extension can be used with the PBX, the IP address of the mother board, password, and extension number must be assigned on the SIP extension and on the PBX. Even if the IP terminal registration mode has been set to full automatic mode or extension input mode, general SIP extensions must be registered manually.
  For details on how to register, refer to the Installation Manual.
- When registering the SIP extension, the user ID must be the extension number of the SIP extension.
- When a SIP extension uses the Call Hold feature, the target call is put on Consultation Hold.
- **DSP Resource Usage**
  Making a call from a SIP extension requires a certain number of DSP resources, depending on the codec used. If all DSP resources are in use, this operation cannot be performed. To ensure a minimum level of performance, DSP resources can be reserved for VoIP communication. (→ 5.5.4 DSP Resource Usage) However, DSP resources are not required for P2P calls. (→ 5.2.3 Peer-to-Peer (P2P) Connection)

Installation Manual References

4.4 Virtual Cards

PC Programming Manual References

9.15 PBX Configuration—[1-1] Configuration—Slot—V-SIPEXT—Card Property
9.16 PBX Configuration—[1-1] Configuration—Slot—V-SIPEXT—Port Property

5.2.2.1 KX-UT Series SIP Phones

Description

KX-UT series telephones are Panasonic SIP phones that provide tighter integration with the PBX than general SIP phones. The following features are available:

Automatic PBX registration

Like IP-PTs (→ 5.2.1 IP Proprietary Telephone (IP-PT)), registration of KX-UT series SIP phones can be made to occur automatically through system programming.

Phonebook integration

A KX-UT series SIP phone will automatically download up to 100 personal speed dialling numbers and up to 300 system speed dialling numbers (→ 2.6.4 Speed Dialling—Personal/System) and store this information in its local phonebook.

Note

A KX-UT series SIP phone downloads the system speed dialling numbers from the KX-NS300 to the local phonebook. The KX-NS300 has two kinds of memory for system speed dialling numbers, they are the basic memory and the expansion for tenant. A KX-UT series downloads the system speed dialling numbers from the basic memory of the KX-NS300. The expansion for tenant of the KX-NS300 will not be downloaded. About the basic memory and the expansion for tenant, refer to the PC Programming Manual14.1 PBX Configuration—[6-1] Feature—System Speed Dial—◆ Select Table.
System speed dial numbers do not download immediately; they download when the telephone is rebooted. The number of system speed dialling numbers that are downloaded from the KX-NS300 can be changed. Refer to 9.2.1 PBX Configuration—[1-1] Configuration—Slot—System Property—Main—Main—◆ System Speed Dial Download For UT Extensions in the PC Programming Manual.

Customisable flexible buttons

The following features can be assigned through system programming to the flexible buttons available on some KX-UT series SIP phones:

Button	Usage
Single-CO (S-CO)	Used to access a specified trunk for making or receiving calls.
DN*1	A type of button specific to KX-UT series SIP phones. For details, refer to the telephone's documentation.
One-Touch*1	Used to dial a specified number or feature. For details, refer to the telephone's documentation.
Headset*1	Used to turn on/off the headset mode while idle. (→ 2.11.4 Headset Operation)
Login/Logout*2	Used to switch between log-in and log-out mode. (→ 2.2.2.7 Log-in/Log-out)
Contact (DSS)	Access another extension with one touch. Unlike a One-Touch button, this type of button cannot be used to dial feature numbers.
Wrap-up*2	Used to switch between the Wrap-up/Not Ready and Ready modes. (→ 2.2.2.7 Log-in/Log-out)

Button	Usage
Call Park	Place the current call into a parking zone of the PBX. (→ 2.13.2 Call Park) There are two modes: • Specific: Place the call into the specified park zone. • Automatic: The telephone searches for an idle park zone from among those assigned to its own flexible buttons.

*1 This feature is not controlled from the PBX.
*2 Not available on all KX-UT series SIP phones.

Conditions

- **Requirement:**
  A V-UTEXT card is required to use a KX-UT series SIP phone.
- Even if phonebook entries downloaded from the PBX are edited, the phonebook entries in the PBX are not updated.
- Only entries in the PBX directories that have phone numbers assigned will be downloaded.
- On KX-UT248 and KX-UT670 phones, the labels displayed on flexible buttons can be customised through system programming.
  → 12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Label Name
- Incoming/Outgoing call log information is stored on the phone, not on the PBX.
- For detailed explanations about a particular KX-UT series SIP phone, refer to the telephone's documentation.
- **DSP Resource Usage**
  Making a call from a SIP extension requires a certain number of DSP resources, depending on the codec used. If all DSP resources are in use, this operation cannot be performed. To ensure a minimum level of performance, DSP resources can be reserved for VoIP communication. (→ 5.5.4 DSP Resource Usage) However, DSP resources are not required for P2P calls. (→ 5.2.3 Peer-to-Peer (P2P) Connection)
- **Phonebook integration**
  Up to 100 entries can be stored in the phonebook of the KX-UT113.

Installation Manual References

4.4 Virtual Cards

PC Programming Manual References

9.2.1 PBX Configuration—[1-1] Configuration—Slot—System Property—Main—Main—◆ System Speed Dial Download For UT Extensions
9.2.2 PBX Configuration—[1-1] Configuration—Slot—System Property—Site—Main—◆ IP Terminal Registration Mode
9.19 PBX Configuration—[1-1] Configuration—Slot—V-UTEXT—Card Property
9.20 PBX Configuration—[1-1] Configuration—Slot—V-UTEXT—Port Property
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button

5.2.2.2 SIP Portable Station (S-PS) and SIP Cell Station (SIP-CS)

Description

This PBX supports the connection of SIP Cell Stations (SIP-CSs). A SIP-CS functions like other CSs, except it uses SIP for managing calls. A PS that connects through a SIP-CS is called a SIP Portable Station (S-PS). This section explains the main features available to SIP-CSs and S-PSs.

Phonebook

An S-PS will automatically download up to 300 system speed dialling numbers (→ 2.6.4 Speed Dialling—Personal/System) and store them in its local phonebook.

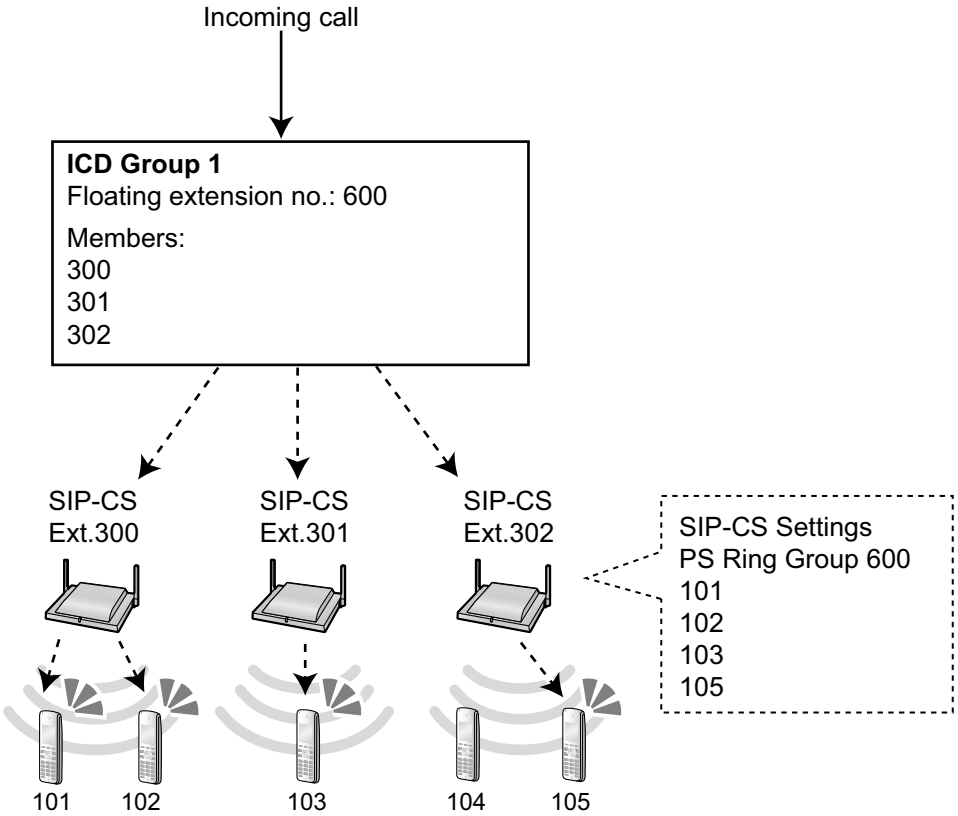
Wireless XDP

An S-PS can be paired with a wired extension. However, only parallel ringing is supported. Other features, such as switching a call between telephones, are not supported. (→ 5.2.4.5 Wireless XDP Parallel Mode)

S-PS Broadcast Call

An S-PS cannot be a member of a PS Ring Group (→ 5.2.4.2 PS Ring Group). Instead, an ICD group can be used to achieve the same functionality.

1. In the ICD group settings, specify the SIP-CSs as the members of an ICD group.
2. Via the Super Master SIP-CS's Web interface, create a PS Ring Group that contains the floating extension number of the ICD group and the S-PSs to broadcast incoming calls to.



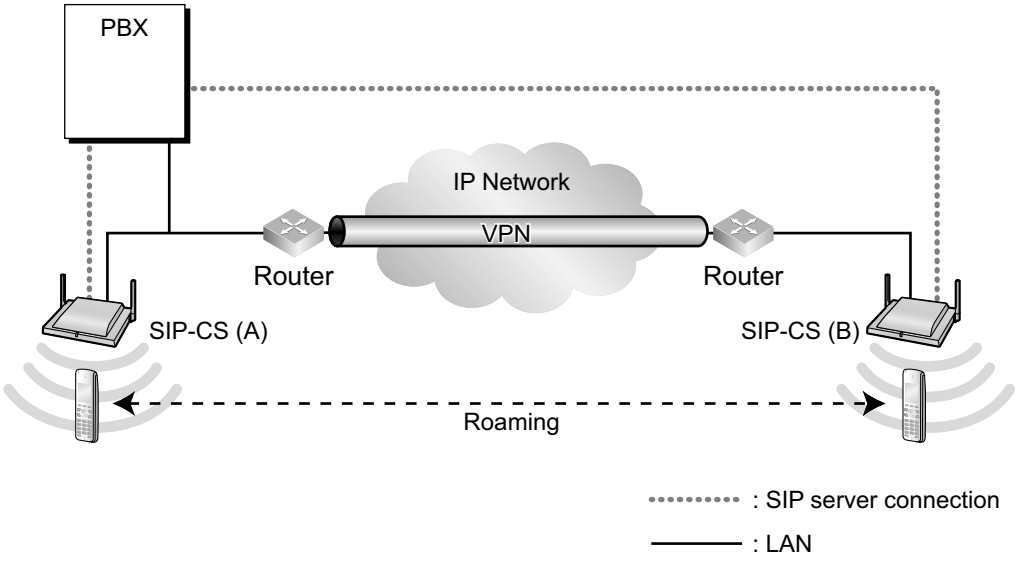
Flexible Keys

The following types of flexible keys can be programmed on an S-PS:

Button	Usage
Single-CO (S-CO)	Used to access a specified trunk for making or receiving calls.
DN	A type of button specific to SIP phones. For details, refer to the telephone's documentation.
One-Touch	Used to dial a specified number or feature. For details, refer to the telephone's documentation.
Login/Logout	Used to switch between log-in and log-out mode. (→ 2.2.2.7 Log-in/ Log-out)
Contact (DSS)	Access another extension with one touch. Unlike a One-Touch button, this type of button cannot be used to dial feature numbers.
Wrap-up	Used to switch between the Wrap-up/Not Ready and Ready modes. (→ 2.2.2.7 Log-in/Log-out)
Call Park	Place the current call into a parking zone of the PBX. (→ 2.13.2 Call Park) There are two modes: • Specific: Place the call into the specified park zone. • Automatic: The telephone searches for an idle park zone from among those assigned to its own flexible buttons.

Conditions

- S-PSs can operate only under a SIP-CS. They are not compatible with other types of CSs (e.g., IP-CSs). Also, other types of PSs cannot operate under a SIP-CS.
- SIP-CSs can operate under the same KX-NS300 as other types of CSs (e.g., IP-CS) so long as each type's wireless range does not overlap with the other.
- To allow roaming to a remote location, a SIP-CS must be installed at the remote location, and the SIP-CS's SIP server must be the same PBX to which the S-PS is registered.
  For example, in the following figure if the S-PS is registered to the PBX, it can roam between SIP-CS (A) and SIP-CS (B) because they both use the PBX as their SIP server.



- **DSP Resource Usage**

Making a call from a PS requires a certain number of DSP resources, depending on the codec used. If all DSP resources are in use, this operation cannot be performed. To ensure a minimum level of performance, DSP resources can be reserved for VoIP communication. (→ 5.5.4 DSP Resource Usage)

[S-PS Broadcast Call]

Note

For the following conditions, "ICD group" refers to an ICD group that is configured to distribute calls to SIP-CSs. These conditions do not necessarily apply to ICD groups in general.

- It is possible to specify extensions other than SIP-CSs as ICD group members.
- If there is an incoming call at an ICD group and a new incoming call arrives at the same ICD group, the new call will be placed in the group's queue. (→ 2.2.2.4 Queuing Feature)
- If there is an incoming call at an ICD group and a new incoming call arrives at a different ICD group, up to 1 call can be queued at the SIP-CS.
- If an S-PS is receiving an incoming broadcast call but then receives an individual call (a call directly to its extension number), the individual call will be given precedence.
- If an S-PS is receiving an incoming broadcast call but then receives a group call from an ICD group of which the S-PS is a member, the group call will be given precedence.
- A supervisor extension in the ICD group can control the log-in/log-out status of a SIP-CS in the group. However, a SIP-CS cannot change its own log-in/log-out status. (→ 2.2.2.8 Supervisory Feature, → 2.2.2.7 Log-in/Log-out)
- Automatic Log-out does not apply to SIP-CSs. (→ 2.2.2.7 Log-in/Log-out)
- A supervisor extension in the ICD group cannot control the wrap-up status of a SIP-CS in the group, nor can a SIP-CS change its own wrap-up status.
- A Wrap-up button configured on an S-PS controls the PS's wrap-up status. It cannot be used to control a SIP-CS's wrap-up status.

PC Programming Manual References

7.12 Utility—CS-Web Connection
9.20 PBX Configuration—[1-1] Configuration—Slot—V-UTEXT—Port Property—KX-UT Series SIP Phone, S-PS, and SIP-CS Registration and De-registration
9.20 PBX Configuration—[1-1] Configuration—Slot—V-UTEXT—Port Property—Main—◆ Telephone Type
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ Telephone Type
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button

Feature Guide References

2.2.2 Incoming Call Distribution Group Features

5.2.2.3 Simple Remote Connection

Description

Using the Built-in Media Relay Gateway

Using the Built-in Media Relay Gateway feature, KX-NT500 series IP-PTs can be connected to a remote location without an additional device, such as an SBC (Session Border Controller). KX-NT500 series IP-PTs can be registered to the remote location after configuring the Built-in Media Relay Gateway. For the KX-NS300, KX-UT-series SIP phones and general SIP phones can also be installed to the remote location easily. KX-UT SIP series phones and general SIP phones can be connected to the Built-in Media Relay Gateway by following the method for using an SBC.

Note

- Peer-to-peer communication is not supported for the built-in media relay gateway.

Using an SBC (Session Border Controller)

KX-UT SIP phones and general SIP phones support simple remote connectivity when the KX-NS300 is networked with an SBC (session border controller). Simple remote connectivity means that even if the SIP phone is located behind a NAT router, firewall, or both, specialised settings such as NAT traversal settings do not need to be configured for each remote extension.
There are 2 scenarios for configuring and connecting a SIP phone:

a. The SIP phone is connected and registered to the PBX on the PBX's local network. The necessary settings are configured automatically by the PBX.

b. The remote IP settings of the SIP phone are configured without first connecting the phone to the PBX. Once programmed, the SIP phone is sent to the remote location, connected to the network and will automatically connect to the PBX.

Conditions

- Extensions that will be configured remotely use HTTPS for transferring the configuration file. However, a maximum of 20 extensions per site can be connected in this way.
- The following types of settings must be configured on the PBX:
  – The remote setting for the SIP phone's port
  – The outside-facing IP address and port of the PBX-side network gateway
  – The necessary NAT traversal settings for the NAT device
- The following types of setting must be configured on the PBX-side network gateway:
  When using an SBC (Session Border Controller)
  – Static port forwarding settings for traversing the SBC (SIP, TR-069, RTP, and NTP)
  For using the built-in Media Relay Gateway
  – Static port forwarding settings for traversing the built-in Media Relay Gateway (SIP, TR-069, RTP, NTP, PTAP, and MGCP)
- General SIP phones must support early media.
- SIP-CSs and S-PSs do not support this feature.

Installation Manual References

5.8.3 Installing SIP Phones at a Remote Site
5.8.4 Installing IP Phones at a Remote Site with a Built-in Media Relay Gateway

PC Programming Manual References

9.2.2 PBX Configuration—[1-1] Configuration—Slot—System Property—Site—SIP Extension
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - CWMP Server IP Address
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - CWMP Server (HTTP) Port No.
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - CWMP Server (HTTPS) Port No.
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - SIP-MLT Data Download Server (HTTP) Port No.
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - SIP-MLT Data Download Server (HTTPS) Port No.
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - SIP Proxy Server IP Address
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - SIP Proxy Server Port No.
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - NTP Server IP Address
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - NTP Server Port No.
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - Keep Alive Packet Type
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - Keep Alive Packet Sending Interval Time (s)
→◆ Setting parameters assigned to Remote SIP-MLT—NAT - SIP Register Expire Time (s)
→◆ Setting parameters for Networking Survivability, assigned to Remote SIP-MLT—NAT - CWMP Server IP Address
→◆ Setting parameters for Networking Survivability, assigned to Remote SIP-MLT—NAT - CWMP Server (HTTP) Port No.
→◆ Setting parameters for Networking Survivability, assigned to Remote SIP-MLT—NAT - CWMP Server (HTTPS) Port No.
→◆ Control Condition of Remote SIP-MLT—PERIODIC Ability
→◆ Control Condition of Remote SIP-MLT—PERIODIC Packet Sending Interval Time (s)
9.2.2 PBX Configuration—[1-1] Configuration—Slot—System Property—Site—Media Relay
→◆ Common—NAT - External IP Address
→◆ IP Extension—NAT - MGCP Server Port No.
→◆ IP Extension—Keep Alive Packet Type
→◆ IP Extension—Keep Alive Packet Sending Interval Time (s)
→◆ SIP Extension / UT Extension—NAT - SIP Proxy Server Port No.
→◆ UT Extension—NAT - CWMP Server (HTTP) Port No.
→◆ UT Extension—NAT - CWMP Server (HTTPS) Port No.
→◆ UT Extension—NAT - CWMP Server (HTTP) Port No. for Network Survivability
→◆ UT Extension—NAT - CWMP Server (HTTPS) Port No. for Network Survivability
→◆ UT Extension—NAT - SIP-MLT Data Download Server (HTTP) Port No.
→◆ UT Extension—NAT - SIP-MLT Data Download Server (HTTPS) Port No.
→◆ UT Extension—NAT - NTP Server Port No.
→◆ UT Extension—Keep Alive Packet Type
→◆ UT Extension—Keep Alive Packet Sending Interval Time (s)
→◆ UT Extension—PERIODIC Ability
→◆ UT Extension—PERIODIC Packet Sending Interval Time (s)
→◆ Option—NAT - RTP IP Address
→◆ Option—NAT - SIP Proxy Server IP Address
→◆ Option—NAT - CWMP Server IP Address
→◆ Option—NAT - CWMP Server IP Address for Network Survivability
→◆ Option—NAT - NTP Server IP Address
9.14 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Port Property—Remote Place
9.16 PBX Configuration—[1-1] Configuration—Slot—V-SIPEXT—Port Property—Remote Place
9.20 PBX Configuration—[1-1] Configuration—Slot—V-UTEXT—Port Property—Remote Place

5.2.3 Peer-to-Peer (P2P) Connection

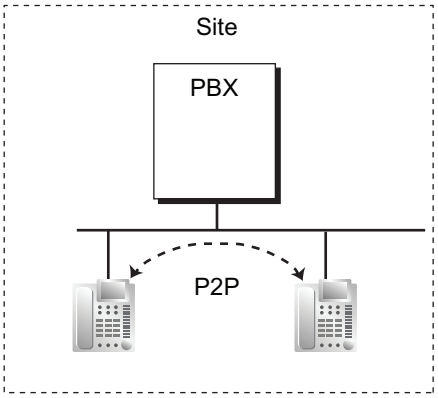
Description

This PBX automatically establishes peer-to-peer communication between peer-to-peer compatible IP extensions (i.e., IP-PTs and SIP extensions) and SIP Trunks that belong to the same P2P group. With peer-to-peer calls, the call is routed directly from one IP extension to another without going through a DSP card, which means that P2P calls are established without using the PBX's resources.

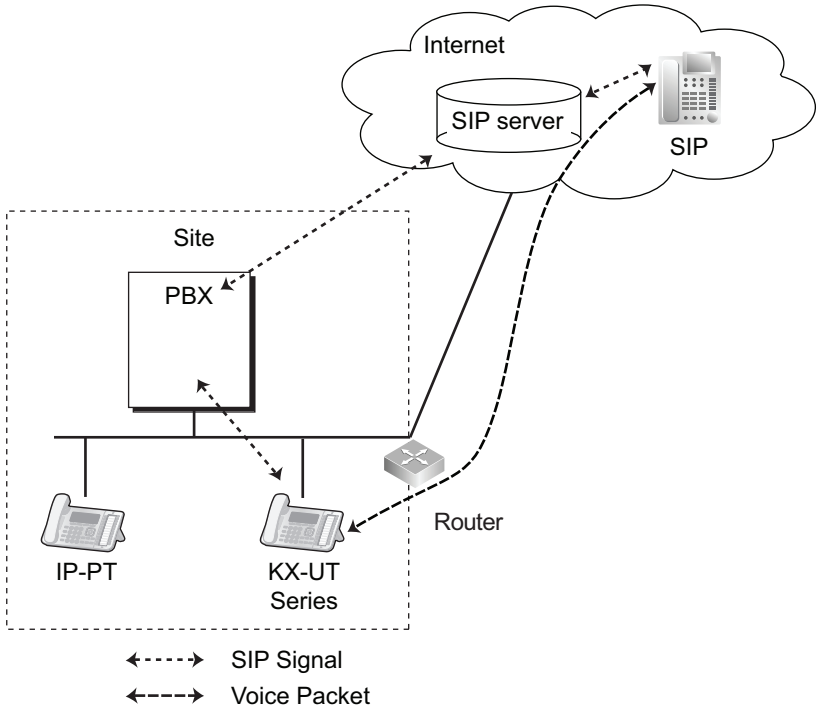
P2P Scenarios

P2P calls can be established between telephones in a site. The following illustrations show P2P calls established in a variety of network configurations. It is assumed that all the devices are connected over a private IP network.

P2P connection within a site

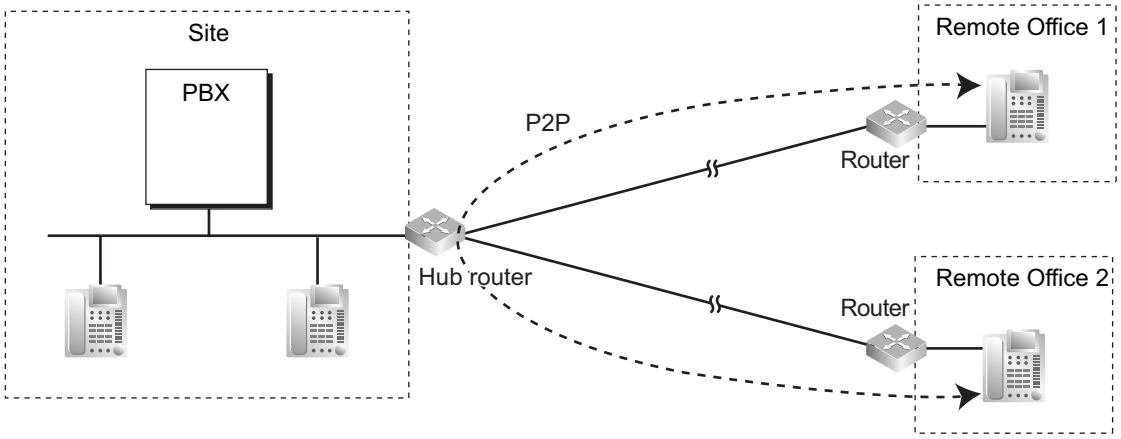


Calls between IP telephones within a site establish a P2P connection.



Example: P2P connection between an IP extension and IP trunk within a site
Voice packets are transmitted and received between the IP extension and the IP trunk within a site.

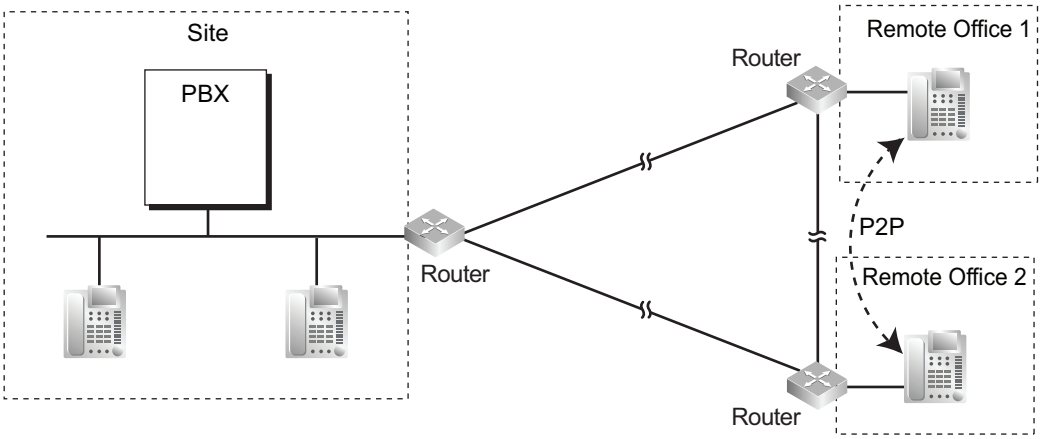
P2P connection between branch offices over a VPN (hub-and-spoke connection)[*1]



Calls between IP telephones in separate branch offices establish a P2P connection, using the hub router at Site to communicate over a VPN.

[*1] When the VPN uses a hub-and-spoke (star) topology, the hub router must be configured to allow U-turn connections.

P2P connection between branch offices over a VPN (mesh connection)



Calls between IP telephones in separate branch offices establish a P2P connection directly through the VPN routers at each branch.

Conditions

[General]

- Three codecs are used for peer-to-peer calls: G.722, G.711, and G.729A. The speech quality of the codecs varies as follows: (High) G.722, G.711, G.729A (Low).
  When the preferred codec of each party differs, the call will be established using the lower codec. For example, if the caller prefers G.711 while the called party prefers G.729A, the call will be established using G.729A.
- G.722 is only available for calls between KX-NT300 series IP-PTs, KX-NT500 series IP-PTs, and some SIP extensions that support this codec during peer-to-peer communication.
- Through system programming, it is possible to assign the preferred codec to use for IP-PTs and KX-UT series SIP phones.
- For non-KX-UT series SIP extensions, the priority of the codec that will be used can be specified via the telephone itself.
- For non-peer-to-peer calls via the DSP card, calls cannot be made or received when all of the card's resources are being used.
- KX-UT series SIP phones and general SIP phones support P2P communication over SIP trunks.
- KX-NT500 series IP-PTs support P2P communication over SIP trunks. Contact your dealer for the compatible software version for each KX-NT500 series IP-PT. Other IP-PTs do not support this feature.
- This PBX supports H.263/H.264 codecs for P2P video communication.
- P2P communication using the T.38 protocol is supported for calls between IP extensions or IP extensions and IP trunks. However, only SIP extensions and SIP trunks can be used.

[P2P Groups]

- Telephones or SIP Trunks must belong to the same P2P group to establish a P2P connection. Telephones or SIP Trunks can be assigned to the same P2P group.
- Calls between IP extensions in different P2P groups are established via the DSP card in "DSP-through mode". DSP-through mode is where only 1 DSP resource is required, regardless of the codec being used.
- Calls between extensions or SIP Trunks in different P2P groups will consume DSP resources.
- To activate P2P communication between IP extensions and IP trunks, set **IP Extension - SIP Trunk P2P** to enable (→ 10.9 PBX Configuration—[2-9] System—System Options—Option 8—◆ Extension - Trunk P2P—IP Extension - SIP Trunk P2P).
- The PBX configures P2P group settings for each SIP trunk port, and P2P communication is only allowed when both parties belong to the same P2P group.

- In an environment where P2P communication between IP extensions and IP trunks is not allowed due to provider restrictions, set **IP Extension - SIP Trunk P2P** to disable, or assign different P2P group settings between callers.

Note

During P2P communication, DTMF signals cannot be detected by the PBX since sound RTP packets are processed directly between IP extensions and IP trunks. As a result, features that use DTMF detection (e.g., DISA Call Transfer to Outside User) cannot be used. To use such features, change the settings to allow for non-P2P calls, to create an environment where DTMF detection is available. Confirm with your ITSP whether there are services which use DTMF detection and whether P2P communication is possible.

Installation Manual References

4.3.3 DSP S Card (KX-NS5110)

PC Programming Manual References

9.2.2 PBX Configuration—[1-1] Configuration—Slot—System Property—Site—VoIP-DSP Options
9.10 PBX Configuration—[1-1] Configuration—Slot—V-SIPGW—Port Property—Main—◆ P2P Group
9.10 PBX Configuration—[1-1] Configuration—Slot—V-SIPGW—Port Property—Main—◆ P2P Group Name
9.14 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Port Property—Main—◆ P2P Group
9.14 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Port Property—Main—◆ P2P Group Name
9.14 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Port Property—Option—◆ IP Codec Priority
9.16 PBX Configuration—[1-1] Configuration—Slot—V-SIPEXT—Port Property—Main—◆ P2P Group
9.16 PBX Configuration—[1-1] Configuration—Slot—V-SIPEXT—Port Property—Main—◆ P2P Group Name
9.16 PBX Configuration—[1-1] Configuration—Slot—V-SIPEXT—Port Property—FAX/T.38
9.20 PBX Configuration—[1-1] Configuration—Slot—V-UTEXT—Port Property—Main—◆ P2P Group
9.20 PBX Configuration—[1-1] Configuration—Slot—V-UTEXT—Port Property—Main—◆ P2P Group Name
9.20 PBX Configuration—[1-1] Configuration—Slot—V-UTEXT—Port Property—Option—◆ UT Codec Priority - 1st—UT Codec Priority - 4th
10.9 PBX Configuration—[2-9] System—System Options—Option 8—◆ Extension - Trunk P2P—IP Extension - SIP Trunk P2P

Feature Guide References

5.2.1 IP Proprietary Telephone (IP-PT)
5.1.2 Group

5.2.4 Portable Station (PS) Features

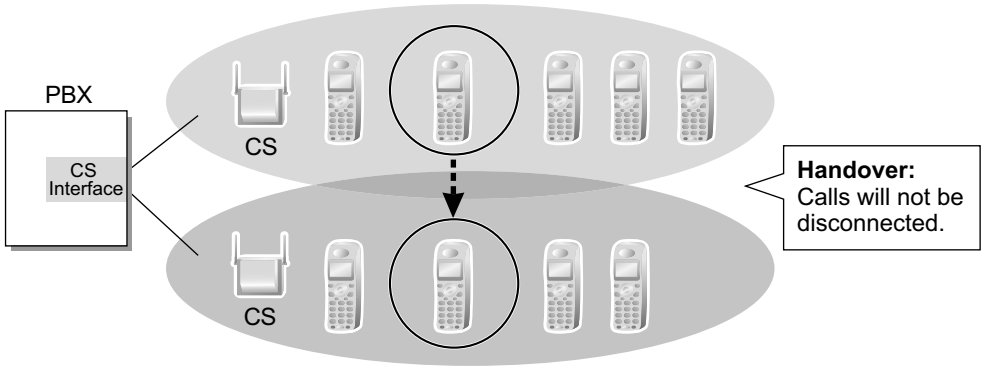
5.2.4.1 Portable Station (PS) Connection

Description

This PBX supports the connection of a PS. Cell Stations (CSs) allow PSs to receive reception within a designated area. It is possible to use the PBX features using the PS like a PT.

Conditions

- For details specific to SIP-based CSs and PSs, see "5.2.2.2 SIP Portable Station (S-PS) and SIP Cell Station (SIP-CS)".
- The PS registration is required through the system programming. To avoid unexpected registration to another PBX, the Personal Identification Number (PIN) for the PBX is necessary to register a PS. The registration can be cancelled.
- **Handover**
  Even if a PS user moves during a conversation, the PS will automatically switch between cells without disconnecting the call (**Handover**).
  Handover is available in any of the following cases:
  a. During a conversation with an extension or outside party.
  b. While a call is ringing at the PS.
  c. While the PS is in idle status.



  However, Handover is not available in any of the following cases:
  a. When the new (Handover) CS is busy.
  b. When there is no CS within range.
  c. While the Live Call Screening (LCS) is activated (→ 3.2.2.16 Live Call Screening (LCS)).
  d. While the PS user is paging other extensions (→ 2.17.1 Paging).
  e. While the PS user is dialling digits to make a trunk call.
  f. If one CS is a SIP-CS and the other is not (e.g., an IP-CS).
- **DSP Resource Usage**
  Making a call through IP-CS with a PS requires a certain number of DSP resources, depending on the codec used. If all DSP resources are in use, this operation cannot be performed. To ensure a minimum level of performance, DSP resources can be reserved for VoIP communication. (→ 5.5.4 DSP Resource Usage)
- When a caller has dialled the extension number of a PS but the CS is busy, the caller hears a busy tone.
- For more information about connecting PSs to CSs, see the Quick Installation Guide for the relevant CS.
- The number of digits allowed for a PS extension number is determined by the PS model. See your PS documentation for details.

PC Programming Manual References

9.34 PBX Configuration—[1-2] Configuration—Portable Station

PT Programming Manual References

[690] PS Registration
[691] PS Termination
[692] Personal Identification Number (PIN) for PS Registration

Feature Guide References

5.5.7 Flexible Numbering/Fixed Numbering

5.2.4.2 PS Ring Group

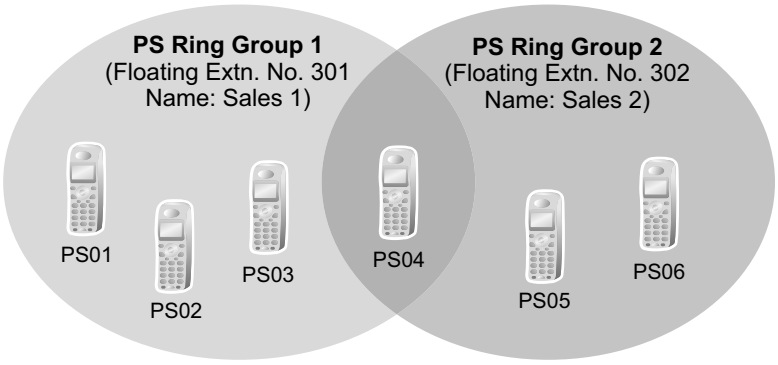
Description

A PS ring group is a group of PS extensions that receives incoming calls. Each group has a floating extension number and name. One PS can belong to multiple groups.

[Programming Example]

	PS Ring Group 01	PS Ring Group 02	PS Ring Group 03	. .
Floating Extn. No.	301	302	303	. .
Group Name	Sales 1	Sales 2	Sales 3	. .
Incoming Trunk Call Information Display	Called Party's Name/ Number	Caller's Name/ Number	Caller's Name/ Number	. .
PS01	✓			. .
PS02	✓			. .
PS03	✓			. .
PS04	✓	✓		. .
PS05		✓		. .
PS06		✓		. .
PS07			✓	. .
:	:	:	:	:

✓: Constituent



Conditions

- **PS Ring Group**
  A maximum of 32 groups can be created.
- **Compatible PSs**
  The following PSs can be assigned to PS ring groups:
  – KX-TCA155
  – KX-TCA175
  – KX-TCA185
  – KX-TCA256

– KX-TCA275
– KX-TCA285
– KX-TCA355
– KX-TCA385
– KX-TCA364

S-PSs cannot be assigned to a PS ring group. For details about S-PSs, see "5.2.2.2 SIP Portable Station (S-PS) and SIP Cell Station (SIP-CS)".

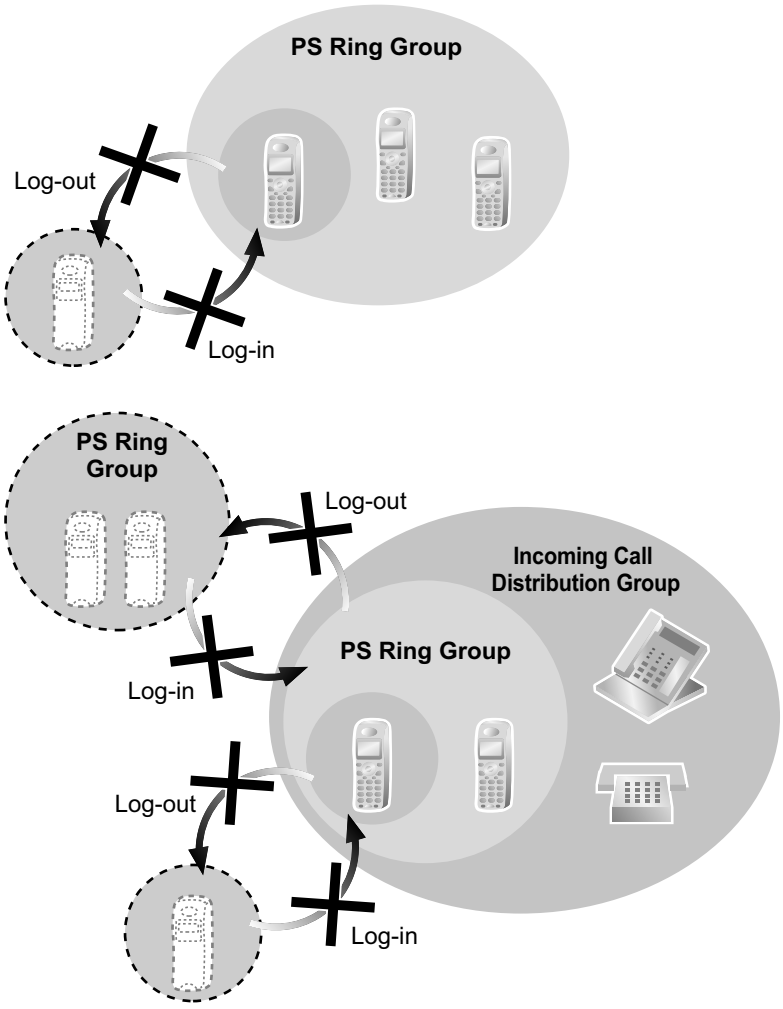
- Incoming trunk call information is shown on a PS display when a trunk call arrives at a PS ring group which the PS joins. The display information can be selected on a PS ring group basis through system programming: Called Party's Name/Number or Caller's Name/Number.
- **Calling Multiple PSs Simultaneously**
  There are two methods to call multiple PSs simultaneously using the floating extension number assigned to the following groups:

Method	Assignment	Merit	Demerit
Incoming Call Distribution Group	Assign all desired PSs to one incoming call distribution group, and set the group call distribution method for the group to "Ring". → 2.2.2 Incoming Call Distribution Group Features	All PS users in the group can use the Log-in/Log-out feature, Wrap-up feature, and ICD Group button for the group.	The CS may often be busy as each PS in the group uses one channel when a call arrives at the group.
PS Ring Group	Assign all desired PSs to one PS ring group.	Only one channel is used when a call arrives at the group.	PS users in the group cannot use the Log-in/Log-out and Wrap-up features.

- When a PS joins a PS ring group, the following personal settings are disregarded:
  a. When the PS ring group is called:
  – Delayed Ringing
  – Display information when the incoming calls arrive;
  The settings (e.g., display priority) are disregarded.
  – The setting which is assigned on the PS (e.g., FWD)
  – The status of the PS (e.g., busy)
  b. Log-in/log-out setting (from the PS ring group/from the incoming call distribution group which the PS ring group belongs to). (→ 2.2.2.7 Log-in/Log-out)

Note

Log-in/log-out setting of the PS ring group from the incoming call distribution group is also disregarded.



- When the PS ring group is called using the floating extension number, the group becomes busy to other callers using the floating extension number. However, the individual group members may be called directly using their extension number.
- If a PS in a PS ring group has set the DND feature for trunk calls, the PS will not ring when an intercom call or a trunk call arrives at the PS ring group. (→ 2.3.3 Do Not Disturb (DND))
- For calls directed to PS ring groups, the PBX will handle at most two calls simultaneously. The third call cannot arrive at a PS ring group until one of the first two calls is answered or a caller hangs up.

PC Programming Manual References

11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Main—◆ Distribution Method
11.8 PBX Configuration—[3-8] Group—PS Ring Group
11.8.1 PBX Configuration—[3-8] Group—PS Ring Group—Member List

PT Programming Manual References

[620] Incoming Call Distribution Group Member

Feature Guide References

6.1 Capacity of System Resources

5.2.4.3 PS Directory

Description

A PS user can store numbers and/or names in the directory. A stored number is dialled by selecting a name or number in the directory.
Depending on the PS model, the PS user can use the following directories for easy operation:

Type	Description
PS Dialling Directory	Makes a call by selecting from a private directory of names and telephone numbers.
System Speed Dialling Directory	Makes a call by selecting from a common directory of names and numbers.
PBX Extension Dialling Directory	Makes a call by selecting from a common directory of extension names.
Shortcut Directory	Accesses a feature by selecting from a private directory of feature names and numbers.
Quick Dialling	Makes a call or accesses a feature easily by selecting from a private directory of names and numbers.

Conditions

- S-PS users can only access the PS Dialling Directory. However, the first 300 entries in the PBX's system speed dialling directory will be automatically downloaded to each S-PS.
  For details about S-PSs, see "5.2.2.2 SIP Portable Station (S-PS) and SIP Cell Station (SIP-CS)".

PC Programming Manual References

14.1 PBX Configuration—[6-1] Feature—System Speed Dial
→◆ Name
→◆ CO Line Access Number + Telephone Number
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Main—◆ Extension Name
12.2.1 PBX Configuration—[4-2-1] Extension—Portable Station—Extension Settings—Main—◆ Extension Name

PT Programming Manual References

[001] System Speed Dialling Number
[002] System Speed Dialling Name
[004] Extension Name

5.2.4.4 PS Feature Buttons

Description

A PS user can use PBX features using a combination of buttons (button + a specified number, ⁎, or #) and/or display operation. The flexible buttons and the display are customised through PS programming. The button assignment is the same for the PT (→ 2.21.2 Flexible Buttons). Some special feature buttons (e.g., WAVESEARCH button) may be customised depending on the PS type.

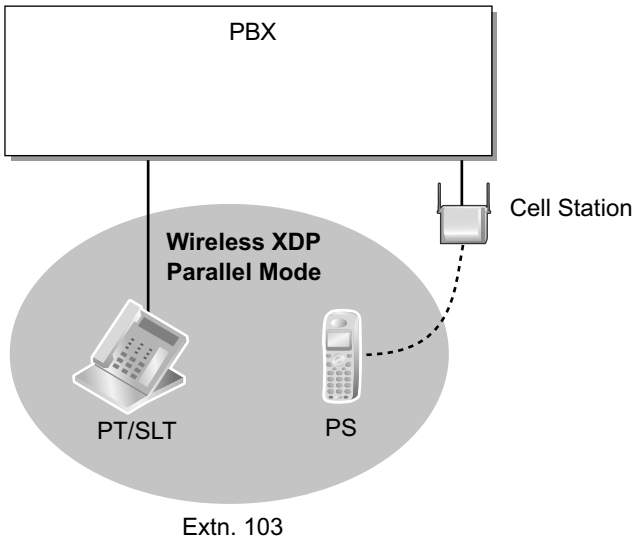
Conditions

- The flexible buttons that can be assigned to an S-PS are different from other types of PSs. For details, see "5.2.2.2 SIP Portable Station (S-PS) and SIP Cell Station (SIP-CS)".

5.2.4.5 Wireless XDP Parallel Mode

Description

A PS can be used in parallel with a wired telephone (PT/SLT). In this case, the wired telephone is the main telephone and the PS is the sub telephone. When Wireless XDP Parallel Mode is enabled, the two telephones share one extension number (main telephone's extension number).



Conditions

- If one of the telephones goes off-hook while the other telephone is on a call, the call is switched to the telephone going off-hook. However, the call is not switched in one of the following conditions:
  - **a.** During a Conference call (→ 2.14 Conference Features).
  - **b.** While Live Call Screening (LCS) or Two-way Record is activated (→ 3.2.2.16 Live Call Screening (LCS) and 3.2.2.30 Two-way Record/Two-way Transfer).
  - **c.** While receiving OHCA (DPT over a stacking connection only) (→ 2.10.4.3 Off-hook Call Announcement (OHCA)).
  - **d.** While being monitored by another extension (→ 2.10.3 Call Monitor).
  - **e.** During Consultation Hold.
  - **f.** During a Conference Group Call (→ 2.15.1 Conference Group Call).
- Wireless XDP Parallel Mode can only be set from a PS. The wired telephone can accept or deny this feature through COS programming. Once this feature is set, the setting at the wired telephone cannot be changed unless a PS changes the setting.
- The following features are not available for extensions in Wireless XDP Parallel Mode while the PS is on a call (however, they are available for extensions in Wireless XDP Parallel Mode while the wired telephone is on a call):
  - Executive Busy Override (→ 2.10.2 Executive Busy Override)
  - Whisper OHCA (→ 2.10.4.4 Whisper OHCA)
  - CCBS (→ 4.1.2.9 Completion of Calls to Busy Subscriber (CCBS))
- If a SIP extension is paired as either the wired extension (e.g., KX-UT series SIP phone) or the wireless extension (e.g., S-PS), both telephones ring when an incoming call arrives. However, no other functions, such as switching the call between telephones, are available.
- If an incoming call arrives while the PS is on a call, the wired extension will indicate an incoming call but will not ring. However, if the wired telephone is a SIP extension, the telephone will ring.
- Most of the extension data (e.g., extension number, extension name) of the wired telephone is used for its PS as well. However, the PS has its own extension data for the following:

- Ring Tone Pattern Table Selection (→ 2.1.3.2 Ring Tone Pattern Selection)
- Preferred Line Assignment—Incoming (→ 2.4.2 Line Preference—Incoming)
- Preferred Line Assignment—Outgoing (→ 2.5.5.2 Line Preference—Outgoing)
- Hot Line Setting (→ 2.6.6 Hot Line)
- Transfer Recall Destination for Call Transfer and Call Park (→ 2.12.1 Call Transfer) (→ 2.13.2 Call Park)
- Display Language (→ 2.21.4 Display Information)
- ISDN Bearer Mode (→ 4.1.2.1 Integrated Services Digital Network (ISDN)—SUMMARY)
- Flexible Button Assignment (→ 2.21.2 Flexible Buttons)

Note

To change the setting of the extension data above, the setting for the wired telephone or the PS must be changed individually. When changing the PS setting, use the PS's original extension number (not the main telephone's extension number), if required.

- When the Wireless XDP Parallel Mode has been set, the following extension data for the wired telephone is copied to the PBX extension data for the PS and the extension data remains there even when the Wireless XDP Parallel Mode is cancelled.
  - Call Waiting Setting (→ 2.1.3.3 Call Waiting)
  - FWD/DND Setting (→ 2.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features)
  - Call Pickup Deny Setting (→ 2.4.3 Call Pickup)
  - Executive Busy Override Deny Setting (→ 2.10.2 Executive Busy Override)
  - Itemised Billing Code for ARS (→ 2.8.1 Automatic Route Selection (ARS))
  - Transfer Recall Destination for Call Transfer and Call Park (→ 2.12.1 Call Transfer) (→ 2.13.2 Call Park)
  - CLIP/COLP Number and CLIP/COLP Number Selection (→ 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP))
  - CLIR and COLR Setting (→ 4.1.2.2 Calling/Connected Line Identification Presentation (CLIP/COLP))
  - Extension Personal Identification Number (PIN) (→ 2.24.1 Extension Personal Identification Number (PIN))
  - COS Programming (→ 5.1.1 Class of Service (COS))
  - Extension User Group (→ 5.1.2 Group)
- If Extension Feature Clear is performed, the corresponding extension data for both the wired telephone and the PS will be cleared. (→ 2.24.2 Extension Feature Clear)
- When a call arrives, both the wired telephone and the PS ring. However, in the following cases only the extension from which the option was set will receive ringing:
  - Automatic Callback Busy (→ 2.10.1 Automatic Callback Busy (Camp-on))
  - Transfer Recall (→ 2.12.1 Call Transfer), Hold Recall (→ 2.13.1 Call Hold), and Call Park Recall (→ 2.13.2 Call Park)

PC Programming Manual References

9.34 PBX Configuration—[1-2] Configuration—Portable Station—PS Registration and De-registration
10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Features—◆ Wireless XDP Parallel Mode Set / Cancel
10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Optional Device & Other Extensions—◆ Accept Wireless XDP Parallel Mode Set by PS
12.1.1 PBX Configuration—[4-1-1] Extension—Wired Extension—Extension Settings—Option 1—◆ Wireless XDP / Shared Extension

PT Programming Manual References

[515] Wireless XDP Parallel Mode for Paired Telephone

Feature Guide References

2.11.10 Parallelled Telephone
5.1.1 Class of Service (COS)

User Manual References

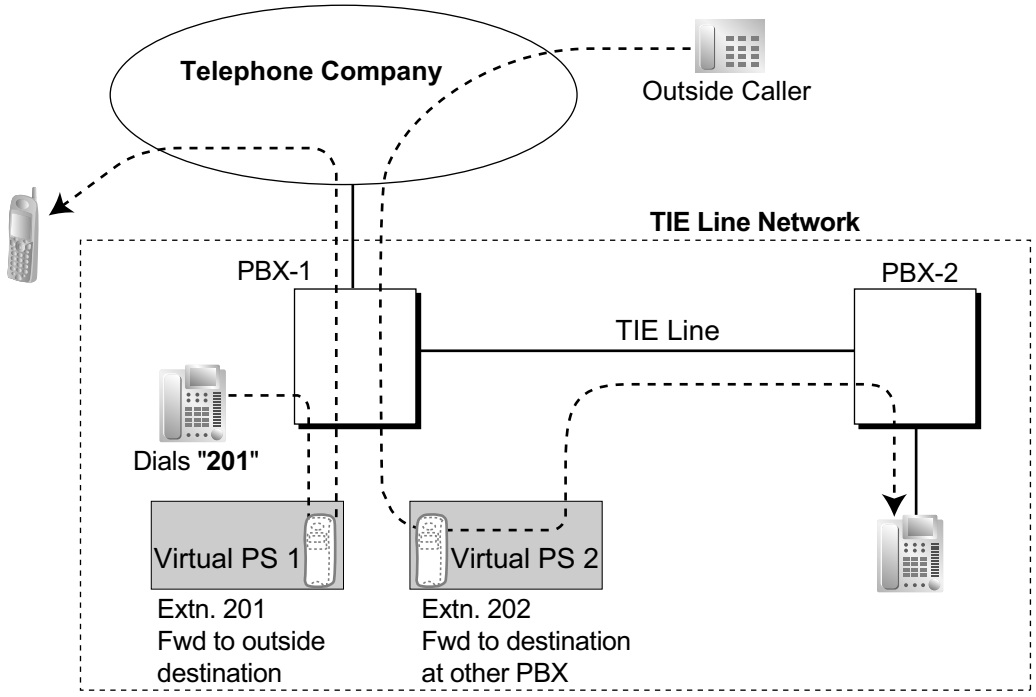
1.9.12 Using Your PS or S-PS in Parallel with a Wired Telephone (Wireless XDP Parallel Mode)

5.2.4.6 Virtual PS

Description

An extension number can be assigned for a portable station (PS) without registering the PS unit itself. This is known as temporary registration. If a forward destination is then assigned for this PS, all calls to that extension number will be forwarded to the assigned destination. Using this setting to forward calls to outside destinations or destinations at another PBX allows those destinations to receive calls as if they were within the PBX. In addition, depending on system programming, the forward destination can use some of the features of the PBX. This can be especially useful for a cellular phone user, who can use his cellular phone as if it were his extension when he is away from his desk.

[Example]



The following features can be accessed using this method:

Feature	Description & Reference
Outside Destinations in Incoming Call Distribution Group	A virtual PS allows calls to an Incoming Call Distribution (ICD) Group to be answered by outside destinations or extensions at another PBX. → 2.2.2.3 Outside Destinations in Incoming Call Distribution Group
Network ICD Group	Using virtual PSs in an ICD Group, up to 4 other PBXs can be called at the same time. → 4.2.6 Network ICD Group
PS Roaming by Network ICD Group	One PS can be registered at up to 4 PBXs. Using virtual PSs in an ICD Group, all 4 PBXs can be called simultaneously to search for the PS. → 4.2.6.1 PS Roaming by Network ICD Group

Feature	Description & Reference
Automatic Fax Transfer	A virtual PS can be used to forward fax calls to a fax machine at another PBX connected by TIE line. → 2.16.2 Automatic Fax Transfer
Built-in Simplified Voice Message	Registering a virtual PS as the first extension of an ICD Group provides the ICD Group with a dedicated message box that is not shared with an actual extension. →2.16.3 Built-in Simplified Voice Message (SVM)

Conditions

- To use this feature, call forwarding to trunks must be enabled through COS programming.

PC Programming Manual References

9.34 PBX Configuration—[1-2] Configuration—Portable Station

PT Programming Manual References

[690] PS Registration

Feature Guide References

2.3.2 Call Forwarding (FWD)

5.2.5 ISDN Extension Features

5.2.5.1 ISDN Extension

Description

An ISDN (PRI) port can be used for either trunk or extension connection. When extension connection is enabled, ISDN terminal devices or a behind PBX can be connected to the port.
The PRI port is point-to-point (P-P) configuration, one terminal device can be connected to the port.

Conditions

- ISDN terminal devices that receive power over the telephone line are not supported.

PC Programming Manual References

9.26 PBX Configuration—[1-1] Configuration—Slot—Port Property - PRI Port—Extension Setting

5.2.6 Extension Port Configuration

Description

There are three types of extension ports as follows:

a. DPT Port: DPT, DSS Console, Panasonic VPS (DPT [Digital] Integration), or PT-interface CS (e.g., KX-TDA0158) can be connected.

b. SLT Port: SLT or Panasonic VPS (DTMF Integration) can be connected.

c. Super Hybrid Port: DPT, APT, SLT, DSS Console, Panasonic VPS, or PT-interface CS can be connected.

EXtra Device Port (XDP) of Super Hybrid Ports:

A DPT and SLT can be connected to one Super Hybrid port (TR: SLT, HL: DPT). In this case, the SLT port (TR) of the Super Hybrid port can be used as an XDP port to connect an SLT as a sub telephone. There are two modes for the XDP port as follows:

Mode	Description
Parallel Mode	The DPT and SLT have the **same extension number** so that they can act as one extension. They use the main telephone's (DPT's) extension data (e.g., extension number, COS). (→ 2.11.10 Parallelled Telephone)
XDP Mode	The DPT and SLT have **different extension numbers** so that they can act as completely different extensions. To use XDP mode, XDP mode must be enabled (on) the port through system programming.

Conditions

- **Automatic Detection on Super Hybrid Port**
  A DPT, SLT, or PT-interface CS connected to a Super Hybrid port can be detected automatically without any programming. An APT connected to a Super Hybrid port can be detected automatically when the XDP mode has been disabled.
- A DSS Console or a Panasonic VPS (DPT [Digital] Integration) can also be connected with an SLT in XDP mode.
- **APT and SLT in Parallel Mode**
  An APT and an SLT can also be connected to a Super Hybrid port and used in parallel mode.
- **Digital XDP**
  A DPT can be connected to another DPT and act as a completely different extension. (→ 2.11.10 Parallelled Telephone)
- **Wireless XDP Parallel Mode**
  A PS can be used in parallel mode with a wired telephone.
  (→ 1.25.5 Wireless XDP Parallel Mode)
- **DSS Console and Paired Telephone Assignment**
  When a DSS Console is connected, a paired extension must be assigned through system programming. Only a PT can be a paired extension.

PC Programming Manual References

9.22 PBX Configuration—[1-1] Configuration—Slot—Extension—Port Property
12.3 PBX Configuration—[4-3] Extension—DSS Console—◆ Pair Extension

PT Programming Manual References

[007] DSS Console Paired Telephone
[600] EXtra Device Port (XDP) Mode

5.3 Legacy Device Connection

5.3.1 Trunk Adaptor Connection

Description

IP trunks (V-SIPGW and V-IPGW) can be connected to a Trunk Adaptor (KX-NS8188/KX-NS8290), which allows the KX-NS300 to connect to E1 trunks and PRI lines.

Conditions

[General]

- One IP trunk port corresponds to 1 channel in the Trunk Adaptor.
- An activation key is not required for ports whose **Connection Attribute** setting is set to **Trunk Adaptor**. Ports without this setting can still be connected to a Trunk Adaptor, but they will require an activation key.
- One KX-NS300 can connect to multiple Trunk Adaptors.
- For details about configuring settings for E1 trunks and PRI lines, refer to the documentation for the Trunk Adaptor.

[Connection via SIP Trunk]

- The SIP trunk authentication ID and the authentication ID of the IP interface on the Trunk Adaptor must be the same.
- The channel attribute of SIP trunks connecting to a Trunk Adaptor must be set to **Basic channel**.

PC Programming Manual References

9.10 PBX Configuration—[1-1] Configuration—Slot—V-SIPGW—Port Property—Programming Port Properties—Trunk Adaptor
9.10 PBX Configuration—[1-1] Configuration—Slot—V-SIPGW—Port Property—Main
→◆ Connection Attribute
→◆ Channel Attribute
9.12 PBX Configuration—[1-1] Configuration—Slot—V-IPGW—Port Property
→Trunk Adaptor
→◆ Connection Attribute

5.4 E-mail Notification Features

5.4.1 E-mail Notification for Extension Users

Description

An e-mail can be sent to extension users, notifying them of events such as when they receive a new voice message. Notifications can be sent for the following events:

- Missed trunk call
- New voice message

Conditions

- For extension users to receive notifications of new voice messages, they must have a mailbox assigned to their extensions. (→ Section 20 UM Configuration—[1] Mailbox Settings)
- For extension users to receive notifications of missed trunk calls, the following settings must be configured:
  → "Contact—Email 1–3" in 8.2.1 Users—Add User—Single User
  → "Email notification" in 8.2.1 Users—Add User—Single User
- To send e-mail notifications, the SMTP server settings must be configured.
- Up to 3 e-mail addresses can be registered for each extension user.

PC Programming Manual References

8.2 Users—Add User
20.1.2 UM Configuration—[1-2] Mailbox Settings—Full Setting—Notification Parameters—◆ E-mail/Text Message Device—Device No. 1, 2, 3—E-mail Address
27.2.5 Network Service—[2-6] Server Feature—SMTP

Feature Guide References

3.2.1.28 Message Waiting Notification—E-mail Device

User Manual References

3.2.1 User Programming—Changing E-mail Notification Settings

5.4.2 E-mail Notification of System-level Events

Description

An e-mail can be sent to administrators or other specified e-mail addresses when certain system-level events occur. Notifications can be sent for the following events:

Event	Details
System alarm	An e-mail is sent to two registered e-mail addresses.
Software update	Notifications can be sent for the following four types of software update events: **1.** A software update has become available on the update FTP server. **2.** A software update has been downloaded from the update FTP server. **3.** A software update has been successfully installed. **4.** A software licence is about to expire.

Conditions

- To send e-mail notifications, the SMTP server settings must be configured.
- The system name in the e-mail notification can be configured through system programming.
  → 27.3.3 Network Service—[3-3] Client Feature—SNMP Agent—◆ MIB info—SysName

PC Programming Manual References

5.1 System Control—Program Update
7.7 Utility—Email Notification
8.2 Users—Add User
27.2.5 Network Service—[2-6] Server Feature—SMTP

Feature Guide References

5.5.9 Software Upgrading

5.4.3 E-mail Notification of Sensor Alarm

Description

An e-mail can be sent to a specified e-mail address when the external sensor detects an alarm. You can put comments in the e-mail. If you put a URL for the network camera in the comments, the recipient of the notification can monitor the video feed from the network camera by clicking URL in the comments.

Conditions

- **Hardware Requirement:**
  An external sensor and a DOORPHONE card.
- To send e-mail notifications for a sensor alarm, a DOORPHONE card must be installed.
- A maximum 8 sensors can be connected to the PBX. An e-mail address can be set for each external sensor.
- For users to receive sensor alarm notifications, the following settings must be configured:
  - Notification e-mail for sensor alarm must be enabled
    13.5 PBX Configuration—[5-5] Optional Device—External Sensor ◆ Sensor Alarm—Email notification
  - E-mail address
    13.5 PBX Configuration—[5-5] Optional Device—External Sensor◆ Sensor Alarm—Email Address
  - Comment for e-mail
    13.5 PBX Configuration—[5-5] Optional Device—External Sensor ◆ Sensor Alarm—Email Comment

PC Programming Manual References

13.5 PBX Configuration—[5-5] Optional Device—External Sensor

Feature Guide References

2.18.3 External Sensor

5.5 System Data Control

5.5.1 User Profiles

Description

Manage the following user information settings on a per-user level.
- User information (name, language, etc.)
- Contact information (extension number, e-mail addresses, etc.)
- Unified Message information (mailbox number, password, etc.)
- E-mail notification information
- Telephony feature information (FWD/DND, Personal Speed Dialling, etc.)
- Login account information

Users can configure certain user information via Web Maintenance Console.
User information for other accounts can be viewed, added, edited, and deleted by logging in with a "User (Administrator)" account or an "Installer" account.

Note

User information (extension number, name, mailbox number, login account, etc.) must be registered in "User Profiles" before configuring personal information in "PBX Configuration—Extension" or "UM Configuration—Mailbox Settings".

Installation Manual References

5.11 Automatic Configuration of Mailboxes

PC Programming Manual References

Section 8 Users
Section 12 PBX Configuration—[4] Extension
Section 20 UM Configuration—[1] Mailbox Settings

Feature Guide References

3.1.2.1 Automatic Configuration of Mailboxes
5.5.2 PC Programming

User Manual References

3.2.1 User Programming

5.5.2 PC Programming

Description

This PBX can be programmed and administered using a PC. There are two programming methods:

1. **On-site Programming:** System programming/diagnosis can be performed locally by connecting a PC to the PBX directly.
2. **Remote Programming:** System programming/diagnosis and data upload can be performed from a remote location.

1. **On-site Programming:**

Method	Description
Using the LAN interface	Available via the LAN port of the MPR card.
Using a modem through an SLT port[*1]	An RMT card must be installed. Assign the floating extension number of the analogue remote maintenance (default: 599), and dial this number from the PC to connect to the PBX.
Using an ISDN TA interface (64 kbps) through an ISDN Extension Line[*1]	Assign the floating extension number of the ISDN remote maintenance (default: 699), and dial this number from the PC to connect to the PBX. The RMT card is not required for this method. This method is available only when a user-supplied ISDN TA that supports CAPI is used.

[*1] If remote access is disabled through system programming, then this on site programming cannot be done.

2. **Remote Programming:**

Method	Description
Using a modem (RMT card)	An RMT card must be installed. The floating extension number of the analogue remote maintenance must be assigned (default: 599). PC programming, using a telephone connected in parallel with the modem, can be done in the following ways: • **Direct Access** Dial the DIL/DID/DDI number whose destination is the floating extension number of the analogue remote maintenance. • **Through DISA** Dial the floating extension number of the analogue remote maintenance using the DISA feature. (→ 2.16.1 Direct Inward System Access (DISA)) • **Call Transfer** Call an extension (probably the operator), and request a transfer to the floating extension number of the analogue remote maintenance. (→ 2.12.1 Call Transfer)
Using an ISDN TA interface (64 kbps) through an ISDN Trunk	The floating extension number of the ISDN remote maintenance must be assigned (default: 699), and dial the DIL/DID/DDI number whose destination is the floating extension number of the ISDN remote maintenance. The RMT card is not required for this method. This method is available only when an user-supplied ISDN TA that supports CAPI is used.

There are three levels of authorisation for programming the PBX, where each level controls which settings the programming is allowed to access and change. The three levels are as follows:

Level	Description	Number of Accounts/Network
Installer	For dealers and system installers	1
User (Administrator)	For on-site managers	8
User (User)	For end users	492

Conditions

- Each account is assigned a password that is required to log in.
- Users can be added using the Add User Wizard.

CAUTION

To the Administrator or Installer regarding account passwords

1. Please provide all system passwords to the customer.
2. To avoid unauthorised 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 no passwords set initially. For security, select 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 10 numbers or characters be used for maximum protection against unauthorised access.

Installation Manual References

4.3.10 RMT card in KX-NS300 (KX-TDA0196)
5.2 PC Connection
5.3 Starting Web Maintenance Console

PC Programming Manual References

2.1.1 Web Maintenance Console Accounts
2.1.2 Access Levels
8.2 Users—Add User

PT Programming Manual References

[801] External Modem Control
[810] Remote Programming
[811] Modem Floating Extension Number
[812] ISDN Remote Floating Extension Number

User Manual References

3.2 System Programming Using Web Maintenance Console

Feature Guide References

2.1.1.2 Direct In Line (DIL)
2.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)

5.5.3 PT Programming

Description

A PT user can perform the following programming:

a. **Personal Programming:** Customising the extension according to his needs.
b. **System Programming:** Customising the PBX according to organisational needs.
c. **Manager Programming:** Customising specified frequently changing items (e.g., Charge Management and Remote Extension Dial Lock).

Conditions

- COS programming determines what programming can be performed:
  – System programming and personal programming
  – Personal programming only
  – No programming
- System programming can be performed only from a multi-line display DPT or IP-PT. Multi-line display APTs are not supported.
- The extension which is connected to the lowest numbered extension port can perform both personal programming and system programming regardless of the COS.
- The extension(s) assigned as the manager COS can perform manager programming.
- During programming, the PT is considered to be busy.
- Only one system programmer or one manager programmer is allowed to perform system or manager programming at one time. The maximum number of simultaneous programmers that each PBX supports is as follows:
  – one system programmer + 63 personal programmers
  – one manager programmer + 63 personal programmers
  – 64 personal programmers
- **System Programming Password Level**
  To access system programming, a valid password must be entered. For more detail information, refer to "1.1.2 Password Security" in the PT Programming Manual.
- **Personal Programming Data Default Set**
  A user can return the items programmed on the telephone to default.

PC Programming Manual References

10.7.1 PBX Configuration—[2-7-1] System—Class of Service—COS Settings—Manager—◆ PT Programming Mode Level
19.1 PBX Configuration—[11-1] Maintenance—Main—Password—◆ System Password - PT Programming—Prog *#: Administrator Level
19.1 PBX Configuration—[11-1] Maintenance—Main—Password—◆ System Password - PT Programming—Prog **: User Level
19.1 PBX Configuration—[11-1] Maintenance—Main—Password—◆ Manager Password - PT Programming—Prog *1

PT Programming Manual References

[516] Programming Mode Limitation

Feature Guide References

5.1.1 Class of Service (COS)
5.1.6 Manager Features
6.1 Capacity of System Resources

User Manual References

1.9.14 Clearing Features Set at Your Extension (Extension Feature Clear)
3.1 Customising Your Phone (Personal Programming)
4.1 Manager Programming

5.5.4 DSP Resource Usage

Description

To digitally process audio signals, such as a telephone call, the PBX must use a certain number of DSP (Digital Signal Processing) resources. DSP resources are provided by the DSP card installed in the PBX. Since there are a limited number of DSP resources, no further operations (e.g., telephone calls, playing an OGM) can be performed if all resources are being used.
The following list shows some of the basic operations that require DSP resources.

- IP extension call
- IP trunk call
- Conference
- Accessing the Unified Messaging system (including recording calls)
- OGM playback
- Echo canceller

For IP extension and trunk calls, the number of required resources differs depending on the codec (G.711 or G.729) used.

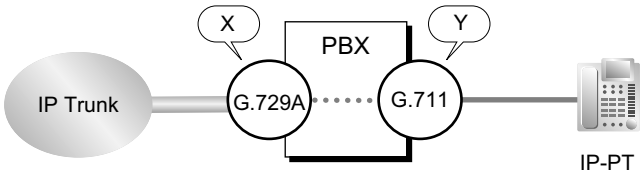
Note

The examples in this section are intended to illustrate the concept of DSP resource usage. More complex situations may necessitate additional resources, and in some cases fewer resources may be necessary.

Examples of DSP resource usage

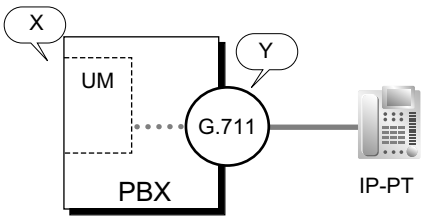
Fundamentally, the number of resources required for a given situation is the sum of the resources required for each individual operation. The following examples illustrate DSP resource usage in various situations.

[IP trunk to IP extension]



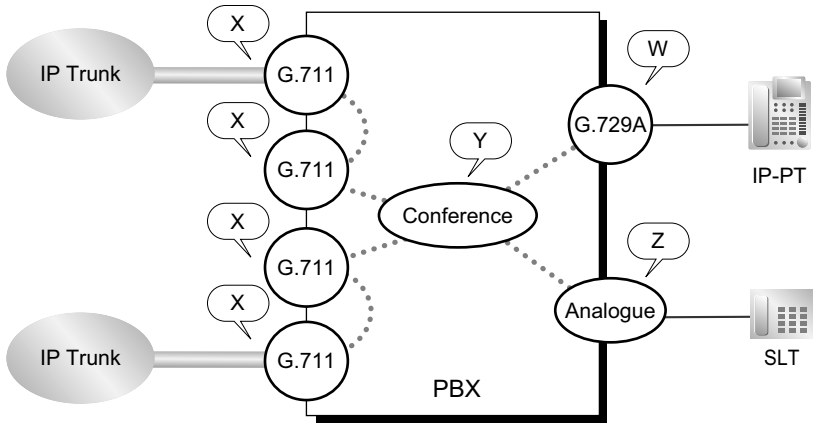
If an IP trunk call using the G.729 codec requires X number of resources, and an IP extension call using the G.711 codec requires Y number of resources, then the number of resources required for a call from an IP extension to an IP trunk requires X + Y number of resources.

[Unified Messaging access]



Playing back messages from or recording messages to the Unified Messaging (UM in the figure above) system requires DSP resources, X in this example, in addition to the resources required for the G.711 codec (Y). The total cost is X + Y.

[Conference call]



A conference requires additional resources, Y, for handling the multiple voice channels. Also, in standard two-way conversations, analogue lines generally do not require any DSP resources, but in a conference they do. In addition, IP trunks in a conference require additional DSP resources.
For this example, then, the number of required resources is X + X + X + X + Y + Z + W.

DSP usage graph

The PBX keeps a record of the maximum DSP usage per hour for each of the following features/services. A graph can then be displayed in Web Maintenance Console showing trends in DSP usage over time, as well as the number of calls and operations that could not be performed due to lack of resources.

- VoIP (IP trunk, IP extension and IP-CS usage)
- Conference
- Unified Messaging
- OGM
- Two-way Recording

The graph also shows the amount of free resources and the total resource use.

Conditions

[General]

- Most internal VoIP calls require no resources because IP phones connect over a P2P connection (→ 5.2.3 Peer-to-Peer (P2P) Connection). That is, the phones themselves do the signal processing without consuming PBX resources. The PBX simply performs the initial connection.
- Telephones and trunk lines use the same number of resources as analogue telephones and trunks connected directly to the PBX.

[DSP usage graph]

- The most recent 30 days of DSP usage is recorded.
- The scale of the graph can be set to 1-hour, 4-hour, or 24-hour intervals.

Installation Manual References

2.3.3 System Capacity—DSP Resources
4.3.3 DSP S Card (KX-NS5110)

PC Programming Manual References

9.2.1 PBX Configuration—[1-1] Configuration—Slot—System Property—Main—V-IPGW–GW Settings–Option 1—◆ IP Codec Priority—1st, 2nd, 3rd
9.2.1 PBX Configuration—[1-1] Configuration—Slot—System Property—Main—Main
→◆ DSP CODEC G.711 only (SIP extension)
→◆ DSP CODEC G.711 only (IP-GW)
→◆ DSP CODEC Priority-1 value only (others)
9.11 PBX Configuration—[1-1] Configuration—Slot—V-IPGW—Shelf Property
→◆ Voice Codec Priority 1st
→◆ Voice Codec Priority 2nd
→◆ Voice Codec Priority 3rd
9.10 PBX Configuration—[1-1] Configuration—Slot—V-SIPGW—Port Property—Voice/FAX—◆ IP Codec Priority—1st, 2nd, 3rd
9.14 PBX Configuration—[1-1] Configuration—Slot—V-IPEXT—Port Property—Option—◆ IP Codec Priority
9.18 PBX Configuration—[1-1] Configuration—Slot—V-IPCS—Port Property—Option—◆ IP Codec Priority
9.37 PBX Configuration—[1-5] Configuration—DSP Resource

Feature Guide References

2.14.2 Conference
2.30.2 Outgoing Message (OGM)
3.2 System and Subscriber Features
3.2.2.30 Two-way Record/Two-way Transfer
3.2.1.4 Automatic Two-way Recording for Manager
5.2.1 IP Proprietary Telephone (IP-PT)
5.2.2 SIP (Session Initiation Protocol) Extension

5.5.4.1 DSP Resource Reservation

Description

A number of resources can be reserved for particular features or services to guarantee a minimum level of service. Resources reserved for a particular service (e.g., conferencing) cannot be used for another service (e.g., Unified Messaging).
For example, one may want to reserve resources for OGM (Outgoing Message) to ensure that recorded messages can be played to incoming calls.
The resources can be reserved for the following types of services:

- VoIP (G.711)
- Conference trunk
- Unified Messaging
- Two-way Recording
- OGM

Note

It is not necessary to reserve resources for a feature to use it. In normal operation, free DSP resources are allocated on a first-come first-serve basis. Resources should be reserved only if a minimum level of performance is required for your system.
For example, reserving resources for Two-way Recording (→ 3.2.1.4 Automatic Two-way Recording for Manager, → 3.2.2.30 Two-way Record/Two-way Transfer) also reserves UM ports. This can have the unintended effect of blocking access to the Unified Messaging system even when no recording is being performed. Therefore, resources for Two-way Recording should be reserved only if it is necessary to guarantee that Two-way Recording can be performed. (→ 3.1.1 Unified Messaging System Overview)

Resource Reservation Example

The following table shows the number and types of resources that must be reserved for a given workload. To calculate the number of free (i.e., non-reserved) resources, a DSP S card (63 DSP resources) is assumed.

Service	Minimum performance[*1]	Resources per unit of performance	No. of required DSP resources
VoIP call (G.711)	20 calls	1	20
Unified Messaging[*2]	4 operations	1.3	5.2
Two-way Recording	2 recordings	2.3	4.6
OGM	4 playbacks	2	8
Conference trunk[*3]	4 conferences	0.5	2
Unified Messaging tone[*4]	2 tones	1	2
		Total Reserved Resources	**41.8**
		Free Resources[*5]	**21**

[*1] Minimum performance refers to the minimum number of simultaneous operations.
[*2] Unified Messaging operations include operations such as users (subscribers) accessing their mailboxes and outside callers leaving messages in subscribers' mailboxes.
[*3] DSP is used when DSP Conference Priority is set to Preferential.
[*4] The resources for Unified Messaging tones (used for Two-way Recording) are reserved automatically and cannot be released.
[*5] Free Resources = DSP capacity – Total Reserved Resources, rounded to the nearest whole number.

If the PBX's resources are reserved as shown in the example above, the resources required to meet the numbers of operations listed in the "Minimum performance" column are guaranteed to be available. Note, however, that for an operation such as a conference call, DSP resources are required for each individual

conference party in addition to the resources required for the conference trunk itself. Therefore, if all 40 VoIP units as well as all free resources are being used, a new conference call cannot be established, even if sufficient conference resources are available.

Conditions

- The total number of resources provided by each type of DSP card is as follows:
  – DSP S card: 63

PC Programming Manual References

9.37 PBX Configuration—[1-5] Configuration—DSP Resource

5.5.4.2 DSP Resource Advisor

Description

Web Maintenance Console provides a tool for calculating the number of resources required for a given set of operating conditions. The Web Maintenance Console user provides information such as the number of ports for a given resource (e.g., 16 extension ports using the G.729 codec) and the expected usage load (e.g., 50% busy), and the resource advisor calculates the number of DSP resources required to meet those conditions. This tool can be used in offline mode to simulate various PBX configurations and usage cases to help determine the size of DSP card required.
This tool will also recommend which and how many resources to reserve for various features (→ 5.5.4.1 DSP Resource Reservation). The recommended settings can be applied immediately from the resource advisor tool.
The resource usage can be calculated using the following types of services and features:

Service/Feature	DSP Resources per Unit
Trunk using G.729 codec	2.2
Trunk using G.711 codec	1
Non-IP trunk (ISDN trunk, analogue trunk, etc.)	1
Extension using G.729 codec	2.2
Extension using G.711 codec	1
IP-CS extension using G.729 codec	2.2
IP-CS extension using G.711 codec	1
Unified Messaging*1	1.3
Two-way Recording*1*2	2.3
OGM*1	2
Conference trunk*1	0.5

*1 The DSP costs of the extensions/trunks involved in the operation are not included in the per-unit DSP resource count.
*2 Two-way Recording also requires a conference trunk. For an example of the required DSP resources, see 5.5.4 DSP Resource Usage.

Example 1: Small Office

In a small office (e.g., 32 employees), the necessary number of trunks and extensions is likely to be relatively small. In addition, the expected load on the system will also be small.
(For clarity, unused services are not included in the table.)

Service	Number of Ports	Load (Busy Ratio %)	DSP Cost*1
Trunk using G.729 codec	3	5%	0.3
Non-IP trunk	1	5%	0.05
Extension using G.729 codec	32	10%	7.0
Unified Messaging	4	—	5.2
OGM	2	—	4.0
Conference	4	—	2.0

Service	Number of Ports	Load (Busy Ratio %)	DSP Cost*1
		Total DSP Cost	**18.55**

*1 DSP Cost = Number of Ports × Resource cost per port (unit) × Load

In the example above, the total DSP cost is 18.55. In such an environment, a PBX with a DSP S card (max. 63 DSP resources) would be sufficient.

Conditions

- Calls that are established via P2P (→ 5.2.3 Peer-to-Peer (P2P) Connection) do not use the PBX's DSP resources, so they may be excluded from the usage calculation.

PC Programming Manual References

9.37 PBX Configuration—[1-5] Configuration—DSP Resource

5.5.5 Automatic Setup

Description

There is an automatic setup features as follows:

1. **Automatic Time Adjustment**
   It is possible to adjust the PBX clock automatically in the following two ways:
   a. **Summer Time (Daylight Saving Time) Setting:**
   The start and end dates of the summer time can be programmed. The PBX clock will be adjusted (one hour forward or backward) at 2:00 AM of the programmed date, if enabled through system programming. It means 2:00 AM will become 3:00 AM on the start date of the summer time, and 2:00 AM will become 1:00 AM on the end date.

   Note

   If the Timed Reminder (Wake-up call) is set;
   – On the summer time start date, the setting between 2:00 AM and 3:00 AM will not happen.
   – On the summer time end date, the setting between 1:00 AM and 2:00 AM will ring twice.

   b. **Time Information from Telephone Company:**
   Time information can be received on the following calls:
   • An incoming or outgoing call through an ISDN line
   • An incoming call through an analogue line with Caller ID which includes the time information.
   The PBX clock will be adjusted every day with the first call after 3:05 AM, if enabled through system programming.

   Note

   If the Timed Reminder (Wake-up call) is set, the setting will not happen or will ring twice depending on the adjustment.

   c. **Time Information through Network Time Protocol (NTP):**
   By connecting the PBX to an NTP server, it is possible to receive and update the time setting.
   The PBX clock will be adjusted every day at 3:05 AM, if enabled through system programming.

Conditions

[General]

- Through system programming, it is possible to specify NTP, ISDN, or neither method as the selected method of automatic time adjustment.
- SMDR will record the call information using the PBX clock so that the recording time will be overlapped at the end of summer time. (→ 2.22.1.1 Station Message Detail Recording (SMDR))

[NTP Time Information]

- The time set through NTP will apply the same to all PTs connected to the PBX, regardless if an IP extension is located in another time zone.
- The PBX provides NTP server information to KX-UT extensions (→ 5.2.2.1 KX-UT Series SIP Phones) if the **NTP server** setting is enabled. If an NTP server has been specified through system programming, KX-UT extensions will retrieve their time from that server. The PBX uses its own IP address as the NTP server for the KX-UT extension.
  If **NTP server** is disabled, the time for KX-UT extensions must be set individually at each telephone.

PC Programming Manual References

10.1.1 PBX Configuration—[2-1-1] System—Date & Time—Date & Time Setting
10.1.2 PBX Configuration—[2-1-2] System—Date & Time—SNTP / Daylight Saving

10.1.2.1 PBX Configuration—[2-1-2] System—Date & Time—SNTP / Daylight Saving—Daylight Saving
27.2.4 Network Service—[2-5] Server Feature—NTP—◆ NTP server

Feature Guide References

4.1.2.1 Integrated Services Digital Network (ISDN)—SUMMARY
2.24.4 Timed Reminder

5.5.6 Dynamic Host Configuration Protocol (DHCP) Server

Description

The PBX has a built-in DHCP server. When the DHCP server is enabled, the PBX will automatically assign IP addresses to other devices on the network, such as IP-PTs.
Using a DHCP server simplifies network management by removing the need to assign IP addresses to devices manually.

Conditions

- The DHCP Server feature cannot be used if the PBX's IP address assignment mode is set to DHCP.
- If the PBX's DHCP server is enabled, make sure that no other DHCP servers are running on the same network. Having more than one DHCP server on a network can result in network errors.
- For the following settings, the PBX delivers the settings of its LAN port to devices: subnet mask, default gateway address, and DNS server addresses. As NTP server information for KX-UT extensions, the PBX delivers its own IP address.

PC Programming Manual References

27.2.1 Network Service—[2-1] Server Feature—DHCP

5.5.7 Flexible Numbering/Fixed Numbering

Description

To dial another extension user or to access PBX features, the access numbers (extension numbers or feature numbers) are required.
There are three types of numbering plans:

1. Flexible Numbering (available while a dial tone is heard)
2. Flexible Numbering (available while a busy, DND, or ringback tone is heard)
3. Fixed Numbering (available while dialling or talking)

1. **Flexible Numbering (available while a dial tone is heard)**
   Extension numbers and feature numbers which are available while a dial tone is heard can be customised for easy use. The numbers must not conflict. It is also possible to use the default settings shown in the following table.
   a. **Extension Numbers:** Extension numbers consist of leading numbers and additional numbers. Extension numbers (consisting of "0" through "9") can be assigned as follows:
      - Numbering schemes: 1-64
      - Leading number: up to three digits
      - Additional number: up to two digits (default: two digits)
   b. **Feature Numbers:** A number of up to four digits, consisting of "0" through "9", "✳", and "#"

   c. **Other PBX Extension Numbers (Other PBX Extension Number [TIE] -1 through 16):** A number of up to three digits, consisting of "0" through "9", "✳", and "#"

[Flexible Numbering Table (available while a dial tone is heard)]

Feature	Default
Extension Numbering Scheme 1—Leading Number	**1 / 2**
Extension Numbering Scheme 2—Leading Number	**2 / 3**
Extension Numbering Scheme 3—Leading Number	**3 / 4**
Extension Numbering Scheme 4—Leading Number	**4 / 1**
Extension Numbering Scheme 5–20—Leading Number	**None**
Extension Numbering Scheme 21—Leading Number	**5**
Extension Numbering Scheme 31—Leading Number	**6**
Extension Numbering Scheme 32–64—Leading Number	**None**
Operator Call*1	**9 / 0**
Idle Line Access (Local Access)*1	**0 / 1 / 9**
Trunk Group Access*1	**8**
TIE Line Access*1	**7**
Redial*1	**#**
Speed Dialling—System/Personal*1	**✳✳**
Personal Speed Dialling—Programming*1	**✳30**
Doorphone Call*1	**✳31**

Feature	Default
Conference Group Call[*1]	✳32
Group Paging[*1]	✳33
External BGM on/off[*1]	✳35
Outgoing Message (OGM) playback/record/clear	✳36
S-CO Line Access[*1]	✳37
Simplified Voice Message Access	✳38
Parallel Telephone (Ring) Mode set/cancel	✳39
Group Call Pickup[*1]	✳40
Directed Call Pickup[*1]	✳41
TAFAS—Calls through an External Pager	✳42
Group Paging answer[*1]	✳43
Automatic Callback Busy cancel/CCBS cancel	✳46
User Remote Operation/Walking COS/Verification Code Entry	✳47
Wireless XDP Parallel Mode set/cancel	✳48
Account Code Entry[*1]	✳49
Call Hold/Call Hold Retrieve	✳50
Call Hold Retrieve—Specified with a Holding Extension Number[*1]	✳51
Call Park/Call Park Retrieve[*1][*2]	✳52
Call Hold Retrieve—Specified with a Held Trunk Number[*1]	✳53
Door Open[*1]	✳55
External Relay	✳56
External Feature Access	✳60
SIP Refer (Blind)[*3]	✳61
ISDN Hold	✳62
COLR set/cancel[*1]	✳7✳0
CLIR set/cancel[*1]	✳7✳1
Switch CLIP/COLP of the Trunk/Extension[*1]	✳7✳2
MCID	✳7✳3
ISDN-FWD set/cancel/confirm	✳7✳5
Message Waiting set/cancel/callback	✳70
FWD/DND set/cancel—Both[*1]	✳710
FWD/DND set/cancel—External[*1]	✳711
FWD/DND set/cancel—Internal[*1]	✳712

Feature	Default
FWD/DND No Answer Timer set[*1]	✳713
Group FWD set/cancel—Both[*1]	✳714
Group FWD set/cancel—External[*1]	✳715
Group FWD set/cancel—Internal[*1]	✳716
Call Pickup Deny set/cancel[*1]	✳720
Paging Deny set/cancel[*1]	✳721
Walking Extension/Enhanced Walking Extension	✳727
Data Line Security set/cancel[*1]	✳730
Manual Call Waiting for Extension Call off/BSS/OHCA/Whisper OHCA[*1]	✳731
Automatic Call Waiting set/cancel[*1]	✳732
Executive Busy Override Deny set/cancel[*1]	✳733
Not Ready Mode on/off[*1]	✳735
Log-in/Log-out[*1]	✳736
Incoming Call Queue Monitor	✳739
Hot Line programme/set/cancel	✳740
Absent Message set/cancel[*1]	✳750
BGM set/cancel	✳751
Remote Wake-up Call	✳76✳
Timed Reminder set/cancel	✳760
Printing Message	✳761
Extension Dial Lock set/cancel[*1]	✳77
Time Service Switch[*1]	✳780
Remote Extension Dial Lock off[*1]	✳782
Remote Extension Dial Lock on[*1]	✳783
NDSS Monitor Release	✳784
Trunk Busy Out Clear	✳785
Extension Feature Clear[*1]	✳790
Extension Personal Identification Number (PIN) set/cancel[*1]	✳799
Dial Information (CTI)	None
Other PBX Extension Number (TIE) 1–16	None

Feature	Default
Quick Dialling*4	**None**

*1 SIP extension users can use these feature numbers.
*2 From a SIP extension, this feature number can be used only for Call Park Retrieve.
*3 This feature number is used to do SIP carrier transfer feature. For more information, see 2.12.2 SIP Refer Transfer.
*4 It is possible to register Quick Dialling numbers that overlap with other registered numbers. This is used for the Automatic Rerouting of VoIP Calls To Public Trunk feature.

2. **Flexible Numbering (available while a busy, DND, or ringback tone is heard)**
Feature numbers which are available while a busy, DND, or ringback tone is heard can be customised for easy use. The numbers should be one digit ("0" through "9", "✳", or "#") and must not conflict. For default, refer to the following table:

[Flexible Numbering Table (available while a busy, DND, or ringback tone is heard)]

Feature	Default
Call Waiting/DND Override*1	**1 or 2***2
Executive Busy Override*1	**3**
Message Waiting set*1	**4**
Call Monitor*1	**5**
Automatic Callback Busy/CCBS	**6**
Alternate Calling—Ring/Voice	**✳**

*1 SIP extensions cannot perform DND Override, Executive Busy Override, Message Waiting, or Call Monitor, but can be the recipient of them.
*2 To use Call Waiting/DND Override, both "1" and "2" are available by default.

3. **Fixed Numbering (available while dialling or talking)**
The features which are available while dialling or talking have fixed numbers as shown in the following table:

[Fixed Numbering Table (available while dialling or talking)]

Feature	Fixed Numbering
Pulse to Tone Conversion	**✳**
Conference*1	**3**
Door Open*2	**5**

*1 SIP extensions cannot establish conferences but can participate in them.
*2 SIP extension users can use these feature numbers.

Conditions

- All features have a default feature number.
- The following are examples of feature number conflicts: 1 and 11, 0 and 00, 2 and 21, 10 and 101, 32 and 321, etc.
- **Feature number + Additional number (Parameter)**

Some flexible feature numbers require additional digits to make the feature active. For example, to set Call Waiting, the feature number for "Call Waiting" must be followed by "1" and to cancel it, the same feature number should be followed by "0".

- If a feature number includes "*" or "#", rotary SLT users cannot use it.
- ISDN extension users cannot use the following features:
  - OGM playback/record
  - Call Hold/Call Hold Retrieve (held at its own extension)
  - ISDN Hold
  - MCID
  - Walking Extension
  - Call Waiting
  - Hot Line
  - Timed Reminder
  - Executive Busy Override
  - Call Monitor
  - Automatic Callback Busy/CCBS
- PS users cannot use the following features:
  - Personal Speed Dialling
  - OGM playback/record
  - S-CO Line Access
  - Parallel Telephone Mode set/cancel
  - Walking Extension
  - BGM set/cancel
  - Timed Reminder

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main
→Extension
→Features
→Other PBX Extension
10.6.2 PBX Configuration—[2-6-2] System—Numbering Plan—Quick Dial
10.6.3 PBX Configuration—[2-6-3] System—Numbering Plan—B/NA DND Call Feature

PT Programming Manual References

[100] Flexible Numbering

Feature Guide References

3.1.1 Unified Messaging System Overview
5.2.4.1 Portable Station (PS) Connection
6.1 Capacity of System Resources

User Manual References

5.3.1 Feature Number Table

5.5.8 Floating Extension

Description

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 destination of incoming calls etc.

Resource		Description	Default
Device	**External Pager**	Used as the destination for TAFAS feature. (→ 2.17.2 Trunk Answer From Any Station (TAFAS))	600 or 6000*1
	Outgoing Message (OGM)	Used to send messages for DISA feature. (→ 2.16.1 Direct Inward System Access (DISA))	5 or 50 + two-digit group number*1
	Built-in Simplified Voice Message (SVM)	Used to access message boxes associated with extensions. (→ 2.16.3 Built-in Simplified Voice Message (SVM)	591 or 5091*1
	Analogue Remote Maintenance	Used to access the PC programming mode through a modem on a PC.	599 or 5099*1
	ISDN Remote Maintenance	Used to access the PC programming mode through the ISDN interface on a PC.	699 or 6099*1
Group	**Incoming Call Distribution Group**	Used to call an incoming call distribution group. (→ 2.2.2.1 Incoming Call Distribution Group Features—SUMMARY)	6 or 60 + two-digit group number*1*2
	PS Ring Group	Used to call a PS ring group. (→ 5.2.4.2 PS Ring Group)	—
	VM (DPT) Group	Used to call a VM (DPT) group.	—
	VM (DTMF) Group	Used to call a VM (DTMF) group.	—
	UM Group	Used to call a UM group.	500 or 5000*1

*1 The default floating extension number depends on the value specified for **Numbering Plan** in Easy Setup.
*2 A default floating extension number is provided only up to group 64. The floating extension number for groups 65 and higher must be set explicitly.

Conditions

- It is possible to give names to floating extension numbers.

Installation Manual References

5.4.1 Easy Setup Wizard

PC Programming Manual References

10.6.1 PBX Configuration—[2-6-1] System—Numbering Plan—Main—Extension
11.5.1 PBX Configuration—[3-5-1] Group—Incoming Call Distribution Group—Group Settings—Main
→◆ Floating Extension Number
→◆ Group Name
11.7.2 PBX Configuration—[3-7-2] Group—UM Group—Unit Settings—◆ Floating Extension No.
11.8 PBX Configuration—[3-8] Group—PS Ring Group—◆ Floating Extension Number
13.2 PBX Configuration—[5-2] Optional Device—External Pager—◆ Floating Extension Number
13.3.2 PBX Configuration—[5-3-2] Optional Device—Voice Message—DISA Message—◆ Floating Extension Number

PT Programming Manual References

[623] Incoming Call Distribution Group Name
[660] UM Group Floating Extension Number
[700] External Pager Floating Extension Number
[730] Outgoing Message (OGM) Floating Extension Number
[731] Outgoing Message (OGM) Name
[811] Modem Floating Extension Number
[812] ISDN Remote Floating Extension Number

Feature Guide References

2.28.1 Voice Mail (VM) Group
3.1.1 Unified Messaging System Overview
5.5.2 PC Programming

5.5.9 Software Upgrading

Description

The main software of the PBX, as well as the software of other connected devices can be updated either manually or automatically.

- **Obtaining updates manually**
  Obtaining software updates (downloading the update to the PBX) can be done manually via Web Maintenance Console. In this case, software updates can be obtained from an FTP server, a USB memory device connected to the PBX, a NAS, or a PC that can access Web Maintenance Console.
- **Obtaining updates automatically**
  The PBX can automatically check for and download updates from an FTP server. Also, a notification e-mail can be sent to specified e-mail addresses when an update becomes available and when it is downloaded. Also, the PBX can be configured to check for updates automatically, but not to download them.

Installing an update can be done either manually via Web Maintenance Console, or on a set schedule.
The software of the following types of devices and components can be updated:

Data Type	Description
Main software data	Operating system data area on the PBX's mother board
Expansion Unit software data	Operating system data area on an Expansion Unit's mother board
LPR (software on a slot card) software data	Flash ROM on a slot card (e.g., DHLC4) This includes the LPR software of Expansion Units.
Cell Station (CS) software data	Flash ROM on a CS This includes the Flash ROM on CSs.
IP-PT/SIP extension software data*1	Firmware of supported IP-PTs and SIP extensions

*1 Only Panasonic telephones are supported. For details about a specific telephone, refer to the telephone's documentation.

Conditions

An SD Memory Card (KX-NS3134, KX-NS3135 or KX-NS3136) or USB memory (commercially-supplied) is required to install upgrades of the software above.
The software version of the mother board can be confirmed through system programming.

Installation Manual References

4.3.1 Mother Board

PC Programming Manual References

5.1 System Control—Program Update
9.2.3 PBX Configuration—[1-1] Configuration—Slot—System Property—Slot Summary—◆ Card Type

PT Programming Manual References

[190] Main Processing (MPR) Software Version Reference

5.6 Fault Recovery/Diagnostics

5.6.1 UPS (Uninterruptible Power Supply) Integration

Description

An uninterruptible power supply unit (UPS) is a device that supplies power for several minutes to a connected device when a power failure occurs.

Conditions

- For details about UPS units that are compatible with the automatic shutdown feature of this PBX, consult your dealer. If an incompatible UPS is connected and the UPS runs out of power, the PBX will turn off without shutting down.
- When power is restored after a power outage, the PBX operates in the following manner:
  - If the PBX did not shut down, normal operation continues uninterrupted.
  - If the PBX shut down and power remains in the UPS, the PBX must be started again manually. (The power switch must be turned off and then on again.)
  - If the PBX shut down and no power remains in the UPS, the PBX starts automatically. (This is because the PBX's power switch is on.)

Installation Manual References

4.10 Connection of Peripherals

PC Programming Manual References

4.1.1 Status—Equipment Status—UPS

5.6.2 Power Failure Transfer

Description

When the power supply to the PBX fails, specific SLTs are automatically connected to specific trunks (**Power Failure Connections**). The PBX will switch from normal operation to the Power Failure Connections, and all existing conversations will be disconnected.
Only the trunks handled by Power Failure Connections can be used during a power failure.

Conditions

- Only trunk calls can be made during a power failure. All other features do not work.
- The following SLT ports and LCOT ports are connected during a power failure. It is possible to allow trunk calls that are established during a power failure to be maintained even when the power returns and the connection is switched back to the normal configuration from the Power Failure Connection.

Main Unit	Port 1 on pre-installed MCSLC16 and port 1 on pre-installed LCOT6 Port 2 on pre-installed MCSLC16 and port 2 on pre-installed LCOT6
Expansion Unit	Port 1-4 on pre-installed MCSLC16 and port 1-4 on LCOT6 (if two cards are installed, the lower numbered card is used).

Installation Manual References

4.12 Power Failure Connections

5.6.3 Power Failure Restart

Description

When turning the electricity back on, the PBX restarts the stored data automatically and the PBX will record the event (System Restart) in the error log.

Conditions

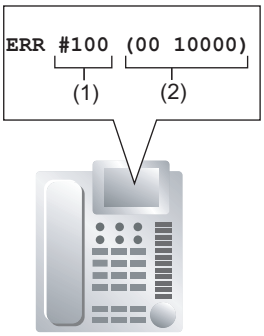
- In the event of a power failure, PBX memory is protected by a factory-provided lithium battery. There is no memory loss except the memories of Automatic Callback Busy (Camp-on) (→ 2.10.1 Automatic Callback Busy (Camp-on)) and Call Park (→ 2.13.2 Call Park).

5.6.4 Local Alarm Information

Description

When a PBX error occurs and the PBX detects it, the System Alarm button light on the PT of an extension, which is allowed to use this feature through system programming (a maximum of two extensions per PBX), turns on red. Pressing the button will show the error number on the display. If multiple errors occur, the error number will be displayed in order of highest priority to lowest. The System Alarm button light turns off automatically after all error numbers have been displayed.
For details about the errors and their solutions, refer to the Installation Manual.

[Error Example]



[Explanation]

Number in the Example	Item	Description
(1)	**Error Code**	Shows three-digit error code.
(2)	**Sub Code**	Shows 8-digit sub code (BBWXYYZZ). **BB:** 00 **W:** Slot type (Physical shelf: blank, Virtual shelf: *) **X:** Unit number/Non-PBX process code **YY:** Slot number/Process code **ZZ:** Port number/Process number

Conditions

- **System Alarm Button**
  A flexible button can be customised as the System Alarm button.
- The alarm information will be recorded on SMDR, if enabled through system programming.
- The PBX can be automatically diagnosed at a preprogrammed time every day.

Installation Manual References

7.1.6 Troubleshooting by Error Log

PC Programming Manual References

7.3.1 Utility—Log—Error Log
12.1.4 PBX Configuration—[4-1-4] Extension—Wired Extension—Flexible Button—◆ Type
12.2.3 PBX Configuration—[4-2-3] Extension—Portable Station—Flexible Button—◆ Type
19.1 PBX Configuration—[11-1] Maintenance—Main

→SMDR—◆ Print Information—Error Log
→Maintenance—◆ Local Alarm Display—Extension 1, Extension 2
→Maintenance—◆ Daily Test Start Time—Set
→Maintenance—◆ Daily Test Start Time—Hour
→Maintenance—◆ Daily Test Start Time—Minute

Feature Guide References

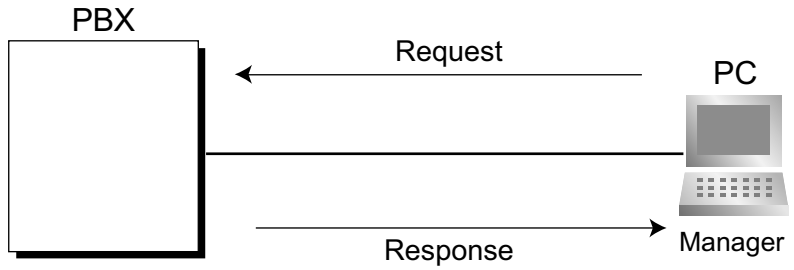
2.21.2 Flexible Buttons
2.22.1.1 Station Message Detail Recording (SMDR)

5.6.5 Simple Network Management Protocol (SNMP) System Monitor

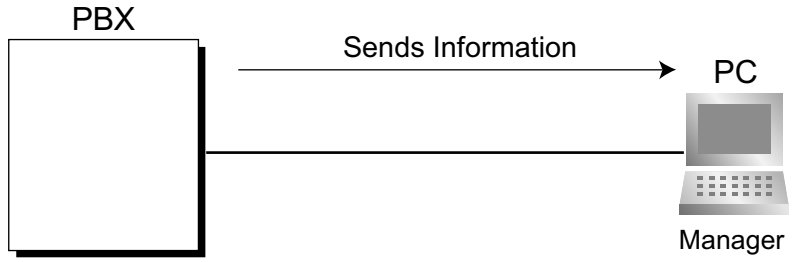
Description

It is possible for a PC assigned as an SNMP manager to manage and receive PBX system status information, such as alarm information and general system activity using SNMP. Management Information Bases (MIBs) are sent to a PC (i.e., the SNMP manager) connected to the PBX over a LAN and can then be stored and analysed using SNMP manager software.
The two features for managing information using SNMP are as follows:

– **Polling:**
A bilateral transaction of information. Polling allows the manager to request information from the PBX.



– **TRAP:**
An automatic relay of information from the PBX when a status change occurs or an alarm is detected.



TRAP Implementation

The PBX will send the two types of TRAP as follows:

Type	TRAP Name	Description
Standard TRAP	coldStart	Information is sent after turning on the power of the PBX or resetting the PBX.
	Authentication Failure	Information is sent when an unregistered Community Name and/or Manager IP address is entered.
Enterprise Specific TRAP*1	Major Alarm	Information is sent when a major alarm is detected.
	Minor Alarm	Information is sent when a minor alarm is detected.

*1 Enterprise Specific TRAPs contain information exclusive to the PBX model (Enterprise Specific MIB).

Conditions

- Through system programming, it is possible to enable or disable this feature.

- Up to 2 SNMP managers can be assigned.
- This PBX supports SNMP Protocol Version 1.0, 2.0c and SNMP Version 1.0-TRAP.
- This PBX can only receive read-only MIBs. Write MIBs are not supported.
- This PBX supports MIB II.
- For more information regarding major and minor alarms, refer to the Installation Manual.
- For a list of the MIB object groups supported by this PBX, refer to 6.4 Supported Management Information Base (MIB) Table in the Appendix.
- Through system programming, it is possible to select whether each type of TRAP (e.g., ColdStart) is sent to the SNMP manager or not.

Installation Manual References

7.1.6 Troubleshooting by Error Log

PC Programming Manual References

27.3.3 Network Service—[3-3] Client Feature—SNMP Agent

Feature Guide References

5.6.4 Local Alarm Information

5.6.6 Dynamic Host Configuration Protocol (DHCP) Assignment

Description

It is possible to assign this PBX as a Dynamic Host Configuration Protocol (DHCP) client, allowing IP addresses to be received from a DHCP server over a LAN.

Conditions

- It is possible to enable this feature through system programming.

Notice

It is important to set your DHCP server to not change the IP addresses of the mother board and DSP cards once IP telephones are registered to the PBX. The IP telephones will not operate properly if these IP addresses are changed.

PC Programming Manual References

27.1 Network Service—[1] IP Address/Ports—DHCP Server
→◆ LAN Setting—Obtain an IP address automatically/Use the following IP address
→◆ LAN Setting—IP Address
→◆ DSP IP Setting—Obtain DSP IP address automatically/Use the following DSP IP address

5.6.7 PING Confirmation

Description

It is possible for this PBX to confirm the connection of IP telephones, routers, and hubs within or outside the private network using PING. The PBX will send an Internet Control Message Protocol (ICMP) echo request through the PC programming terminal and receive an ICMP message confirming connection.

Conditions

- This PBX performs PING as follows:
  - Test packet length: 56 bytes
  - Ping attempts: 5
  - Time out length: 1 second
  - Ping interval time: 1 second

PC Programming Manual References

7.1.2 Utility—Diagnosis—Ping

5.6.7 PING Confirmation

Section 6

Appendix

6.1 Capacity of System Resources

System

Item	Capacity
Absent Message—Extension	1 × 16 characters
Absent Message—System	8 × 16 characters
Call Park Zone	100
Conference	3 – 8 parties per conference 32 parties total
COS	64
DID/DDI Table	32 digits, 1000 entries
Extension number	1 – 5 digits
Extension Personal Identification Number (PIN)	10 digits, 1 entry/extension
Host PBX Access Code	10 digits, 10 entries/trunk group
Number of Characters of Name	20
Printing message	8
Queuing Time Table	64
Ring Tone Pattern Plan	8
Simultaneous Programmers (PT Programming)	• one system programmer + 63 personal programmers • one manager programmer + 63 personal programmers • 64 personal programmers
SMDR Call Storage	1000 calls (Without SD card) / 40,000 calls (with SD card)
Special Carrier Access Code	16 digits, 20 entries
Tenant	8
Time Service Holiday	24
Verification Code	4 digits, 1000 entries
Verification Code Personal Identification Number (PIN)	10 digits, 1000 entries

Dialling

Item	Capacity
Emergency Call	32 digits, 10 entries
Hot Line	32 digits
Key Pad Protocol Dial (ISDN Service Access)	32 digits

Item	Capacity
Personal Speed Dialling	32 digits, 100 entries/extn.
Quick Dialling	8 digits, 4000 entries
Redial	32 digits
System Speed Dialling	32 digits, 1000 entries/tenant
One-touch Dialling—PT	32 digits, 5000 entries/system
One-touch Dialling—PS	32 digits, 1000 entries/system

Groups

Item	Capacity
Conference Group	8 (31 members/group for Conference Group Mode, 31 members/group for Broadcast Mode)
User Group	32
Call Pickup Group	64
Idle Extension Hunting Group	64 (16 extensions/group)
Incoming Call Distribution Group	128 (128 extensions/group)
Paging Group	32
PS Ring Group	32
Trunk Group	64
UM Group	1
VM (DPT) Group	2 units × 12 ports (24 channels)
VM (DTMF) Group	2 groups × 32 channels
P2P Group	32

TRS/Barring

Item	Capacity
TRS/Barring Level	7
TRS/Barring Denied Code	16 digits, 100 entries/level
TRS/Barring Exception Code	16 digits, 100 entries/level

ARS

Item	Capacity
Routing Plan Table	48 entries
Leading Number Table	16 digits, 1000 entries
Leading Number Exception Table	16 digits, 200 entries
ARS Carrier	48
Itemised Billing Code	10 digits
Authorisation Code for Tenant	16 digits
Authorisation Code for Trunk Group	10 digits

Call Log and Message Waiting

Item	Capacity
Outgoing Call Log—PT	100 records/extn. 1520 records/system
Outgoing Call Log—PS	100 records/extn. 640 records/system
Incoming Call Log—PT	100 records/extn. 3040 records/system
Incoming Call Log—PS + Incoming Call Distribution Group	100 records/extn. or group Total 2560 records/system
Message Waiting—PS + Incoming Call Distribution Group	256
Message Waiting—PT + SLT	256

Voice Message

Item	Capacity
Outgoing Message (OGM)	64
OGM Total Recording Time	Approx. 20 minutes
Build-in Simplified Voice Message(SVM)	125 messages
SVM Total Recording Time	120 minutes

Hospitality and Charge Management Features

Item	Capacity
Billing items for guest rooms	1000 records/PBX (Without SD card) / 10,000 records/PBX (with SD card)

Item	Capacity
Hotel Operator	4
Charge Rate	7 digits including a decimal
Charge Denomination	3 currency characters/symbols

Networking

Item	Capacity
TIE Line Routing and Modification Table	32 entries
Leading Number	3 digits
PBX Code	7 digits
NDSS: Monitored PBXs	8
NDSS: Registered Extensions for Monitor PBX	250

Unified Messaging

Item	Capacity
Mailboxes	500 subscriber mailboxes 1 System Manager mailbox 1 Message Manager mailbox
Group Distribution List	User: 4 groups, 40 members per group System: 20 groups, 200 members per group
Service Group	64 entries
Unified Messaging Ports	24 ports

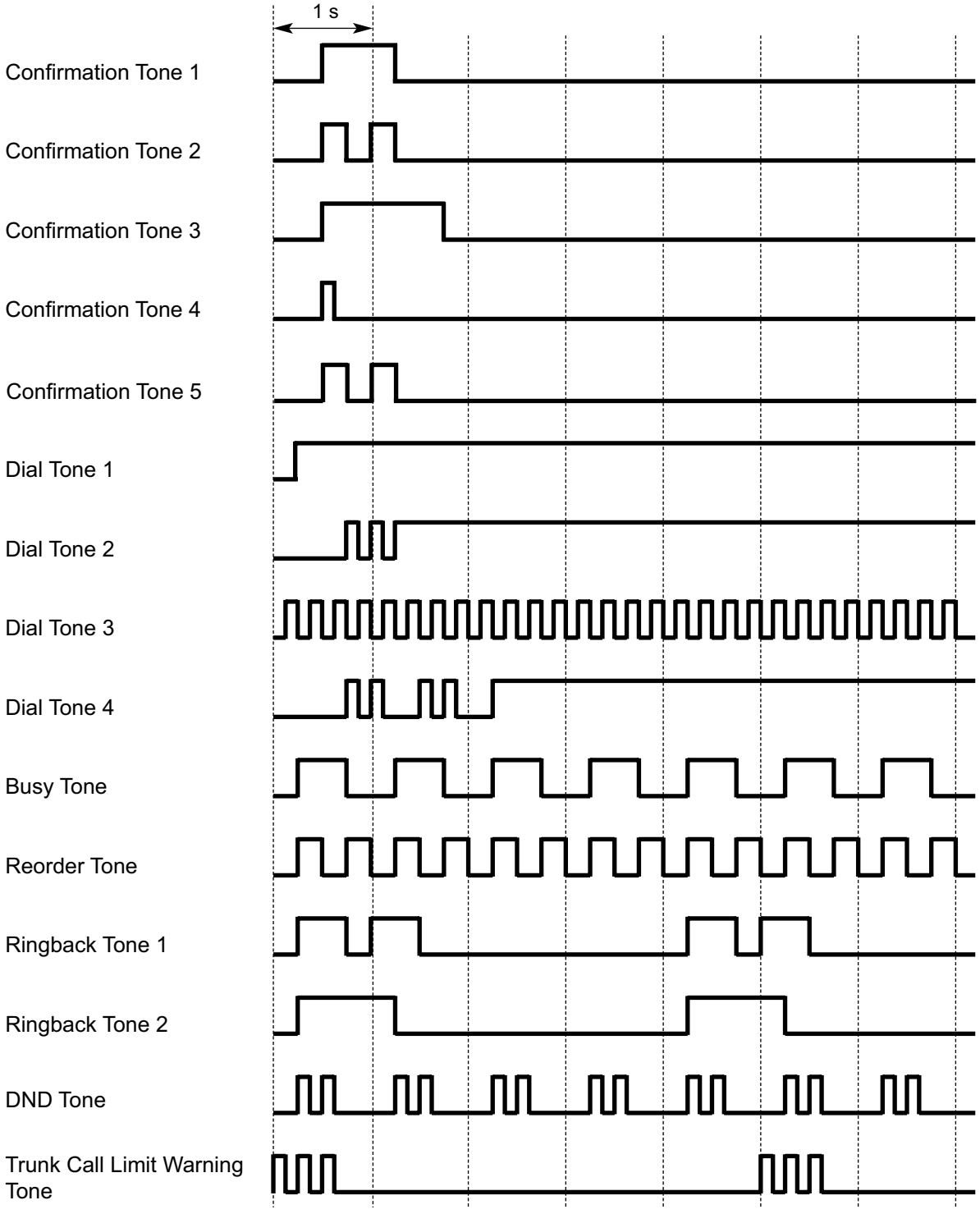
Web Maintenance Console Accounts

Item	Capacity
Users (User)	492
Users (Administrator)	8 accounts
Installer	1 account
Password (all account types)	4 – 16 characters

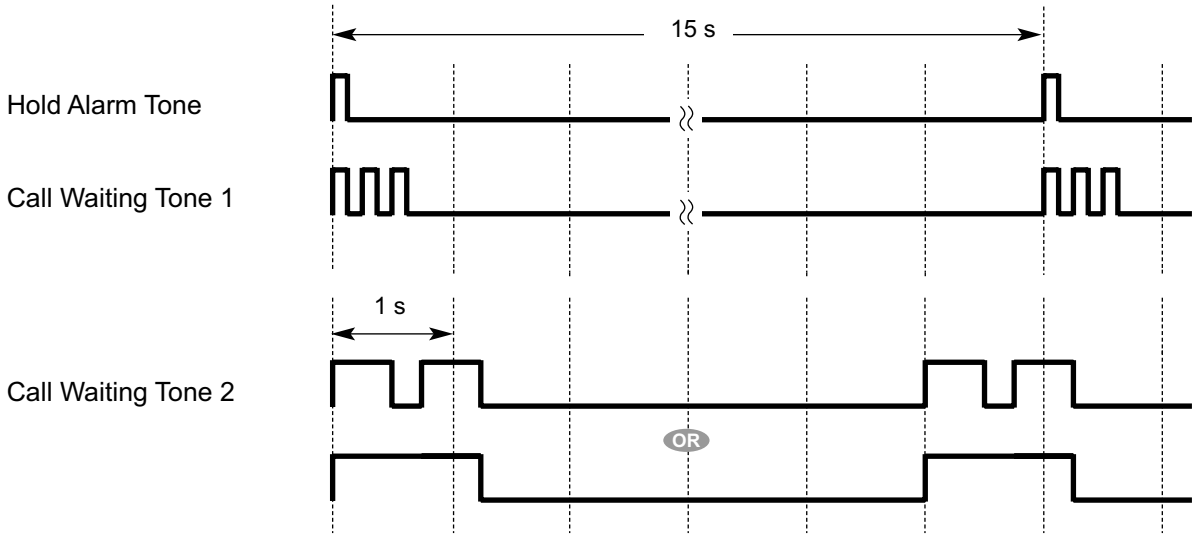
6.2 Tones/Ring Tones

6.2.1 Tones/Ring Tones

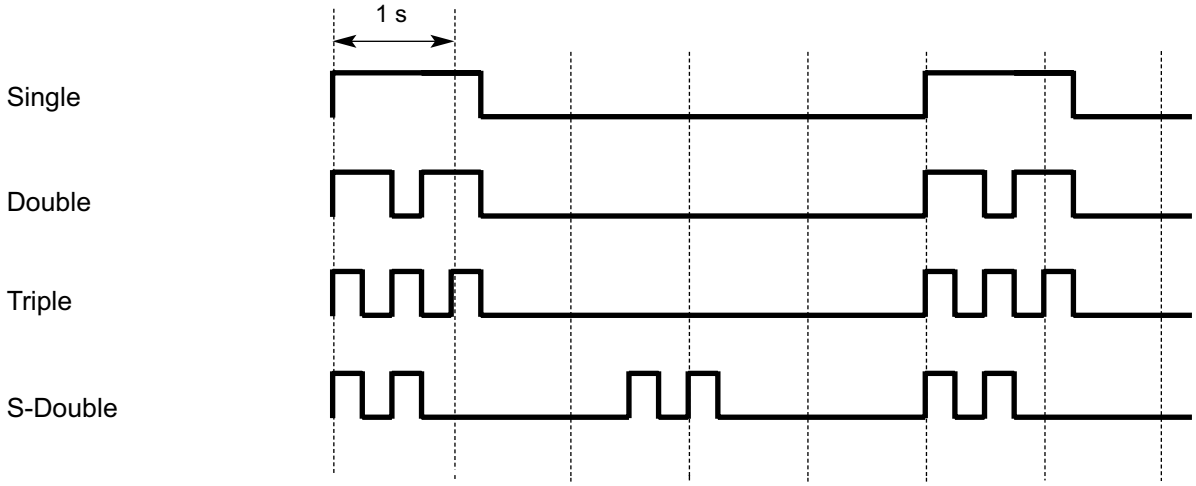
Tone Patterns (Default)



Tone Patterns (Default)



Ring Tone Patterns (Default)*



* The duration of a ring tone may vary by country/area.

6.3 Features that Require Activation Keys

Feature	Required Activation Keys
Outside Destinations in Incoming Call Distribution Group (→ 2.2.2.3 Outside Destinations in Incoming Call Distribution Group)	Activation Key for Mobile Extension (KX-NSE101, KX-NSE105, KX-NSE110, KX-NSE120)
Announces queuing status (number of calls in the waiting queue/number of calls in the waiting queue and estimated waiting time) (→ 2.2.2.4 Queuing Feature)	Activation Key for Call center Enhanced Feature (KX-NSF201)
ACD Supervisory Feature (→ 2.2.2.9 Supervisory Feature (ACD))	Activation Key for Call center Enhanced Feature (KX-NSF201)
Parallel Ringing When Forwarding to Trunk (→ 2.3.2 Call Forwarding (FWD))	Activation Key for Mobile Extension (KX-NSE101, KX-NSE105, KX-NSE110, KX-NSE120)
DISA Automatic Walking COS (→ 2.16.1 Direct Inward System Access (DISA))	Activation Key for Mobile Extension (KX-NSE101, KX-NSE105, KX-NSE110, KX-NSE120)
Computer Telephony Integration (CTI) (→ 2.26.1 Computer Telephony Integration (CTI))	Activation Key for CTI interface (KX-NSF101)
CA (Communication Assistant) (→ 2.26.2 CA (Communication Assistant))	→ Refer to the documentation for CA.
UM Port Expansion (→ 3.1.1 Unified Messaging System Overview)	2-Channel/4-Channel Unified Messaging Activation Key (KX-NSU102, KX-NSU104)
Scheduled Backup for Unified Messaging (→ 3.1.2.5 System Backup/Restore)	Activation Key for Message Backup (KX-NSU003)
Automatic Two-way Recording for Manager (→ 3.2.1.4 Automatic Two-way Recording for Manager)	Activation Key for Two-way Recording Control (KX-NSU002)
Message Waiting Notification—E-mail Device (→ 3.2.1.28 Message Waiting Notification—E-mail Device)	Activation Key for Unified Messaging E-mail Notification (KX-NSU201, KX-NSU205, KX-NSU210, KX-NSU220, KX-NSU299)
Two-way Record/Two-way Transfer (→ 3.2.2.30 Two-way Record/Two-way Transfer)	Activation Key for Two-way Recording (KX-NSU301, KX-NSU305, KX-NSU310, KX-NSU320, KX-NSU399)
Microsoft Outlook Integration (→ 3.3.1 Integration with Microsoft Outlook)	Activation Key for Unified Messaging E-mail Notification (KX-NSU201, KX-NSU205, KX-NSU210, KX-NSU220, KX-NSU299)
IMAP Integration (→ 3.3.2 IMAP Integration)	Activation Key for Unified Messaging E-mail Notification (KX-NSU201, KX-NSU205, KX-NSU210, KX-NSU220, KX-NSU299)

Feature	Required Activation Keys
Common Extension Numbering for 2 PBXs (→ 4.2.1.4 Common Extension Numbering for 2 PBXs)	Activation Key for QSIG Network (KX-NSN002)
Common Extension Numbering for Multiple PBXs (→ 4.2.2.2 Common Extension Numbering for Multiple PBXs)	Activation Key for QSIG Network (KX-NSN002)
QSIG Enhanced Features (→ 4.2.5 QSIG Enhanced Features)	Activation Key for QSIG Network (KX-NSN002)
Network Direct Station Selection (NDSS) (→ 4.2.5.1 Network Direct Station Selection (NDSS))	Activation Key for QSIG Network (KX-NSN002)
Centralised Voice Mail (→ 4.2.5.2 Centralised Voice Mail)	Activation Key for QSIG Network (KX-NSN002)
Network ICD Group (→ 4.2.6 Network ICD Group)	Activation Key for Mobile Extension (KX-NSE101, KX-NSE105, KX-NSE110, KX-NSE120)
PS Roaming by Network ICD Group (→ 4.2.6.1 PS Roaming by Network ICD Group)	Activation Key for Mobile Extension (KX-NSE101, KX-NSE105, KX-NSE110, KX-NSE120)

6.4 Supported Management Information Base (MIB) Table

System Group (1.3.6.1.2.1.1)

Object ID	Item	Description
1	sysDescr	Information of Hardware type and Software version of the Device.
2	sysObjectID	Object identifier of this product.
3	sysUpTime	Elapsed time since the system was restarted.
4	sysContact	Device Administrator.
5	sysName	Name of Device.
6	sysLocation	Installation Location of Device.
7	sysService	Support Layer.

Interface Group (1.3.6.1.2.1.2)

Object ID	Item	Description
1	ifNumber	The number of Network Devices.
2	IfTable (NA)	Management Table by each Network Device.
2.1	IfEntry (NA)	Components of ifTable.
2.1.1	ifIndex	Each interface identifier.
2.1.2	ifDescr	Explanation of Interface.
2.1.3	ifType	Type of Interface.
2.1.4	ifMtu	Maximum Datagram Length which can be sent/received.
2.1.5	ifSpeed	Maximum Transfer Speed.
2.1.6	ifPhysAddress	Physical Address (MAC Address).
2.1.7	ifAdminStatus	The desired state of the interface.
2.1.8	ifOperStatus	The current operational state of the interface.
2.1.9	ifLastChange	The value of sysUpTime at the time the interface entered its current operational state (up or down).
2.1.10	ifInOctets	The number of Octets received.
2.1.11	ifInUcastPkts	The number of Unicast Packets delivered to a higher-layer protocol.
2.1.12	ifInNUcastPkts	The number of Non Unicast Packets delivered to a higher-layer protocol.
2.1.14	ifInErrors	The number of inbound Packets that contained errors.
2.1.15	ifInUnKnownProtos	The number of Packet received which are discarded because of an unknown/unsupported protocol.

Object ID	Item	Description
2.1.16	ifOutOctets	The number of Octets transmitted.
2.1.17	ifOutUcastPkts	The number of Unicast Packets which are received from upper protocol.
2.1.18	ifOutNUcastPkts	The number of Non Unicast Packets which are received from upper protocol.
2.1.21	ifOutQLen	The length of the output packet queue (in packets).
2.1.22	ifSpecific	Relevant MIB object identifier.

IP Group (1.3.6.1.2.1.4)

Object ID	Item	Description
1	ipForwarding	The value which indicates operation availability as a router (whether Datagram is transferred or not).
2	ipDefaultTTL	Default value for IP Packet TTL (Time to Live).
3	ipInReceives	The total number of Packets received (including packet received in error).
4	ipnHdrErrors	The number of Packets discarded due to errors in their header.
5	ipInAddrError	The number of Packets discarded because IP Address of the destination was invalid.
7	ipInUnknownProtos	The number of Packets discarded because the protocol was unknown/unsupported.
8	ipInDiscards	The number of incoming Packets discarded because of an insufficient reception buffer.
9	ipInDelivers	The total number of Packets received (including ICMP) normally.
10	ipOutRequests	The total number of IP Packets (ICMP) which are tried to be transmitted (relay Packet is not included).
13	ipReasmTimeout	The maximum number of seconds required in the buffer to rebuild a fragmented Packet.
14	ipReasmReqds	The number of Packets that required rebuilding from a fragmented state.
15	ipReasmOKs	The number of Packets that were rebuilt correctly from a fragmented state.
16	ipReasmFails	The number of Packets that could not be rebuilt correctly from a fragmented state.
17	ipFragOKs	The number of Packets that were fragmented correctly.
18	ipFragFails	The number of Packets that could not be fragmented correctly.
19	ipFragCreates	The number of IP datagrams created due to fragmentation.
20	ipAddrTable (NA)	Management Table of addressing information relevant to this entity's IP addresses.
20.1	IpAddrEntry (NA)	Components of ipAddrTable.

Object ID	Item	Description
20.1.1	IpAdEntAddr	IP Address.
20.1.2	IpAdEntIfindex	Index value of the Interface which is assigned to IP address.
20.1.3	IpAdEntNetMask	The Subnet Mask associated with IP address.
20.1.4	ipAdEntBcastAddr	Broadcast Address Value associated with IP Address.
20.1.5	IpAdEntReasmMaxSize	The size of the largest IP Datagram which can be sent/received through IP Address.

ICMP Group (1.3.6.1.2.1.5)

Object ID	Item	Description
1	cmpInMsgs	The total number of ICMP messages received (excluded, with error).
2	icmpInErrors	The total number of ICMP messages received which contained error.
8	icmpInEchos	The total number of ICMP echo request messages received.
9	icmpInEchoReps	The total number of ICMP echo answering messages received.
14	icmpOutMsgs	The number of ICMP messages which were sent.
15	icmpOutErrors	The number of ICMP messages which were not sent because of error.
21	icmpOutEchos	The number of ICMP Echo request messages sent.
22	icmpOutEchoReps	The number of ICMP Echo Reply messages sent.

TCP Group (1.3.6.1.2.1.6)

Object ID	Item	Description
1	tcpRtoAlgorithm	The algorithm used to determine the timing of retransmitting when a response was unacknowledged.
2	tcpRtoMin	Minimum value permitted for retransmission timeout (in milliseconds).
3	tcpRtoMax	Maximum value permitted for retransmission timeout (in milliseconds).
4	tcpMaxConn	Maximum number of TCP connections which can be supported.
5	tcpActiveOpens	The total number of Active open TCP connections.
6	tcpPassiveOpens	The total number of Passive open TCP connections.
7	tcpAttemptFails	The total number of connections error.
8	tcpEstabResets	The total number of resets.
10	tcpInSegs	The total number of segments received.
11	tcpOutSegs	The total number of segments sent.
12	tcpRetransSegs	The total number of segments retransmitted.

Object ID	Item	Description
14	tcpInErrs	The total number of segments received in error.
15	tcpOutRsts	The total number of TCP segments sent containing the RST flag (reset connection).

UDP Group (1.3.6.1.2.7)

Object ID	Item	Description
1	udpInDatagrams	The total number of UDP Datagrams received.
2	udpNoPorts	The total number of received UDP Datagrams for which there was no application at the destination port.
3	udpInError	The total number of received UDP Datagrams which contained error.
4	udpOutDatagrams	The total number of UDP Datagrams sent.

SNMP Group (1.3.6.1.2.1.11)

Object ID	Item	Description
1	snmpInPkts	The total number of SNMP messages received.
2	snmpOutPkts	The total number of SNMP messages sent.
3	snmpInBadVersions	The total number of received SNMP messages of which version is unsupported.
4	snmpInBadCommunityNames	The total number of SNMP messages with unknown Community Name.
6	snmpInASNParseErrs	The total number of SNMP messages with incorrect OID type.
13	snmpInTotalReqVars	The total number of Objects which have been retrieved value successfully.
15	snmpInGetRequests	The total number of Get-Request which have been accepted and processed (e.g. Data extract by using snmpget/snmpwalk command).
16	snmpInGetNexts	The total number of Get-Next which have been accepted and processed (for at the time of following a layer-tree by using snmpwalk command).
20	snmpOutTooBigs	The total number of sent SNMP messages which returned an error of "TooBig".
21	snmpOutNoSuchNames	The total number of sent SNMP messages which returned an error of "NoSuchName".
24	snmpOutGenErrs	The total number of sent SNMP messages which returned an error of "GenErr".
28	snmpOutGetResponses	The total number of GetResponse sent.
29	snmpOutTraps	The total number of TRAP sent.

Object ID	Item	Description
30	snmpEnableAuthenTraps	Indicates whether the SNMP agent process is permitted to generate authentication failure traps.

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Web Site: http://www.panasonic.net/

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