Thank you for purchasing a Panasonic Pure IP-PBX. Please read this manual carefully before using this product and save this manual for future use.

KX-NCP500/KX-NCP1000: PBMPR Software File Version 1.0000 or later
Introduction

About this Feature Guide

This Feature Guide is designed to serve as an overall feature reference for the Panasonic Pure 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:

Section 1, Call Handling Features
Provides details about the call handling features.

Section 2, System Configuration and Administration Features
Provides details about the system configuration and administration features.

Section 3, 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.

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 PTs, SLTs, PSs, or DSS Consoles.

**Safety Notices**
Please observe the safety notices in this manual in order to avoid danger to users or other people, and prevent damage to property.
The notices are classified as follows, according to the severity of injury or damage:

<table>
<thead>
<tr>
<th>WARNING</th>
<th>This notice means that misuse could result in death or serious injury.</th>
</tr>
</thead>
<tbody>
<tr>
<td>CAUTION</td>
<td>This notice means that misuse could result in injury or damage to property.</td>
</tr>
</tbody>
</table>

The KX-NCP500UK/KX-NCP1000UK, KX-NCP500NE/KX-NCP1000NE, and KX-NCP500GR/KX-NCP1000GR are designed to interwork with the:
- Analogue Public Switched Telephone Network (PSTN) of European countries
- Pan-European Integrated Services Digital Network (ISDN) using ISDN basic rate access
- Pan-European Integrated Services Digital Network (ISDN) using ISDN primary rate access
- ONP 2048 kbit/s digital structured leased lines (D2048S)

Panasonic Communications Co., Ltd./Panasonic Communications Company (U.K.) Ltd. declares that this equipment is in compliance with the essential requirements and other relevant provisions of Radio & Telecommunications Terminal Equipment (R&TTE) Directive 1999/5/EC.

Declarations of Conformity for the relevant Panasonic products described in this manual are available for download by visiting:

[http://www.doc.panasonic.de](http://www.doc.panasonic.de)

Contact to Authorised Representative:
Panasonic Testing Centre
Panasonic Marketing Europe GmbH
Winsbergering 15, 22525 Hamburg, Germany

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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 2.7.1 Frequently Asked Questions (FAQ) of the PC Programming Manual, or [190] Main Processing (MPR) Software Version Reference in the PT Programming Manual.
• Some optional service cards, PTs, and features are not available in some areas. Please consult your certified Panasonic dealer for more information.
• Product specifications are subject to change without notice. In some cases, additional information, including updates to this and other manuals, is included in the Maintenance Console’s Information before programming. Install the latest version of Maintenance Console to view this 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, model number suffixes (e.g., KX-NCP500NE) are omitted unless necessary.
• All system programming can be performed through PC programming (→ 2.3.1 PC Programming). However, only a subset can be performed through PT programming (→ 2.3.2 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.

→ 8.1 [6-1] 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.
Feature Highlights

IPCMPR Card Features
The IPCMPR card (the preinstalled Main Processing card) includes the following features and resources:

VoIP Trunks/VoIP Extensions
The IPCMPR card allows the use of IP trunks (H.323/SIP) and IP telephones (IP-PTs, IP softphones, and SIP extensions). By connecting a DSP card to the IPCMPR card, these IP trunks and extensions can be used. It is also possible to increase the number of IP trunks and extensions through the purchase of activation key codes.

LAN Connection
The IPCMPR card is equipped with a LAN port allowing for the PBX to be connected to a LAN.

Computer Telephony Integration (CTI) Features
Connecting a PC to a DPT, or connecting a CTI Server to this PBX, allows functions of the PC, PBX and extensions to be integrated. For example, detailed caller information can be taken from a database and displayed on the PC as a call arrives, or the PC can dial numbers for the extension automatically. (→ 1.32.1 Computer Telephony Integration (CTI))

Built-in ESVM Card Features
The IPCMPR card comes preinstalled with the ESVM card feature (2 channels), allowing for simple answering machine services to be supported. (→ 1.24.1 Built-in Simplified Voice Message (SVM))

Built-in Communication Assistant (CA) Server
The IPCMPR card comes preinstalled with an Activation Key for CA Basic for 5 Users, allowing for serverless 1st Party CTI call control. If a PC containing the CA application is connected to the PBX, the user can access many useful call features such as Presence (viewing the current status of other CA users). (→ 1.32.3 CA (Communication Assistant))

Internet Protocol (IP) Telephone Features
This PBX supports the following telephone features that use IP:

IP-PT Features
This PBX supports the connection of IP-PTs which function almost identically to normal PTs. However, they connect to the PBX over a LAN and make and receive calls using IP. In addition, KX-NX300 series IP-PTs support the following features:

– Self Labelling (KX-NX366 only): The LCDs featured next to the 12 flexible buttons of the KX-NX366 can be programmed to display information regarding the setting of the corresponding flexible button. In addition, the flexible buttons can be organised into 4 "pages" allowing you to assign a total of 4 items to each physical button. (→ 1.31.4 Self Labelling (KX-NX366 only))

– Optional Modules: (→ 1.31.1 IP Proprietary Telephone (IP-PT))
  • KX-NX307(PSLP1528) Bluetooth® Module: Allows for a Bluetooth wireless headset to be connected to a KX-NX300 series (except KX-NX321) IP-PT.
  • KX-NX303 Add-on 12 Key Module: Allows for 12 additional flexible buttons to be connected to a KX-NX346 or KX-NX343.
  • KX-NX305 Add-on 60 Key Module: Allows for 60 additional flexible buttons to be connected to a KX-NX346 or KX-NX343.

– High Sound Quality: Calls between KX-NX300 series IP-PTs support G.722 CODEC allowing for high sound quality conversations. (→ 1.31.3 Peer-to-Peer Connection)

SIP (Session Initiation Protocol) Extensions
This PBX supports the connection of third party SIP supported IP telephones (hardphones and softphones). SIP Extensions make and receive calls using IP. (→ 1.31.2 SIP (Session Initiation Protocol) Extension)

IP Softphones
This PBX supports the connection of Panasonic IP softphones, providing IP-PT operations and features using software on a PC.
Internet Protocol (IP) Related Features

Server Connection
This PBX supports the connection to the following servers, allowing for enhanced efficiency and record keeping:

– **SNMP Server:** Allows for users assigned as SNMP managers to manage and receive PBX system status information, such as local alarm information and general system activity using Simple Network Management Protocol (SNMP). (→ 2.4.4 Simple Network Management Protocol (SNMP) System Monitor)

– **SNTP Server:** Allows for the PBX to automatically receive and update the time setting using Simple Network Time Protocol (SNTP). (→ 2.3.5 Automatic Setup)

– **DHCP Server:** Allows for this PBX to receive IP addresses from a Dynamic Host Configuration Protocol (DHCP) server. (→ 2.4.5 Dynamic Host Configuration Protocol (DHCP) Assignment)

Peer-to-peer Connection
Intercom calls can be made between peer-to-peer compatible IP extensions allowing for communication without using the PBX’s VoIP resources. (→ 1.31.3 Peer-to-Peer Connection)

Networking Features
This PBX supports the following networking features:

Voice over Internet Protocol (VoIP) Network
The PBX can connect to another PBX via a private IP network. In this case, voice signals are converted into IP packets and sent through this network. Automatic rerouting of VoIP calls to public trunks is also available, in case of network difficulties. (→ 1.30.2 Voice over Internet Protocol (VoIP) Network)

TIE Line Service
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. (→ 1.30.1 TIE Line Service)

Virtual Private Network (VPN)
VPN is a service provided by the VPN service provider. It uses an existing public line as if it were a private line. (→ 1.30.3 Virtual Private Network (VPN))

QSIG Network
QSIG is a protocol based on ISDN (Q.931) and offers enhanced PBX features in a private network of two or more connected PBXs. (→ 1.30.4 QSIG Standard Features)

External Device Support
The following are just a few of the many useful optional devices available to enhance your telephony system:

Portable Station (PS) Features
PSs can be connected to this PBX. It is possible to use the PBX features using the PS like a PT. A PS can also be used in parallel with a wired telephone (Wireless XDP Parallel Mode). In this case, the wired telephone is the main telephone and the PS is the sub telephone. (→ 1.25 Portable Station (PS) Features)

Cellular Phone Features (KX-NCS3910 required)
This PBX supports the use of cellular phones destinations with the PBX. In combination with the use of certain features such as Virtual PS, DISA, and XDP Parallel mode, cellular phones can make and receive calls as if they were registered extensions. (→ 1.33.1 Cellular Phone Features—SUMMARY)

Phone Assistant (PA) Pro Series
This PBX supports Phone Assistant Pro Series applications, a collection of highly-intuitive CTI applications that allow you to use your PC to access a variety of PBX features conveniently and efficiently. PA Pro series applications combine the efficiency and user-friendliness of CTI applications with the convenience of server-based applications, allowing User IDs, passwords and PBX settings to be configured over a web browser.
Voice Mail Features
This PBX supports Voice Processing Systems (VPS) with DTMF Integration as well as DPT (Digital) Integration, connected locally and over a network. (→ 1.24 Voice Message Features)

ACD Report Server (KX-NCV200)
This PBX supports VPSs with an ACD Report Server. In addition to VPS features, the ACD Report Server provides you with a variety of features, such as monitoring, reporting, call logging features and creating performance graphs, to manage call information efficiently.

Other Useful Features
Easy Installation and Setup
– With Physical Cards:
  By simply connecting telephones and outside lines to analogue extension and trunk cards, you can make and receive calls. For digital trunks (e.g., E1, T1, etc.), it may be necessary to first make the required settings and restart the PBX.
– With Virtual Cards:
  By connecting the PBX to the LAN, installing each virtual card and registering the IP telephones through system programming, you can make and receive calls.

Enhanced Walking Extension Feature
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 users to use their extensions at home and at work, or for multiple users who work in shifts to use their own extension settings on a single telephone. (→ 1.28.3.2 Enhanced Walking Extension)

Built-in Small Call Centre Features
An incoming call distribution (ICD) group (→ 1.2.2 Incoming Call Distribution Group Features) can be used as a small call centre with the following features:
– Queuing Feature
  When a preprogrammed number of extensions in an ICD group are busy, additional incoming calls can wait in a queue. While callers are waiting in the queue, they can be sent an outgoing message (OGM) informing the caller of the current situation, or be redirected to another destination or disconnected depending on the amount of time the caller has been waiting. (→ 1.2.2.4 Queuing Feature)
– Log-in/Log-out
  ICD group members can join (Log-in) or leave (Log-out) their groups manually. After a call is completed, a preprogrammed time period where users will not receive calls can be assigned (Wrap-up). This gives time for users to make reports about the previous call without being interrupted or allows users to not receive calls while they are taking a break. (→ 1.2.2.7 Log-in/Log-out)
– VIP Call
  The VIP Call feature is one method of making sure that calls from preferred customers or callers are answered quickly. When using VIP Call mode, ICD groups are assigned a priority, allowing calls in higher-priority groups to be answered before calls in lower-priority groups. (→ 1.2.2.5 VIP Call).

Hospitality Features
This PBX has several features allowing it to be used effectively and efficiently in a hotel-type environment. These features allow extensions assigned as hotel operators to change and monitor the room status of guest rooms, check or set wake-up calls, and print out records of guest charges. (→ 1.27.1 Hospitality Features—SUMMARY)
# List of Abbreviations

<table>
<thead>
<tr>
<th>Abbreviation</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>Automated Attendant</td>
</tr>
<tr>
<td>AA</td>
<td>Automatic Call Distribution</td>
</tr>
<tr>
<td>ANI</td>
<td>Automatic Number Identification</td>
</tr>
<tr>
<td>AOC</td>
<td>Advice of Charge</td>
</tr>
<tr>
<td>APT</td>
<td>Analogue Proprietary Telephone</td>
</tr>
<tr>
<td>ARS</td>
<td>Automatic Route Selection</td>
</tr>
<tr>
<td>B</td>
<td>Background Music</td>
</tr>
<tr>
<td>BRI</td>
<td>Basic Rate Interface</td>
</tr>
<tr>
<td>C</td>
<td>Communication Assistant</td>
</tr>
<tr>
<td>CCBs</td>
<td>Completion of Calls to Busy Subscriber</td>
</tr>
<tr>
<td>CDPG</td>
<td>Call Distribution Port Group</td>
</tr>
<tr>
<td>CF</td>
<td>Call Forwarding—by ISDN</td>
</tr>
<tr>
<td>CLI</td>
<td>Calling Line Identification</td>
</tr>
<tr>
<td>CLIP</td>
<td>Calling Line Identification Presentation</td>
</tr>
<tr>
<td>CLIR</td>
<td>Calling Line Identification Restriction</td>
</tr>
<tr>
<td>CNIP</td>
<td>Calling Name Identification Presentation</td>
</tr>
<tr>
<td>CNIR</td>
<td>Calling Name Identification Restriction</td>
</tr>
<tr>
<td>COLP</td>
<td>Connected Line Identification Presentation</td>
</tr>
<tr>
<td>COLR</td>
<td>Connected Line Identification Restriction</td>
</tr>
<tr>
<td>CONP</td>
<td>Connected Name Identification Presentation</td>
</tr>
<tr>
<td>CONR</td>
<td>Connected Name Identification Restriction</td>
</tr>
<tr>
<td>COS</td>
<td>Class of Service</td>
</tr>
<tr>
<td>CPC</td>
<td>Calling Party Control</td>
</tr>
<tr>
<td>CS</td>
<td>Cell Station</td>
</tr>
<tr>
<td>CT</td>
<td>Call Transfer—by ISDN</td>
</tr>
<tr>
<td>CTI</td>
<td>Computer Telephony Integration</td>
</tr>
<tr>
<td>D</td>
<td>Direct Dialling In</td>
</tr>
<tr>
<td>DDI</td>
<td>Dynamic Host Configuration Protocol</td>
</tr>
<tr>
<td>DID</td>
<td>Direct Inward Dialling</td>
</tr>
<tr>
<td>DIL</td>
<td>Direct In Line</td>
</tr>
<tr>
<td>DISA</td>
<td>Direct Inward System Access</td>
</tr>
<tr>
<td>DND</td>
<td>Do Not Disturb</td>
</tr>
<tr>
<td>DPT</td>
<td>Digital Proprietary Telephone</td>
</tr>
<tr>
<td>DSS</td>
<td>Direct Station Selection</td>
</tr>
<tr>
<td>DTMF</td>
<td>Dual Tone Multi-Frequency</td>
</tr>
<tr>
<td>E</td>
<td>External Feature Access</td>
</tr>
<tr>
<td>EFA</td>
<td></td>
</tr>
</tbody>
</table>
List of Abbreviations

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
MSN
Multiple Subscriber Number

N
NDSS
Network Direct Station Selection

O
OGM
Outgoing Message
OHCA
Off-hook Call Announcement

OPX
Off Premise Extension

P
PDN
Primary Directory Number
PIN
Personal Identification Number
PING
Packet Internet Groper
P-MP
Point-to-multipoint
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
SNTP
Simple Network Time Protocol
SVM
Built-in Simplified Voice Message

T
TAFAS
Trunk Answer from Any Station
TEI
Terminal Endpoint Identifier
List of Abbreviations

TRG
Trunk Group

TRS/Barring
Toll Restriction/Call Barring

UCD
Uniform Call Distribution

VM
Voice Mail

VoIP
Voice over Internet Protocol

VPN
Virtual Private Network

VPS
Voice Processing System

XDP
EXtra Device Port

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Scheduling: 810-395-9111

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# Table of Contents

## 1 Call Handling Features

### 1.1 Incoming Call Features

| 1.1.1 | Incoming Trunk Call Features | 18 |
| 1.1.1.1 | Incoming Trunk Call Features—SUMMARY | 18 |
| 1.1.1.2 | Direct In Line (DIL) | 21 |
| 1.1.1.3 | Direct Inward Dialling (DID)/Direct Dialling In (DDI) | 23 |
| 1.1.1.4 | Multiple Subscriber Number (MSN) Ringing Service | 26 |
| 1.1.1.5 | Calling Line Identification (CLI) Distribution | 29 |
| 1.1.1.6 | Intercept Routing | 31 |
| 1.1.1.7 | Intercept Routing—No Destination | 36 |
| 1.1.2 | Internal Call Features | 37 |
| 1.1.2.1 | Internal Call Features—SUMMARY | 37 |
| 1.1.2.2 | Internal Call Block | 39 |
| 1.1.3 | Incoming Call Indication Features | 41 |
| 1.1.3.1 | Incoming Call Indication Features—SUMMARY | 41 |
| 1.1.3.2 | Ring Tone Pattern Selection | 42 |
| 1.1.3.3 | Call Waiting | 44 |

### 1.2 Receiving Group Features

| 1.2.1 | Idle Extension Hunting | 46 |
| 1.2.2 | Incoming Call Distribution Group Features | 48 |
| 1.2.2.1 | Incoming Call Distribution Group Features—SUMMARY | 48 |
| 1.2.2.2 | Group Call Distribution | 52 |
| 1.2.2.3 | Outside Destinations in Incoming Call Distribution Group | 56 |
| 1.2.2.4 | Queuing Feature | 58 |
| 1.2.2.5 | VIP Call | 61 |
| 1.2.2.6 | Overflow Feature | 62 |
| 1.2.2.7 | Log-in/Log-out | 64 |
| 1.2.2.8 | Supervisory Feature | 67 |
| 1.2.3 | Call Distribution Port Group | 69 |
| 1.3 | Call Forwarding (FWD)/Do Not Disturb (DND) Features | 71 |
| 1.3.1 | Call Forwarding (FWD)/Do Not Disturb (DND) | 71 |
| 1.3.1.1 | Call Forwarding (FWD)/Do Not Disturb (DND)—SUMMARY | 71 |
| 1.3.1.2 | Call Forwarding (FWD) | 72 |
| 1.3.1.3 | Do Not Disturb (DND) | 76 |
| 1.3.1.4 | FWD/DND Button, Group FWD Button | 78 |

### 1.4 Answering Features

| 1.4.1 | Answering Features | 81 |
| 1.4.1.1 | Answering Features—SUMMARY | 81 |
| 1.4.1.2 | Line Preference—Incoming | 82 |
| 1.4.1.3 | Call Pickup | 83 |
| 1.4.1.4 | Hands-free Answerback | 85 |

### 1.5 Making Call Features

| 1.5.1 | Predialling | 87 |
| 1.5.2 | Automatic Extension Release | 88 |
| 1.5.3 | Intercom Call | 89 |
| 1.5.4 | Trunk Call Features | 91 |
| 1.5.4.1 | Trunk Call Features—SUMMARY | 91 |
| 1.5.4.2 | Emergency Call | 92 |
| 1.5.4.3 | Account Code Entry | 93 |
| 1.5.4.4 | Dial Type Selection | 95 |
| 1.5.4.5 | Reverse Circuit | 96 |
| 1.5.4.6 | Trunk Busy Out | 97 |
# Table of Contents

1.5.4.7 Pause Insertion ................................................................. 98
1.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX) ................................................................. 99
1.5.4.9 Special Carrier Access Code ........................................... 101
1.5.5 Seizing a Line Features ...................................................... 102
1.5.5.1 Seizing a Line Features—SUMMARY .............................. 102
1.5.5.2 Line Preference—Outgoing ............................................. 103
1.5.5.3 Trunk Access ................................................................. 104

1.6 Memory Dialling Features .................................................. 106
1.6.1 Memory Dialling Features ................................................. 106
1.6.1.1 Memory Dialling Features—SUMMARY ......................... 106
1.6.1.2 One-touch Dialling ....................................................... 108
1.6.1.3 KX-T7710 One-touch Dialling ....................................... 109
1.6.1.4 Last Number Redial ...................................................... 110
1.6.1.5 Speed Dialling—Personal/System ................................. 112
1.6.1.6 Hot Line ................................................................. 115

1.7 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Features ............................................................. 116
1.7.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension ................................................................. 116

1.8 Busy Line/Busy Party Features ........................................... 122
1.8.1 Automatic Callback Busy (Camp-on) ................................. 122
1.8.2 Executive Busy Override .................................................. 123
1.8.3 Call Monitor ................................................................. 124
1.8.4 Second Call Notification to Busy Extension ...................... 126
1.8.4.1 Second Call Notification to Busy Extension—SUMMARY .... 126
1.8.4.2 Call Waiting Tone ....................................................... 128
1.8.4.3 Off-hook Call Announcement (OHCA) ......................... 129
1.8.4.4 Whisper OHCA .......................................................... 130

1.9 Toll Restriction (TRS)/Call Barring (Barring) Features .......... 131
1.9.1 Toll Restriction (TRS)/Call Barring (Barring) ....................... 131
1.9.2 Budget Management ....................................................... 139
1.9.3 Extension Dial Lock ....................................................... 140
1.9.4 Dial Tone Transfer .......................................................... 141
1.9.5 Walking COS ............................................................... 143
1.9.6 Verification Code Entry .................................................... 145

1.10 Automatic Route Selection (ARS) Features .......................... 147
1.10.1 Automatic Route Selection (ARS) ..................................... 147

1.11 Conversation Features ...................................................... 153
1.11.1 Hands-free Operation ..................................................... 153
1.11.2 Off-hook Monitor .......................................................... 154
1.11.3 Mute ................................................................. 155
1.11.4 Headset Operation ........................................................ 156
1.11.5 Data Line Security ........................................................ 157
1.11.6 Flash/Recall/Terminate .................................................. 158
1.11.7 External Feature Access (EFA) ........................................ 160
1.11.8 Trunk Call Limitation ..................................................... 161
1.11.9 Paralled Telephone ....................................................... 163
1.11.10 Calling Party Control (CPC) Signal Detection .................. 166

1.12 Transferring Features ....................................................... 167
1.12.1 Call Transfer ............................................................... 167

1.13 Holding Features ............................................................. 171
1.13.1 Call Hold ................................................................. 171
1.13.2 Call Park ................................................................. 174
1.24.2.3 Voice Mail DPT (Digital) Integration ................................................................. 274
1.25 Portable Station (PS) Features .............................................................................. 280
  1.25.1 Portable Station (PS) Connection ................................................................. 280
  1.25.2 PS Ring Group .............................................................................................. 282
  1.25.3 PS Directory ................................................................................................. 286
  1.25.4 PS Feature Buttons ....................................................................................... 287
  1.25.5 Wireless XDP Parallel Mode .......................................................................... 288
  1.25.6 Virtual PS ..................................................................................................... 291
1.26 Administrative Information Features ................................................................. 293
  1.26.1 Record Log Features ...................................................................................... 293
  1.26.1.1 Station Message Detail Recording (SMDR) .................................................. 293
  1.26.1.2 Syslog Record Management ....................................................................... 302
  1.26.2 Printing Message .......................................................................................... 303
  1.26.3 Call Charge Services .................................................................................... 304
1.27 Hospitality Features ............................................................................................. 307
  1.27.1 Hospitality Features—SUMMARY ................................................................. 307
  1.27.2 Room Status Control .................................................................................... 308
  1.27.3 Call Billing for Guest Room ........................................................................... 311
1.28 Extension Controlling Features .......................................................................... 314
  1.28.1 Extension Personal Identification Number (PIN) ........................................... 314
  1.28.2 Extension Feature Clear ................................................................................ 316
  1.28.3 Walking Extension Features ......................................................................... 318
  1.28.3.1 Walking Extension ...................................................................................... 318
  1.28.3.2 Enhanced Walking Extension ..................................................................... 319
  1.28.4 Timed Reminder ........................................................................................... 321
1.29 Audible Tone Features ......................................................................................... 323
  1.29.1 Dial Tone ..................................................................................................... 323
  1.29.2 Confirmation Tone ........................................................................................ 325
1.30 Networking Features ............................................................................................ 327
  1.30.1 TIE Line Service ........................................................................................... 327
  1.30.2 Voice over Internet Protocol (VoIP) Network ................................................ 349
  1.30.3 Virtual Private Network (VPN) .................................................................. 353
  1.30.4 QSIG Standard Features .............................................................................. 355
  1.30.4.1 QSIG Standard Features—SUMMARY ...................................................... 355
  1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG ... 357
  1.30.4.3 Call Forwarding (CF)—by QSIG ............................................................... 360
  1.30.4.4 Call Transfer (CT)—by QSIG ................................................................. 362
  1.30.4.5 Completion of Calls to Busy Subscriber (CCBS)—by QSIG ......................... 364
  1.30.5 QSIG Enhanced Features .......................................................................... 365
  1.30.5.1 Network Direct Station Selection (NDSS) ................................................. 365
  1.30.5.2 Centralised Voice Mail ............................................................................. 370
  1.30.6 Network ICD Group .................................................................................... 374
  1.30.6.1 PS Roaming by Network ICD Group .......................................................... 376
1.31 IP Telephone Features ......................................................................................... 378
  1.31.1 IP Proprietary Telephone (IP-PT) .................................................................. 378
  1.31.2 SIP (Session Initiation Protocol) Extension .................................................... 380
  1.31.3 Peer-to-Peer Connection .............................................................................. 382
  1.31.4 Self Labelling (KX-NT366 only) ................................................................. 384
1.32 Computer Telephony Integration (CTI) Features ................................................ 386
  1.32.1 Computer Telephony Integration (CTI) ...................................................... 386
  1.32.2 PC Phone/PC Console ............................................................................... 389
  1.32.3 CA (Communication Assistant) ................................................................. 391
1.33 Cellular Phone Features ..................................................................................... 392
  1.33.1 Cellular Phone Features—SUMMARY ....................................................... 392
2 System Configuration and Administration Features .........................393
  2.1 System Configuration—Hardware ..................................................394
    2.1.1 Extension Port Configuration ....................................................394
  2.2 System Configuration—Software ..................................................396
    2.2.1 Class of Service (COS) ...............................................................396
    2.2.2 Group .......................................................................................398
    2.2.3 Tenant Service ...........................................................................402
    2.2.4 Time Service .............................................................................406
    2.2.5 Operator Features ......................................................................410
    2.2.6 Manager Features ......................................................................411
  2.3 System Data Control .................................................................413
    2.3.1 PC Programming .......................................................................413
    2.3.2 PT Programming .......................................................................416
    2.3.3 Password Security ......................................................................418
    2.3.4 Quick Setup ...............................................................................420
    2.3.5 Automatic Setup ........................................................................421
    2.3.6 Flexible Numbering/Fixed Numbering ........................................423
    2.3.7 Floating Extension .....................................................................429
    2.3.8 Software Upgrading .................................................................431
  2.4 Fault Recovery/Diagnostics .........................................................432
    2.4.1 Power Failure Transfer ...............................................................432
    2.4.2 Power Failure Restart .................................................................433
    2.4.3 Local Alarm Information .............................................................434
    2.4.4 Simple Network Management Protocol (SNMP) System Monitor ..................................................................................436
    2.4.5 Dynamic Host Configuration Protocol (DHCP) Assignment ..................................................................................438
    2.4.6 Packet Internet Groper (PING) Confirmation ..................................................................................439
  3 Appendix .......................................................................................441
    3.1 Capacity of System Resources .....................................................442
    3.2 Tones/Ring Tones .................................................................446
      3.2.1 Tones/Ring Tones ....................................................................446
    3.3 Supported Management Information Base (MIB) Table .................448
  Index ................................................................................................453
Section 1

Call Handling Features
## 1.1.1 Incoming Trunk Call Features

### 1.1.1.1 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 IPCMPR card can be assigned a networking type: Public, Private, or VPN (Virtual Private Network).

**Table:**

<table>
<thead>
<tr>
<th>Card Type</th>
<th>Channel/Protocol Type</th>
<th>Networking Type</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Public (DIL/DID/DDI/MSN)</td>
</tr>
<tr>
<td>IPCMPR (V-IPGW)</td>
<td>H.323</td>
<td>✓*</td>
</tr>
<tr>
<td>IPCMPR (V-SIPGW)</td>
<td>SIP</td>
<td>✓*</td>
</tr>
<tr>
<td>LCOT</td>
<td>—</td>
<td>✓*</td>
</tr>
<tr>
<td>T1</td>
<td>LCOT</td>
<td>✓*</td>
</tr>
<tr>
<td></td>
<td>GCOT</td>
<td>✓*</td>
</tr>
<tr>
<td></td>
<td>DID</td>
<td>✓*</td>
</tr>
<tr>
<td></td>
<td>TIE (E &amp; M)</td>
<td>✓</td>
</tr>
<tr>
<td></td>
<td>OPX (EXTN.)</td>
<td>✓</td>
</tr>
<tr>
<td>E1</td>
<td>DR2</td>
<td>✓*</td>
</tr>
<tr>
<td></td>
<td>E &amp; M-C</td>
<td>✓</td>
</tr>
<tr>
<td></td>
<td>E &amp; M-P</td>
<td>✓</td>
</tr>
<tr>
<td>BRI/PRI</td>
<td>CO</td>
<td>✓*</td>
</tr>
</tbody>
</table>

✓*: Enable (default); ✓: Enable

1. 3.33 [1-1] Slot—Port Property - T1 Port—Trunk Property
2. 3.39 [1-1] Slot—Port Property - E1 Port—Trunk Property
3. 12.2 [10-2] DIL Table & Port Settings—DIL—Trunk Property

---

**: Enable (default); ✓: Enable

1. 1.30.1 TIE Line Service
2. 1.30.3 Virtual Private Network (VPN)
2. Distribution Method
One of the following methods can be assigned to each trunk port:

<table>
<thead>
<tr>
<th>Method</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Direct In Line (DIL)</td>
<td>Directs a call to a preprogrammed single destination (e.g., the operator).</td>
</tr>
<tr>
<td></td>
<td>→ 1.1.1.2 Direct In Line (DIL)</td>
</tr>
<tr>
<td>Direct Inward Dialling (DID)</td>
<td>Directs a call with a DID number from a DID line to a preprogrammed destination.</td>
</tr>
<tr>
<td></td>
<td>DID is also known as Direct Dialling In (DDI).</td>
</tr>
<tr>
<td></td>
<td>→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)</td>
</tr>
<tr>
<td>Multiple Subscriber Number (MSN) Ringing Service</td>
<td>Directs a call with an MSN from an ISDN line to a preprogrammed destination.</td>
</tr>
<tr>
<td></td>
<td>→ 1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service</td>
</tr>
</tbody>
</table>

3. Destination Change with the Caller’s Identification Number
The Calling Line Identification (CLI) Distribution feature works in conjunction with the DIL/DID/DDI/MSN features.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Line Identification (CLI) Distribution</td>
<td>Directs a call to a CLI destination if the caller’s identification number has been assigned in the Caller ID Table.</td>
</tr>
<tr>
<td></td>
<td>→ 1.1.1.5 Calling Line Identification (CLI) Distribution</td>
</tr>
</tbody>
</table>

4. Available Distribution Feature for Each Optional Trunk Card Type

<table>
<thead>
<tr>
<th>Trunk Card Type</th>
<th>Channel Type</th>
<th>Feature</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>DIL</td>
</tr>
<tr>
<td>LCOT</td>
<td>—</td>
<td>✓*</td>
</tr>
<tr>
<td>T1</td>
<td>LCOT</td>
<td>✓*</td>
</tr>
<tr>
<td></td>
<td>GCOT</td>
<td>✓*</td>
</tr>
<tr>
<td></td>
<td>DID</td>
<td>✓</td>
</tr>
<tr>
<td></td>
<td>TIE (E &amp; M)</td>
<td>✓*</td>
</tr>
<tr>
<td>E1</td>
<td>DR2</td>
<td>✓</td>
</tr>
<tr>
<td></td>
<td>E &amp; M-C</td>
<td>✓*</td>
</tr>
<tr>
<td></td>
<td>E &amp; M-P</td>
<td>✓*</td>
</tr>
<tr>
<td>BRI</td>
<td>CO</td>
<td>✓</td>
</tr>
<tr>
<td>PRI</td>
<td>CO</td>
<td>✓</td>
</tr>
</tbody>
</table>

✓*: Enable (default); ✓: Enable

5. Available Destinations
1.1.1 Incoming Trunk Call Features

<table>
<thead>
<tr>
<th>Destination</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/SIP Extension/ISDN Extension/T1-OPX)</td>
<td>✓</td>
</tr>
<tr>
<td>PS</td>
<td>✓</td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td>✓</td>
</tr>
<tr>
<td>PS Ring Group</td>
<td>✓</td>
</tr>
<tr>
<td>Floating Extension no. for SVM</td>
<td>✓</td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td>✓</td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td>✓</td>
</tr>
<tr>
<td>DISA</td>
<td>✓</td>
</tr>
<tr>
<td>Analogue/ISDN Remote Maintenance</td>
<td>✓</td>
</tr>
<tr>
<td>Idle Line Access no. + Phone no.</td>
<td>✓</td>
</tr>
<tr>
<td>Trunk Group Access no. + Trunk Group no. + Phone no.</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with no PBX Code)</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with PBX Code)</td>
<td>✓</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
</table>
| Intercept Routing| If a called party does not answer a call within a preprogrammed time period (Intercept time), it is redirected to the preprogrammed destination.  
→ 1.1.1.6 Intercept Routing |
| No Answer (IRNA) | If a called party is busy or in DND mode, the call is redirected to the preprogrammed destination.  
→ 1.1.1.6 Intercept Routing |
| Busy/DND         | If a destination is not assigned, the call is redirected to the operator.  
→ 1.1.1.7 Intercept Routing—No Destination |

PC Programming Manual References
3.33 [1-1] Slot—Port Property - T1 Port—◆ Trunk Property
3.39 [1-1] Slot—Port Property - E1 Port—◆ Trunk Property
12.2 [10-2] DIL Table & Port Settings—DIL—◆ Trunk Property

PT Programming Manual References
None
1.1.1.2 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]

A trunk call is received.

Does the call have its CLI* information and is CLI mode enabled for the trunk and the time mode?

No

Yes

CLI works.

Is the CLI destination assigned?

No

The call is routed to the CLI destination.

Yes

Is the DIL destination of the time mode assigned?

No

The call is routed to the operator (Intercept Routing—No Destination).

Yes

The call is routed to the DIL destination.

*: 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.

<table>
<thead>
<tr>
<th>Trunk No.</th>
<th>CLI</th>
<th>Destination*</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Day</td>
<td>Lunch</td>
</tr>
<tr>
<td>01</td>
<td>Enable</td>
<td>Disable</td>
</tr>
<tr>
<td>02</td>
<td>Enable</td>
<td>Disable</td>
</tr>
<tr>
<td>...</td>
<td>:</td>
<td>:</td>
</tr>
</tbody>
</table>

*1 \[12.2 \{10-2\} DIL Table & Port Settings—DIL— DIL Destination—Day, Lunch, Break, Night
1.1.1 Incoming Trunk Call Features

→ [450] DIL 1:1 Destination

**Note**
Tenant number and VM trunk group number can also be assigned in the DIL table. Tenant number is used to determine the time mode (day/lunch/break/night) for the corresponding trunk. VM trunk group number is used in Voice Mail DPT (Digital) Integration.

**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**
12.2 [10-2] DIL Table & Port Settings—DIL
→ DIL Destination—Day, Lunch, Break, Night
→ Tenant Number
→ VM Trunk Group No.

**PT Programming Manual References**
[421] BRI DIL/DDI/MSN Selection
[450] DIL 1:1 Destination

**Feature Guide References**
1.1.1.5 Calling Line Identification (CLI) Distribution
2.2.3 Tenant Service
2.2.4 Time Service
3.1 Capacity of System Resources
1.1.1.3 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]**

A trunk call is received.

Is the DID/DDI number found in the DID/DDI table?

Yes

Does the call have its CLI* information and is CLI mode enabled for the time mode?

No

The call is routed to the DID/DDI destination.

Yes

Does the DID/DDI number match an extension number?

No

The call is routed to the extension.

Yes

The call is routed to the TIE line or trunk.

No

Is the CLI destination assigned?

Yes

CLI works.

No

Is the DID/DDI destination for the time mode assigned?

Yes

The call is routed to the CLI destination.

No

The call is routed to the DID/DDI destination.

*: 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.
### 1.1.1 Incoming Trunk Call Features

<table>
<thead>
<tr>
<th>Location</th>
<th>No.</th>
<th>Name</th>
<th>CLI</th>
<th>Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>Day</td>
<td>Lunch</td>
</tr>
<tr>
<td>0001</td>
<td>123-4567</td>
<td>John White</td>
<td>Enable</td>
<td>Disable</td>
</tr>
<tr>
<td>0002</td>
<td>123-2468</td>
<td>Tom Smith</td>
<td>Enable</td>
<td>Disable</td>
</tr>
<tr>
<td>0003</td>
<td>123-456</td>
<td>A company</td>
<td>Enable</td>
<td>Disable</td>
</tr>
</tbody>
</table>

Note

Tenant number and VM trunk group number can also be assigned in the DID/DDI table. Tenant number is used to determine the time mode (day/lunch/break/night) for the corresponding DID/DDI number. VM trunk group number is used in Voice Mail DPT (Digital) Integration (→ 1.24.2.3 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]

<table>
<thead>
<tr>
<th>Removed number of digits: 6</th>
<th>Modified DID/DDI number: 876543 21 = 1021</th>
</tr>
</thead>
<tbody>
<tr>
<td>Added number: 10</td>
<td></td>
</tr>
<tr>
<td>Received DID/DDI number: 87654321</td>
<td>1) Remove the first 6 digits.</td>
</tr>
<tr>
<td></td>
<td>2) Add &quot;10&quot;.</td>
</tr>
</tbody>
</table>

- **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:

<table>
<thead>
<tr>
<th>Received Number</th>
<th>Destination</th>
<th>Explanation</th>
</tr>
</thead>
<tbody>
<tr>
<td>123-4567</td>
<td>Extn. 100</td>
<td>The PBX finds the match in location 0001 in the table after receiving &quot;7&quot;. So the call is routed to extension 100.</td>
</tr>
<tr>
<td>123-456</td>
<td>Extn. 101</td>
<td>The Inter-digit time expired after receiving &quot;6&quot;. The PBX finds the match in location 0003 in the table. So the call is routed to extension 101.</td>
</tr>
</tbody>
</table>

**PC Programming Manual References**

4.5 [2-3] Timers & Counters—Miscellaneous—Incoming Call Inter-digit Timer—DDI / DID
12.2 [10-2] DIL Table & Port Settings—DDI / DID / TIE / MSN
   → Distribution Method
   → DDI/DID/TIE/MSN—Remove Digit
   → DDI/DID/TIE/MSN—Additional Dial
12.3 [10-3] DDI / DID Table

**PT Programming Manual References**

[421] BRI DIL/DDI/MSN Selection
[451] DID Number
[452] DID Name
[453] DID Destination

**Feature Guide References**

1.1.1.5 Calling Line Identification (CLI) Distribution
2.2.3 Tenant Service
2.2.4 Time Service
3.1 Capacity of System Resources
1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service

Description

Provides automatic direction of an incoming ISDN-BRI (Basic Rate Interface) line call with an MSN to a preprogrammed destination. One ISDN-BRI port can support a maximum of 10 MSNs. Each MSN has a destination for each time mode (day/lunch/break/night). Point-to-multipoint must be selected for the ISDN configuration.

[Method Flowchart]

A trunk call is received.

Are any MSNs assigned in the MSN table?

Yes

Is the MSN found in the MSN table?

Yes

Does the call have its CLI* information and is CLI mode enabled for the time mode?

Yes

CLI works.

Is the CLI destination assigned?

No

The call is routed to the operator (Intercept Routing—No Destination).

Yes

Is the MSN destination for the time mode assigned?

No

The call is ignored.

No

The call is routed to the MSN destination.

*: 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 MSN destination, but routed to the CLI destination.

[Programming Example of MSN Table for ISDN BRI Port 1]

A table can be programmed for each ISDN-BRI port. Each BRI port has 10 MSN locations.

→ 12.6 [10-4] MSN Table
### 1.1.1 Incoming Trunk Call Features

<table>
<thead>
<tr>
<th>Location</th>
<th>No.</th>
<th>Name</th>
<th>CLI</th>
<th>Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>Day</td>
<td>Lunch</td>
</tr>
<tr>
<td>01</td>
<td>123-4567</td>
<td>A Company</td>
<td>Enable</td>
<td>Disable</td>
</tr>
<tr>
<td>02</td>
<td>123-2468</td>
<td>C Company</td>
<td>Enable</td>
<td>Disable</td>
</tr>
</tbody>
</table>

#### Note
Tenant number and VM trunk group number can also be assigned in the MSN table. Tenant number is used to determine the time mode (day/lunch/break/night) for the corresponding MSN. VM trunk group number is used in Voice Mail DPT (Digital) Integration.

→ 1.24.2.3 Voice Mail DPT (Digital) Integration

#### Explanation:
If the MSN "123-4567" is received from BRI port 1:

1. Checks the number in the table.
   → Matches the number in location 01.
2. Checks the time mode.
   - In Day mode: CLI is enabled. Route to CLI destination.
   - In Lunch mode: CLI is disabled. Route to MSN destination, extension 100.

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

##### [Modification Example]

<table>
<thead>
<tr>
<th>Removed number of digits: 6</th>
<th>Added number: 10</th>
<th>Received MSN: 87654321</th>
<th>Modified MSN: 876543 21 = 1021</th>
</tr>
</thead>
<tbody>
<tr>
<td>1) Remove the first 6 digits.</td>
<td>2) Add &quot;10&quot;.</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- When using point-to-multipoint configuration with a BRI, do not connect another ISDN terminal device in parallel with the PBX. As only two channels can be used at one time with the BRI, the other ISDN terminal device may monopolise both channels.

#### PC Programming Manual References

12.2 [10-2] DIL Table & Port Settings—DDI / DID / TIE / MSN
   → ♦ Distribution Method
   → ♦ DDI/DID/TIE/MSN—Remove Digit
1.1.1 Incoming Trunk Call Features

→ DDI/DID/TIE/MSN—Additional Dial
12.6 [10-4] MSN Table

PT Programming Manual References

[421] BRI DIL/DDI/MSN Selection
[426] BRI Configuration

Feature Guide References

1.1.1.5 Calling Line Identification (CLI) Distribution
2.2.3 Tenant Service
2.2.4 Time Service
3.1 Capacity of System Resources
### 1.1.1.5 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.

<table>
<thead>
<tr>
<th>CLI Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID</td>
<td>Caller's number is sent from an analogue trunk. → 1.18.1 Caller ID</td>
</tr>
<tr>
<td>Calling Line Identification Presentation (CLIP)</td>
<td>Caller's number is sent from an ISDN line. → 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP)</td>
</tr>
<tr>
<td>Automatic Number Identification (ANI)</td>
<td>Caller's number is sent from an E1 or T1 line. → 1.22.1 E1 Line Service → 1.23.1 T1 Line Service</td>
</tr>
</tbody>
</table>

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

a. DIL  
b. DID/DDI  
c. MSN Ringing Service

Each trunk (for DIL) and the DID/DDI/MSN number can enable or disable the CLI feature for each time mode (day/lunch/break/night) (→ 2.2.4 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]

<table>
<thead>
<tr>
<th>Location (System Speed Dialling No.)</th>
<th>System Speed Dialling Name*¹</th>
<th>Telephone No.*²</th>
<th>CLI Destination*³</th>
</tr>
</thead>
<tbody>
<tr>
<td>000</td>
<td>ABC Company</td>
<td>901234567890</td>
<td>200</td>
</tr>
<tr>
<td>001</td>
<td>:</td>
<td>:</td>
<td>:</td>
</tr>
<tr>
<td>:</td>
<td>:</td>
<td>:</td>
<td>:</td>
</tr>
</tbody>
</table>

*¹ → 8.1 [6-1] System Speed Dial— Name  
   → [002] System Speed Dialling Name  
*² → 8.1 [6-1] System Speed Dial— CO Line Access Number + Telephone Number  
   → [001] System Speed Dialling Number  
*³ → 8.1 [6-1] 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. (→ 1.18.1 Caller ID)

### PC Programming Manual References

12.2 [10-2] DIL Table & Port Settings—CLI for DIL—◆ CLI Ring for DIL—Day, Lunch, Break, Night
12.3 [10-3] DDI / DID Table—◆ CLI Ring for DDI/DID—Day, Lunch, Break, Night
12.6 [10-4] MSN Table—◆ CLI Ring for MSN—Day, Lunch, Break, Night
8.1 [6-1] 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

1.1.1.2 Direct In Line (DIL)
1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)
1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service
1.6.1.5 Speed Dialling—Personal/System
1.1.1.6 Intercept Routing

Description

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

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercept Routing—No Answer (IRNA)</td>
<td>If a called party does not answer a call within a preprogrammed time period (IRNA Timer), the call is redirected to the preprogrammed destination.</td>
</tr>
</tbody>
</table>
| 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 or T1 [LCOT/GCOT] 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.

→ 6.10 [4-2-1] Portable Station—Extension Settings—Intercept Destination—◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night
→ [604] Extension Intercept Destination

<table>
<thead>
<tr>
<th>When the original destination is:</th>
<th>The Available Intercept Destination is:</th>
</tr>
</thead>
</table>
| • Wired Extension (PT/SLT/SIP Extension/ T1-OPX)  
• PS | The destination assigned to the original extension.  
6.1 [4-1-1] 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  
6.10 [4-2-1] 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  
→ [604] Extension Intercept Destination |
### 1.1.1 Incoming Trunk Call Features

<table>
<thead>
<tr>
<th>When the original destination is:</th>
<th>The Available Intercept Destination is:</th>
</tr>
</thead>
<tbody>
<tr>
<td>• ICD Group</td>
<td>The ICD Group Overflow destination assigned to the group. (→ 1.2.2.6 Overflow Feature)</td>
</tr>
<tr>
<td></td>
<td>→ [625] Destination for Overflow Time Expiration</td>
</tr>
<tr>
<td>• VM Group (DTMF/DPT)</td>
<td>The destination assigned to the first extension of the VM group.</td>
</tr>
<tr>
<td>• DISA*¹</td>
<td>If all DISA ports are busy when a call is made using DISA, one of the following can be selected through system programming:</td>
</tr>
<tr>
<td>• PS Ring Group²</td>
<td>• <strong>Disable</strong>: Busy tone is sent to the caller. When using an analogue trunk, a ringback tone is sent.</td>
</tr>
<tr>
<td>• External Pager (TAFAS)²</td>
<td>• <strong>Operator</strong>: The call will be redirected to the operator.</td>
</tr>
<tr>
<td>• Analogue/ISDN Remote Maintenance</td>
<td>• <strong>AA-0, AA-9</strong>: The call will be redirected to the destination assigned to that AA number.</td>
</tr>
<tr>
<td>• Other PBX Extension (TIE with no PBX Code)</td>
<td>→ 7.3 [5-3-1] Voice Message—DISA System—Option 1— DISA Intercept—Intercept when all DISA ports are busy</td>
</tr>
<tr>
<td></td>
<td>Not assignable (Intercept Routing is not available.)</td>
</tr>
</tbody>
</table>

---

*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.
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]

<table>
<thead>
<tr>
<th>Intercept Destination</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/SIP Extension/ISDN Extension/T1-OPX)</td>
<td>✓</td>
</tr>
<tr>
<td>PS</td>
<td>✓</td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td>✓</td>
</tr>
<tr>
<td>PS Ring Group</td>
<td>✓</td>
</tr>
<tr>
<td>Floating Extension no. for SVM</td>
<td>✓</td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td>✓</td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td>✓</td>
</tr>
<tr>
<td>DISA</td>
<td>✓</td>
</tr>
<tr>
<td>Analogue/ISDN Remote Maintenance</td>
<td>✓</td>
</tr>
<tr>
<td>Idle Line Access no. + Phone no.</td>
<td>✓</td>
</tr>
<tr>
<td>Trunk Group Access no. + Trunk Group no. + Phone no.</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with no PBX Code)</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with PBX Code)</td>
<td>✓</td>
</tr>
</tbody>
</table>

Conditions

- **Intercept Routing—DND on/off**
  Intercept Routing—DND can be enabled or disabled system programming.

"600" is an example of a Floating Extension No. for the External Pager.
If disabled, one of the following is activated depending on the type of line that a call arrives through:

a. **LCOT or T1 (LCOT/GCOT) 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 or T1 (LCOT/GCOT) 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.**

→ **12.7 [10-5] 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).


**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 VM groups are decided on a tenant basis.**

**When an intercom, trunk, or DISA call is received by an extension and forwarded (e.g., FWD—All Calls) to an outside destination, and the outside destination is busy or does not answer, the Intercept Routing feature can be used.** This can be useful when calls are forwarded to cellular phones. This feature may not be available depending on the specifications of some trunks. Also, this feature is not available when the original call was made from a SIP extension.

**PC Programming Manual References**

4.5 [2-3] Timers & Counters

→ **Dial / IRNA / Recall / Tone—Intercept Routing No Answer (IRNA)—Day, Lunch, Break, Night**

→ **DISA / Door / Reminder / U. Conf—DISA Intercept Timer—Day, Lunch, Break, Night**


6.1 [4-1-1] Wired Extension—Extension Settings

→ **Intercept Destination**

→ **Intercept No Answer Time**

6.10 [4-2-1] Portable Station—Extension Settings

→ **Intercept Destination**

→ **Intercept No Answer Time**

7.3 [5-3-1] Voice Message—DISA System—Option 1—DISA Intercept—Intercept when all DISA ports are busy

12.7 [10-5] 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

1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)
1.1.1.7 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, or T1 (LCOT/GCOT) card even when disabled.
- **If an operator (tenant/PBX) is not assigned:**
  The extension connected to the lowest-numbered jack will be the intercept destination.
- Intercept Routing—No Destination also applies to calls from doorphones.

PC Programming Manual References
12.7 [10-5] Miscellaneous—◆ Intercept—Routing to Operator - No Destination (Destination is not programmed.)

PT Programming Manual References
[006] Operator Assignment

Feature Guide References
2.2.5 Operator Features
1.1.2 Internal Call Features

1.1.2.1 Internal Call Features—SUMMARY

Description

The following types of internal calls are available:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercom Call</td>
<td>A call from one extension to another.</td>
</tr>
<tr>
<td></td>
<td>→ 1.5.3 Intercom Call</td>
</tr>
<tr>
<td>Doorphone Call</td>
<td>When a call from a doorphone reaches its destination, the recipient can talk to the visitor.</td>
</tr>
<tr>
<td></td>
<td>→ 1.17.1 Doorphone Call</td>
</tr>
</tbody>
</table>

[Available Destination]

The destinations of doorphone calls can be assigned for each time mode (day/lunch/break/night) (→ 2.2.4 Time Service) on a doorphone port basis.

<table>
<thead>
<tr>
<th>Destination</th>
<th>Extension</th>
<th>Doorphone</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/SIP Extension/ISDN Extension/T1-OPX)</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>PS</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>PS Ring Group</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Floating Extension no. for SVM</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>DISA</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Analogue/ISDN Remote Maintenance</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Idle Line Access no. + Phone no.</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Trunk Group Access no. + Trunk Group no. + Phone no.</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with no PBX Code)</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with PBX Code)</td>
<td>✓</td>
<td>✓</td>
</tr>
</tbody>
</table>

✓: Available

PC Programming Manual References

1.1.2 Internal Call Features

PT Programming Manual References

[720] Doorphone Call Destination
1.1.2.2 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]**

<table>
<thead>
<tr>
<th>Caller</th>
<th>Called Party</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>COS 1</td>
</tr>
<tr>
<td>COS 1</td>
<td></td>
</tr>
<tr>
<td>COS 2</td>
<td></td>
</tr>
<tr>
<td>COS 3</td>
<td>✓</td>
</tr>
<tr>
<td>...</td>
<td></td>
</tr>
</tbody>
</table>

✓: 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 (→ 2.2.5 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. (→ 1.17.1 Doorphone Call)
1.1.2 Internal Call Features

PC Programming Manual References

4.14 [2-7-3] Class of Service—Internal Call Block—◆ COS Number of the Extension Which Receive the Call from Other Extension 1–64
5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Main—◆ COS
6.1 [4-1-1] Wired Extension—Extension Settings—Main—◆ COS
6.10 [4-2-1] Portable Station—Extension Settings—Main—◆ COS
7.1 [5-1] Doorphone—◆ COS
7.6 [5-4] External Relay—◆ COS Number

PT Programming Manual References

None

Feature Guide References

2.2.1 Class of Service (COS)
### 1.1.3 Incoming Call Indication Features

#### 1.1.3.1 Incoming Call Indication Features—SUMMARY

**Description**

Incoming calls are indicated by various methods as follows:

<table>
<thead>
<tr>
<th>Type</th>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ring Tone</td>
<td>Ring Tone Pattern Selection</td>
<td>A telephone rings when receiving a call. The ring tone patterns can be changed for each incoming call type.</td>
</tr>
<tr>
<td>Voice-calling</td>
<td>Alternate Receiving—Ring/Voice</td>
<td>A PT user can select to receive intercom calls by ring tone or by voice, through personal programming.</td>
</tr>
<tr>
<td>LED (Light Emitting Diode)</td>
<td>LED Indication</td>
<td>The light shows line conditions with a variety of light patterns.</td>
</tr>
<tr>
<td>Display (Caller’s Information)</td>
<td>Display Information</td>
<td>The display shows the caller’s information.</td>
</tr>
<tr>
<td>External Pager</td>
<td>Trunk Answer from Any Station (TAFAS)</td>
<td>The external pager sends a ring tone when receiving a call.</td>
</tr>
<tr>
<td>Tone/Voice during a Conversation</td>
<td>Call Waiting</td>
<td>A busy extension hears a tone, or voice from the handset/built-in speaker indicating that another incoming call is waiting.</td>
</tr>
</tbody>
</table>
1.1.3.2 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]**

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Duration</th>
</tr>
</thead>
<tbody>
<tr>
<td>Single</td>
<td>1 280 ms*</td>
</tr>
<tr>
<td>Double</td>
<td></td>
</tr>
<tr>
<td>Triple</td>
<td></td>
</tr>
<tr>
<td>S-Double</td>
<td></td>
</tr>
</tbody>
</table>

* 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.
  → 4.15 [2-8-1] Ring Tone Patterns—Call from CO
- **Incoming Doorphone Calls**: each pattern plan can assign a ring tone pattern for each doorphone.
  → 4.16 [2-8-2] 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).
  → 4.17 [2-8-3] 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>
<thead>
<tr>
<th>Table No.</th>
<th>Intercom Call/Hold Recall</th>
<th>Trunk Call/Hold Recall</th>
<th>Doorphone Call</th>
<th>Timed Reminder</th>
<th>Call Back</th>
<th>LCS</th>
<th>External Sensor</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>TRG1</td>
<td>TRG2</td>
<td>...</td>
<td>Port 1</td>
<td>Port 2</td>
<td>...</td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>Double</td>
<td>Single</td>
<td>Single</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>Single</td>
<td>Double</td>
<td>Double</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**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.
1.1.3 Incoming Call Indication Features

PC Programming Manual References

4.15 [2-8-1] Ring Tone Patterns—Call from CO—◆ Ring Tone Pattern Plan 1–8
4.16 [2-8-2] Ring Tone Patterns—Call from Doorphone—◆ Ring Tone Pattern Plan 1–8
4.17 [2-8-3] Ring Tone Patterns—Call from Others—◆ Extension—Ring Tone Pattern Plan 1–8
4.18 [2-9] System Options—Option 1—◆ PT Operation—PT Ring Off Setting
6.1 [4-1-1] Wired Extension—Extension Settings
   →Option 1—◆ Ring Pattern Table
   →Option 6—◆ ICM Tone
6.5 [4-1-4] Wired Extension—Flexible Button—◆ Optional Parameter (Ringing Tone Type Number) (for Loop CO, Single CO, Group CO, ICD Group, SDN)
6.10 [4-2-1] Portable Station—Extension Settings—Option 1—◆ Ring Pattern Table
6.17 [4-3] DSS Console—◆ Optional Parameter (Ringing Tone Type Number) (for Loop CO, Single CO, Group CO, ICD Group, SDN)

PT Programming Manual References

None

Feature Guide References

3.1 Capacity of System Resources

User Manual References

3.1.3 Customising the Buttons—To distinguish the ringing tones for each CO, ICD Group, PDN, SDN, or INTERCOM button (Digital proprietary telephone only)
1.1.3.3 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.

<table>
<thead>
<tr>
<th>Call Type</th>
<th>Notification Method</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><strong>DPT</strong></td>
</tr>
<tr>
<td>Intercom Call</td>
<td>Call Waiting tone/OHCA/Whisper OHCA/Off</td>
</tr>
<tr>
<td>Trunk Call*</td>
<td>Call Waiting tone/Off</td>
</tr>
</tbody>
</table>

* 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 call for an extension in a VM group (DPT/DTMF)** is not available.
- **Data Line Security**
  Setting Data Line Security cancels the Call Waiting setting. (→ 1.11.5 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 (→ 3.2.1 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. 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

3.24 [1-1] Slot—Card Property - LCO type—◆ Caller ID—Caller ID Signalling
4.9 [2-6-1] Numbering Plan—Main—Features
  ➔◆ Manual Call Waiting for Extension Call
  ➔◆ Automatic Call Waiting
4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature
  ➔◆ BSS / OHCA / Whisper OHCA / DND Override
  ➔◆ BSS / OHCA / Whisper OHCA / DND Override-2
4.18 [2-9] System Options—Option 5—◆ Call Waiting—Automatic Call Waiting for Extension Call

6.1 [4-1-1] Wired Extension—Extension Settings
  ➔Option 2—◆ Manual C. Waiting for Extension Call
  ➔Option 2—◆ Automatic C. Waiting
  ➔Option 4—◆ Call Waiting Tone Type
6.10 [4-2-1] Portable Station—Extension Settings
  ➔Option 2—◆ Manual C. Waiting for Extension Call
  ➔Option 2—◆ Automatic C. Waiting
  ➔Option 4—◆ Call Waiting Tone Type

PT Programming Manual References

None

Feature Guide References

1.8.4 Second Call Notification to Busy Extension
1.18.2 Incoming Call Log
1.26.1.1 Station Message Detail Recording (SMDR)

User Manual References

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

1.2.1 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.

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Circular Hunting</td>
<td>An idle extension is searched for in the order specified in the idle extension hunting group in a circular way.</td>
</tr>
<tr>
<td></td>
<td>![Circular Hunting Diagram]</td>
</tr>
<tr>
<td>Terminated Hunting</td>
<td>An idle extension is searched for in the order specified in the idle extension hunting group until reaching the last assigned extension.</td>
</tr>
<tr>
<td></td>
<td>![Terminated Hunting Diagram]</td>
</tr>
</tbody>
</table>

**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) (→ 2.2.4 Time Service).

### Available Destination

<table>
<thead>
<tr>
<th>Destination</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/SIP Extension/ISDN Extension/T1-OPX)</td>
<td>✓</td>
</tr>
<tr>
<td>PS</td>
<td>✓</td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td>✓</td>
</tr>
</tbody>
</table>
### 1.2.1 Idle Extension Hunting

<table>
<thead>
<tr>
<th>Destination</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>PS Ring Group</td>
<td>✔</td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td>✔</td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td>✔</td>
</tr>
<tr>
<td>DISA</td>
<td>✔</td>
</tr>
<tr>
<td>Analogue/ISDN Remote Maintenance</td>
<td>✔</td>
</tr>
<tr>
<td>Idle Line Access no. + Phone no.</td>
<td>✔</td>
</tr>
<tr>
<td>Trunk Group Access no. + Trunk Group no. + Phone no.</td>
<td>✔</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with no PBX Code)</td>
<td>✔</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with PBX Code)</td>
<td>✔</td>
</tr>
</tbody>
</table>

- **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**

5.17 [3-6] Extension Hunting Group
5.18 [3-6] Extension Hunting Group—Member List

**PT Programming Manual References**

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

**Feature Guide References**

1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)
3.1 Capacity of System Resources
1.2.2 Incoming Call Distribution Group Features

1.2.2.1 Incoming Call Distribution Group Features—SUMMARY

Description

An incoming call distribution group is a group of extensions programmed through system programming.

→ 5.14 [3-5-1] Incoming Call Distribution Group—Group Settings—Member List
→ [620] Incoming Call Distribution Group Member

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).

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.

→ 5.13 [3-5-1] 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.

<table>
<thead>
<tr>
<th>Group No.</th>
<th>Floating Extn. No.⁴</th>
<th>Group Name²</th>
<th>Distribution Method³</th>
<th>Max. No. of Busy Extensions⁴</th>
<th>Queuing Call Capacity⁵</th>
<th>Queuing Hurry-up Level⁶</th>
<th>Overflow Time⁷</th>
<th>Overflow Destination⁸</th>
<th>Tenant No.⁹</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>601</td>
<td>Sales</td>
<td>Ring</td>
<td>3</td>
<td>60</td>
<td>100</td>
<td>Day</td>
<td>...</td>
<td>100</td>
</tr>
<tr>
<td>2</td>
<td>602</td>
<td>Engineering</td>
<td>UCD</td>
<td>Max.</td>
<td>8</td>
<td>200</td>
<td>...</td>
<td>...</td>
<td>200</td>
</tr>
<tr>
<td>3</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>5</td>
</tr>
</tbody>
</table>

¹→ [622] Incoming Call Distribution Group Floating Extension Number
²→ [623] Incoming Call Distribution Group Name
³→ [624] Incoming Call Distribution Group Distribution Method
⁴→ [632] Maximum Number of Agents
⁵→ [628] Queuing Call Capacity
⁶→ [629] Queuing Hurry-up Level
⁷→ [626] Overflow Time
⁸→ [625] Destination for Overflow Time Expiration/[627] Destination When All Busy
⁹The tenant number is required to determine the time mode (day/lunch/break/night) (→ 2.2.4 Time Service) and the music source (for Music on Hold) for each group.
1.2.2 Incoming Call Distribution Group Features

### 1. Group Call Distribution
Incoming calls are distributed using one of the following methods:

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

### 2. 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
It is possible to assign a priority to incoming call distribution groups so that incoming calls can be received in priority order.

---

* → 1.2.2.4 Queuing Feature
*2 → 1.2.2.8 Supervisory Feature
*3 → 1.2.2.7 Log-in/Log-out
4. **Overflow Feature** (→ 1.2.2.6 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**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Log-in/Log-out</td>
<td>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. → 1.2.2.7 Log-in/Log-out</td>
</tr>
<tr>
<td>Supervisory Feature</td>
<td>Incoming Call Queue Monitor</td>
</tr>
<tr>
<td></td>
<td>Log-in/Log-out Monitor and Remote Control</td>
</tr>
</tbody>
</table>

**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 (→ 1.1.2.2 Internal Call Block).

### PC Programming Manual References

5.13 [3-5-1] Incoming Call Distribution Group—Group Settings
  →Main
  →Overflow Queuing Busy
  →Overflow No Answer
  →Miscellaneous

5.16 [3-5-3] Incoming Call Distribution Group—Miscellaneous— Options—ICD Group Key Mode

5.14 [3-5-1] Incoming Call Distribution Group—Group Settings—Member List

### 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

1.3.1.2 Call Forwarding (FWD)
1.20.2 Flexible Buttons
3.1 Capacity of System Resources
1.2.2.2 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 (→ 1.2.2.4 Queuing Feature).

1. Distribution Method

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

<table>
<thead>
<tr>
<th>Distribution Method</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Uniform Call Distribution (UCD)</td>
<td>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).</td>
</tr>
<tr>
<td>Priority Hunting</td>
<td>An idle extension is searched for using the preprogrammed order for the group.</td>
</tr>
</tbody>
</table>

Extn. A

1st Priority 2nd Priority 3rd ....

Extn. B

Extn. C

Extn. D

Always starts searching from the first assigned extension.
### Distribution Method

<table>
<thead>
<tr>
<th>Distribution Method</th>
<th>Description</th>
</tr>
</thead>
</table>
| 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. |

![Diagram](image-url)

**Extn. A**  
**Extn. B**  
**Extn. C**  
**Extn. D**  

**Rings immediately simultaneously.**  
**Delayed Ringing:** Rings after a specified time delay.

---

### 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. (→ 1.1.3.3 Call Waiting)

#### [How the Group Call Waiting Feature Activates]

<table>
<thead>
<tr>
<th>Programming Conditions</th>
<th>Result</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Group Call Waiting Mode</strong></td>
<td><strong>Group Call Waiting Distribution Method</strong></td>
</tr>
<tr>
<td>Distribution</td>
<td>UCD</td>
</tr>
<tr>
<td></td>
<td>Priority Hunting</td>
</tr>
<tr>
<td></td>
<td>Ring</td>
</tr>
<tr>
<td>All</td>
<td>UCD/Priority Hunting/Ring</td>
</tr>
</tbody>
</table>

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

#### [Example]

- Group Call Waiting mode: All  
- Group call distribution method for idle extensions: UCD

All extensions hear the Call Waiting tone (Ring).

---

### [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.

![Diagram of incoming call distribution groups](image)

### 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)**
  - When a KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is installed in the PBX, and the distribution type is set to Uniform Call Distribution, it is possible to select whether incoming calls are distributed to idle extensions evenly in order (UCD), or to the extension that has been idle the longest (ACD).
  - 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. (→ 1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND))
  - The Group Call Waiting feature cannot be used with the VIP Call feature (→ 1.2.2.5 VIP Call) and/or Wrap-up feature (→ 1.2.2.7 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
- 5.1 [3-1-1] Trunk Group—TRG Settings—Main—◆ Line Hunting Order
- 5.13 [3-5-1] Incoming Call Distribution Group—Group Settings
  - →Main—◆ Distribution Method
  - →Main—◆ Call Waiting Distribution
  - →Miscellaneous—◆ Extension No Answer Redirection Time
1.2.2 Incoming Call Distribution Group Features

PT Programming Manual References

[621] Incoming Call Distribution Group Delayed Ringing
[624] Incoming Call Distribution Group Distribution Method
[632] Maximum Number of Agents
1.2.2.3 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

- The KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is required to 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. (→ 1.25.6 Virtual PS)
  - The forwarding type of the virtual PS is set to All Calls. (→ 1.3.1.2 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.
– The distribution method for the ICD Group is set to Ring. (→ 1.2.2.2 Group Call Distribution)

- 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.
- LCOT trunks without reverse circuit detection (→ 1.5.4.5 Reverse Circuit) and channels of a T1 trunk set to LCOT type do not support this feature.
- 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 is 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. (→ 1.17.6 Direct Inward System Access (DISA))

**PC Programming Manual References**

4.18 [2-9] System Options—Option 4—◆ Send CLIP of CO Caller to ISDN—when ICD Group with Cellular Phone

**PT Programming Manual References**

None

**Feature Guide References**

1.25.6 Virtual PS
1.30.6 Network ICD Group
1.2.2.4 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) (→ 2.2.4 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]**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
<th>Condition</th>
</tr>
</thead>
<tbody>
<tr>
<td>OGM xx</td>
<td>An outgoing message is sent to the caller. &quot;xx&quot; applies to the OGM number.</td>
<td>After the OGM, Music on Hold will be sent and the next event in the sequence will be activated.</td>
</tr>
<tr>
<td>b × 5 s</td>
<td>Puts the caller in the waiting queue for b (01-16) × 5 seconds.</td>
<td>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.</td>
</tr>
<tr>
<td>Sequence c</td>
<td>Redirects to sequence c. &quot;c&quot; applies to the sequence number.</td>
<td>None</td>
</tr>
<tr>
<td>Overflow</td>
<td>Redirects to the overflow destination.</td>
<td>None</td>
</tr>
<tr>
<td>Disconnect</td>
<td>Disconnects the line.</td>
<td>None</td>
</tr>
<tr>
<td>None</td>
<td>Redirects to the next sequence.</td>
<td>If assigned as sequence 01, the Queuing Time Table will not be activated.</td>
</tr>
<tr>
<td>(No command)</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**[Programming Example of Queuing Time Table]**

<table>
<thead>
<tr>
<th>Queuing Time Table No.</th>
<th>Sequence 01</th>
<th>Sequence 02</th>
<th>Sequence 03</th>
<th>Sequence 04</th>
<th>...</th>
</tr>
</thead>
<tbody>
<tr>
<td>01</td>
<td>OGM 01</td>
<td>6 × 5 s</td>
<td>OGM 03</td>
<td>Overflow</td>
<td></td>
</tr>
<tr>
<td>02²</td>
<td>OGM 02</td>
<td>6 × 5 s</td>
<td>OGM 04</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>03</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

¹ → 5.15 [3-5-2] Incoming Call Distribution Group—Queuing Time Table—Queueing Sequence—Sequence 01–16

² → [631] Sequences in Queuing Time Table

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:

The call queues.

Sequence 01
OGM 01 is sent.
Thank you for calling Panasonic. The department you are calling is busy. Please hold the line. We will answer your call shortly.

Sequence 02
OGM 03 is sent.
We are sorry to keep you holding. The department is still busy. We are transferring you to the operator.

Sequence 03
Redirects to the overflow destination.

Overflow destination answers.

The call is connected to the member extension as soon as the extension becomes available.

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:

<table>
<thead>
<tr>
<th>Light Pattern</th>
<th>Calls in the Waiting Queue</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td>No queued call</td>
</tr>
<tr>
<td>Red on</td>
<td>At or under the assigned number for Hurry-up</td>
</tr>
<tr>
<td>Rapid red flashing</td>
<td>Over the assigned number for Hurry-up</td>
</tr>
</tbody>
</table>

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—Incoming Call Queue Monitor
4.18 [2-9] System Options—Option 4—Transfer—Transfer to busy Extension without BSS Operation
5.13 [3-5-1] 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
5.19 [3-7-1] VM(DPT) Group—System Settings—Call Waiting on VM Group
5.22 [3-8-1] VM(DTMF) Group—System Settings—Others—Call Waiting on VM Group
6.5 [4-1-4] Wired Extension—Flexible Button
  →Type
1.2.2 Incoming Call Distribution Group Features

Port assignments:

6.13 [4-2-3] Portable Station—Flexible Button
   →.Type
   →.Ext. No. / Floating Ext. No. (for Hurry-up)

6.17 [4-3] DSS Console
   →.Type
   →.Ext. No. / Floating Ext. No. (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

1.2.2.6 Overflow Feature
1.17.5 Outgoing Message (OGM)
1.20.2 Flexible Buttons
3.1 Capacity of System Resources

User Manual References

1.9.3 Forwarding a Waiting Call (Manual Queue Redirection)
1.2.2.5 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 to extensions uniformly.

**[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.

---

**PC Programming Manual References**

5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Miscellaneous—◆ VIP Call Mode

**PT Programming Manual References**

None
1.2.2.6 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. An Overflow command is assigned to the Queuing Time Table.
   d. The Overflow time expires.
   e. 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) (→ 2.2.4 Time Service). The destination can be assigned as follows, depending on the above conditions.

• For a) and b):
  → 5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Overflow Queuing Busy—Queuing Busy—Destination-Day, Lunch, Break, Night
  → [627] Destination When All Busy

• For c), d), and e):
  → [625] Destination for Overflow Time Expiration

<table>
<thead>
<tr>
<th>Destination</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/SIP Extension/ISDN Extension/T1-OPX)</td>
<td>✓</td>
</tr>
<tr>
<td>PS</td>
<td>✓</td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td>✓</td>
</tr>
<tr>
<td>PS Ring Group</td>
<td>✓</td>
</tr>
<tr>
<td>Floating Extension no. for SVM</td>
<td>✓</td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td>✓</td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td>✓</td>
</tr>
<tr>
<td>DISA</td>
<td>✓</td>
</tr>
<tr>
<td>Analogue/ISDN Remote Maintenance</td>
<td>✓</td>
</tr>
<tr>
<td>Idle Line Access no. + Phone no.</td>
<td>✓</td>
</tr>
<tr>
<td>Trunk Group Access no. + Trunk Group no. + Phone no.</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with no PBX Code)</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with PBX Code)</td>
<td>✓</td>
</tr>
</tbody>
</table>

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 or T1 (LCOT/GCOT) 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 (→ 1.17.6 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 or T1 (LCOT/GCOT) card, a busy tone will not be sent to the caller.

PC Programming Manual References

5.13 [3-5-1] Incoming Call Distribution Group—Group Settings
  → Overflow Queuing Busy
  → Overflow No Answer

5.15 [3-5-2] Incoming Call Distribution Group—Queuing Time Table

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

1.2.2.4 Queuing Feature
1.2.2.7 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>

<table>
<thead>
<tr>
<th>Extension</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>101</td>
<td>Ready</td>
</tr>
<tr>
<td>102</td>
<td>Log-out</td>
</tr>
<tr>
<td>103</td>
<td>Ready</td>
</tr>
<tr>
<td>104</td>
<td>Not Ready</td>
</tr>
<tr>
<td>105</td>
<td>Wrap-up</td>
</tr>
<tr>
<td>106</td>
<td>Ready</td>
</tr>
</tbody>
</table>

**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:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Usage</th>
<th>Light Pattern</th>
</tr>
</thead>
<tbody>
<tr>
<td>No parameter</td>
<td>Used with an ICD Group button, or with the floating extension number of an incoming call distribution group, or with ≠ (All).</td>
<td>Red on</td>
</tr>
<tr>
<td>Floating extension number of a specified incoming call distribution group</td>
<td>Used to log in to or out of the specified incoming call distribution group.</td>
<td>Log-out Status</td>
</tr>
<tr>
<td>Parameter</td>
<td>Usage</td>
<td>Light Pattern</td>
</tr>
<tr>
<td>-----------</td>
<td>-------</td>
<td>---------------</td>
</tr>
<tr>
<td>✉ (All)</td>
<td>Used to log in to or out of all incoming call distribution groups to which the extension user belongs.</td>
<td>Red on after Log-out Operation, Off after Log-in Operation</td>
</tr>
</tbody>
</table>

- 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 SIP extensions, 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:

<table>
<thead>
<tr>
<th>Light pattern</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Slow red flashing</td>
<td>Wrap-up</td>
</tr>
<tr>
<td>Red on</td>
<td>Not Ready</td>
</tr>
<tr>
<td>Off</td>
<td>Ready (Wrap-up mode cancel)</td>
</tr>
</tbody>
</table>

- When a PS in Wireless XDP Parallel Mode completes a call, neither the PS nor its wired telephone can have Wrap-up time. (→ 1.25.5 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 (→ 1.2.2.2 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. (→ 1.2.2.8 Supervisory Feature)

- **Log-in/Log-out Information on SMDR**
  Log-in/Log-out information can be printed out on SMDR. (→ 1.26.1.1 Station Message Detail Recording (SMDR))

**PC Programming Manual References**

4.9 [2-6-1] Numbering Plan—Main—Features
  → ◆ Log-in / Log-out
  → ◆ Not Ready (Manual Wrap-up) Mode On / Off

5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Miscellaneous
1.2.2 Incoming Call Distribution Group Features

→ ◆ No. of Unanswered Calls for Automatic Log-out
→ ◆ Last Extension Log-out
5.16 [3-5-3] Incoming Call Distribution Group—Miscellaneous—◆ Options—Wrap-up Timer based on
5.14 [3-5-1] Incoming Call Distribution Group—Group Settings—Member List—◆ Wrap-up Timer
6.1 [4-1-1] Wired Extension—Extension Settings—Option 8—◆ Wrap-up Timer
6.5 [4-1-4] Wired Extension—Flexible Button
  → ◆ Type
  → ◆ Parameter Selection (for Log-in/Log-out)
  → ◆ Ext. No. / Floating Ext. No. (for Log-in/Log-out)
6.10 [4-2-1] Portable Station—Extension Settings—Option 8—◆ Wrap-up Timer
6.13 [4-2-3] Portable Station—Flexible Button
  → ◆ Type
  → ◆ Parameter Selection (for Log-in/Log-out)
  → ◆ Ext. No. / Floating Ext. No. (for Log-in/Log-out)
6.17 [4-3] DSS Console
  → ◆ Type
  → ◆ Parameter Selection (for Log-in/Log-out)
  → ◆ Ext. No. / Floating Ext. No. (for Log-in/Log-out)

PT Programming Manual References

None

Feature Guide References

1.20.2 Flexible Buttons

User Manual References

1.9.1 Leaving an Incoming Call Distribution Group (Log-in/Log-out, Wrap-up)
1.2.2.8 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.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Call Queue Monitor</td>
<td>The supervisor extension can monitor the status of an incoming call distribution group with the display.</td>
</tr>
</tbody>
</table>
| 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>*

--- Date and time
--- Floating extension number/name of incoming call distribution group
--- The number of queuing calls
--- The longest queuing time

Since JAN.29 09:10AM
--- Monitoring starting date and time
--- Total number of incoming calls
--- Total number of overflowed calls
--- The number of lost calls
--- Average queuing time

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

--- Log-in/Log-out Monitor: DSS buttons of the incoming call distribution group members show their status.
--- Log-in/Log-out Remote Control: Pressing the button changes the status as follows:

<table>
<thead>
<tr>
<th>Light pattern</th>
<th>Status</th>
<th>Light pattern</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Green on</td>
<td>Log-in (Ready)</td>
<td>Red on</td>
<td>Log-out</td>
</tr>
<tr>
<td>Slow Green Flashing</td>
<td>Log-in (Not Ready)</td>
<td>Log-in (Ready)</td>
<td>Green on</td>
</tr>
<tr>
<td>Red on</td>
<td>Log-out</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Off</td>
<td>Extension in another incoming call distribution group</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
1.2.2 Incoming Call Distribution Group Features

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-T7440, and KX-T7441.

- **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

- **4.9 [2-6-1] Numbering Plan—Main—Features—» Incoming Call Queue Monitor**
- **5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Miscellaneous—» Supervisor Extension Number**

PT Programming Manual References

None

User Manual References

- **1.9.2 Monitoring and Controlling the Call Status of an Incoming Call Distribution Group (Incoming Call Distribution Group Monitor)**
1.2.3 Call Distribution Port Group

Description

When using the Virtual 16-Channel VoIP Gateway Card (V-IPGW16) to receive calls, 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]

<table>
<thead>
<tr>
<th>V-IPGW Card Number</th>
<th>Port Number</th>
<th>Call Distribution Port Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
<td>CDPG 1</td>
</tr>
<tr>
<td>1</td>
<td>2</td>
<td>CDPG 2</td>
</tr>
<tr>
<td>1</td>
<td>8</td>
<td>CDPG 2</td>
</tr>
<tr>
<td>2</td>
<td>1</td>
<td>CDPG 1</td>
</tr>
<tr>
<td>2</td>
<td>8</td>
<td>CDPG 3</td>
</tr>
</tbody>
</table>

[Programming Example of the Hunt Pattern Table]

<table>
<thead>
<tr>
<th>No.</th>
<th>Leading Number</th>
<th>Call Distribution Port Group (1st)</th>
<th>Call Distribution Port Group (2nd)</th>
<th>...</th>
<th>Call Distribution Port Group (16th)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>10</td>
<td>CDPG 1</td>
<td>CDPG 4</td>
<td>...</td>
<td>-</td>
</tr>
<tr>
<td>2</td>
<td>20</td>
<td>CDPG 1</td>
<td>CDPG 2</td>
<td>...</td>
<td>CDPG 3</td>
</tr>
<tr>
<td>:</td>
<td>:</td>
<td>:</td>
<td>:</td>
<td>...</td>
<td>:</td>
</tr>
<tr>
<td>32</td>
<td>300</td>
<td>CDPG 8</td>
<td>CDPG 11</td>
<td>...</td>
<td>-</td>
</tr>
</tbody>
</table>

*1 → 3.7 [1-1] Slot—Port Property - Virtual IP Gateway Port—Call Distribution Port Group

*2 → 3.13 [1-1] Slot—Shelf Property - Virtual IP Gateway—Hunt Pattern—Hunt Pattern 1–16—Leading Number

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:

- The leading number (10) is searched for in the Hunt Pattern Table.
- CDPG 1 is the 1st priority CDPG.
- The call is routed to Port 1 of the first card.
- Port 1 of the first card is busy.
- The call is rerouted to Port 1 of the second card.
- Port 1 of the second card is available.
- The call is answered.

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.

**Installation Manual References**

3.4.1 Virtual Cards

**PC Programming Manual References**

3.7 [1-1] Slot—Port Property - Virtual IP Gateway Port
3.13 [1-1] Slot—Shelf Property - Virtual IP Gateway—Hunt Pattern

**PT Programming Manual References**

None

**Feature Guide References**

1.30.2 Voice over Internet Protocol (VoIP) Network
1.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features

1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)

1.3.1.1 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. (→ 1.3.1.2 Call Forwarding (FWD))

2. **DND**
   Callers to an extension will hear a tone to inform them that the extension user is not available. (→ 1.3.1.3 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. (→ 1.3.1.4 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.
## 1.3.1.2 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:

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

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

**Intercom Calls to Extension**

**Extension**

- Forwards to Another Extension
- Forwards to Outside Party

**Available Forwarding Type:** All Calls, Busy, No Answer, Busy/No Answer

**Incoming Call Distribution Group**

- Forwards to Another Extension
- Forwards to Outside Party

**Available Forwarding Type:** All Calls
## Available Destinations

<table>
<thead>
<tr>
<th>Destination</th>
<th>Availability</th>
<th>Condition for Original Extension/Incoming Call Distribution Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/SIP Extension/ISDN Extension/T1-OPX)</td>
<td>✓</td>
<td>Only available when FWD to extension is allowed through COS programming.¹</td>
</tr>
<tr>
<td>PS</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>PS Ring Group</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Floating Extension no. for SVM</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>DISA</td>
<td>✓</td>
<td>Only available for incoming trunk calls. Incoming intercom and door-phone calls cannot be forwarded to a DISA floating extension number.</td>
</tr>
<tr>
<td>Analogue/ISDN Remote Maintenance</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Idle Line Access no. + Phone no.</td>
<td>✓</td>
<td>Only available when FWD to trunk is allowed through COS programming.</td>
</tr>
<tr>
<td>Trunk Group Access no. + Trunk Group no. + Phone no.</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Other PBX Extension (TIE with no PBX Code)</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Other PBX Extension (TIE with PBX Code)</td>
<td>✓</td>
<td>Only available when FWD to trunk is allowed through COS programming.</td>
</tr>
</tbody>
</table>

¹ If an extension user is not permitted by COS to call a certain extension (→ 1.1.2.2 Internal Call Block), the FWD feature will not function if that extension is set as the forwarding destination.

## 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. (→ 1.11.8 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 (when the destination extension is busy or in DND mode)
1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)

- Incoming Call Distribution Group—Overflow

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 destination extension rings, and then the call is redirected to the intercept destination by the Intercept Routing—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. (→ 1.2.2.4 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. (→ 1.19.1 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, or T1 (LCOT/GCOT) cards
[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.

PC Programming Manual References
4.9 [2-6-1] Numbering Plan—Main—Features—FWD No Answer Timer Set
4.12 [2-7-1] Class of Service—COS Settings
  →CO & SMDR—Call Forward to CO
  →Programming & Manager—Group Forward Set
  →Optional Device & Other Extensions—Remote Operation by Other Extension
5.1 [3-1-1] Trunk Group—TRG Settings—Main
  →CO-CO Duration Time
  →Extension-CO Duration Time
5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Group Log / Group FWD
6.3 [4-1-2] Wired Extension—FWD/DND
6.12 [4-2-2] 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
1.1.1.6 Intercept Routing
1.2.1 Idle Extension Hunting
1.2.2.6 Overflow Feature
2.2.1 Class of Service (COS)

User Manual References
1.6.1 Forwarding Calls
1.3.1.3 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 (→ 1.2.1 Idle Extension Hunting) or the Intercept Routing—Busy/DND feature (→ 1.1.1.6 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. (→ 1.15.1 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, or T1 (LCOT/GCOT) 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**

4.5 [2-3] Timers & Counters—Dial / IRNA / Recall / Tone—◆ Tone Length—Busy Tone / DND Tone
4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature
  →◆ BSS / OHCA / Whisper OHCA / DND Override
  →◆ BSS / OHCA / Whisper OHCA / DND Override-2
4.12 [2-7-1] Class of Service—COS Settings—Extension Feature—◆ DND Override
6.3 [4-1-2] Wired Extension—FWD/DND
  →◆ Call from CO—DND Status Availability
  →◆ Call from Extension—DND Status Availability

**PT Programming Manual References**

[507] DND Override

**Feature Guide References**

2.2.1 Class of Service (COS)
User Manual References

1.2.4 When the Dialed Line is Busy or There is No Answer
1.8.2 Refusing Incoming Calls (Do Not Disturb [DND])
1.3.1.4 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.

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>FWD/DND for Extension</td>
<td>FWD/DND—Internal Works for incoming intercom calls</td>
</tr>
<tr>
<td></td>
<td>FWD/DND—External Works for incoming trunk calls</td>
</tr>
<tr>
<td></td>
<td>FWD/DND—Both Works for all incoming calls</td>
</tr>
</tbody>
</table>

[Button Status]

The FWD/DND button shows the current status as follows:

<table>
<thead>
<tr>
<th>Light Pattern</th>
<th>Status (default)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Red on</td>
<td>FWD on</td>
</tr>
<tr>
<td>Slow red flashing</td>
<td>DND on</td>
</tr>
<tr>
<td>Off</td>
<td>FWD/DND off</td>
</tr>
</tbody>
</table>

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.

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>FWD for Incoming Call Group</td>
<td>Group FWD—Internal Works for incoming intercom calls</td>
</tr>
<tr>
<td></td>
<td>Group FWD—External Works for incoming trunk calls</td>
</tr>
<tr>
<td></td>
<td>Group FWD—Both Works for all incoming calls</td>
</tr>
</tbody>
</table>

[Button Status]

The Group FWD button shows the current status as follows:

<table>
<thead>
<tr>
<th>Light Pattern</th>
<th>Status (default)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Red on</td>
<td>FWD on</td>
</tr>
<tr>
<td>Off</td>
<td>FWD off</td>
</tr>
</tbody>
</table>

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. (→ 1.25.6 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:
  ![FWD DND Off]

- 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**

4.9 [2-6-1] 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

4.18 [2-9] System Options—Option 1
   → ◆ PT Fwd / DND—Fwd LED
   → ◆ PT Fwd / DND—DND LED
   → ◆ PT Fwd / DND—Fwd/DND key mode when Idle

6.1 [4-1-1] Wired Extension—Extension Settings—FWD / DND
1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)

6.5 [4-1-4] Wired Extension—Flexible Button
→ Type
→ Ext. No. / Floating Ext. No. (for Group Fwd (Both))
→ Ext. No. / Floating Ext. No. (for Group Fwd (External))
→ Ext. No. / Floating Ext. No. (for Group Fwd (Internal))

6.10 [4-2-1] Portable Station—Extension Settings—FWD / DND Reference

6.13 [4-2-3] Portable Station—Flexible Button
→ Type
→ Ext. No. / Floating Ext. No. (for Group Fwd (Both))
→ Ext. No. / Floating Ext. No. (for Group Fwd (External))
→ Ext. No. / Floating Ext. No. (for Group Fwd (Internal))

6.17 [4-3] DSS Console
→ Type
→ Ext. No. / Floating Ext. No. (for Group Fwd (Both))
→ Ext. No. / Floating Ext. No. (for Group Fwd (External))
→ Ext. No. / Floating Ext. No. (for Group Fwd (Internal))

PT Programming Manual References
None

Feature Guide References

1.20.1 Fixed Buttons
1.20.2 Flexible Buttons

User Manual References

3.1.2 Settings on the Programming Mode
1.4 Answering Features

1.4.1 Answering Features

1.4.1.1 Answering Features—SUMMARY

Description

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

<table>
<thead>
<tr>
<th>Destination</th>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>At the own extension (PT only)</td>
<td>Line Preference—Incoming</td>
<td>A user can select the line seized when going off-hook.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>→ 1.4.1.2  Line Preference—Incoming</td>
</tr>
<tr>
<td></td>
<td>Direct One-touch Answering</td>
<td>A user can answer an incoming call simply by pressing the flashing button.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>→ 1.4.1.3  Call Pickup</td>
</tr>
<tr>
<td>At another extension</td>
<td>Call Pickup—Directed/Group</td>
<td>A user can pick up a specified extension’s call or a call in a specified call pickup group.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>→ 1.4.1.4  Hands-free Answerback</td>
</tr>
</tbody>
</table>
1.4.1.2 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).

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>No Line</td>
<td>Select a line by pressing the desired Line Access button to answer an incoming call after you go off-hook.</td>
</tr>
<tr>
<td>PDN</td>
<td>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. (→ 1.7.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension)</td>
</tr>
<tr>
<td>Prime Line</td>
<td>Answer a call arriving at a Flexible CO or ICD Group button (on which the &quot;Prime Line&quot; is assigned) simply by going off-hook. This works even when multiple calls are received simultaneously.</td>
</tr>
<tr>
<td>Ringing Line (default)</td>
<td>Answer the longest ringing call at one’s telephone simply by going off-hook when multiple calls arrive.</td>
</tr>
</tbody>
</table>

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

6.1 [4-1-1] Wired Extension—Extension Settings—Option 4 — Incoming Preferred Line
6.10 [4-2-1] Portable Station—Extension Settings—Option 4 — Incoming Preferred Line

PT Programming Manual References

None

User Manual References

3.1.2 Settings on the Programming Mode
1.4.1.3 Call Pickup

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

<table>
<thead>
<tr>
<th>Type</th>
<th>Picking up Call Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Directed</td>
<td>A specified extension’s call.</td>
</tr>
<tr>
<td>Group</td>
<td>A call within a specified call pickup group.</td>
</tr>
</tbody>
</table>

Call Pickup Deny:
Preventing other extensions from picking up calls ringing at your extension is also possible.

Conditions
- **Call Pickup applies to:**
  Intercom, trunk, and doorphone calls
- **Internal Call Block**
  An extension that is restricted by COS from calling certain extensions (→ 1.1.2.2 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. (→ 2.2.2 Group)

[Example]

PC Programming Manual References
4.9 [2-6-1] Numbering Plan—Main—Features
→ Group Call Pickup
PT Programming Manual References

[650] Extension User Groups of a Pickup Group

Feature Guide References

2.2.1 Class of Service (COS)
3.1 Capacity of System Resources

User Manual References

1.3.3 Answering a Call Ringing at Another Telephone (Call Pickup)
1.4.1.4 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:

<table>
<thead>
<tr>
<th>Type</th>
<th>Answering Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercom Call</td>
<td>Established immediately after a beep tone at the called extension and the caller hears a confirmation tone.</td>
</tr>
<tr>
<td>Trunk Call</td>
<td>Established after a specified number of rings, a called extension hears a beep tone.</td>
</tr>
</tbody>
</table>

**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. (→ 1.2.2.2 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.

- **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**

4.5 [2-3] Timers & Counters—Dial / IRNA / Recall / Tone—◆ Tone Length—Reorder Tone for PT Hands-free

4.18 [2-9] 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

6.1 [4-1-1] Wired Extension—Extension Settings

  →Option 5—◆ Automatic Answer for CO Call
  →Option 6—◆ Forced Automatic Answer

**PT Programming Manual References**

None
1.4.1 Answering Features

Feature Guide References
1.5.3 Intercom Call

User Manual References
1.3.2 Answering Hands-free (Hands-free Answerback)
1.5 Making Call Features

1.5.1 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. (→ 1.6.1.5 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
1.5.2 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:**
  - **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


   → ◆ Dial—Extension First Digit
   → ◆ Dial—Extension Inter-digit
   → ◆ Tone Length—Reorder Tone for PT Handset
   → ◆ Tone Length—Reorder Tone for PT Hands-free

PT Programming Manual References

None
1.5.3 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.

- **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 (→ 1.7.1 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:

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ringback Tone</td>
<td>Indicates the called party is being called.</td>
</tr>
<tr>
<td>Confirmation Tone</td>
<td>Indicates the called party has set voice-calling.</td>
</tr>
<tr>
<td>Busy Tone</td>
<td>Indicates the called party is busy.</td>
</tr>
<tr>
<td>DND Tone</td>
<td>Indicates the called party has set DND.</td>
</tr>
</tbody>
</table>

**PC Programming Manual References**

4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature—Alternate Calling - Ring / Voice
6.1 [4-1-1] Wired Extension—Extension Settings
  ➔Main ➔Extension Number
  ➔Main ➔Extension Name
  ➔Option 3 ➔Intercom Call by Voice
6.10 [4-2-1] Portable Station—Extension Settings—Main
  ➔Extension Number
  ➔Extension Name
1.5.3 Intercom Call

PT Programming Manual References
[003] Extension Number
[004] Extension Name

Feature Guide References
1.20.2 Flexible Buttons

User Manual References
1.2.1 Basic Calling
1.2.6 Alternating the Calling Method (Alternate Calling—Ring/Voice)
1.13.2 Using the Directories
3.1.2 Settings on the Programming Mode
1.5.4 Trunk Call Features

1.5.4.1 Trunk Call Features—SUMMARY

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

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Emergency Call</td>
<td>A user can dial the preprogrammed emergency numbers regardless of the restrictions imposed on the extension.</td>
</tr>
<tr>
<td></td>
<td>→ 1.5.4.2 Emergency Call</td>
</tr>
<tr>
<td>Account Code Entry</td>
<td>A user can enter an account code to identify outgoing calls for accounting and billing purposes.</td>
</tr>
<tr>
<td></td>
<td>→ 1.5.4.3 Account Code Entry</td>
</tr>
<tr>
<td>Pulse to Tone Conversion</td>
<td>A user can temporarily switch from Pulse mode to DTMF mode to access special services.</td>
</tr>
<tr>
<td></td>
<td>→ 1.5.4.4 Dial Type Selection</td>
</tr>
<tr>
<td>Pause Insertion</td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>→ 1.5.4.7 Pause Insertion</td>
</tr>
<tr>
<td></td>
<td>→ 1.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)</td>
</tr>
<tr>
<td></td>
<td>→ 1.5.4.9 Special Carrier Access Code</td>
</tr>
</tbody>
</table>
1.5.4.2 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 (→ 1.5.4.3 Account Code Entry)
  - in any TRS/Barring levels (→ 1.9.1 Toll Restriction (TRS)/Call Barring (Barring))
  - after the preprogrammed call charge limit is reached (→ 1.9.2 Budget Management)
  - in Extension Dial Lock (→ 1.9.3 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. (→ 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP))
  The CLIP number assigned to the extension will be sent regardless of the settings such as CLIR or CLIP number assigned to an ISDN port to be used. This feature is only available when using a PRI (PRI23) line with E911-compatible services.

**PC Programming Manual References**

9.4 [7-4] Emergency Dial

**PT Programming Manual References**

[304] Emergency Number

**Feature Guide References**

3.1 Capacity of System Resources
1.5.4.3 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.

<table>
<thead>
<tr>
<th>Mode</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Option</td>
<td>A user can enter an account code if needed at any time desired.</td>
</tr>
<tr>
<td>Forced</td>
<td>A user must always enter an account code before seizing a trunk.</td>
</tr>
</tbody>
</table>

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. (→ 1.5.4.2 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. (→ 1.9.6 Verification Code Entry)

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—Account Code Entry
4.12 [2-7-1] Class of Service—COS Settings—CO & SMDR—Account Code Mode

PT Programming Manual References

[508] Account Code Mode

Feature Guide References

1.20.2 Flexible Buttons
1.26.1.1 Station Message Detail Recording (SMDR)
2.2.1 Class of Service (COS)
1.5.4 Trunk Call Features

User Manual References

1.2.1 Basic Calling
1.5.4.4 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:

<table>
<thead>
<tr>
<th>Mode</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DTMF (Dual Tone Multi-Frequency)</td>
<td>The dialling signal from an extension is converted to tone dialling.</td>
</tr>
<tr>
<td></td>
<td>DTMF signals are transmitted to the trunk.</td>
</tr>
<tr>
<td>Pulse Dial (Rotary)</td>
<td>The dialling signal from an extension is converted to rotary dialling.</td>
</tr>
<tr>
<td></td>
<td>Rotary pulses are transmitted to the trunk.</td>
</tr>
</tbody>
</table>

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 \( \star \). 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

3.25 [1-1] Slot—Port Property - LCO Port  
   → ◆ Dialling Mode  
   → ◆ DTMF Width  
   → ◆ Pulse Speed  

3.33 [1-1] Slot—Port Property - T1 Port  
   → ◆ CO Dial Mode  
   → ◆ DTMF Width  
   → ◆ CO Pulse Speed  

3.39 [1-1] Slot—Port Property - E1 Port  
   → ◆ CO Dial Mode  
   → ◆ DTMF Width  
   → ◆ CO Pulse Speed  

PT Programming Manual References

[410] LCOT Dialling Mode  
[411] LCOT Pulse Rate  
[412] LCOT DTMF Minimum Duration
1.5.4.5 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 (→ 1.26.1.1 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
3.25 [1-1] Slot—Port Property - LCO Port—_reverse Detection

PT Programming Manual References
[415] LCOT Reverse Circuit
1.5.4.6 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**

- This feature is not available for the following LCOT cards: KX-NCP1180NE (LCOT4)
- 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.

**PC Programming Manual References**

3.25 [1-1] Slot—Port Property - LCO Port— Busy Out Status
4.9 [2-6-1] Numbering Plan—Main—Features— Busy Out Cancel
4.12 [2-7-1] Class of Service—COS Settings—Programming & Manager— Manager
4.18 [2-9] System Options—Option 5— Busy Out—Busy Out for Analogue CO

**PT Programming Manual References**

[511] Manager Assignment

**Feature Guide References**

2.2.6 Manager Features

**User Manual References**

2.1.6 Allowing Users to Seize an Unavailable Outside Line (Trunk Busy Out)
1.5.4.7 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 (→ 1.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX))
- **b.** Special Carrier Access code (→ 1.5.4.9 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. (1.10.1 Automatic Route Selection (ARS))

PC Programming Manual References

3.25 [1-1] Slot—Port Property - LCO Port—♦ Pause Time
3.33 [1-1] Slot—Port Property - T1 Port—♦ Pause Time
3.39 [1-1] Slot—Port Property - E1 Port—♦ Pause Time
8.4 [6-4] Second Dial Tone

PT Programming Manual References

[416] LCOT Pause Time
1.5.4.8 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 (1.5.4.7 Pause Insertion).

[Example]

```
Dials “9-0-01-23-4567”.
```

<table>
<thead>
<tr>
<th>Idle Line Access No.: 9</th>
</tr>
</thead>
<tbody>
<tr>
<td>Extn. 101</td>
</tr>
<tr>
<td>Extn. 102</td>
</tr>
</tbody>
</table>

```
Dials “9-0-01-23-4567”.
```

<table>
<thead>
<tr>
<th>Host PBX Access Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>Telephone No.</td>
</tr>
</tbody>
</table>

```
Dials “9-101”.
```

<table>
<thead>
<tr>
<th>Idle Line Access No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Telephone No. of the Host PBX</td>
</tr>
</tbody>
</table>

**Note**

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

**Conditions**

- TRS/Barring
TRRS/Barring checks only the dialled telephone number excluding the Host PBX Access code when accessing the telephone company through the host PBX. (→ 1.9.1 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. (1.10.1 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**

5.1 [3-1-1] Trunk Group—TRG Settings—Host PBX Access Code  

**PT Programming Manual References**

[471] Host PBX Access Code

**Feature Guide References**

1.26.1.1 Station Message Detail Recording (SMDR)  
3.1 Capacity of System Resources
1.5.4.9 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. (→ 1.5.4.7 Pause Insertion)

Conditions

- **TRS/Barring**
  TRS/Barring checks only the dialled telephone number excluding the Special Carrier Access code. (→ 1.9.1 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. (1.10.1 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. (→ 1.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX))

PC Programming Manual References

9.3 [7-3] Special Carrier

PT Programming Manual References

[303] Special Carrier Access Code

Feature Guide References

1.10.1 Automatic Route Selection (ARS)
3.1 Capacity of System Resources
1.5.5 Seizing a Line Features

1.5.5.1 Seizing a Line Features—SUMMARY

**Description**

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

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Line Preference—Outgoing</td>
<td>A user can select the line to be seized when going off-hook.</td>
</tr>
<tr>
<td></td>
<td>→ 1.5.5.2 Line Preference—Outgoing</td>
</tr>
<tr>
<td>Trunk Access</td>
<td>A user can select the Trunk Access method every time he makes a trunk call.</td>
</tr>
<tr>
<td></td>
<td>→ 1.5.5.3 Trunk Access</td>
</tr>
</tbody>
</table>
1.5.5.2 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:

<table>
<thead>
<tr>
<th>Line Preference</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ICM/PDN</td>
<td>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. (→ 1.7.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension)</td>
</tr>
<tr>
<td>Idle Line</td>
<td>When an extension user goes off-hook, an idle trunk is selected automatically from the assigned trunk groups.</td>
</tr>
<tr>
<td>No Line</td>
<td>When an extension user goes off-hook, no line is selected. The extension user must select the desired line to make a call.</td>
</tr>
<tr>
<td>Prime Line</td>
<td>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.</td>
</tr>
</tbody>
</table>

**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**

4.13 [2-7-2] Class of Service—External Call Block
5.2 [3-1-2] Trunk Group—Local Access Priority
6.1 [4-1-1] Wired Extension—Extension Settings—Option 4—♦ Outgoing Preferred Line
6.10 [4-2-1] 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
1.5.5 Seizing a Line Features

1.5.5.3 Trunk Access

Description

The following methods can be used to access a trunk:

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

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:

<table>
<thead>
<tr>
<th>Type</th>
<th>Parameter</th>
</tr>
</thead>
<tbody>
<tr>
<td>Loop-CO (L-CO)</td>
<td>No parameter (all assigned trunk groups through system programming are applied.)</td>
</tr>
<tr>
<td>Group-CO (G-CO)</td>
<td>A specified trunk group.</td>
</tr>
<tr>
<td>Single-CO (S-CO)</td>
<td>A specified trunk.</td>
</tr>
</tbody>
</table>

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.

- 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

3.21 [1-1] Slot—Port Property - Extension Port— Connection
3.25 [1-1] Slot—Port Property - LCO Port— Connection
3.28 [1-1] Slot—Port Property - BRI Port— Connection
3.30 [1-1] Slot—Port Property - PRI Port— Connection
3.33 [1-1] Slot—Port Property - T1 Port— Connection
3.39 [1-1] Slot—Port Property - E1 Port— Connection

4.9 [2-6-1] Numbering Plan—Main—Features
  → Idle Line Access (Local Access)
  → Trunk Group Access
  → Single CO Line Access

4.13 [2-7-2] Class of Service—External Call Block

5.1 [3-1-1] Trunk Group—TRG Settings—Main— Line Hunting Order
5.2 [3-1-2] Trunk Group—Local Access Priority

6.5 [4-1-4] Wired Extension—Flexible Button
  → Type
  → Parameter Selection (for Single CO)
  → Parameter Selection (for Group CO)

6.13 [4-2-3] Portable Station—Flexible Button
  → Type
  → Parameter Selection (for Single CO)
  → Parameter Selection (for Group CO)

6.17 [4-3] DSS Console
  → Type
  → Parameter Selection (for Single CO)
  → Parameter Selection (for Group CO)

10.1 [8-1] System Setting— ARS Mode
12.1 [10-1] CO Line Settings— CO Name

PT Programming Manual References

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

Feature Guide References

1.20.2 Flexible Buttons
2.2.1 Class of Service (COS)

User Manual References

1.2.1 Basic Calling
## 1.6 Memory Dialling Features

### 1.6.1 Memory Dialling Features

#### 1.6.1.1 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**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Storing Method &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>One-touch Dialling</strong></td>
<td>• Personal Programming&lt;br&gt;• System Programming&lt;br&gt;(PC Programming only)</td>
</tr>
<tr>
<td></td>
<td>→ 1.6.1.2 One-touch Dialling</td>
</tr>
<tr>
<td><strong>KX-T7710 One-touch Dialling</strong></td>
<td>System Programming&lt;br&gt;(PC Programming only)</td>
</tr>
<tr>
<td></td>
<td>→ 1.6.1.3 KX-T7710 One-touch Dialling</td>
</tr>
<tr>
<td><strong>Last Number Redial (Outgoing Call Log)</strong></td>
<td>Recently dialled telephone numbers are automatically stored.</td>
</tr>
<tr>
<td></td>
<td>→ 1.6.1.4 Last Number Redial</td>
</tr>
<tr>
<td><strong>Speed Dialling</strong></td>
<td><strong>Personal</strong>&lt;br&gt;• Personal Programming&lt;br&gt;• Personal Operation with the Feature Number&lt;br&gt;• System Programming&lt;br&gt;(PC Programming only)</td>
</tr>
<tr>
<td></td>
<td>→ 1.6.1.5 Speed Dialling—Personal/System</td>
</tr>
<tr>
<td></td>
<td><strong>System</strong>&lt;br&gt;System Programming</td>
</tr>
<tr>
<td></td>
<td>→ 1.6.1.5 Speed Dialling—Personal/System</td>
</tr>
<tr>
<td><strong>Quick Dialling</strong></td>
<td>System Programming&lt;br&gt;(PC Programming only)</td>
</tr>
<tr>
<td></td>
<td>→ 1.6.1.6 Quick Dialling</td>
</tr>
<tr>
<td><strong>Hot Line</strong></td>
<td>• Personal Programming&lt;br&gt;• Personal Operation with the Feature Number&lt;br&gt;• System Programming&lt;br&gt;(PC Programming only)</td>
</tr>
<tr>
<td></td>
<td>→ 1.6.1.7 Hot Line</td>
</tr>
</tbody>
</table>
2. Valid Input

<table>
<thead>
<tr>
<th>Input</th>
<th>Display while Entering</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>0–9/×/#</td>
<td>0–9/×/#</td>
<td>Store the digits × and #.</td>
</tr>
<tr>
<td>PAUSE</td>
<td>P</td>
<td>Store a pause by pressing the PAUSE button. (→ 1.5.4.7  Pause Insertion)</td>
</tr>
<tr>
<td>FLASH/RECALL</td>
<td>F</td>
<td>Store a flash/recall signal (EFA mode) by pressing the FLASH/RECALL button at the beginning of the number. (→ 1.11.7  External Feature Access (EFA))</td>
</tr>
<tr>
<td>INTERCOM</td>
<td>[ ]</td>
<td>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.</td>
</tr>
<tr>
<td>TRANSFER</td>
<td>T</td>
<td>Store a transfer command by pressing the TRANSFER button at the beginning of the number (used only for a One-touch Dialing). (→ 1.12.1  Call Transfer)</td>
</tr>
</tbody>
</table>

[Example] Storing "T + 305" = Transferring a call to extension 305.

Available only when in system/personal programming mode

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.
1.6.1.2 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.

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

6.5 [4-1-4] Wired Extension—Flexible Button
  → Type
  → Dial (for One-touch)
6.10 [4-2-1] Portable Station—Extension Settings—Option 6—Flexible Button Programming Mode
6.13 [4-2-3] Portable Station—Flexible Button
  → Type
  → Dial (for One-touch)
6.17 [4-3] DSS Console
  → Type
  → Dial (for One-touch)

PT Programming Manual References

None

Feature Guide References

1.20.2 Flexible Buttons
3.1 Capacity of System Resources

User Manual References

1.2.2 Easy Dialling
1.6.1.3 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]**

<table>
<thead>
<tr>
<th>Button</th>
<th>Desired Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>MESSAGE</td>
<td>702 (Message Waiting [To Call Back])</td>
</tr>
<tr>
<td>One-touch Dial 01</td>
<td>100 (Hotel Operator)</td>
</tr>
<tr>
<td>One-touch Dial 02</td>
<td>7601 (Wake-up Call)</td>
</tr>
<tr>
<td>One-touch Dial 03</td>
<td>102 (Restaurant)</td>
</tr>
</tbody>
</table>

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**

4.9 [2-6-1] Numbering Plan—Main—KX-T7710

**PT Programming Manual References**

None

**Feature Guide References**

1.20.2 Flexible Buttons

**User Manual References**

1.2.2 Easy Dialling
1.6.1.4 Last Number Redial

Description

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

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 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 outside 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.
• Interrupt Redial
  When an outside party or seized trunk 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 a number 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.

PC Programming Manual References

  ➔ Automatic Redial—Repeat Counter
  ➔ Automatic Redial—Repeat Interval
  ➔ Automatic Redial—Redial Call Ring Duration
  ➔ Automatic Redial—Analogue CO Mute / Busy Detection Timer
4.9 [2-6-1] Numbering Plan—Main—Features—◆ Redial
4.18 [2-9] 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
6.1 [4-1-1] Wired Extension—Extension Settings—Option 7—◆ Outgoing Call Log Memory
6.10 [4-2-1] Portable Station—Extension Settings—Option 7—◆ Outgoing Call Log Memory
1.6.1 Memory Dialling Features

PT Programming Manual References
[205] Automatic Redial Repeat Times
[206] Automatic Redial Interval

Feature Guide References
3.1 Capacity of System Resources

User Manual References
1.2.3 Redial
1.13.1 Using the Call Log

Visit our Web Site: www.targettelcom.com
1.6.1.5 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.

**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.
- **Call Directory—Speed Dialling**
  Display PT users can make a call by selecting stored names on the display.

**[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 and the SVM Log display are also locked, and the voice messages in the user’s message box cannot be played back. An extension personal identification number (PIN) is required to use this feature. (→ 1.28.1 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. (→ 1.9.1 Toll Restriction (TRS)/Call Barring (Barring))
- **System Speed Dialling Display by AUTO DIAL/STORE Button**
  Pressing the AUTO DIAL/STORE button on a display DPT while on-hook can display the System Speed Dialling Directory.

**PC Programming Manual References**

4.9 [2-6-1] Numbering Plan—Main—Features
→ ◆ System Speed Dialling / Personal Speed Dialling
→ ◆ Personal Speed Dialling - Programming
4.12 [2-7-1] Class of Service—COS Settings—TRS—◆ TRS Level for System Speed Dialling
6.4 [4-1-3] Wired Extension—Speed Dial
8.1 [6-1] System Speed Dial
8.6 [6-6] Tenant—◆ System Speed Dial

**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**

3.1 Capacity of System Resources
User Manual References

1.2.2 Easy Dialling
1.13.1 Using the Call Log
1.13.2 Using the Directories
3.1.2 Settings on the Programming Mode
3.3 Customising Your System (System Programming)
1.6.1 Memory Dialling Features

1.6.1.6 Quick Dialling

Description

An extension user can access an extension or feature by simply dialling a 1-4 or 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.
  (→ 2.3.6 Flexible Numbering/Fixed Numbering)
- The following example shows how Quick Dialling numbers can be stored and utilised:

<table>
<thead>
<tr>
<th>Location No.</th>
<th>Quick Dialling No.</th>
<th>Desired Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quick Dialling 01</td>
<td>110</td>
<td>9110 (Trunk Call)</td>
</tr>
<tr>
<td>Quick Dialling 02</td>
<td>5</td>
<td>3016 (Room Service)</td>
</tr>
<tr>
<td>Quick Dialling 03</td>
<td>2011</td>
<td>90123456789 (Another Branch)</td>
</tr>
</tbody>
</table>

PC Programming Manual References

4.10  [2-6-2] Numbering Plan—Quick Dial

PT Programming Manual References

None

Feature Guide References

1.30.3  Virtual Private Network (VPN)
3.1  Capacity of System Resources

User Manual References

1.2.2  Easy Dialling
1.6.1.7 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, T1-OPX, and PS

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—◆ Hot Line (Pickup Dial) Program Set / Cancel
6.1 [4-1-1] Wired Extension—Extension Settings—Option 2
  ◆ Pickup Dial Set
  ◆ Pickup Dial No.
6.10 [4-2-1] Portable Station—Extension Settings—Option 2
  ◆ Pickup Dial Set
  ◆ Pickup Dial No.

PT Programming Manual References

[204] Hot Line Waiting Time

Feature Guide References

3.1 Capacity of System Resources

User Manual References

1.2.2 Easy Dialling
1.7 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Features

1.7.1 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 (→ 1.9.5 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:

<table>
<thead>
<tr>
<th>Light Pattern</th>
<th>PDN Button Status</th>
<th>SDN Button Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td>This extension is idle.</td>
<td>The corresponding PDN extension is idle.</td>
</tr>
</tbody>
</table>
### 1.7.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension

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

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

<table>
<thead>
<tr>
<th>Extn. 101 (Boss)</th>
<th>Extn. 102 (Boss)</th>
<th>Extn. 103 (Secretary)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PDN buttons</td>
<td>SDN buttons</td>
<td></td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Rapid Green Flashing)</td>
<td>(Off)</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Extn. 101 (Boss)</th>
<th>Extn. 102 (Boss)</th>
<th>Extn. 103 (Secretary)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PDN buttons</td>
<td>SDN buttons</td>
<td></td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Red On)</td>
<td>(Off)</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Extn. 101 (Boss)</th>
<th>Extn. 102 (Boss)</th>
<th>Extn. 103 (Secretary)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PDN buttons</td>
<td>SDN buttons</td>
<td></td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Off)</td>
<td>(Rapid Red Flashing)</td>
<td></td>
</tr>
<tr>
<td>(Slow Red Flashing)</td>
<td>(Off)</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Extn. 101 (Boss)</th>
<th>Extn. 102 (Boss)</th>
<th>Extn. 103 (Secretary)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PDN buttons</td>
<td>SDN buttons</td>
<td></td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Off)</td>
<td>(Green On)</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Extn. 101 (Boss)</th>
<th>Extn. 102 (Boss)</th>
<th>Extn. 103 (Secretary)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PDN buttons</td>
<td>SDN buttons</td>
<td></td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Off)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
<tr>
<td>(Rapid Green Flashing)</td>
<td>(Off)</td>
<td></td>
</tr>
<tr>
<td>(Green On)</td>
<td>(Off)</td>
<td>(Off)</td>
</tr>
</tbody>
</table>

---

**Feature Guide**

1.7.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension
6. A Call From 333-3333 Arrives at Ext. 102

Extn. 101 (Boss)          Extn. 102 (Boss)          Extn. 103 (Secretary)
PDN buttons          SDN buttons          PDN buttons          SDN buttons          PDN buttons          SDN buttons
(Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)
(Green On) (Extn. 102) (Extn. 102) (Extn. 102) (Extn. 102) (Extn. 102) (Extn. 102)

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

Extn. 101 (Boss)          Extn. 102 (Boss)          Extn. 103 (Secretary)
PDN buttons          SDN buttons          PDN buttons          SDN buttons          PDN buttons          SDN buttons
(Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)
(Rapid Green Flashing) (Red On) (Extn. 102) (Rapid Red Flashing) (Rapid Red Flashing) (Rapid Red Flashing)
(Green On) (Extn. 101) (Extn. 101) (Extn. 101) (Extn. 101) (Extn. 101)

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

Extn. 101 (Boss)          Extn. 102 (Boss)          Extn. 103 (Secretary)
PDN buttons          SDN buttons          PDN buttons          SDN buttons          PDN buttons          SDN buttons
(Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)    (Off) (Extn. 103)
(Rapid Green Flashing) (Red On) (Extn. 102) (Rapid Red Flashing) (Rapid Red Flashing) (Rapid Red Flashing)
(Green On) (Extn. 101) (Extn. 101) (Extn. 101) (Extn. 101) (Extn. 101)

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 newest 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 last, 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 (→ 1.2.2.1 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 (→ 1.5.5.2 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 (→ 1.4.1.2 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 (→ 1.28.3 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 (→ 1.25.5 Wireless XDP Parallel Mode).

**OHCA/Whisper OHCA**
A PDN extension cannot receive OHCA or Whisper OHCA, unless the call is made using a corresponding SDN button (→ 1.8.4.3 Off-hook Call Announcement (OHCA), → 1.8.4.4 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 (→ 1.5.3 Intercom Call).

### [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**

4.12 [2-7-1] Class of Service—COS Settings—PDN/SDN
4.17 [2-8-3] Ring Tone Patterns—Call from Others—◆ Extension—Ring Tone Pattern Plan 1–8
4.18 [2-9] System Options—Option 4—◆ System Wireless—SDN Delayed Ringing with LCD
6.5 [4-1-4] Wired Extension—Flexible Button
   ◆ Type
   ◆ Parameter Selection (for SDN)
   ◆ Ext. No. / Floating Ext. No. (for SDN)
1.7.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension

→ ◆ Optional Parameter (Ringing Tone Type Number) (for Loop CO, Single CO, Group CO, ICD Group, SDN)

6.10 [4-2-1] Portable Station—Extension Settings—Option 9—◆ PDN Delayed Ringing
6.13 [4-2-3] Portable Station—Flexible Button
  → ◆ Type
  → ◆ Parameter Selection (for SDN)
  → ◆ Ext. No. / Floating Ext. No. (for SDN)
6.17 [4-3] DSS Console
  → ◆ Type
  → ◆ Parameter Selection (for SDN)
  → ◆ Ext. No. / Floating Ext. No. (for SDN)
  → ◆ Optional Parameter (Ringing Tone Type Number) (for Loop CO, Single CO, Group CO, ICD Group, SDN)

PT Programming Manual References
None

Feature Guide References
1.2.2.2 Group Call Distribution
1.20.3 LED Indication
2.2.1 Class of Service (COS)
3.1 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)
1.8 Busy Line/Busy Party Features

1.8.1 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 or an ISDN extension.

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—◆ Automatic Callback Busy Cancel
4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature—◆ Automatic Callback Busy

PT Programming Manual References

None

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer
1.8.2 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.

Conditions

• 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 (→ 1.11.5 Data Line Security) has been set.
  b. While being monitored by another extension (→ 1.8.3 Call Monitor).
  c. While receiving OHCA (→ 1.8.4.3 Off-hook Call Announcement (OHCA)) or Whisper OHCA (→ 1.8.4.4 Whisper OHCA).
  d. During a Conference call (→ 1.14.1 Conference Features).
  e. During a doophone call (→ 1.17.1 Doophone Call).
  f. While Live Call Screening (LCS) or Two-way Record is activated (→ 1.24.2.3 Voice Mail DPT (Digital) Integration).
  g. During Consultation Hold.

• This feature is not available for a trunk-to-trunk call via DISA.

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—Executive Override Deny Set / Cancel
4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature—Executive Busy Override
4.12 [2-7-1] Class of Service—COS Settings
  →Extension Feature—Executive Busy Override
  →Programming & Manager—Executive Busy Override Deny
6.1 [4-1-1] Wired Extension—Extension Settings—Option 3—Executive Override Deny
6.10 [4-2-1] 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

2.2.1 Class of Service (COS)

User Manual References

1.2.4 When the Dialed Line is Busy or There is No Answer
1.8.8 Preventing Other People from Joining Your Conversation (Executive Busy Override Deny)
1.8.3 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 (→ 1.8.2 Executive Busy Override) or Data Line Security (→ 1.11.5 Data Line Security) has been set.
  b. While receiving OHCA (→ 1.8.4.3 Off-hook Call Announcement (OHCA)) or Whisper OHCA (→ 1.8.4.4 Whisper OHCA).
  c. During a Conference call (→ 1.14.1 Conference Features).
  d. During a doorphone call (→ 1.17.1 Doorphone Call).
  e. While Live Call Screening (LCS) or Two-way Record is activated (→ 1.24.2.3 Voice Mail DPT (Digital Integration)).
  f. During Consultation Hold.
- This feature stops when the busy extension user presses the following buttons during a conversation (→ 1.20.1 Fixed Buttons and 1.20.2 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

4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature—Call Monitor
4.12 [2-7-1] Class of Service—COS Settings—Extension Feature—Call Monitor
6.1 [4-1-1] Wired Extension—Extension Settings
  → Option 2—Data Mode
  → Option 3—Executive Override Deny
6.10 [4-2-1] Portable Station—Extension Settings—Option 3—Executive Override Deny

PT Programming Manual References

None
Feature Guide References

2.2.1 Class of Service (COS)

User Manual References

1.2.4 When the Dialled Line is Busy or There is No Answer
1.8.4 Second Call Notification to Busy Extension

1.8.4.1 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:

<table>
<thead>
<tr>
<th>Notification Receiving Method</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Waiting Tone</td>
<td>Sends the Call Waiting tone to the busy extension.</td>
</tr>
<tr>
<td></td>
<td>→ 1.8.4.2 Call Waiting Tone</td>
</tr>
<tr>
<td>Off-hook Call Announcement (OH-CA)</td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>→ 1.8.4.3 Off-hook Call Announcement (OHCA)</td>
</tr>
<tr>
<td>Whisper OHCA</td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>→ 1.8.4.4 Whisper OHCA</td>
</tr>
</tbody>
</table>

**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.

<table>
<thead>
<tr>
<th>Calling Extension’s OHCA COS Mode</th>
<th>Called Extension’s Call Waiting Mode</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>OFF</td>
</tr>
<tr>
<td></td>
<td>ON</td>
</tr>
<tr>
<td>Cancel</td>
<td>Call Waiting Tone</td>
</tr>
<tr>
<td></td>
<td>Call Waiting tone</td>
</tr>
<tr>
<td></td>
<td>OHCA</td>
</tr>
<tr>
<td></td>
<td>Whisper OHCA</td>
</tr>
<tr>
<td>Disable</td>
<td>Call Waiting disabled</td>
</tr>
<tr>
<td></td>
<td>Call Waiting tone</td>
</tr>
<tr>
<td></td>
<td>Call Waiting tone</td>
</tr>
<tr>
<td></td>
<td>Call Waiting tone</td>
</tr>
<tr>
<td>Enable</td>
<td>Call Waiting disabled</td>
</tr>
<tr>
<td></td>
<td>Call Waiting tone</td>
</tr>
<tr>
<td></td>
<td>OHCA</td>
</tr>
<tr>
<td></td>
<td>Whisper OHCA</td>
</tr>
<tr>
<td></td>
<td>(or Call Waiting tone)</td>
</tr>
<tr>
<td></td>
<td>(or Call Waiting tone)</td>
</tr>
</tbody>
</table>

- 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

1.1.3.3 Call Waiting

User Manual References

1.2.4 When the Dialed Line is Busy or There is No Answer
1.8.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)
1.8.4.2 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).

PC Programming Manual References

4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature
  → Caller ID—Visual Caller ID Display
  → Caller ID—Visual Caller ID Display
  → BSS / OHCA / Whisper OHCA / DND Override
  → BSS / OHCA / Whisper OHCA / DND Override-2
6.1 [4-1-1] Wired Extension—Extension Settings
  → Option 2— Manual C. Waiting for Extension Call
  → Option 2— Automatic C. Waiting
  → Option 4— Call Waiting Tone Type
6.10 [4-2-1] Portable Station—Extension Settings
  → Option 2— Manual C. Waiting for Extension Call
  → Option 2— Automatic C. Waiting
  → Option 4— Call Waiting Tone Type

PT Programming Manual References

None

User Manual References

1.8.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)
3.1.2 Settings on the Programming Mode
1.8.4.3 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-T7536
  - KX-T7436
  - KX-T7235 (except KX-T7235G/FR/SL/NE)
- If the KX-T7235G/FR/SL/NE are connected to the PBX, the OHCA feature for the KX-T7235 should be disabled through system programming.
- 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 connected to a PC (PC Console or PC Phone) via the USB Module.
  c. The called extension (DPT) is in the Digital XDP connection.

The Call Waiting tone is sent to the called extension. (→ 1.8.4.2 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

4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature
   → BSS / OHCA / Whisper OHCA / DND Override
   → BSS / OHCA / Whisper OHCA / DND Override-2
4.12 [2-7-1] Class of Service—COS Settings—Extension Feature—OHCA / Whisper OHCA

PT Programming Manual References

None

Feature Guide References

2.2.1 Class of Service (COS)

User Manual References

1.8.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)
1.8.4.4 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-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. (→ 1.8.4.2 Call Waiting Tone)
- If the called extension does not use a KX-DT300, 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.)
- 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
4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature
→ BSS / OHCA / Whisper OHCA / DND Override
→ BSS / OHCA / Whisper OHCA / DND Override-2
4.12 [2-7-1] Class of Service—COS Settings—Extension Feature—OHCA / Whisper OHCA

PT Programming Manual References
None

Feature Guide References
2.2.1 Class of Service (COS)

User Manual References
1.8.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)
1.9 Toll Restriction (TRS)/Call Barring (Barring) Features

1.9.1 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.

→ 4.12 [2-7-1] Class of Service—COS Settings—TRS—绿色 TRS Level for System Speed Dialling

→ [300] TRS/Barring Override by 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.

<table>
<thead>
<tr>
<th>Level</th>
<th>Denied Code Tables</th>
<th>Exception Code Tables</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Not Programmable</td>
<td>Not Programmable</td>
</tr>
<tr>
<td>2</td>
<td>Table for Level 2</td>
<td>Tables for Levels 2 through 6</td>
</tr>
<tr>
<td>3</td>
<td>Tables for Levels 2 and 3</td>
<td>Tables for Levels 3 through 6</td>
</tr>
<tr>
<td>4</td>
<td>Tables for Levels 2 through 4</td>
<td>Tables for Levels 4 through 6</td>
</tr>
</tbody>
</table>
[Usage Example] Using this method, certain outgoing trunk calls (e.g., international/cellular phone/long distance) can be restricted as in the example below:

<table>
<thead>
<tr>
<th>Level</th>
<th>Restricted</th>
<th>Allowed</th>
</tr>
</thead>
<tbody>
<tr>
<td>Level 1</td>
<td>No restriction</td>
<td></td>
</tr>
<tr>
<td>Level 2 (Boss)</td>
<td>• International Calls</td>
<td>• Countries where Clients are Located&lt;br&gt;• Cellular Phone Calls&lt;br&gt;• Long Distance Calls&lt;br&gt;• Local Calls</td>
</tr>
<tr>
<td>Level 3 (Secretary)</td>
<td>• International Calls&lt;br&gt;• Cellular Phone Calls</td>
<td>• Boss’s Cellular Phone&lt;br&gt;• Long Distance Calls&lt;br&gt;• Local Calls</td>
</tr>
<tr>
<td>Level 4 (Operator)</td>
<td>• International Calls&lt;br&gt;• Cellular Phone Calls&lt;br&gt;• Long Distance Calls</td>
<td>• Local Calls</td>
</tr>
</tbody>
</table>

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:

<table>
<thead>
<tr>
<th>Level</th>
<th>Denied Code Tables</th>
<th>Exception Code Tables</th>
</tr>
</thead>
<tbody>
<tr>
<td>Level 1</td>
<td>Not Programmable</td>
<td>Not Programmable</td>
</tr>
<tr>
<td>Level 2</td>
<td>00</td>
<td>00xxxx</td>
</tr>
<tr>
<td>Level 3</td>
<td>090</td>
<td>090xxxxxx</td>
</tr>
<tr>
<td>Level 4</td>
<td>0</td>
<td>–</td>
</tr>
</tbody>
</table>

[Programming Example: COS Settings]
### 1.9.1 Toll Restriction (TRS)/Call Barring (Barring)

#### Class of Service—COS Settings—TRS—

**TRS Level—Day, Lunch, Break, Night**

**Miscellaneous—** TRS Override by System Speed Dialling

<table>
<thead>
<tr>
<th>COS No.</th>
<th>Level for Time Mode&lt;sup&gt;1&lt;/sup&gt;</th>
<th>Level for System Speed Dialling&lt;sup&gt;2&lt;/sup&gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Day</td>
<td>Lunch</td>
</tr>
<tr>
<td>1</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>2</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td></td>
<td>:</td>
<td>:</td>
</tr>
</tbody>
</table>

<sup>1</sup> [2-7-1] Class of Service—COS Settings—TRS—◆ TRS Level—Day, Lunch, Break, Night

<sup>2</sup> [7-5] Miscellaneous—◆ TRS Override by System Speed Dialling

→ [501] TRS/Barring Level
→ [509] TRS/Barring Level for System Speed Dialling
An extension user makes a trunk call.

Is the call made by System Speed Dialling?

Yes

Is TRS/Barring Override by System Speed Dialling enabled?

Yes

Checks the TRS/Barring level for the time mode of the extension's COS.

Level 7

Level 1

Levels 2, 3, 4, 5, 6

Is the dialled number found in applicable Denied Code Tables?

No

Yes

Is the dialled number found in applicable Exception Code Tables?

No

Yes

The call is denied. The user hears reorder tone.

The call is made.

**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.

<table>
<thead>
<tr>
<th>Level</th>
<th>Denied Code Tables</th>
<th>Exception Code Tables</th>
</tr>
</thead>
<tbody>
<tr>
<td>Level 1</td>
<td>Not Programmable</td>
<td>Not Programmable</td>
</tr>
</tbody>
</table>
### Denied Code Tables

<table>
<thead>
<tr>
<th>Level 2</th>
<th>Table for Level 2</th>
<th>Table for Level 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Level 3</td>
<td>Table for Level 3</td>
<td>Table for Level 3</td>
</tr>
<tr>
<td>Level 4</td>
<td>Table for Level 4</td>
<td>Table for Level 4</td>
</tr>
<tr>
<td>Level 5</td>
<td>Table for Level 5</td>
<td>Table for Level 5</td>
</tr>
<tr>
<td>Level 6</td>
<td>Table for Level 6</td>
<td>Table for Level 6</td>
</tr>
<tr>
<td>Level 7</td>
<td>Not Programmable</td>
<td>Not Programmable</td>
</tr>
</tbody>
</table>

### Exception Code Tables

- [301] TRS/Barring Denied Code

- [302] TRS/Barring Exception Code

### 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:

<table>
<thead>
<tr>
<th>Level 1</th>
<th>No restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Level 2 (Engineering)</td>
<td>• International Calls</td>
</tr>
<tr>
<td></td>
<td>• Country where Factory is Located</td>
</tr>
<tr>
<td></td>
<td>• Cellular Phone Calls</td>
</tr>
<tr>
<td></td>
<td>• Long Distance Calls</td>
</tr>
<tr>
<td></td>
<td>• Local Calls</td>
</tr>
<tr>
<td>Level 3 (Overseas Sales)</td>
<td>• Cellular Phone Calls</td>
</tr>
<tr>
<td></td>
<td>• Company Cellular Phone</td>
</tr>
<tr>
<td></td>
<td>• International Calls</td>
</tr>
<tr>
<td></td>
<td>• Long Distance Calls</td>
</tr>
<tr>
<td></td>
<td>• Local Calls</td>
</tr>
<tr>
<td>Level 4 (Accounting)</td>
<td>• International Calls</td>
</tr>
<tr>
<td></td>
<td>• Cities where Clients are Located</td>
</tr>
<tr>
<td></td>
<td>• Cellular Phone Calls</td>
</tr>
<tr>
<td></td>
<td>• Local Calls</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Level 1</th>
<th>Not Programmable</th>
</tr>
</thead>
<tbody>
<tr>
<td>Level 2</td>
<td>00 Leading number to deny international calls</td>
</tr>
<tr>
<td>Level 3</td>
<td>090 Leading number to deny cellular phone calls</td>
</tr>
</tbody>
</table>

### Exception Code Tables

<table>
<thead>
<tr>
<th>Level 1</th>
<th>Not Programmable</th>
</tr>
</thead>
<tbody>
<tr>
<td>Level 2</td>
<td>00xx Leading number for country to be allowed</td>
</tr>
<tr>
<td>Level 3</td>
<td>090xxx Leading number for cellular phones to be allowed</td>
</tr>
</tbody>
</table>
Denied Code Tables

<table>
<thead>
<tr>
<th>Level 4</th>
<th>0</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Leading number to deny both international and long distance calls</td>
</tr>
</tbody>
</table>

Exception Code Tables

<table>
<thead>
<tr>
<th></th>
<th>03</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Long distance numbers for cities to be allowed, and leading number of cellular phones</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th></th>
<th>06</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>090</td>
</tr>
</tbody>
</table>

### 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.
  - 6.1 [4-1-1] Wired Extension—Extension Settings—Main— ◆ COS
  - 6.10 [4-2-1] Portable Station—Extension Settings—Main— ◆ COS
  - [602] Class of Service
- TRS/Barring checks are applied to the following:
  - ARS
  - Trunk Access (Idle Line/Trunk Group/S-CO Line)
- It is programmable whether "x" or "#" is checked by the TRS/Barring. This is useful in preventing unauthorised calls which could be possible through certain telephone company exchanges.
  - 9.5 [7-5] Miscellaneous—◆ TRS Check for Dial "* #"
- It is programmable whether TRS/Barring checks the digits dialled after the External Feature Access during a trunk call. (→ 1.11.7 External Feature Access (EFA))
  - 9.5 [7-5] 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 (→ 1.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)) or a Special Carrier Access code (→ 1.5.4.9 Special Carrier Access Code) in the following cases:

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

- 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.
  

- **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.**
  → 9.5 [7-5] 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 (→ 1.9.4 Dial Tone Transfer)
  2. Budget Management (→ 1.9.2 Budget Management)
  3. TRS/Barring Override by System Speed Dialling
  4. Walking COS/Verification Code Entry
     (→ 1.9.5 Walking COS, 1.9.6 Verification Code Entry)
  5. Extension Dial Lock

**PC Programming Manual References**

4.12 [2-7-1] Class of Service—COS Settings—TRS
   → ◆ TRS Level—Day, Lunch, Break, Night
   → ◆ TRS Level for System Speed Dialling
6.1 [4-1-1] Wired Extension—Extension Settings—Main— ◆ COS
6.5 [4-1-4] Wired Extension—Flexible Button
   → ◆ Type
   → ◆ Parameter Selection (for TRS Level Change)
6.10 [4-2-1] Portable Station—Extension Settings—Main— ◆ COS
6.13 [4-2-3] Portable Station—Flexible Button
   → ◆ Type
   → ◆ Parameter Selection (for TRS Level Change)
6.17 [4-3] DSS Console
   → ◆ Type
   → ◆ Parameter Selection (for TRS Level Change)
9.1 [7-1] Denied Code
9.2 [7-2] Exception Code
9.3 [7-3] Special Carrier
9.5 [7-5] 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
1.9.1 Toll Restriction (TRS)/Call Barring (Barring)

[602] Class of Service

Feature Guide References

1.5.5.3 Trunk Access
1.6.1.5 Speed Dialling—Personal/System
1.9.3 Extension Dial Lock
1.10.1 Automatic Route Selection (ARS)
2.2.1 Class of Service (COS)
2.2.4 Time Service
3.1 Capacity of System Resources
1.9.2 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. (→ 1.9.1 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). (→ 1.9.6 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). (1.9.5 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
6.1 [4-1-1] Wired Extension—Extension Settings—Option 3— Charge Limit
6.10 [4-2-1] Portable Station—Extension Settings—Option 3— Charge Limit
8.3 [6-3] Verification Code— Budget Management
8.2 [6-2] Hotel & Charge—Charge— Charge Options—Action at Charge Limit

PT Programming Manual References
None

User Manual References
3.2.2 Manager Programming
1.9.3 Extension Dial Lock

Description
An extension user can change the TRS/Barring level of the telephone (→ 1.9.1 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 (→ 1.28.1 Extension Personal Identification Number (PIN)). This feature is also known as Electronic Station Lockout.

Conditions
• This feature also restricts changing the FWD destination. (→ 1.3.1.2 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
4.9 [2-6-1] Numbering Plan—Main—Features
→ ◆ Extension Dial Lock Set / Cancel
→ ◆ Remote Extension Dial Lock Off
→ ◆ Remote Extension Dial Lock On
4.12 [2-7-1] 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
2.2.1 Class of Service (COS)

User Manual References
1.6.4 Preventing Other People from Using Your Telephone (Extension Dial Lock)
2.1.1 Extension Control
1.9.4 Dial Tone Transfer

**Description**

An extension assigned as the manager can change the TRS/Barring level (→ 1.9.1 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).

1. Call
2. Change TRS/Barring level
3. Make a trunk call

**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**

6.5 [4-1-4] Wired Extension—Flexible Button
  → ◆ Type
  → ◆ Parameter Selection (for TRS Level Change)

6.13 [4-2-3] Portable Station—Flexible Button
  → ◆ Type
  → ◆ Parameter Selection (for TRS Level Change)

6.17 [4-3] DSS Console
  → ◆ Type
  → ◆ Parameter Selection (for TRS Level Change)

**PT Programming Manual References**

None

**Feature Guide References**

1.20.2 Flexible Buttons
User Manual References

2.1.3 Restriction Level Control (Dial Tone Transfer)
1.9.5 Walking COS

Description

A user can enter his extension number and 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 (→ 1.17.9 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 (→ 1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND))
• Incoming Call Distribution Group Log-in/Log-out (→ 1.2.2.7 Log-in/Log-out)
• Absent Message setting (→ 1.19.2 Absent Message)
• Extension Dial Lock (→ 1.9.3 Extension Dial Lock)
• Time Service—Changing the Time Mode (day/lunch/break/night) (→ 2.2.4 Time Service)
• CLIP number setting (CLIP ID) (→ 1.21.1.2 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 (→ 2.2.1 Class of Service (COS))
  – the budget of the specified extension is applied (→ 1.9.2 Budget Management)
  – the Itemised Billing code of the specified extension is applied (→ 1.10.1 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 (→ 1.26.1.1 Station Message Detail Recording (SMDR)).
• Walking COS is also available through DISA. (→ 1.17.6 Direct Inward System Access (DISA))
• Extension PIN

An extension personal identification number (PIN) is required to use this feature. (→ 1.28.1 Extension Personal Identification Number (PIN)) If the wrong PIN is entered three times, the line will be disconnected.

• This feature can not be used for extensions which the extension being operated is prevented from calling by Internal Call Block. (→ 1.1.2.2 Internal Call Block)

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—◆ User Remote Operation / Walking COS / Verification Code
4.12 [2-7-1] Class of Service—COS Settings—Optional Device & Other Extensions—◆ Remote Operation by Other Extension
6.1 [4-1-1] Wired Extension—Extension Settings—Main—◆ Extension PIN
6.10 [4-2-1] Portable Station—Extension Settings—Main—◆ Extension PIN

PT Programming Manual References

[005] Extension Personal Identification Number (PIN)
1.9.5 Walking COS

Feature Guide References

1.17.6 Direct Inward System Access (DISA)
2.2.1 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)
1.9.6 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 (→ 1.9.1 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 (→ 2.2.1 Class of Service (COS))
  – the budget of the specified extension is applied (→ 1.9.2 Budget Management)
  – the Itemised Billing code of the specified extension is applied (→ 1.10.1 Automatic Route Selection
  (ARS))
  – verification code is recorded on SMDR as the call originator, instead of the extension number of
    the actual extension used (→ 1.26.1.1 Station Message Detail Recording (SMDR)).

• Verification Code Entry through DISA
  This feature is also available through DISA. (→ 1.17.6 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]

<table>
<thead>
<tr>
<th>Location</th>
<th>Code(^1)</th>
<th>Name(^2)</th>
<th>PIN(^3)</th>
<th>COS(^4)</th>
<th>Itemised Billing Code for ARS(^5)</th>
<th>Budget(^6)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0001</td>
<td>1111</td>
<td>Tom Smith</td>
<td>1234</td>
<td>1</td>
<td>2323</td>
<td>5000Euro</td>
</tr>
<tr>
<td>0002</td>
<td>2222</td>
<td>John White</td>
<td>9876543210</td>
<td>3</td>
<td>4545</td>
<td>3000Euro</td>
</tr>
</tbody>
</table>

\(^1\) \[8.3 [6-3] Verification Code—\(\text{Verification Code}\)
\(^2\) \[8.3 [6-3] Verification Code—\(\text{User Name}\)
\(^3\) \[8.3 [6-3] Verification Code—\(\text{Verification Code PIN}\)

Feature Guide 145
1.9.6 Verification Code Entry

→ [122] Verification Code Personal Identification Number (PIN)

"  → 8.3 [6-3] Verification Code—◆ COS Number

→ [123] Verification Code COS Number

"  → 8.3 [6-3] Verification Code—◆ Itemised Billing Code for ARS

"  → 8.3 [6-3] Verification Code—◆ Budget Management

PC Programming Manual References


4.9 [2-6-1] Numbering Plan—Main—Features—◆ User Remote Operation / Walking COS / Verification Code

6.1 [4-1-1] Wired Extension—Extension Settings

→ Option 1—◆ ARS Itemised Code

→ Option 3—◆ Charge Limit

6.10 [4-2-1] Portable Station—Extension Settings

→ Option 1—◆ ARS Itemised Code

→ Option 3—◆ Charge Limit

8.3 [6-3] 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

1.9.2 Budget Management

1.10.1 Automatic Route Selection (ARS)

2.2.1 Class of Service (COS)

2.2.6 Manager Features

3.1 Capacity of System Resources

User Manual References

1.2.7 Calling without Restrictions

3.2.2 Manager Programming
1.10 Automatic Route Selection (ARS) Features

1.10.1 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  Assignment
1.10.1 Automatic Route Selection (ARS)

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. (→ 1.5.5.3 Trunk Access)

→ 10.1 [8-1] System Setting—◆ ARS Mode
→ [320] ARS Mode

2. Leading Number Exception Table ② Assignment
Store the telephone numbers that will avoid using the ARS feature.

→ 10.7 [8-6] Leading Number Exception
→ [325] ARS Exception Number

ARS Leading Number Exception Table

<table>
<thead>
<tr>
<th>Location No.</th>
<th>Leading No. Exception</th>
</tr>
</thead>
<tbody>
<tr>
<td>001</td>
<td>03355</td>
</tr>
<tr>
<td>002</td>
<td>06456</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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 ③ 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).

→ 10.2 [8-2] Leading Number—◆ Leading Number
→ [321] ARS Leading Number
→ 10.2 [8-2] Leading Number—◆ Additional Number of Digits
→ 10.2 [8-2] Leading Number—◆ Routing Plan Number
→ [322] ARS Routing Plan Table Number

ARS Leading Number Table

<table>
<thead>
<tr>
<th>Location No.</th>
<th>Leading No.</th>
<th>Additional (Remain) No. of Digits</th>
<th>Routing Plan Table No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>0001</td>
<td>039</td>
<td>7</td>
<td>1</td>
</tr>
<tr>
<td>0002</td>
<td>03</td>
<td>0</td>
<td>4</td>
</tr>
<tr>
<td>0003</td>
<td>0444</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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.
### 4. Routing Plan Table Assignment

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

#### Time Table

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.

- **10.4 [8-3] Routing Plan Time—Time Setting**
- **[330] ARS Routing Plan Time Table**

#### Carrier Priority

Assign the appropriate carrier (refer to "5. Carrier Table Assignment") and their priority in each time block. The carrier is selected in the entry order (the order in which entries are listed).

- **10.5 [8-4] Routing Plan Priority**
- **[331–346] ARS Routing Plan Table (1–16)**

### 3. ARS Leading Number Table

<table>
<thead>
<tr>
<th>Location No.</th>
<th>Leading No.</th>
<th>Additional (Remain) No. of Digits</th>
<th>Routing Plan Table No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>0001</td>
<td>03</td>
<td>8</td>
<td>1</td>
</tr>
</tbody>
</table>

### 4. ARS Routing Plan Table

#### 5. Time Table

<table>
<thead>
<tr>
<th>Time</th>
<th>Carrier</th>
</tr>
</thead>
<tbody>
<tr>
<td>9:00</td>
<td>1 (A telecom) 4 (D telecom)</td>
</tr>
<tr>
<td>12:00</td>
<td>1 (A telecom) 2 (B telecom)</td>
</tr>
<tr>
<td>15:00</td>
<td>1 (A telecom) 2 (B telecom)</td>
</tr>
<tr>
<td>21:00</td>
<td>1 (A telecom) 3 (C telecom)</td>
</tr>
</tbody>
</table>

#### 6. Carrier Priority

<table>
<thead>
<tr>
<th>Time</th>
<th>Priority 1</th>
<th>Priority 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>SUN</td>
<td>1</td>
<td>4</td>
</tr>
<tr>
<td>Time-B</td>
<td>1</td>
<td>2</td>
</tr>
<tr>
<td>Time-C</td>
<td>1</td>
<td>2</td>
</tr>
<tr>
<td>Time-D</td>
<td>1</td>
<td>3</td>
</tr>
<tr>
<td>SAT</td>
<td>3</td>
<td>2</td>
</tr>
<tr>
<td>Time-B</td>
<td>3</td>
<td>1</td>
</tr>
<tr>
<td>Time-C</td>
<td>3</td>
<td>1</td>
</tr>
<tr>
<td>Time-D</td>
<td>3</td>
<td>1</td>
</tr>
</tbody>
</table>
5. **Carrier Table** ① Assignment

A specified number of carriers can be programmed. Assign the following items for each Carrier Table:

> 10.6 [8-5] Carrier—Carrier

- **Carrier Name** ②: Assign the carrier name.
  > 10.6 [8-5] Carrier—Carrier—Carrier Name

- **Removed Number of Digits** ③: Assign the number of digits to remove from the beginning of the user-dialled number.
  > 10.6 [8-5] Carrier—Carrier—Carrier Name

- **Carrier Access Code** ④: Assign the code to access the carrier.
  > 10.6 [8-5] Carrier—Carrier—Carrier Name

- **Trunk Group** ⑤: Assign the trunk groups which connect to each carrier.
  > 10.6 [8-5] Carrier—Carrier—Carrier Name

- **Modify Command** ⑥: Assign the commands to modify the dialled number to access the carrier.
  > 10.6 [8-5] Carrier—Carrier—Carrier Name

### [Command Explanation]

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number</td>
<td>Add the number.</td>
</tr>
<tr>
<td>C</td>
<td>Add the Carrier Access code.</td>
</tr>
<tr>
<td>P</td>
<td>Analogue Line: Insert a pause. ISDN/T1/E1 Line: Insert a pause and change to tone (DTMF) signal.</td>
</tr>
<tr>
<td>A</td>
<td>Add the Authorisation code for a tenant (③).</td>
</tr>
<tr>
<td>G</td>
<td>Add the Authorisation code for a trunk group (⑤).</td>
</tr>
<tr>
<td>I</td>
<td>Add the Itemised Billing code (③).</td>
</tr>
<tr>
<td>H</td>
<td>Add the dialled number after the digits are removed (Home position).</td>
</tr>
</tbody>
</table>
[Programming Example]

<table>
<thead>
<tr>
<th></th>
<th>Carrier Table</th>
<th>1</th>
<th>2</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>A telecom</td>
<td>B telecom</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>Removed Number of Digits</td>
<td>6</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>Carrier Access Code</td>
<td>0077</td>
<td>0088</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>Trunk Group</td>
<td>1, 2, 3</td>
<td>1, 2</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>Modify Command</td>
<td>CH#12</td>
<td>CH</td>
<td></td>
</tr>
</tbody>
</table>

[Example]
Dialled number: 0123456789
(Trunk Access no. is ignored.)
Modification:

1. Remove 6 digits.
2. Add the Carrier Access code (CH).
3. Add the number.

6. Optional Assignment

Authorisation Code for a Tenant
An Authorisation code can be assigned for each carrier and each tenant.
→ 10.6 [8-5] Carrier—Authorisation Code for Tenant

Authorisation Code for a Trunk Group
An Authorisation code can be assigned for each trunk group and each carrier.
→ 10.8 [8-7] Authorisation Code for TRG

Itemised Billing Code
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.
→ 6.10 [4-2-1] Portable Station—Extension Settings—Option 1—ARS Itemised Code
→ 8.3 [6-3] Verification Code—ARS Itemised Code

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. (→ 1.26.1.1 Station Message Detail Recording (SMDR))


- **ARS Data Download/Upload**
  
  It is possible to download or upload the following ARS data to the PBX using PC programming:
  
  - ◆ ARS Leading Number Exception Table
  - ◆ ARS Leading Number Table
  - ◆ ARS Routing Plan Table
    
    → 2.5.9 Tool—Import
    → 2.5.10 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. (→ 1.9.1 Toll Restriction (TRS)/Call Barring (Barring))**

**PC Programming Manual References**

2.5.9 Tool—Import

→ ARS - Leading Digit
→ ARS - Except Code
→ ARS - Routing Plan

2.5.10 Tool—Export


4.18 [2-9] System Options—Option 3—◆ Dial Tone—Dial Tone for ARS


6.10 [4-2-1] Portable Station—Extension Settings—Option 1—◆ ARS Itemised Code

8.3 [6-3] Verification Code—◆ Itemised Billing Code for ARS

Section 10 [8] ARS


**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)
[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**

3.1 Capacity of System Resources
1.11 Conversation Features

1.11.1 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**

4.5 [2-3] Timers & Counters—Dial / IRNA / Recall / Tone—◆ Tone Length—Reorder Tone for PT Hands-free
6.1 [4-1-1] Wired Extension—Extension Settings—Option 4—◆ LCS Answer Mode

**PT Programming Manual References**

None
1.11.2 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-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.

PC Programming Manual References

4.18 [2-9] System Options—Option 1— ◆ PT Operation—Off Hook Monitor for KX-T74xx/T75xx/T76xx

PT Programming Manual References

None

User Manual References

1.4.7 Letting Other People Listen to the Conversation (Off-hook Monitor)
1.11.3 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
1.11.4 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, 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 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 (→ 1.6.1.4 Last Number Redial)
  - Receiving OHCA (→ 1.8.4.3 Off-hook Call Announcement (OHCA))
  - Receiving Whisper OHCA (→ 1.8.4.4 Whisper OHCA)

PC Programming Manual References

3.21 [1-1] Slot—Port Property - Extension Port— Headset OFF/ON
6.5 [4-1-4] Wired Extension—Flexible Button— Type
6.17 [4-3] DSS Console— Type

PT Programming Manual References

None

Feature Guide References

1.20.2 Flexible Buttons

User Manual References

1.3.5 Using the ANSWER/RELEASE Button
1.4.8 Using the Headset (Headset Operation)
3.1.2 Settings on the Programming Mode
1.11.5 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

4.9 [2-6-1] Numbering Plan—Main—Features—◆ Data Line Security Set / Cancel
6.1 [4-1-1] Wired Extension—Extension Settings—Option 2—◆ Data Mode

PT Programming Manual References

None

User Manual References

1.8.10 Protecting Your Line against Notification Tones (Data Line Security)
1.11.6 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. (→ 1.5.5.2 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. (→ 1.11.7 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 (→ 1.26.1.1 Station Message Detail Recording (SMDR)), restarts the call timer, inserts the automatic pause, and checks the TRS/Barring level (→ 1.9.1 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. (→ 1.10.1 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.

PC Programming Manual References

3.25 [1-1] Slot—Port Property - LCO Port— Disconnect Time
3.33 [1-1] Slot—Port Property - T1 Port— Disconnect Time
3.39 [1-1] Slot—Port Property - E1 Port— Disconnect Time
6.1 [4-1-1] Wired Extension—Extension Settings—Option 7— Flash Mode during CO Conversation
6.5 [4-1-4] Wired Extension—Flexible Button— Type
6.10 [4-2-1] Portable Station—Extension Settings—Option 7— Flash Mode during CO Conversation
6.13 [4-2-3] Portable Station—Flexible Button— Type
6.17 [4-3] DSS Console— Type

PT Programming Manual References

[418] LCOT Disconnect Time
Feature Guide References

1.20.2 Flexible Buttons
1.11.7 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 (→ 1.5.4.8 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 (→ 1.11.6 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

3.25 [1-1] Slot—Port Property - LCO Port—Flash Time
3.33 [1-1] Slot—Port Property - T1 Port—Flash Time
3.39 [1-1] Slot—Port Property - E1 Port—Flash Time
4.9 [2-6-1] Numbering Plan—Main—Features—External Feature Access
6.1 [4-1-1] Wired Extension—Extension Settings—Option 7—Flash Mode during CO Conversation
6.5 [4-1-4] Wired Extension—Flexible Button—Type
6.10 [4-2-1] Portable Station—Extension Settings—Option 7—Flash Mode during CO Conversation
6.13 [4-2-3] Portable Station—Flexible Button—Type
6.17 [4-3] DSS Console—Type
9.5 [7-5] Miscellaneous—TRS Check after EFA

PT Programming Manual References

[417] LCOT Flash/Recall Time

Feature Guide References

1.20.2 Flexible Buttons

User Manual References

1.10.4 If a Host PBX is Connected
# 1.11.8 Trunk Call Limitation

## Description

Trunk calls are limited by the following features:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Extension-to-Trunk Call Duration</strong></td>
<td>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. 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.</td>
</tr>
<tr>
<td><strong>Trunk-to-Trunk Call (except Unattended Conference Call) Duration</strong></td>
<td>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. 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.</td>
</tr>
<tr>
<td><strong>Budget Management</strong></td>
<td>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 (→ 1.9.2 Budget Management).</td>
</tr>
<tr>
<td><strong>Dialling Digit Restriction during Conversation</strong></td>
<td>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.</td>
</tr>
</tbody>
</table>

## Conditions

- During an Unattended Conference Call, the Unattended Conference Recall time is applied. (→ 1.14.1.2 Conference)
- When using LCO trunks that do not support Calling Party Control (CPC) signal detection (→ 1.11.10 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. → [473] Trunk-to-Trunk Call Duration
- For SIP Extension users, the line will be disconnected without hearing any warning tones when the trunk call limitation expires.

## PC Programming Manual References

- 4.12 [2-7-1] Class of Service—COS Settings—CO & SMDR—.Extension-CO Line Call Duration Limit
- 4.18 [2-9] System Options—Option 2—.Extension - CO Call Limitation—For Incoming Call
- 5.1 [3-1-1] Trunk Group—TRG Settings—Main →.CO-CO Duration Time
1.11.8 Trunk Call Limitation

→◆ Extension-CO Duration Time

PT Programming Manual References

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

Feature Guide References

2.2.1 Class of Service (COS)
1.11.9 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 paralleled telephones are described below.

<table>
<thead>
<tr>
<th>Features</th>
<th>Descriptions</th>
<th>Connections</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parallel Mode</td>
<td>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.</td>
<td>APT/DPT + SLT</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DPT</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Extn. 101</td>
</tr>
<tr>
<td></td>
<td></td>
<td>APT</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SLT</td>
</tr>
<tr>
<td>EXtra Device Port (XDP) Mode</td>
<td>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. (→ 2.1.1 Extension Port Configuration)</td>
<td>DPT + SLT</td>
</tr>
<tr>
<td></td>
<td></td>
<td>PBX</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DPT</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SLT</td>
</tr>
<tr>
<td>Digital XDP</td>
<td>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 &quot;master DPT&quot;, and the DPT connected to the master DPT is called the &quot;slave DPT&quot;. 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.</td>
<td>DPT + DPT</td>
</tr>
<tr>
<td></td>
<td></td>
<td>PBX</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DPT</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Extn. 101</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DPT</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SLT</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Extn. 101 (in Parallel Mode) or Extn. 105 (in XDP Mode)</td>
</tr>
</tbody>
</table>
## 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 (→ 1.4.1.4 Hands-free Answerback) or voice-calling mode (Alternate Receiving Ring/Voice) (→ 1.5.3 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:
  - While being monitored by another extension. (→ 1.8.3 Call Monitor)
  - While receiving OHCA (→ 1.8.4.3 Off-hook Call Announcement (OHCA)) or Whisper OHCA. (→ 1.8.4.4 Whisper OHCA)
  - During a Conference call (→ 1.14.1 Conference Features).
  - While Live Call Screening (LCS) or Two-way Record is activated (→ 1.24.2.3 Voice Mail DPT (Digital) Integration).
[DPT + DPT]

- **Capable Telephones**
  KX-DT300 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**
1.4.3 System Capacity
3.8.2 Parallel Connection of the Extensions
3.8.3 Digital EXtra Device Port (Digital XDP) Connection

**PC Programming Manual References**
3.21 [1-1] Slot—Port Property - Extension Port
   → ◆ XDP Mode
   → ◆ Parallel Telephone Ringing
4.9 [2-6-1] Numbering Plan—Main—Features—◆ Parallel Telephone (Ring) Mode Set / Cancel

**PT Programming Manual References**
[600] EXtra Device Port (XDP) Mode

**User Manual References**
1.8.12 Setting the Parallelled Telephone to Ring (Parallelled Telephone)
1.11.10 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 (→ 1.14.1.2 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 (→ 1.17.6 Direct Inward System Access (DISA)) and an extension or an outside party, the line is disconnected.

PC Programming Manual References
3.25 [1-1] Slot—Port Property - LCO Port—◆ CPC Signal Detection Time—Outgoing, Incoming
3.33 [1-1] Slot—Port Property - T1 Port
   →◆ CPC Signal Detection (DID)—Outgoing, Incoming
   →◆ CPC Signal Detection (LCO/GCO)—Outgoing, Incoming
3.39 [1-1] Slot—Port Property - E1 Port—◆ CPC Detection Time—Out, In

PT Programming Manual References
[413] LCOT CPC Signal Detection Time—Outgoing
[414] LCOT CPC Signal Detection Time—Incoming
1.12 Transferring Features

1.12.1 Call Transfer

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

<table>
<thead>
<tr>
<th>Feature</th>
<th>Transferring method</th>
</tr>
</thead>
<tbody>
<tr>
<td>With Announcement</td>
<td>Transfer is completed after announcing the destination party.</td>
</tr>
<tr>
<td>Without Announcement</td>
<td>Transfer is completed without an announcement. After dialling the destination, while hearing a ringback tone, the originator can replace the handset.</td>
</tr>
</tbody>
</table>

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.

A call is transferred without announcement.

Does the transferrer have a Transfer Recall destination assigned?
Yes

Is the first transfer destination an extension?
No

Is the intercept destination an extension, ICD group, or VM group?
No

Is the recall destination an extension, ICD group, or VM group?
Yes

The set extension, ICD group, or VM group is memorised as the Transfer Recall destination.

No

The transferrer is memorised as the Transfer Recall destination.

No No

Yes

The set extension, ICD group, or VM group is memorised as the Transfer Recall destination.

[Available destination]

<table>
<thead>
<tr>
<th>Destination</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/SIP Extension/ISDN Extension/T1-OPX)</td>
<td>✓</td>
</tr>
<tr>
<td>PS</td>
<td>✓</td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td>✓</td>
</tr>
<tr>
<td>PS Ring Group</td>
<td>✓</td>
</tr>
<tr>
<td>Floating Extension no. for SVM</td>
<td>✓ (DPT only)*</td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td>✓ (DPT only)*</td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td>✓</td>
</tr>
</tbody>
</table>
### 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. (→ 1.13.4 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. (→ 1.3.1.2 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) (→ 1.30.1 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. (→ 1.6.1 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 (→ 1.7.1 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.

- 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.

### PC Programming Manual References

- 4.4 [2-2] Operator & BGM— BGM and Music on Hold—Sound on Transfer
- 4.12 [2-7-1] Class of Service—COS Settings—CO & SMDR— Transfer to CO
- 4.18 [2-9] System Options—Option 4
  - DSS Key—Automatic Transfer for Extension Call
  - Transfer—Transfer to busy Extension without BSS Operation
- 6.1 [4-1-1] Wired Extension—Extension Settings—Option 1— Transfer Recall Destination
1.12.1 Call Transfer

6.10  [4-2-1] 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
2.2.1 Class of Service (COS)

User Manual References
1.4.1 Transferring a Call (Call Transfer)
1.13 Holding Features

1.13.1 Call Hold

Description

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

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Regular Hold</td>
<td>Any extension can retrieve a held call.</td>
</tr>
<tr>
<td>Exclusive Call Hold</td>
<td>Only the extension user who held the call can retrieve it.</td>
</tr>
</tbody>
</table>

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. ([1.13.2 Call Park]

- **Music on Hold**
  Music, if available, is sent to the held party. ([1.13.4 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 ([1.1.2.2 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:

<table>
<thead>
<tr>
<th>Mode 1</th>
<th>Hold</th>
<th>Hold (to be Retrieved from Another Extension)*</th>
<th>Transfer to Trunk</th>
<th>Transfer to Extension</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Flashing the hookswitch</td>
<td>Flashing the hookswitch + Hold Feature No.</td>
<td>Flashing the hookswitch</td>
<td>Flashing the hookswitch</td>
</tr>
<tr>
<td></td>
<td>+ Going on-hook</td>
<td>+ Going on-hook</td>
<td>+ Trunk Access No.</td>
<td>+ Extension No.</td>
</tr>
</tbody>
</table>
### 1.13.1 Call Hold

<table>
<thead>
<tr>
<th>Mode</th>
<th>Hold (to be Retrieved from Another Extension)</th>
<th>Transfer to Trunk</th>
<th>Transfer to Extension</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mode 2 (Default)</td>
<td>Flashing the hookswitch + Hold Feature No. + Going on-hook</td>
<td>Flashing the hookswitch + Hold Feature No. + Going on-hook</td>
<td>Flashing the hookswitch + Trunk Access No.</td>
</tr>
<tr>
<td>Mode 3</td>
<td>Flashing the hookswitch + Hold Feature No. + Going on-hook</td>
<td>Flashing the hookswitch + Hold Feature No. + Hold Feature No. + Going on-hook</td>
<td>Flashing the hookswitch + Extension No.</td>
</tr>
<tr>
<td>Mode 4</td>
<td>Flashing the hookswitch + Hold Feature No. + Going on-hook</td>
<td>Flashing the hookswitch + Hold Feature No. + Hold Feature No. + Going on-hook</td>
<td>Flashing the hookswitch + Extension No.</td>
</tr>
</tbody>
</table>

* 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. (→ 3.2.1 Tones/Ring Tones)

### PC Programming Manual References

→ ◆ Recall—Hold Recall  
→ ◆ Recall—Disconnect after Recall

4.9 [2-6-1] 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

4.18 [2-9] 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
1.13.2 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:

<table>
<thead>
<tr>
<th>Light pattern</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Slow red flashing</td>
<td>Parked in the preset parking zone</td>
</tr>
<tr>
<td>Off</td>
<td>No parked call</td>
</tr>
</tbody>
</table>

- **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.

- **Call Park Retrieve Deny**
  If an extension user cannot call certain extensions on a COS basis (→ 1.1.2.2 Internal Call Block), he cannot retrieve the parked call which the extensions made.

PC Programming Manual References

4.5 [2-3] Timers & Counters
  → ◆ Recall—Call Park Recall
  → ◆ Recall—Disconnect after Recall

4.9 [2-6-1] Numbering Plan—Main—Features—◆ Call Park / Call Park Retrieve

6.5 [4-1-4] Wired Extension—Flexible Button
  → ◆ Type
  → ◆ Parameter Selection (for Call Park)
  → ◆ Optional Parameter (Ringing Tone Type Number) (for Call Park)

6.13 [4-2-3] Portable Station—Flexible Button
  → ◆ Type
  → ◆ Parameter Selection (for Call Park)
  → ◆ Optional Parameter (or Ringing Tone Type Number) (for Call Park)
6.17  [4-3] DSS Console  
   →◆ Type  
   →◆ Parameter Selection (for Call Park)  
   →◆ Optional Parameter (Ringing Tone Type Number) (for Call Park)  

PT Programming Manual References  
None  

Feature Guide References  
  1.20.2  Flexible Buttons  
  3.1  Capacity of System Resources  

User Manual References  
  1.4.2  Holding a Call
1.13.3 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)
1.13.4 Music on Hold

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

a. Internal music source
b. External music source
c. Cyclic tone

Music Source Assignment
One external music port is available. If BGM is selected, it is possible to select whether the external music port, internal music port 1, or internal music port 2 is assigned to BGM (→ 1.17.4 Background Music (BGM)).

Conditions

- **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.
- Each tenant can select a music source to use for Music on Hold. The type of call determines which tenant’s music source is used, as follows:

<table>
<thead>
<tr>
<th>Type</th>
<th>Music Source</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Intercom Calls/Outgoing Calls</td>
<td>Selected based on the tenant setting to which the extension user belongs.</td>
</tr>
<tr>
<td>Incoming Trunk Calls</td>
<td>Selected based on the tenant setting of the distribution method (DIL/DID/DDI/MSN).</td>
</tr>
</tbody>
</table>

- When using an IP-PT connected to a V-IPEXT card, only cyclic tone can be selected as Music on Hold. The cyclic tone pattern is shown below:

```
1 s
```

Installation Manual References
3.13.1 Connection of Peripherals

PC Programming Manual References

4.4 [2-2] Operator & BGM
  → ◆ BGM and Music on Hold—Music Source of BGM
  → ◆ BGM and Music on Hold—Music on Hold


PT Programming Manual References

[710] Music Source Selection for BGM
1.13.4 Music on Hold

[711] Music on Hold

Feature Guide References

2.2.3 Tenant Service
1.14 Conference Features

1.14.1 Conference Features

1.14.1.1 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:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference</td>
<td>During a two-party conversation, an extension user can add other parties to establish a conference call with up to eight parties.</td>
</tr>
<tr>
<td></td>
<td>→ 1.14.1.2 Conference</td>
</tr>
<tr>
<td>Executive Busy Override</td>
<td>An extension user can interrupt an existing call to establish a three-party conference call.</td>
</tr>
<tr>
<td></td>
<td>→ 1.8.2 Executive Busy Override</td>
</tr>
<tr>
<td>Privacy Release</td>
<td>During a conversation with an outside party on the S-CO button, a PT/PS user can allow another extension to join the conversation.</td>
</tr>
<tr>
<td></td>
<td>→ 1.14.1.3 Privacy Release</td>
</tr>
</tbody>
</table>

Conditions

- This PBX supports a maximum of 32 parties being engaged in simultaneous conference calls. (e.g., 4 eight-party conferences, 8 three-party conferences + 2 four-party conferences, 10 three-party conferences)
1.14.1.2 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.
- For a Conference With Six or More Parties
  The ECHO card is required. Also the echo cancelling function should be enabled through system programming.

Installation Manual References

3.7.5 ECHO16 Card (KX-TDA0166)

PC Programming Manual References

  → Unattended Conference—Recall Start Timer
  → Unattended Conference—Warning Tone Start Timer
  → Unattended Conference—Disconnect Timer
4.12 [2-7-1] Class of Service—COS Settings—CO & SMDR— Transfer to CO
4.18 [2-9] System Options—Option 3
  → Confirmation Tone—Tone 4-1 : Start Conference
  → Confirmation Tone—Tone 4-2 : Finish Conference
  → Echo Cancel—Conference
6.5 [4-1-4] Wired Extension—Flexible Button—◆ Type
6.13 [4-2-3] Portable Station—Flexible Button—◆ Type
6.17 [4-3] DSS Console—◆ Type

PT Programming Manual References
None

Feature Guide References
1.20.2 Flexible Buttons
3.1 Capacity of System Resources

User Manual References
1.4.5 Multiple Party Conversation
1.14.1.3 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 (→ 1.11.5 Data Line Security) and Executive Busy Override Deny (→ 1.8.2 Executive Busy Override).

PC Programming Manual References

4.18 [2-9] System Options—Option 1—◆PT Operation—Privacy Release by SCO key
6.5 [4-1-4] Wired Extension—Flexible Button—◆Type
6.13 [4-2-3] Portable Station—Flexible Button—◆Type
6.17 [4-3] DSS Console—◆Type

PT Programming Manual References

None

Feature Guide References

1.20.2 Flexible Buttons

User Manual References

1.4.5 Multiple Party Conversation
1.15 Paging Features

1.15.1 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 an external speaker (external pager) which belong to the paging group. 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 an external pager. One extension user group or external pager can belong to several paging groups. (→ 2.2.2 Group)

[Example]

[Programming Example]

<table>
<thead>
<tr>
<th>Paging Group No.</th>
<th>Extension User Group No.</th>
<th>External Pager</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>001 002 003 ...</td>
<td></td>
</tr>
<tr>
<td>01</td>
<td>✓               ...</td>
<td></td>
</tr>
<tr>
<td>02</td>
<td>✓ ✓  ✓ ...</td>
<td></td>
</tr>
<tr>
<td>03</td>
<td>✓ ✓ ... ✓</td>
<td>✓</td>
</tr>
<tr>
<td>04</td>
<td>✓ ... ...</td>
<td></td>
</tr>
<tr>
<td>05</td>
<td>✓ ✓ ✓ ... ✓</td>
<td>✓</td>
</tr>
<tr>
<td></td>
<td>✓ ✓ ✓ ✓ ... ✓</td>
<td></td>
</tr>
</tbody>
</table>

✓: Constituent

1: → 5.10 [3-4] Paging Group
or
5.11 [3-4] Paging Group—All Setting
→ [640] Extension User Groups of a Paging Group
2: → 5.12 [3-4] Paging Group—External Pager
→ [641] External Pagers of a Paging Group
Conditions

- Extensions which cannot be paged are:
  - PSs
  - SLTs
  - Ringing or busy PTs
  - PTs in Paging Deny mode
  - PTs in Paging DND mode
  - IP-PTs connected to a V-IPEXT card, that are assigned to extension user group 31 (default)\(^1\)

  These extensions cannot be paged but can answer the page.

\(^1\) 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 IPCMPR card.

- **External Pager Priority**
  An external pager can be used with the following priorities:
  TAFAS → Paging → BGM
  (→ 1.17.3 Trunk Answer From Any Station (TAFAS), 1.17.4 Background Music (BGM))

- **Volume Control**
  Paging volume from the PTs and the external pager can be changed through system programming.

- **Paging DND**
  When DND (→ 1.3.1.3 Do Not Disturb (DND)) is set for incoming calls, it is programmable whether your extension receives paging through system programming.

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features
  → ◆ Group Paging
  → ◆ Group Paging Answer
  → ◆ Paging Deny Set / Cancel

4.18 [2-9] System Options
  → Option 1 — ◆ PT Fwd / DND—Paging to DND Extension
  → Option 3 — ◆ Confirmation Tone—Tone 2 : Paged / Automatic Answer

4.20 [2-11-1] Audio Gain—Paging/MOH
  → ◆ Paging—EPG (External Pager)
  → ◆ Paging—Paging Level from PT Speaker

5.10 [3-4] Paging Group
5.11 [3-4] Paging Group—All Setting
5.12 [3-4] Paging Group—External Pager

PT Programming Manual References

[640] Extension User Groups of a Paging Group
[641] External Pagers of a Paging Group

Feature Guide References

3.1 Capacity of System Resources

User Manual References

1.7.1 Paging
1.7.2 Answering/Denying a Paging Announcement
1.16 Broadcasting Features

1.16.1 Broadcasting

Description
A PT user can call multiple parties, assigned as members of a group (broadcasting group), to make a voice announcement. After one member answers the call, the caller can begin to make an announcement. Other members' telephones will continue to ring, and each member can listen to the announcement by answering the call. The caller will hear a confirmation tone every time the members answer the call. During the announcement, the voices of members will not be heard. However, the caller can allow specific members to speak, making a conference call. A maximum of four parties can participate in the conference call simultaneously. This conversation can be heard by the other members.

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

<table>
<thead>
<tr>
<th>Destination</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/SIP Extension/ISDN Extension/T1-OPX)</td>
<td>✓</td>
</tr>
<tr>
<td>PS</td>
<td>✓</td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td></td>
</tr>
<tr>
<td>PS Ring Group</td>
<td></td>
</tr>
<tr>
<td>Floating Extension no. for SVM</td>
<td></td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td></td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td></td>
</tr>
<tr>
<td>DISA</td>
<td></td>
</tr>
<tr>
<td>Analogue/ISDN Remote Maintenance</td>
<td></td>
</tr>
<tr>
<td>Idle Line Access no. + Phone no.</td>
<td></td>
</tr>
<tr>
<td>Trunk Group Access no. + Trunk Group no. + Phone no.</td>
<td>✓*1</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with no PBX Code)</td>
<td>✓*1</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with PBX Code)</td>
<td>✓*1</td>
</tr>
</tbody>
</table>

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

Conference Call
The caller can allow members to speak, and can control the conversation using the following buttons. Pressing the other buttons during the conversation will be ignored.

<table>
<thead>
<tr>
<th>Button</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSS</td>
<td>Establishes a conversation with the corresponding member extension, or remove the extension from the conversation.</td>
</tr>
</tbody>
</table>
### 1.16.1 Broadcasting

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

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.

### Conditions
- Only PTs that are permitted by COS programming can originate broadcasting calls.
- **Ring Duration**
  A system timer is available to limit the call ring duration at the members. When this limit is reached, members’ telephones will stop ringing. If no members have answered the call, the caller will hear a busy tone.
- The broadcasting 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 broadcasting call is made, a call waiting tone will be sent to the extension.
- The call information of the caller (not members) will be recorded on SMDR.
- A caller cannot make a broadcasting call with a call on hold.
- Call Pickup is not available for a broadcasting call. (→ 1.4.1.3 Call Pickup)
- The broadcasting 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. (→ 1.25.5 Wireless XDP Parallel Mode)

### PC Programming Manual References
- 4.5 [2-3] Timers & Counters—Miscellaneous—Broadcasting—Ring Duration
- 4.9 [2-6-1] Numbering Plan—Main—Features—Broadcasting Operation
- 4.12 [2-7-1] Class of Service—COS Settings—Miscellaneous—Broadcasting Operation
- 5.27 [3-10] Broadcasting Group
- 5.28 [3-10] Broadcasting Group—Member List

### PT Programming Manual References
- None

### Feature Guide References
- 3.1 Capacity of System Resources
User Manual References

1.7.3 Making an Announcement and Having a Multiple Party Conversation (Broadcasting)
1.17 Optional Device Features

1.17.1 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, the OPB card and DPH card
- Each doorphone port can only be assigned to one tenant. The Time Table (day/lunch/break/night) of the tenant applies. (→ 2.2.4 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. (→ 1.1.2.1 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. (→ 1.1.2.2 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. (→ 1.17.2 Door Open)
- A doorphone number can be referenced for each doorphone port.

Installation Manual References
3.7.1 OPB3 Card (KX-NCP1190)
3.7.2 DPH4 Card (KX-TDA0161)
3.7.3 DPH2 Card (KX-TDA0162)
3.12.1 Connection of Doorphones, Door Openers, External Sensors, and External Relays

PC Programming Manual References
  → ◆ Doorphone—Call Ring Duration
  → ◆ Doorphone—Call Duration
4.9 [2-6-1] Numbering Plan—Main—Features—◆ Doorphone Call
4.12 [2-7-1] Class of Service—COS Settings—TRS—◆ TRS Level—Day, Lunch, Break, Night
4.16 [2-8-2] Ring Tone Patterns—Call from Doorphone
4.18 [2-9] System Options—Option 3—◆ Confirmation Tone—Tone 1 : Called by Voice
7.1 [5-1] Doorphone
PT Programming Manual References

[720] Doorphone Call Destination
[729] Doorphone Number Reference

User Manual References

1.10.1 If a Doorphone/Door Opener is Connected
1.17.2 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 (→ 1.17.1 Doorphone Call).

Conditions
- Hardware Requirement: A user-supplied door opener on each door.
- 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.

Installation Manual References
3.12.1 Connection of Doorphones, Door Openers, External Sensors, and External Relays

PC Programming Manual References
3.41 [1-1] Slot—OPB3 Card Property—Device Type
4.9 [2-6-1] Numbering Plan—Main—Features—Door Open
4.12 [2-7-1] 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
2.2.1 Class of Service (COS)

User Manual References
1.10.1 If a Doorphone/Door Opener is Connected
1.17.3 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). 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.

Installation Manual References
3.13.1 Connection of Peripherals

PC Programming Manual References
4.9 [2-6-1] Numbering Plan—Main—Features—◆ TAFAS Answer
7.2 [5-2] External Pager

PT Programming Manual References
[700] External Pager Floating Extension Number

User Manual References
1.3.4 Answering a Call via an External Speaker (Trunk Answer From Any Station [TAFAS])
1.17.4 Background Music (BGM)

Description

A PT user can listen to BGM through the built-in speaker while on-hook and idle.

BGM—External:
BGM can also be broadcast in the office through the external pager, this can be turned on and off by an extension assigned as the manager.

Conditions

[BGM]
- **Hardware requirement**: A user-supplied external music source, such as a radio
- 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 connected to a V-IPEXT card that can simultaneously perform the BGM feature (→ 1.17.4 Background Music (BGM)). Changing this setting may affect the number of simultaneous IP extension and IP trunk calls available on the IPCMPR card.

[BGM—External]
- **Hardware requirement**: A user-supplied external pager
- An external pager can be used with the following priorities:
  - TAFAS → Paging → BGM
  (→ 1.15.1  Paging, 1.17.3 Trunk Answer From Any Station (TAFAS))

Installation Manual References

- 3.13.1 Connection of Peripherals

PC Programming Manual References

- 3.4 [1-1] Slot—Card Property - IPCMPR—VoIP-DSP Option—◆ IP Extension Count of BGM
- 4.9 [2-6-1] Numbering Plan—Main—Features
  - → ◆ External BGM On / Off
  - → ◆ BGM Set / Cancel
- 7.2 [5-2] External Pager

PT Programming Manual References

None

User Manual References

- 1.8.9 Turning on the Background Music (BGM)
- 2.1.4 Turning on the External Background Music (BGM)
1.17.5 Outgoing Message (OGM)

Description

An extension assigned as the manager (manager extension) can record outgoing messages (OGMs) for the following features:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Usage &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Direct Inward System Access (DISA)</td>
<td>When a call arrives on a DISA line, the caller will hear a message.</td>
</tr>
<tr>
<td></td>
<td>→ 1.17.6 Direct Inward System Access (DISA)</td>
</tr>
<tr>
<td>Queuing Feature</td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>→ 1.2.2.4 Queuing Feature</td>
</tr>
<tr>
<td>Timed Reminder</td>
<td>When answering the Timed Reminder Alarm, the user will hear a message.</td>
</tr>
<tr>
<td></td>
<td>→ 1.28.4 Timed Reminder</td>
</tr>
</tbody>
</table>

Conditions

- This feature will function using the preinstalled IPCMPR card. To increase the capacity of the feature it is possible to add an MSG4 card, ESVM2 card or ESVM4 card mounted on an OPB3 card.
- **Number of Messages and Recording Time**
  A maximum of 64 messages can be recorded.
  - A floating extension number is assigned to each outgoing message (OGM) (default: 5 + two-digit OGM number).
  - The SVM/OGM Block on the IPCMPR card or one ESVM2 card allows two messages to play simultaneously, and one MSG4 or ESVM4 card allows four messages to play simultaneously. Increasing the number of cards does not increase the total recording time or the number of messages; it increases the number of messages that can be played simultaneously.

**[Example]**

![Diagram showing message recording time and capacity with different card configurations.]

The maximum recording time for each card depends on the recording quality and the type of card used, as follows:

<table>
<thead>
<tr>
<th>Recording Quality</th>
<th>Recording Time with the IPCMPR/ESVM2/ESVM4 Card</th>
<th>Recording Time with the MSG4 Card</th>
</tr>
</thead>
<tbody>
<tr>
<td>High (default)</td>
<td>about 20 minutes</td>
<td>about 8 minutes</td>
</tr>
</tbody>
</table>
### Recording Quality

<table>
<thead>
<tr>
<th>Recording Quality</th>
<th>Recording Time with the IPCMPR/ESVM2/ESVM4 Card</th>
<th>Recording Time with the MSG4 Card</th>
</tr>
</thead>
<tbody>
<tr>
<td>Middle</td>
<td>about 60 minutes</td>
<td>about 8 minutes</td>
</tr>
<tr>
<td>Low</td>
<td>about 120 minutes</td>
<td>about 8 minutes</td>
</tr>
</tbody>
</table>

When cards of different types are installed, the shortest recording time is used. For example, if an ESVM4 card and an MSG4 card are installed, the maximum recording time at default will be about 8 minutes.

**Note**

When MSG/ESVM card expansion has been completed, the new card is blank—all necessary messages will need to be stored in the card. This can be done in two ways:

a. By installing the new MSG/ESVM card then recording each message again. The PBX will automatically store the new recordings in each card.

b. By transferring messages from the PBX to a PC using the Maintenance Console, then transferring them back to the PBX with the new MSG/ESVM card installed.

- 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.

**Note**

Record voice messages only; avoid the recording of music. However, it is possible to record music when using the IPCMPR/ESVM card in High Recording Quality mode.

- 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 ports of the card(s) become idle, he will hear the progress tone for a preprogrammed time period. After that, the PBX will automatically proceed into the recording mode.

### Installation Manual References

- 3.3.1 IPCMPR Card
- 3.7.1 OPB3 Card (KX-NCP1190)
- 3.7.6 MSG4 Card (KX-TDA0191)
- 3.7.7 ESVM2 Card (KX-TDA0192) and ESVM4 Card (KX-TDA0194)

### PC Programming Manual References

- 2.6.6 Utility—Message File Transfer PC to PBX
- 2.6.7 Utility—Message File Transfer PBX to PC
- 3.5 [1-1] Slot—OPB3 Card Property—MSG Feature
- 3.41 [1-1] Slot—OPB3 Card Property—MSG Feature
- 4.9 [2-6-1] Numbering Plan—Main—Features—OGM Record / Clear / Playback
5.15 [3-5-2] Incoming Call Distribution Group—Queuing Time Table—◆ Queuing Sequence—Sequence 01–16
7.3 [5-3-1] Voice Message—DISA System—Option 2—◆ Recording Mode
7.4 [5-3-2] 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
1.24.1 Built-in Simplified Voice Message (SVM)
3.1 Capacity of System Resources

User Manual References
2.1.5 Recording Outgoing Messages (OGM)
### 1.17.6 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., modem for remote system administration, 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.


→ 7.3 [5-3-1] 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.

→ 7.4 [5-3-2] Voice Message—DISA Message—◆ 1 Digit AA Destination (Extension Number)—Dial 0–9


#### 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). (→ 1.17.5 Outgoing Message (OGM))

### Programming Example

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>01</td>
<td>501</td>
<td>100 301 200 103 202 101 102 400 104 205</td>
<td>04</td>
</tr>
<tr>
<td>02</td>
<td>502</td>
<td></td>
<td>05</td>
</tr>
<tr>
<td>:</td>
<td>:</td>
<td>:</td>
<td>:</td>
</tr>
</tbody>
</table>

*1 → 7.4 [5-3-2] Voice Message—DISA Message—◆ Floating Extension Number

*2 → 7.4 [5-3-2] Voice Message—DISA Message—◆ 1 Digit AA Destination (Extension Number)—Dial 0–9

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.

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

<table>
<thead>
<tr>
<th>Security Mode</th>
<th>Intercom</th>
<th>TIE Line Call</th>
<th>Trunk Call</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Without PBX Code</td>
<td>With PBX Code</td>
</tr>
<tr>
<td>All Security</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Trunk Security</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>No Security</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
</tbody>
</table>

✓: Available

*: If trunk call is available, Account Code Entry (→ 1.5.4.3 Account Code Entry) is also available.

Note

DISA AA service and Operator Call (→ 2.2.5 Operator Features) are available for any security mode.

Security Mode Override by Verification Code Entry

If the caller performs Verification Code Entry (→ 1.9.6 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.

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.

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.
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 (→ 1.9.5 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 (→ 1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND))
- Incoming Call Distribution Group Log-in/Log-out (→ 1.2.2.7 Log-in/Log-out)
- Absent Message setting (→ 1.19.2 Absent Message)
- Extension Dial Lock (→ 1.9.3 Extension Dial Lock)
- Time Service Switching Mode (→ 2.2.4 Time Service)

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]

<table>
<thead>
<tr>
<th>Trunk No.</th>
<th>CLI</th>
<th></th>
<th>Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Day</td>
<td>Lunch</td>
<td>…</td>
</tr>
<tr>
<td>01</td>
<td>Disable</td>
<td>Disable</td>
<td>…</td>
</tr>
<tr>
<td>:</td>
<td>:</td>
<td>:</td>
<td>:</td>
</tr>
</tbody>
</table>

"CLI" must be set to Disable, to allow incoming calls to be received by DISA.

[Programming Example of System Speed Dialling Table]

<table>
<thead>
<tr>
<th>Location</th>
<th>Name</th>
<th>Trunk Access + Telephone Number</th>
<th>CLI Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td>000</td>
<td>J. Smith</td>
<td>912341115678</td>
<td>200</td>
</tr>
<tr>
<td>001</td>
<td>:</td>
<td>:</td>
<td>:</td>
</tr>
<tr>
<td>:</td>
<td>:</td>
<td>:</td>
<td>:</td>
</tr>
</tbody>
</table>

In this example, calls received on trunk 01 are routed to the DISA OGM with floating extension number 501. 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.
The call information for DISA is recorded as the one of the DISA floating extension numbers. (→ 1.26.1.1 Station Message Detail Recording (SMDR))

[Flowchart]

A DISA call from an outside party is received.

Is there a port available?

Yes

The PBX answers the call.

(DIRA Mute & OGM Start Time after Answering expires)

The OGM plays and the PBX starts to receive the DTMF signalling.

No

The call is routed to an operator.

(DIRA Delayed Answer time expires)

Is there a port available?

Yes

The PBX receives the dialled digits and checks the dialled number.

What is the DISA security mode?

Continued on next page

No

Is a second digit dialled?

Yes

The call is routed to an operator.

(DIRA First Digit Time When No Dial expires)

Operator

No

What method is assigned for DISA Intercept Routing—No Dial?

Operator Disable

AA-0, AA-9

The call is disconnected.

The call is redirected to the destination assigned to AA-0 or AA-9.

No

Is the first digit dialled?

Yes

The OGM stops.

To C

Is a second digit dialled?

Yes

To C

No

Is the first dialled digit assigned a destination for the DISA AA service?

Yes

The call is routed to the destination.

No

What is the DISA security mode?
1.17.6 Direct Inward System Access (DISA)

Continued from previous page

No Security

Trunk Security

All Security

Is the correct PIN entered?

Does the caller press while hearing the reorder tone (Call Retry)?

Is the Walking COS/Verification Code Entry feature number dialed for Security Mode Override?

Yes

No

Yes

No

Is the dialed number an extension number or floating extension number?

Yes

To C

No

Is the dialed number an extension number or floating extension number?

To B

What is the dialed number?

Extension No./Floating Extension No.

Trunk Access No. + Telephone No.

Feature No.* (Absent Message, FWD, etc.)

The feature is set.

To D

The dialed number is sent to the trunk.

Is the extension in DND mode?

Yes

No

Is the extension busy?

Yes

To A

The call is disconnected.

No

The call is directed to the extension. The caller hears a ringback tone.

Does the caller press * while hearing a ringback tone (Call Retry)?

Yes

To A

(DISA Intercept time expires)

No

Does the destination answer the call?

Yes

To A

The call is established.

No

Does the destination answer the call?

Yes

To E

(DISA Disconnect Time after Intercept expires)

The call is established.

Continued on next page

The call is disconnected.

Continued on next page

* Feature numbers are available only when the Walking COS feature is used.
1.17.6 Direct Inward System Access (DISA)

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.

- This feature will function using the preinstalled IPCMR card. To increase the capacity of the feature it is possible to add an MSG4 card, ESVM2 card or ESVM4 card mounted on an OPB3 card.

- 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).

→ [209] DISA Delayed Answer Time

- Call Retry

While hearing a ringback, reorder, or busy tone, retrying the call is possible by pressing "*". System programming selects whether pressing "x" 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.


- **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
    → 5.1 [3-1-1] Trunk Group—TRG Settings—Tone Detection— ◆ DISA Tone Detection—Silence
    → [475] DISA Silence Detection
  - Continuous Signal Detection
    → 5.1 [3-1-1] Trunk Group—TRG Settings—Tone Detection— ◆ DISA Tone Detection—Continuous
    → [476] DISA Continuous Signal Detection
  - Cyclic Signal Detection
    → 5.1 [3-1-1] Trunk Group—TRG Settings—Tone Detection— ◆ DISA Tone Detection—Cyclic
    → [477] DISA Cyclic Signal Detection

- **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. (→ 1.11.8 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.


  → ◆ DISA—CO-to-CO Call Prolong Counter
  → ◆ DISA—CO-to-CO Call Prolong Time

  → [210] DISA Trunk-to-Trunk Call Prolong Time

- **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 at the PBX by pressing "+extension number", if DISA is connected by the Automatic DISA Activation feature. This feature can be enabled or disabled through system programming.

  - The KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is required to use this feature.
  - DISA security mode should be set to No Security or Trunk 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.
  - End of Call Detection does not operate after dialling "+".
  - Calls can only be transferred to extensions within the PBX.
  - Transfer Recall, Call Splitting and Call Transfer with Announcement are not available.
• **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.

• **Call Deny**
  Extensions can deny DISA calls on a COS basis.
  → [4.12] [2-7-1] 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.

• **DISA Automatic Walking COS**
  The KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is required to use this feature.
  → Each outgoing message (OGM) can be assigned a name through system programming for programming reference.
  → [7.4] [5-3-2] Voice Message—DISA Message—◆ Name
  → [731] Outgoing Message (OGM) Name

**Installation Manual References**

- 3.7.1 OPB3 Card (KX-NCP1190)
- 3.7.6 MSG4 Card (KX-TDA0191)
- 3.7.7 ESVM2 Card (KX-TDA0192) and ESVM4 Card (KX-TDA0194)

**PC Programming Manual References**

  → ◆ DISA—Delayed Answer Timer
  → ◆ DISA—Mute & OGM Start Timer after answering
  → ◆ DISA—No Dial Intercept Timer
  → ◆ DISA—2nd Dial Timer for AA
  → ◆ DISA—Intercept Timer—Day, Lunch, Break, Night
  → ◆ DISA—Disconnect Timer after Intercept
  → ◆ DISA—CO-to-CO Call Prolong Counter
  → ◆ DISA—CO-to-CO Call Prolong Time
  → ◆ DISA—Progress Tone Continuation Time before Recording Message
  → ◆ DISA—Reorder Tone Duration


4.12 [2-7-1] Class of Service—COS Settings—Optional Device & Other Extensions—◆ Accept the Call from DISA

5.1 [3-1-1] Trunk Group—TRG Settings
  → Tone Detection—◆ DISA Tone Detection—Silence
  → Tone Detection—◆ DISA Tone Detection—Continuous
  → Tone Detection—◆ DISA Tone Detection—Cyclic

6.10 [4-2-1] Portable Station—Extension Settings—Intercept Destination—◆ Intercept Destination—When called party does not answer—Day, Lunch, Break, Night
7.3 [5-3-1] Voice Message—DISA System
7.4 [5-3-2] 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

1.1.1.6 Intercept Routing
1.1.1.7 Intercept Routing—No Destination
1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)
2.2.1 Class of Service (COS)

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)
1.17.7 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 (→ 1.17.5 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]

<table>
<thead>
<tr>
<th>Destination</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/ISDN Extension/T1-OPX)</td>
<td>✓</td>
</tr>
<tr>
<td>PS</td>
<td>✓*1</td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td>✓</td>
</tr>
<tr>
<td>SIP Extension</td>
<td>✓</td>
</tr>
<tr>
<td>PS Ring Group</td>
<td>✓</td>
</tr>
<tr>
<td>Floating Extension no. for SVM</td>
<td>✓</td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td>✓</td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td>✓</td>
</tr>
<tr>
<td>DISA</td>
<td>✓</td>
</tr>
<tr>
<td>Analogue/ISDN Remote Maintenance</td>
<td>✓</td>
</tr>
<tr>
<td>Idle Line Access no. + Phone no.</td>
<td>✓</td>
</tr>
<tr>
<td>Trunk Group Access no. + Trunk Group no. + Phone no.</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with no PBX Code)</td>
<td>✓</td>
</tr>
<tr>
<td>Other PBX Extension (TIE with PBX Code)</td>
<td>✓</td>
</tr>
</tbody>
</table>

*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. (→ 1.25.6 Virtual PS)

Conditions

- **Hardware Requirement:**
  - An OPB card, and ESVM2 card or ESVM4 card
- When using this feature, it is necessary to disable the **MSG Feature** setting of the IPCMPR card.
- 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.
- It is not recommended to use the CNG feature when an MSG card is installed in the PBX.

Installation Manual References

3.3.1 IPCMPR Card
1.17.7 Automatic Fax Transfer

3.7.1 OPB3 Card (KX-NCP1190)
3.7.7 ESVM2 Card (KX-TDA0192) and ESVM4 Card (KX-TDA0194)

PC Programming Manual References

3.5 [1-1] Slot—OPB3 Card Property—◆ MSG Feature
7.4 [5-3-2] Voice Message—DISA Message—◆ Fax Extension

PT Programming Manual References

None

Feature Guide References

1.17.6 Direct Inward System Access (DISA)
1.17.8 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]

<table>
<thead>
<tr>
<th>Destination</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wired Extension (PT/SLT/ISDN Extension/T1-OPX)</td>
<td>✓</td>
</tr>
<tr>
<td>PS</td>
<td>✓</td>
</tr>
<tr>
<td>SIP Extension</td>
<td></td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td></td>
</tr>
<tr>
<td>PS Ring Group</td>
<td></td>
</tr>
<tr>
<td>Floating Extension no. for SVM</td>
<td></td>
</tr>
<tr>
<td>VM Group (DTMF/DPT)</td>
<td></td>
</tr>
<tr>
<td>External Pager (TAFAS)</td>
<td></td>
</tr>
<tr>
<td>DISA</td>
<td></td>
</tr>
<tr>
<td>Analogue/ISDN Remote Maintenance</td>
<td></td>
</tr>
<tr>
<td>Idle Line Access no. + Phone no.</td>
<td></td>
</tr>
<tr>
<td>Trunk Group Access no. + Trunk Group no. + Phone no.</td>
<td></td>
</tr>
<tr>
<td>Other PBX Extension (TIE with no PBX Code)</td>
<td></td>
</tr>
<tr>
<td>Other PBX Extension (TIE with PBX Code)</td>
<td></td>
</tr>
</tbody>
</table>

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.

Conditions

- Hardware Requirement:
  An external sensor, OPB card and EIO 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. (→ 1.3.1.2 Call Forwarding (FWD))
• The following features cannot be used when a sensor call is received:
  – Alternate Receiving—Voice (→ 1.5.3 Intercom Call)
  – Hands-free Answerback (→ 1.4.1.4 Hands-free Answerback)
  – Consultation Hold/Call Hold (→ 1.13.1 Call Hold)
  – Call Transfer (→ 1.12.1 Call Transfer)
  – Executive Busy Override (→ 1.8.2 Executive Busy Override)
• Sensor call information is output on SMDR.

Installation Manual References
3.7.1 OPB3 Card (KX-NCP1190)
3.7.4 ElO4 Card (KX-TDA0164)
3.12.1 Connection of Doorphones, Door Openers, External Sensors, and External Relays

PC Programming Manual References
3.41 [1-1] Slot—OPB3 Card Property
  → ◆ Input Signal Decision Time
  → ◆ Input Signal Detection Reopening Time
4.17 [2-8-3] Ring Tone Patterns—Call from Others—◆ External Sensor—Ring Tone Pattern Plan 1–8
7.7 [5-5] External Sensor

PT Programming Manual References
None
1.17.9  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), reorder tone will be heard at the extension.

Conditions

- Hardware Requirement:
  
  An external relay device, OPB card and EIO card

- 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

3.7.1  OPB3 Card (KX-NCP1190)
3.7.4  EIO4 Card (KX-TDA0164)
3.12.1  Connection of Doorphones, Door Openers, External Sensors, and External Relays

PC Programming Manual References

3.41  [1-1] Slot—OPB3 Card Property—Device Type
4.9  [2-6-1] Numbering Plan—Main—Features—External Relay Access
4.12  [2-7-1] Class of Service—COS Settings—Optional Device & Other Extensions—External Relay Access
7.6  [5-4] External Relay

PT Programming Manual References

None

Feature Guide References

2.2.1  Class of Service (COS)

User Manual References

1.10.2  If an External Relay is Connected
1.18 Caller ID Features

1.18.1 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:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID</td>
<td>Caller’s information which is sent from an analogue trunk.</td>
</tr>
<tr>
<td></td>
<td>The following Caller ID signalling types are supported: FSK and DTMF.</td>
</tr>
<tr>
<td>Calling Line Identification Presentation (CLIP)</td>
<td>Caller’s information which is sent from an ISDN line.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP)</td>
</tr>
<tr>
<td>Automatic Number Identification (ANI)</td>
<td>Caller’s information which is sent from an E1 or T1 line.</td>
</tr>
<tr>
<td></td>
<td>→ 1.22.1 E1 Line Service</td>
</tr>
<tr>
<td></td>
<td>→ 1.23.1 T1 Line Service</td>
</tr>
</tbody>
</table>

2. Service Features

<table>
<thead>
<tr>
<th>Features</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Line Identification (CLI) Distribution</td>
<td>Directs a DIL/DID/DDI/MSN call to a CLI destination if the caller’s identification (Caller ID/CLIP/ANI) has been assigned to the Caller ID Table.</td>
</tr>
<tr>
<td></td>
<td>→ 1.1.1.5 Calling Line Identification (CLI) Distribution</td>
</tr>
<tr>
<td>Incoming Call Log</td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>→ 1.18.2 Incoming Call Log</td>
</tr>
</tbody>
</table>

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.

[Example]

<table>
<thead>
<tr>
<th>&lt;Table Selection&gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trunk Group No.</td>
</tr>
<tr>
<td>1</td>
</tr>
<tr>
<td>2</td>
</tr>
<tr>
<td>:</td>
</tr>
</tbody>
</table>

<Modification Table>

<table>
<thead>
<tr>
<th>Area Code</th>
<th>Removed No. of Digits</th>
<th>Added No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Local/International Call Data 1</td>
<td>012</td>
<td>3</td>
</tr>
<tr>
<td>Local/International Call Data 2</td>
<td>00</td>
<td>2</td>
</tr>
<tr>
<td>Long Distance Call Data</td>
<td>Not programmable</td>
<td>0</td>
</tr>
</tbody>
</table>

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>

<table>
<thead>
<tr>
<th>Call Type</th>
<th>Removed No. of Digits</th>
<th>Added No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Subscriber Call Data</td>
<td>0</td>
<td>Blank</td>
</tr>
<tr>
<td>National Call Data</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>International Call Data</td>
<td>0</td>
<td>00</td>
</tr>
</tbody>
</table>
### 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]

<table>
<thead>
<tr>
<th>Location (System Speed Dialling No.)</th>
<th>Telephone No.</th>
<th>System Speed Dialling Name</th>
<th>CLI Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td>000</td>
<td>90123456789</td>
<td>ABC Company</td>
<td>200</td>
</tr>
<tr>
<td>001</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

*1 → 8.1  [6-1] System Speed Dial— ◆ CO Line Access Number + Telephone Number
→ [001] System Speed Dialling Number
  ² → 8.1 [6-1] System Speed Dial—◆ Name
  → [002] System Speed Dialling Name
  ³ → 8.1 [6-1] 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:**
  A DHLC4 card, SLC8 card, or SLC16 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.
- A caller ID name received from the network via a PRI23 card will not be displayed on an SLT. However, if a System Speed Dialling name is registered, it will be displayed.

**PC Programming Manual References**

4.5 [2-3] Timers & Counters—Miscellaneous
  → ◆ Caller ID—Waiting to receive
  → ◆ Caller ID—Visual Caller ID Display
4.18 [2-9] System Options—Option 4—◆ Private Network—Public Call through Private Network—Minimum Public Caller ID Digits
4.19 [2-10] Extension CID Settings
5.1 [3-1-1] Trunk Group—TRG Settings—Main—◆ Caller ID Modification Table
6.1 [4-1-1] Wired Extension—Extension Settings—Option 8
  → ◆ Extension Caller ID Sending
  → ◆ Incoming Call Wait Timer for Extension Caller ID
8.1 [6-1] System Speed Dial
1.18.1 Caller ID

5.3 [3-1-3] 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

1.6.1.5 Speed Dialling—Personal/System
1.26.1.1 Station Message Detail Recording (SMDR)
1.18.2 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. 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]

<table>
<thead>
<tr>
<th>Caller’s name (20 digits max.)</th>
<th>Date and time of a call received</th>
<th>Answering Status*</th>
<th>Caller’s number (16 digits max.)</th>
</tr>
</thead>
<tbody>
<tr>
<td>John White</td>
<td>DEC.12 10:00AM MON</td>
<td>NEW: Not Answered</td>
<td>123456789</td>
</tr>
</tbody>
</table>

* "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:

<table>
<thead>
<tr>
<th>Light pattern</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Red on</td>
<td>There is unchecked information.</td>
</tr>
<tr>
<td>Off</td>
<td>All information has been checked.</td>
</tr>
</tbody>
</table>

- **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.

- **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, the Personal Speed Dialling number display and the SVM Log display are also locked and the voice...
messages in the user’s message box cannot be played back. An extension personal identification number (PIN) is required to use this feature. (→ 1.28.1 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 is automatically attached to the telephone number.

- **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 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.

**PC Programming Manual References**

5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Miscellaneous— Supervisor Extension Number
5.16 [3-5-3] Incoming Call Distribution Group—Miscellaneous— Options—Call Log to ICD Group for Answered Call
6.1 [4-1-1] Wired Extension—Extension Settings
  → Option 5— Incoming Call Display
  → Option 6— Display Lock / SVM Lock
  → Option 7— Incoming Call Log Memory
6.5 [4-1-4] Wired Extension—Flexible Button— Type
6.10 [4-2-1] Portable Station—Extension Settings
  → Option 5— Incoming Call Display
  → Option 7— Incoming Call Log Memory
6.17 [4-3] DSS Console— Type

**PT Programming Manual References**

None

**Feature Guide References**

1.6.1.5 Speed Dialling—Personal/System
1.20.2 Flexible Buttons
3.1 Capacity of System Resources

**User Manual References**

1.13.1 Using the Call Log
3.1.2 Settings on the Programming Mode
1.19 Message Features

1.19.1 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 Voice Processing System (VPS) or 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]

105: Tom Smith

--- Extension no. and name of who left the message

* Own extension
* Other extension

*: For example, this button is useful when the secretary checks the message for the boss (Boss & Secretary Feature).

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. (→ 1.29.1 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 on the VPS, whether the notification is cleared or not depends on the VPS settings.

- This feature is not available for SIP extensions. However, KX-HGT100 users can use this feature.

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—Message Waiting Set / Cancel / Call Back
4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature—Message Waiting Set
4.18 [2-9] System Options—Option 3—Dial Tone—Distinctive Dial Tone
5.22 [3-8-1] VM(DTMF) Group—System Settings—VM DTMF Command—Listening Message
1.19.1 Message Waiting

6.5 [4-1-4] Wired Extension—Flexible Button—◆ Type
6.13 [4-2-3] Portable Station—Flexible Button—◆ Type
6.17 [4-3] DSS Console—◆ Type

PT Programming Manual References
None

Feature Guide References
1.20.2 Flexible Buttons
3.1 Capacity of System Resources

User Manual References
1.2.4 When the Dialled Line is Busy or There is No Answer
1.10.5 If a Voice Processing System is Connected
1.19.2 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:

<table>
<thead>
<tr>
<th>Type</th>
<th>Message No.</th>
<th>Message (Example)</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>System message</td>
<td>1</td>
<td>Will Return Soon</td>
<td>Messages may be edited through system programming. They are used for every extension user commonly.</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>Gone Home</td>
<td></td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>At Ext %%% (Extension Number)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>4</td>
<td>Back at %:%%% (Hour:Minute)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>Out until %/% (Month/Day)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>6</td>
<td>In a Meeting</td>
<td></td>
</tr>
<tr>
<td></td>
<td>7</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>8</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Personal message</td>
<td>9</td>
<td></td>
<td>A message is programmable at each extension through personal programming (Personal Absent Message), which can only be used by that extension user.</td>
</tr>
</tbody>
</table>

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.

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—Absent Message Set / Cancel
6.1 [4-1-1] Wired Extension—Extension Settings—Option 3—Absent Message
6.10 [4-2-1] Portable Station—Extension Settings—Option 3—Absent Message
8.5 [6-5] Absent Message

PT Programming Manual References

[008] Absent Message

Feature Guide References

3.1 Capacity of System Resources
1.19.2 Absent Message

**User Manual References**

1.6.3 Showing a Message on the Caller’s Telephone Display (Absent Message)
3.1.2 Settings on the Programming Mode
1.20 Proprietary Telephone (PT) Features

1.20.1 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]

<table>
<thead>
<tr>
<th>Button</th>
<th>Usage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Navigator Key, Jog Dial, Volume Key</td>
<td>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.</td>
</tr>
<tr>
<td>ENTER</td>
<td>Used to confirm the selected item.</td>
</tr>
<tr>
<td>CANCEL</td>
<td>Used to cancel the selected item.</td>
</tr>
<tr>
<td>PROGRAM</td>
<td>PROG.</td>
</tr>
<tr>
<td>FLASH/RECALL</td>
<td>R</td>
</tr>
</tbody>
</table>

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.
<table>
<thead>
<tr>
<th>Button</th>
<th>Usage</th>
</tr>
</thead>
<tbody>
<tr>
<td>HOLD</td>
<td>Used to place a call on hold.</td>
</tr>
<tr>
<td>SP-PHONE (Speakerphone)</td>
<td>Used for hands-free operation. Also used to switch between handset and hands-free operation.</td>
</tr>
<tr>
<td>MONITOR</td>
<td>Used for a hands-free dialling. Also used to monitor the party’s voice in hands-free mode.</td>
</tr>
<tr>
<td>MESSAGE</td>
<td>Used to leave a message waiting indication or call back the party who left the message waiting indication.</td>
</tr>
<tr>
<td>REDIAL</td>
<td>Used to redial the last dialled number.</td>
</tr>
<tr>
<td>TRANSFER</td>
<td>Used to transfer a call to another party.</td>
</tr>
<tr>
<td>Flexible CO (Trunk)</td>
<td>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.</td>
</tr>
<tr>
<td>INTERCOM</td>
<td>Used to make or receive intercom calls.</td>
</tr>
<tr>
<td>AUTO ANS (Auto Answer)/MUTE</td>
<td>Used to receive an incoming call in hands-free mode, or used for microphone or handset mute during a conversation. (Dual feature button)</td>
</tr>
<tr>
<td>VOICE CALL/MUTE</td>
<td>Used to monitor an intercom call automatically (a hands-free conversation is not possible). Also used for handset microphone mute during a conversation.</td>
</tr>
<tr>
<td>AUTO DIAL/STORE</td>
<td>Used for System/Personal Speed Dialling and storing programme changes.</td>
</tr>
<tr>
<td>CONF (Conference)</td>
<td>Used to establish a multiple-party conversation.</td>
</tr>
<tr>
<td>FWD/DND</td>
<td>Used to perform FWD or DND. (Dual feature button)</td>
</tr>
<tr>
<td>PAUSE</td>
<td>Used to insert a pause in a stored number. With an APT, it is used as the PROGRAM button.</td>
</tr>
<tr>
<td>Soft</td>
<td>Used to select the item displayed on the bottom line of the display.</td>
</tr>
<tr>
<td>SELECT</td>
<td>Used to select the displayed item or to call the displayed number.</td>
</tr>
<tr>
<td>SHIFT</td>
<td>Used to access the second level of Soft button items.</td>
</tr>
<tr>
<td>MODE</td>
<td>Used to shift the display to access various features.</td>
</tr>
<tr>
<td>NEXT PAGE</td>
<td>Used to switch the page for the Self Labelling feature (KX-NT366 only).</td>
</tr>
</tbody>
</table>
### [DSS Console]

<table>
<thead>
<tr>
<th>Button</th>
<th>Usage</th>
</tr>
</thead>
<tbody>
<tr>
<td>ANSWER</td>
<td>Used to answer an incoming call or place the current call on hold and answer another call with one touch.</td>
</tr>
<tr>
<td>RELEASE</td>
<td>Used to disconnect the line during or after a conversation or to complete a Call Transfer.</td>
</tr>
<tr>
<td>Flexible CO (Trunk)</td>
<td>Used to make or receive a trunk call or can be reassigned to a different trunk or to another feature button.</td>
</tr>
<tr>
<td>Flexible DSS (Direct Station Selection)</td>
<td>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.</td>
</tr>
<tr>
<td>PF (Programmable Feature)</td>
<td>Used to access a preprogrammed feature with one touch. (no default)</td>
</tr>
</tbody>
</table>

#### Conditions
- Certain buttons are equipped with a light to show line or feature status.
1.20.2 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:

<table>
<thead>
<tr>
<th>Button Usage</th>
<th>Usage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Single-CO (S-CO)</td>
<td>Used to access a specified trunk for making or receiving calls.</td>
</tr>
<tr>
<td>Group-CO (G-CO)</td>
<td>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.</td>
</tr>
<tr>
<td>Loop-CO (L-CO)</td>
<td>Used to access an idle trunk for making calls. Incoming calls from any trunk arrive at this button.</td>
</tr>
<tr>
<td>Direct Station Selection (DSS)</td>
<td>Used to access an extension with one touch.</td>
</tr>
<tr>
<td>Network Direct Station Selection (NDSS)</td>
<td>Used to access an extension at another PBX within the same network.</td>
</tr>
<tr>
<td>One-touch Dialling</td>
<td>Used to access a preprogrammed party or feature with one touch.</td>
</tr>
<tr>
<td>Incoming Call Distribution (ICD) Group</td>
<td>Used to access a specified incoming call distribution group for making or receiving calls.</td>
</tr>
<tr>
<td>Message</td>
<td>Used to leave a message waiting indication or call back the party who left the message waiting indication.</td>
</tr>
<tr>
<td>FWD/DND (External/Internal/Both)*</td>
<td>Used to perform the FWD or DND feature for the extension. The feature is applied to trunk calls, intercom calls, or both.</td>
</tr>
<tr>
<td>Group FWD (External/Internal/Both)</td>
<td>Used to perform the FWD feature for a specified incoming call distribution group. The feature is applied to trunk calls, intercom calls, or both.</td>
</tr>
<tr>
<td>Account Code Entry (Account)</td>
<td>Used to enter an account code.</td>
</tr>
<tr>
<td>Conference</td>
<td>Used to establish a multiparty conversation.</td>
</tr>
<tr>
<td>Terminate</td>
<td>Used to disconnect the current call and make another call without hanging up.</td>
</tr>
<tr>
<td>External Feature Access (EFA)</td>
<td>Used to send a flash/recall signal to the telephone company or host PBX to access their features.</td>
</tr>
<tr>
<td>Charge Reference</td>
<td>Used to check the total call charge for your own extension.</td>
</tr>
<tr>
<td>Call Park</td>
<td>Used to park or retrieve a call in a preset PBX parking zone.</td>
</tr>
<tr>
<td>Call Park (Automatic Park Zone)</td>
<td>Used to park a call in an idle PBX parking zone automatically.</td>
</tr>
<tr>
<td>Call Log</td>
<td>Used to show the incoming call information.</td>
</tr>
<tr>
<td>Log-in/Log-out*</td>
<td>Used to switch between log-in and log-out mode.</td>
</tr>
<tr>
<td>Hurry-up</td>
<td>Used to redirect the longest waiting call in the queue of an incoming call distribution group to the overflow destination.</td>
</tr>
<tr>
<td>Button</td>
<td>Usage</td>
</tr>
<tr>
<td>---------------------------------------------</td>
<td>----------------------------------------------------------------------</td>
</tr>
<tr>
<td>Wrap-up&lt;sup&gt;1&lt;/sup&gt;</td>
<td>Used to switch the Wrap-up/Not Ready and Ready modes.</td>
</tr>
<tr>
<td>System Alarm</td>
<td>Used to confirm a PBX error.</td>
</tr>
<tr>
<td>Time Service&lt;sup&gt;1&lt;/sup&gt;</td>
<td>Used to switch the assigned time modes: day, lunch, break or night. Also used to check the current time mode status.</td>
</tr>
<tr>
<td>Answer</td>
<td>Used to answer an incoming call.</td>
</tr>
<tr>
<td>Release</td>
<td>Used to disconnect the line during or after a conversation, or to complete a Call Transfer.</td>
</tr>
<tr>
<td>Toll Restriction/Call Barring</td>
<td>Used to change the TRS/Barring level of other extension users tempo- rarily.</td>
</tr>
<tr>
<td>ISDN Service</td>
<td>Used to access an ISDN service.</td>
</tr>
<tr>
<td>Calling Line Identification Restriction (CLIR)&lt;sup&gt;1&lt;/sup&gt;</td>
<td>Used to switch between the CLIP and CLIR service.</td>
</tr>
<tr>
<td>Connected Line Identification Restriction (COLR)&lt;sup&gt;1&lt;/sup&gt;</td>
<td>Used to switch between the COLP and COLR service.</td>
</tr>
<tr>
<td>ISDN Hold</td>
<td>Used to transfer a call using the telephone company.</td>
</tr>
<tr>
<td>Headset</td>
<td>Used to turn on/off the headset mode while idle.  Used to switch between hands-free mode and headset modes during a conversation.</td>
</tr>
<tr>
<td>Time Service Switching Mode (Automatic/Manual)&lt;sup&gt;1&lt;/sup&gt;</td>
<td>Used to switch between the Automatic Switching and Manual Switching mode.</td>
</tr>
<tr>
<td>Two-way Record</td>
<td>Used to record a conversation into your own mailbox.</td>
</tr>
<tr>
<td>Two-way Transfer</td>
<td>Used to record a conversation into the mailbox of a specified exten- sion.</td>
</tr>
<tr>
<td>One-touch Two-way Transfer</td>
<td>Used to record a conversation into the mailbox of a specified exten- sion with one touch.</td>
</tr>
<tr>
<td>Live Call Screening (LCS)</td>
<td>Used to monitor your own voice mailbox while an incoming caller is leaving a message and, if desired, intercept the call.</td>
</tr>
<tr>
<td>Voice Mail (VM) Transfer</td>
<td>Used to transfer a call to the mailbox of a specified extension.</td>
</tr>
<tr>
<td>Check-in</td>
<td>Used to switch the status of extensions from Check-out to Check-in.</td>
</tr>
<tr>
<td>Check-out</td>
<td>Used to switch the status of extensions from Check-in to Check-out.</td>
</tr>
<tr>
<td>Cleaned-up</td>
<td>Used to switch the room status of extensions between Ready and Not Ready.</td>
</tr>
<tr>
<td>CTI</td>
<td>Used to access CTI features.</td>
</tr>
<tr>
<td>Primary Directory Number (PDN)</td>
<td>Used to make and receive both outside and intercom calls.</td>
</tr>
</tbody>
</table>
### 1.20.2 Flexible Buttons

<table>
<thead>
<tr>
<th>Button</th>
<th>Usage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secondary Directory Number (SDN)</td>
<td>Used to show the current status of another extension, call the extension, and pick up or transfer calls to it. (→ 1.7.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension)</td>
</tr>
</tbody>
</table>

*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.

### Conditions

- The Flexible DSS button lights of the KX-T7040 and KX-T7240 show only "Off" or "Red On". If the button is customised as a button which shows other light patterns, the other light patterns also show only as "Red On".

### PC Programming Manual References

- 6.5 [4-1-4] Wired Extension—Flexible Button
- 6.6 [4-1-4] Wired Extension—Flexible Button—Flexible button data copy
- 6.7 [4-1-5] Wired Extension—PF Button
- 6.13 [4-2-3] Portable Station—Flexible Button
- 6.17 [4-3] DSS Console

### PT Programming Manual References

None

### User Manual References

- 3.1.3 Customising the Buttons
1.20.3 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

[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>
<thead>
<tr>
<th>Light Pattern</th>
<th>Line Status Button</th>
<th>Trunk Status</th>
<th>Intercom Line Status</th>
<th>Incoming Call Distribution Group Line Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td></td>
<td>S-CO</td>
<td>G-CO</td>
<td>L-CO INTERCOM ICD Group</td>
</tr>
<tr>
<td>Green on</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Slow green flashing</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Moderate green flashing</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Rapid green flashing</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Red on</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Trunk Status</th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>S-CO</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>G-CO</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>L-CO</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>INTERCOM</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>ICD Group</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- **Trunk Status**
  - Off: Idle
  - Green on: This extension is using the line.
  - Slow green flashing: This extension is holding the line.
  - Moderate green flashing: This extension is holding the line using Exclusive Hold or using the line for an Unattended Conference.
  - Rapid green flashing: Incoming call/Privacy Release
  - Red on: Incoming call for another extension/Another extension is using the line/Another extension has the line on Exclusive Hold.
  - Other extensions are using all trunks in the trunk group.

- **Incoming Call Distribution Group Line Status**
  - —
  - This extension is logged out of the incoming call distribution group.

Feature Guide
1.20.3 LED Indication

<table>
<thead>
<tr>
<th>Light Pattern</th>
<th>Trunk Status</th>
<th>Intercom Line Status</th>
<th>Incoming Call Distribution Group Line Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Slow red flashing</td>
<td>Another extension is holding the line.</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>Rapid red flashing</td>
<td>Incoming call to the incoming call distribution group in Ring distribution method</td>
<td>—</td>
<td>—</td>
</tr>
</tbody>
</table>

For information on the light patterns of PDN and SDN buttons, refer to 1.7.1 Primary Directory Number (PDN)/Secondary Directory Number (SDN) Extension.

3. **Light Pattern of the Corresponding Extension Status Button**

<table>
<thead>
<tr>
<th>Light Pattern</th>
<th>Corresponding Extension Status Button (DSS)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td>Idle</td>
</tr>
<tr>
<td>Red on</td>
<td>Busy/Incoming call*/DND for trunk calls</td>
</tr>
<tr>
<td>Rapid red flashing</td>
<td>Incoming call*2</td>
</tr>
</tbody>
</table>

*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**

- **Slow Flashing**
- **Moderate Flashing**
- **Rapid Flashing**

**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**

4.18 [2-9] 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

**PT Programming Manual References**

None

**Feature Guide References**

1.2.2.2 Group Call Distribution  
1.2.2.7 Log-in/Log-out  
1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)  
1.13.1 Call Hold  
1.14.1.2 Conference  
1.14.1.3 Privacy Release
1.20.4 Display Information

Description

A display PT shows the user the following information while making or receiving calls if they are available:

<table>
<thead>
<tr>
<th>Display Item</th>
<th>Display Example</th>
<th>Condition</th>
</tr>
</thead>
<tbody>
<tr>
<td>The extension number and name of the calling or called extension, or incoming call distribution group</td>
<td>123: Tom Smith</td>
<td>–</td>
</tr>
<tr>
<td>Status of the called extension</td>
<td>123: Busy</td>
<td>–</td>
</tr>
<tr>
<td>The number and name of the optional device</td>
<td>D02: 1st Door</td>
<td>–</td>
</tr>
<tr>
<td>The dialled telephone number</td>
<td>1234567890</td>
<td>–</td>
</tr>
<tr>
<td>The received call information</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a. Caller’s name</td>
<td>ABC Company</td>
<td></td>
</tr>
<tr>
<td>b. Caller’s number</td>
<td>12345678</td>
<td></td>
</tr>
<tr>
<td>c. Trunk number/name</td>
<td>Line 001: Sales</td>
<td></td>
</tr>
<tr>
<td>d. Original Destination, if the call is forwarded</td>
<td>102: Mike Panasonic</td>
<td></td>
</tr>
<tr>
<td>e. DDI/DID/MSN name</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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

4.18 [2-9] System Options—Option 5—PT Feature Access—No. 1–8
6.1 [4-1-1] Wired Extension—Extension Settings
  →Option 5—Display Language
  →Option 5—Incoming Call Display
  →Option 5—Automatic LCD Switch when Start Talking
6.10 [4-2-1] Portable Station—Extension Settings
1.20.4 Display Information

→Option 5—◆ Display Language
→Option 5—◆ Incoming Call Display
→Option 5—◆ Automatic LCD Switch when Start Talking

8.2  [6-2] Hotel & Charge—Charge
→◆ Charge Options—Digits After Decimal Point
→◆ Charge Options—Currency
→◆ Charge Options—Currency Display Position

12.1  [10-1] 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
1.21 Integrated Services Digital Network (ISDN) Service Features

1.21.1 Integrated Services Digital Network (ISDN)

1.21.1.1 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

<table>
<thead>
<tr>
<th>Interface Type</th>
<th>Description</th>
<th>Port Mode</th>
<th>Configuration Type¹</th>
</tr>
</thead>
<tbody>
<tr>
<td>Basic Rate Interface (BRI)</td>
<td>Provides two 64 kbps B channels for communication and one 16 kbps D channel for signalling (2B + D).</td>
<td>Trunk, Extension¹</td>
<td>Point-to-Point/Point-to-multipoint</td>
</tr>
<tr>
<td></td>
<td></td>
<td>QSIG (Master, Slave)¹</td>
<td></td>
</tr>
<tr>
<td>Primary Rate Interface (PRI)</td>
<td>Provides thirty or twenty-three 64 kbps B channels for communication and one 64 kbps D channel for signalling (30B + D/23B + D).</td>
<td>Trunk, QSIG (Master, Slave), Extension²</td>
<td>Point-to-Point</td>
</tr>
</tbody>
</table>

¹ For PRI only:
→ 3.28 [1-1] Slot—Port Property - PRI Port—Main— Port Type
² For BRI only:
→ 3.28 [1-1] Slot—Port Property - BRI Port—Network— Access Mode
→ 426 BRI Configuration

Note

Point-to-Point (P-P):
One ISDN terminal device can be connected to one ISDN port.

Point-to-multipoint (P-MP):
A maximum of eight ISDN terminal devices can be connected to one ISDN port.

2. ISDN Supplementary Service Table for Public Network

<table>
<thead>
<tr>
<th>Service</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Direct Dialling In (DDI)</td>
<td>Directs a call with a DDI number to a preprogrammed destination. (P-P only)</td>
</tr>
<tr>
<td></td>
<td>→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)</td>
</tr>
<tr>
<td>Multiple Subscriber Number (MSN)</td>
<td>One ISDN port can support a maximum of 10 MSNs for routing incoming calls. (P-MP only)</td>
</tr>
<tr>
<td></td>
<td>→ 1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service</td>
</tr>
<tr>
<td>Service</td>
<td>Description &amp; Reference</td>
</tr>
<tr>
<td>----------------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>Calling Line Identification Presentation (CLIP)</strong></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP)</td>
</tr>
<tr>
<td><strong>Connected Line Identification Presentation (COLP)</strong></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP)</td>
</tr>
<tr>
<td><strong>Calling Line Identification Restriction (CLIR)</strong></td>
<td>Prevents the caller’s CLI being presented to the called party by the caller.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP)</td>
</tr>
<tr>
<td><strong>Connected Line Identification Restriction (COLR)</strong></td>
<td>Prevents COLP being sent by the answered party.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP)</td>
</tr>
<tr>
<td><strong>Subaddressing (SUB)</strong></td>
<td>You may add digits after the telephone number. These digits will be passed to ISDN terminal device.</td>
</tr>
<tr>
<td><strong>User-to-user Signalling Type 1 (UUS-1)</strong></td>
<td>Transmits a specified volume of the data through the D channel between the ISDN terminal devices of different PBXs using UUS types 1.</td>
</tr>
<tr>
<td><strong>Advice of Charge (AOC)</strong></td>
<td>The PBX can receive the call charge information on ISDN lines from the telephone company.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.3 Advice of Charge (AOC)</td>
</tr>
<tr>
<td><strong>Call Forwarding (CF)—by ISDN</strong></td>
<td>Forwards an incoming call to another outside party using the ISDN service of the telephone company.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.4 Call Forwarding (CF)—by ISDN (P-MP)</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.5 Call Forwarding (CF)—by ISDN (P-P)</td>
</tr>
<tr>
<td><strong>Call Hold (HOLD)—by ISDN</strong></td>
<td>Puts one ISDN call on hold.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.6 Call Hold (HOLD)—by ISDN</td>
</tr>
<tr>
<td><strong>Call Transfer (CT)—by ISDN</strong></td>
<td>Transfers an ISDN call to an outside party. Call Transfer with Announcement and Call Transfer without Announcement are available.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.7 Call Transfer (CT)—by ISDN</td>
</tr>
<tr>
<td><strong>Three-party Conference (3PTY)—by ISDN</strong></td>
<td>Establishes a three-party conference call using the ISDN service of the telephone company.</td>
</tr>
<tr>
<td></td>
<td>→ 1.21.1.8 Three-party Conference (3PTY)—by ISDN</td>
</tr>
</tbody>
</table>
### 1.21.1 Integrated Services Digital Network (ISDN)

<table>
<thead>
<tr>
<th>Service</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
</table>
| 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.  
→ 1.21.1.9 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.  
→ 1.21.1.10 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.

→ 3.28 [1-1] Slot—Port Property - BRI Port—ISDN CO—◆ ISDN Centrex  
→ 3.30 [1-1] 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 (→ 1.21.1.11 ISDN Extension)
An ISDN (BRI/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. When the ISDN port is in P-MP configuration, a maximum of eight ISDN terminal devices can be connected to the port. However, only a maximum of two devices can be used simultaneously.

### 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.

  → 3.28 [1-1] Slot—Port Property - BRI Port—ISDN CO—◆ ISDN Outgoing Call Type  
  → 3.30 [1-1] 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).
  → 4.18 [2-9] System Options—Option 2—◆ ISDN en Bloc Dial—[#] as End of Dial for en Bloc mode  
  - the dialled number is a preprogrammed telephone number.
  → 5.4 [3-1-4] Dialling Plan  
  - the inter-digit timer expires.

- **Some supplementary services are provided by the key protocol (→ 1.21.1.12 ISDN Service Access by Keypad Protocol).**

- **Extension Number**
  An extension number can be assigned to each ISDN port.
  → 6.1 [4-1-1] Wired Extension—Extension Settings—Main—◆ Extension Number
• **Network Type (BRI only)**
  Each BRI port can be set to the requirements of each country/area.
  → 3.28 [1-1] Slot—Port Property - BRI Port—Network— ◆ Network Type
  → [420] BRI Network Type

• **L1 Active Mode and L2 Data Link Mode (BRI only)**
  The active mode of Layer 1 (L1) and the data link mode of Layer 2 (L2) can be selected individually (Permanent or Call by Call) on an ISDN port basis.
  3.28 [1-1] Slot—Port Property - BRI Port—Network
  → ◆ L1 Mode
  → ◆ L2 Mode
  → [425] BRI Layer 2 Active Mode

• **TEI Assignment Mode (BRI only)**
  The Terminal Endpoint Identifier (TEI) assignment mode can be selected on an ISDN port basis.
  → 3.28 [1-1] Slot—Port Property - BRI Port—Network— ◆ TEI Mode
  → [427] BRI TEI Mode

• **Bearer Mode**
  The bearer mode can be assigned on an extension basis.
  → 6.1 [4-1-1] Wired Extension—Extension Settings—Option 7— ◆ ISDN Bearer
  → 6.10 [4-2-1] Portable Station—Extension Settings—Option 7— ◆ ISDN Bearer

• **Automatic ISDN Configuration**
  ISDN (BRI) port configuration can be set automatically through system programming.
  → 2.5.2 Tool—BRI Automatic Configuration
  The following items will be set by making and receiving a call using the subscriber number assigned for each ISDN (BRI) port:
  – L1 Active Mode
  – L2 Data Link Mode
  – Access Mode (Point-to-Point/Point-to-Multipoint)
  – TEI Assignment Mode (Fix 00–63/Automatic)

---

**PC Programming Manual References**

2.5.2 Tool—BRI Automatic Configuration
3.27 [1-1] Slot—Card Property - BRI type/PRI type
3.28 [1-1] Slot—Port Property - BRI Port
3.29 [1-1] Slot—Port Property - BRI Port—Connection Command
3.30 [1-1] Slot—Port Property - PRI Port
3.31 [1-1] Slot—Port Property - PRI Port—Connection Command
4.18 [2-9] System Options—Option 2— ◆ ISDN en Bloc Dial—[#] as End of Dial for en Bloc mode
6.1 [4-1-1] Wired Extension—Extension Settings
  → Main— ◆ Extension Number
  → Option 7— ◆ ISDN Bearer
6.10 [4-2-1] Portable Station—Extension Settings—Option 7— ◆ ISDN Bearer
5.5 [3-1-4] Dialling Plan—Auto Assign

---

**PT Programming Manual References**

[420] BRI Network Type
[424] BRI Layer 1 Active Mode
[425] BRI Layer 2 Active Mode
[426] BRI Configuration
[427] BRI TEI Mode
1.21.1 Integrated Services Digital Network (ISDN)

Feature Guide References

1.30.4 QSIG Standard Features
1.21.1.2 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]

1) Dials "87654321".

called party (CLIP/COLP No.: 87654321)

caller (CLIP/COLP No.: 12345678)

2) "12345678" is displayed.

[COLP Example]

1) Dials "111222333".

caller (CLIP/COLP No.: 111222333)

2) Answers the call.

3) "111222444" is displayed.

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. (→1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG)

PC Programming Manual References

3.28 [1-1] Slot—Port Property - BRI Port
   →ISDN CO—◆ Subscriber Number
   →Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY
3.30 [1-1] 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
3.33 [1-1] Slot—Port Property - T1 Port—◆ Subscriber Number
3.39 [1-1] Slot—Port Property - E1 Port—◆ Subscriber Number
4.9 [2-6-1] Numbering Plan—Main—Features
   →◆ COLR Set / Cancel
   →◆ CLIR Set / Cancel
   →◆ Switch CLIP of CO Line / Extension
4.18 [2-9] System Options—Option 4—◆ Send CLIP of CO Caller to ISDN—when Transfer (CLIP of Held Party)
5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Main—◆ CLIP on ICD Group Button
6.1 [4-1-1] Wired Extension—Extension Settings
   →Main—◆ Extension Number
   →ISDN CLIP—◆ CLIP ID
   →ISDN CLIP—◆ CLIP on Extension/CO
   →ISDN CLIP—◆ CLIR
   →ISDN CLIP—◆ COLR
6.5 [4-1-4] Wired Extension—Flexible Button—◆ Type
6.10 [4-2-1] Portable Station—Extension Settings
   →ISDN CLIP—◆ CLIP ID
PT Programming Manual References

[003] Extension Number
[422] BRI Subscriber Number
[606] CLIP/COLP Number

Feature Guide References

1.20.2 Flexible Buttons

User Manual References

1.8.4 Displaying Your Telephone Number on the Called Party and Caller’s Telephone (Calling/Connected Line Identification Presentation [CLIP/COLP])
1.8.5 Preventing Your Telephone Number Being Displayed on the Caller’s Telephone (Connected Line Identification Restriction [COLR])
1.8.6 Preventing Your Number Being Displayed on the Called Party’s Telephone (Calling Line Identification Restriction [CLIR])
1.21.1.3 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:

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Advice of Charge During Call (AOC-D)</td>
<td>AOC is received during the call and when the call is completed.</td>
</tr>
<tr>
<td>Advice of Charge At End of Call (AOC-E)</td>
<td>AOC is received when the call is completed.</td>
</tr>
</tbody>
</table>

**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. (→ 1.9.2 Budget Management)
- **AOC for ISDN extension**
  - An ISDN extension also receives AOC.

**PC Programming Manual References**

3.28 [1-1] Slot—Port Property - BRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY
3.30 [1-1] Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

**PT Programming Manual References**

None
1.21.1.4 Call Forwarding (CF)—by ISDN (P-MP)

**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. Extension users can set the forward destination to the network on an MSN basis. The setting must have been done beforehand to use this feature. There are the following types:

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Forwarding—Unconditional (CFU)</td>
<td>Incoming calls are forwarded unconditionally.</td>
</tr>
<tr>
<td>Call Forwarding—Busy (CFB)</td>
<td>An incoming call is forwarded when the destination in the PBX is busy.</td>
</tr>
<tr>
<td>Call Forwarding—No Reply (CFNR)</td>
<td>An incoming call is forwarded when the destination in the PBX does not answer within a preprogrammed time period.</td>
</tr>
</tbody>
</table>

When the CFB or CFNR has been set, the network determines to forward the incoming call to the preprogrammed destination after the call reached the PBX once. When the CFU has been set, the network forwards the call directly to the preprogrammed destination.

[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.
The feature requires the MSN service. (→ 1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service)
COS programming determines the extensions that are able to use this feature.

**PC Programming Manual References**

4.9 [2-6-1] Numbering Plan—Main—Features—◆ ISDN-FWD (MSN) Set / Cancel / Confirm
4.12 [2-7-1] Class of Service—COS Settings—CO & SMDR—◆ CF (MSN)

**PT Programming Manual References**

None

**Feature Guide References**

1.3.1.2 Call Forwarding (FWD)
2.2.1 Class of Service (COS)

**User Manual References**

1.6.1 Forwarding Calls
1.21.1.5 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

3.28 [1-1] Slot—Port Property - BRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY
3.30 [1-1] Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY
4.9 [2-6-1] Numbering Plan—Main—Features—◆ ISDN-FWD (MSN) Set / Cancel / Confirm
4.12 [2-7-1] Class of Service—COS Settings—CO & SMDR—ﬂ CF (MSN)

PT Programming Manual References
None

Feature Guide References
1.3.1.1 Call Forwarding (FWD)/Do Not Disturb (DND)—SUMMARY
1.3.1.2 Call Forwarding (FWD)

User Manual References
1.6.1 Forwarding Calls
1.21.1.6 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 (→ 1.21.1.7 Call Transfer (CT)—by ISDN) and Three-party Conference (3PTY)—by ISDN (→ 1.21.1.8 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. (→ 1.9.1 Toll Restriction (TRS)/Call Barring (Barring))
- ARS cannot be applied to the call dialled after activating this feature. (→ 1.10.1 Automatic Route Selection (ARS))
- It is impossible to seize any other trunk during this feature.

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—◆ ISDN Hold
6.5 [4-1-4] Wired Extension—Flexible Button—◆ Type
6.13 [4-2-3] Portable Station—Flexible Button—◆ Type
6.17 [4-3] DSS Console—◆ Type

PT Programming Manual References

None

Feature Guide References

1.20.2 Flexible Buttons
1.21.1.7 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. (→ 1.12.1 Call Transfer)
- The call charges after completing this feature will not be recorded by the PBX.

PC Programming Manual References
3.28 [1-1] Slot—Port Property - BRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY
3.30 [1-1] Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

PT Programming Manual References
None

User Manual References
1.4.1 Transferring a Call (Call Transfer)
1.21.1.8 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**

3.28 [1-1] Slot—Port Property - BRI Port—Supplementary Service— ◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY

3.30 [1-1] Slot—Port Property - PRI Port—Supplementary Service— ◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

**PT Programming Manual References**

None

**User Manual References**

1.4.5 Multiple Party Conversation
1.21.1.9 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
4.9 [2-6-1] Numbering Plan—Main—Features—◆ MCID

PT Programming Manual References
None

User Manual References
1.3.6 Identifying Malicious Calling Parties (Malicious Call Identification [MCID])
1.21.1.10 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. (→ 1.6.1.4 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
3.28 [1-1] Slot—Port Property - BRI Port—Supplementary Service
   → ◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY
   → CCBS Option
3.30 [1-1] 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

PT Programming Manual References
None

User Manual References
1.2.4 When the Dialled Line is Busy or There is No Answer
1.21.1 Integrated Services Digital Network (ISDN)

1.21.1.11 ISDN Extension

Description

An ISDN (BRI/PRI) port can be used for either trunk or extension connection. When 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.

If the ISDN port is in point-to-point (P-P) configuration, one terminal device can be connected to the port. If the ISDN port is in point-to-multipoint (P-MP) configuration, a maximum of eight terminal devices can be connected to the port. However, only a maximum of two devices can be used simultaneously.

Individual terminal devices on a behind PBX can be addressed with Multiple Subscriber Number (MSN). The MSN consists of the ISDN extension number and an additional digit, 0 through 9 or 00 through 99.

[Example]

Conditions

- **Subaddressing**
  Subaddressing is possible between ISDN terminal devices. The subaddress goes through the PBX to the ISDN terminal devices directly.

- **Call Transfer (PBX feature)** is available only for ISDN extensions in P-MP configuration. (→ 1.12.1 Call Transfer)

- ISDN extensions can belong to an incoming call distribution group (→ 1.2.2 Incoming Call Distribution Group Features) or idle extension hunting group (→ 1.2.1 Idle Extension Hunting). In this case, an MSN can be assigned.
• If the last digit of the MSN is "0", all terminal devices on the same ISDN port receive the call simultaneously, if the specification of each terminal device is available. To use the MSN whose last digit is "0" as an individual MSN, system programming is required for each ISDN port.

PC Programming Manual References
3.28 [1-1] Slot—Port Property - BRI Port—ISDN Extension
3.30 [1-1] Slot—Port Property - PRI Port—Extension Setting

PT Programming Manual References
None

Feature Guide References
1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service
1.21.1.12 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
- 6.5 [4-1-4] Wired Extension—Flexible Button
  - Type
  - Dial (for ISDN Service)
- 6.13 [4-2-3] Portable Station—Flexible Button
  - Type
  - Dial (for ISDN Service)
- 6.17 [4-3] DSS Console
  - Type
  - Dial (for ISDN Service)

PT Programming Manual References
None

Feature Guide References
- 1.20.2 Flexible Buttons
- 3.1 Capacity of System Resources

User Manual References
- 1.2.5 Accessing the ISDN Service (ISDN Service Access)
1.22 E1 Line Service Features

1.22.1 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

[DR2 (Digital System R2)/E & M-C (Continuous E & M)/E & M-P (Pulsed E & M)]

The E1 card supports three different channel types to provide connection at minimum expense. One of the three channel types can be assigned to each of the 30 channels of the E1 card individually according to tariffs and customer needs.

2. E1 Features

The following table shows the features available for each channel type:

[Feature Table]

<table>
<thead>
<tr>
<th>Channel Type</th>
<th>(1)</th>
<th>(2)</th>
<th>(3)</th>
<th>(4)</th>
<th>(5)</th>
</tr>
</thead>
<tbody>
<tr>
<td>DR2</td>
<td></td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>E &amp; M-C</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>E &amp; M-P</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
</tbody>
</table>

✓: Available

*1 Receiving Dial Mode should be MFC-R2.
*2 Available when this PBX is connected to a KX-NCP series/KX-TDE series/KX-TDA series PBX, or KX-TD500.

[Explanation]

<table>
<thead>
<tr>
<th>Number in the Table</th>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>(1)</td>
<td>TIE Line Service</td>
<td>Refer to the TIE Line Service feature. (→ 1.30.1 TIE Line Service)</td>
</tr>
<tr>
<td>(2)</td>
<td>Direct Inward Dialling (DID)</td>
<td>Refer to the DID feature. (→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI))</td>
</tr>
<tr>
<td>(3)</td>
<td>Direct In Line (DIL)</td>
<td>Refer to the DIL feature. (→ 1.1.1.2 Direct In Line (DIL))</td>
</tr>
</tbody>
</table>
| (4)                 | 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. (→ 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP))
|                     |                                  | 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. (→ 1.18.1 Caller ID) |
### 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

3.39 [1-1] Slot—Port Property - E1 Port

### PT Programming Manual References

None
1.23 T1 Line Service Features

1.23.1 T1 Line Service

Description

The T1 line carries twenty-four 64 kbps-voice channels at 1.5 Mbps transmission speed as a trunk or private line. Voice is digitised by Pulse Code Modulation (PCM).

1. Channel Type

[LCOT/GCOT/DID/TIE/OPX]

The T1 card supports five different channel types to provide connection at minimum expense. One of the five channel types can be assigned to one of the 24 channels of the T1 card individually according to tariffs and customer needs.

2. T1 Features

The following table shows the features available for each channel type:

[Feature Table]

<table>
<thead>
<tr>
<th>Channel Type</th>
<th>(1)</th>
<th>(2)</th>
<th>(3)</th>
<th>(4)</th>
<th>(5)</th>
</tr>
</thead>
<tbody>
<tr>
<td>TIE</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>DID</td>
<td></td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>DIL</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>TIE (E &amp; M)</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>OPX</td>
<td>✓</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

✓: Available

[Explanation]

<table>
<thead>
<tr>
<th>Number in the Table</th>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>(1)</td>
<td>TIE Line Service</td>
<td>Refer to the TIE Line Service feature. (→ 1.30.1 TIE Line Service)</td>
</tr>
<tr>
<td>(2)</td>
<td>Direct Inward Dialling (DID)</td>
<td>Refer to the DID feature. (→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI))</td>
</tr>
<tr>
<td>(3)</td>
<td>Direct In Line (DIL)</td>
<td>Refer to the DIL feature. (→ 1.1.1.2 Direct In Line (DIL))</td>
</tr>
</tbody>
</table>
### Number in the Table

<table>
<thead>
<tr>
<th>Number</th>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>(4)</td>
<td><strong>Off Premise Extension (OPX)</strong></td>
<td>An SLT which is installed off site can perform as if it were installed locally. The SLT makes and receives a call through the PBX. An extension number can be given to the OPX, and extension data such as COS is also applied.</td>
</tr>
<tr>
<td></td>
<td></td>
<td><strong>[Example]</strong></td>
</tr>
<tr>
<td></td>
<td></td>
<td>[Diagram showing SLT, T1 LCOT, T1 OPX, T1 Card, PBX, Extn.101, Extn.102, and SLT dialing “103”]</td>
</tr>
<tr>
<td>(5)</td>
<td><strong>Automatic Number Identification (ANI)</strong></td>
<td>Receives the caller’s number from the T1 line. When the ANI number is received, it can be treated the same as a Caller ID number. (→ 1.18.1 Caller ID)</td>
</tr>
</tbody>
</table>

**Conditions**
- If a T1 line is used as a trunk, the channel type depends on the contract with the telephone company.

**PC Programming Manual References**
- 3.33 [1-1] Slot—Port Property - T1 Port
- 3.34 [1-1] Slot—Port Property - T1 Port—Connection Command

**PT Programming Manual References**
- None
1.24 Voice Message Features

1.24.1 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 IPCMPR Card

This block can be used either for the SVM feature, for the OGM feature (→ 1.17.5 Outgoing Message (OGM)), or both.

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]

IPCMPR

SVM: Floating Extension No. 591 (default)

Message Box for Extn. 101
Message Box for Extn. 102
Message Box for Extn. 103
Message Box for Extn. 210
2 CH

ESVM2/ESVM4 Card

Like the SVM/OGM block on the IPCMPR card, an ESVM2/ESVM4 card, mounted on an OPB3 card, can be used either for the SVM feature, for the OGM feature, or both.

An ESVM4 card (4 channels) contains two blocks of data, and the ESVM2 card (2 channels) contains one block of data. Up to four ESVM cards (max. 16 channels) can be installed in the PBX.

Only up to 2 cards (max. 8 channels) can be used for the SVM feature per system. In this case, each card is assigned a floating extension number rather than the SVM/OGM block on the IPCMPR card. The rest of the channels must be used for OGM feature purposes.

When using an ESVM card for the SVM feature, it must be mounted on sub-slot 3 (leftmost slot) of the OPB3 card.
1.24.1 Built-in Simplified Voice Message (SVM)

[Example]

---

### SVM Card 1 (ESVM4): Floating Extension No. 591 (example)

**Block A**
- Message Box for Extn. 101
- Message Box for Extn. 102
- Message Box for Extn. 103
- Message Box for Extn. 210

**Block B**
- Message Box for Extn. 104
- Message Box for Extn. 105

**SVM Card 2 (ESVM2): Floating Extension No. 592 (example)**
- Message Box for Extn. 301

---

### Message Recording Limit

Up to 125 (IPCMPR)/250 (ESVM2/ESVM4) messages (greeting messages and voice messages for extensions) with a maximum total recording time of 60 minutes (default) can be recorded per card/block. This memory is shared between the message boxes of all extensions assigned to that card/block. The recording quality, which can be selected through system programming, determines the total recording time as follows:

<table>
<thead>
<tr>
<th>Recording Quality</th>
<th>Recording Time with the IPCMPR/ESVM2/ESVM4 Card</th>
</tr>
</thead>
<tbody>
<tr>
<td>High</td>
<td>about 20 minutes</td>
</tr>
<tr>
<td>Middle (default)</td>
<td>about 60 minutes</td>
</tr>
<tr>
<td>Low</td>
<td>about 120 minutes</td>
</tr>
</tbody>
</table>

When a card/block is used for the OGM feature, the recording quality can be set separately (→ 1.17.5 Outgoing Message (OGM)).
When using a card/block for both the SVM feature and OGM feature, the first 64 messages in the card/block are saved for OGM use and the remaining 186 messages will be for SVM use, as shown below:

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) (→ 1.3.1.2 Call Forwarding (FWD))
- Intercept Routing—No Answer/Busy/DND (→ 1.1.1.6 Intercept Routing)

**[Programming Example for Intercept Routing]**

<table>
<thead>
<tr>
<th>Extension No.</th>
<th>Intercept Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Day</td>
</tr>
<tr>
<td>101</td>
<td>102</td>
</tr>
<tr>
<td>102</td>
<td>103</td>
</tr>
<tr>
<td>301 (Operator)</td>
<td>–</td>
</tr>
</tbody>
</table>

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) (→ 2.2.4 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 (→ 1.2.2.6 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. (→ 1.25.6 Virtual PS)

Conditions

[General]

- This feature will function using the preinstalled IPCMPR card. To increase the capacity of the feature it is possible to add an ESVM2 card or ESVM4 card mounted on an OPB3 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 the ESVM card to which an extension belongs is changed through system programming, all the recorded voice messages and greeting messages for the extension will be cleared.
- When using the IPCMPR 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 (→ 1.27.1 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. (Default: 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) (→ 1.28.1 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). However, a user can only access the message boxes of other extensions assigned to the same card/block. For example, if the user accesses a message box belonging to card/block 1 first, then the user can only access the message boxes of other extensions belonging to card/block 1. If the user tries to access the message box of another extension belonging to a different card, the user will hear a reorder tone. While a reorder tone is heard, it is possible to retry another call by pressing "×".

*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 (1.30.4.3 Call Forwarding (CF)—by QSIG).

**Installation Manual References**
3.3.1 IPCMPR Card
3.7.1 OPB3 Card (KX-NCP1190)
3.7.7 ESVM2 Card (KX-TDA0192) and ESVM4 Card (KX-TDA0194)

**PC Programming Manual References**
2.5.5 Tool—Simplified Voice Message—Delete All Recording
2.5.6 Tool—Simplified Voice Message—Check Current Usage
3.5 [1-1] Slot—OPB3 Card Property—◆ SVM Feature
3.41 [1-1] Slot—OPB3 Card Property—◆ SVM Feature
4.5 [2-3] Timers & Counters—Miscellaneous
4.9 [2-6-1] Numbering Plan—Main—Features—Simplified Voice Message Access

5.1 [3-1-1] Trunk Group—TRG Settings—Tone Detection

6.9 [4-1-7] Wired Extension—Simplified Voice Message
6.10 [4-2-1] Portable Station—Extension Settings—Option 6—SVM Lock
6.16 [4-2-5] Portable Station—Simplified Voice Message
7.5 [5-3-3] Voice Message—SVM

PT Programming Manual References

None

Feature Guide References

1.9.5 Walking COS
1.17.5 Outgoing Message (OGM)
1.29.1 Dial Tone
3.1 Capacity of System Resources

User Manual References

1.6.2 Using Voice Messaging (Built-in Simplified Voice Message [SVM])
3.1.2 Settings on the Programming Mode
1.24.2 Voice Mail Features

1.24.2.1 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**

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>VM (DTMF) Group</td>
<td>A group of SLT ports which use the Voice Mail DTMF Integration features.</td>
</tr>
<tr>
<td></td>
<td>A maximum of 2 groups can be assigned.</td>
</tr>
<tr>
<td>VM (DPT) Group</td>
<td>A group of DPT ports which use the Voice Mail DPT (Digital) Integration features.</td>
</tr>
<tr>
<td></td>
<td>• A maximum of 2 KX-TVS/KX-TVP systems can be connected to the PBX.</td>
</tr>
<tr>
<td></td>
<td>• A maximum of one VM (DPT) group per KX-TVS/KX-TVP can be assigned.</td>
</tr>
<tr>
<td></td>
<td>• A maximum of 12 ports (24 channels) of the VPS can form each group.</td>
</tr>
</tbody>
</table>

[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]

<table>
<thead>
<tr>
<th>Slot</th>
<th>Port</th>
<th>Port Type</th>
<th>Extn. No.</th>
<th>DPT Property</th>
</tr>
</thead>
<tbody>
<tr>
<td>3</td>
<td>1</td>
<td>DPT</td>
<td>101</td>
<td>VM (DPT) 1</td>
</tr>
<tr>
<td>3</td>
<td>2</td>
<td>DPT</td>
<td>102</td>
<td>VM (DPT) 2</td>
</tr>
<tr>
<td>4</td>
<td>1</td>
<td>S-Hybrid</td>
<td>201</td>
<td>VM (DPT) 2</td>
</tr>
<tr>
<td>4</td>
<td>2</td>
<td>S-Hybrid</td>
<td>202</td>
<td>VM (DPT) 2</td>
</tr>
<tr>
<td>5</td>
<td>1</td>
<td>SLT</td>
<td>301</td>
<td>Not assignable</td>
</tr>
<tr>
<td>5</td>
<td>2</td>
<td>SLT</td>
<td>302</td>
<td>Not assignable</td>
</tr>
<tr>
<td>6</td>
<td>1</td>
<td>SLT</td>
<td>401</td>
<td>Not assignable</td>
</tr>
<tr>
<td>6</td>
<td>2</td>
<td>SLT</td>
<td>402</td>
<td>Not assignable</td>
</tr>
</tbody>
</table>

[Programming Example of VM (DTMF) Group]

<table>
<thead>
<tr>
<th>VM (DTMF) Group No.</th>
<th>Floating Extension No.</th>
<th>Group Name</th>
<th>Service Mode</th>
<th>Extension No. of Port Connected to VPS Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>300</td>
<td>Company C</td>
<td>AA</td>
<td>301 302 …</td>
</tr>
<tr>
<td>2</td>
<td>400</td>
<td>Company D</td>
<td>VM</td>
<td>401 402 …</td>
</tr>
</tbody>
</table>

→ 5.23 [3-8-2] VM(DTMF) Group—Group Settings

[Programming Example of VM (DPT) Group]

<table>
<thead>
<tr>
<th>VM (DPT) Group No.</th>
<th>Floating Extension No.</th>
<th>Group Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>500</td>
<td>Company A</td>
</tr>
<tr>
<td>2</td>
<td>200</td>
<td>Company B</td>
</tr>
</tbody>
</table>

→ 5.20 [3-7-2] VM(DPT) Group—Unit Settings—Floating Extension No.
→ [660] VM Group Floating Extension Number
→ 5.20 [3-7-2] 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 (→ 1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)) 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 destination, assigned on the trunk group which receives the call, by the Intercept Routing feature.
1.24.2 Voice Mail Features

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 Dialing button for direct access to a Voice Mail feature. (→
1.6.1.2 One-touch Dialing) For example, to access a mailbox (mailbox number 123) of the VPS (extension
number 165) directly, assign “165#6123” to a One-touch Dialing 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

3.21 [1-1] Slot—Port Property - Extension Port
→ DPT Type—Type
→ DPT Type—VM Unit No.
→ DPT Type—VM Port No.
5.19 [3-7-1] VM(DPT) Group—System Settings
5.20 [3-7-2] VM(DPT) Group—Unit Settings
5.22 [3-8-1] VM(DTMF) Group—System Settings
5.23 [3-8-2] VM(DTMF) Group—Group Settings

PT Programming Manual References

[601] Terminal Device Assignment
[660] VM Group Floating Extension Number

Feature Guide References

1.24.2.2 Voice Mail DTMF Integration
1.24.2.3 Voice Mail DPT (Digital) Integration
3.1 Capacity of System Resources
1.24.2.2 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.
   → 5.23 [3-8-2] VM(DTMF) Group—Group Settings—◆ Type

2. **DTMF Command Assignment**
   Assign the DTMF command to suit the VPS settings.
5.22 [3-8-1] VM(DTMF) Group—System Settings

→ ◆ VM DTMF Command—Recording Message
→ ◆ VM DTMF Command—Listening Message
→ ◆ VM DTMF Command—Switching to AA
→ ◆ VM DTMF Command—Switching to VM

<table>
<thead>
<tr>
<th>Command (Default)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Switching to VM</td>
</tr>
<tr>
<td>Switching to AA</td>
</tr>
<tr>
<td>Recording message</td>
</tr>
<tr>
<td>Listening message</td>
</tr>
</tbody>
</table>

**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.

→ 5.22 [3-8-1] VM(DTMF) Group—System Settings—◆ Others—FWD to the VPS Sequence

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Transmitted Command</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>In AA Service Mode</td>
</tr>
<tr>
<td>Answer by Mailbox</td>
<td>Switching to VM command + Recording message command (#6 + H [H = Mailbox No.])</td>
</tr>
<tr>
<td>(Default)</td>
<td></td>
</tr>
<tr>
<td>AA</td>
<td>Switching to AA command (#8)</td>
</tr>
<tr>
<td>None</td>
<td>(DTMF commands are not sent. Work with default of the VPS.)</td>
</tr>
</tbody>
</table>

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.

→ 5.22 [3-8-1] VM(DTMF) Group—System Settings—◆ Others—Intercept Routing to the VPS Sequence
### Transmitted Command

<table>
<thead>
<tr>
<th>Parameter</th>
<th>In AA Service Mode</th>
<th>In VM Service Mode</th>
</tr>
</thead>
<tbody>
<tr>
<td>Answer by Mailbox</td>
<td>Switching to VM command + Recording message command (#6 + H [H = Mailbox No.])</td>
<td>Recording message command (H [H = Mailbox No.])</td>
</tr>
<tr>
<td>AA</td>
<td>Switching to AA command (#8)</td>
<td>Switching to AA command (#8)</td>
</tr>
<tr>
<td>None (Default)</td>
<td>(DTMF commands are not sent. Work with default of the VPS.)</td>
<td></td>
</tr>
</tbody>
</table>

#### [Example of a) & b)]

![Diagram](image)

**c. Transferring to a Mailbox of the VPS**

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. (→ 1.12.1 Call Transfer) The extension user should use the Voice Mail (VM) Transfer button, when transferring a call to the mailbox of the VPS.

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.

6.5 [4-1-4] Wired Extension—Flexible Button

→ ◆ Type

→ ◆ Ext. No. / Floating Ext. No. (for Voice Mail Transfer)

6.13 [4-2-3] Portable Station—Flexible Button

→ ◆ Type

→ ◆ Ext. No. / Floating Ext. No. (for Voice Mail Transfer)

6.17 [4-3] DSS Console

→ ◆ Type

→ ◆ Ext. No. / Floating Ext. No. (for Voice Mail Transfer)
Performance of Pressing the VM Transfer Button and Entering an Extension Number

<table>
<thead>
<tr>
<th>Transmitted Command</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>In AA Service Mode</strong></td>
</tr>
<tr>
<td>Switching to VM command + Recording message command (#6 + H [H = Mailbox No.])</td>
</tr>
</tbody>
</table>

Example

Trunk Call

PBX

Extension (Operator)

Sent "+6 + 103": mailbox number

Switching to VM command

Transfer with VM Transfer button + extension number (103)

VPS (In AA service mode)

VM Port 1

VM (DTMF) Group

VM Port X

VM Port 2

VM Port 3

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 (→ 1.19.1 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).

→ 5.22 [3-8-1] 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

<table>
<thead>
<tr>
<th>Transmitted Command</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>In AA Service Mode</strong></td>
</tr>
<tr>
<td>Switching to VM command + Listening message command (#6 + *H [H = Mailbox No.])</td>
</tr>
</tbody>
</table>
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]

<table>
<thead>
<tr>
<th>Status</th>
<th>Condition</th>
<th>Default Command</th>
</tr>
</thead>
<tbody>
<tr>
<td>RBT (ringback tone)</td>
<td>The PBX is ringing the corresponding extension.</td>
<td>1</td>
</tr>
<tr>
<td>BT (busy tone)</td>
<td>The called extension is busy.</td>
<td>2</td>
</tr>
<tr>
<td>ROT (reorder tone)</td>
<td>The dialled number is invalid.</td>
<td>3</td>
</tr>
<tr>
<td>DND (DND tone)</td>
<td>The called extension has set DND. (→ 1.3.1.3 Do Not Disturb (DND))</td>
<td>4</td>
</tr>
<tr>
<td>Answer</td>
<td>The called extension has answered the call.</td>
<td>5</td>
</tr>
<tr>
<td>FWD VM RBT (FWD to Voice Mail ringback tone)</td>
<td>The called extension has set FWD to VPS and the PBX is calling another port of the VPS.</td>
<td>6</td>
</tr>
<tr>
<td>FWD VM BT (FWD to Voice Mail busy tone)</td>
<td>The called extension has set FWD to VPS and all ports of the VPS are busy.</td>
<td>7</td>
</tr>
<tr>
<td>FWD EXT RBT (FWD to extension ringback tone)</td>
<td>The PBX is calling an extension other than the one dialled. FWD or Idle Extension Hunting (→ 1.2.1 Idle Extension Hunting) may be assigned by the called extension.</td>
<td>8</td>
</tr>
<tr>
<td>Confirm (confirmation tone)</td>
<td>The PBX receives confirmation that the feature has been successfully set or cancelled (e.g., Message Waiting) on the extension.</td>
<td>9</td>
</tr>
<tr>
<td>Disconnect</td>
<td>The caller has hung up.</td>
<td>#9</td>
</tr>
</tbody>
</table>
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..."

The caller dials the extension number (extn. 102). The VPS will transfer the call to the extension via the PBX.

If the extension is not available, the PBX sends DTMF status signal of the extension (busy status) to the VPS.

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.
  
  → 5.22 [3-8-1] 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.
  
  → 5.22 [3-8-1] 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).
  

- The Inter-digit time for the DTMF command and for DTMF status signal is programmable.
  
  5.22 [3-8-1] VM(DTMF) Group—System Settings
  → Timing—Inter-digit Time

- Data Line Security is set automatically on the extensions in the VM (DTMF) group to achieve proper recording. (→ 1.11.5 Data Line Security)
PC Programming Manual References

4.5 [2-3] Timers & Counters—Miscellaneous—Voice Mail (Caller from VM to CO)—On-hook Wait Time
5.22 [3-8-1] VM(DTMF) Group—System Settings
5.23 [3-8-2] VM(DTMF) Group—Group Settings
6.5 [4-1-4] Wired Extension—Flexible Button
   →◆ Type
   →◆ Ext. No. / Floating Ext. No. (for Voice Mail Transfer)
6.10 [4-2-1] Portable Station—Extension Settings—Option 1—Programmed Mailbox No.
6.13 [4-2-3] Portable Station—Flexible Button
   →◆ Type
   →◆ Ext. No. / Floating Ext. No. (for Voice Mail Transfer)
6.17 [4-3] DSS Console
   →◆ Type
   →◆ Ext. No. / Floating Ext. No. (for Voice Mail Transfer)

PT Programming Manual References

None

Feature Guide References

1.1.1.6 Intercept Routing
1.3.1.2 Call Forwarding (FWD)
1.12.1 Call Transfer
1.20.2 Flexible Buttons
1.24.2.1 Voice Mail (VM) Group

User Manual References

1.10.5 If a Voice Processing System is Connected
1.24.2.3 Voice Mail DPT (Digital) Integration

Description
A Panasonic VPS that supports DPT (Digital) Integration (e.g., the KX-TVP200) 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. (→ 1.3.1.2 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. (→ 1.1.1.6 Intercept Routing) If the "Intercept to Mailbox" setting is enabled through system programming, the PBX sends the mailbox number of the intercepted extension to the VPS. Therefore the caller can leave a message in the mailbox of the extension without knowing the mailbox number.

4. **Transferring to a Mailbox of the VPS**
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. (→ 1.12.1 Call Transfer) The extension user should use the Voice Mail (VM) Transfer button, when transferring a call to the mailbox of the VPS. 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 number with the required command (after the VPS answers the call). Therefore the caller can leave a message without knowing the mailbox number.

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.

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. (→ 1.19.1 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 COS programming on the VPS.

7. **VPS Trunk Service & Automatic Time Mode Notification for Incoming Call**
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 (→ 2.2.4 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 (→ 1.1.1.2 Direct In Line (DIL), 1.30.1 TIE Line Service)

   12.2 [10-2] DIL Table & Port Settings—DIL
   → ◆ Tenant Number
   → ◆ VM Trunk Group No.

b. **DID/DDI**: the setting of each location number for DID/DDI (→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI))

   12.3 [10-3] DDI / DID Table
   → ◆ Tenant Number
   → ◆ VM Trunk Group No.

c. **MSN**: the setting of each MSN (→ 1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service)

   12.6 [10-4] MSN Table—MSN
   → ◆ 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]**

<table>
<thead>
<tr>
<th>Location No.</th>
<th>DID No.</th>
<th>DID Destination</th>
<th>DID Name</th>
<th>VM Trunk Group No.</th>
<th>Tenant No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>0001</td>
<td>123-4567</td>
<td>Day 105 Lunch 100 Break 105 Night 100</td>
<td>John White</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>0002</td>
<td>123-2468</td>
<td></td>
<td></td>
<td></td>
<td>2</td>
</tr>
</tbody>
</table>

**[VPS Programming—Programming Example of Trunk Group Assignment]**

<table>
<thead>
<tr>
<th>Trunk Group No.</th>
<th>Company Greeting No.</th>
<th>Incoming Call Service</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Day 1</td>
<td>Custom Service 11</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Lunch 2</td>
<td>Custom Service 29</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Break 3</td>
<td>Custom Service 31</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Night 4</td>
<td>Custom Service 12</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>Day 5</td>
<td>Custom Service 21</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Lunch 6</td>
<td>Custom Service 15</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Break 7</td>
<td>Custom Service 42</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Night 8</td>
<td>Custom Service 30</td>
<td></td>
</tr>
</tbody>
</table>
**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 or an MSN, the PBX sends the DDI/DID number or MSN 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. (→ 1.15.1 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.
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
The date and time settings of the VPS are controlled by the PBX.

15. Remote FWD Setting by the VM
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

[General]
• When programming a KX-TVM series VPS, if there is no option to connect to a KX-NCP500/KX-NCP1000 PBX, select the KX-TDA100/KX-TDA200 PBX.
• 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.

[LCS Recording Mode]

→ 6.1 [4-1-1] Wired Extension—Extension Settings—Option 4—LCS Answer Mode

→ 6.10 [4-2-1] Portable Station—Extension Settings—Option 4—LCS Recording Mode

[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. (→ 1.11.9 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. (→ 1.28.1 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]).
→ 6.1 [4-1-1] Wired Extension—Extension Settings—Option 4—LCS Recording Mode
→ 6.10 [4-2-1] 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. (→ 1.25.5 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.
[Transferring to a Mailbox of the VPS]

- When transferring a trunk call to a mailbox of the VPS, users should use the VM Transfer button.
- If using a standard Transfer button, users must dial "#6" after accessing the VPS, and confirm that the desired mailbox has been reached, before going on hook to complete the transfer.

PC Programming Manual References

5.19  [3-7-1] VM(DPT) Group—System Settings
5.20  [3-7-2] VM(DPT) Group—Unit Settings
6.1  [4-1-1] Wired Extension—Extension Settings—Option 4
   ◆ LCS Recording Mode
   ◆ LCS Answer Mode
6.5  [4-1-4] Wired Extension—Flexible Button
   ◆ Type
   ◆ Ext. No. / Floating Ext. No. (for Two-way Record)
   ◆ Ext. No. / Floating Ext. No. (for Two-way Transfer)
   ◆ Ext. No. / Floating Ext. No. (for Voice Mail Transfer)
   ◆ Ext. No. of Mailbox (for Two-way Transfer)
6.10 [4-2-1] Portable Station—Extension Settings—Option 4—◆ LCS Recording Mode
6.13 [4-2-3] Portable Station—Flexible Button
   ◆ Type
   ◆ Ext. No. / Floating Ext. No. (for Two-way Record)
   ◆ Ext. No. / Floating Ext. No. (for Two-way Transfer)
   ◆ Ext. No. / Floating Ext. No. (for Voice Mail Transfer)
   ◆ Ext. No. of Mailbox (for Two-way Transfer)
6.17 [4-3] DSS Console
   ◆ Type
   ◆ Ext. No. / Floating Ext. No. (for Two-way Record)
   ◆ Ext. No. / Floating Ext. No. (for Two-way Transfer)
   ◆ Ext. No. / Floating Ext. No. (for Voice Mail Transfer)
   ◆ Ext. No. of Mailbox (for Two-way Transfer)
7.1  [5-1] Doorphone—◆ VM Trunk Group Number
12.2 [10-2] DIL Table & Port Settings—DIL
   ◆ Tenant Number
   ◆ VM Trunk Group No.
12.3 [10-3] DDI / DID Table
   ◆ Tenant Number
   ◆ VM Trunk Group No.
12.6 [10-4] MSN Table—MSN
   ◆ Tenant Number
   ◆ VM Trunk Group No.

PT Programming Manual References

[201] Transfer Recall Time

Feature Guide References

1.20.2 Flexible Buttons
1.24.2.1 Voice Mail (VM) Group
2.3.6 Flexible Numbering/Fixed Numbering

User Manual References

1.10.5 If a Voice Processing System is Connected
3.1.2 Settings on the Programming Mode
3.2.2 Manager Programming
1.25 Portable Station (PS) Features

1.25.1 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

- 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) or Two-way Record is activated (→ 1.24.2.3 Voice Mail DPT (Digital) Integration).
- d. While the PS user is paging other extensions (→ 1.15.1 Paging).
- e. During a doorphone call (→ 1.17.1 Doorphone Call).
- f. While the PS user is dialling digits to make a trunk call.
- g. During a conversation with an extension using PC Console or PC Phone, to which the conversation is being recorded.
- h. While participating as a member of a broadcasting call (→ 1.16.1 Broadcasting).
- i. When a caller has dialled the extension number of a PS but the CS is busy, the caller hears a busy tone.
- j. A PS can only be assigned an extension number of 4 digits or less.
Installation Manual References
3.9 Connection of DECT Portable Stations
3.10 Connection of 2.4 GHz Portable Stations
3.11 Connection of DECT 6.0 Portable Stations

PC Programming Manual References
3.44 [1-2] Portable Station

PT Programming Manual References
[690] PS Registration
[691] PS Termination
[692] Personal Identification Number (PIN) for PS Registration

Feature Guide References
2.3.6 Flexible Numbering/Fixed Numbering
1.25.2 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]

<table>
<thead>
<tr>
<th>Floating Extn. No.</th>
<th>PS Ring Group 01</th>
<th>PS Ring Group 02</th>
<th>PS Ring Group 03</th>
<th>..</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group Name</td>
<td>Sales 1</td>
<td>Sales 2</td>
<td>Sales 3</td>
<td>..</td>
</tr>
<tr>
<td>Incoming Trunk Call Information Display</td>
<td>Called Party's Name/Number</td>
<td>Caller's Name/Number</td>
<td>Caller's Name/Number</td>
<td>..</td>
</tr>
<tr>
<td>PS01</td>
<td>✓</td>
<td></td>
<td></td>
<td>..</td>
</tr>
<tr>
<td>PS02</td>
<td>✓</td>
<td></td>
<td></td>
<td>..</td>
</tr>
<tr>
<td>PS03</td>
<td>✓</td>
<td></td>
<td></td>
<td>..</td>
</tr>
<tr>
<td>PS04</td>
<td>✓</td>
<td>✓</td>
<td></td>
<td>..</td>
</tr>
<tr>
<td>PS05</td>
<td></td>
<td>✓</td>
<td></td>
<td>..</td>
</tr>
<tr>
<td>PS06</td>
<td></td>
<td>✓</td>
<td></td>
<td>..</td>
</tr>
<tr>
<td>PS07</td>
<td></td>
<td></td>
<td>✓</td>
<td>..</td>
</tr>
</tbody>
</table>

✓: 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-TD7580
  - KX-TCA155
  - KX-TCA255
  - KX-TCA256
- KX-TCA355
- KX-TD7680
- KX-TD7684
- KX-TD7685
- KX-TD7690
- KX-TD7694
- KX-TD7695

• 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:

<table>
<thead>
<tr>
<th>Method</th>
<th>Assignment</th>
<th>Merit</th>
<th>Demerit</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Incoming Call Distribution Group</strong></td>
<td>Assign all desired PSs to one incoming call distribution group, and set the group call distribution method for the group to “Ring”.</td>
<td>All PS users in the group can use the Log-in/Log-out feature, Wrap-up feature, and ICD Group button for the group.</td>
<td>The CS may often be busy as each PS in the group uses one channel when a call arrives at the group.</td>
</tr>
<tr>
<td></td>
<td>→ 1.2.2 Incoming Call Distribution Group Features</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>PS Ring Group</strong></td>
<td>Assign all desired PSs to one PS ring group.</td>
<td>Only one channel is used when a call arrives at the group.</td>
<td>PS users in the group cannot use the Log-in/Log-out and Wrap-up features.</td>
</tr>
</tbody>
</table>

• 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). (→ 1.2.2.7 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. (→ 1.3.1.3 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
5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Main— Distribution Method
5.25 [3-9] PS Ring Group
5.26 [3-9] PS Ring Group—Member List

PT Programming Manual References
[620] Incoming Call Distribution Group Member
Feature Guide References

3.1 Capacity of System Resources
1.25.3 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:

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>PS Dialling Directory</td>
<td>Makes a call by selecting from a private directory of names and telephone numbers.</td>
</tr>
<tr>
<td>System Speed Dialling Directory</td>
<td>Makes a call by selecting from a common directory of names and numbers.</td>
</tr>
<tr>
<td>PBX Extension Dialling Directory</td>
<td>Makes a call by selecting from a common directory of extension names.</td>
</tr>
<tr>
<td>Shortcut Directory</td>
<td>Accesses a feature by selecting from a private directory of feature names and numbers.</td>
</tr>
<tr>
<td>Quick Dialling</td>
<td>Makes a call or accesses a feature easily by selecting from a private directory of names and numbers.</td>
</tr>
</tbody>
</table>

PC Programming Manual References

8.1 [6-1] System Speed Dial
   → ◆ Name
   → ◆ CO Line Access Number + Telephone Number
6.1 [4-1-1] Wired Extension—Extension Settings—Main—◆ Extension Name
6.10 [4-2-1] Portable Station—Extension Settings—Main—◆ Extension Name

PT Programming Manual References

[001] System Speed Dialling Number
[002] System Speed Dialling Name
[004] Extension Name
1.25.4 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 (→ 1.20.2 Flexible Buttons). Some special feature buttons (e.g., WAVESEARCH button) may be customised depending on the PS type.
1.25.5 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), as with XDP Parallel mode for a DPT and an SLT.

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:
  b. While Live Call Screening (LCS) or Two-way Record is activated (→ 1.24.2.3 Voice Mail DPT (Digital Integration)).
  c. While receiving OHCA (→ 1.8.4.3 Off-hook Call Announcement (OHCA)).
  d. While being monitored by another extension (→ 1.8.3 Call Monitor).
  e. During Consultation Hold.
  f. During a broadcasting call (→ 1.16.1 Broadcasting).
- 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 (→ 1.8.2 Executive Busy Override)
  - Whisper OHCA (→ 1.8.4.4 Whisper OHCA)
  - CCBS (→ 1.21.1.10 Completion of Calls to Busy Subscriber (CCBS))
- 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 (→ 1.1.3.2 Ring Tone Pattern Selection)
  - Preferred Line Assignment—Incoming (→ 1.4.1.2 Line Preference—Incoming)
  - Preferred Line Assignment—Outgoing (→ 1.5.5.2 Line Preference—Outgoing)
  - Hot Line Setting (→ 1.6.1.7 Hot Line)
Transfer Recall Destination for Call Transfer and Call Park (→ 1.12.1 Call Transfer) (→ 1.13.2 Call Park)

Display Language (→ 1.20.4 Display Information)

ISDN Bearer Mode (→ 1.21.1.1 Integrated Services Digital Network (ISDN)—SUMMARY)

Flexible Button Assignment (→ 1.20.2 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 (→ 1.1.3.3 Call Waiting)
  – FWD/DND Setting (→ 1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND))
  – Call Pickup Deny Setting (→ 1.4.1.3 Call Pickup)
  – Executive Busy Override Deny Setting (→ 1.8.2 Executive Busy Override)
  – Itemised Billing Code for ARS (→ 1.10.1 Automatic Route Selection (ARS))
  – Transfer Recall Destination for Call Transfer and Call Park (→ 1.12.1 Call Transfer) (→ 1.13.2 Call Park)
  – CLIP/COLP Number and CLIP/COLP Number Selection (→ 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP))
  – CLIR and COLR Setting (→ 1.21.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP))
  – Extension Personal Identification Number (PIN) (→ 1.28.1 Extension Personal Identification Number (PIN))
  – COS Programming (→ 2.2.1 Class of Service (COS))
  – Extension User Group (→ 2.2.2 Group)

• If Extension Feature Clear is performed, the corresponding extension data for both the wired telephone and the PS will be cleared. (→ 1.28.2 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 (→ 1.8.1 Automatic Callback Busy (Camp-on))
  – Transfer Recall (→ 1.12.1 Call Transfer), Hold Recall (→ 1.13.1 Call Hold), and Call Park Recall (→ 1.13.2 Call Park)

PC Programming Manual References

3.21 [1-1] Slot—Port Property - Extension Port—◆ DPT Type—Type
3.44 [1-2] Portable Station—PS Registration and De-registration
4.9 [2-6-1] Numbering Plan—Main—Features—◆ Wireless XDP Parallel Mode Set / Cancel
4.12 [2-7-1] Class of Service—COS Settings—Optional Device & Other Extensions—◆ Accept Wireless XDP Parallel Mode Set by PS
6.1 [4-1-1] Wired Extension—Extension Settings—Option 1—◆ Wireless XDP

PT Programming Manual References

[515] Wireless XDP Parallel Mode for Paired Telephone
Feature Guide References

1.11.9 Parallelled Telephone
2.2.1 Class of Service (COS)

User Manual References

1.8.13 Using Your PS in Parallel with a Wired Telephone (Wireless XDP Parallel Mode)
1.25.6 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]**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Outside Destinations in Incoming Call Distribution Group</td>
<td>A virtual PS allows calls to an Incoming Call Distribution (ICD) Group to be answered by outside destinations or extensions at another PBX.</td>
</tr>
<tr>
<td>Network ICD Group</td>
<td>Using virtual PSs in an ICD Group, up to 4 other PBXs can be called at the same time.</td>
</tr>
<tr>
<td>PS Roaming by Network ICD Group</td>
<td>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.</td>
</tr>
</tbody>
</table>
### Feature Guide References

- **Automatic Fax Transfer**
  - A virtual PS can be used to forward fax calls to a fax machine at another PBX connected by TIE line.
  - → 1.17.7 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.
  - → 1.24.1 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

3.44 [1-2] Portable Station

#### PT Programming Manual References

[690] PS Registration

#### Feature Guide References

1.3.1.2 Call Forwarding (FWD)
1.26 Administrative Information Features

1.26.1 Record Log Features

1.26.1.1 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:

<table>
<thead>
<tr>
<th>Output Method</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Serial Interface (RS-232C) port</td>
<td>SMDR information is sent to a connected PC, printer, etc.</td>
</tr>
<tr>
<td>Telnet compatible terminal emulator</td>
<td>SMDR information is sent to a Telnet compatible terminal emulator via LAN.</td>
</tr>
</tbody>
</table>

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 (→ 2.4.3 Local Alarm Information)
   e. Hospitality feature information (→ 1.27.1 Hospitality Features—SUMMARY)
   f. Printing Message information (→ 1.26.2 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

<table>
<thead>
<tr>
<th>Date</th>
<th>Time</th>
<th>Ext</th>
<th>CO</th>
<th>Dial Number</th>
<th>Ring</th>
<th>Duration</th>
<th>ACC Code</th>
<th>CD</th>
</tr>
</thead>
<tbody>
<tr>
<td>01/02/10:03AM</td>
<td>1200 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>5'15</td>
<td>00:01:05</td>
<td>9876543210</td>
<td>NA</td>
</tr>
<tr>
<td>01/02/10:07AM</td>
<td>1200 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>0'05</td>
<td>00:01:05</td>
<td>9876543210</td>
<td>TR</td>
</tr>
<tr>
<td>01/02/10:15AM</td>
<td>1200 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>00:01:05</td>
<td>00:01:05</td>
<td>9876543210</td>
<td>TR</td>
</tr>
<tr>
<td>01/02/10:30AM</td>
<td>*123 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>00:01:05</td>
<td>00:01:05</td>
<td>9876543210</td>
<td>TR</td>
</tr>
</tbody>
</table>

Pattern B: 80 digits with call charge information

<table>
<thead>
<tr>
<th>Date</th>
<th>Time</th>
<th>Ext</th>
<th>CO</th>
<th>Dial Number</th>
<th>Duration</th>
<th>Cost</th>
<th>ACC Code</th>
<th>CD</th>
</tr>
</thead>
<tbody>
<tr>
<td>01/02/10:07PM</td>
<td>1234 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>00:01:05</td>
<td>00:01:05</td>
<td>9876543210</td>
<td>TR</td>
</tr>
<tr>
<td>01/02/10:07PM</td>
<td>1234 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>00:01:05</td>
<td>00:01:05</td>
<td>9876543210</td>
<td>TR</td>
</tr>
<tr>
<td>01/02/10:07PM</td>
<td>1234 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>00:01:05</td>
<td>00:01:05</td>
<td>9876543210</td>
<td>TR</td>
</tr>
<tr>
<td>01/02/10:07PM</td>
<td>1234 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>00:01:05</td>
<td>00:01:05</td>
<td>9876543210</td>
<td>TR</td>
</tr>
</tbody>
</table>

Pattern C: 120 digits

<table>
<thead>
<tr>
<th>Date</th>
<th>Time</th>
<th>Ext</th>
<th>CO</th>
<th>Dial Number</th>
<th>Ring</th>
<th>Duration</th>
<th>Cost</th>
<th>ACC Code</th>
<th>CD</th>
</tr>
</thead>
<tbody>
<tr>
<td>01/02/10:03AM</td>
<td>1210 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>00:01:05</td>
<td>00:01:05</td>
<td>00560.00EU</td>
<td>9876543210</td>
<td>TR</td>
</tr>
<tr>
<td>01/02/10:07AM</td>
<td>2005 01</td>
<td>^7</td>
<td>2</td>
<td>12345678901234567890</td>
<td>00:01:05</td>
<td>00:01:05</td>
<td>00560.00EU</td>
<td>9876543210</td>
<td>TR</td>
</tr>
</tbody>
</table>

[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].
<table>
<thead>
<tr>
<th>Number in the Pattern</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>(1)</td>
<td>Date</td>
<td>Shows the date of the call.</td>
</tr>
<tr>
<td>(2)</td>
<td>Time</td>
<td>Shows the end time of a call as Hour/Minute/AM or PM.</td>
</tr>
<tr>
<td>(3)</td>
<td>Ext (Extension)</td>
<td>Shows the extension number, floating extension number, etc., which was engaged in the call.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Also shows the following codes:</td>
</tr>
<tr>
<td></td>
<td></td>
<td><strong>Dxxx</strong>: Outgoing trunk call from a doorphone (xxx=doorphone number) (→ 1.17.1 Doorphone Call)</td>
</tr>
<tr>
<td></td>
<td></td>
<td><strong>Txxx</strong>: Outgoing trunk call by TIE line service (xxx=trunk group number)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>*<strong>xxx</strong>: Verified call (xxx=verification code) (→ 1.9.6 Verification Code Entry)</td>
</tr>
<tr>
<td>(4)</td>
<td>CO (Trunk)</td>
<td>Shows the trunk number used for the call. For patterns A and B, &quot;00&quot; will be shown for trunk numbers over hundred.}</td>
</tr>
<tr>
<td>Number in the Pattern</td>
<td>Item</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>-----------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| (5)                   | Dial Number                 | **[Trunk Call]**<br>Outgoing Trunk Call<br>Shows the dialled telephone number.<br>Valid digits are as follows:<br>0 through 9, *, #<br>P: Pause<br>F: EFA signal<br>=: A Host PBX Access code (→ 1.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX))<br>. (dot): Secret dialling<br>X: Privacy dial<br>→: Transferred call<br>If the transfer destination extension enters some digits, the entered digits will be added after “→”.
-Incoming Trunk Call<br>Shows <I> + the caller’s identification name/number.<br>It is also possible to show the DDI/DID/MSN call information. In this case, <D> + DDI/DID/MSN name/number is added before <I>.

**[Outgoing Intercom Call]**<br>Shows the dialled extension number followed by "EXT".

**[Log-in/Log-out]**<br>Shows the log-in or log-out status.

**[Check-in/Check-out]**<br>Shows the check-in or check-out status. (→ 1.27.2 Room Status Control)

**[Timed Reminder]**<br>Shows the status of a timed reminder, either "Start", "No Answer", or "Answer". (→ 1.28.4 Timed Reminder)

**[Printing Message]**<br>Shows the selected message. (→ 1.26.2 Printing Message)

**[Sensor Call]**<br>Shows calls from an external sensor as follows:<br><I> S + sensor number. (→ 1.17.8 External Sensor)

<table>
<thead>
<tr>
<th>(6)</th>
<th>Ring</th>
<th>Shows the ring duration before answering a call in Minutes/Seconds.</th>
</tr>
</thead>
<tbody>
<tr>
<td>(7)</td>
<td>Duration</td>
<td>Shows the duration of the trunk call in Hours/Minutes/Seconds.</td>
</tr>
<tr>
<td>(8)</td>
<td>Acc Code (Account Code)</td>
<td>Shows the account code appended to the call. (→ 1.5.4.3 Account Code Entry)</td>
</tr>
</tbody>
</table>
### Number in the Pattern

<table>
<thead>
<tr>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
</table>
| (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) (→ 2.3.1 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

<table>
<thead>
<tr>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
</table>
| Outgoing trunk call | Controls whether the outgoing trunk calls are shown. This setting is common throughout the PBX. COS programming is also required.  
  → [804] SMDR Outgoing Call Printing |
| Incoming trunk call | Controls whether the incoming trunk calls are shown.  
  → [805] SMDR Incoming Call Printing |
| Outgoing intercom call | Controls whether the outgoing intercom calls are recorded.  
| Log-in/Log-out status | Controls whether the log-in/log-out status is recorded.  
| 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. (→ 1.10.1 Automatic Route Selection (ARS))  
| 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.  
  → 13.1 [11-1] Main—SMDR Options—Option—Caller ID Number & Name |
<table>
<thead>
<tr>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DID/DDI number</td>
<td>Controls whether the DID/DDI number, name, number and name, or nothing is shown. If &quot;none&quot; is selected, &lt;D&gt; will not be shown. → 13.1 [11-1] Main—SMDR Options—◆ Option—DDI/DDI Number &amp; Name</td>
</tr>
<tr>
<td>Secret dialling</td>
<td>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. → 13.1 [11-1] Main—SMDR Options—◆ Option—Secret Dial</td>
</tr>
<tr>
<td>Privacy dial</td>
<td>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 &quot;X&quot;. (e.g., 123-456-XXXX) → 13.1 [11-1] Main—SMDR Options—◆ Option—Privacy Mode</td>
</tr>
<tr>
<td>Date order</td>
<td>The date order is changeable: month/day/year, day/month/year, year/month/day, year/day/month. → 13.1 [11-1] Main—SMDR—◆ SMDR Format—Date Format</td>
</tr>
<tr>
<td>Received call</td>
<td>Controls whether the time of receiving an incoming trunk call is shown. → 13.1 [11-1] Main—SMDR Options—◆ Option—Condition Code &quot;RC&quot;</td>
</tr>
<tr>
<td>Answered call</td>
<td>Controls whether the time of answering an incoming trunk call is shown. → 13.1 [11-1] Main—SMDR Options—◆ Option—Condition Code &quot;AN&quot;</td>
</tr>
<tr>
<td>Room status</td>
<td>Controls whether room status changes are shown. → 8.2 [6-2] Hotel &amp; Charge—Main—◆ SMDR for External Hotel Application 1—Room Status Control</td>
</tr>
<tr>
<td>Timed Reminder call</td>
<td>Controls whether Timed Reminder calls are shown (→ 1.28.4 Timed Reminder). → 8.2 [6-2] Hotel &amp; Charge—Main—◆ SMDR for External Hotel Application 1—Timed Reminder (Wake-up Call)</td>
</tr>
<tr>
<td>Printing Message</td>
<td>Specifies the messages that can be selected from an extension (→ 1.26.2 Printing Message). → 8.2 [6-2] Hotel &amp; Charge—Main—◆ SMDR for External Hotel Application 2—Printing Message 1–8</td>
</tr>
<tr>
<td>Time format</td>
<td>Controls whether time is displayed in 12-hour or 24-hour format. → 13.1 [11-1] Main—SMDR—◆ SMDR Format—Time Format (12H / 24H)</td>
</tr>
</tbody>
</table>

**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.
   - → [802] SMDR Page Length

b. **Skip Perforation:** determines the number of lines to be skipped at the end of every page.
   - → [803] SMDR Skip Perforation

The page length should be at least four lines longer than the skip perforation length.

**Explanation:**

![Diagram](image)

- 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.
- 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")

  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.

  → [208] Call Duration Count Starting Time for LCOT

  If the reverse signal detection has been set (→ 1.5.4.5 Reverse Circuit), the PBX will start counting the call after detecting the reverse signal from the telephone company regardless of the above timer.

- **Serial Interface (RS-232C) Parameters**

  The following communication parameters can be assigned for the Serial Interface (RS-232C) port.

  a. **New Line Code:** Select the code for the PC or printer. If the PC or printer automatically feeds lines with carriage return, select "CR". If not, select "CR+LF".


  b. **Baud Rate:** A baud rate code indicates the data transmission speed from the PBX to the PC or printer.

     → [800] RS-232C Parameter—Baud Rate

  c. **Word Length:** A word length code indicates how many bits compose a character.

     → [800] RS-232C Parameter—Word Length
d. **Parity Bit:** A parity code indicates what type of parity is used to detect an error in the string of bits composing a character. Make an appropriate selection depending on the requirements of the PC or printer.

   → [800] RS-232C Parameter—Parity Bit

e. **Stop Bit Length:** A stop bit code indicates the end of a bit string which composes a character. Select an appropriate value depending on the requirements of the PC or printer.

   → [800] RS-232C Parameter—Stop Bit Length

   • If a call is transferred to an ICD group using Automatic Transfer, the condition code "TR" will not be recorded on SMDR (→ 1.12.1 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 IPCMPR card, 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 abnormally, it can take up to 7 minutes for the PBX to detect the termination. 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.

### Installation Manual References

3.13.1 Connection of Peripherals

### PC Programming Manual References


4.12 [2-7-1] Class of Service—COS Settings—CO & SMDR— ◆ Outgoing CO Call Printout (SMDR)

8.2 [6-2] Hotel & Charge
   → Main— ◆ SMDR for External Hotel Application 2—Printing Message 1–8
   → Charge— ◆ Charge Options—Currency

   → SMDR
   → SMDR Options
   → RS-232C
PT Programming Manual References

[800] RS-232C Parameter—Baud Rate
[800] RS-232C Parameter—Word Length
[800] RS-232C Parameter—Parity Bit
[800] RS-232C Parameter—Stop Bit Length
[802] SMDR Page Length
[803] SMDR Skip Perforation
[804] SMDR Outgoing Call Printing
[805] SMDR Incoming Call Printing

Feature Guide References

1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)
1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service
1.2.2.7 Log-in/Log-out
1.5.4.8 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)
1.12.1 Call Transfer
1.17.6 Direct Inward System Access (DISA)
1.30.1 TIE Line Service
2.2.1 Class of Service (COS)
3.1 Capacity of System Resources
1.26.1.2 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, register the IP address of the Syslog server, and to select whether major/minor are sent.

PC Programming Manual References


PT Programming Manual References

None

Feature Guide References

2.4.3 Local Alarm Information
1.26.2 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
4.9 [2-6-1] Numbering Plan—Main—Features— Printing Message
8.2 [6-2] Hotel & Charge—Main— SMDR for External Hotel Application 2—Printing Message 1–8

PT Programming Manual References
None

Feature Guide References
1.26.1.1 Station Message Detail Recording (SMDR)
3.1 Capacity of System Resources

User Manual References
1.11.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)
1.26.3 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:

<table>
<thead>
<tr>
<th>Trunk</th>
<th>Service</th>
</tr>
</thead>
<tbody>
<tr>
<td>Analogue trunk</td>
<td>Pay Tone*1</td>
</tr>
<tr>
<td>ISDN line</td>
<td>Advice of Charge (AOC) (→ 1.21.1.3 Advice of Charge (AOC))</td>
</tr>
<tr>
<td>E1 line</td>
<td>Meter Pulse</td>
</tr>
</tbody>
</table>

*1 When the pay tone service is used, pay tone type (12 kHz/16 kHz) should be selected.

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 (→ 1.9.2 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
******************************************************
**CO Line**
**001:** 00194.00  **002:** 00073.00  **003:** 00161.00  **004:** 00033.00
Total Charge: 00175.95
```

```
******************************************************
* Charge Meter Print Out - All Extensions
******************************************************
**775:** 00194.00  **102:** 00073.00  **776:** 00161.00  **107:** 00033.00
**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.
- It is possible to enable the PBX to send the flash/recall signal to the telephone company after a conversation (in on-hook status) to receive the call charge information.

PC Programming Manual References

3.24 [1-1] Slot—Card Property - LCO type
  ➔ Pay Tone—Pay Tone Frequency
  ➔ Pay Tone—Sending Flash while end talk
1.26.3 Call Charge Services

5.6  [3-1-5] Trunk Group—Charge Rate
6.5  [4-1-4] Wired Extension—Flexible Button—◆ Type
6.13 [4-2-3] Portable Station—Flexible Button—◆ Type
6.17 [4-3] DSS Console—◆ Type
8.2  [6-2] 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
[491] Pay Tone Signal Type

Feature Guide References
1.9.6 Verification Code Entry
1.20.2 Flexible Buttons
3.1 Capacity of System Resources

User Manual References
1.11.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)
3.2.2 Manager Programming
1.27 Hospitality Features

1.27.1 Hospitality Features—SUMMARY

Description

This PBX has several features that support its use in a hotel-type environment, where extensions correspond to guest rooms.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Room Status Control</td>
<td>An extension designated as the hotel operator can set the check-in status of rooms remotely.</td>
</tr>
<tr>
<td>Call Billing for Guest Room</td>
<td>Charges for calls from guest rooms can be logged and output as a guest bill.</td>
</tr>
<tr>
<td>Remote Wake-up Call</td>
<td>An extension designated as the hotel operator can set a timed reminder for a room remotely.</td>
</tr>
<tr>
<td>SMDR for External Hotel Application</td>
<td>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.</td>
</tr>
</tbody>
</table>
1.27.2 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 three 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:

<table>
<thead>
<tr>
<th>Light Pattern</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td>Checked-out and Ready</td>
</tr>
<tr>
<td>Flashing Red</td>
<td>Checked-out and Not Ready</td>
</tr>
<tr>
<td>Red on</td>
<td>Checked-in</td>
</tr>
</tbody>
</table>
[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 (including IP-PT), SLT, ISDN Extension, T1-OPX
- 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

6.5 [4-1-4] Wired Extension—Flexible Button—◆ Type
6.17 [4-3] DSS Console—◆ Type
8.2 [6-2] Hotel & Charge
   →Main—◆ Hotel Operator—Extension 1–4
   →Bill—◆ Checkout Billing—Billing for Guest

PT Programming Manual References

None

Feature Guide References

1.6.1.4 Last Number Redial
1.9.3 Extension Dial Lock
1.26.1.1 Station Message Detail Recording (SMDR)
1.28.4 Timed Reminder
User Manual References

1.11.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)
1.27.3 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.
  
  8.2 [6-2] 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.
  
  8.2 [6-2] Hotel & Charge—Bill
  → ◆ Checkout Billing—LCD for "Telephone"
  → ◆ Checkout Billing—LCD for "Minibar"
  → ◆ Checkout Billing—LCD for "Others"

- A tax rate.
  
  8.2 [6-2] 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.

→ 8.2 [6-2] 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

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>(1)</strong></td>
<td><strong>Hotel</strong></td>
</tr>
<tr>
<td><strong>(2)</strong></td>
<td><strong>Check in:</strong> 01.JAN.00 06:31PM</td>
</tr>
<tr>
<td><strong>(3)</strong></td>
<td><strong>Check out:</strong> 03.JAN.00 07:03AM</td>
</tr>
<tr>
<td><strong>(4)</strong></td>
<td><strong>Room:</strong> 202 : Mr. Smith</td>
</tr>
<tr>
<td><strong>(5)</strong></td>
<td><strong>Telephone:</strong> 104.30 (Tax 10.000% = 9.48)</td>
</tr>
<tr>
<td></td>
<td><strong>Minibar:</strong> 4.00 (Tax 10.000% = 0.36)</td>
</tr>
<tr>
<td></td>
<td><strong>Others:</strong> 0.00 (Tax 15.000% = 0.00)</td>
</tr>
<tr>
<td><strong>(6)</strong></td>
<td><strong>Total:</strong> FR 108.30 (Tax Total = 9.84)</td>
</tr>
<tr>
<td><strong>(7)</strong></td>
<td><strong>Sheet:</strong> 002</td>
</tr>
<tr>
<td><strong>(8)</strong></td>
<td><strong>Telephone:</strong> +41 3 12 34 56 78</td>
</tr>
</tbody>
</table>

1. A programmable title (e.g., hotel name).
2. The check-in time.
3. The check-out time.
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 [→ 1.26.1.1 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).

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 (→ 1.9.5 Walking COS).

→ 6.1 [4-1-1] 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

6.1 [4-1-1] Wired Extension—Extension Settings—Main—◆ Extension PIN
8.2 [6-2] Hotel & Charge—Bill—◆ Checkout Billing—LCD for "Telephone"
8.2 [6-2] 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" (%)

PT Programming Manual References

None

Feature Guide References

1.6.1.4 Last Number Redial
1.9.3 Extension Dial Lock
1.26.1.1 Station Message Detail Recording (SMDR)
1.28.4 Timed Reminder
3.1 Capacity of System Resources

User Manual References

1.11.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)
1.28 Extension Controlling Features

1.28.1 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)\(^1\) (→ 1.24.2.3 Voice Mail DPT (Digital) Integration)
b. Display Lock (→ 1.6.1.5 Speed Dialling—Personal/System, 1.24.1 Built-in Simplified Voice Message (SVM), 1.18.2 Incoming Call Log)
c. Walking Extension (→ 1.28.3 Walking Extension Features)
d. Extension Dial Lock (→ 1.9.3 Extension Dial Lock)
e. Walking COS (→ 1.9.5 Walking COS)
f. Walking COS through DISA (→ 1.17.6 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

4.9 [2-6-1] Numbering Plan—Main—Features—◆ Extension PIN Set / Cancel
4.18 [2-9] System Options—Option 1—◆ PT LCD—Password / PIN Display
6.1 [4-1-1] Wired Extension—Extension Settings—Main—◆ Extension PIN
6.10 [4-2-1] 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
3.2.2 Manager Programming
1.28.2 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.

<table>
<thead>
<tr>
<th>Features</th>
<th>After Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Absent Message</td>
<td>Off</td>
</tr>
<tr>
<td>BGM</td>
<td>Off</td>
</tr>
<tr>
<td>FWD*/DND*</td>
<td>Off</td>
</tr>
<tr>
<td>Call Pickup Deny</td>
<td>Allow</td>
</tr>
<tr>
<td>Call Waiting*</td>
<td>Disable (In Canada, the default setting is &quot;Enable&quot; [Call Waiting tone].)</td>
</tr>
<tr>
<td>Data Line Security</td>
<td>Off</td>
</tr>
<tr>
<td>Executive Busy Override Deny</td>
<td>Allow</td>
</tr>
<tr>
<td>Log-in/Log-out</td>
<td>Log-in</td>
</tr>
<tr>
<td>Message Waiting</td>
<td>All messages left by other extensions will be cleared.</td>
</tr>
<tr>
<td>Paging Deny</td>
<td>Allow</td>
</tr>
<tr>
<td>Paralleled Telephone</td>
<td>Paired SLT will ring.</td>
</tr>
<tr>
<td>Hot Line*</td>
<td>Off</td>
</tr>
<tr>
<td>Timed Reminder</td>
<td>Cleared</td>
</tr>
</tbody>
</table>

Note
The features with "*" can be programmed not to be cancelled by this feature.

Conditions
- Extension Dial Lock (→ 1.9.3 Extension Dial Lock) and the extension personal identification number (PIN) (→ 1.28.1 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. (→ 1.29.1 Dial Tone)

PC Programming Manual References
4.9 [2-6-1] Numbering Plan—Main—Features—◆ Extension Feature Clear
4.18 [2-9] System Options—Option 2
  →◆ Extension Clear—Call Waiting
  →◆ Extension Clear—Fwd/DND
  →◆ Extension Clear—Hot Line (Pickup Dial)
PT Programming Manual References

None

User Manual References

1.8.14 Clearing Features Set at Your Extension (Extension Feature Clear)
1.28.3 Walking Extension Features

1.28.3.1 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 (including IP-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. (→ 1.28.1 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 (→ 2.3.1 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.

PC Programming Manual References

4.9 [2-6-1] Numbering Plan—Main—Features—Walking Extension
6.1 [4-1-1] Wired Extension—Extension Settings—Main—Extension PIN
6.17 [4-3] DSS Console—Pair Extension

PT Programming Manual References

[007] DSS Console Paired Telephone

Feature Guide References

1.31.1 IP Proprietary Telephone (IP-PT)

User Manual References

1.12.1 Walking Extension
1.28.3.2 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:

![Diagram showing extension settings]

**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, T1 OPX extensions, and SIP extensions.
- An extension personal identification number (PIN) is required to use this feature. (→ 1.28.1 Extension Personal Identification Number (PIN))

**PC Programming Manual References**
4.9 [2-6-1] Numbering Plan—Main—Features—◆ Walking Extension

**PT Programming Manual References**
None

**Feature Guide References**
1.28.3.1 Walking Extension

**User Manual References**
1.12.2 Enhanced Walking Extension
1.28.4 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 (→ 1.17.5 Outgoing Message (OGM)). A different message can be assigned for each time mode (day/lunch/break/night) (→ 2.2.4 Time Service).

PC Programming Manual References

  → ◆ Timed Reminder—Repeat Counter
  → ◆ Timed Reminder—Interval Time
  → ◆ Timed Reminder—Alarm Ringing Duration
4.9 [2-6-1] Numbering Plan—Main—Features
  → ◆ Remote Timed Reminder (Remote Wakeup Call)
  → ◆ Timed Reminder Set / Cancel
4.17 [2-8-3] Ring Tone Patterns—Call from Others— ◆ Timed Reminder—Ring Tone Pattern Plan 1–8
4.18 [2-9] System Options—Option 1— ◆ PT LCD—Time Display
8.2 [6-2] Hotel & Charge—Main— ◆ SMDR for External Hotel Application 1—Timed Reminder (Wake-up Call)

PT Programming Manual References

None

Feature Guide References

1.27.2 Room Status Control

User Manual References

1.8.1 Setting the Alarm (Timed Reminder)
1.28.4 Timed Reminder

1.11.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)
1.29 Audible Tone Features

1.29.1 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).

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tone 1A/1B</td>
<td>A normal dial tone is heard when:</td>
</tr>
<tr>
<td></td>
<td>a. No features listed for dial tones 2 through 4 has been set, or</td>
</tr>
<tr>
<td></td>
<td>b. ARS is used.</td>
</tr>
<tr>
<td>Tone 2A/2B</td>
<td>This tone is heard when:</td>
</tr>
<tr>
<td></td>
<td>• There are messages that have previously been listened to and no new messages for the Built-in Simplified Voice Message (SVM) feature.¹</td>
</tr>
<tr>
<td></td>
<td>• Any of the features below are set.</td>
</tr>
<tr>
<td></td>
<td>• Absent Message</td>
</tr>
<tr>
<td></td>
<td>• BGM</td>
</tr>
<tr>
<td></td>
<td>• FWD</td>
</tr>
<tr>
<td></td>
<td>• Call Pickup Deny</td>
</tr>
<tr>
<td></td>
<td>• Call Waiting</td>
</tr>
<tr>
<td></td>
<td>• DND</td>
</tr>
<tr>
<td></td>
<td>• Extension Dial Lock</td>
</tr>
<tr>
<td></td>
<td>• Executive Busy Override Deny</td>
</tr>
<tr>
<td></td>
<td>• Hot Line</td>
</tr>
<tr>
<td></td>
<td>• Timed Reminder</td>
</tr>
<tr>
<td>Tone 3A/3B</td>
<td>This tone is heard when:</td>
</tr>
<tr>
<td></td>
<td>• A called PS is being searched for.</td>
</tr>
<tr>
<td></td>
<td>• The recording time used by the Built-in Simplified Voice Message (SVM) feature reaches the limit.¹</td>
</tr>
<tr>
<td></td>
<td>• Any of the features below are performed.</td>
</tr>
<tr>
<td></td>
<td>• Account Code Entry</td>
</tr>
<tr>
<td></td>
<td>• Consultation Hold</td>
</tr>
<tr>
<td></td>
<td>• Answering a Timed Reminder call with no message</td>
</tr>
<tr>
<td></td>
<td>• Answering a sensor call</td>
</tr>
<tr>
<td>Tone 4A/4B</td>
<td>This tone is heard when new messages have been recorded for the extension.</td>
</tr>
</tbody>
</table>

¹ 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.

- Distinctive dial tones can be disabled. When disabled, dial tone 1 will be heard in all cases except those marked with "*1" in the table above.

- **Dial Tone Patterns**
  All dial tone patterns have a default (→ 3.2.1 Tones/Ring Tones).

- Only dial tone 1 is sent to the extensions in the VM (DPT/DTMF) group. (→ 1.24.2.1 Voice Mail (VM) Group)

### PC Programming Manual References

4.18 [2-9] System Options—Option 3
- ➔ Dial Tone—Distinctive Dial Tone
- ➔ Dial Tone—Dial Tone for Extension
- ➔ Dial Tone—Dial Tone for ARS

### PT Programming Manual References

None
1.29.2 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.

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tone 1</td>
<td>a. Sent when the setting is accepted.</td>
</tr>
<tr>
<td></td>
<td>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.</td>
</tr>
<tr>
<td>Tone 2</td>
<td>a. Sent from an external paging device or an extension before being paged.</td>
</tr>
<tr>
<td></td>
<td>b. Sent when a call is received in Hands-free Answerback mode.</td>
</tr>
<tr>
<td>Tone 3-1</td>
<td>a. Sent before a conversation is established when using the Paging feature.</td>
</tr>
<tr>
<td></td>
<td>b. Sent when a conversation is established with the extension in the following modes after the call making operation:</td>
</tr>
<tr>
<td></td>
<td>• Hands-free Answerback mode</td>
</tr>
<tr>
<td></td>
<td>• Voice-calling mode (Alternate Receiving—Ring/Voice)</td>
</tr>
<tr>
<td></td>
<td>c. Sent when making a call to or from a doorphone.</td>
</tr>
<tr>
<td>Tone 3-2</td>
<td>Sent just before a conversation is established when accessing the following features by the feature numbers:</td>
</tr>
<tr>
<td></td>
<td>• Call Park Retrieve</td>
</tr>
<tr>
<td></td>
<td>• Call Pickup</td>
</tr>
<tr>
<td></td>
<td>• Hold Retrieve</td>
</tr>
<tr>
<td></td>
<td>• Paging Answer</td>
</tr>
<tr>
<td></td>
<td>• TAFAS</td>
</tr>
<tr>
<td>Tone 4-1</td>
<td>Sent when moving from a two-party call to a three-party call. (e.g., Executive Busy Override, Conference, Privacy Release, Two-way Record.)</td>
</tr>
<tr>
<td>Tone 4-2</td>
<td>Sent when moving from a three-party call to a two-party call. (e.g., Executive Busy Override, Conference, Privacy Release, Two-way Record.)</td>
</tr>
<tr>
<td>Tone 5</td>
<td>Sent when a call is placed on hold (including Consultation Hold).</td>
</tr>
</tbody>
</table>

**Conditions**

- **Confirmation Tone Patterns**
  
  All confirmation tone patterns have a default (→ 3.2.1 Tones/Ring Tones).

- It is possible to eliminate each tone.
1.29.2 Confirmation Tone

PC Programming Manual References

4.18 [2-9] 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

PT Programming Manual References

None
1.30 Networking Features

1.30.1 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. The TIE lines can be used to call through the PBX to reach another switching system (PBX or telephone company). By using TIE lines, the PBX can support not only communications with the public network but with other company members in the private network where the PBX is included.

Interface

The following interfaces can be used to establish a private network:

<table>
<thead>
<tr>
<th>Interface</th>
<th>Network Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>T1 (TIE [E &amp; M])</td>
<td>Digital (64 kbps × 24 channels)</td>
</tr>
<tr>
<td>E1 (E &amp; M)</td>
<td>Digital (64 kbps × 30 channels)</td>
</tr>
<tr>
<td>BRI/PRI (QSIG)</td>
<td>Digital (ISDN 2B+D/30B+D/23B+D)</td>
</tr>
<tr>
<td>VoIP</td>
<td>Internet Protocol (IP)</td>
</tr>
</tbody>
</table>

The available interfaces depend on the cards installed in the PBX.

→ 3.28 [1-1] Slot—Port Property - BRI Port—Network Numbering Plan— Trunk Property
→ 3.30 [1-1] Slot—Port Property - PRI Port—Network Numbering Plan— Trunk Property
→ 3.33 [1-1] Slot—Port Property - T1 Port— Trunk Property
→ 3.39 [1-1] Slot—Port Property - E1 Port— Trunk Property

Explanation

1. Making a TIE Line Call

One of the following two methods can be used to make a TIE line call.

a. Extension Number Method (Access without PBX Code)

Dial the [Extension Number] only.

[Example]
1.30.1 TIE Line Service

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.

b. PBX Code Method (Access with PBX Code)
Dial the [TIE Line Access Number] + [PBX Code] + [Extension Number].
→ 4.9 [2-6-1] Numbering Plan—Main—Features— ◈ TIE Line Access
→ 11.1 [9-1] TIE Table— ◈ Own PBX Code

[Example]

![Diagram showing TIE Line and PBX connections]

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.

2. TIE Line and Trunk Connection
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]

```
Outside Caller
Dials "123-4567".

Extn. 1011
Extn. 2011
```

**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. (→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI))

→ 12.3 [10-3] DDI / DID Table— DD/ID DID Destination—Day, Lunch, Break, Night
→ [453] DID Destination

b. **FWD/Call Transfer/Intercept Routing to the TIE Line**
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—by Making a TIE Line Call Method**

[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".

   → 4.9 [2-6-1] 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".
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".

b. Trunk Call through Other PBXs—by the ARS feature

[Example]
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".
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".

**c. 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.

→ 4.12 [2-7-1] Class of Service—COS Settings—TRS—◆ TRS Level—Day, Lunch, Break, Night
→ 4.13 [2-7-2] Class of Service—External Call Block
→ 5.1 [3-1-1] Trunk Group—TRG Settings—Main—◆ COS
→ 8.3 [6-3] Verification Code
→ [500] Trunk Group Number
[Programming Example of PBX-2]

<table>
<thead>
<tr>
<th>Trunk Group No.</th>
<th>COS No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>3</td>
</tr>
<tr>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>3</td>
<td>2</td>
</tr>
<tr>
<td>:</td>
<td>:</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>TRG of Incoming Call</th>
<th>Outgoing Call</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>TRG 1</td>
</tr>
<tr>
<td>COS 1</td>
<td>✔</td>
</tr>
<tr>
<td>COS 2</td>
<td>✔</td>
</tr>
<tr>
<td>COS 3</td>
<td>✔</td>
</tr>
<tr>
<td>:</td>
<td>:</td>
</tr>
</tbody>
</table>

✔ : Block

[Example]
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.
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.

d. FWD/Call Transfer/Intercept Routing to the Trunk
[Example]

Explaination:

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]**

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".

1. 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.
2. PBX-2 sends the modified call to the outside party "23-4567" according to its ARS programming.

3. **TIE Line Routing and Modification Table**

   **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.
   - **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.

   **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)

4.9 [2-6-1] Numbering Plan—Main—Other PBX Extension—Dial

If you dial:

a) 2xxx
b) 3xxx
c) 4xxx

(2, 3, 4: Other PBX Extension Number [TIE 3] in the Flexible Numbering Plan)

**[TIE Line Routing and Modification Table of PBX-1]**

<table>
<thead>
<tr>
<th>Location No.</th>
<th>Leading No.</th>
<th>TRG</th>
<th>Dial Modification</th>
<th>Priority 2</th>
<th>Dial Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>Removed No. of Digits</td>
<td>Added No.</td>
<td>TRG</td>
</tr>
<tr>
<td>01</td>
<td>2</td>
<td>1</td>
<td>0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>02</td>
<td>3</td>
<td>1</td>
<td>0</td>
<td>2</td>
<td>0</td>
</tr>
<tr>
<td>03</td>
<td>4</td>
<td>2</td>
<td>0</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Explanation:**

Location 01:
The hunt sequence by dialing [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)

→ 4.9 [2-6-1] Numbering Plan—Main—Features—TIE Line Access
11.1 [9-1] TIE Table — Own PBX Code

**TIE Line Network**

If you dial:
- a) 7-952-xxxx
- b) 7-953-xxxx
- c) 7-954-xxxx

*(7: TIE Line Access Number in the Flexible Numbering Plan)*

---

### [TIE Line Routing and Modification Table of PBX-1]

<table>
<thead>
<tr>
<th>Location No.</th>
<th>Leading No.</th>
<th>Priority 1</th>
<th>Priority 2</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Dial Modification</td>
<td>Dial Modification</td>
</tr>
<tr>
<td></td>
<td>Leading No.</td>
<td>TRG</td>
<td>Removed No. of Digits</td>
</tr>
<tr>
<td>01</td>
<td>952</td>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>02</td>
<td>953</td>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>03</td>
<td>954</td>
<td>2</td>
<td>3</td>
</tr>
</tbody>
</table>

**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]

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 andModified Table of PBX-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).

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)
1.30.1 TIE Line Service

[Example]

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.

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).

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.
4. TIE Line Routing Flowchart
[Making a TIE Line Call from an Extension]

A TIE line call is made as follows:

PBX Code Method: 7-abc-xxxx
Extension No. Method: dexx

Is the dialled number identified as a TIE line access no.
or an other PBX extension no. in the flexible numbering plan of the own PBX?

Yes:
TIE line access no.: 7
Other PBX extension no.: de

Is the leading number (abc or de) found in the TIE Line Routing and Modification Table of the own PBX?

Yes
Selects the corresponding trunk group, and the dialled number is modified if a removed number of digits and/or added number is assigned.

Is the trunk group available?

Yes
Is there an idle trunk in the trunk group?

Yes
Routes to other PBX or trunk.

No
Reorder tone

No
Reorder tone

No
Busy tone

Not treated as a TIE line call.
Receiving a Call through a TIE Line

**<Extension Number Method (Access without PBX Code)>**

A call is received through a TIE line as follows: # 1021

The received number is modified as programmed for each trunk port.

- Removed number of digits: 1
- Added number: None
- Received number: # 1021
- Modified number: # 1021 = 1021

Remove the first 1 digit.

- Check the modified number with the flexible numbering plan of the own PBX.

**<PBX Code Method (Access with PBX Code)>**

A call is received through a TIE line as follows: ## 0511033

The received number is modified as programmed for each trunk port.

- Removed number of digits: 3
- Added number: 9
- Received number: ## 0511033
- Modified number: ## 0511033 = 9511033

1) Remove the first 3 digits.
2) Add "9".

Does the modified number have the own PBX code "951"?

- Yes: 9511033
- Goes to A

Removes the own PBX code "951".

(A is in the flowchart of [Making a TIE Line Call from an Extension].)

Operator Call No. of the Own PBX

- Directs the call to the operator.

Extension No. of Other PBX

- Does the corresponding extension exist?
  - No
  - Is the corresponding extension idle?
    - Yes
      - Calls the extension.
    - No
      - Reorder tone.

Idle Line Access No. or Trunk Group Access No.

- Is the trunk group of the outgoing call from the own PBX enabled against the COS of the trunk group of the incoming call to the own PBX?
  - No
    - Reorder tone
  - Yes
    - TRS/Barring applies.

Others

- Reorder tone.

1.30.1 TIE Line Service
## Conditions

- A trunk which is used for a private network should be assigned "Private" as the networking type. (→ 1.1.1.1 Incoming Trunk Call Features—SUMMARY)
- To establish a QSIG network (→ 1.30.4 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-NCP series, KX-TDE series, and KX-TDA series PBXs.

### [TIE Line Network Connection Example]

![TIE Line Network Diagram]

- **[TIE Call by Own PBX Extension Number]**
  When only two PBXs are connected in a network (not including a VoIP network), both PBXs can have extensions with the same leading number.

### 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.

→ The KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is required to use this feature.
→ This feature is not available when the PBXs are connected by a VoIP network.
→ System programming is required to enable this feature.

• 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.
→ 4.5 [2-3] Timers & Counters—Miscellaneous—◆ Incoming Call Inter-digit Timer—TIE

**PC Programming Manual References**
3.28 [1-1] Slot—Port Property - BRI Port—Network Numbering Plan—◆ Trunk Property
3.30 [1-1] Slot—Port Property - PRI Port—Network Numbering Plan—◆ Trunk Property
3.33 [1-1] Slot—Port Property - T1 Port—◆ Trunk Property
3.39 [1-1] Slot—Port Property - E1 Port—◆ Trunk Property
4.5 [2-3] Timers & Counters—Miscellaneous—◆ Incoming Call Inter-digit Timer—TIE
4.9 [2-6-1] Numbering Plan—Main—Features
→ ◆ Idle Line Access (Local Access)
→ ◆ Trunk Group Access
→ ◆ TIE Line Access
4.9 [2-6-1] Numbering Plan—Main—Other PBX Extension
4.12 [2-7-1] Class of Service—COS Settings—TRS—◆ TRS Level—Day, Lunch, Break, Night
4.18 [2-9] System Options—Option 4—◆ Private Network—TIE Call by Extension Numbering (Optional SD Card Required)
5.1 [3-1-1] Trunk Group—TRG Settings—Main—◆ COS
11.1 [9-1] TIE Table
12.2 [10-2] DIL Table & Port Settings
→ DIL—◆ Trunk Property
→ DIL—◆ DIL Destination—Day, Lunch, Break, Night
→ DDI / DID / TIE / MSN
12.3 [10-3] DDI / DID Table—◆ DDI / DID Destination—Day, Lunch, Break, Night

**PT Programming Manual References**
[453] DID Destination
[500] Trunk Group Number

**Feature Guide References**
1.1.1.6 Intercept Routing
1.3.1.2 Call Forwarding (FWD)
1.9.1 Toll Restriction (TRS)/Call Barring (Barring)
1.10.1 Automatic Route Selection (ARS)
1.12.1 Call Transfer
1.17.6 Direct Inward System Access (DISA)
2.3.6 Flexible Numbering/Fixed Numbering
3.1 Capacity of System Resources
1.30.2  Voice over Internet Protocol (VoIP) Network

**Description**

The PBX can connect to another PBX via a private IP network. In this case, voice signals are converted into IP packets (VoIP) and sent through this network. VoIP networks support private network communications using TIE line service.

**[Example]**

Required Programming

- **PBX**
  For making a call:
  ARS programming (→ 1.10.1 Automatic Route Selection (ARS)) or TIE line service programming
  For receiving a call:
  TIE line service programming

**[TIE Line Routing and Modification Table]**

<table>
<thead>
<tr>
<th>Location No.</th>
<th>Leading No.</th>
<th>Priority 1</th>
<th>Priority 2</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>TRG</td>
<td>TRG</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Dial Modification</td>
<td>Dial Modification</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Removed No. of Digits</td>
<td>Added No.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Removed No. of Digits</td>
<td>Added No.</td>
</tr>
<tr>
<td>01</td>
<td>2</td>
<td>2 (VoIP port)</td>
<td>0</td>
</tr>
<tr>
<td>02</td>
<td>3</td>
<td>2</td>
<td>0</td>
</tr>
</tbody>
</table>
Calls to destinations with leading number "2" or "3" are automatically routed through the VoIP ports, designated as trunk group 2.

- **V-IPGW Card**
  IP address assignment for the local PBX and other PBXs.

**[Programming Example]**

<table>
<thead>
<tr>
<th>Destination</th>
<th>Leading No.</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>2</td>
<td>200.45.11.35</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>199.176.64.1</td>
</tr>
</tbody>
</table>

**[Explanation]**
Calls are routed to the IP address of each V-IPGW card based on the leading number dialled.

**Gateway Groups**
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.
When "123XXX" is dialled, the call is routed to GW Group-1. However the lowest-numbered device (V-IPGW(1)) is busy or not available, so the call is rerouted to V-IPGW(2).

**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.

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]**
1.30.2 Voice over Internet Protocol (VoIP) Network

<table>
<thead>
<tr>
<th>Quick Dialling No.</th>
<th>Destination No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>720</td>
<td>9012345</td>
</tr>
</tbody>
</table>

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.

**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. (→ 1.30.4 QSIG Standard Features)
- TRS/Call Barring settings apply to calls rerouted to public trunks.

**Installation Manual References**

3.4.1 Virtual Cards

**PC Programming Manual References**

3.7 [1-1] Slot—Port Property - Virtual IP Gateway Port
11.1 [9-1] TIE Table

**Feature Guide References**

1.30.1 TIE Line Service
1.30.3 Virtual Private Network (VPN)

Description

Virtual Private Network (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 (→ 1.10.1 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/MSN) which is assigned on the trunk.

[Example]

Conditions

- Each BRI/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 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 (→ 1.6.1.6 Quick Dialling).
[Quick Dialling Programming Example]

<table>
<thead>
<tr>
<th>Location No.</th>
<th>Quick Dialling No.</th>
<th>Desired No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quick Dialling 01</td>
<td>2345 (extension no. of other PBX)</td>
<td>9-123-4321 (Public no. of extension 2345)</td>
</tr>
</tbody>
</table>

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
3.28 [1-1] Slot—Port Property - BRI Port—Network Numbering Plan—◆ Trunk Property
3.30 [1-1] Slot—Port Property - PRI Port—Network Numbering Plan—◆ Trunk Property
10.2 [8-2] Leading Number—◆ Leading Number
11.1 [9-1] TIE Table
   →◆ Leading Number
   →◆ Removed Number of Digits
   →◆ Added Number

PT Programming Manual References
None

Feature Guide References
1.30.1 TIE Line Service
1.30.4 QSIG Standard Features

1.30.4.1 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]

<table>
<thead>
<tr>
<th>Service</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Line Identification Presentation (CLIP)</td>
<td>Sends the caller’s number to the QSIG network when making a call.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG</td>
</tr>
<tr>
<td>Connected Line Identification Presentation (COLP)</td>
<td>Sends the number of the answered party to the QSIG network when answering a call.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG</td>
</tr>
<tr>
<td>Calling Line Identification Restriction (CLIR)</td>
<td>Prevents the caller’s CLI being presented to the called party by the caller.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG</td>
</tr>
<tr>
<td>Connected Line Identification Restriction (COLR)</td>
<td>Prevents COLP being sent by the answered party.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG</td>
</tr>
<tr>
<td>Calling Name Identification Presentation (CNIP)</td>
<td>Sends the caller’s name to the QSIG network when making a call.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG</td>
</tr>
<tr>
<td>Connected Name Identification Presentation (CONP)</td>
<td>Sends the name of the answered party to the QSIG network when answering a call.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG</td>
</tr>
<tr>
<td>Service</td>
<td>Description &amp; Reference</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>-----------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Calling Name Identification Restriction (CNIR)</td>
<td>Prevents the caller’s name being presented to the called party by the caller.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG</td>
</tr>
<tr>
<td>Connected Name Identification Restriction (CONR)</td>
<td>Prevents CONP being sent by the answered party.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.2 Calling/Connected Line Identification Presentation (CLIP/ COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG</td>
</tr>
<tr>
<td>Call Forwarding (CF)—by QSIG</td>
<td>Forwards a call to the QSIG network.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.3 Call Forwarding (CF)—by QSIG</td>
</tr>
<tr>
<td>Call Transfer (CT)—by QSIG</td>
<td>Transfers a call to the QSIG network.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.4 Call Transfer (CT)—by QSIG</td>
</tr>
<tr>
<td>Completion of Calls to Busy Subscriber (CCBS)—by QSIG</td>
<td>Receives callback ringing when a busy called party on the QSIG network becomes free.</td>
</tr>
<tr>
<td></td>
<td>→ 1.30.4.5 Completion of Calls to Busy Subscriber (CCBS)—by QSIG</td>
</tr>
</tbody>
</table>

**Feature Guide References**

1.21.1 Integrated Services Digital Network (ISDN)
1.30.1 TIE Line Service
1.30.4.2 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]

1) Dials "202".
2) "John
   101"
   is displayed.

[COLP/CONP Example]

1) Dials "203".
2) Answers the call.

3) "Paul
   204"
   is displayed.

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 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**

- 3.28 [1-1] Slot—Port Property - BRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY
- 3.30 [1-1] Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY
- 4.9 [2-6-1] Numbering Plan—Main—Features
  →◆ COLR Set / Cancel
  →◆ CLIR Set / Cancel
- 6.1 [4-1-1] Wired Extension—Extension Settings—Main
  →◆ Extension Number
  →◆ Extension Name
- 6.10 [4-2-1] Portable Station—Extension Settings—Main
  →◆ Extension Number
  →◆ Extension Name

**PT Programming Manual References**

- [003] Extension Number
- [004] Extension Name

**Feature Guide References**

- 1.20.2 Flexible Buttons

**User Manual References**

- 1.8.4 Displaying Your Telephone Number on the Called Party and Caller’s Telephone (Calling/Connected Line Identification Presentation [CLIP/COLP])
- 1.8.5 Preventing Your Telephone Number Being Displayed on the Caller’s Telephone (Connected Line Identification Restriction [COLR])
1.8.6 Preventing Your Number Being Displayed on the Called Party's Telephone (Calling Line Identification Restriction [CLIR])
1.30.4.3 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 (→ 1.3.1.2 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

3.28  [1-1] Slot—Port Property - BRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY
3.30  [1-1] Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

PT Programming Manual References

None

User Manual References

1.6.1  Forwarding Calls
1.30.4.4 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 (→ 1.12.1 Call Transfer).
PC Programming Manual References

3.28  [1-1] Slot—Port Property - BRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY
3.30  [1-1] Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

PT Programming Manual References

None

User Manual References

1.4.1  Transferring a Call (Call Transfer)
1.30.4.5 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

3.28 [1-1] Slot—Port Property - BRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, 3PTY
3.30 [1-1] Slot—Port Property - PRI Port—Supplementary Service—◆ COLP, CLIR, COLR, CNIP, CONP, CNIR, CONR, CF (Rerouting), CT, CCBS, AOC-D, AOC-E, E911, 3PTY

PT Programming Manual References

None

User Manual References

1.2.4 When the Dialed Line is Busy or There is No Answer
1.30.5 QSIG Enhanced Features

1.30.5.1 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:

<table>
<thead>
<tr>
<th>Light pattern</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td>The monitored extension is idle.</td>
</tr>
<tr>
<td>Red on</td>
<td>The monitored extension is busy or has set DND for trunk calls.</td>
</tr>
</tbody>
</table>

[Example Network]

```
PBX-1
Network ID 1
Extn. 2001
Extn. 2002
Extn. 4001

PBX-2
Network ID 2
Extn. 2001
Extn. 2002

PBX-3
Network ID 0
Extn. 1001

PBX-4
Network ID 3
Extn. 4001
```

[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 (→ 1.30.1 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.

   → 11.2 [9-2] 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 BRI or 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.

- 3.28 [1-1] Slot—Port Property - BRI Port—ISDN CO— Networking Data Transfer
- 3.30 [1-1] Slot—Port Property - PRI Port—CO Setting— Networking Data Transfer

<table>
<thead>
<tr>
<th>Transmit</th>
<th>Extension Status Information Functions</th>
</tr>
</thead>
</table>
| 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-GW-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.
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.

Netwok Monitor Extensions Table

<table>
<thead>
<tr>
<th>Index No.</th>
<th>Network Extn. No.</th>
<th>Network Extn. Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>001</td>
<td>2001</td>
<td>Branch 1: T. Durden</td>
</tr>
<tr>
<td>002</td>
<td>2002</td>
<td>Branch 1: M. Singer</td>
</tr>
<tr>
<td>003</td>
<td>4001</td>
<td>Branch 2: R. Paulson</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>250</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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**

- The KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is required for all KX-NCP series PBXs that will monitor extensions or have extensions monitored.
- All PBXs in the network must be KX-NCP series, KX-TDE series, or KX-TDA series PBXs. For information on the hardware requirements for KX-TDE/KX-TDA series 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.

6.5 [4-1-4] Wired Extension—Flexible Button
- ◆ Type
- ◆ Dial (for NDSS)

6.13 [4-2-3] Portable Station—Flexible Button
- ◆ Type
- ◆ Dial (for NDSS)

6.17 [4-3] DSS Console
- ◆ 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.
  → 4.12 [2-7-1] Class of Service—COS Settings—Programming & Manager—◆ Manager Assignment
  → [511] Manager Assignment
- A maximum of two ports of each BRI card can be assigned to transmit extension status information ("ON").
- 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.
  → 11.2 [9-2] 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.
  → 6.8 [4-1-6] Wired Extension—NDSS Link Data - Send
  → 6.15 [4-2-4] 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 can not be used when two PBXs are networked using the [TIE Call by Own PBX Extension Number] feature (→ 1.30.1 TIE Line Service).

PC Programming Manual References

3.28 [1-1] Slot—Port Property - BRI Port—ISDN CO—◆ Networking Data Transfer
3.30 [1-1] Slot—Port Property - PRI Port—CO Setting—◆ Networking Data Transfer
4.12 [2-7-1] Class of Service—COS Settings—Programming & Manager—◆ Manager
6.5 [4-1-4] Wired Extension—Flexible Button
  → ◆ Type
  → ◆ Dial (for NDSS)
6.8 [4-1-6] Wired Extension—NDSS Link Data - Send
6.13 [4-2-3] Portable Station—Flexible Button
  → ◆ Type
  → ◆ Dial (for NDSS)
6.15 [4-2-4] Portable Station—NDSS Link Data - Send
6.17 [4-3] DSS Console
  → ◆ Type
  → ◆ Dial (for NDSS)
11.2 [9-2] Network Data Transmission
11.3 [9-3] Network Operator (VoIP)
11.4 [9-4] NDSS Key Table

PT Programming Manual References

[511] Manager Assignment

Feature Guide References

1.30.1 TIE Line Service
1.30.2 Voice over Internet Protocol (VoIP) Network
1.30.4 QSIG Standard Features

User Manual References

1.2.1 Basic Calling
2.1.7 Releasing Network Direct Station Selection (NDSS) Monitor
1.30.5 QSIG Enhanced Features

1.30.5.2 Centralised Voice Mail

Description

Up to 8 PBXs connected in a VoIP or ISDN private network can share the services of a VPS connected to another PBX in DPT integration. This VPS can provide voice mail for extensions attached to any of the PBXs in the network, as if the VPS were connected to the local PBX. In addition, the VPS can send Message Waiting notifications to extensions at any PBX, and users can access their mailboxes directly using the Message Waiting button.

The VPS features available at extensions in a network are shown below.

<table>
<thead>
<tr>
<th>No.</th>
<th>Feature</th>
<th>Remote VPS</th>
<th>Local VPS</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Automatic Configuration—Quick Setup</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>2</td>
<td>FWD to a Mailbox of the VPS</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>3</td>
<td>Intercept Routing to a Mailbox of the VPS</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>4</td>
<td>Transferring to a Mailbox of the VPS</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>5</td>
<td>Transfer Recall to a Mailbox of the VPS</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>6</td>
<td>Listening to a Recorded Message (Direct Mailbox Access)</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>7</td>
<td>VPS Trunk Service &amp; Automatic Time Mode Notification for Incoming Call</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>8</td>
<td>Caller’s Identification Notification to the VPS</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>9</td>
<td>DID Number Notification to the VPS</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>10</td>
<td>Status Notification to the VPS</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>11</td>
<td>Paging by the VPS</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>12</td>
<td>Live Call Screening (LCS)</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>13</td>
<td>Two-way Recording into the VPS</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>14</td>
<td>VPS Data Control by the PBX</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>15</td>
<td>Remote FWD Setting by the VM</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>16</td>
<td>VM Menu</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>17</td>
<td>Message clear on Check Out</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>18</td>
<td>Voice message number display</td>
<td>Yes</td>
<td>Yes</td>
</tr>
</tbody>
</table>
[Trunk Call Answered by the VPS, Transferred to Extension at Other PBX (Extension Number Method)]

A trunk call is answered by the AA service of the VPS. 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 in the VPS 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 VPS, and answered by mailbox 201. If the outside caller leaves a message, the VPS 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 VM 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 VPS, using a VPS 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 VPS.

Multiple VPSs
More than one PBX in the network can have a VPS connected, and each VPS can provide voice mail services for extensions connected to other PBXs in the network.

Conditions
- The KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is required for the KX-NCP series PBX connected to the VPS, and for all KX-NCP series PBXs that will use this feature.
• All PBXs in the network must be KX-NCP series, KX-TDE series, or KX-TDA series PBXs. For information on the hardware requirements for KX-TDE/KX-TDA series PBXs that share the same VPS, refer to the corresponding Feature Guide.
• Only KX-TVM series VPSs connected using DPT integration support this feature.
• Mailboxes for extensions connected to other PBXs in the network must be set up manually.
• An extension can receive Message Waiting notifications from multiple VPSs 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 VPSs, 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.
• When PBXs are connected by PRI23 cards, the card must be set to ISDN Standard mode. If set to T1 mode, the card must be removed and reinstalled to allow the mode to be changed. When the card is reinserted into the PBX, it is possible to choose whether the card will be set to T1 mode or ISDN Standard mode. The default is ISDN Standard.

**PBX Code Method**
The number used in this method must be no more than 8 digits.
• When using a mailbox number containing more than 5 digits, the version of the KX-TVM series VPS must be 2.1 (Main Software Version 2.50) or later.

**Transfer to Mailbox**
It is possible to transfer calls to a mailbox of a VPS at another PBX, by creating a flexible button containing the following:

[T] + extension number of VPS at other PBX + [P] + [#] + 6

Then, the extension user can transfer calls to a mailbox by pressing this key, entering the mailbox number, and going on hook.
• The VM Menu feature cannot be used in a Centralised Voice Mail network. When extensions belonging to a PBX that is not connected to the VPS have a mailbox in the VPS, the VM Menu feature should be disabled for the mailbox using the Class of Service (COS) setting of the VPS. Otherwise, there may be an answering delay of about 6 seconds.

**PC Programming Manual References**
11.1 [9-1] TIE Table—Centralised VM
11.2 [9-2] Network Data Transmission
→◆ Error Log for Centralised VM—Network MSW Transmission (Counter)
→◆ Error Log for Centralised VM—Network MSW Transmission (Buffer)

**Feature Guide References**
1.24.2.3 Voice Mail DPT (Digital) Integration
1.30.1 TIE Line Service
1.30.6 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.

Conditions

- The KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is required for all PBXs that will support virtual PSs.
- The conditions for 1.2.2.3 Outside Destinations in Incoming Call Distribution Group also apply to this feature.

PC Programming Manual References

4.18 [2-9] System Options—Option 4—Send CLIP of CO Caller to ISDN—when ICD Group with Cellular Phone
PT Programming Manual References
None

Feature Guide References
1.2.2.3 Outside Destinations in Incoming Call Distribution Group
1.25.6 Virtual PS
1.30.6 Network ICD Group

1.30.6.1 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**

- The KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is required for all PBXs that will support virtual PSs.
- 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**

4.18 [2-9] System Options—Option 4—◆ System Wireless—Out of Range Registration

**PT Programming Manual References**

None

**Feature Guide References**

1.2.2.3 Outside Destinations in Incoming Call Distribution Group
1.31 IP Telephone Features

1.31.1 IP Proprietary Telephone (IP-PT)

Description
This PBX supports the connection of IP-PTs, which function almost identically to normal PTs. However, they connect to the PBX over a LAN, and make and receive calls using IP.

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
- A V-IPEXT card is required to use IP-PTs.
- 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. For details on how to register IP-PTs, see the IP Networking Guide.
- The following features cannot be used with an IP-PT:
  - XDP (→ 1.11.9 Paralleled Telephone)
  - Digital XDP (→ 1.11.9 Paralleled Telephone)
  - OHCA (→ 1.8.4.3 Off-hook Call Announcement (OHCA))
  - Using PC Phone or PC Console on a connected PC (→ 1.32.2 PC Phone/PC Console)
- 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)
- **KX-NT307(PSLP1528) Bluetooth Module**
  A Bluetooth wireless headset can be registered to an 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.

• **Automatic Rerouting to Secondary PBX**
  A KX-NT300 series IP-PT can automatically connect to a secondary PBX, when the primary PBX becomes disconnected.

**Installation Manual References**

3.4.1 Virtual Cards

**PC Programming Manual References**

3.14 [1-1] Slot—Card Property - Virtual IP Extension
3.15 [1-1] Slot—Port Property - Virtual IP Extension
3.16 [1-1] Slot—Port Property - Virtual IP Extension—Connection Command

**PT Programming Manual References**

None
1.31.2 SIP (Session Initiation Protocol) Extension

Description

This PBX supports the connection of SIP supported IP telephones (hardphones and softphones). SIP extensions make and receive calls using Internet Protocol (IP).

SIP extensions support the following features:

- Account Code Entry (∆ 1.5.4.3 Account Code Entry)
- Automatic Route Selection (ARS) (∆ 1.10 Automatic Route Selection (ARS) Features)
- Broadcasting* (∆ 1.16.1 Broadcasting)
- Call Forwarding (FWD)* (∆ 1.3.1.2 Call Forwarding (FWD))
- Call Hold* (∆ 1.13.1 Call Hold)
- Call Transfer with Announcement (∆ 1.12.1 Call Transfer)
- Conference* (∆ 1.14.1 Conference Features)
- Direct Inward System Access (DISA) (∆ 1.17.6 Direct Inward System Access (DISA))
- Doorphone Call (∆ 1.17.1 Doorphone Call)
- Emergency Call (∆ 1.5.4.2 Emergency Call)
- Idle Line Access (∆ 1.5.3 Trunk Access)
- Operator Call (∆ 2.2.5 Operator Features)
- Personal Speed Dialling* (∆ 1.6.1.5 Speed Dialling—Personal/System)
- Quick Dialling (∆ 1.6.1.6 Quick Dialling)
- S-CO Line Access (∆ 1.6.1.5 Trunk Access)
- System Speed Dialling* (∆ 1.6.1.5 Speed Dialling—Personal/System)
- TIE Line Call (∆ 1.30.1 TIE Line Service)
- Trunk Group Access (∆ 1.5.3 Trunk Access)
- Verification Code Entry (∆ 1.9.6 Verification Code Entry)

*1 As a member only (not as an originator).
*2 Assigned through system programming only.
*3 Feature number is not supported.

KX-HGT100 extensions support the following additional features:

- Message Waiting
- Personal Speed Dialling (Internal)
- Voice Mail*1
- Automatic Date and Time Adjustment
- Showing/preventing the telephone number when making outside calls

For details, refer to the Quick Reference Guide of the KX-HGT100.

*1 To perform VPS integration, it is necessary to assign Call Forwarding or Intercept Routing to the VPS extension through system programming.
It is also possible to perform Remote FWD Setting by the VM (∆ 1.24.2.3 Voice Mail DPT (Digital) Integration).

Conditions

- 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 IPCMPR card, password, and extension number must be assigned on the SIP extension and on the PBX.
  For details on how to register, refer to the Information on IP Networking Guide.
- 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.
Installation Manual References

3.4.1 Virtual Cards

PC Programming Manual References

3.17 [1-1] Slot—Card Property - Virtual SIP Extension
3.18 [1-1] Slot—Port Property - Virtual SIP Extension Port
3.19 [1-1] Slot—Port Property - Virtual SIP Extension Port—Connection Command

PT Programming Manual References

None
1.31.3 Peer-to-Peer Connection

Description

This PBX automatically establishes peer-to-peer communication between peer-to-peer compatible IP extensions (i.e., IP-PTs and SIP extensions). Non-peer-to-peer IP extension calls are routed through the DSP card. However, with peer-to-peer calls, the call is routed directly from one IP extension to another without going through a DSP card. In other words, peer-to-peer calls can be made without using the PBX’s resources.

Conditions

- Peer-to-peer calls can only be made between extensions connected to the PBX through the IPCMPR card.
- The following combinations of IP telephones are peer-to-peer compatible:

<table>
<thead>
<tr>
<th></th>
<th>KX-NT265 (version 2.00 or later)*</th>
<th>KX-NT300 Series</th>
<th>SIP Extension</th>
</tr>
</thead>
<tbody>
<tr>
<td>KX-NT265 (version 2.00 or later)*</td>
<td>✓</td>
<td></td>
<td></td>
</tr>
<tr>
<td>KX-NT300 Series</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>SIP Extension</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
</tbody>
</table>

✓: Available
*: For details about checking the version of the KX-NT265, refer to the Installation Manual.

- The types of CODEC available for peer-to-peer calls depend on the IP telephones being used, as follows:

<table>
<thead>
<tr>
<th></th>
<th>KX-NT265 (version 2.00 or later)</th>
<th>KX-NT300 Series</th>
<th>SIP Extension</th>
</tr>
</thead>
<tbody>
<tr>
<td>KX-NT265 (version 2.00 or later)</td>
<td>G.711/G.729A</td>
<td></td>
<td></td>
</tr>
<tr>
<td>KX-NT300 Series</td>
<td>G.711/G.729A</td>
<td>G.722/G.711/G.729A</td>
<td></td>
</tr>
<tr>
<td>SIP Extension</td>
<td>G.711/G.729A</td>
<td>G.711/G.729A</td>
<td>G.711/G.729A</td>
</tr>
</tbody>
</table>

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 the KX-NT300 series IP-PTs and some SIP extensions that support this CODEC during peer-to-peer communication.
- Through system programming, it is possible to assign a specific type of CODEC to be used for each extension.
- For SIP extensions, the priority of the CODEC that will be used can be specified on a SIP extension basis.
- 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.
Installation Manual References
1.4.3 System Capacity

PC Programming Manual References
3.15 [1-1] Slot—Port Property - Virtual IP Extension—Option—◆ IP Codec Priority

Feature Guide References
1.31.1 IP Proprietary Telephone (IP-PT)
1.31.4 Self Labelling (KX-NT366 only)

**Description**

The KX-NT366 IP-PT features 12 flexible buttons, with an LCD screen next to the buttons. Each item can be named on the LCD screen through personal/system programming to reflect the function of its corresponding flexible button. Additionally, the flexible buttons can be organised into 4 "pages", allowing you to assign a total of 48 items to the 12 flexible buttons. You can toggle between pages by pressing the NEXT PAGE key, as follows:

![Diagram showing LCD screen and flexible buttons with examples of labels]

**Conditions**

- Up to 12 characters can be assigned to the LCD of each flexible button through personal/system programming.
- 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.

**PC Programming Manual References**

6.5 [4-1-4] Wired Extension—Flexible Button—Label Name

**PT Programming Manual References**

None
Feature Guide References

1.20.2 Flexible Buttons

User Manual References

1.14.1 Self Labelling (KX-NT366 only)
1.32 Computer Telephony Integration (CTI) Features

1.32.1 Computer Telephony Integration (CTI)

Description
Connecting a PC to a DPT, or connecting a CTI Server to this PBX, enables extension users to make use of advanced features by using the stored data in the PC or the CTI Server. The features that are available through CTI depend on the CTI application software used on connected PCs. There are two types of CTI as follows:

1. First Party Call Control
2. Third Party Call Control

1. First Party Call Control
A PC is connected to a DPT using a USB port (USB Module) attached to the DPT. The PC monitors the status of the DPT and controls the DPT.

2. Third Party Call Control
A CTI Server is connected to the PBX using a LAN port of the IPCMPR card as a CTI interface. PCs monitor the status of the PBX and control the PBX via the CTI Server. When a call is received, the CTI Server can be used to decide call routing, and to send information about the call to the destination extension user’s PC. So, for example, if the call is from a customer, that customer’s account information can be shown...
automatically on the screen of the PC at the same time as the call is received, depending on the CTI software used. In Third Party Call Control, the PC does not control extensions directly.

**Conditions**

**[General]**

- **Capable telephone:**
  
  DPT: KX-DT300 series or KX-T7600 series PT, and USB Module

  IP-PT
  
- **CTI application software** must be installed on the connected PC. In addition, some features may require a KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version).

- **Application Programming Interface (API)/Protocol**
  
  The following interfaces are required for First Party Call Control and Third Party Call Control:

<table>
<thead>
<tr>
<th>Type</th>
<th>API/Protocol</th>
</tr>
</thead>
<tbody>
<tr>
<td>First Party Call Control</td>
<td>TAPI 2.1</td>
</tr>
<tr>
<td>Third Party Call Control</td>
<td>ECMA CSTA Phase 3</td>
</tr>
<tr>
<td></td>
<td>TAPI 2.1</td>
</tr>
</tbody>
</table>

- The operating system of PC or CTI Server required for First Party Call Control or Third Party Call Control depends on your CTI application software. For details, refer to the manual for your CTI application software.

- Third Party Call Control cannot be performed when two or more CTI Servers are connected to the PBX.

**[IP-PT Call Control]**

- A PC connected directly to an IP-PT cannot be used for First Party Call Control.

- IP-PTs do not support the use of PC Phone or PC Console software on a connected PC.

**Installation Manual References**

3.8.4 First Party Call Control CTI Connection

3.13.1 Connection of Peripherals

**PC Programming Manual References**

4.9 [2-6-1] Numbering Plan—Main—Features— Dial Information (CTI)
1.32.1 Computer Telephony Integration (CTI)

4.18 [2-9] System Options—Option 6 (CTI)

PT Programming Manual References
None
1.32.2 PC Phone/PC Console

Description

CTI First Party Call Control can be conducted by connecting a PC with a KX-DT343/KX-DT346 or KX-T7633/KX-T7636 DPT (equipped with USB Module) through USB ports. By using the KX-TDA0350 PC Phone or KX-TDA0300 PC Console CTI application, users can enjoy many PBX features while utilising the benefits of a CTI solution. PC Console gives users access to certain administrative level features that are not available with PC Phone.

The extensions that use PC Console must be assigned through system programming.

Along with many PBX features, PC Phone and PC Console offer many specific features, including the following:

- **Phone Book:**
  A list of telephone numbers including detailed information pertaining to each entry. This feature allows the user to make calls and send e-mails to each entry.

- **Outlook Integration:**
  The Microsoft® Outlook® Address Book can be integrated to be used with the PC Phone or PC Console.

- **Pop-up Screen:**
  An image containing information about the caller appears when a call is received.

- **Voice Memo:**
  Conversations are recorded at any time and can be saved, sent by e-mail, etc.

- **Text Memo:**
  A user can type a short message during a conversation which can be saved, sent by e-mail, etc.

[For PC Phone Only]

- **TAM (Telephone Answering Machine):**
  If a call is not answered within a preprogrammed time, callers can leave a message after hearing a greeting message.

- **Message:**
  TAM messages are played back and/or automatically transferred by phone or by e-mail.

[For PC Console Only]

- **Incoming Group Screen:**
  Names, extension numbers, and the status (e.g., busy) of each extension can be monitored on a single screen. Calls can also be made and received on this screen.

Conditions

- **Hardware Requirements:** KX-DT343/KX-DT346 or KX-T7633/KX-T7636 DPT, and USB Module

Installation Manual References

3.8.4 First Party Call Control CTI Connection

PC Programming Manual References

3.21 [1-1] Slot—Port Property - Extension Port
  → DPT Type—Type
  → DPT Type—Location No.

PT Programming Manual References

[601] Terminal Device Assignment
Feature Guide References

3.1 Capacity of System Resources
1.32.3 CA (Communication Assistant)

Description

CTI First Party Call Control can be conducted by connecting a PC containing the CA application to the PBX. Activation keys are required to enable the application, some of which are preinstalled on the IPCMPR card. CA is a serverless application that runs on a LAN network. CA has 3 operating modes: Basic, Pro, and Supervisor.

As well as many of the same features as PC Phone/PC Console, CA offers some new CTI features such as:

- **Presence:**
  CA users can view the present status (Available, Busy, etc.) of other CA users in their contact list. This feature is not available for CA Basic.

- **ICD Group Supervisor:**
  A supervisor can use this feature to monitor users within an ICD group from a PC.

- **Call Accounting:**
  Calls can be monitored from a PC in the same way as SMDR for the PBX.

Installation Manual References

3.8.4 First Party Call Control CTI Connection

PC Programming Manual References

3.4 [1-1] Slot—Card Property - IPCMPR—Port Number—◆ Built-in Communication Assistant Server
6.1 [4-1-1] Wired Extension—Extension Settings—Option 9—◆ Built-in Communication Assistant
6.10 [4-2-1] Portable Station—Extension Settings—Option 9—◆ Built-in Communication Assistant
1.33 Cellular Phone Features

1.33.1 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.

The following features can be used with cellular phones and other outside extensions:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Outside Destinations in Incoming Call Distribution Group</td>
<td>Up to 4 cellular phones can be assigned as members of an Incoming Call Distribution (ICD) Group, and receive calls to the group.</td>
</tr>
<tr>
<td>Cellular Phone XDP Parallel Mode</td>
<td>A PT user can set up to 4 cellular phones to ring in parallel for incoming calls.</td>
</tr>
<tr>
<td>DISA Automatic Walking COS</td>
<td>Registered cellular phones are automatically recognised as PBX extensions when calling through DISA.</td>
</tr>
<tr>
<td>DISA Call Transfer From Outside Destination</td>
<td>A cellular phone user who answers a trunk call forwarded from the PBX using DISA can transfer that call to an extension within the PBX.</td>
</tr>
</tbody>
</table>

**Conditions**

- The KX-NCS3910 (Activation Key for Software Upgrade to Enhanced Version) is required to use this feature.

**Feature Guide References**

1.25.6 Virtual PS
Section 2

System Configuration and Administration Features
2.1 System Configuration—Hardware

2.1.1 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-TDA0141) 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:

<table>
<thead>
<tr>
<th>Mode</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parallel Mode</td>
<td>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). (→ 1.11.9 Parallelled Telephone)</td>
</tr>
<tr>
<td>XDP Mode</td>
<td>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.</td>
</tr>
</tbody>
</table>

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. (→ 1.11.9 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

3.21 [1-1] Slot—Port Property - Extension Port
6.17 [4-3] DSS Console— Pair Extension

PT Programming Manual References

[007] DSS Console Paired Telephone
[600] EXtra Device Port (XDP) Mode
2.2 System Configuration—Software

2.2.1 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.** → 1.1.2.2 Internal Call Block
- **b.** → 1.3.1.2 Call Forwarding (FWD)
- **c.** → 1.3.1.3 Do Not Disturb (DND)—DND Override
- **d.** → 1.4.1.3 Call Pickup
- **e.** → 1.5.4.3 Account Code Entry
- **f.** → 1.5.5.3 Trunk Access
- **g.** → 1.8.2 Executive Busy Override
- **h.** → 1.8.3 Call Monitor
- **i.** → 1.8.4.3 Off-hook Call Announcement (OHCA)
- **j.** → 1.8.4.4 Whisper OHCA
- **k.** → 1.9.1 Toll Restriction (TRS)/Call Barring (Barring)
- **l.** → 1.9.3 Extension Dial Lock
- **m.** → 1.9.5 Walking COS
- **n.** → 1.11.8 Trunk Call Limitation
- **o.** → 1.12.1 Call Transfer
- **p.** → 1.17.2 Door Open
- **q.** → 1.17.6 Direct Inward System Access (DISA)
- **r.** → 1.21.1.4 Call Forwarding (CF)—by ISDN (P-MP)
- **s.** → 1.25.5 Wireless XDP Parallel Mode
- **t.** → 1.26.1.1 Station Message Detail Recording (SMDR)—SMDR for Outgoing Trunk Calls
- **u.** → 2.2.4 Time Service—Time Service Switching
- **v.** → 2.2.6 Manager Features
- **w.** → 2.3.2 PT Programming

**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**

4.12 [2-7-1] Class of Service—COS Settings
6.1 [4-1-1] Wired Extension—Extension Settings—Main—◆ COS
6.10 [4-2-1] Portable Station—Extension Settings—Main—◆ COS
PT Programming Manual References

2.1.8 COS Programming
[602] Class of Service

Feature Guide References

3.1 Capacity of System Resources

User Manual References

1.2.7 Calling without Restrictions
2.2.2 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.
   → 5.1 [3-1-1] Trunk Group—TRG Settings
   → [402] LCOT/BRI Trunk Group Number
   One trunk can belong to only one trunk group on a port or channel basis.

   Port basis: LCOT/ISDN-BRI/ISDN-PRI23/ISDN-PRI30
   Channel basis: E1/T1

2. Extension User Group
   The PBX supports extension user groups, each of which is used to compose the following groups:
   a. Tenant (→ 2.2.3 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.
   → 5.7 [3-2] User Group
   → [603] Extension User Group
   Assignable Extensions: PT/SLT/PS/SIP Extension/ISDN Extension/T1-OPX

[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. (→ 1.4.1.3 Call Pickup)
   → 5.8 [3-3] Call Pickup Group
   → 5.9 [3-3] 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.

[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.

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) and name. One extension can belong to multiple groups.

5. VM Group
There are two types of VM groups as follows:
### 2.2.2 Group

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>VM (DTMF) Group</td>
<td>A group of SLT ports which use the Voice Mail DTMF Integration features.</td>
</tr>
<tr>
<td></td>
<td>One SLT port can belong to only one group.</td>
</tr>
<tr>
<td>VM (DPT) Group</td>
<td>A group of DPT ports which use the Voice Mail DPT (Digital) Integration features.</td>
</tr>
<tr>
<td></td>
<td>One DPT port can belong to only one group.</td>
</tr>
</tbody>
</table>

(→ 1.24.2.1 Voice Mail (VM) Group)  
→ 5.19 [3-7-1] VM(DPT) Group—System Settings  
→ 5.20 [3-7-2] VM(DPT) Group—Unit Settings  
→ 5.22 [3-8-1] VM(DTMF) Group—System Settings  
→ 5.23 [3-8-2] VM(DTMF) Group—Group Settings  
→ 5.24 [3-8-2] VM(DTMF) Group—Group Settings—Member List  
→ 5.25 [3-9] PS Ring Group

#### [Example]

<table>
<thead>
<tr>
<th>PBX</th>
</tr>
</thead>
<tbody>
<tr>
<td>DPT Port</td>
</tr>
<tr>
<td>SLT Port</td>
</tr>
<tr>
<td>DPT Port</td>
</tr>
<tr>
<td>DPT Port</td>
</tr>
<tr>
<td>DPT Port</td>
</tr>
<tr>
<td>DPT Port</td>
</tr>
<tr>
<td>SLT Port</td>
</tr>
<tr>
<td>SLT Port</td>
</tr>
<tr>
<td>SLT Port</td>
</tr>
<tr>
<td>SLT Port</td>
</tr>
</tbody>
</table>

![Diagram of PBX with VM (DPT) and VM (DTMF) Groups]

#### 6. 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.

(→ 1.25.2 PS Ring Group)  
→ 5.25 [3-9] PS Ring Group  
→ 5.26 [3-9] PS Ring Group—Member List

![Diagram of PS Ring Groups]

---

**[660] VM Group Floating Extension Number**

**[Example]**
2.2.2 Group

PC Programming Manual References
Section 5 [3] Group

PT Programming Manual References
[402] LCOT/BRI 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] VM Group Floating Extension Number
[680] Idle Extension Hunting Type
[681] Idle Extension Hunting Group Member

Feature Guide References
3.1 Capacity of System Resources
2.2.3 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.
   (→ 2.2.2 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.
   (→ 2.2.4 Time Service)

   [Example]
   Tenant 1
   Use Time Table 1
   Extension User Group 1
   Extension User Group 2
   Extension User Group 3
   Extension User Group 4

   Tenant 2
   Use Time Table 2
   Extension User Group 5
   Extension User Group 6

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) (→ 2.2.5 Operator Features)
      → 8.6 [6-6] Tenant— Operator (Extension Number)
   b. ARS Mode (Off/ Local Access/ All Access/ System) (→ 1.10.1 Automatic Route Selection (ARS))
      → 8.6 [6-6] Tenant— ARS Mode
   c. Music Source for Music on Hold (System/ BGM Number/ Cyclic Tone)
      (→ 1.13.4 Music on Hold)
      → 8.6 [6-6] Tenant— Music On Hold
   d. System Speed Dialling (System/ Tenant Exclusive)
      (→ 1.6.1.5 Speed Dialling— Personal/ System)
      → 8.6 [6-6] Tenant— System Speed Dial

   [Programming Example]

<table>
<thead>
<tr>
<th>Tenant No.</th>
<th>Operator</th>
<th>ARS Mode</th>
<th>Music Source</th>
<th>System Speed Dialling</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Extn.101</td>
<td>Local Access</td>
<td>System(^3)</td>
<td>System(^4)</td>
</tr>
<tr>
<td>2</td>
<td>None(^1)</td>
<td>System(^2)</td>
<td>Cyclic Tone</td>
<td>Extended/ Tenant Exclusive</td>
</tr>
<tr>
<td>3</td>
<td>Floating extn. no. 200</td>
<td>Off</td>
<td>BGM1</td>
<td>Extended/ Tenant Exclusive</td>
</tr>
<tr>
<td>Tenant No.</td>
<td>Operator</td>
<td>ARS Mode</td>
<td>Music Source</td>
<td>System Speed Dialling</td>
</tr>
<tr>
<td>-----------</td>
<td>----------</td>
<td>----------</td>
<td>--------------</td>
<td>-----------------------</td>
</tr>
</tbody>
</table>

*1 Follows the system assignment of a PBX operator.
   → [006] Operator Assignment

*2 Follows the system assignment of the ARS mode.
   → Section 10 [8] ARS
   → [320] ARS Mode

*3 Follows the system assignment of the music source for the Music on Hold.
   → [711] Music on Hold

*4 Follows the system assignment for System Speed Dialling.
   → 8.1 [6-1] System Speed Dial
   → [001] System Speed Dialling Number

### Conditions

- **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 (→ 1.1.2.2 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]
2.2.3 Tenant Service

[Programming Example]

| Caller | 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 (→ 1.2.2.1 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

4.4 [2-2] Operator & BGM
   →◆ PBX Operator—Day, Lunch, Break, Night
   →◆ BGM and Music on Hold—Music on Hold
4.6 [2-4] Week Table
4.8 [2-5] Holiday Table
4.14 [2-7-3] Class of Service—Internal Call Block
5.7 [3-2] User Group
5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Main—◆ Tenant Number
6.1 [4-1-1] Wired Extension—Extension Settings—Main—◆ User Group
6.10 [4-2-1] Portable Station—Extension Settings—Main—◆ User Group
7.1 [5-1] Doorphone—◆ Tenant Number
8.1 [6-1] System Speed Dial
8.6 [6-6] Tenant
Section 10 [8] ARS
10.6 [8-5] Carrier—Authorisation Code for Tenant
2.2.3 Tenant Service

12.2 [10-2] DIL Table & Port Settings—DIL—◆ Tenant Number
12.3 [10-3] DDI / DID Table—◆ Tenant Number
12.6 [10-4] MSN Table—MSN—◆ Tenant Number

PT Programming Manual References

[001] System Speed Dialling Number
[006] Operator Assignment
[320] ARS Mode
[711] Music on Hold

Feature Guide References

3.1 Capacity of System Resources
2.2.4 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.

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Automatic</td>
<td>The PBX will switch mode according to the preprogrammed Time Table.</td>
</tr>
<tr>
<td>Manual</td>
<td>A manager, or preprogrammed extension on a COS basis can switch mode by dialling the feature number or pressing the Time Service button.</td>
</tr>
</tbody>
</table>

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]

<table>
<thead>
<tr>
<th>Time Schedule</th>
<th>Time Table No. (Tenant No.)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1</td>
</tr>
<tr>
<td>MON</td>
<td>Day 1 start</td>
</tr>
<tr>
<td></td>
<td>Lunch start</td>
</tr>
<tr>
<td></td>
<td>Day 2 start</td>
</tr>
<tr>
<td></td>
<td>Break 1 start</td>
</tr>
<tr>
<td></td>
<td>Break 1 end (Day restart)</td>
</tr>
<tr>
<td></td>
<td>Night start</td>
</tr>
<tr>
<td>TUE</td>
<td>Day 1 start</td>
</tr>
<tr>
<td></td>
<td>Lunch start</td>
</tr>
<tr>
<td></td>
<td>Day 2 start</td>
</tr>
<tr>
<td></td>
<td>Break 1 start</td>
</tr>
<tr>
<td></td>
<td>Break 1 end (Day restart)</td>
</tr>
<tr>
<td></td>
<td>Night start</td>
</tr>
</tbody>
</table>

<Time Service Image of Monday>

<table>
<thead>
<tr>
<th>Time Table No.</th>
<th>00:00</th>
<th>08:00</th>
<th>11:00</th>
<th>12:00</th>
<th>01:00</th>
<th>13:00</th>
<th>16:00</th>
<th>20:00</th>
<th>24:00</th>
<th>08:00</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Night</td>
<td>Day 1</td>
<td>Lunch</td>
<td>Day 2</td>
<td>Night</td>
<td>Day 1</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>Night</td>
<td>Day 1</td>
<td></td>
<td></td>
<td>Night</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>Night</td>
<td>Day 1</td>
<td>Night</td>
<td>Lunch</td>
<td>Day 1</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>Night</td>
<td>Day 1</td>
<td></td>
<td>Lunch</td>
<td>Day 1</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
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/MSN) (→ 1.1.1 Incoming Trunk Call Features)
- b. Destination of the Intercept Routing (→ 1.1.1.6 Intercept Routing)
- c. Queuing Time Table for incoming call distribution groups (→ 1.2.2.4 Queuing Feature)
- d. Overflow destination for incoming call distribution groups (→ 1.2.2.6 Overflow Feature)
- e. Destination of incoming doorphone calls (→ 1.17.1 Doorphone Call)
- f. PBX operator (→ 2.2.5 Operator Features)
- g. COS for TRS/Barring and for Trunk Access
- h. Outgoing Message (OGM) for Timed Reminder (→ 1.28.4 Timed Reminder)
- i. Intercept time for Intercept Routing—No Answer (→ 1.1.1.6 Intercept Routing) and for DISA Intercept Routing—No Answer (→ 1.17.6 Direct Inward System Access (DISA))

**[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>

<table>
<thead>
<tr>
<th>Location</th>
<th>DID/DDI No.</th>
<th>Tenant (Time Table) No.</th>
<th>DID/DDI Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>Day</td>
</tr>
<tr>
<td>001</td>
<td>123-4567</td>
<td>1</td>
<td>105</td>
</tr>
<tr>
<td>002</td>
<td>123-2468</td>
<td>1</td>
<td>102</td>
</tr>
</tbody>
</table>

### <DIL Table>

<table>
<thead>
<tr>
<th>Trunk No.</th>
<th>Tenant (Time Table) No.</th>
<th>DIL Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Day</td>
</tr>
<tr>
<td>01</td>
<td>1</td>
<td>101</td>
</tr>
<tr>
<td>02</td>
<td>2</td>
<td>102</td>
</tr>
</tbody>
</table>

**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 the extension 100 (VPS).

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
2.2.4 Time Service

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:

<table>
<thead>
<tr>
<th>Light Pattern</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td>Day mode</td>
</tr>
<tr>
<td>Red on</td>
<td>Night mode</td>
</tr>
<tr>
<td>Green on</td>
<td>Lunch mode</td>
</tr>
<tr>
<td>Slow green flashing</td>
<td>Break mode</td>
</tr>
<tr>
<td>Slow red flashing</td>
<td>Holiday mode</td>
</tr>
</tbody>
</table>

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

- PT programming can set the Start and/or End time of the following:
  - Day-1 (Day Start time)
  - Lunch (Lunch Start time)
  - Day-2 (Lunch End time)
  - Night (Night Start time)

- PC programming can also set the following three time periods for break mode per day:
  - 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

4.6 [2-4] Week Table
4.8 [2-5] Holiday Table
4.9 [2-6-1] Numbering Plan—Main—Features—◆ Time Service (Day / Lunch / Break / Night) Switch
4.12 [2-7-1] Class of Service—COS Settings—Programming & Manager—◆ Time Service Switch
6.5 [4-1-4] 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)
6.13 [4-2-3] Portable Station—Flexible Button
  ➔◆ Type
  ➔◆ Parameter Selection (for Time Service)
  ➔◆ Parameter Selection (for Time Service - Automatic/Manual)
2.2.4 Time Service

→◆ Optional Parameter (or Ringing Tone Type Number) (for Time Service)
6.17 [4-3] DSS Console
   →◆ Type
   →◆ Parameter Selection (for Time Service)
   →◆ Parameter Selection (for Time Service - Automatic/Manual)
   →◆ Optional Parameter (Ringing Tone Type Number) (for Time Service)

PT Programming Manual References

[101] Time Service Switching Mode
[102] Time Service Starting Time
[514] Time Service Manual Switching

Feature Guide References

1.9.1 Toll Restriction (TRS)/Call Barring (Barring)
1.20.2 Flexible Buttons
2.2.1 Class of Service (COS)
2.2.3 Tenant Service
3.1 Capacity of System Resources

User Manual References

1.8.11 Checking the Time Service Status
2.1.2 Time Service Mode Control
2.2.5 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:

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>PBX operator</td>
<td>An extension or incoming call distribution group can be assigned as a PBX operator for each time mode (day/lunch/break/night).</td>
</tr>
<tr>
<td>Tenant operator</td>
<td>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.</td>
</tr>
</tbody>
</table>

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
8.6 [6-6] Tenant— ◆ Operator (Extension Number)

PT Programming Manual References
[006] Operator Assignment

Feature Guide References
2.2.3 Tenant Service
3.1 Capacity of System Resources

User Manual References
1.2.1 Basic Calling
1.11.1 Using the Telephones in a Hotel-type Environment (Hospitality Features)
### 2.2.6 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:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description &amp; Reference</th>
<th>Manager Password</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Manager Programming</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Manager Password Change</td>
<td>Changes the manager password.</td>
<td>Required</td>
</tr>
<tr>
<td><strong>Call Charge Management</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sets, displays, clears, and prints the call charge data.</td>
<td></td>
<td>Required</td>
</tr>
<tr>
<td>→ 1.26.3 Call Charge Services</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Verification Code Personal Identification Number (PIN) Set</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sets a verification code PIN for each verification code.</td>
<td></td>
<td>Required</td>
</tr>
<tr>
<td>→ 1.9.6 Verification Code Entry</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Remote PIN Clear</strong></td>
<td>Clears the extension PIN of an extension remotely, and a verification code PIN. PIN Lock is also unlocked.</td>
<td>Required</td>
</tr>
<tr>
<td>→ 1.28.1 Extension Personal Identification Number (PIN)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>→ 1.9.6 Verification Code Entry</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Remote Extension Dial Lock</strong></td>
<td>Sets or cancels the Extension Dial Lock on an extension remotely.</td>
<td>Required</td>
</tr>
<tr>
<td>→ 1.9.3 Extension Dial Lock</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Dial Tone Transfer</strong></td>
<td>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).</td>
<td>Not required</td>
</tr>
<tr>
<td>→ 1.9.4 Dial Tone Transfer</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Outgoing Message (OGM)</strong></td>
<td>Records and plays back outgoing messages (OGMs).</td>
<td>Not required</td>
</tr>
<tr>
<td>→ 1.17.5 Outgoing Message (OGM)</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Time Service</strong></td>
<td>Switches the time mode (day/lunch/break/night) manually.</td>
<td>Not required</td>
</tr>
<tr>
<td>→ 2.2.4 Time Service</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>BGM—External</strong></td>
<td>Sets the External BGM on and off.</td>
<td>Not required</td>
</tr>
<tr>
<td>→ 1.17.4 Background Music (BGM)</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
### 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:

- Keeping PINs secret.
- Selecting complex, random PINs that cannot be easily guessed.
- Changing PINs regularly.

- **Manager Password**
  
  One manager password can be assigned per PBX.

### PC Programming Manual References

4.12 [2-7-1] Class of Service—COS Settings—Programming & Manager— Manager


### PT Programming Manual References

[112] Manager Password  
[511] Manager Assignment

### Feature Guide References

2.2.1 Class of Service (COS)

### User Manual References

2.1 Control Features

3.2.2 Manager Programming
2.3 System Data Control

2.3.1 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:**

<table>
<thead>
<tr>
<th>Method</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Using the Serial Interface (RS-232C) port</td>
<td>The PBX has a Serial Interface (RS-232C) port which can be used for either system administration or SMDR (→ 1.26.1.1 Station Message Detail Recording (SMDR)).</td>
</tr>
<tr>
<td>Using the USB port</td>
<td>The PC is connected to a USB port (USB Module) attached to a DPT.</td>
</tr>
<tr>
<td>Using the LAN interface</td>
<td>Available via the MNT port of the IPCMPR card.</td>
</tr>
<tr>
<td>Using a modem through an SLT port*1</td>
<td>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.</td>
</tr>
<tr>
<td>Using an ISDN TA interface (64 kbps) through an ISDN Extension Line*1</td>
<td>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.</td>
</tr>
</tbody>
</table>

*1 If remote access is disabled through system programming, then this on site programming cannot be done.

2. **Remote Programming:**

<table>
<thead>
<tr>
<th>Method</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Using a modem (RMT card)</td>
<td>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:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Direct Access</strong></td>
</tr>
<tr>
<td></td>
<td>Dial the DIL/DID/DDI number whose destination is the floating extension number of the analogue remote maintenance.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Through DISA</strong></td>
</tr>
<tr>
<td></td>
<td>Dial the floating extension number of the analogue remote maintenance using the DISA feature. (→ 1.17.6 Direct Inward System Access (DISA))</td>
</tr>
<tr>
<td></td>
<td>• <strong>Call Transfer</strong></td>
</tr>
<tr>
<td></td>
<td>Call an extension (probably the operator), and request a transfer to the floating extension number of the analogue remote maintenance. (→ 1.12.1 Call Transfer)</td>
</tr>
</tbody>
</table>
Method | Description
--- | ---
**Using an external modem (Not RMT card)** | An external modem can be used instead of an RMT card to perform the remote programming. An external modem can be connected to the Serial Interface (RS-232C) of the PBX. Connect the modem to an extension port which is assigned as the DIL/DID/DDI/MSN destination or to a trunk directly to connect to the PBX from the PC. An AT command can be sent to the modem automatically when it is connected to the Serial Interface (RS-232C) port. AT commands can be programmed beforehand through system programming to initialise the modem. An AT command can also be sent manually in the PT system programming mode.

**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/MSN 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.

**Conditions**
- Only one system programming session can be in progress at a time, whether using a PC or a PT.
- **Programmer Code and Password for System Programming**
  To access system programming, a programmer code and its password must be entered. For more details, see Section 2.3.3 Password Security in this Feature Guide.
- **Remote Programming Limitation**
  It is possible to disable remote access through system programming.

**Installation Manual References**
- 3.3.1 IPCMPR Card
- 3.3.3 RMT Card (KX-TDA0196)
- 3.13.1 Connection of Peripherals
- 4.1 Overview
- 4.2 PC Connection
- 4.3 Installation of the Maintenance Console

**PC Programming Manual References**
  - →Remote
  - →Password— System Password - PC Programming—User Level
  - →Password— System Password - PC Programming—Administrator Level
  - →Password— System Password - PC Programming—Installer Level

**PT Programming Manual References**
- [801] External Modem Control
- [810] Remote Programming
- [811] Modem Floating Extension Number
- [812] ISDN Remote Floating Extension Number
Feature Guide References

1.1.1.2 Direct In Line (DIL)
1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)
1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service
2.3.2 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
- 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 details, see Section 2.3.3 Password Security in this Feature Guide.
- **Personal Programming Data Default Set**
  A user can return all of the items programmed on the telephone to default.

PC Programming Manual References

4.12 [2-7-1] Class of Service—COS Settings—Programming & Manager—Programming Mode Level
  ➔ System Password - PT Programming—Prog ** : User Level
  ➔ System Password - PT Programming—Prog # : Administrator Level
  ➔ Manager Password - PT Programming—Prog *1

PT Programming Manual References

2.1 PT Programming
[516] Programming Mode Limitation

Feature Guide References

2.2.1 Class of Service (COS)
2.2.6 Manager Features
3.1 Capacity of System Resources
User Manual References

3.1 Customising Your Phone (Personal Programming)
3.2 Manager Programming
3.3 Customising Your System (System Programming)
2.3.3 Password Security

Description

To maintain system security, system passwords are required to access certain programming functions of the PBX. By giving different users access to different passwords, it is possible to control the amount of programming that each user is able to perform.

The following types of system passwords are available:

<table>
<thead>
<tr>
<th>Password</th>
<th>Description</th>
<th>Format</th>
</tr>
</thead>
<tbody>
<tr>
<td>PT</td>
<td>System Password (PT) for User</td>
<td>4 – 10 digits</td>
</tr>
<tr>
<td></td>
<td>Used to access user-level PT programming. The specific PT programming items</td>
<td></td>
</tr>
<tr>
<td></td>
<td>that may be programmed at user level can be selected through system</td>
<td></td>
</tr>
<tr>
<td></td>
<td>programming item.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>System Password (PT) for Administrator</td>
<td>4 – 10 digits</td>
</tr>
<tr>
<td></td>
<td>Used to access administrator-level PT programming. All PT programming</td>
<td></td>
</tr>
<tr>
<td></td>
<td>settings are available.</td>
<td></td>
</tr>
<tr>
<td>PC</td>
<td>System Password (PC) for User</td>
<td>4 – 10 cahrac-</td>
</tr>
<tr>
<td></td>
<td>Used with the user-level programmer code to access user-level PC programming.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>The installer can specify which system programming settings are available.</td>
<td>ters</td>
</tr>
<tr>
<td></td>
<td>System Password (PC) for Administrator</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Used with the administrator-level programmer code to access</td>
<td></td>
</tr>
<tr>
<td></td>
<td>administrator-level PC programming. The installer can</td>
<td></td>
</tr>
<tr>
<td></td>
<td>specify which system programming settings are available.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>System Password (PC) for Installer</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Used with the installer-level programmer code to access installer-level</td>
<td></td>
</tr>
<tr>
<td></td>
<td>PC programming. All system programming settings are available.</td>
<td></td>
</tr>
</tbody>
</table>

The three programmer codes used for PC programming can be set through Maintenance Console. For more information about programmer codes, see Section 2.1.2 Access Levels in the PC Programming Manual. The required format of each code is as follows:

<table>
<thead>
<tr>
<th>Item</th>
<th>Length</th>
</tr>
</thead>
<tbody>
<tr>
<td>User Level Programmer Code</td>
<td>0 – 16 characters</td>
</tr>
<tr>
<td>Administrator Level Programmer Code</td>
<td>4 – 16 characters</td>
</tr>
<tr>
<td>Installer Level Programmer Code</td>
<td>4 – 16 characters</td>
</tr>
</tbody>
</table>

CAUTION

To the Administrator or Installer regarding the system password

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 default passwords preset. For security, change these passwords the first time that you programme the PBX.
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. For a list of numbers and characters that can be used in system passwords, refer to Section 1.1.2 Entering Characters in the PC Programming Manual.
2.3.3 Password Security

PC Programming Manual References

2.1.2 Access Levels

PT Programming Manual References

None

Feature Guide References

3.1 Capacity of System Resources
2.3.4 Quick Setup

Description

It is possible to set up the basic PBX parameters using a PC. When a PC accesses the PBX for the first time with the Installer Level Programmer Code (→ 2.3.1 PC Programming), Quick Setup display will appear automatically. The following items can be programmed as desired:

<table>
<thead>
<tr>
<th>Item</th>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Date &amp; Time</td>
<td>Year/Month/Date/ Hour/ Minute</td>
<td>The date and time set on the PC will be used.</td>
</tr>
<tr>
<td>System Password</td>
<td>4–10 characters</td>
<td>Enter the system password for installer (→ 2.3.3 Password Security).</td>
</tr>
<tr>
<td>Operator &amp; Manager</td>
<td>Extension Number</td>
<td>Assign the PBX operator to all time modes (day/lunch/break/night) (→ 2.2.5 Operator Features). The extension assigned as the PBX operators is automatically allowed to perform manager operation (→ 2.2.6 Manager Features).</td>
</tr>
<tr>
<td>Flexible Numbering Feature</td>
<td>1. Type 1 (with ×)</td>
<td>If “Type 1 (with ×)” is selected, “×” must prefix all feature numbers (except access numbers) when an extension user wants to use a feature. [Example] Call Pickup feature number Type 1 (with ×): ×41 Type 2 (without ×): 41 For default of the flexible numbers, refer to the [Flexible Numbering Table (available while a dial tone is heard)] (→ 2.3.6 Flexible Numbering/Fixed Numbering).</td>
</tr>
<tr>
<td>Extension</td>
<td>From 101/from 201</td>
<td>Determines the number from which default extension numbers begin.</td>
</tr>
<tr>
<td>Operator</td>
<td>0/9</td>
<td>The feature number for Operator Call can be selected.</td>
</tr>
<tr>
<td>CO Access</td>
<td>0/1/9</td>
<td>The feature number for Idle Line Access can be selected.</td>
</tr>
<tr>
<td>Remote Maintenance Number</td>
<td>Remote Maintenance Dial Number</td>
<td>Enter the complete telephone number of the PBX (including the country code). When necessary, this number will be used to access the PBX from a remote location for maintenance purposes.</td>
</tr>
<tr>
<td>DHCP Client</td>
<td>Enable/Disable</td>
<td>Determines whether to use a DHCP server to assign IP address information automatically or not.</td>
</tr>
<tr>
<td>LAN Connection(^1)</td>
<td>• IP Address for IPCMPR Card</td>
<td>Manually enter the LAN information when not using a DHCP server.</td>
</tr>
<tr>
<td></td>
<td>• IP Address for VoIP-DSP</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Subnet Mask</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Default Gateway</td>
<td></td>
</tr>
</tbody>
</table>

\(^1\) If the DHCP Client setting is set to Enable, this parameter can not be set.

Note that the programmable items may vary depending on the version of Maintenance Console being used.
2.3.5 Automatic Setup

Description
There are two automatic setup features as follows:
1. Automatic ISDN Configuration
2. Automatic Time Adjustment

1. Automatic ISDN Configuration
The ISDN (BRI) port configuration can be set automatically through system programming.
The following items will be set by making and receiving a call using the subscriber number assigned for each ISDN (BRI) port:
   a. L1 Active Mode
   b. L2 Data Link Mode
   c. Access Mode (Point-to-Point/Point-to-Multipoint)
   d. TEI Assignment Mode (Fix 00–63/Automatic)

2. 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 everyday 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 Simple Network Time Protocol (SNTP):
By connecting the PBX to an SNTP server, it is possible to receive and update the time setting.
The PBX clock will be adjusted everyday at 3:05 AM, if enabled through system programming.

Conditions
[General]
• Through system programming, it is possible to specify SNTP, 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. (→ 1.26.1.1 Station Message Detail Recording (SMDR))
2.3.5 Automatic Setup

[SNTP Time Information]
- The time set through SNTP will apply the same to all PTs and IP-PTs connected to the PBX, regardless if an IP extension is located in another time zone.

PC Programming Manual References
2.5 Tool—2.5.2 Tool—BRI Automatic Configuration
4.1 [2-1-1] Date & Time—Date & Time Setting
4.2 [2-1-2] Date & Time—SNTP / Daylight Saving
4.3 [2-1-2] Date & Time—SNTP / Daylight Saving—Daylight Saving

PT Programming Manual References
None

Feature Guide References
1.21.1.1 Integrated Services Digital Network (ISDN)—SUMMARY
1.28.4 Timed Reminder
2.3.6 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 default (Type 1 or Type 2) 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", "x", 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", "x", and "#"

[Flexible Numbering Table (available while a dial tone is heard)]

<table>
<thead>
<tr>
<th>Feature</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Extension Numbering Scheme 1—Leading Number</td>
<td>1 / 2</td>
</tr>
<tr>
<td>Extension Numbering Scheme 2—Leading Number</td>
<td>2 / 3</td>
</tr>
<tr>
<td>Extension Numbering Scheme 3—Leading Number</td>
<td>3 / 4</td>
</tr>
<tr>
<td>Extension Numbering Scheme 4—Leading Number</td>
<td>4 / 1</td>
</tr>
<tr>
<td>Extension Numbering Scheme 5—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 6—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 7—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 8—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 9—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 10—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 11—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 12—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 13—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 14—Leading Number</td>
<td>None</td>
</tr>
</tbody>
</table>

Type 1 (with #) | Type 2 (without #)

10 | 11 | 12 | 13 | 14 | 15 | 16 | 17 | 18 | 19 | 20 | 21 | 22 | 23 |
## 2.3.6 Flexible Numbering/Fixed Numbering

<table>
<thead>
<tr>
<th>Feature</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Type 1 (with *)</td>
</tr>
<tr>
<td>Extension Numbering Scheme 15—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 16—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 17—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 18—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 19—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 20—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Extension Numbering Scheme 21—Leading Number</td>
<td>5</td>
</tr>
<tr>
<td>Extension Numbering Scheme 22—Leading Number</td>
<td>6</td>
</tr>
<tr>
<td>Extension Numbering Scheme 23–64—Leading Number</td>
<td>None</td>
</tr>
<tr>
<td>Operator Call</td>
<td>9 / 0</td>
</tr>
<tr>
<td>Idle Line Access (Local Access)</td>
<td>0 / 1 / 9</td>
</tr>
<tr>
<td>Trunk Group Access</td>
<td>8</td>
</tr>
<tr>
<td>TIE Line Access</td>
<td>7</td>
</tr>
<tr>
<td>Redial</td>
<td>#</td>
</tr>
<tr>
<td>Speed Dialling—System/Personal</td>
<td>* *</td>
</tr>
<tr>
<td>Personal Speed Dialling—Programming</td>
<td>*30</td>
</tr>
<tr>
<td>Doorphone Call</td>
<td>*31</td>
</tr>
<tr>
<td>Broadcasting</td>
<td>*32</td>
</tr>
<tr>
<td>Group Paging</td>
<td>*33</td>
</tr>
<tr>
<td>External BGM on/off</td>
<td>*35</td>
</tr>
<tr>
<td>Outgoing Message (OGM) playback/record/clear</td>
<td>*36</td>
</tr>
<tr>
<td>S-CO Line Access</td>
<td>*37</td>
</tr>
<tr>
<td>SVM—Personal Greeting Message playback/record/clear</td>
<td>*38</td>
</tr>
<tr>
<td>Parallel Telephone Mode set/cancel</td>
<td>*39</td>
</tr>
<tr>
<td>Group Call Pickup</td>
<td>*40</td>
</tr>
<tr>
<td>Directed Call Pickup</td>
<td>*41</td>
</tr>
<tr>
<td>TAFAS—Calls through an External Pager</td>
<td>*42</td>
</tr>
<tr>
<td>Group Paging answer</td>
<td>*43</td>
</tr>
<tr>
<td>Automatic Callback Busy cancel/CCBS cancel</td>
<td>*46</td>
</tr>
<tr>
<td>User Remote Operation/Walking COS/Verification Code Entry</td>
<td>*47</td>
</tr>
<tr>
<td>Wireless XDP Parallel Mode set/cancel</td>
<td>*48</td>
</tr>
<tr>
<td>Feature</td>
<td>Default</td>
</tr>
<tr>
<td>-----------------------------------------------------------------------</td>
<td>-------------------</td>
</tr>
<tr>
<td></td>
<td>Type 1 (with √)</td>
</tr>
<tr>
<td>Account Code Entry</td>
<td>49 (with √)</td>
</tr>
<tr>
<td>Call Hold/Call Hold Retrieve</td>
<td>50 (with √)</td>
</tr>
<tr>
<td>Call Hold Retrieve—Specified with a Holding Extension Number</td>
<td>51 (with √)</td>
</tr>
<tr>
<td>Call Park/Call Park Retrieve</td>
<td>52 (with √)</td>
</tr>
<tr>
<td>Call Hold Retrieve—Specified with a Held Trunk Number</td>
<td>53 (with √)</td>
</tr>
<tr>
<td>Door Open</td>
<td>55 (with √)</td>
</tr>
<tr>
<td>External Relay</td>
<td>56 (with √)</td>
</tr>
<tr>
<td>External Feature Access</td>
<td>60 (with √)</td>
</tr>
<tr>
<td>ISDN Hold</td>
<td>62 (with √)</td>
</tr>
<tr>
<td>Preventing (not showing) the telephone number when making outside calls (KX-HGT100 only)(^1)</td>
<td>67 (with √)</td>
</tr>
<tr>
<td>Showing the telephone number when making outside calls (KX-HGT100 only)(^1)</td>
<td>68 (with √)</td>
</tr>
<tr>
<td>COLR set/cancel</td>
<td>7×0 (with √)</td>
</tr>
<tr>
<td>CLIR set/cancel</td>
<td>7×1 (with √)</td>
</tr>
<tr>
<td>Switch CLIP/COLP of the Trunk/Extension</td>
<td>7×2 (with √)</td>
</tr>
<tr>
<td>MCID</td>
<td>7×3 (with √)</td>
</tr>
<tr>
<td>ISDN-FWD set/cancel/confirm</td>
<td>7×5 (with √)</td>
</tr>
<tr>
<td>Message Waiting set/cancel/callback</td>
<td>70 (with √)</td>
</tr>
<tr>
<td>FWD/DND set/cancel—Both</td>
<td>710 (with √)</td>
</tr>
<tr>
<td>FWD/DND set/cancel—External</td>
<td>711 (with √)</td>
</tr>
<tr>
<td>FWD/DND set/cancel—Internal</td>
<td>712 (with √)</td>
</tr>
<tr>
<td>FWD/DND No Answer Timer set</td>
<td>713 (with √)</td>
</tr>
<tr>
<td>Group FWD set/cancel—Both</td>
<td>714 (with √)</td>
</tr>
<tr>
<td>Group FWD set/cancel—External</td>
<td>715 (with √)</td>
</tr>
<tr>
<td>Group FWD set/cancel—Internal</td>
<td>716 (with √)</td>
</tr>
<tr>
<td>Call Pickup Deny set/cancel</td>
<td>720 (with √)</td>
</tr>
<tr>
<td>Paging Deny set/cancel</td>
<td>721 (with √)</td>
</tr>
<tr>
<td>Walking Extension/Enhanced Walking Extension</td>
<td>727 (with √)</td>
</tr>
<tr>
<td>Data Line Security set/cancel</td>
<td>730 (with √)</td>
</tr>
<tr>
<td>Manual Call Waiting for Extension Call off/BSS/OHCA/Whisper OHCA</td>
<td>731 (with √)</td>
</tr>
<tr>
<td>Automatic Call Waiting set/cancel</td>
<td>732 (with √)</td>
</tr>
</tbody>
</table>
### 2.3.6 Flexible Numbering/Fixed Numbering

<table>
<thead>
<tr>
<th>Feature</th>
<th>Type 1 (with *)</th>
<th>Type 2 (without *)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Executive Busy Override Deny set/cancel</td>
<td>733</td>
<td>733</td>
</tr>
<tr>
<td>Not Ready Mode on/off</td>
<td>735</td>
<td>735</td>
</tr>
<tr>
<td>Log-in/Log-out</td>
<td>736</td>
<td>736</td>
</tr>
<tr>
<td>Incoming Call Queue Monitor</td>
<td>739</td>
<td>739</td>
</tr>
<tr>
<td>Hot Line programme/set/cancel</td>
<td>740</td>
<td>740</td>
</tr>
<tr>
<td>Absent Message set/cancel</td>
<td>750</td>
<td>750</td>
</tr>
<tr>
<td>BGM set/cancel</td>
<td>751</td>
<td>751</td>
</tr>
<tr>
<td>Remote Wake-up Call</td>
<td>76</td>
<td>76</td>
</tr>
<tr>
<td>Timed Reminder set/cancel</td>
<td>760</td>
<td>760</td>
</tr>
<tr>
<td>Printing Message</td>
<td>761</td>
<td>761</td>
</tr>
<tr>
<td>Extension Dial Lock set/cancel</td>
<td>77</td>
<td>77</td>
</tr>
<tr>
<td>Time Service Switch</td>
<td>780</td>
<td>780</td>
</tr>
<tr>
<td>Remote Extension Dial Lock off</td>
<td>782</td>
<td>782</td>
</tr>
<tr>
<td>Remote Extension Dial Lock on</td>
<td>783</td>
<td>783</td>
</tr>
<tr>
<td>NDSS Monitor Release</td>
<td>784</td>
<td>784</td>
</tr>
<tr>
<td>Trunk Busy Out Clear</td>
<td>785</td>
<td>None</td>
</tr>
<tr>
<td>Extension Feature Clear</td>
<td>790</td>
<td>790</td>
</tr>
<tr>
<td>Extension Personal Identification Number (PIN) set/cancel</td>
<td>799</td>
<td>799</td>
</tr>
<tr>
<td>Dial Information (CTI)</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>Other PBX Extension Number (TIE) 1–16</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>Quick Dialling(^2)</td>
<td>None</td>
<td>None</td>
</tr>
</tbody>
</table>

\(^1\) Settings for the KX-HGT100 are unrelated to PBX feature numbers and are set directly via the telephone. If the number set to the telephone is also used as a PBX feature number, the number functions as the telephone’s feature number, and not as a PBX feature number. See 4.3.1 Registering IP Telephones in the IP Networking Guide for programming details.

\(^2\) 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:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Waiting/DND Override</td>
<td>1 or 2(^1)</td>
</tr>
<tr>
<td>Executive Busy Override</td>
<td>3</td>
</tr>
</tbody>
</table>

\(^1\) 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.
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:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Message Waiting set</td>
<td>4</td>
</tr>
<tr>
<td>Call Monitor</td>
<td>5</td>
</tr>
<tr>
<td>Automatic Callback Busy/CCBS</td>
<td>6</td>
</tr>
<tr>
<td>Alternate Calling—Ring/Voice</td>
<td>¥</td>
</tr>
</tbody>
</table>

* To use Call Waiting/DND Override, both “1” and “2” are available by default.

### 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
2.3.6 Flexible Numbering/Fixed Numbering

- Walking Extension
- BGM set/cancel
- Timed Reminder

• A PS can only be assigned an extension number of 4 digits or less.

PC Programming Manual References
4.9 [2-6-1] Numbering Plan—Main
→Extension
→Features
→Other PBX Extension
4.10 [2-6-2] Numbering Plan—Quick Dial
4.11 [2-6-3] Numbering Plan—B/NA DND Call Feature

PT Programming Manual References
[100] Flexible Numbering

Feature Guide References
1.24.2.1 Voice Mail (VM) Group
1.25.1 Portable Station (PS) Connection
3.1 Capacity of System Resources

User Manual References
4.2.1 Feature Number Table
2.3.7 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.

<table>
<thead>
<tr>
<th>Resource</th>
<th>Description</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device</td>
<td>External Pager</td>
<td>Used as the destination for TAFAS feature. ((\rightarrow 1.17.3) Trunk Answer From Any Station (TAFAS))</td>
</tr>
<tr>
<td></td>
<td>Outgoing Message (OGM)</td>
<td>Used to send messages for DISA feature. ((\rightarrow 1.17.6) Direct Inward System Access (DISA)) 5 + two-digit OGM number</td>
</tr>
<tr>
<td></td>
<td>Built-in Simplified Voice Message (SVM)</td>
<td>Used to access message boxes associated with extensions. ((\rightarrow 1.24.1) Built-in Simplified Voice Message (SVM))</td>
</tr>
<tr>
<td></td>
<td>Analogue Remote Maintenance</td>
<td>Used to access the PC programming mode through a modem on a PC.</td>
</tr>
<tr>
<td></td>
<td>ISDN Remote Maintenance</td>
<td>Used to access the PC programming mode through the ISDN interface on a PC.</td>
</tr>
<tr>
<td>Group</td>
<td>Incoming Call Distribution Group</td>
<td>Used to call an incoming call distribution group. ((\rightarrow 1.2.2.1) Incoming Call Distribution Group Features—SUMMARY) 6 + two-digit group number</td>
</tr>
<tr>
<td></td>
<td>PS Ring Group</td>
<td>Used to call a PS ring group. ((\rightarrow 1.25.2) PS Ring Group)</td>
</tr>
<tr>
<td></td>
<td>VM (DPT) Group</td>
<td>Used to call a VM (DPT) group. 500 (VM [DPT] Group 1)</td>
</tr>
<tr>
<td></td>
<td>VM (DTMF) Group</td>
<td>Used to call a VM (DTMF) group.</td>
</tr>
</tbody>
</table>

Conditions
- It is possible to give names to floating extension numbers.

PC Programming Manual References
4.9 [2-6-1] Numbering Plan—Main—Extension
5.13 [3-5-1] Incoming Call Distribution Group—Group Settings—Main
  \(\rightarrow\) Floating Extension Number
  \(\rightarrow\) Group Name
5.20 [3-7-2] VM(DPT) Group—Unit Settings—Floating Extension No.
5.25 [3-9] PS Ring Group—Floating Extension Number
7.2 [5-2] External Pager—Page Number 1, Page Number 2—Floating Extension Number
7.4 [5-3-2] Voice Message—DISA Message—Floating Extension Number
  \(\rightarrow\) Remote—Analogue Remote (Modem) Floating Extension Number
2.3.7 Floating Extension

Remote—ISDN Remote Floating Extension Number

PT Programming Manual References
[623] Incoming Call Distribution Group Name
[660] VM 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
1.24.2.1 Voice Mail (VM) Group
2.3.1 PC Programming
2.3.8 Software Upgrading

Description

It is possible to upload software from a PC to the PBX to upgrade the following:

<table>
<thead>
<tr>
<th>Data File</th>
<th>Storage Area</th>
</tr>
</thead>
<tbody>
<tr>
<td>Main Processing (IPCMPR) software data</td>
<td>Operating system data area on IPCMPR card.</td>
</tr>
<tr>
<td>Default data of system programming for each country/area</td>
<td>Country/area data area on the IPCMPR card.</td>
</tr>
<tr>
<td>Language data for PT display</td>
<td>Each language data area on the IPCMPR card.</td>
</tr>
<tr>
<td></td>
<td>Type 1: PT display except for PT system programming (→ 2.3.2 PT Programming) (5 languages max.)</td>
</tr>
<tr>
<td></td>
<td>Type 2: PT system programming data (one language max.)</td>
</tr>
<tr>
<td>LPR (software on a slot card) software data</td>
<td>Flash ROM on the slot card</td>
</tr>
<tr>
<td>Cell Station (CS) software data</td>
<td>Flash ROM on the CS</td>
</tr>
</tbody>
</table>

Conditions

- The software version of the IPCMPR can be confirmed through system programming.

Installation Manual References

3.3.1 IPCMPR Card

PC Programming Manual References

3.2 [1-1] Slot—Summary—◆ Card Type

PT Programming Manual References

[190] Main Processing (MPR) Software Version Reference
2.4 Fault Recovery/Diagnostics

2.4.1 Power Failure Transfer

Description
When the power supply to the PBX fails, specific SLTs and/or ISDN terminal devices 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 analogue trunk cards and extension cards, as well as the number of PFT ports (ports which can be used for Power Failure Connections) are as follows:
  Trunk Card: LCOT4 (1 PFT port)
  Extension Card: DHLC4 (1 PFT port)

Note
Through system programming, 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. However, if no special programming is performed, the connection will be dropped when power returns.

Installation Manual References
3.15.1 Power Failure Connections

PC Programming Manual References
13.3 [11-3] Power Failure Transfer

PT Programming Manual References
None
2.4.2 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) (→ 1.8.1 Automatic Callback Busy (Camp-on)) and Call Park (→ 1.13.2 Call Park).
2.4.3 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]

![Error Example Image]

[Explanation]

<table>
<thead>
<tr>
<th>Number in the Example</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>(1)</td>
<td>Error Code</td>
<td>Shows three-digit error code.</td>
</tr>
<tr>
<td>(2)</td>
<td>Sub Code</td>
<td>Shows six-digit sub code (X1YYZZ).</td>
</tr>
<tr>
<td></td>
<td></td>
<td>X: Slot type (Physical slot: blank, Virtual slot: *)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>1: Cabinet number</td>
</tr>
<tr>
<td></td>
<td></td>
<td>YY: Slot number (00–08)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>ZZ: Physical port number (01–16)</td>
</tr>
</tbody>
</table>

Conditions

- **System Alarm Button**
  A flexible button can be customised as the System Alarm button.
- If an extension allowed to utilise this feature uses PC Console or PC Phone, the alarm information will be displayed on the PC, and can be sent to the preprogrammed destination via PC Console or PC Phone. PC Console and PC Phone use e-mail to send the information.
- The alarm information will be recorded on SMDR, if enabled through system programming.
- The PBX can be automatically diagnosed at a preprogrammed time everyday.

Installation Manual References

5.1.5 Troubleshooting by Error Log
PC Programming Manual References

2.6.7 Utility—Message File Transfer PBX to PC
6.5 [4-1-4] Wired Extension—Flexible Button—◆ Type
6.13 [4-2-3] Portable Station—Flexible Button—◆ Type
6.17 [4-3] DSS Console—◆ Type
   →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

PT Programming Manual References

None

Feature Guide References

1.20.2 Flexible Buttons
1.26.1.1 Station Message Detail Recording (SMDR)
2.4.4 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:

<table>
<thead>
<tr>
<th>Type</th>
<th>TRAP Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Standard TRAP</td>
<td>coldStart</td>
<td>Information is sent after turning on the power of the PBX or resetting the PBX.</td>
</tr>
<tr>
<td></td>
<td>Authentication Failure</td>
<td>Information is sent when an unregistered Community Name and/or Manager IP address is entered.</td>
</tr>
<tr>
<td>Enterprise Specific TRAP*1</td>
<td>Major Alarm</td>
<td>Information is sent when a major alarm is detected.</td>
</tr>
<tr>
<td></td>
<td>Minor Alarm</td>
<td>Information is sent when a minor alarm is detected.</td>
</tr>
</tbody>
</table>

*1 Enterprise Specific TRAPs are original KX-NCP-exclusive information (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 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 5.1.5 Troubleshooting by Error Log in the Installation Manual.
- For a list of the MIB object groups supported by this PBX, refer to 3.3 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.

PC Programming Manual References

13.4 [11-4-1] SNMP—System Setting
13.5 [11-4-2] SNMP—Manager

PT Programming Manual References

None

Feature Guide References

2.4.3 Local Alarm Information
2.4.5 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 IPCMPR 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**

3.4 [1-1] Slot—Card Property - IPCMPR—LAN Setting

- DHCP Client
- IP Address for IPCMPR Card
- IP Address for VoIP-DSP

**PT Programming Manual References**

None
2.4.6 Packet Internet Groper (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 a 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

2.6.17 Utility—Ping

PT Programming Manual References

None
2.4.6 Packet Internet Groper (PING) Confirmation
Section 3
Appendix

Target Telcom Sales & Service Inc.
4400 Martin Rd.
Capac, MI 48014-2911

Important Phone Numbers
Sales: 248-752-3611
Service: 810-395-8555
Scheduling: 810-395-9111

Visit our Web Site:
www.targettelcom.com
## 3.1 Capacity of System Resources

### System

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Absent Message—Extension</td>
<td>1 x 16 characters</td>
</tr>
<tr>
<td>Absent Message—System</td>
<td>8 x 16 characters</td>
</tr>
<tr>
<td>Call Park Zone</td>
<td>100</td>
</tr>
<tr>
<td>Conference</td>
<td>3-8 parties per conference (32 parties total)</td>
</tr>
<tr>
<td>COS</td>
<td>64</td>
</tr>
<tr>
<td>DID/DDI Table</td>
<td>32 digits, 1000 entries</td>
</tr>
<tr>
<td>Extension number</td>
<td>1-5 digits (1-4 digits for PSs)</td>
</tr>
<tr>
<td>Extension Personal Identification Number (PIN)</td>
<td>10 digits, 1 entry/extension</td>
</tr>
<tr>
<td>Host PBX Access Code</td>
<td>10 digits, 10 entries/trunk group</td>
</tr>
<tr>
<td>Multiple Subscriber Number (MSN)</td>
<td>10 entries per ISDN-BRI port</td>
</tr>
<tr>
<td>Number of Characters of Name</td>
<td>20</td>
</tr>
<tr>
<td>Printing message</td>
<td>8</td>
</tr>
<tr>
<td>Queuing Time Table</td>
<td>64</td>
</tr>
<tr>
<td>Ring Tone Pattern Plan</td>
<td>8</td>
</tr>
<tr>
<td>Simultaneous Programmers</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• one system programmer + 63 personal programmers</td>
</tr>
<tr>
<td></td>
<td>• one manager programmer + 63 personal programmers</td>
</tr>
<tr>
<td></td>
<td>• 64 personal programmers</td>
</tr>
<tr>
<td>SMDR Call Storage</td>
<td>200 calls</td>
</tr>
<tr>
<td>Special Carrier Access Code</td>
<td>16 digits, 20 entries</td>
</tr>
<tr>
<td>Tenant</td>
<td>8</td>
</tr>
<tr>
<td>Time Service Holiday</td>
<td>24</td>
</tr>
<tr>
<td>Verification Code</td>
<td>4 digits, 1000 entries</td>
</tr>
<tr>
<td>Verification Code Personal Identification Number (PIN)</td>
<td>10 digits, 1000 entries</td>
</tr>
</tbody>
</table>

### Dialling

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Emergency Call</td>
<td>32 digits, 10 entries</td>
</tr>
<tr>
<td>Hot Line</td>
<td>32 digits</td>
</tr>
<tr>
<td>Key Pad Protocol Dial (ISDN Service Access)</td>
<td>32 digits</td>
</tr>
</tbody>
</table>
### Groups

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Broadcasting Group</td>
<td>8 (32 members/group)</td>
</tr>
<tr>
<td>Call Pickup Group</td>
<td>64</td>
</tr>
<tr>
<td>Idle Extension Hunting Group</td>
<td>64 (16 extensions/group)</td>
</tr>
<tr>
<td>Incoming Call Distribution Group</td>
<td>128 (32 extensions/group)</td>
</tr>
<tr>
<td>Paging Group</td>
<td>32</td>
</tr>
<tr>
<td>PS Ring Group</td>
<td>32</td>
</tr>
<tr>
<td>Trunk Group</td>
<td>64</td>
</tr>
<tr>
<td>VM (DPT) Group</td>
<td>2 groups x 12 ports (24 channels)</td>
</tr>
<tr>
<td>VM (DTMF) Group</td>
<td>2 groups x 32 channels</td>
</tr>
</tbody>
</table>

### TRS/Barring

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>TRS/Barring Level</td>
<td>7</td>
</tr>
<tr>
<td>TRS/Barring Denied Code</td>
<td>16 digits, 100 entries/level</td>
</tr>
<tr>
<td>TRS/Barring Exception Code</td>
<td>16 digits, 100 entries/level</td>
</tr>
</tbody>
</table>

### ARS

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Routing Plan Table</td>
<td>16 entries</td>
</tr>
<tr>
<td>Leading Number Table</td>
<td>16 digits, 1000 entries</td>
</tr>
<tr>
<td>Leading Number Exception Table</td>
<td>16 digits, 200 entries</td>
</tr>
<tr>
<td>ARS Carrier</td>
<td>10</td>
</tr>
<tr>
<td>Itemised Billing Code</td>
<td>10 digits</td>
</tr>
<tr>
<td>Authorisation Code for Tenant</td>
<td>16 digits</td>
</tr>
<tr>
<td>Authorisation Code for Trunk Group</td>
<td>10 digits</td>
</tr>
</tbody>
</table>
### Call Log and Message Waiting

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Outgoing Call Log—PT</td>
<td>100 records/extn.</td>
</tr>
<tr>
<td></td>
<td>520 records/system</td>
</tr>
<tr>
<td>Outgoing Call Log—PS</td>
<td>100 records/extn.</td>
</tr>
<tr>
<td></td>
<td>320 records/system</td>
</tr>
<tr>
<td>Incoming Call Log—PT</td>
<td>100 records/extn.</td>
</tr>
<tr>
<td></td>
<td>1040 records/system</td>
</tr>
<tr>
<td>Incoming Call Log—PS + Incoming Call Distribution Group</td>
<td>100 records/extn. or group</td>
</tr>
<tr>
<td></td>
<td>Total 1536 records/system</td>
</tr>
<tr>
<td>Message Waiting—PS + Incoming Call Distribution Group</td>
<td>192</td>
</tr>
<tr>
<td>Message Waiting—PT + SLT</td>
<td>512</td>
</tr>
</tbody>
</table>

### Voice Message

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Outgoing Message (OGM)</td>
<td>64</td>
</tr>
<tr>
<td>OGM Total Recording Time</td>
<td>MSG4 Card: 8 minutes; IPCMPR/ESVM2/ESVM4 Card: 20 minutes (at High recording quality)</td>
</tr>
<tr>
<td>Built-in Simplified Voice Message (SVM)</td>
<td>IPCMPR Card: 125 messages; ESVM2/ESVM4 Card: 250 messages per card/block</td>
</tr>
<tr>
<td>SVM Total Recording Time</td>
<td>20, 60, or 120 minutes (depending on the recording quality)</td>
</tr>
</tbody>
</table>

### Hospitality and Charge Management Features

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Billing items for guest rooms</td>
<td>1000 records/PBX</td>
</tr>
<tr>
<td>Hotel Operator</td>
<td>4</td>
</tr>
<tr>
<td>Charge Rate</td>
<td>7 digits including a decimal</td>
</tr>
<tr>
<td>Charge Denomination</td>
<td>3 currency characters/symbols</td>
</tr>
</tbody>
</table>

### Networking

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>TIE Line Routing and Modification Table</td>
<td>32 entries</td>
</tr>
<tr>
<td>Leading Number</td>
<td>3 digits</td>
</tr>
<tr>
<td>PBX Code</td>
<td>7 digits</td>
</tr>
</tbody>
</table>
### NDSS: Monitored PBXs

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>NDSS: Monitored PBXs</td>
<td>8</td>
</tr>
<tr>
<td>NDSS: Registered Extensions for Monitor PBX</td>
<td>250</td>
</tr>
</tbody>
</table>

### CTI

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500</th>
<th>KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>PC Console</td>
<td>8</td>
<td></td>
</tr>
<tr>
<td>PC Phone</td>
<td>20</td>
<td>36</td>
</tr>
<tr>
<td>CA Basic</td>
<td></td>
<td>128 users</td>
</tr>
<tr>
<td>CA Pro</td>
<td></td>
<td>128 users</td>
</tr>
<tr>
<td>CA Supervisor</td>
<td></td>
<td>4 users</td>
</tr>
</tbody>
</table>

### Password

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>System Password for Installer</td>
<td>4 – 10 characters</td>
</tr>
<tr>
<td>System Password for Administrator—for PC Programming</td>
<td>4 – 10 characters</td>
</tr>
<tr>
<td>System Password for User—for PC Programming</td>
<td>4 – 10 characters</td>
</tr>
<tr>
<td>System Password for Administrator—for PT Programming</td>
<td>4 – 10 digits</td>
</tr>
<tr>
<td>System Password for User—for PT Programming</td>
<td>4 – 10 digits</td>
</tr>
<tr>
<td>Manager Password</td>
<td>4 – 10 digits</td>
</tr>
</tbody>
</table>

### Programmer Code

<table>
<thead>
<tr>
<th>Item</th>
<th>KX-NCP500/KX-NCP1000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Installer Level Programmer Code</td>
<td>4 – 16 characters</td>
</tr>
<tr>
<td>Administrator Level Programmer Code</td>
<td>4 – 16 characters</td>
</tr>
<tr>
<td>User Level Programmer Code</td>
<td>0 – 16 characters</td>
</tr>
</tbody>
</table>
3.2 Tones/Ring Tones

3.2.1 Tones/Ring Tones

Tone Patterns (Default)

Confirmation Tone 1

Confirmation Tone 2

Confirmation Tone 3

Confirmation Tone 4

Confirmation Tone 5

Dial Tone 1

Dial Tone 2

Dial Tone 3

Dial Tone 4

Busy Tone

Reorder Tone

Ringback Tone 1

Ringback Tone 2

DND Tone

Trunk Call Limit Warning Tone
**Tone Patterns (Default)**

- Hold Alarm Tone: 15 s
- Call Waiting Tone 1: 1 s
- Call Waiting Tone 2: 1 s
- Cyclic Tone

**Ring Tone Patterns (Default)**

- Single
- Double
- Triple
- S-Double

*The duration of a ring tone may vary by country/area.*
### 3.3 Supported Management Information Base (MIB) Table

#### System Group (1.3.6.1.2.1.1)

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>sysDescr</td>
<td>Information of Hardware type and Software version of the Device.</td>
</tr>
<tr>
<td>2</td>
<td>sysObjectID</td>
<td>Object identifier of this product.</td>
</tr>
<tr>
<td>3</td>
<td>sysUpTime</td>
<td>Elapsed time since the system was restarted.</td>
</tr>
<tr>
<td>4</td>
<td>sysContact</td>
<td>Device Administrator.</td>
</tr>
<tr>
<td>5</td>
<td>sysName</td>
<td>Name of Device.</td>
</tr>
<tr>
<td>6</td>
<td>sysLocation</td>
<td>Installation Location of Device.</td>
</tr>
<tr>
<td>7</td>
<td>sysService</td>
<td>Support Layer.</td>
</tr>
</tbody>
</table>

#### Interface Group (1.3.6.1.2.1.2)

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>ifNumber</td>
<td>The number of Network Devices.</td>
</tr>
<tr>
<td>2</td>
<td>ifTable (NA)</td>
<td>Management Table by each Network Device.</td>
</tr>
<tr>
<td>2.1</td>
<td>ifEntry (NA)</td>
<td>Components of ifTable.</td>
</tr>
<tr>
<td>2.1.1</td>
<td>ifIndex</td>
<td>Each interface identifier.</td>
</tr>
<tr>
<td>2.1.2</td>
<td>ifDescr</td>
<td>Explanation of Interface.</td>
</tr>
<tr>
<td>2.1.3</td>
<td>ifType</td>
<td>Type of Interface.</td>
</tr>
<tr>
<td>2.1.4</td>
<td>ifMtu</td>
<td>Maximum Datagram Length which can be sent/received.</td>
</tr>
<tr>
<td>2.1.5</td>
<td>ifSpeed</td>
<td>Maximum Transfer Speed.</td>
</tr>
<tr>
<td>2.1.6</td>
<td>ifPhysAddress</td>
<td>Physical Address (MAC Address).</td>
</tr>
<tr>
<td>2.1.7</td>
<td>ifAdminStatus</td>
<td>The desired state of the interface.</td>
</tr>
<tr>
<td>2.1.8</td>
<td>ifOperStatus</td>
<td>The current operational state of the interface.</td>
</tr>
<tr>
<td>2.1.9</td>
<td>ifLastChange</td>
<td>The value of sysUpTime at the time the interface entered its current operational state (up or down).</td>
</tr>
<tr>
<td>2.1.10</td>
<td>ifInOctets</td>
<td>The number of Octets received.</td>
</tr>
<tr>
<td>2.1.11</td>
<td>ifInUcastPkts</td>
<td>The number of Unicast Packets delivered to a higher-layer protocol.</td>
</tr>
<tr>
<td>2.1.12</td>
<td>ifInNUcastPkts</td>
<td>The number of Non Unicast Packets delivered to a higher-layer protocol.</td>
</tr>
<tr>
<td>2.1.14</td>
<td>ifInErrors</td>
<td>The number of inbound Packets that contained errors.</td>
</tr>
<tr>
<td>2.1.15</td>
<td>ifInUnknownProtos</td>
<td>The number of Packet received which are discarded because of an unknown/unsupported protocol.</td>
</tr>
<tr>
<td>2.1.16</td>
<td>ifOutOctets</td>
<td>The number of Octets transmitted.</td>
</tr>
</tbody>
</table>
### 3.3 Supported Management Information Base (MIB) Table

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>2.1.17</td>
<td>ifOutUcastPkts</td>
<td>The number of Unicast Packets which are received from upper protocol.</td>
</tr>
<tr>
<td>2.1.18</td>
<td>ifOutNUcastPkts</td>
<td>The number of Non Unicast Packets which are received from upper protocol.</td>
</tr>
<tr>
<td>2.1.21</td>
<td>ifOutQLen</td>
<td>The length of the output packet queue (in packets).</td>
</tr>
<tr>
<td>2.1.22</td>
<td>ifSpecific</td>
<td>Relevant MIB object identifier.</td>
</tr>
</tbody>
</table>

### IP Group (1.3.6.1.2.1.4)

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>ipForwarding</td>
<td>The value which indicates operation availability as a router (whether Datagram is transferred or not).</td>
</tr>
<tr>
<td>2</td>
<td>ipDefaultTTL</td>
<td>Default value for IP Packet TTL (Time to Live).</td>
</tr>
<tr>
<td>3</td>
<td>ipInReceives</td>
<td>The total number of Packets received (including packet received in error).</td>
</tr>
<tr>
<td>4</td>
<td>ipnHdrErrors</td>
<td>The number of Packets discarded due to errors in their header.</td>
</tr>
<tr>
<td>5</td>
<td>ipInAddrError</td>
<td>The number of Packets discarded because IP Address of the destination was invalid.</td>
</tr>
<tr>
<td>7</td>
<td>ipInUnknownProtos</td>
<td>The number of Packets discarded because the protocol was unknown/unsupported.</td>
</tr>
<tr>
<td>8</td>
<td>ipInDiscards</td>
<td>The number of incoming Packets discarded because of an insufficient reception buffer.</td>
</tr>
<tr>
<td>9</td>
<td>ipInDelivers</td>
<td>The total number of Packets received (including ICMP) normally.</td>
</tr>
<tr>
<td>10</td>
<td>ipOutRequests</td>
<td>The total number of IP Packets (ICMP) which are tried to be transmitted (relay Packet is not included).</td>
</tr>
<tr>
<td>13</td>
<td>ipReasmTimeout</td>
<td>The maximum number of seconds required in the buffer to rebuild a fragmented Packet.</td>
</tr>
<tr>
<td>14</td>
<td>ipReasmReqds</td>
<td>The number of Packets that required rebuilding from a fragmented state.</td>
</tr>
<tr>
<td>15</td>
<td>ipReasmOKs</td>
<td>The number of Packets that were rebuilt correctly from a fragmented state.</td>
</tr>
<tr>
<td>16</td>
<td>ipReasmFails</td>
<td>The number of Packets that could not be rebuilt correctly from a fragmented state.</td>
</tr>
<tr>
<td>17</td>
<td>ipFragOKs</td>
<td>The number of Packets that were fragmented correctly.</td>
</tr>
<tr>
<td>18</td>
<td>ipFragFails</td>
<td>The number of Packets that could not be fragmented correctly.</td>
</tr>
<tr>
<td>19</td>
<td>ipFragCreates</td>
<td>The number of IP datagrams created due to fragmentation.</td>
</tr>
<tr>
<td>20</td>
<td>ipAddrTable (NA)</td>
<td>Management Table of addressing information relevant to this entity’s IP addresses.</td>
</tr>
<tr>
<td>20.1</td>
<td>IpAddrEntry (NA)</td>
<td>Components of ipAddrTable.</td>
</tr>
<tr>
<td>20.1.1</td>
<td>IpAdEntAddr</td>
<td>IP Address.</td>
</tr>
</tbody>
</table>
### 3.3 Supported Management Information Base (MIB) Table

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>20.1.2</td>
<td>IpAdEntIfindex</td>
<td>Index value of the Interface which is assigned to IP address.</td>
</tr>
<tr>
<td>20.1.3</td>
<td>IpAdEntNetMask</td>
<td>The Subnet Mask associated with IP address.</td>
</tr>
<tr>
<td>20.1.4</td>
<td>ipAdEntBcastAddr</td>
<td>Broadcast Address Value associated with IP Address.</td>
</tr>
<tr>
<td>20.1.5</td>
<td>IpAdEntReasmMaxSize</td>
<td>The size of the largest IP Datagram which can be sent/received through IP Address.</td>
</tr>
</tbody>
</table>

**ICMP Group (1.3.6.1.2.1.5)**

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>cmpInMsgs</td>
<td>The total number of ICMP messages received (excluded, with error).</td>
</tr>
<tr>
<td>2</td>
<td>icmpInErrors</td>
<td>The total number of ICMP messages received which contained error.</td>
</tr>
<tr>
<td>8</td>
<td>icmpInEchos</td>
<td>The total number of ICMP echo request messages received.</td>
</tr>
<tr>
<td>9</td>
<td>icmpInEchoReps</td>
<td>The total number of ICMP echo answering messages received.</td>
</tr>
<tr>
<td>14</td>
<td>icmpOutmsgs</td>
<td>The number of ICMP messages which were sent.</td>
</tr>
<tr>
<td>15</td>
<td>icmpOutErrors</td>
<td>The number of ICMP messages which were not sent because of error.</td>
</tr>
<tr>
<td>21</td>
<td>icmpOutEchos</td>
<td>The number of ICMP Echo request messages sent.</td>
</tr>
<tr>
<td>22</td>
<td>icmpOutEchoReps</td>
<td>The number of ICMP Echo Reply messages sent.</td>
</tr>
</tbody>
</table>

**TCP Group (1.3.6.1.2.1.6)**

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>tcpRtoAlgorithm</td>
<td>The algorithm used to determine the timing of retransmitting when a response was unacknowledged.</td>
</tr>
<tr>
<td>2</td>
<td>tcpRtoMin</td>
<td>Minimum value permitted for retransmission timeout (in milliseconds).</td>
</tr>
<tr>
<td>3</td>
<td>tcpRtoMax</td>
<td>Maximum value permitted for retransmission timeout (in milliseconds).</td>
</tr>
<tr>
<td>4</td>
<td>tcpMaxConn</td>
<td>Maximum number of TCP connections which can be supported.</td>
</tr>
<tr>
<td>5</td>
<td>tcpActiveOpens</td>
<td>The total number of Active open TCP connections.</td>
</tr>
<tr>
<td>6</td>
<td>tcpPassiveOpens</td>
<td>The total number of Passive open TCP connections.</td>
</tr>
<tr>
<td>7</td>
<td>tcpAttemptFails</td>
<td>The total number of connections error.</td>
</tr>
<tr>
<td>8</td>
<td>tcpEstabResets</td>
<td>The total number of resets.</td>
</tr>
<tr>
<td>10</td>
<td>tcpInSegs</td>
<td>The total number of segments received.</td>
</tr>
<tr>
<td>11</td>
<td>tcpOutSegs</td>
<td>The total number of segments sent.</td>
</tr>
<tr>
<td>12</td>
<td>tcpRetransSegs</td>
<td>The total number of segments retransmitted.</td>
</tr>
<tr>
<td>14</td>
<td>tcpInErrs</td>
<td>The total number of segments received in error.</td>
</tr>
</tbody>
</table>
### 3.3 Supported Management Information Base (MIB) Table

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>15</td>
<td>tcpOutRsts</td>
<td>The total number of TCP segments sent containing the RST flag (reset connection).</td>
</tr>
</tbody>
</table>

#### UDP Group (1.3.6.1.2.7)

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>udpInDatagrams</td>
<td>The total number of UDP Datagrams received.</td>
</tr>
<tr>
<td>2</td>
<td>udpNoPorts</td>
<td>The total number of received UDP Datagrams for which there was no application at the destination port.</td>
</tr>
<tr>
<td>3</td>
<td>udpInError</td>
<td>The total number of received UDP Datagrams which contained error.</td>
</tr>
<tr>
<td>4</td>
<td>udpOutDatagrams</td>
<td>The total number of UDP Datagrams sent.</td>
</tr>
</tbody>
</table>

#### SNMP Group (1.3.6.1.2.1.11)

<table>
<thead>
<tr>
<th>Object ID</th>
<th>Item</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>snmpInPkts</td>
<td>The total number of SNMP messages received.</td>
</tr>
<tr>
<td>2</td>
<td>snmpOutPkts</td>
<td>The total number of SNMP messages sent.</td>
</tr>
<tr>
<td>3</td>
<td>snmpInBadVersions</td>
<td>The total number of received SNMP messages of which version is unsupported.</td>
</tr>
<tr>
<td>4</td>
<td>snmpInBadCommunityNames</td>
<td>The total number of SNMP messages with unknown Community Name.</td>
</tr>
<tr>
<td>6</td>
<td>snmpInASNParseErrs</td>
<td>The total number of SNMP messages with incorrect OID type.</td>
</tr>
<tr>
<td>13</td>
<td>snmpInTotalReqVars</td>
<td>The total number of Objects which have been retrieved value successfully.</td>
</tr>
<tr>
<td>15</td>
<td>snmpInGetRequests</td>
<td>The total number of Get-Request which have been accepted and processed (e.g. Data extract by using snmpget/snmpwalk command).</td>
</tr>
<tr>
<td>16</td>
<td>snmpInGetNexts</td>
<td>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).</td>
</tr>
<tr>
<td>20</td>
<td>snmpOutTooBigs</td>
<td>The total number of sent SNMP messages which returned an error of &quot;TooBig&quot;.</td>
</tr>
<tr>
<td>21</td>
<td>snmpOutNoSuchNames</td>
<td>The total number of sent SNMP messages which returned an error of &quot;NoSuchName&quot;.</td>
</tr>
<tr>
<td>24</td>
<td>snmpOutGenErrs</td>
<td>The total number of sent SNMP messages which returned an error of &quot;GenErr&quot;.</td>
</tr>
<tr>
<td>28</td>
<td>snmpOutGetResponses</td>
<td>The total number of GetResponse sent.</td>
</tr>
<tr>
<td>29</td>
<td>snmpOutTraps</td>
<td>The total number of TRAP sent.</td>
</tr>
<tr>
<td>30</td>
<td>snmpEnableAuthenTraps</td>
<td>Indicates whether the SNMP agent process is permitted to generate authentication failure traps.</td>
</tr>
</tbody>
</table>
3.3 Supported Management Information Base (MIB) Table
A
AA → Automated Attendant 196
Absent Message 219
Account Code Entry 93
Advice of Charge (AOC) 240
Alarm Information, Local 434
Alternate Calling—Ring/Voice 89
Alternate Receiving—Ring/Voice 89
Alternate Receiving/Calling Mode (Ring/Voice) Override 85
ANI → Automatic Number Identification 210
Answering Features 81
Answering, One-touch Direct 81
AOC → Advice of Charge 240
Appendix 441
ARS → Automatic Route Selection 147
Audible Tone Features 323
Authorisation Code 151
Automated Attendant 196
Automatic Call Hold 171
Automatic Call Park 174
Automatic Call Waiting 44
Automatic Callback Busy (Camp-on) 122
Automatic Extension Release 88
Automatic Fax Transfer 205
Automatic Log-out 65
Automatic Number Identification (ANI) 210
Automatic Privacy 182
Automatic Redial 110
Automatic Rerouting of VoIP Calls to Public Trunks 351
Automatic Rerouting to Secondary PBX 379
Automatic Route Selection (ARS) 147
Automatic Setup 421

B
Background Music (BGM) 192
Barring 131
BGM → Background Music 192
BGM—External 192
Billing Code, Itemised 151
Billing for Guest Room Calls 311
Block, Internal Call 39
Bluetooth Module 378
Boss & Secretary feature 74
Broadcasting 185
Broadcasting Group 185
Budget Management 139
Built-in Simplified Voice Message (SVM) 257
Busy on Busy 62
Busy Override Deny, Executive 123
Busy Override, Executive 123
Buttons, Fixed 221
Buttons, Flexible 224
Buttons, PS Feature 287

C
CA → Communication Assistant 391
Call Billing for Guest Room 311
Call Charge Management 305
Call Charge Services 304
Call Directory—Extension Dialling 89
Call Directory—Speed Dialling 112
Call Distribution Port Group 69
Call Forwarding (CF)—by ISDN (P-MP) 241
Call Forwarding (CF)—by ISDN (P-P) 243
Call Forwarding (CF)—by QSIG 360
Call Forwarding (FWD) 71, 72
Call Hold 171
Call Hold (HOLD)—by ISDN 245
Call Hold Retrieve Deny 171
Call Hold, Automatic 171
Call Log, Incoming 215
Call Log, Outgoing 110
Call Monitor 124
Call Park 174
Call Park Recall 174
Call Park Retrieve Deny 174
Call Pickup 83
Call Pickup Group 83, 398
Call Pickup, Directed 83
Call Splitting 176
Call Transfer 167
Call Transfer (CT)—by ISDN 246
Call Transfer (CT)—by QSIG 362
Call Transfer with Announcement 167
Call Transfer without Announcement 167
Call Transfer—Screened → Call Transfer with Announcement 167
Call Transfer—Unscreened → Call Transfer without Announcement 167
Call Waiting 44
Call Waiting Caller ID (Visual Caller ID) 44
Call Waiting Tone 128
Callback Busy, Automatic (Camp-on) 122
Caller ID 210
Calling Line Identification (CLI) Distribution 29
Calling Line Identification Presentation (CLIP) 210, 237
Calling Line Identification Presentation (CLIP)—by QSIG 357
Calling Line Identification Restriction (CLIR) 237
Calling Name Identification Presentation (CNIP)—by QSIG 357
Calling Party Control (CPC) Signal Detection 166
Capacity of System Resources 442
Carrier Access Code, Special 101
CCBS → Completion of Calls to Busy Subscriber 248
CCBS by QSIG → Completion of Calls to Busy Subscriber—by QSIG 364
CDPG → Call Distribution Port Group 69
Cellular Phone Features—SUMMARY 392
Cellular Phone XDP Parallel 56
Centralised Voice Mail 370
Centrex Service, ISDN 234
CF by ISDN (P-MP) → Call Forwarding—by ISDN (P-MP) 241
CF by ISDN (P-P) → Call Forwarding—by ISDN (P-P) 243
CF by QSIG → Call Forwarding—by QSIG 360
Charge Management, Call 305
Charge Services, Call 304
Circular Hunting 46
Class of Service (COS) 396
CLI → Calling Line Identification Distribution 29
CLIP → Calling Line Identification Presentation 210, 237
CLIP by QSIG → Calling Line Identification Presentation—by QSIG 357
CLIR → Calling Line Identification Restriction 237, 358
CNIP by QSIG → Calling Name Identification Presentation—by QSIG 357
CNIR → Calling Name Identification Restriction 358
Code Entry, Verified 145
COLP → Connected Line Identification Presentation 237
COLP by QSIG → Connected Line Identification Presentation—by QSIG 357
COLR → Connected Line Identification Restriction 237, 358
Communication Assistant (CA) 391
Completion of Calls to Busy Subscriber (CCBS) 249
Completion of Calls to Busy Subscriber (CCBS)—by QSIG 364
Computer Telephony Integration (CTI) 386
Conference 179, 180, 185
Confirmation Tone 325
Connected Line Identification Presentation (COLP) 237
Connected Line Identification Presentation (COLP)—by QSIG 357
Connected Line Identification Restriction (COLR) 237
Connected Name Identification Presentation (CONP)—by QSIG 357
CONR → Connected Name Identification Restriction 358
Consultation Hold 160, 169, 176
COS → Class of Service 396
CPC → Calling Party Control Signal Detection 166
CTI → Computer Telephony Integration 386

D
Data Line Security 157
DDI → Direct Dialling In 23
Delayed Ringing 53
Denied Code Tables 131
DHCP → Dynamic Host Configuration Protocol 420, 438
Diagnostics 432
Dial Tone 323
Dial Tone Transfer 141
Dial Type Selection 95
Dialling Digit Restriction during Conversation 161
Dialling, One-touch 108
DID → Direct Inward Dialling 23
Digital XDP 163
DIL → Direct In Line 21
Direct Dialling In (DDI) 23
Direct In Line (DIL) 21
Direct Inward Dialling (DID) 23
Direct Inward System Access (DISA) 196
Direct One-touch Answering 81
Direct Recording 260
Direct Trunk Access 104
Directed Call Pickup 83
Directory, Call—Extension Dialling 89
Directory, Call—Speed Dialling 112
Directory, PS 286
DISA → Direct Inward System Access 196
DISA AA Service → DISA Built-in Automated Attendant Number 196
DISA Automatic Walking COS 198

DISA Built-in Automated Attendant Number (DISA AA Service) 198
DISA Interconnect Routing—Busy 197
DISA Interconnect Routing—DND 197
DISA Interconnect Routing—No Answer 197
DISA Interconnect Routing—No Dial 196
DISA Security Mode 197
Display Information 230
Display Lock 112, 215, 262
Distribution, Group Call 52
DND → Do Not Disturb 71, 76
DND Override 76
DND, Paging 76, 184
Do Not Disturb (DND) 71, 76
Door Open 190
Doorphone Call 188
Dynamic Host Configuration Protocol (DHCP) Assignment 438

E
E1 Line Service 253
EFA → External Feature Access 160
Electronic Station Lockout → Extension Dial Lock 140
Emergency Call 92
Enhanced Walking Extension 319
Exception Code Tables 131
Exclusive Call Hold 171
Executive Busy Override 123
Executive Busy Override Deny 123
Extension Controlling Features 314
Extension Dial Lock 140
Extension Feature Clear 316
Extension Personal Identification Number (PIN) 314
Extension PIN Lock 314
Extension Port Configuration 394
Extension Release, Automatic 88
Extension User Group 398
Extension-to-Trunk Call Duration 161
External Feature Access (EFA) 160
External Relay Control 209
External Sensor 207
EXtra Device Port (XDP) 394

F
Fault Recovery/Diagnostics 432
Fax Transfer, Automatic 205
Feature Clear, Extension 316
Feature Highlights 5
First Party Call Control 386
Fixed Buttons 221
Flash/Recall/Terminate 158
Flexible Buttons 224
Flexible Numbering/Fixed Numbering 423
Floating Extension 429
Full One-touch Dialling 108
FWD → Call Forwarding 71, 72
FWD—All Calls 72
FWD—Busy 72
FWD—Busy/No Answer 72
FWD—Follow Me 72
FWD—No Answer 72
No Line Preference 82, 103
Numbering, Fixed/Flexible 423

Off-hook Call Announcement (OHCA) 129
Off-hook Monitor 154
OGM → Outgoing Message 193
OHCA (Off-hook Call Announcement) 129
OHCA, Whisper 130
One-touch Answering, Direct 81
One-touch Dialling 108
One-touch Dialling, Full 108
One-touch Transfer 169
Operator Call 410
Operator Features 410
Operator, Hotel 308
Outgoing Call Log 110
Outgoing Message (OGM) 193
Off-hook Monitor 154
OGM → Outgoing Message 193
Outgoing Call Log 110
Overflow Feature 62

Paging 183
Paging Deny 183
Paging DND 76, 184
Paging Group 183, 399
Paralleled Telephone 163
Password 418
Pause Insertion 98
PC Console 389
PC Phone 389
PC Programming 413
PDN → Primary Directory Number 116
Peer-to-Peer Connection 382
Personal Identification Number (PIN), Extension 314
Personal Identification Number (PIN), Verification Code 145
Personal Speed Dialling 112
Pickup, Group Call 83, 398
PIN → Extension Personal Identification Number 314
PIN → Verification Code Personal Identification Number 145
PIN Lock, Extension 314
PING → Packet Internet Groper 439
Port Configuration, Extension 394
Portable Station (PS) Connection 280
Portable Station (PS) Features 280
Power Failure Connections 432
Power Failure Restart 433
Power Failure Transfer 432
Predialling 87
Primary Directory Number (PDN) 116
Prime Line Preference 82, 103
Printing Message 303
Priority Hunting 52
Privacy Release 182
Proprietary Telephone (PT) Features 221
PS → Portable Station Features 280
PS Directory 286
PS Feature Buttons 287
PS Ring Group 282, 400
PS Roaming by Network ICD Group 376
PT → Proprietary Telephone Features 221
PT Programming 416
Pulse to Tone Conversion 95

QSIG Enhanced Features 365
QSIG Standard Features 355
Queue Redirection, Manual 59
Queuing Feature 58
Quick Dialling 114
Quick Setup 420

Recall 158
Record Log Features 293
Recording into the VPS, Two-way 277
Recording, Direct 260
Redial, Automatic 110
Redial, Interrupt 110
Redial, Last Number 110
Relay Control, External 209
Remote Control 67
Remote Extension Dial Lock 140
Reverse Circuit 96
Ring 53
Ring Tone Pattern Selection 42
Ring Tones 446
Ringing Line Preference 82
Room Status Control 308
Route Selection, Automatic (ARS) 147

S-CO Line Access 104
Screening, Live Call (LCS) 276
SDN → Secondary Directory Number 116
SDN Direct Dial 116
SDN Walking COS 116
Second Call Notification to Busy Extension 126
Secondary Directory Number (SDN) 116
Security Mode, DISA 197
Self Labelling (KX-NT366 only) 384
Sending Faxes Through the Network 352
Sensor, External 207
Service-out mode 319
Setup, Automatic 421
Setup, Quick 420
Simple Network Management Protocol (SNMP) System Monitor 436
Simple Network Time Protocol (SNTP) 421
SIP (Session Initiation Protocol) Extension 380
SMO → Station Message Detail Recording 293
SNMP → Simple Network Management Protocol 436
SNTP → Simple Network Time Protocol 421
Software Upgrading 431
Special Carrier Access Code 101
Speed Dialling—Personal/System 112
Splitting, Call 176
Index

Station Hunting → Idle Extension Hunting 46
Station Message Detail Recording (SMDR) 293
Station Programme Clear → Extension Feature Clear 316
Station Speed Dialling → Personal Speed Dialling 112
Subaddressing 233
Supervisory Feature 67
Supported Management Information Base (MIB) Table 448
SVM → Built-in Simplified Voice Message 257
SVM Log 260
Syslog Record Management 302
System Configuration and Administration Features 393
System Configuration—Hardware 394
System Configuration—Software 396
System Data Control 413
System Speed Dialling 112
System Speed Dialling, TRS/Barring Override by 131

T
T.38 Protocol 352
T1 Line Service 255
TAFAS (Trunk Answer From Any Station) 191
Tenant Service 402
Tenant-to-Tenant Call Block 403
Terminate 158
Terminated Hunting 46
Third Party Call Control 386
Three-party Conference (3PTY)—by ISDN 247
TIE Line Service 327
Time Service 406
Timed Reminder 321
Toll Restriction (TRS) 131
Toll Restriction (TRS)/Call Barring (Barring) 131
Tone, Audible 323
Tone, Call Waiting 128
Tone, Confirmation 325
Tone, Dial 323
Tones/Ring Tones 446
Transfer Recall 168
Transfer, Call 167
Transfer, One-touch 169
Transferring Features 167
TRAP Implementation 436
TRS/Barring → Toll Restriction/Call Barring Features 131
TRS/Barring Level 131
TRS/Barring Override by System Speed Dialling 131
Trunk Access 104
Trunk Answer From Any Station (TAFAS) 191
Trunk Busy Out 97
Trunk Call Features 91
Trunk Call Limitation 161
Trunk Group 398
Trunk Group Access 104
Trunk-to-Trunk Call Duration 161
Two-way Recording into the VPS 277

U
UCD → Uniform Call Distribution 52
Unattended Conference 180
Uniform Call Distribution (UCD) 52
Upgrading Software 431
User-to-user Signalling Type 1 (UUS-1) 233

V
Verification Code Entry 145
Verification Code Personal Identification Number (PIN) 145
VIP Call 61
Virtual Locations 319
Virtual Private Network (VPN) 353
Virtual PS 291
VM → Voice Mail Features 264
Voice Mail (VM) Group 264, 399
Voice Mail DPT (Digital) Integration 274
Voice Mail DTMF Integration 267
Voice Mail Features 264
Voice Mail, Centralised 370
Voice Message Features 257
Voice Message, Built-in Simplified (SVM) 257
Voice over Internet Protocol (VoIP) Network 349
VoIP → Voice over Internet Protocol Network 349
VPN → Virtual Private Network 353
VPS Data Control by the PBX 277

W
Walking COS 143
Walking COS Through DISA 198
Walking COS, DISA Automatic 198
Walking Extension 318
Walking Extension Features 318
Walking Extension, Enhanced 319
Whisper OHCA 130
Wireless XDP Parallel Mode 288
Wrap-up 64

X
XDP → EXtra Device Port 394
XDP Parallel Mode, Wireless 288
XDP, Digital 163
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