



**LG-ERICSSON**



# ipLDK-60

## Feature Description & Operation Manual



## *Revision History*

<b>Issue</b>	<b>Date</b>	<b>Contents of Changes</b>
1.0	2006.10	Preliminary Release for ipLDK-60
2.0	2007.12	Second Release for ipLDK-60
2.1		Reformatting through-out General update
2.2		Update for ipLDK-60 Version 2

## ***Table of Contents***

<b>1</b>	<b>Introduction.....</b>	<b>1</b>
1.1	Feature Organization .....	1
1.2	Feature Sections .....	1
<b>2</b>	<b>Incoming Call Services.....</b>	<b>2</b>
2.1	Call Forward.....	2
2.2	Call Forward, Preset .....	5
2.2.1	Station Preset Forward .....	5
2.2.2	CO Line Preset Forward .....	6
2.3	Caller Id Display, Analogue CO Line.....	7
2.4	CO Line Name .....	8
2.5	CO Line Ring Assignment.....	9
2.6	Caller Controlled Routing (CCR) with VMIB.....	11
2.7	Data Line Security.....	13
2.8	Direct Inward Dialing (DID).....	14
2.9	Direct Inward System Access (DISA).....	16
2.10	Distinctive CO Line Ring .....	18
2.11	E1-R2 & LCO Collect Call Blocking, Brazil.....	19
2.12	Intercom Signal Mode .....	20
2.13	Preferred Line Answer (PLA) .....	21
2.14	Station Call Pickup .....	22
2.14.1	Directed Call Pick Up .....	22
2.14.2	Group Call Pick Up.....	23
2.15	Universal Night Answer (UNA).....	25
2.16	Voice Over .....	26
<b>3</b>	<b>Outgoing Call Services.....</b>	<b>27</b>
3.1	Authorization Code.....	27
3.2	Automatic Call Release.....	29
3.3	Basic Outbound Line Access .....	30
3.4	Call Time Restriction .....	32
3.5	CO Line Queuing .....	33
3.6	CO Step Call .....	34
3.7	Class Of Service (COS) .....	35
3.8	Emergency Call Service.....	37
3.9	Flash .....	38
3.10	Hot Line & Warm Line .....	39
3.11	Least Call Routing (LCR) .....	40

3.12	Memory Dialing .....	41
3.12.1	Auto Call Number Redial (ACNR) .....	41
3.12.2	Dial by Name .....	43
3.12.3	Dialing Security .....	45
3.12.4	Last Number Redial .....	46
3.12.5	Save Number Redial .....	47
3.12.6	Station Speed Dial .....	48
3.12.7	System Speed Dial.....	50
3.12.7.1	System Speed Zone.....	52
3.13	Private Line .....	53
3.14	Station Serial Call.....	54
3.15	Walking COS.....	55
<b>4</b>	<b>Call Handling Services .....</b>	<b>56</b>
4.1	Absent Text Message .....	56
4.1.1	Custom Message .....	56
4.1.2	Pre-selected Message .....	58
4.2	Automatic Fax Transfer .....	59
4.3	Automatic Hold.....	60
4.4	Automatic Privacy .....	61
4.5	Barge In .....	62
4.6	Call Log.....	63
4.7	Call Park .....	64
4.8	Call Transfer.....	65
4.8.1	Call Transfer, CO line.....	65
4.8.2	Call Transfer, Internal.....	67
4.9	Camp-on .....	69
4.10	CO Line Call Hold .....	71
4.11	Emergency Intrusion .....	73
4.12	Forced Handsfree Mode .....	74
4.13	Forced Trunk Disconnect .....	75
4.14	Hold Preference .....	76
4.15	Intercom Call Hold.....	77
4.16	Music On Hold (MOH).....	78
4.17	On-Hook dialing .....	79
4.18	Station and Group Call Coverage .....	80
4.19	Station Message Wait/Call Back .....	82
4.20	Time & Date .....	84
<b>5</b>	<b>Digital Key Telephone Features .....</b>	<b>85</b>
5.1	Do-Not-Disturb (DND) .....	85
5.2	Call Screening.....	87
5.3	Flexible Buttons.....	89
5.4	Headset.....	92

5.5	Hot Desk .....	93
5.6	Multi-lingual Display .....	95
5.7	Mute .....	96
5.8	One-Time DND .....	97
5.9	Station Relocation .....	98
5.10	User Station Programming .....	99
<b>6</b>	<b>Station &amp; Hunt Groups .....</b>	<b>102</b>
6.1	Hunt Groups .....	102
6.1.1	ACD (Automatic Call Distribution) Group & Supervisor .....	105
6.1.2	ACD (Automatic Call Distribution) Statistic Reports .....	109
6.1.3	External Voice Mail .....	112
6.2	Intercom Tenancy Group .....	113
<b>7</b>	<b>Conference Services .....</b>	<b>114</b>
7.1	Conference Room .....	114
7.2	Multi Party Conference .....	117
7.3	Paging Conference .....	118
7.4	SLT 3-party Conference .....	119
7.5	SLT Broker's Call .....	120
7.6	Unsupervised Conference .....	121
<b>8</b>	<b>Paging Services .....</b>	<b>123</b>
8.1	Internal, External Zone and All Call Paging .....	123
8.2	Meet-Me-Page .....	125
8.3	Push-to-Talk (PTT) Page .....	126
8.4	SOS Page .....	127
8.5	VMIB Page Message .....	128
8.6	Automatic Park and Page .....	130
<b>9</b>	<b>Station Pairs .....</b>	<b>131</b>
9.1	Executive/Secretary .....	131
9.2	Linked-Pair Station .....	133
9.3	Mobile Phone Support .....	134
9.3.1	Mobile Phone Remote Menu .....	137
<b>10</b>	<b>Miscellaneous Services .....</b>	<b>138</b>
10.1	Alarm .....	138
10.2	BGM (Background Music) .....	139
10.3	Chime Bell .....	140
10.4	Door Phone .....	141
10.5	External Control Contacts .....	143
10.5.1	Loud Bell .....	143
10.5.2	Door Open .....	143
10.5.3	External Page Control .....	144

10.6	In-Room Indication .....	145
10.7	Station Port Blocking.....	146
10.8	Wake-up Alarm .....	147
<b>11</b>	<b>Voice Services .....</b>	<b>149</b>
11.1	Company Directory .....	149
11.2	Two-way Recording .....	151
11.3	VMIB Auto Attendant Announcement.....	152
11.4	VMIB System Greetings and Prompts.....	153
11.5	VMIB Voice Mailbox .....	156
11.5.1	Incoming VMIB Voice Mail .....	156
11.5.2	Announcement only Mailbox.....	158
11.5.3	Administrative Mailbox .....	159
11.5.4	Managing VMIB Voice Mail.....	161
11.5.5	New Message Notification.....	164
11.5.6	Message Reply .....	165
11.5.7	Forward A Message .....	166
11.5.8	Add a Comment Message.....	167
11.5.9	Send A Message.....	168
11.5.10	Managing Personal Options.....	169
11.5.11	Virtual Mailbox.....	170
11.5.12	Remote VMIB Mailbox Control.....	171
11.5.13	Recording User Greetings.....	172
<b>12</b>	<b>Call Statistics &amp; Traffic Analysis .....</b>	<b>173</b>
12.1	Account Code.....	173
12.2	Lost Call Print-out.....	174
12.3	SMDR (Station Message Detail Recording) .....	176
12.4	Traffic Analysis.....	178
12.4.1	Attendant Reports .....	180
12.4.2	Call Reports .....	182
12.4.3	CO Reports .....	183
12.4.4	Hardware Unit Reports.....	185
<b>13</b>	<b>Attendant Services .....</b>	<b>186</b>
13.1	Attendant Date/Time Display Control.....	186
13.2	Attendant Call & Queuing.....	187
13.3	Attendant Forward.....	188
13.4	Attendant Intrusion .....	190
13.5	Attendant Override .....	191
13.6	Attendant Positions .....	192
13.7	Attendant Recall.....	193
13.8	CO Call Forward Off-Net.....	194
13.9	Day/Night Service .....	196
13.10	DSS/DLS Consoles.....	198

13.11	Door Phone BGM Control .....	199
13.12	Outgoing Access Control .....	200
13.13	Station Feature Cancel .....	201
<b>14</b>	<b>ISDN Service .....</b>	<b>202</b>
14.1	CLI Message Wait.....	202
14.2	CLI Transit Service.....	204
14.3	CLIP/COLP (Calling/Connected Line Identification Presentation) .....	206
14.3.1	Incoming Call ID Service.....	206
14.3.2	Outgoing Call ID Service.....	207
14.4	CLIR/COLR (Calling/Connected Line Identification Restriction).....	209
14.5	Incoming Calling Line Id (ICLID) Routing of DID Calls .....	210
14.6	ISDN Supplementary Services.....	211
14.6.1	ISDN Call Deflection .....	211
14.6.2	ISDN Call Reroute, CFNR (Call Forward No-response) .....	212
14.7	Keypad Facility.....	214
14.8	Multiple Subscriber Number (MSN).....	216
14.9	Prefix Code Dialing Service (Ukraine).....	217
<b>15</b>	<b>VoIP Service .....</b>	<b>218</b>
15.1	H.323 .....	218
15.1.1	Configuring H.323 GateKeeper Mode.....	219
15.2	IP address Dialing .....	221
15.3	IP Phone Support.....	222
15.4	SIP (Session Initiation Protocol).....	224
15.4.1	Configuring SIP Trunks.....	224
15.4.2	Configuring SIP Registered Lines .....	225
<b>16</b>	<b>Private Networking .....</b>	<b>227</b>
16.1	Busy Lamp Field (BLF) Manager .....	228
16.2	CO Transit - In.....	229
16.3	CO Transit - Out.....	230
16.4	Centralized Attendant Service (CAS).....	231
16.5	Centralized VM (Voice Mail) Service.....	232
16.6	Common Network Services.....	233
16.6.1	Network Absent Text Message .....	233
16.6.2	Network Call Back (Call Completion) .....	233
16.6.3	Network Call Forward.....	234
16.6.4	Network Call Transfer .....	235
16.6.5	Network Camp-on (Call Offer).....	236
16.6.6	Network Conference .....	236
16.6.7	Network Identification Service.....	237
16.6.8	Network MWI, CLI Message Wait .....	237
16.6.9	Network Station to Station Call .....	238
16.7	Network Configuration.....	239

16.7.1	ISDN QSig.....	239
16.7.2	VOIP Networking.....	244
16.8	Network Firewall routing.....	250
<b>17</b>	<b>Admin &amp; Maintenance .....</b>	<b>251</b>
17.1	Digital Keypad Admin .....	251
17.2	Lock-key operation.....	253
17.3	PC Admin, On-Line .....	254
17.3.1	Configuration Text File .....	256
17.3.2	LDK Utilities, Database Upload/Download.....	257
17.3.3	LDK Utilities, Software Upgrade.....	258
<b>18</b>	<b>Short Message Service.....</b>	<b>259</b>
18.1	Public SMS.....	259



# 1 INTRODUCTION

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## 1.1 Feature Organization

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In the following sections, features have been divided in to major groupings. For example, features associated with ISDN CO lines are grouped under ISDN Services. Effort has been taken to divide features in to meaningful groupings so that individual features can be quickly located within the manual. Under each major grouping, features are divided alphabetically for ease of use.

## 1.2 Feature Sections

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Each section is a discussion of a feature available with the ipLDK-60 system. The sections are divided into 5 categories. Where a particular category is not relevant, it is not included. The five (5) categories are:

- Description explains the nature of the feature.
- Operation describes how to use the feature.
- Consideration explains requirements or constraints of the feature and its configuration.
- Reference is a list related topics in this manual that aid in understanding the feature.
- Admin Programming is a reference list of admin programming which may influence operation of this feature.

## **2 INCOMING CALL SERVICES**

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### **2.1 Call Forward**

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#### Description

A user can forward (send) their calls to another Station, Station Group or a user's VMIB mailbox. The user may forward all calls immediately, if the user does not answer the call, or if the station is busy. In addition, the user may select to activate 'Follow-me' forward, forwarding calls from a location other than the user's normal station.

In some cases, the user may be allowed to forward calls over a CO Line, 'Off-net'. With Off-net forward, the system places a call over an idle CO Line and, after dialing, connects the caller to the CO Line. Several choices of call type (CO or Station call) and forward condition (All, busy and no-answer) are available.

A Flexible button can be assigned to activate the Forward feature.

#### Operation

To activate call forward,

1. Lift the handset or press the On/Off button
2. Press the [DND/FOR] button or dial the Forward code '5 5 4'
3. Dial the Call Forward type code (1-4)
  - 1 – Unconditional, all calls forward immediately
  - 2 – Busy, calls forward if the station is busy
  - 3 – No answer, calls forward if not answered in the Call Forward No- Answer time
  - 4 – Busy/No-answer, calls forward if busy or not answered after Call Forward No-answer time
4. Dial the station or Group number, or '#' for the VMIB to receive calls
5. Return to idle.

To activate Off-Net Call Forward,

1. Lift the handset or press the On/Off button
2. Press the [DND/FOR] button or dial the Forward code '5 5 4'
3. Dial the Call Forward type code 5 - 6 and 8 - 9
  - 5 – Unconditional, all calls forward immediately to a Speed Dial number
  - 6 – No answer, calls forward if not answered in the Call Forward No-Answer time to a Speed Dial number
  - 8 – Unconditional, all calls forward immediately to a user dialed telephone number, which is temporarily stored as the last available Speed bin
  - 9 –No-answer, calls forward if not answered after Call Forward No-answer time to the number in a Speed Dial bin
4. Seize a CO Line, optional
5. Dial the Speed Dial bin or telephone number
6. Return to idle.

To activate Follow-me Call Forward from any station,

1. Lift the Handset or press the On/Off button
2. Press the [DND/FOR] button or dial '5 5 4'
3. Dial Call Forward Code 0
4. Dial forwarding station number
5. Dial forwarding station Authorization code
6. Return to idle.

To deactivate Call Forward,

1. Press the [DND/FWD] button
- Or
1. Lift the handset or press the On/Off button
  2. Dial 5 5 9, the DND/FWD Cancel code.

### Consideration

- To use Call Forward, a station must be permitted to forward calls.
- Authorization codes are required for Follow-me Call Forward; assure the Authorization code is known before activating.
- A station must be separately allowed Call Forward and Off-Net Call Forward.
- Incoming analog CO lines must provide Disconnect Supervision and the Open Loop Detect timer must be set for proper operation of Off-Net connections.
- Station calls may automatically forward when assigned Preset Call Forward or may forward to the user's VMIB voice mailbox after the VMIB Forward time, if Auto Forward to VMIB is enabled. For proper operation, the station must be allowed access to the VMIB.
- Off-Net forward allows the user to dial the telephone number to receive calls. When used, the number dialed is temporarily stored in the last available Station Speed Dial bin. The assigned Station Speed bin is deleted automatically when the user cancels Off-net Forward.
- A station in DND cannot be the Call Forward destination; attempting to forward to a station in DND returns error tone.
- A dial pulse SLT cannot forward to the VMIB.

### Reference

- A. [Preset Call Forward](#)
- B. [Unsupervised Conference](#)
- C. [Authorization Codes](#)
- D. [DND \(Do Not Disturb\)](#)

### Admin Programming

System Numbering Plan (PGM 106, Flexible button 11)

Defines the Call Forward activation code for SLTs

System Numbering Plan (PGM 106, Flexible button 16)

Defines the code to cancel DND/FWD and Absent Message

Call Forward (PGM 111, Flexible button 2)

Permits stations access to Call Forward

Call Forward No Answer Timer (PGM 181, Flexible button 1)

Sets the timer for No answer Forward

Allow Off-Net Call Forward (PGM 111, Flexible button 18)

Permits stations access to Off-net Call Forward

Open Loop Detect Timer (PGM 142, Flexible button 13)

Sets the duration of an open loop for disconnect supervision on analog lines

Authorization Code Table (PGM 227),

Defines Authorization codes which might be required for Off-net Call Forward

Off-net Call Mode (PGM 112, Flexible button 12)

Defines call type that can Forward Off-net, external and/or internal

Unsupervised Conference Timer (PGM 182, Flexible button 6)

Sets the duration an Unsupervised conference is allowed

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

Auto Forward to VMIB (PGM 113, Flexible button 14)

Configures Stations to forward calls automatically to the user's VMIB mail-box

Auto Forward to VMIB timer (PGM 181, Flexible button 20)

Sets the time calls will forward to the user's VMIB voice mailbox

## **2.2 Call Forward, Preset**

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### 2.2.1 Station Preset Forward

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#### Description

If a station, configured for Preset Forward, does not answer a call in the Preset Forward time, the call is sent to the Preset Call Forward destination.

#### Consideration

- Preset Call Forward can form a chain; one station can be Preset Forward to another station, which is Preset Forward to another and so on. However, Preset Forward cannot be configured to forward a call to a previous station in the chain, which would form a loop.
- The Preset Call Forward destination can be a station or a Hunt group.
- User activated Call Forward will override Preset Call Forward.
- Other automatic forward functions (Auto Forward to VMIB) may interact with Preset forward operation based on timer settings.

#### Reference

- A. [Call Forward](#)
- B. [VMIB Voice Mailbox](#)

#### Admin Programming

Preset Call Forward (PGM 121)

Defines the destination for Preset Forward for a range of stations

Preset Call Forward Timer (PGM 181, Flexible button 12)

Sets the time before calls will forward

## 2.2.2 CO Line Preset Forward

---

### Description

If a call on a CO Line, configured for Preset Forward, is not answered in the CO Preset Forward time, the call is sent to the Preset Call Forward destination.

### Consideration

- The CO Line Preset Call Forward destination can be a station or a Hunt group, Station or Virtual Voice Mailbox, or a System Greeting with or without CCR.
- Other automatic forward features such as Forward to VMIB may interact with Preset Forward based on timer settings.
- Calls to a hunt group are not subject to CO Line Preset Forward.
- CO Line Preset Forward will override Station Preset Forward.

### Reference

- A. [Call Forward](#)
- B. Station [Voice Mailbox](#)
- C. Virtual Voice Mailbox

### Admin Programming

Preset Forward Timer (PGM 141, Flexible button 13)

Defines the time before CO Preset Forward is activated

Preset Forward (PGM 141, Flexible button 14)

Selects the destination for CO Preset Forward

## **2.3 Caller Id Display, Analogue CO Line**

---

### Description

When an incoming call is received, the sending network normally provides the contact number for the caller. This Caller Id is displayed in the DKT LCD and in the LCD of compatible SLTs, upon receipt. The phone displays the telephone number, name or both based on the system configuration. The ipLDK-60 system supports the following analog Caller Id protocols:

- Bell core GR-30-CORE & SR-TSV-002476 & ETSI ETS 300 659, ETSI ETS 300 778
- Denmark, TDK-TS 900 216
- Sweden, Telecom specification 8211-A112, Standard SS 63 63 25
- India DTMF, TEC Standard No. S/ASF-01/02
- Brazil DTMF, 220-250-713 (STANDARD) Issue 01, November 1993

If CLI Name Display is enabled for a station, the system will compare the CLI for an incoming call with the Station's Speed Dial numbers, if a match is found, the name associated with the Station Speed Dial, if any, is displayed.

### Consideration

- ipLDK sends Caller Id to SLTs during the silent period of the ring cycle. In some cases, SLTs may require at least 4 seconds between ringing to detect the Caller Id. In such cases, it may be necessary to extend the 'Off' or silent period of the SLT Ring Phase timer.
- A maximum of 12 digits can be displayed for CLI.

### Reference

- A. CLI Message Wait
- B. Call Log
- C. ISDN CLI

### Admin Programming

CO CID ATTR (PGM 147)

Defines the Caller Id type for CO lines

CID Name/Number (PGM 147, Flexible button 2)

Determines the displayed CID information (Name and/or Number)

SLT Ring Phase timer (PGM 182, Flexible button 4)

Defines the On/Off cycle for ring signals sent to SLTs

CLIP Display (PGM 114, Flexible button 1)

Enables display of CLI for incoming calls to a station

CLI Name Display (PGM 114, Flexible button 11)

Enables display of matching Speed Dial name for a station

## **2.4 CO Line Name**

---

### Description

When placing or receiving a call, the DKT display will indicate LINE xx, where xx is the CO line number. A name of up to 12 characters can be assigned to the CO line for display in place of the number.

### Consideration

- SMDR will display the CO line number not the programmed name (refer to Ref. A).
- CO Line Name Display must be enabled and CLIP Display disabled to display the CO Line Name.

### Reference

- A. Station Message Detail Recording (SMDR)

### Admin Programming

CO Line Name Display (PGM 142, Flexible button 1)

Enables the display for a range of CO lines

CO Line Name Assignment (PGM 142, Flexible button 2)

Assigns a name to a CO Line

CLIP Display (PGM 114, Flexible button 1)

Enables display of CLI for incoming calls to a station



## 2.5 CO Line Ring Assignment

---

### Description

A CO Line configured with the “Normal” service type can be assigned to notify (ring) specific destinations when an incoming call is received. The destination can be one or more stations, a Hunt Group, a VMIB Announcement, or a station in a networked system. Separate destinations can be defined for Day, Night, Weekend Lunch, and On-demand operation (Ref. C). In addition, the notification can be immediate upon detection of the incoming call or delayed in three (3) second increments from 3 to 27 seconds.

### Consideration

- Every CO line is assigned to ring at Station 100 by default (Ref. D).  
A busy DKT may receive muted ‘Off-hook’ ring to minimize interference with the active call while still providing notification (ref. Off-hook Ring).
- To receive incoming CO line calls, the DKTU should have a {CO} or {LOOP} button available.
- If no path is available to connect a call across a network of ipLDK systems, the receiving ipLDK routes the call to the Error destination assigned in the DID/DISA Destination table.
- Stations must have DID Call Wait enabled to receive signaling for a DID/DISA call while busy.
- A VMIB Announcement can be associated with a CCR Table to allow caller to dial digits in response to the greeting.

### Reference

- A. Hunt Group
- B. VMIB announcement
- C. Day/Night Service
- D. System Attendant
- E. Day/Night Service
- F. Private Networking
- G. Caller Controlled Routing (CCR) with VMIB

### Admin Programming

DID Call Wait (PGM 114, Flexible button 17)

Permits Call Wait signal for DID/DISA calls

CO Service Type (PGM 140, Flexible button 1)

Configures the service type for the CO Line, Normal or DID

CO Ring Assignment (PGM 144)

Assigns the ringing destination for a Normal CO Line

Hunt Group (PGM 190)

Configures members and type for a Hunt group

Hunt Group Attributes (PGM 191)

Defines the characteristics of each Hunt Group

Weekly Time Table (PGM 233)

Establishes the time of day for automatic Day, Night and Weekend system operation

DID/DISA Error Destination (PGM 167, Flexible button 2)

Assigns destination for calls that encounter a busy condition for Networked Station ring

## **2.6 Caller Controlled Routing (CCR) with VMIB**

---

### Description

When a call comes into the system through a DID or DISA line, the call may be routed to one of 70 VMIB System Greetings as an Auto Atd. The greeting is assigned to answer the call with a user-recorded menu corresponding to a CCR table. The CCR table configures a destination for each digit the caller might dial. For example, "For Sales dial 1" the CCR table would assign the digit '1' to route to the Sales group. Each digit (0-9) can be assigned to route the call to a:

- Station
- Hunt Group
- VMIB announcement
- Speed Dial
- Page
- Conference Room
- Network number
- Station or Virtual Mailbox
- Caller selected Mailbox
- Company Directory
- Remote System Greeting Recording

When routed to a VMIB greeting, the greeting may have an associated CCR table forming a multi-level interactive Auto Atd. No more than 10 levels are allowed in a multi-level CCR menu structure. In a multi-level structure dialing "\*" returns to the previous level and dialing "#" returns to the first level.

In addition, the system will monitor digits for multiple digits, allowing the user to dial a station or group number. When a station or group is dialed, the Auto Attendant will complete an unsupervised call transfer to the destination.

### Operation

When a call is answered by a VMIB announcement with CCR, during and for the Inter-digit time (default 5 seconds) after the announcement, the system will monitor the call for DTMF digits. If the user dials a digit, the system will wait the CCR Inter-digit time (default 3 seconds). If no additional digits are dialed, the system routes the call using the CCR table.

If the user dials additional digits, the call is routed to the station or hunt group dialed. If no digits are dialed the call is routed based on the CCR Error/Time-out or CO Line Ring Assignments.

### Consideration

- CO Lines must be assigned for DISA service.
- CO Lines should provide disconnect supervision for proper operation.
- For DID a VMIB System Greeting with a CCR table should be assigned as the Flexible DID Destination.

- If the CCR table has no assignments for an announcement, the call is routed based on the DID/DISA destination.
- If the CCR destination is busy, the caller may re-attempt entry based on the DISA Retry Counter. An additional entry attempt will reroute the call to the CCR Busy destination.
- If the caller dials an invalid digit or does not dial within the CCR Inter-digit time, the caller may re-attempt entry based on the DISA Retry Counter. An additional reentry will reroute the call to the CCR Error/Time-out destination.
- If the destination does not answer in the DID/DISA No Answer time, the call is routed to the CCR No Answer destination
- A System Speed Dial may be assigned as a CCR destination to route the call to an external destination over a CO line. This establishes an Unsupervised Conference (Ref. I).

### Reference

- A. Direct Inward Dial (DID)
- B. Direct Inward System Access (DISA)
- C. DISA Authorization Code
- D. System Speed Dial
- E. Hunt Group
- F. Conference Room
- G. Networking
- H. Paging
- I. [Unsupervised Conference](#)
- J. Voice Services
  - Company Directory
  - System Greetings & Prompts

### Admin Programming

DISA Retry Counter (PGM 160, Flexible button 4)

Defines the number of times a DISA caller may re-attempt an invalid entry

CCR Inter-digit Timer (PGM 180, Flexible button 15)

Defines the time the systems allows between digits during a CCR session

Inter-digit Timer (PGM 181, Flexible button 8)

Defines the time the system waits for a DISA caller to dial the first digit

DID/DISA No-Answer Timer (PGM 181, Flexible button 2)

Sets the no-answer time for DID and DISA calls.

Caller Controlled Routing Table (PGM 228)

Establishes the destinations for a CCR session

## **2.7 Data Line Security**

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### Description

In-band signals such as Camp-on (Refer to Ref. A) and Override (refer to Ref. B) tone, will interfere and cause errors in FAX and data transmission. To disable use of these tones for SLT ports used for FAX or data modems, Data Line Security can be enabled.

### Consideration

- If a device connected to an SLT port, which is assigned Data Line Security, makes an external call via an ISDN Line, the 3.1 KHz Setup message will be sent to the ISDN instead of the speech Setup message.
- If a busy station, assigned Data Line Security, receives a DID/DISA call, the call is disconnected regardless of the DID/DISA Destination setting. Also, normal off-hook ring and call wait tones are disabled.

### Reference

- A. Camp-On
- B. Attendant Override
- C. Call Transfer
- D. DID
- E. DISA

### Admin Programming

Data Line Security (PGM 111, Flexible button 4)

Assigns stations as secure data or FAX station

## **2.8 Direct Inward Dialing (DID)**

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### Description

A carrier service, known as Direct Inward Dial (DID), sends digits to the system so that the call may be routed directly to a specific station, Hunt group, VMIB announcement, Speed, or Page, bypassing the Attendant. Digits sent to the system are generally the last digits (3 or 4) dialed by the caller. DID service is available for digital and IP CO lines only and not analog CO lines.

For digital CO Lines, after collecting the digits from the carrier, ipLDK converts the digits employing one of three DID Conversion Types:

- Type 0 collect incoming digits and convert using the DID digit mask, resulting in the DID destination number.
- Type 1 use the incoming DID digits as the destination number without converting.
- Type 2 use DID conversion Type 0 as index to the Flexible DID Conversion Table.

### Consideration

- For ISDN DID CO calls, if the received Caller Id matches an ID in the ICLID Table the call routing is determined from the Flexible DID Conversion table index. If Caller Id does not match an entry in the ICLID Table, the normal DID call processes are used.
- DID calls that encounter a busy or DND signal, are not answered in the DID/DISA No Answer Timer, or are received at a vacant or invalid number can be routed to the Attendant, a tone, Station group, or VMIB announcement as defined by the DID/DISA Destination routes. When the Attendant receives such calls, the call is appropriately identified by the Attendant Phone display.
- A station which activates an Absence Text Message may automatically be placed in DND and DID calls routed based on the DID/DISA Destination DND route.
- For a station that is part of a non-pilot Hunt group, if the station is busy or does not answer the call, DID calls will follow the group hunt process.
- DID calls are subject to Group Call Pick-up and Directed Call Pick-up.
- If an announcement is defined as the destination in the Flexible DID Table, a Caller Controlled Routing Table for the announcement can be defined.
- DID calls to a busy station can be placed in a waiting state according to admin programming.

### Reference

- A. Day/Night Service
- B. System Speed Dialing
- C. Paging Features
- D. Caller Controlled Routing
- E. Station Call Pickup
- F. Absent Text Message
- G. VoIP Services

### Admin Programming

DID Restriction (PGM 114 – FLEX 16)

Restricts a station from receiving a DID call, DID calls to the station route to the Attendant

DID Call Wait (PGM 114, Flexible button 17)

Configures Call Wait for DID/DISA calls to a busy station, assigned to a station range

DID Conversion Type (PGM 143, Flexible button 4)

Configures the type of Conversion for a range of CO Lines

DID Digit Conversion Mask (PGM 146, Flexible button 6)

Configures the basic digit map to convert the last several DID digits

DID/DISA Destination (PGM 167)

Assigns destination of DID/DISA calls to a number that is invalid, busy, etc.

DID/DISA No-Answer Timer (PGM 181, Flexible button 2)

Sets the no-answer time for DID and DISA calls.

Flexible DID Table (PGM 231)

Destination table for Type 2 conversion

Weekly Time Table (PGM 233)

Establishes the time of day for automatic Day, Night and Weekend system operation

ICLID Table (PGM 237)

Assigns CLI and matching DID Conversion Table index for call routing

## **2.9 Direct Inward System Access (DISA)**

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### Description

DISA service permits an incoming caller to access system resources and features. For 'Normal' CO Lines that are enabled for DISA service, the system will answer the call and deliver internal dial tone or a VMIB announcement. During and for the Inter-digit time (default 5 seconds) after the announcement, the connection is monitored for digits from the caller. The caller can dial any number from the System Numbering Plan code available to an SLT including CO Line access codes, if allowed. DISA service is enabled separately for the Day/Night service modes (refer to Ref. F).

The caller may be required to enter a valid Station or System Authorization code, which will affect the dialing restrictions associated with the caller. If no codes are required or the user dials a System code, the CO Line COS and the CO-to-CO COS determine restrictions. For a Station Authorization code, the Station COS and CO COS determine any restrictions.

### Consideration

- If DISA Account Code is enabled, the caller receives a DND warning tone or a VMIB announcement to indicate an Authorization Code entry is required (refer to Ref. C).
- The caller may attempt entry of an Authorization code based on the DISA Retry Count.
- A Station can be configured not receive DISA calls. In this case, if the DISA caller dials the station, the call is routed to stations assigned to receive ring for the CO line or, if no stations are assigned ring, the call routes to the attendant.
- DID calls that encounter a busy or DND signal, are not answered in the DID/DISA No Answer Timer, or are received at a vacant or invalid number can be routed to the Attendant, a tone, Station Group, or VMIB announcement as defined by the DID/DISA Destination routes. When the Attendant receives such calls, the call is appropriately identified by the Attendant station LCD.
- A Station which activates an Absence Text Message may automatically be placed in DND and DID calls routed based on the DID/DISA Destination DND route.

### Reference

- A. Direct Inward Dialing (DID)
- B. VMIB System Greetings and Prompts
- C. Authorization Code
- D. Class of Service (COS)
- E. Day/Night Service

### Admin Programming

P-Msg DND (PGM 113, Flexible button 16)

Enables DND activation when an Absence Text message is activated at the Station

DISA Restriction (PGM 114, Flexible button 10)

Defines stations allowed to receive a DISA call

DISA Line Assignment (PGM 140, Flexible button 2)

Configures a 'Normal' CO Line for DISA service



DISA Account Code (PGM 141, Flexible button 3)

Configures DISA to require an Authorization code for access to system resources

DISA Retry Counter (PGM 160 – FLEX 4)

Configures the number of times the system will allow the user to enter a code

CO-to-CO COS Assignment (PGM 166)

Allows outgoing CO access by the DISA user

DID/DISA Destination (PGM 167)

Error destination table

DID/DISA No-Answer Timer (PGM 181, Flexible button 2)

Sets the no-answer time for DID and DISA calls.

Flexible DID Table (PGM 231)

Destination table for Type 2 DID Conversion

Weekly Time Table (PGM 233)

Establishes the time of day for automatic Day, Night and Weekend system operation

## **2.10 Distinctive CO Line Ring**

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### Description

The ring tone signal used to notify stations of an incoming call can be changed to provide distinctive ringing on a per CO line basis. Ring for each CO line can be assigned as one of four (4) signals. The signal consists of two continuous tones, which vary based on 'Country Code'; for North America, the tones available are shown in the chart below.

Ring tone 1	480 Hz and 0Hz
Ring tone 2	400 Hz and 0 Hz
Ring tone 3	620 Hz and 0 Hz
Ring tone 4	770 Hz and 0 Hz

### Reference

- A. CO Line Ring Assignment

### Admin Programming

CO Distinct Ring (PGM 142, Flexible button 5)

Assigns one of four ring tones to a range of CO Lines

Ring Frequency (PGM 422)

Assigns the frequency of the signals used for ring tones.

## **2.11 E1-R2 & LCO Collect Call Blocking, Brazil**

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### Description

E1 and LCO lines may support automatic Collect Call Blocking. When a call is received on an E1 CO line assigned for Collect Call Blocking, the system answers the call and generates the collect call-blocking signal. If the call is a collect call, the network will disconnect the call. If the call is not a collect call, the network does not disconnect the call and the ipLDK will process the call in the normal fashion.

There are two Collect Call blocking service types, w/Indicator, and Double Answer. For LCO lines, only 'Double Answer' is supported. For E1R2, 'Double Answer' is employed for R2 signaling and 'w/Indicator' is employed with R2MFC signaling.

### Consideration

- If 'Double Answer' is assigned, the ipLDK answers the call and maintains the connection for the Answer timer, generates an open loop signal for the Idle Timer and then reconnects. If the PSTN maintains the call, the ipLDK processes the call normally.

### Admin Programming

Collect Call Blocking (PGM 146, Flexible button 7)

Defines the type of Collect Call service for the CO line, w/Indicator or Double Answer

Collect Call Answer Timer (PGM 146, Flexible button 8)

Sets the delay before generating the Open loop on the CO Line

Collect Call Idle Timer (PGM 146, Flexible button 9)

Sets the duration of the open loop, signaling the E1 line to drop a collect call

## **2.12 Intercom Signal Mode**

---

### Description

Users can control the signaling mode used for incoming internal calls. Intercom calls will signal the user and are answered in one of three modes:

- HF (Hands Free) The station receives a tone burst followed by the voice of the calling party over the speaker and the user can reply hands free.
- TN (Tone) The station receives ringing for an incoming intercom call. The called party answers by lifting the handset or pressing the On/Off button.
- PV (Privacy) The station receives a tone burst and the voice of the calling party over the speaker. The microphone is muted for privacy. The called party must lift the handset or press the [MUTE] button to answer the call.

### Operation

To assign the Intercom Signal mode,

1. Press the [TRANS/PGM] button.
2. Dial 1 2.
3. Dial the Signal Mode (1 = HF, 2 =TN, 3 = PV).
4. Press the [HOLD/SAVE] button and confirmation tone is heard.

### Consideration

- Call backs, Call Transfers, and Attendant Override always signal the user with ring and must be answered by lifting the handset or pressing the On/Off button regardless of the assigned Intercom Signal mode (refer to Ref. A, B, and C).
- An intercom call in the Hands Free or Privacy mode is considered as an answered call. Thus, features such as Call Back, Call Forward No answer, etc. will not operate for intercom calls if in the HF or PV signaling mode.
- Intercom signal mode will not affect the Page announcements, which are always received over the speaker of idle stations.
- A station permitted Forced Handsfree, can change the Intercom Signal mode of a call to the HF mode overriding the signal mode assigned at the called Station.
- A station assigned Auto Answer will connect to incoming call transfers in the assigned HF or PV mode.

### Reference

- A. CO Line Queuing
- B. Message Wait
- C. Call Transfer
- D. Attendant Override
- E. Paging
- F. Station User Programming
- G. Forced Handsfree

### Admin Programming

Forced Hands Free (PGM 111, Flexible button 19)

Enables the user to force an intercom call to change from Tone to Hands free mode

## **2.13 Preferred Line Answer (PLA)**

---

### Description

When Preferred Line Answer (PLA) is enabled, lifting the handset or pressing the Speakerphone button automatically answers the ringing call. If several types of incoming calls (transfer, recall, queue callback, CO, or Intercom call) are ringing, the PLA priority determines the call that is answered.

The default PLA priority is:

- Transferred Call
- Recalled Call
- Normal Incoming Call
- CO Line Queued Call
- Intercom Call

### Operation

1. Lift the handset or press the speakerphone button.
2. ipLDK-60 connects the ringing call.

### Consideration

- Automatic Speaker Select should be enabled for proper operation.
- For DKTs that do not have a speakerphone, pressing the On/Off button will answer the call; however, you must lift the handset to be heard.
- Intercom calls are always given the lowest priority.

### Admin Programming

Preferred Line Answer (PGM 112, Flexible button 7)

Configures the status of PLA for a station range, 'On' or 'Off'

Automatic Speaker Selection (PGM 111, Flexible button 1)

Configures the status of Auto Speaker Operation for a station range, 'On' or 'Off'

PLA Priority (PGM 173)

Defines the priority for incoming CO call types for the system

## 2.14 Station Call Pickup

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### Description

A station user can pick up, answer, a call ringing at another Station. There are two types of Station Call Pick-up:

- Directed Call Pick-up
- Group Call Pick-up

### 2.14.1 Directed Call Pick Up

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#### Description

A station user can pick up, answer, a call ringing another station by dialing the direct call pick up code (refer to Ref. A) and the ringing station number.

#### Operation

To answer a call ringing at another station,

1. Lift the handset or press the [On/Off] button.
2. Dial Direct Call Pick-up code 7 (refer to Ref. A) or press the programmed {Direct Call Pick-up} Flexible button and dial the Intercom number of the ringing Station.

To assign a {Direct Call Pick-Up} button:

1. Press the [TRANS/PGM] button.
2. Press the Flexible button to be assigned.
3. Dial 7.
4. Press the [HOLD/SAVE] button to accept changes.

#### Consideration

- A {CO Line}, {POOL} or {LOOP} button is required to pick up a CO Line call.
- When several calls are available for pick-up, PLA Priority (PGM 173) determines the call that is answered. When several calls of the same type are available, the oldest call is given priority.
- Queue callback and Private line calls cannot be picked up.
- Only ringing intercom calls are subject to Directed Call Pick-up. An intercom call to a station that is in the Handsfree or Privacy Intercom Signaling Mode cannot be accessed using Station Call Pick-up.
- Direct Call Pick-up is allowed within the same Intercom Tenancy Group. A station may be allowed to pick up a call to stations in other Tenancy Groups (PGM 120).
- ICM box calls are subject to Directed Call Pick-up.

#### Reference

- A. Intercom Signaling Mode
- B. Intercom Tenancy Group
- C. Flexible CO Appearance
- D. Private Lines

### Admin Programming

System Numbering Plan (PGM 107, Flexible button 6)

Assigns the code for Direct Call Pick-up

PLA Priority (PGM 173)

Defines the priority for incoming CO call types for the system

## 2.14.2 Group Call Pick Up

---

### Description

A station can pick up a call ringing at another station in the same pick-up group. Ringing intercom calls, incoming CO Lines, recalling CO Lines and transferred CO Lines can be answered using Grp Call Pick-up if the Stations belong to the same pick-up group (refer to Ref. A).

### Operation

To answer a call ringing at a Station in the same pick-up group,

1. Lift the handset or press the [MON] button.
2. Dial the Group Pick-up code '5 6 6' (Ref. B) or press the {Group Call Pick-up} button.

To assign a {Group Call Pick-up} button:

1. Press [TRANS/PGM] button.
2. Press the Flexible button to be assigned.
3. Type '5 6 6' (refer to Ref. A).
4. Press the [HOLD/SAVE] button.

### Consideration

- Only ringing intercom calls are subject to Group Call Pick-up. An intercom call to a station that is in the Handsfree or Privacy Intercom Signaling Mode cannot be accessed using Station Call Pick-up.
- A {CO Line}, {LOOP} or {POOL} button is required to pick-up a CO Line call.
- Queue callback and Private line calls cannot be picked up.
- A station can answer an intercom call placed to a Door box using Group Call Pick-up.
- When several calls are queued at a station or Hunt Group, the pick-up order depends on PLA Priority (PGM 173).
- When CO line calls of the same type are queued, the first queued CO line has priority.
- Group Call Pick-up is allowed within the Intercom Tenancy Group. A station may be permitted to pick up calls to stations in other Intercom Tenancy Group (refer to Ref. B).

### Reference

- A. Flexible Numbering Plan
- B. Intercom Tenancy Group
- C. Private Lines

Admin Programming

System Numbering Plan (PGM 107, Flexible button 2)

Assigns the code for Group Call Pick Up

Pick-up Attribute (PGM 190, Flexible button 2)

Enables pick-up for a Station Group

PLA Priority (PGM 173)

Defines the priority for incoming CO call types for the system



## **2.15 Universal Night Answer (UNA)**

---

### Description

When the system is in the Night service mode, incoming calls on CO Lines configured for UNA can be answered by any station in the system with the Universal Night Answer (UNA) code (refer to Ref. A). Calls on a CO line assigned Night Answer are also generally configured to ring over External Paging (refer to Ref. B) equipment as a Night Bell. This is achieved using External Night Ring, which will ring according to the Loud Bell Controls (refer to Ref. C).

### Operation

To pick-up a call in Night mode,

1. Dial 5 6 9, the UNA code.

### Consideration

- If no call is available when the UNA code is dialed, the station receives error tone.
- Once answered, the call can be held, transferred, etc. in the normal fashion.
- UNA is not restricted to specific station or Intercom Tenant Groups, any station can answer any UNA call no matter the group assignments.

### Reference

- A. Day/Night Service Mode
- B. External Page
- C. Loud Bell

### Admin Programming

System Numbering Plan (PGM 107, Flexible button 4)

Assigns the code for UNA

Universal Night Answer (PGM 141, Flexible button 8)

Enables UNA operation for a range of CO Lines

External Night Ring (PGM 160, Flexible button 7)

Enables Loud Bell operation for calls on UNA enabled CO Lines in Night service mode

External Control Contact (PGM 168, Flexible button 1)

Assigns contact function and Loud Bell Control Station

Weekly Time Table (PGM 233)

Establishes the time of day for automatic Day, Night and Weekend system operation

## 2.16 Voice Over

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### Description

With Voice Over, a busy user can receive announcements from other users without interrupting the existing conversation. The announcement is received over the existing conversation so that only the busy station hears both incoming parties. The busy station user can alternately talk back to both parties.

### Operation

To activate a Voice-Over call,

1. While receiving busy on an intercom call, press ‘\*’ to camp-on (refer to Ref. A)
2. The busy station hears camp on tone and is connected to both parties. The busy station sends and receives voice to/from both simultaneously; the new and current caller will not send or receive voice to/from each other.
3. Press the flashing [HOLD/SAVE] button of the busy station; the active call is placed on hold and the busy station will be connected with the Voice-Over caller.
4. To alternate between calls, press [HOLD/SAVE] button.

### Consideration

- Placing a Station in DND or One-time DND disables the voice over function except that the Attendant can override DND at other Stations.
- Both calls will be dropped if the busy station hangs-up with an active Voice Over call, both the Voice Over and active call are terminated.
- If either party connected to the busy station hangs up, the remaining call is connected to the station as in a normal talking state.
- If the busy station responds to the Voice Over using the Hold button, the active call will be placed on hold and receive MOH, if provided (refer to Ref. B). The station and Voice Over caller are connected in a normal conversation.
- The busy station user can alternate between the calls using the Hold/Save button.
- The Voice Over announcement is received at the busy station over the active device, speaker or handset.
- Voice over must be enabled for both the called and calling (caller) stations. The calling station must be allowed “Caller Voice Over” and the called (busy station) must be allowed “Voice Over”.

### Reference

- A. Camp-On
- B. MOH

### Admin Programming

Caller Voice Over (PGM 111, Flexible button 22)

Enables stations to place a Voice Over call

Voice Over (PGM 113, Flexible button 6)

Enables a busy station to receive Voce Over calls

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## 3 OUTGOING CALL SERVICES

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### 3.1 Authorization Code

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#### Description

Authorization Codes (passwords) provide a means to control access to Walking COS, or DISA and may be required for outgoing CO line, LCR, and Voice Mail In-box access. An Authorization Code can be either a fixed length of five (5) digits or a variable length of 3 to 11 digits.

There are two types of Authorization Codes, Station and System. A Station Authorization Code is specifically related to a given station and intended for a single user. The System Authorization Codes are intended for use by any station in the system. Authorization Codes are assigned a Day and Night COS. When users dial an Authorization Code, the system invokes the COS assigned to Authorization Code or, if no COS is assigned, the Station COS is invoked.

The Attendant is permitted to assign any Authorization Code including codes for another station. Normal users may only assign the Station Authorization Codes (password) for the specific station.

#### Operation

To access a CO line when an Authorization Code is required,

1. When attempting CO line access, a DND warning tone or announcement is heard.
2. Dial the Authorization Code, and '#'. If 5-digit codes are assigned, '#' is not needed.
3. If valid, the CO line connects; if not valid, error tone is presented.

To register an Authorization Code when none has been assigned,

1. Press the [TRANS/PGM] button.
2. Dial 3 1.
3. Dial the Authorization Code, and '#'. If 5-digit codes are assigned, '#' is not needed.
4. Press the [HOLD/SAVE] button to accept changes.

To change an existing Authorization Code,

1. Press the [TRANS/PGM] button.
2. Dial 3 2.
3. Dial the existing Authorization Code, and '#'. If 5-digit codes are assigned, '#' is not needed.
4. Dial the new Authorization Code.
5. Press [HOLD/SAVE] to accept changes.

Consideration

- Up to 200 Authorization Codes can be entered, the first 48 codes are for stations the remaining are system codes. Codes 001 to 048 are employed for stations 100 to 147 respectively.
- Duplicate Authorization Codes are not allowed.
- If Account Codes are required for Loop LCR ACNT for the station, the user must enter a valid Authorization code to access Loop LCR (refer to Ref. A).
- If a DISA user enters an invalid code, error tone is heard then the user may attempt re-entry of the code. The user may attempt reentry based on the DISA Retry counter.

Reference

- A. LCR: 2.2.7
- B. DISA

Admin Programming

Authorization Code Table (PGM 227)

Assigns Station and System Authorization Codes

DISA Account Code (PGM 141, Flexible button 3)

Forces DISA callers to enter an Authorization Code for access to the System

DISA Retry Count (PGM 160, Flexible button 4)

Assigns the DISA re-try counter

CO Line Group Account (PGM 141, Flexible button 9)

Forces Authorization Code entry to access a CO line

Loop LCR Account Code (PGM 111, Flexible button 16)

Forces Authorization Codes for LCR access

5 Digit Authorization Code Usage (PGM 161, Flexible button 21)

Defines Authorization Code length, fixed 5 digit or variable 3 to 11 digits

Station Account (PGM112, Flexible button 20)

Forces the user to enter an Authorization Code to place a CO line call

## 3.2 Automatic Call Release

---

### Description

When a Station does not initiate dialing on an outgoing CO line or receives a no-answer on an intercom call, the system will disconnect the call based on the assigned Auto Release Timer. If the station is in speakerphone or monitor mode, the station will return to idle otherwise the station will receive error tone over the handset.

### Consideration

- Hands free Answer Intercom calls (refer to Ref. A) are considered answered and Auto Call Release is not applied.
- When the Automatic Release time is assigned as 0, Auto Call Release is disabled.

### Reference

- A. Intercom Signal Mode

### Admin Programming

Automatic CO Release Timer (PGM 180, Flexible button 14)

    Sets the timer for automatic release of a CO line call

Station Auto Release Timer (PGM 182, Flexible button 5)

    Sets the timer for automatic release of an intercom call

### 3.3 Basic Outbound Line Access

---

#### Description

Each station is allowed or denied access to particular CO lines or lines in a CO Group. Stations may further be limited access by Intercom Tenancy Group (refer to Ref. A) configurations. Flexible buttons assigned as a CO line or CO Group button, including the Pool and Loop buttons, provide the DKT user with access to Lines. These buttons function as below.

CO Line – access the specified CO Line

Pool –access outgoing CO Line from the specified group

Loop – access 1<sup>st</sup> available CO Line

The System Numbering Plan incorporates codes to access any available CO line, a specific CO line, or a line from a specific CO Group. As a default,

Dial '9' for any available CO Line

Dial '88' and the CO line number (01 - 36) for a specific CO line

Dial '8' and the CO Group number (01 – 24) for a CO line from the group

#### Operation

To access a CO line from a DKT,

1. Lift the handset or press the [MON] button.
2. Press the desired CO line, Pool button, or Loop button or dial the individual CO line, CO Group or the first available CO line code.

To access a CO line from a SLT,

1. Lift the handset.
2. Dial the individual CO line, CO Group or the 1st available CO line code.

To assign the [LOOP] button, perform the following Steps:

1. Press the [TRANS/PGM] button
2. Press the desired Flexible button
3. Press [TRANS/PGM]
4. Dial 8 4
5. Press the [HOLD/SAVE] button to accept changes.

#### Consideration

- An SLT user must use the dial codes to place CO line calls.
- A DKT must have an available appearance for incoming or outgoing CO line calls and may use a button or dial codes to place CO line calls.
- When the user dials '9' for any CO line and 'Override 1st CO Line Group' is enabled, the system will search the first CO line Group for an available line. If no Line is available in the group, the next group is searched until an available CO line is found or all groups, except group 00, have been searched. If no CO line is available, busy tone is provided and the user may queue for the next available Line.

- Error tone is provided should the user attempt to place a call on a CO line that the Station is not permitted outgoing access. The user may receive transferred and incoming calls on the CO line even though outgoing access is not allowed.
- When selecting a CO line from a CO Group, the ipLDK employs the Round robin or Last Choice technique to select the line as determined by PGM 160, Flexible button 3.
- For a BRI, the system can change the selected bearer channel if the Station has an appearance for each channel, or a [LOOP] button.
- CO Group 25 is used to define a CO line as not equipped, unused. To prevent access all unused CO lines must be assigned to CO Group 25.
- CO line Group 00 is the Private line group. Lines in this group can only be accessed with a button for the CO line.
- If CO PGM is allowed in PGM 112, Flexible button 6, a user can program a CO line or CO Group (Pool or Loop function) to a Flexible button.

### Reference

Intercom Tenancy Group

### Admin Programming

System Numbering Plan (PGM 107, Flexible button 7)

Assigns code for access to a CO Group

System Numbering Plan (PGM 107, Flexible button 8)

Assigns code for access to a specific CO line

System Numbering Plan (PGM 107, Flexible button 12)

Assigns code for first available CO line

CO PGM (PGM 112, Flexible button 6)

Permits the user to assign CO Line and Group buttons

CO Line Choice (PGM 160, Flexible button 3)

Defines the technique for CO line selection from a CO Group

CO Line Group Access (PGM 117)

Assigns CO Group access for a range of stations

CO Line Group (PGM 141, Flexible button 1)

Assigns CO Lines to a CO Group

Override 1st CO Line Group (PGM 161, Flexible button 3)

Allows search of all CO Groups for an available CO line

CO Line Attributes (PGM 141, Flexible button 10)

Assigns Tenancy groups allowed access to a range of CO lines

### **3.4 Call Time Restriction**

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#### Description

Stations can be configured to limit the duration of outgoing calls. Stations are enabled for Call Time Restriction and the Call Cut-off timer is defined to limit the length of the call. When the user places a call, the Call Cut-off timer starts. Fifteen (15) seconds before the cut-off time is reached all parties receive a warning tone and, at the cut-off time, the call is disconnected.

#### Consideration

- The cut-off timer continues timing the outgoing call even while the call is on hold or if transferred to another station.
- Call Time Restriction is applicable to all outgoing calls including conference calls.

#### Admin Programming

CO Warning Tone (PGM 112, Flexible button 1)

Assigns a warning tone fifteen seconds before the CO Call Time Restriction timer

CO Call Time Restriction (PGM 112, Flexible button 3)

Enables outgoing call duration restrictions for a range of stations

Call Cut-Off Timer (PGM 113, Flexible button 12)

Sets the allowed duration of outgoing calls for restricted stations



### **3.5 CO Line Queuing**

---

#### Description

If busy tone is received when attempting to access a CO line, the user may request to queue for the CO line and receive a call back when the line becomes available. When the CO line or line from the same CO Group becomes available, the System will reserve the line and call the station. The user must answer the call back within 15 seconds or the queue is cancelled and the next station in queue, if any, is offered the line.

#### Operation

To activate CO line queuing while receiving busy tone,

1. Press the [CALL BACK] button or dial the Call Back code '566'. For an SLT, hook-flash to receive dial tone prior to dialing the Call Back code.
2. When the confirmation tone is heard, replace the handset.

When a line is available

1. The station receives Call Back ring.
2. Lift the handset, to access the CO dial tone and place your call as normal.

#### Consideration

- An SLT user must dial the Call Back code.
- A CO line may have any number of queues at one time however, a station may have only one CO line queue active at a time. Activating a second queue request from a station cancels the previous request.
- If the waiting station is busy and the queued CO line becomes available, the available CO line will be directed to the next queued station, if any.
- Callbacks are not subject to Pick-up.

#### Reference

- A. Message Wait Enable
- B. Station Call Pick-up

#### Admin Programming

CO Line Queuing (PGM 112, Flexible button 5)

Enables CO line Queuing for a range of stations

### **3.6 CO Step Call**

---

#### Description

The DKT user may terminate and place a new call to the same number replacing the last digit of the number.

#### Operation

To use CO Step Call when receiving a busy tone,

1. Press the [Speed] button
2. Dial the last digit to use with the previously called number.
3. The active call is terminated and a new call is established.

#### Consideration

- CO Step Call is available only for analog CO lines.

### 3.7 Class Of Service (COS)

#### Description

Stations and CO lines may be assigned a Class of Service (COS) to restrict outgoing call service. Dialing restrictions are applied as the result of the interaction of COS assignments for the station and CO line as shown in the chart below.

Class of service

Station COS	CO Line COS				
	1	2	3	4	5
1	Unrestricted	Unrestricted	Unrestricted	Canned Table No LD code & limit 8 digits	Unrestricted
2	Table A	Table A	Unrestricted	Canned Table No LD code & limit 8 digits	Unrestricted
3	Table B	Unrestricted	Table B	Canned Table No LD code & limit 8 digits	Unrestricted
4	Table A,B	Table A	Table B	Canned Table No LD code & limit 8 digits	Unrestricted
5	Canned Table No LD code	Canned Table No LD code	Canned Table No LD code	Canned Table No LD code & limit 8 digits	Unrestricted
6	Canned Table No LD code & limit 8 digits	Canned Table No LD code & limit 8 digits	Canned Table No LD code & limit 8 digits	Canned Table No LD code & limit 8 digits	Unrestricted
7	Intercom Only	Intercom Only	Intercom Only	Intercom Only	Intercom Only
8	Table C	Table C	Unrestricted	Canned Table No LD code & limit 8 digits	Unrestricted
9	Table D	Table D	Unrestricted	Canned Table No LD code & limit 8 digits	Unrestricted
10	Table C & D	Table C & D	Unrestricted	Canned Table No LD code & limit 8 digits	Unrestricted
11	Table A/B/C/D	Table A/B/C/D	Unrestricted	Canned Table No LD code & limit 8 digits	Unrestricted

When the user places an outgoing CO line call, the System will compare the digits to the appropriate Toll Tables as shown in the chart above. Each Toll Table consists of an Allow and Deny Table pair; if the dialed number matches an Allow Table entry, the number is allowed. If the number matches a Deny Table entry, the call is terminated and error tone sent to the user.

#### Consideration

- If there are entries only in the Deny table, then the restriction is on a Deny Only basis.
- If there are entries only in the Allow table, then the restriction is on an Allow Only basis.
- If there are entries in both the Allow and Deny Tables, the Allow Table is searched, if the dialed number matches an entry in the Allow Table, the call is allowed. If a match is not

found, the Deny Table is searched and if a match is found in the Deny Table, the number is restricted. Otherwise, the number is allowed.

- If Incoming CO Call Toll Check is set, COS is applied if the station dials digits after answering a call on the CO line.
- When dialing on a CO line configured for behind PBX service, COS is not applied until the user dials a PBX trunk access code.

### Admin Programming

Station COS (PGM 116),

Assigns a COS to a range of Stations

CO line COS (PGM 141, Flexible button 2)

Assigns a COS to a range of CO Lines

CO Line Type (PGM 141, Flexible button 5)

Assigns the CO line for behind PBX operation

Incoming Toll Check (PGM 161, Flexible button 16)

Enables COS to be applied when users dial after answering a CO line call

CO-to-CO COS (PGM 166)

Enables a CO line to CO line connection (Unsupervised Conference)

PBX Access Code (PGM 172)

Assigns codes for PBX trunk access

Toll Exception Table (PGM 224)

Assigns Allow and Deny numbers in Toll Tables A to D

Canned Toll Tables (PGM 225)

Assigns Allow and Deny numbers to the Canned Toll Tables

### **3.8 Emergency Call Service**

---

#### Description

If a user dials an Emergency Service number, the call is processed regardless of the Station COS.

#### Consideration

- An emergency call can be dialed by pressing an available CO line at the station that is assigned to COS 7.
- If the Emergency number matches a station number or code in the LCR table, the call is routed as a station call or LCR call with preference to the LCR call. If the Station is assigned COS 7, a station call is processed.

#### Admin Programming

Emergency Service Call (PGM 226)

### **3.9 Flash**

---

#### Description

Flash is employed to request new dial tone from analog CO lines or from the ipLDK (Intercom dial tone). For analog CO lines the type of Flash, open loop or ground and the duration of the Flash can be assigned.

#### Operation

To activate a Flash from a DKT while on a call,

1. Press the [FLASH] softkey.

To activate a flash while on a CO line from a SLT;

1. Momentarily press the hook-switch
2. Dial 5 5 1.

#### Consideration

- The Flash command is not available on ISDN CO Lines.
- If a station is restricted from outgoing calls on a CO line, Flash is not permitted.

#### Admin Programming

System Numbering Plan (PGM 106, Flexible button 8)

Assigns the dial code for an SLT to Flash on a CO Line.

Flash Type (PGM 141, Flexible button 7)

Defines the type of Flash for CO lines, open loop or ground

CO Flash Timer (PGM 142, Flexible button 12)

Sets the duration of the Flash for CO lines

SLT Hook-switch Bounce Timer (PGM 182, Flexible button 1)

Sets the time the system will recognize as contact bounce for the SLT hook-switch

SLT Maximum Hook-switch Flash Timer (PGM 182, Flexible button 2)

Sets the maximum time an SLT can press the hook-switch for a Flash

SLT Minimum Hook-switch Flash Timer (PGM 182, Flexible button 3)

Sets the minimum time an SLT can press the hook-switch for a Flash

### **3.10 Hot Line & Warm Line**

---

#### Description

When a user goes 'off-hook', the system normally provides ICM dial tone. If desired, a station can be assigned to access a pre-assigned Idle Line Selection system resource (Hot Line). The Hot Line can be any of the Idle Line settings:

- Seize a CO Line,
- Call another station,
- Activate a Flexible button feature.

As a Hot Line, the Idle Line is accessed immediately. As a Warm Line, the user receives internal dial tone and after the Warm Line time, the Idle Line is accessed.

#### Operation

To activate a Hot Line/Warm Line,

1. Lift the handset or press the On/Off button.
2. The Hot Line is activated immediately. The Warm Line is accessed if there is no activity for the Warm Line time.

#### Consideration

- For proper operation, the Warm Line Timer should be less than the Dial Tone time-out Timer.
- An SLT may be assigned a Hot/Warm Line.
- Preferred Line Answer takes precedence over Hot/Warm Line.

#### Reference

- A. Preferred Line Answer (PLA)

#### Admin Programming

Warm Line (PGM 113, Flexible button 7)

Assigns Idle Line Selection as Hot or Warm

Idle Line Selection (PGM 122)

Assigns Idle Line Selection to a station range

Warm Line Timer (PGM 182, Flexible button 8)

Sets the Warm Line time

### **3.11 Least Call Routing (LCR)**

---

#### Description

LCR automatically selects the least expensive available route for an outgoing CO call. LCR compares the dialed digits to various tables, modifies the digits, and places the call based on configured treatments. LCR can be configured to apply different routing based on the time of day and day of week. In addition, LCR access can be automatic monitoring all outgoing calls or accessed for specific CO Lines or dial codes. LCR is also applied to Memory dialing and Off-net forward calls.

#### Consideration

- While direct CO LCR is applied to ISDN CO, an ISDN Information message with called party Information Element, which includes only the numbering plan and numbering type, is sent to the network when a user dials a digit. This process maintains the network connection.
- For direct CO LCR, leading digits should be programmed to minimize the chance of dial tone time-out by the carrier network.

#### Reference

- A. Basic Outgoing CO line Access
- B. Memory Dialing
- C. Call Forward Off-Net
- D. ISDN

#### Admin Programming

LCR Attributes (PGM 220)

Establishes LCR attributes

Leading Digit Table (PGM 221)

Defines digits which are monitored

Digit Modification Table (PGM 222)

Defines the modification process for LCR

LCR Table Initialization (PGM 223)

Initializes the LCR tables returning them to the default



## **3.12 Memory Dialing**

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### **3.12.1 Auto Call Number Redial (ACNR)**

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#### Description

If a CO line call destination is busy or no-answer, the number can be redialed automatically using ACNR. The system will retry the number based on the assigned ACNR Retry counter, default is three (3) attempts.

#### Operation

To use ACNR while receiving a busy/no-answer indication on a CO line call,

1. Press the [REDIAL] button or ACNR softkey.
2. Replace the handset or go on-hook.
3. The system will automatically retry the call at programmed intervals.
4. When the called party answers, lift handset.

To cancel ACNR, perform the following Steps:

1. Press the flashing [REDIAL] button or Stop softkey.  
OR
2. Lift the handset or press the [Speaker] button during an ACNR call attempt.

#### Consideration

- ACNR is only available to DKTs with a [REDIAL] button or ACNR softkey. Use code '97' to assign the Redial feature to a Flexible button
- Analog CO Lines must provide call progress tones so the system can detect the status of the CO line during the ACNR attempt.
- If the originally used CO line used to place the call is busy, an ACNR call is placed on a CO line from the same group. If no line is available, the system will restart the call attempt after the ACNR delay time without incrementing the ACNR Retry Counter.

#### Reference

- A. Flexible Button Assignment

#### Admin Programming

Override 1st CO Line Group (PGM 161, Flexible button 3)

Allows search of all CO Groups for an available CO line

ACNR Pause Timer (PGM 180, Flexible button 10)

Sets the time between ACNR call attempts and delay if all CO lines are busy

ACNR Delay Timer (PGM 180, Flexible button 8)

Sets the delay time to restart if the station is busy

ACNR Tone Detect Timer (PGM 180, Flexible button 13)

Sets analog CO line call progress tone detection time

ACNR No Answer Timer (PGM 180, Flexible button 9),

Duration the system will await answer before terminating an ACNR call attempt

ACNR Retry Counter (PGM 180, Flexible button 11)

Defines the number of times ACNR is attempted before it is automatically cancelled

**ACNR No-Tone Retry Counter (PGM 180, Flexible button 12)**

Defines the number of times access to a CO line is attempted when no dial tone is received.

### 3.12.2 Dial by Name

#### Description

Users of a DKT with an LCD may use Dial by Name to call other users in the system or to dial System or Station Speed Dial numbers. For proper operation, each station and Speed Dial should be registered with a Name. Each Station Name can be up to 12 characters where each character is represented by 2 digits as shown in the table below. The name for a Speed Dial number may be up to 16 characters in length.

The user selects a Directory (Intercom, System Speed Dial, or Station Speed Dial) then enters characters to search for a Name, or scroll through the list of Names in the Directory. Once located, the user selects the Hold/Save button to place the call. When entering characters, the DKT LCD will display the matching entries as each character is entered.

#### Operation

To register a DKT Station Name,

1. Press the [TRANS/PGM] button.
2. Dial '7 4'.
3. Dial the Name using 2 digits for each character; refer to the digit map chart below.
4. Press [HOLD/SAVE] to accept changes.

Digit Map Chart

Q - 11	A - 21	D - 31
Z - 12	B - 22	E - 32
. - 13	C - 23	F - 33
1 - 10	2 - 20	3 - 30
G - 41	J - 51	M - 61
H - 42	K - 52	N - 62
I - 43	L - 53	O - 63
4 - 40	5 - 50	6 - 60
P - 71	T - 81	W - 91
Q - 72	U - 82	X - 92
R - 73	V - 83	Y - 93
S - 74	8 - 80	Z - 94
7 - 70		9 - 90
*1 - Blank		
*2 - :	0-00	
*3 - ,		

To register an SLT Station Name,

1. Lift the handset.
2. Dial '5 6 3' (refer to Ref. C); confirmation tone is heard.
3. Dial '7 4', the Name Register code.
4. Dial the Name using 2 digits for each character; refer to the digit map chart below.
5. Momentarily press the hook-switch to save the Name.

To use Dial by Name,

1. Press the [SPEED] button twice.
2. Use the Volume Up/Down buttons to highlight the Directory.
  - 1 = Intercom
  - 2 = Station Speed Dial
  - 3 = System Speed Dial
3. Enter characters to search the Directory  
or, use the Up/Down buttons to scroll through the list of Names.  
Note when entering Names, use the [Call Back] button to delete the last character.
4. When the LCD cursor points to the desired Name, press the [HOLD/SAVE] button or  
press the [SEND] softkey.

#### Consideration

- Dial by Name is available only on a DKT with LCD.
- The name must be registered to use Dial by Name.
- A station or Speed Dial with no Name is not listed in the Dial by Name directories.

#### Reference

- A. System Speed Dial
- B. Station Speed Dial
- C. Station User Programming

#### Admin Programming

System Numbering Plan (PGM 106, Flexible button 20)

Assigns the code for an SLT to access Station Programming

### 3.12.3 Dialing Security

---

#### Description

Normally, when Speed Dial (refer to Ref. A) is used, the number stored in the Speed Dial bin is displayed in the DKT LCD. If desired, the user can disable the number display by preceding the number with an '\*' when entering the Speed Dial number.

#### Operation

To program a speed dial number with security dialing,

1. Press the [TRANS/PGM] button.
2. Press the [SPEED] button.
3. Dial the Speed Dial bin number.
4. Press the '\*' button.
5. Dial the desired number and complete programming the Speed Dial as normal.

#### Consideration

- Dialed phone numbers with Dialing Security are included in SMDR information (refer to Ref. C).
- When a call is placed with a Speed Dial assigned Display Security, the number is not displayed if the call is transferred or recalls. Also, if Redial (refer to Ref. D) is used to redial a previously used Speed Dial with Display Security, the number will not be displayed.

#### Reference

- A. Station Speed Dialing
- B. System Speed Dial
- C. SMDR
- D. Last Number Redial

### 3.12.4 Last Number Redial

---

#### Description

The last dialed number on a CO line can be stored (up to 32 digits) in the station's Last Number Redial buffer. The user may select to redial the last number dialed on the system. Each DKT with an LCD has 10 individual last dialed number directory locations.

#### Operation

To use Last Number Redial at a DKT,

1. Lift the handset or press the [On/Off] button.
  2. Press the [REDIAL] button
- OR
3. Press the [SPEED] button and dial \*.
  4. Scroll using the Volume Up/Down buttons to find the phone number
  5. Press the [HOLD/SAVE] button or [Ok] softkey to accept.

To use Last Number Redial at an SLT,

1. Lift the handset.
2. Dial 5 5 2

#### Consideration

- When the CO line used to place the original call is busy, an idle CO line in the group is accessed and the last dialed number is dialed.
- The last dialed number directory allows duplicate phone numbers.
- If Last Number Redial is activated while the ACNR is in use, the ACNR is canceled.

#### Admin Programming

System Numbering Plan (PGM 106, Flexible button 9)

Assigns the code for an SLT to access LNR

### 3.12.5 Save Number Redial

---

#### Description

The last dialed number on a CO/IP call may be stored (up to 32 digits) in a buffer for future redial. This number is saved in memory until the user requests a new number be stored. Numbers dialed for subsequent calls do not affect the Save Number buffer.

#### Operation

To save a number in the Save Number Redial buffer from a DKT,

1. While on an active outgoing call, press the [SPEED] button twice,
2. When the active call is complete, replace the handset or go on-hook.

To place a Save Number Redial call,

1. Lift the handset or press the On/Off button.
2. Press the [SPEED] button.
3. Press the # button.

#### Consideration

- When the CO line originally used is busy, an idle CO line in the group is accessed and the saved number is dialed.
- The stored Save Number is not deleted if the system should lose power.

### 3.12.6 Station Speed Dial

---

#### Description

Each Station user can store up to 100 frequently used numbers to Station Speed Bins (000 – 099). Each Station Speed Dial consists of up to 24 digits including special function digits.

#### Operation

To make a call using Station Speed Dial from a DKT,

1. Lift the handset or press the [On/Off] button.
2. Press the [SPEED] button.
3. Dial the Station Speed Dial bin (000 – 099).

To store Station Speed Dial numbers from a DKT,

1. Press the [TRANS/PGM] button.
2. Press the [SPEED] button.
3. Dial the Station Speed Dial bin (000 – 099).
4. If desired, press the CO line or Group button.
5. Dial the desired telephone number, up to 24 digits including special codes:

Special codes include:

- **[CALL BACK]** – Insert Pause key
- **★** – If stored as the first digit, it will function as Display Security. Otherwise, it will function as a Pulse to DTMF Switchover.
- **[DND/FOR]** – If CO Dial Tone Detect is enabled and stored as the first or second digit, and the accessed CO Line is behind the PBX mode, its function is Dial Tone Detect. Otherwise, its function is Pause.
- **[FLASH]** –  
If the CO Line is analog, a 'Flash' is inserted in the Speed Dial number.  
For an ISDN line (refer to Ref. A), when stored as the first digit, the remaining digits are sent in a keypad facility IE (Information Element) not in the normal Calling Party Number IE.

6. Press the [HOLD/SAVE] button.
7. If desired, enter the name (Max. 16 characters) using the 2-digit code for each character.
8. Press the [HOLD/SAVE] button.
9. To store another Speed Dial number, repeat this procedure from Step 3.

To delete a Station Speed Dial bin,

1. Press the [TRANS/PGM] button.
2. Press the [SPEED] button.
3. Dial the Speed Dial bin number to be erased.
4. Press the [HOLD/SAVE] button.
5. The stored Speed Dial number should be erased from the speed bin.

To display and enter a Speed Dial bin by scrolling,

1. Press the [TRANS/PGM] button.



2. Press the [SPEED] button.
3. Dial the Speed Dial bin number.
4. Press the [UP] or [DOWN] key to display the next/previous Speed Dial number.

To store Station Speed Dial numbers from a SLT, perform the following Steps:

1. Lift the handset.
2. Dial the Speed Dial program code 5 5 5.
3. Dial the Speed Dial bin number (000 – 099).
4. Dial the desired phone number (up to 24 digits).
5. Press and release the hook-switch.

To make a call using Station Speed Dial from a SLT,

1. Lift the handset.
2. Dial the Speed Dial access code 5 5 8.
3. Dial the Station Speed dial bin (000 – 099).

To delete a Station Speed Dial from a SLT,

1. Lift the handset.
2. Dial the Speed Dial access code 5 5 5.
3. Dial the appropriate Station Speed Dial bin (000 - 099).
4. Press and release the hook-switch.

### Consideration

- The Station Speed Dial is battery protected from power loss.
- Attempts to place a call using an empty Speed Dial bin will return error tone.
- Selecting a CO line before dialing a Speed Dial will use the selected CO line for the call.
- A name of up to 12 characters can be entered for a station using Speed Dial bin 000. In this case, after entering the bin (000), press the [Speed] button then enter the name. Speed Dial bin 000 may still be used to store a number but the name should not be assigned.

### Reference

- A. ISDN Line

### Admin Programming

System Numbering Plan (PGM 106, Flexible button 15)

Assigns the code for SLT Speed Dial access

System Numbering Plan (PGM 106, Flexible button 12)

Assigns the code for SLT to program Speed Dial numbers

Speed Dial Access (PGM 112, Flexible button 9)

Enables access to Speed Dial feature for a range of stations

CO Dial Tone Detect (PGM 160, Flexible button 6)

Enables dial tone detection for a CO Line

### 3.12.7 System Speed Dial

---

#### Description

Memory in the ipLDK-60 is allocated for 500 System Speed Dial bins with up to 24 digits each. System Speed Dial numbers are entered by the System Attendant. The first 200 numbers (2000 to 2199) are available for access by all stations in the system. The remaining 300 numbers are subject to COS restrictions (refer to Ref. A).

#### Operation

To store a number in a System Speed Dial from the System Attendant,

1. Press the [TRANS/PGM] button.
2. Press the [SPEED] button.
3. Dial the System Speed Dial bin (2000 to 2499).
4. If desired, press the CO line or Group button.
5. Dial the desired phone number, which may include special codes (up to 24 digits).

Special codes include:

- **[CALL BACK]** – Insert Pause key
- **\*** – If stored as the first digit, it will function as Display Security. Otherwise, it will function as a Pulse to DTMF Switchover.
- **[DND/FOR]** – If CO Dial Tone Detect (refer to Ref. A) is On and it is stored as the first or second digit, and the accessed CO Line is behind the PBX mode, its function is Dial Tone Detect. Otherwise, its function is Pause.
- **[FLASH]** –  
If the CO Line is analog, a 'Flash' is inserted in the Speed Dial number.  
For an ISDN line (refer to Ref. A), when stored as the first digit, the remaining digits are sent in a keypad facility IE (Information Element) not in the normal Calling Party Number IE.

6. Press the [HOLD/SAVE] button.
7. If desired, enter the name (up to 16 characters) by dialing 2-digits for each character.
8. Press the [HOLD/SAVE] button.
9. To store another Speed Dial number, repeat this procedure from Step 3.

To make a call using System Speed Dial from a DKT,

1. Lift the Handset or press the [On/Off] button.
2. Press the [SPEED] button.
3. Dial the System Speed Dial bin (2000 to 2499).

To make a call using System Speed Dial from a SLT,

1. Lift the handset.
2. Dial the Speed Dial access code 5 5 8.
3. Dial the System Speed Dial bin (2000 to 2499).

#### Consideration

- If you dial an empty System Speed bin, an error tone will be heard.

- If a CO line is selected before dialing a System Speed Dial bin number, the selected CO line is used regardless of any CO line assignment in the System Speed Dial bin.
- When accessing a Speed Dial number, busy tone is received if all CO Lines in the group are busy. You may use Callback (Ref. C) to queue for the next available CO line from the group.
- System Speed Dial numbers are protected from loss of system power.

### Reference

- A. Class of Service (COS)
- B. ISDN Service
- C. Call Back

### Admin Programming

System Numbering Plan (PGM 106, Flexible button 15)

Assigns code for SLT access to Speed Dial

Speed Dial Access (PGM 112, Flexible button 9)

Enables access to Speed Dial feature for a range of stations

CO Dial Tone Detect (PGM 160, Flexible button 6)

Enables dial tone detection for a CO line

System Speed Zone (PGM 232)

Assigns System Speed Dial bins into groups or zones

---

### 3.12.7.1 System Speed Zone

#### Description

System Speed Dial bins can be divided into zones (groups) and stations allowed or denied access to the Speed Dial numbers in a zone. Up to 10 Speed zones can be defined. Toll checks based on COS can be applied to zones (refer to Ref. A). Speed Dial bins not allocated to a zone can be accessed by all stations and no toll checks are applied.

#### Reference

- A. COS
- B. System Speed Dial

#### Admin Programming

Speed Dial Access (PGM 112, Flexible button 9)

Allows Stations access to System Speed Dial

System Speed Zone (PGM 232)

Allows Stations to access Speed Dials in a Zone

### **3.13 Private Line**

---

#### Description

CO lines in the system can be assigned for exclusive use by one or more DKT users. Private lines are assigned to CO Group 00. Also, at the DKT, a button must be assigned for the appearance of the specific Private Line, a Loop or Pool keys cannot be used.

#### Operation

A Private Line operates as a normal CO line except access is limited to stations using a Flexible button to access the CO line.

#### Consideration

- Incoming Private Line calls cannot be answered with Direct or Group Pick-up unless the station has an appearance of the Private Line.

#### Admin Programming

Flexible Buttons (PGM 115)

CO Line Group (PGM 141, Flexible button1)

### **3.14 Station Serial Call**

---

#### Description

Using DSS buttons, users can place consecutive Intercom calls without returning the station to an idle state (no need to hang-up) between calls.

#### Operation

To use serial calling,

1. Press the appropriate DSS flexible button
2. The active call is disconnected and the new call established.

#### Reference

- A. Flexible Buttons

### **3.15 Walking COS**

---

#### Description

A user may temporarily override the toll restriction of a station to make toll calls from a normally toll restricted Station. The user must input an Authorization Code (refer to Ref. A) in order to activate Walking COS and is subject to the COS assigned to the station associated with the Authorization Code. The Walking COS is applied after the code is entered for the next CO line call.

#### Operation

To activate Walking COS from a DKT,

1. Press the [TRANS/PGM] button.
2. Dial 2 3.
3. Dial the Authorization Code; intercom dial tone is received and the COS (Ref. C) is temporarily changed.

To activate Walking COS from an SLT,

1. Lift the handset.
2. Dial 5 6 3.
3. Dial 2 3.
4. Dial the Authorization Code; intercom dial tone is received and the COS (refer to Ref. C) is temporarily changed.

#### Consideration

- A Flexible button may be assigned to activate Walking COS.
- Releasing the CO line after activating Walking COS terminates Walking COS and the station returns to the original COS.
- Flash may be used to maintain access to a CO line after a call or if a wrong number is dialed otherwise, the user must re-activate Walking COS.
- The SMDR record for a call placed with Walking COS will show the station number associated with the Authorization code not the actual Station.
- Walking COS employs the COS associated with the Authorization code even if the station COS has been set at COS 7 with Temporary COS..

#### Reference

- A. Authorization
- B. SMDR
- C. Class Of Service

#### Admin Programming

Authorization Code Table (PGM 227)

Assigns Authorization Codes for each station and system codes

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## 4 CALL HANDLING SERVICES

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### 4.1 Absent Text Message

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#### Description

A user can choose a Pre-selected (refer to Ref. C) or Custom (refer to Ref. D) message to be shown in the DKT LCD of a calling user. When a user calls to a station with Absent Text Message active, the display of the calling users indicates the selected message.

#### Operation

To activate Absent Text Message from the DKT,

1. Press the [TRANS/PGM] button.
2. Dial 5 1.
3. Dial the 2-digit message code 00 – 20.
4. Press the [HOLD/SAVE] button.

To cancel Absent Text Message from the DKT,

1. Press the flashing [DND/FOR] button or dial '5 5 3'.

#### Consideration

- Activating Call Forward (refer to Ref. A) cancels any active Absent Text Message.
- When Absent Text Message is active, the [DND/FOR] button will flash as applicable.
- If Pre Set Msg DND is assigned, when activating Absent Text Message, the station is placed in DND.

#### Reference

- A. Call Forward
- B. DND (Do Not Disturb)
- C. Pre-selected Message
- D. Custom Message

#### Admin Programming

P-Msg DND (PGM113, Flexible button 16)

Assigns Absent Text Message to activate DND

#### 4.1.1 Custom Message

---

#### Description

A user can select from one of 10 Custom messages entered by the System Att'd (refer to Ref. A) or one (1) entered by the user. Each Custom message can be up to 24 characters.

#### Operation

To program Custom Message 00 from a station,



1. Press the [TRANS/PGM] button.
2. Dial 5 2.
3. Enter the message using the character chart below, up to 24 characters.
4. Press the [HOLD/SAVE] button and confirmation tone should be heard.

Digit Map Chart

Q - 11	A - 21	D - 31
Z - 12	B - 22	E - 32
. - 13	C - 23	F - 33
1 - 10	2 - 20	3 - 30
G - 41	J - 51	M - 61
H - 42	K - 52	N - 62
I - 43	L - 53	O - 63
4 - 40	5 - 50	6 - 60
P - 71	T - 81	W - 91
Q - 72	U - 82	X - 92
R - 73	V - 83	Y - 93
S - 74	8 - 80	Z - 94
7 - 70	9 - 90	
*1 - Blank		
*2 - :	0-00	
*3 - ,		

To program Custom Messages 11 - 20 from the System Attendant,

1. Press the [TRANS/PGM] button.
2. Dial 0 5 3 and the message number (11-20).
3. Enter the message; refer to the Digit Map above, up to 24 characters.
4. Press the [HOLD/SAVE] button and confirmation tone is heard.

To activate LCD Messages,

1. Press the [TRANS/PGM] button.
2. Dial 5 1.
3. Dial the 2-digit message code 00 - 20.
4. Press the [HOLD/SAVE] button.

To cancel LCD Messages,

1. Press the flashing [DND/FOR] button.

To cancel LCD Messages (Custom/Pre-selected) from the System Attendant,

1. Press the [TRANS/PGM] button.
2. Dial 0 5 2.
3. Dial the desired station range.
4. Press the [HOLD/SAVE] button.

### Reference

Attendant Positions

#### 4.1.2 Pre-selected Message

---

##### Description

There are ten Pre-selected messages some of which provide for user detail entry such as time or date. The Pre-selected messages include:

- Message 01: LUNCH, RETURN AT *HH:MM*
- Message 02: ON VACATION /RETURN AT DATE *DD:MM*
- Message 03: OUT OF OFFICE/RETURN TIME AT *HH:MM*
- Message 04: OUT OF OFFICE/RETURN AT DATE *DD:MM*
- Message 05: OUT OF OFFICE/RETURN UNKNOWN
- Message 06: CALL: (*Up to 17 digits*)
- Message 07: IN OFFICE: STA *XXXX*
- Message 08: IN A MEETING/RETURN TIME AT *HH:MM*
- Message 09: AT HOME
- Message 10: AT BRANCH OFFICE

##### Operation

To activate LCD Messages,

1. Press the [TRANS/PGM] button.
2. Dial 5 1.
3. Dial the 2-digit message code 00 - 20.
4. Press the [HOLD/SAVE] button.

To cancel LCD Messages,

1. Press the flashing [DND/FOR] button.

## **4.2 Automatic Fax Transfer**

---

### Description

An analog CO line assigned Automatic FAX Transfer will transfer incoming FAX calls to Station 107. When a call is received, the system will answer the call and, if FAX tone (1100Hz, 0.5s On/3s Off repeated) is detected, the call is transferred to the FAX machine attached to Station 107. If FAX tone is not detected within the FAX Tone Detect time, the System will route the call to the appropriate Station(s).

### Consideration

- For proper operation, a CO line must be assigned as the FAX CO line. Only one CO line can be assigned for Automatic FAX Transfer.
- Station 107 (port 8) is used as the FAX Station. The FAX machine should be connected to port 8 for proper operation.
- If the FAX CO line is not answered within the FAX CO call time, the incoming call will be disconnected.
- An outside caller connected to the FAX CO line will hear ring back tone while the system attempts to detect FAX tone.
- In order to transfer calls from the FAX to a FAX station, do not assign CO ring to the FAX Station 107.
- DISA, if assigned to the FAX CO line, will override operation of Automatic FAX Transfer operation.
- When the FAX machine goes idle after a FAX call, the associated CO line is released.
- If the FAX CO line is disconnected during a FAX call, the CO line is released and the FAX machine should return to an idle state.

### Admin Programming

Auto FAX Transfer CO (PGM 161, Flexible button 17)

Enables operation of Auto FAX Transfer for the CO line

FAX tone Detect Timer (PGM 182, Flexible button 13)

Sets the time for detecting FAX tone before rerouting the call

FAX CO Call Timer (PGM 182, Flexible button 14)

Sets the time the ipLDK will await answer before disconnecting a call on the FAX CO line, if no FAX tone is detected

### **4.3 Automatic Hold**

---

#### Description

While on a call, a station can activate features that will place the active call on Hold. Features such as Transfer, Call Park and calling another station will activate Automatic Hold. In addition, the system can be configured to place a CO line on Hold if another CO Line is accessed by the user.

#### Consideration

- Automatic Hold activates the preferred Hold.

#### Reference

- A. CO Line Call Hold
- B. Intercom Call Hold
- C. Hold Preference

#### Admin Programming

Automatic Hold (PGM 112, Flexible button 2)

Assigns Automatic Hold when a second CO Line is accessed

Hold Preference (PGM 160, Flexible button 8)

Assigns the preferred hold as System or Exclusive

Attendant Recall Timer (PGM 180, Flexible button 1)

Sets the time the Attendant and holding station ring before the call is dropped

Exclusive Hold Recall Timer (PGM 180, Flexible button 4)

Sets the time a call will remain on Exclusive Hold

I-Hold Recall Timer (PGM 180, Flexible button 5)

Sets the time the holding station will ring before the Attendant also receives recall

System Hold Recall Timer (PGM 180, Flexible button 6)

Sets the time a call will remain on System Hold before recalling the holding station

Transfer Recall Timer (PGM 180, Flexible button 7)

Sets the time a call will remain on Transfer Hold before recalling the station

Call Park Recall Timer (PGM 180, Flexible button 2)

Sets the time a call will remain on Park Hold before recalling the station

## **4.4 Automatic Privacy**

---

### Description

The system automatically invokes privacy on Intercom and conference calls. As a default, Automatic Privacy is implemented on all CO line calls. If desired, Privacy on CO line calls can be disabled, allowing other stations in the system to access CO line calls active at other stations.

This capability is often used in a small office or group to allow the active user to ask a nearby user to join or answer the call without the need to transfer or place the call on hold. The user need only press the CO line button to access the call. When a station enters a conversation in this way, a Warning tone can be provided to others in the conversation.

### Consideration

- When Automatic Privacy is enabled, selecting a busy CO line button will return busy tone.
- A station can only override a privacy-disabled Station.
- The station will present an Intrusion tone when another Station accesses the line.

### Reference

- A. Conference
- B. Barge-in
- C. Emergency Intrusion

### Admin Programming

Override Privilege (PGM 113, Flexible button 4)

Permits a station to override an active call at another station

Auto Privacy (PGM 161, Flexible button 5)

Disables Automatic Privacy on CO line calls

Privacy Warning Tone (PGM 161, Flexible Button 6)

Assigns a warning tone when intruding on an active CO line call

## **4.5 Barge In**

---

### Description

An authorized station can intrude on an active call at another Station. The intruding station can monitor the call and may be allowed to join and disconnect the active call.

### Operation

To Barge-in from a DKT,

1. Call the busy station and receive busy tone.
2. Use the navigation buttons to expose and select the [MONITOR] softkey
3. Warning tone is given to the existing conversation and the intruding user can monitor the conversation.
4. Select the [JOIN] softkey to join the conversation.
5. Once joined in the conversation, use the [DROP] softkey to disconnect from the call.

### Consideration

- Barge-in is not permitted on a call placed by an Attendant.

### Admin Programming

Barge In mode (PGM 113, Flexible button 13)

Enables a Station to barge-in on calls at other Stations

## 4.6 Call Log

---

### Description

The LDP-6212 and 6230 DKT maintain logs of received, dialed, and lost CO line calls. The log displays the telephone number (Caller Id or dialed), the time and date of the call. Also, if the number matches a Speed Dial, the Name assigned to the Speed Dial is displayed. User can view and recall numbers from the logs. The station must have a Flexible button assigned for the Call Log feature

### Operation

To assign a Call Log feature button,

1. Press the [TRANS/PGM] button.
2. Press the Flexible button to be assigned.
3. Press the [TRANS/PGM] button.
4. Dial 5 7, the Call Log code.
5. Press the [HOLD/SAVE] button.

To use the Call Log feature of the LDP-6230D/6212D,

1. Press the Call Log button.
2. Dial the desired log
  - 1. Received
  - 2. Dialed
  - 3. Lost
3. Use the Volume Up/Down buttons to scroll through the list.
4. To call a displayed number, press the OK softkey.
5. Select the Lost Call Menu to view CLI MSG-Wait numbers. A number in the Lost Call list can be answered, deleted, or saved by pressing the Select softkey.

### Consideration

- The number of records available to a station is assigned as the Call Log List number and can be from 15 to 50 records.
- The maximum number of records available in the system is 500. The system pre-allocates memory for each available record and allocates records to stations beginning with the lowest station port until records are exhausted or the last port is allocated records.
- When the Call Log List Number is assigned in PGM 160, all Call Log records are erased as the system reallocates memory.
- CLI MSG-Wait must be enabled for the station to view and use the Lost Call Log.

### Admin Programming

Call Log List Number (PGM 160, Flexible button 19)

Assigns the number of records allocated for each station

CLI MSG-Wait (PGM 114, Flexible button 4)

Enables logging of lost CO line calls with Caller ID.

## **4.7 Call Park**

---

### Description

Call Park establishes a virtual location called an Orbit to hold a call where other users can retrieve the call through a dial code. Typically, a call is parked at a location (601 to 610) and a page made to inform the desired user of the call and Park orbit. The paged user can then access the call from any phone connected to the system by dialing the Park orbit.

While Parked, the held party receives MOH. Also, Parked calls recall after the Call Park Recall timer expires. If not answered in the Station Auto Release Timer, the recall will stop.

### Operation

To park a call,

1. Press the [TRANS/PGM] button.
2. Dial the Park location 601-610.
3. Replace the handset or go on-hook.
4. Page the desired user to retrieve the call (refer to Ref. A).

To retrieve the parked call from a system phone,

1. Lift the Handset or press the On/Off button.
2. Dial the Park location to retrieve the call.

### Consideration

- If the call remains unanswered for the duration of the Call Park Recall time, the station that parked the call will receive a recall. For CO line calls, if the call is still unanswered, then the attendant will receive the recall. If the Attendant does not answer again in the Attendant Recall time, the CO line call will be disconnected and the line will be returned to an idle state.
- In case of an internal call, the call terminates after the I-Hold Recall time.

### Reference

- A. Paging
- B. CO Line Call Hold
- C. Music On Hold (MOH)

### Admin Programming

System Numbering Plan (PGM 107, Flexible button 5)

Assigns dial codes to park a call

Call Park Recall Timer (PGM 180, Flexible button 2)

Sets the time the call will remain parked before recall to the station

Station Auto Release Timer (PGM 182, Flexible button 5)

Sets the duration of the Call Park Recall



## 4.8 Call Transfer

---

### Description

A user can answer and then send a call to another destination. A Transfer may be screened, the call is announced to the receiving party, or unscreened, the call is transferred without an announcement. A call can be transferred to another station, Station Group, or CO line call.

If the receiving destination does not answer the call in the Transfer Recall time, the transferring station and the transferred station will receive recall. If the call remains unanswered, the attendant will also receive a recall for the Attendant Recall time. After that, the transferred call will be disconnected.

### Reference

- A. Call Transfer CO line
- B. Call Transfer Intercom

### Admin Programming

Transfer Recall Timer (PGM 180, Flexible button 7)

Sets the time a call will remain on Transfer Hold before recalling the transferring station

I-Hold Recall Timer (PGM 180, Flexible button 5)

Sets the time the holding station will ring before the Attendant also receives recall

Attendant Recall Timer (PGM 180, Flexible button 1)

Sets the time the Attendant and holding station ring before the call is dropped

### 4.8.1 Call Transfer, CO line

---

#### Description

A Station user may transfer an active call to a CO line call.

#### Operation

Unscreened CO line call transfer while on an active call DKT,

1. Press the [TRANS/PGM] button, the active call is placed in Transfer Hold and intercom dial tone is received.
2. Place a call to the external number to receive the transfer.
3. Replace the handset and go on-hook to complete the transfer.

Screened CO line call transfer DKT,

1. Press the [TRANS/PGM] button, the active call is placed in Transfer Hold and intercom dial tone is received.
2. Place a call to the external number to receive the transfer.
3. When the called party answers, announce the call transfer.
4. Replace the handset and go on-hook to complete the transfer.

To transfer an active CO call from an SLT, unscreened,

1. Press and release the hook-switch, the active call is placed in Exclusive Hold and intercom dial tone is received.
2. Place a call to the external number to receive the transfer.
3. Replace the handset and go on-hook to complete the transfer.

To transfer an active call from an SLT screened,

1. Press and release the hook-switch, the active call is placed in Exclusive Hold and intercom dial tone is received.
2. Place a call to the external number to receive the transfer.
3. When the station answers, announce the call transfer.
4. Replace the handset and go on-hook to complete the transfer.

### Consideration

- CO line to CO line connection must be enabled to support transfers of a CO line to a CO line.
- To support a CO line to CO line transfer, analog CO lines must provide and detect 'open-loop' disconnect supervision. If disconnect supervision is not supported, the CO line may stay in a busy state after the call is completed.
- For proper operation of CO line to CO line unscreened transfer, the receiving CO line should support 'Answer supervision'. Answer supervision is commonly supported by ISDN lines and not analog CO lines. If Answer supervision is supported, after the Transfer Hold Recall time, the transferring station receives recall. If the station does not respond to the Recall after the I-Hold Recall time, the Attendant also receives recall. If the call remains unanswered after the Attendant recall time both CO lines disconnected. If Answer Supervision is not supported, the transferring station does not receive recall.
- CO line to CO line transfers set-up an Unsupervised Conference and are subject to the conditions of an Unsupervised Conference. Calls are subject to disconnect after expiration of the Unsupervised Conference.
- Pressing a CO line button before completing the transfer will cause the outgoing call to disconnect and failure of the transfer.

### Reference

- A. Unsupervised Conference

### Admin Programming

Transfer Recall Timer (PGM 180, Flexible button 7)

Sets the time a call will remain on Transfer Hold before recalling the transferring Station

I-Hold Recall Timer (PGM 180, Flexible button 5)

Sets the time the holding station will ring before the Attendant also receives recall

Attendant Recall Timer (PGM 180, Flexible button 1)

Sets the time the Attendant and holding station ring before the call is dropped

Open Loop Detect Timer (PGM 142, Flexible button 13)

Sets the duration of an for disconnect supervision for analog lines

Unsupervised Conference Timer (PGM 182, Flexible button 6)

Sets the duration that two CO lines may be connected without supervision from a station

#### 4.8.2 Call Transfer, Internal

---

##### Description

A user can transfer an active call to another station or Hunt Group within the system. The transfer can be screened (announced) or unscreened. The transferred call is placed on Transfer Hold and, if the receiving station is a DKT, the CO line button LED will flash indicating an Exclusive Hold state.

If the receiving station does not answer the call in the Transfer Recall time, the transferring station and the transferred station will receive recall. If the call still remains unanswered, the attendant will also receive a recall for the Attendant Recall time. After that, the transferred call will be disconnected.

In addition, a DKTU user can transfer a call directly to a Voice Mail box.

##### Operation

To transfer an active call from a DKT unscreened,

1. Press the [TRANS/PGM] button, the active call is placed in Transfer Hold and intercom dial tone is received.
2. Dial the station number to receive the transfer.
3. Replace the handset or go on-hook to complete the transfer.

To transfer an active call from a DKT screened,

1. Press the [TRANS/PGM] button, the active call is placed in Transfer Hold and intercom dial tone is received.
2. Dial the station number to receive the transfer.
3. When the station answers, announce the call transfer.
4. Replace the handset or go on-hook.

To transfer an active call to a Voice Mailbox from a DKTU,

1. Press the [TRANS/PGM] button, the active call is placed in Transfer Hold and intercom dial tone is received.
2. Press the [CALLBACK] button.
3. Dial the station number to receive the transfer.
4. Replace the handset or go on-hook to complete the transfer.

To transfer an active call from an SLT unscreened,

1. Press and release the hook-switch, the active call is placed in Exclusive Hold and intercom dial tone is received.
2. Dial the station number to receive the transfer.
3. Replace the handset and go on-hook to complete the transfer.

To transfer an active call from an SLT screened,

1. Press and release the hook-switch, the active call is placed in Exclusive Hold and intercom dial tone is received.

2. Dial the station number to receive the call transfer.
3. When the station answers, announce the call transfer.
4. Replace the handset and go on-hook to complete the transfer.

#### Consideration

- An SLT conducting a transfer can speak to either party by momentarily pressing the hook-switch to activate a Broker's Call.
- It is impossible to transfer a call to another busy SLT from an SLT. When receiving busy tone, the SLT user can be connected to the CO Line with hook-flash.

#### Reference

- A. Broker's Call
- B. CO line Call Hold

#### Admin Programming

Voice Over (PGM 113, Flexible button 6)

Enables a station to receive a Voice Over call when busy

Caller Voice Over (PGM 111, Flexible button 22)

Enables a station to place a Voice over call to a busy station

Transfer Recall Timer (PGM 180, Flexible button 7)

Sets the time a call will remain on Transfer Hold before recalling the transferring station

I-Hold Recall Timer (PGM 180, Flexible button 5)

Sets the time the holding station will ring before the Attendant also receives recall

Attendant Recall Timer (PGM 180, Flexible button 1)

Sets the time the Attendant and holding station ring before the call is dropped

## **4.9 Camp-on**

---

### Description

When a user dials a busy Station, busy tone is received. The user may leave a Call Back request (refer to Ref. A) or may camp-on to the busy station. When the user camps on, the busy station receives a double beep tone as an indication of the call waiting. The waiting user receives MOH.

### Operation

To activate Camp-on with a DSS/BLF button,

1. Press the DSS button for the busy Station.

To activate Camp-on while receiving Intercom busy tone,

1. Dial '\*’.

To answer a waiting call after receiving Camp-on tone at a DKT,

1. Press the [HOLD/SAVE] button; the active call is placed on exclusive hold and the call waiting is connected.
2. Press the [HOLD/SAVE] button, to alternate between parties

To answer a waiting call after receiving Camp-on tone at an SLT,

1. Press and release the hook-switch, Flash.
2. Dial 560.

To return to the original call at an SLT,

1. Press and release the hook-switch, Flash.
2. Dial 560.

### Consideration

- While in a Conference, receiving a Page or in DND, a Station cannot receive a Call Waiting (refer to Ref. B, C and D).
- The Attendant can override a Station using the Camp-on feature (refer to Ref. E).
- If Stop Camp-on Tone is enabled, no camp-on tone is heard by the station receiving the camp-on and only the [Hold/Save] button LED will flash.

### Reference

- A. Station Message Wait/Call Back
- B. Conference
- C. Paging
- D. DND
- E. Attendant Override
- F. Voice Over

### Admin Programming

System Numbering Plan (PGM 106, Flexible button 16)

---

Assigns the code for an SLT to activate System Hold

Stop Camp-on Tone (PGM 112, Flexible button 15)

Disables camp-on tone to the busy station

Voice Over (PGM 113, Flexible button 6)

Enables a station to receive a Voice Over call when busy

Caller Voice Over (PGM 111, Flexible button 22)

Enables a station to place a Voice over call to a busy station

## **4.10 CO Line Call Hold**

---

### Description

A user can place a CO line call on hold. The held call will receive MOH from the assigned source. The call may be placed on Exclusive Hold where only the holding station can pick-up the held call or the call may be placed on System Hold and any station with access to the CO line may pick-up the held call. For Exclusive hold, the CO line LED at other stations will be On, for System Hold, the LED will flash at all stations. Pressing the [HOLD/SAVE] button on the DKT places the call on Preferred Hold (refer to Ref. A), default System Hold. Pressing the [HOLD/SAVE] button twice places the call on the non-Preferred Hold, Exclusive Hold.

The call will remain on hold until picked-up or the Exclusive or System Hold Recall timer, as appropriate, expires. After the Hold Recall time expires (Exclusive or System), the holding station receives recall ring for the I-Hold Recall time. After the I-Hold Recall time, a call on Exclusive Hold is placed on System hold and the station and Attendant will receive ring. If the call remains unanswered for the Attendant Recall time, the call is dropped.

### Operation

To place a CO Line on Hold from a DKT,

1. Press the [HOLD/SAVE] button once, the call is placed in the preferred (Exclusive or System) Hold.
2. Press the Hold button a second time, the call is placed in the non-preferred Holds.

To place a CO Line on Hold from an SLT,

1. Press and release the hook-switch.
2. Dial the Hold code 5 6 0.

To access a held CO Line from the holding DKT,

1. Lift the handset.
2. Press the {CO} button.

To access a held CO Line from the holding SLT,

1. Lift the handset.
2. Dial '8 \*'.

### Reference

- A. Hold Preference
- B. Automatic Hold
- C. Music on Hold

### Admin Programming

System Numbering Plan (PGM 106, Flexible button 17)

Assigns the code to place a call on hold from an SLT

System Numbering Plan (PGM 107, Flexible button 10)

Code to access a Held CO Line from a CO Line Group

System Numbering Plan (PGM 107, Flexible button 11)

Code to access a specific Held CO Line

Hold Preference (PGM 160, Flexible button 8)

Assigns the preferred hold as System or Exclusive

Attendant Recall Timer (PGM 180, Flexible button 1)

Sets the time the Attendant and holding station ring before the call is dropped

Exclusive Hold Recall Timer (PGM 180, Flexible button 4)

Sets the time a call will remain on Exclusive Hold

I-Hold Recall Timer (PGM 180, Flexible button 5)

Sets the time the holding station will ring before the Attendant also receives recall

System Hold Recall Timer (PGM 180, Flexible button 6)

Sets the time a call will remain on System Hold before recalling the holding station



## **4.11 Emergency Intrusion**

---

### Description

An Attendant or Emergency Supervisor station can be allowed to intrude on a conversation of a busy station. The busy station receives a short warning tone then the existing call is disconnected and the intruding station is connected.

### Operation

To activate from an Emergency Intrusion DKT,

1. Place call to busy Station
2. From the LCD menu select Monitor then Join to join the conversation.

### Consideration

- A station assigned Emergency Intrusion cannot intrude on an Attendant call, a station in DND, or a station in a networked system.
- If both stations of a Linked Pair are busy, the Emergency Intrusion is directed to the Master of the Linked pair.
- If both the Executive and Secretary of a pair are busy, the Emergency Intrusion is directed to the Secretary.

### Reference

- A. Attendant
- B. Do Not Disturb (DND)
- C. Linked Pairs
- D. Executive/Secretary Pairs

### Admin Programming

Emergency Supervisor (PGM 112, Flexible button 24)

Assigns the Emergency Supervisor allowing Emergency Intrusion from the station

## **4.12 Forced Handsfree Mode**

---

### Description

A DKT station user can change the Intercom Signal Mode of the called station from Tone to Handsfree mode for a call. When receiving ring back tone, the calling station can change the call to Handsfree. The calling party is then connected to the called station with audio over the speaker of the called station and the called station user may reply hands free.

### Operation

To activate Forced Handsfree mode,

1. Dial '#' while receiving ring back tone on an Intercom call to another DKT.
2. After confirmation tone, the caller's voice is heard over the called DKT station speaker and the called user can respond Handsfree.

### Consideration

- Forced Hands-free is not supported when calling a Linked-pair Station or if the Mobile Extension feature is active.
- Use of Forced Hands-free requires the calling station be enabled for Forced Hands-free use.
- A Hands free intercom call is considered as an answered call. Thus, features such as Call Back, Call Forward No answer, etc. will not operate in the hands free signaling mode.

### Reference

- A. Station Message Wait/Call Back
- B. Call Forward, No Answer
- C. Intercom Signaling Mode
- D. Linked Pair
- E. Mobile Extension

### Admin Programming

Forced Hands-Free Mode (PGM 111, Flexible button 19)

Enables the user to force an intercom call to change from Tone to Hands free mode

### **4.13 Forced Trunk Disconnect**

---

#### Description

An Attendant or Emergency Supervisor station can be allowed to force a busy CO line to return to idle disconnecting the active call. Once the CO line returns to idle, it is connected to the Attendant or Emergency Supervisor station.

#### Operation

To activate Forced Trunk Disconnect feature:

1. Seize a CO line.
2. At busy tone and dial '#'.
3. Busy line goes to idle and connects to the Station.

#### Consideration

- Forced Trunk Disconnect Feature is only possible for analog CO lines.

#### Reference

- A. Emergency Intrusion

#### Admin Programming

Emergency Supervisor (PGM112, Flexible button24)

Assigns the Emergency Supervised allowing Forced Trunk Disconnect for the Station

#### **4.14 Hold Preference**

---

##### Description

When a CO line is placed on hold, the DKT user may place the call on Preferred Hold (Exclusive or System) by a single press of the [Hold/Save] button. Pressing the [Hold/Save] button again toggles the Hold to the non-preferred Hold state.

##### Reference

- A. CO Line Call Hold

##### Admin Programming

Hold Preference (PGM 160, Flexible button 8)

Defines the preferred Hold, Exclusive or System

## **4.15 Intercom Call Hold**

---

### Description

A user can place an intercom call on hold. The held call will receive MOH from the assigned source. When the station initiating the hold returns to idle, the station will receive Hold recall for the I-Hold Recall time then the call is terminated.

### Operation

To place an Intercom call on Hold from a DKT,

1. Press the [HOLD/SAVE] button once, the call is placed in Intercom Hold.

To place a Intercom call on Hold from an SLT,

1. Press and release the hook-switch.
2. Dial the Hold code '5 6 0'.

To access a held intercom call from the holding station

1. Return to idle, the station will receive recall for the held Intercom call,
2. Lift the handset.

### Consideration

- A held intercom call is only available to the station placing the call on Hold.
- A held intercom call will be released if not answered by the holding station in the I-Hold Recall time.

### Reference

- A. Music On Hold

### Admin Programming

System Numbering Plan (PGM 106, Flexible button 17)

Assigns the code to place a call on hold from an SLT

I-Hold Recall Timer (PGM 180, Flexible button 5)

Sets the time the holding station will ring before the holding intercom call is dropped

## **4.16 Music On Hold (MOH)**

---

### Description

When a call is placed on hold manually (refer to Ref. A) or automatically (refer to Ref. B), the party on hold will hear music. Music On Hold (MOH) is supplied through various music sources. MOH is generally provided by an external source connected to the MOH connector of the BKSU. Alternatively, the internal music or hold tone can be used or an external source can be connected to an MOHU connected to an SLT port.

### Consideration

- Only one (1) MOH channel is supported.
- The MOH source can be used in place of Ringback tone when calling a busy Attendant or to a camped on party.
- SLT ports connected with MOHU can be used as the music source.

### Reference

- A. CO Line Call Hold
- B. Automatic Hold
- C. Intercom Call Hold

### Admin Programming

CO Line MOH (PGN 142, Flexible button 6)

Assigns the MOH source for the CO line

Music Type (PGM 171, Flexible button 2)

Assigns the system MOH source

Attendant Call Queue (PGM 160, Flexible button 1)

Assigns source for ringback tone when calling a busy Attendant

Camp-on RBT (PGM160, Flexible button 2)

Assigns source for ringback tone when camping a call to a busy station

Transfer Tone (PGM 160, Flexible button15)

Assigns either ringback tone or MOH for a transferred call

Internal Music Type (PGM171, Flexible button 8)

Assigns one of the 13 available music types as the internal MOH

SLT MOH (PGM171, Flexible button 4)

Assigns the SLT port to use for MOH

## **4.17 On-Hook dialing**

---

### Description

A station can place a call without lifting the handset. When a digit is dialed, the speakerphone is automatically activated.

### Operation

To use On-hook dialing,

1. Dial the number as usual.

### Consideration

- When dialing Off-hook, dialing must begin within the ICM Dial Tone timer. Whether dialing On or Off-hook, subsequent digits must be dial in the inter-digit time or error tone is received and the LCD displays 'Dialing too late'.

### Admin Programming

Auto Speaker Selection (PGM 111, Flexible button 1)

Enable the speakerphone to activate when a digit is dialed or button pressed

ICM Dial Tone Timer (PGM 181, Flexible button 7)

Sets the duration the user receives dial-tone before time-out

Inter-digit Timer (PGM 181, Flexible button 8)

Sets the time allowed between dialed digits

## **4.18 Station and Group Call Coverage**

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### Description

This feature permits a user (the covering station) to receive ring and answer calls directed to a covered station or Station Group. A Flexible button must be assigned at the covering Station for calls to the covered station or group.

The covered station must activate call coverage and can set a delay before the covering station(s) receives notification for the incoming call. For Station Group Call Coverage, the group must have coverage activated. A station may provide Call Coverage for multiple stations or groups, requiring a separate button for each. Also, multiple stations can provide Call Coverage for a station or group.

When a covered station or group rings, the {Call Coverage} button LED will flash and the covering station may receive ring (immediate or delayed) for the call. The covering station can answer the call using the {Call Coverage} button, terminating ring at other stations. Once answered, the LED of {Call Coverage} buttons for the station at any covering stations will extinguish.

### Operation

To assign a Flexible button for Call Coverage, at the covering DKT,

1. Press [TRANS/PGM] button
2. Press the desired Flexible button.
3. Press [TRANS/PGM] button
4. Dial '4 6'.
5. Enter Station or Group number to cover.
6. Press [HOLD/SAVE] button.

To enable Call Coverage at the covered DKT

1. Press [TRANS/PGM] button.
2. Dial '4 5'.
3. Dial '1' to select cover operation.
4. Enter '1' to enable or 0 to disable Call Coverage.
5. Press [HOLD/SAVE]

To set Call-Coverage ring Delay at the covered DKT

1. Press [TRANS/PGM] button.
2. Dial '4 5'.
3. Dial '2' to assign a delay.
4. Dial delay in ring cycles (0 ~ 9).
5. Press [HOLD/SAVE]

To answer a Coverage call from the covering station,

1. Press the rapidly flashing Coverage button or lift the handset.
2. The call is answered and other stations stop receiving notification for the call.



Consideration

- A station user can cover an SLT extension. An SLT cannot perform the Call Coverage function.
- If the Coverage station is in DND, no audible ring will be heard.
- Call coverage is not available for Hunt group members.
- Call Coverage applies to all calls to a covered station or group.

Reference

- A. Station User Programming
- B. Flexible buttons

Admin Programming

Station Call Coverage (PGM 113, Flexible button 18)

Enables Call Coverage for the Station

Group Call Coverage (PGM 191)

Enables Call Coverage and sets delay ring

## **4.19 Station Message Wait/Call Back**

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### Description

When a user calls a station that is busy, in DND (refer to Ref. B), or does not answer, the calling user can activate Station Message Wait or Call Back. The ipLDK determines if Call Back or Message Wait is activated based on the status of the called station. If the called station is busy, a Call Back is activated. If the called station is in DND or does not answer, a Station Message Wait is activated.

When a user activates Call Back to a busy station, the ipLDK will monitor the status of the station. When the busy station returns to idle, the station that activated the Call Back will receive recall ring and the Call Back button LED flashes. When the recall is answered, the originally busy station is called. A station can activate any number of Call Back requests. A station can receive a Call Back request from any number of stations however, only the first Call Back request is stored as a Call Back, the remaining are stored as Message Waits.

When a user calls a station that does not answer or is in DND, Call Back activates a Station Message Wait. The DKT station will indicate the message with a flashing Call Back button and the LCD will indicate the message status. When the station responds to a Station Message Wait, an intercom call is placed to the station that left the message. A station can receive a maximum of five (5) Station Message Waits. Stations attempting to activate additional Message Waits to a station with five station messages will receive error tone.

In addition to the flashing Call Back button, a DKT station with a Station Message Waiting may receive periodic audible reminders of the waiting message. The tone is sent over the speaker only while the station is idle. For SLTs, an audible reminder is available in the form of stutter dial tone when the user goes 'off-hook'.

### Operation

To activate a Station Message Wait/Call Back from a DKT,

1. Press the [CALLBK] button.
2. Hang-up to return to idle. Note, if the called station has a VMIB mailbox, after the User greeting, a voice message can be left.
3. If the called station is busy, a Call Back is activated. If the called station does not answer or is in DND, a Station Message Wait is activated.

To answer a Station Message Wait at a DKT,

1. Press the flashing [CALLBK] button.
2. An intercom call is placed to the station that left the message.

To activate a Station Message Wait/Call Back from a SLT,

1. Hook-flash.
2. Dial 5 5 6'.
3. Replace the handset. Note, if the called station has a VMIB mailbox, after the User greeting, a voice message can be left.

4. If the called station is busy, a Call Back is activated. If the called station does not answer or is in DND, a Station Message Wait is activated.

To respond to a Station Message Wait from an SLT,

1. Lift the handset; stutter dial tone indicates a message waiting.
2. Dial 5 5 7, the Call Back/Station Message Wait response code.
3. An intercom call is placed to the station that left the message.

### Consideration

- A station may receive a Call Back from any or all stations however, after the first Call Back, additional Call Back requests are stored as Message Waits.
- A station may have a maximum of five (5) Station Message Waits.
- Station Message Waits are protected from a power failure.
- The user activating Call Back must respond to the Call Back recall in 30 seconds or the Call Back request is terminated and a Message Wait is activated to the station from the originally called station.
- The ipLDK-60 does not support SLT MWI lamps.
- When calling a station assigned a VMIB Voice mailbox (refer to Ref. C), activating message wait will connect the user to the station's mailbox and the greeting is played. The user may hang-up to leave a Station Message Wait or after the greeting may leave a voice message. If a voice message is left, the Station Message Wait is cancelled and the Voice Message is registered. In either case, the station receives the message.
- Placing a call to a station after leaving a Station Message Wait/Call Back will cancel the Station Message Wait/Call Back.
- Message Wait reminder tone repeats at a programmable interval, 01 to 60 minutes. Message Wait reminder remains active until all the messages are retrieved.
- For Call Backs and returning a Station Message Wait, the ipLDK employs tone ringing intercom calls, overriding the assigned Intercom Signaling Mode (refer to Ref. A).

### Reference

- A. Intercom Signaling Mode
- B. DND
- C. VMIB Voice Mail

### Admin Programming

System Numbering Plan (PGM 106, Flexible button 13)

Assigns the code for SLTs to activate Station Message Wait/Call Back

System Numbering Plan (PGM 106, Flexible button 14)

Assigns code for SLTs to respond to a Message Wait

Station ID Assignment (PGM 110, Flexible button 1)

Assigns the type of station terminal

MW Reminder Timer (PGM 181, Flexible button 9)

Assigns and sets reminder tone for message waiting

## 4.20 Time & Date

---

### Description

The Time and Date can be set by the Attendant, in Admin or automatically retrieved from the ISDN or PSTN when available. The ISDN network will send the local time and date in the CONNECT message when the called party answers. The system can employ this information to set the system time and date. In addition, where desired, the systems supports automatic Daylight Savings Time Adjustment, at the assigned start and end times, the system will adjust the time back and forward by 1 hour.

### Operation

To use the ISDN time and date for the system from the Attendant,

1. Press the [TRANS/PGM] button.
2. Dial 0 4 6
3. Dial 1 to enable or 0 to disable.
4. Press the [HOLD/SAVE] button.

### Consideration

- This feature may be set at the Attendant Station or through Admin.
- For an analog CO line, the time and date is sent as part of the CLI.

### Reference

- A. Attendant

### Admin Programming

PX Time/Day/Month (PGM 161, Flexible button 1)

Enables use of Time and Date from ISDN or Analog CO using CLI

Time and Date (Pgm 178, Flexible buttons 1 & 2)

Assigns the system Time and Date

DST (PGM 178 Flexible button 3)

Enables automatic DST adjustment

Set DST Start (PGM178, Flexible button 4)

Configure DST start date and time

Set DST End (Flexible button 4)

Configure DST end date and time

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## 5 DIGITAL KEY TELEPHONE FEATURES

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### 5.1 Do-Not-Disturb (DND)

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#### Description

While in DND, the station will not receive ringing for CO line calls or Page announcements and stations calling or transferring calls to a station in DND will receive fast busy tone. CO Line calls to a station in DND are sent to the Attendant or, if enabled, calls forward to the station's Voice Mailbox. While in DND, the station user can place calls in the normal fashion.

The station must be permitted use of DND to activate DND. The Attendant can override a station in DND (refer to Ref. A). By default, DND is available at all stations.

#### Operation

To toggle DND from a DKT,

1. Press the [DND/FOR] button.

To toggle DND from an SLT,

1. Lift the handset
2. Dial 5 5 3.

#### Consideration

- Pressing the [DND/FOR] button with Call Forward or Absent Text Message active will deactivate the forward or message and will not activate DND
- When a station is assigned as part of a Preset Call Forward chain and is in DND mode, the call is sent to the next station in the Call Forward chain.
- The user must record a greeting for calls to Forward to the Voice Mail, otherwise the call is sent to the DND announcement.
- Executive/Secretary assignments (refer to Ref H) will override DND forward to Voice Mail.
- The Attendant may cancel DND state at other Stations.
- DND is available to SLTs as well as DKT and IP Phone stations. For the SLT, DND is activated/deactivated by dialing the DND code '5 5 3.

#### Reference

- A. Attendant Intrusion, Override
- B. Ring Assignment
- C. Preset Call Forward
- D. Call Forward
- E. Absent Text Message
- F. Camp-on
- G. One-Time DND
- H. Executive/Secretary

Admin Programming

System Numbering Plan (PGM106, Flexible button 10)

Assigns the code to activate DND

Do-Not-Disturb (PGM 111, Flexible button 3)

Assigns DND to a range of Stations

DND Forward to Voice Mail (PGM 113, Flexible button 19)

Assigns calls to forward to the Voice Mail when the station is in DND

## 5.2 Call Screening

---

### Description

A keyset can be configured to emulate an Answering Machine to allow the user to screen calls that are sent to the station's Voice Mailbox. When a call is sent to the user's mailbox, the caller's voice can be heard over the speaker of the phone, if idle. The user can answer the call, disconnecting from the mailbox without leaving a message, or can leave the caller connected to the mailbox to complete the voice message.

There are two modes of operation for Call Screening. In the Speaker mode, the caller is automatically connected to the speaker when the call connects to the mailbox; in the silent (Ring Mode) the assigned Call Screen button flashes and the user can select to hear the caller by pressing the Call Screen button.

### Operation

To assign a Call Screening button,

1. Press the [Trans/PGM] button.
2. Press the desired Flexible button.
3. Press the [Trans/PGM] button.
4. Dial the Call Screen button code, 48.
5. Dial 1 for Ring mode.  
Dial 2 for Speaker mode, confirmation tone is received.
6. Press the [Hold/Save] button.

To activate/deactivate Call Screening while idle,

1. Press the Call Screen button, the LED lights when active.

### Call Screening Ring Mode

To screen the Voice Mail call,

1. Press the flashing Call Screen button.

To answer the screened call,

1. Press the Call Screen button.

To reject the screened call and allow the Voice mail to take the message

1. Press the Speaker button.

### Call Screening Speaker Mode

To answer a screened call,

1. Press the Call Screen button.

To reject the screened call and allow the Voice mail to take the message

1. Press the Speaker button.

### Consideration

- The user must be assigned with VMIB Access.

- When the user assigns a Call Screen button, Call Screening is inactive and the LED is Off.
- If the user selects to answer a call by pressing the Call Screen button, the voice message is not saved.
- Call Screening does not operate while the keyset is busy or in DND.

#### Reference

- A. Incoming VMIB Mailbox
- B. Call Forward
- C. Station Call Forward Preset
- D. DND

#### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

Do-Not-Disturb (PGM 111, Flexible button 3)

Assigns DND to a range of Stations

DND Forward to Voice Mail (PGM 113, Flexible button 13)

Assigns calls to forward to the Voice Mail when the station is in DND



### 5.3 Flexible Buttons

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#### Description

The Flexible buttons of a DKT or DSS/DLS Console can be assigned for access to CO lines as well as one-button access to other stations and features of the system. The Flexible buttons can be defined by the user or through System Admin. Functions available for Flexible buttons include:

- CO line                      Accesses the assigned CO Line
- CO line Group            Accesses the highest numbered CO Line from a specified CO Line Group
- Loop                        Access incoming and outgoing calls on CO Lines that are not assigned to a another button of the DKT
- DSS/BLF                   Calls and displays status (idle, busy, ringing, DND, etc.) of the assigned Station
- Mailbox                    Calls or transfers calls to the Mailbox and displays message status for the Mailbox
- Numbering Plan           Dial codes shown below for one-button feature access,
  - Paging codes including Meet-Me
  - Internal All Call Page
  - Meet Me Page
  - External Page Zone
  - All Call Page (Int & Ext)
  - SMDR Account Code Enter
  - Call Forward, must include type and destination
  - Cancel DND/FWD/Pre-MSG
  - ACD Reroute
  - Alarm Reset
  - Group Call Pickup
  - Night Answer
  - Call Park
  - Direct Call Pickup
  - CO Line Group, if allowed in PGM 112, Flexible button 6
  - CO Line, if allowed in PGM 112, Flexible button 6
  - Access CO Line In the 1<sup>st</sup> CO Line
  - Attendant Call
  - Door Open Code
  - Conference Room
- Speed Dial                Accesses a memory dialing feature (System or Station Speed Dial, Saved Number Redial, Last number Redial)
- Station Group            Call a Station/Hunt Group number
- Telephone Number       Dials an outside Telephone Number, for LDP-6230D/6212D only.
- Feature button            Feature buttons including Call Coverage, Conference, Call Back, DND/FOR, Flash, Mute, On/Off, or Redial

#### Operation

To assign a Feature button at the DKT,

1. Press the [TRANS/PGM] button.
2. Press the Flexible button to be assigned.
3. Press the [TRANS/PGM] button.
4. Dial the Feature Code for the desired Feature button.

Feature Button	Feature Code
Call Coverage	46 + Station
Mailbox	77 + Mailbox
ACD On/Off Duty	87 & group number
ACD Group, Supv.	8* & group number
Conference	91
Call Back	92
DND/Forward	93
Flash	94
Mute	95
MON	96
Redial	97

5. Press the [HOLD/SAVE] button to save changes.

To assign a CO line or DSS/BLF button,

1. Press the [TRANS/PGM] button.
2. Press the Flexible Button to be assigned.
3. Enter the CO line, CO line Group or station number.
4. Press the [HOLD/SAVE] button.

To assign Telephone Number

1. Press the [TRANS/PGM] button.
2. Press the Flexible Button to be assigned.
3. Press the [TEL NUM] soft button.
4. Enter the desired telephone number.
5. Press the [HOLD/SAVE] button.

To assign a code from the System Numbering Plan

1. Press the [TRANS/PGM] button.
2. Press the Flexible Button to be assigned.
3. Dial the desired Numbering Plan code.
4. Press the [HOLD/SAVE] button.

Consideration

- The Station must be allowed the ability to assign a CO line or CO line Group to a Flexible button. If not allowed CO PGM, the Station can only program features to buttons not assigned as a CO line or CO line Group.

Reference

Flexible buttons have broad references, for additional information, reference the feature assigned to the Flexible button.

*Admin Programming*

Flexible Button Assignment (PGM 115)

Assigns functions to Flexible buttons

CO Programming (PGM 112, Flexible button 6)

Permits a user to assign CO lines and CO line Groups to the DKT

## 5.4 Headset

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### Description

An industry standard headset can be connected to a DKT instead of the handset. The station is programmed for headset operation in place of speakerphone operation, which allows the On/Off button to control the headset state, On or Off-hook. Headset operation and the device to receive ring (headset or speaker) can be assigned through System Admin or through Station User Programming.

### Operation

To set Speakerphone or Headset operation,

1. Press the [TRANS/PGM] button.
2. Dial code '7 5'.
3. The Station Answer mode should be displayed on the LCD.
4. Dial '1' (Speaker mode) or '0' (Headset mode).

To assign device to ring,

1. Press the [TRANS/PGM] button.
2. Dial code '7 6'.
3. The Station Answer mode should be displayed on the LCD.
4. Dial
  - '1' for Speaker ring only
  - '2' for Headset only
  - '3' for Both

### Consideration

- Page announcements are always received over the DKT speaker and not the Headset.

### Reference

- A. Station User Programming
- B. Intercom Signal Mode
- C. Paging

### Admin Programming

Headset Ring Mode (PGM 111, Flexible button 10)

Defines the device (headset or speaker) to receive ring at the station

Speakerphone/Headset (PGM 111, Flexible button 11)

Assigns Speakerphone or Headset operation

## 5.5 Hot Desk

---

### Description

A DKT may be assigned as Hot Desk phone allowing users (Agents) to login with the ipLDK-60. When an Agent logs in, the Hot Desk will become active and will take on the attributes defined for the Agent's station number. When the Agent logs off, the Hot Desk station becomes inactive and the Agent's calls are forward to the user-entered destination. A different Agent may then login through the inactive Hot Desk station.

### Operation

To login at a Hot Desk,

1. Lift the handset or press the On/Off button.
2. Dial the Agent's authorization code followed by '#'.
3. The Hot Desk becomes active with the Agent's station configuration (station number, COS, ring assign, etc.) and receives incoming calls.

To logout a User,

1. Press the [TRANS/PGM]
2. Dial '\* \*', or press the programmed Agent Logout button.
3. Use the Up/Down buttons to select a Call Forward type.
  - Station
  - Off-net speed
  - Mobile extension
  - VMIB Voice Mail
  - VM group
4. Press the [HOLD/SAVE] button, the system saves the Agent's station configuration and the Hot Desk returns to inactive.

### Consideration

- An inactive Hot Desk station displays "DUMMY STATION (xxx)", where xxx is the physical station number of the Hot Desk station.
- A Hot Desk station only allows login operation; all other operations are not allowed.
- A user may only logout from the Agent's active Hot Desk. Attempting to logout using a different Hot Desk will return error tone.
- The total number of users is restricted by the station capacity of the system. The total number of Hot-desk agents = 47 –number of installed stations.
- When logged off, the Agent's active database, including the following items, are saved.
  - Station Number
  - Station Attributes, PGM 111~124
  - CO Routing, Ring assignments, DID routing, etc.
  - Voice Mail
  - Station Group assignment
- An active (logged in) Agent can login to another inactive Hot Desk. However, this will log the Agent off the previously active Hot Desk.

- The Hot Desk station can be programmed to log-out an active Agent automatically if no action has been taken by the Agent for the Auto Log-out timer.
- The Flexible buttons of the Hot Desk station follow the assignments for the logged in agent.
- Only DKTs with more than 12 buttons can be used as Hot Desk stations.
- The modem associated with the station cannot log in (PGM 170).
- If the system is reset, all active Agents are logged out and all Hot Desk stations return to the inactive state.
- If the Hot Desk Agent station number is an ACD group member, logging in to a Hot Desk also logs the Agent in the ACD group.

### Reference

- A. Station User Programming
- B. Station Groups
- C. Do Not Disturb (DND)
- D. VMIB Voice Mail
- E. External Voice Mail

### Admin Programming

Station Id (PGM 110, Flexible button 1)

Assigns the type of station

Dummy Station (PGM 112, Flexible button 23)

Assigns Hot Desk stations

Number of Agents (PGM 250, Flexible button 1)

Assigns the number of Agents that can use the Hot Desk

Assigned Station Number of Agents (PGM 250, Flexible button 2)

Displays the station numbers associated with Agents

Agent Auto Logout Timer (PGM 250, Flexible button 3)

Sets the Agent no-action log-out time

## **5.6 Multi-lingual Display**

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### Description

The LCD language is normally displayed in the language selected for the system. If desired, the user may select to use English as the display language.

### Operation

To Select an LCD language

1. Press [TRANS/PGM
2. Dial 7 for Supplementary User Programs
3. Dial 1 to change the LCD Language

### Consideration

- If the system is initialized, the LCD language will return to the system language.

### Admin Programming

LCD Language (PGM 169, Flexible button 3)

Establishes the language displayed in LCDs system-wide

## 5.7 Mute

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### Description

During a conversation, activating Mute will disable the handset microphone or the speakerphone for privacy while continuing to listen to the other party on the phone through the handset or speaker. The DKT will remain muted until the call is terminated or Mute is deactivated.

### Operation

To mute the transmission audio while on a call,

1. Press the [MUTE] softkey.
2. The DKT is muted and the [MUTE] softkey changes to [SPEAK]. The connected party will not be able to hear audio from the muted station.

To restore transmission,

1. Press the [SPEAK] softkey.
2. Mute is terminated and the connected party can hear the station user.

### Consideration

- When changing from the speakerphone to the handset, mute is released.
- When pressing another DSS button, the mute state will not be changed.
- A Flexible button can be assigned to activate/deactivate Mute.

### Reference

- A. Flexible buttons
- B. Station User Programming



## **5.8 One-Time DND**

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### Description

One-Time DND allows a DKT user to activate DND during a call for the duration of the call. One-Time DND can be activated at any time while during a call, even while receiving a second call. In this case, the phone will stop muted ringing and the call receives DND treatment. Intercom callers receive DND tone and outside callers route to the Attendant, or all calls forward to the user's Voice Mailbox.

When the station returns to idle, One-Time DND is terminated.

### Operation

To activate One-time DND from a DKTU,

1. While on an active Intercom call or connected to a CO line, press the [DND/FOR] button.
2. The station enters DND and muted ring for any second call is terminated.
3. The station must be allowed DND to activate One-Time DND.

### Reference

- A. Do-Not-Disturb (DND)

### Consideration

- The user must have VMIB Access enabled and record a greeting for calls to Forward to the Voice Mail, otherwise the call is sent to the Attendant.

### Admin Programming

Do-Not-Disturb (PGM 111, Flexible button 3)

Enables DND for a range of Stations

DND Forward to Voice Mail (PGM 113, Flexible button 19)

Forwards calls to Voice Mail when the station activates DND

## **5.9 Station Relocation**

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### Description

A station can be moved from one location (station port) to another without loss of any configuration information using Station Relocation. The user activates Station Relocation, which stores the configuration in a temporary buffer. The phone is then moved to the new location and the configuration information copied to the new station port.

### Operation

To store the station configuration to the Station Relocation buffer,

1. Press the [TRANS/PGM] button
2. Dial '\* 1', the Station Relocation Backup code.
3. Unplug the station.

To retrieve stored station attributes,

1. Plug the phone in at the new location.
2. Press the [TRANS/PGM] button
3. Dial '\* 2' the Station Relocation Retrieve code.
4. The Station Attributes are copied to the current station location.

### Consideration

- All information for the port of the destination station will be retained so that it may be copied or relocated to another port.
- If a different station type (12-button versus 24-button) retrieves the Relocation buffer, Flexible button assignments may not be correct and may require reconfiguration.
- DKTs must be relocated to another hybrid or digital port; a DKT cannot be relocated to an SLT port.

### Reference

- A. Station User Programming

## 5.10 User Station Programming

### Description

The ipLDK incorporates a multi-level Station menu to allow users to view and change various aspects and features of their station. The full menu is available to users of a DKT with LCD. A user of a DKT without display and SLT users can access a limited number of the menu items as shown in the chart below. In addition, the Attendant has access to additional menu items as discussed in the Attendant Programming (refer to Ref. A). To use User Station Programming, the user dials a series of digits corresponding to the desired menu items and, if needed, any input as shown in the chart.

Main Menu	Sub Menu	Entry	Remark
[1] RING	[1] TYPE	Select Ring Type 1 to 4, ring heard over speaker as Type is dialed Dial 1,2,3 or 4	DKT
	[2] ANSWER MODE	Select Intercom Signal Mode Dial 1=HF, 2= Tone or 3=Privacy	DKT
	[3] SMS MSG DISPLAY	Show SMS Messages, scroll to display messages	DKT w/ LCD
	[4] ENBLOCK MODE	Enable/disable Enblock dialing, dial all digits then send or send each digit when dialed Dial 1= On or 0=Off	DKT
	[5] SMS/NOTICE DISPLAY	Not used	DKT w/ LCD
	[6] SCROLL SPEED	Not used	
	[7] EAR-MIC HEADSET	Enable/Disable operation of an external mic & headset Dial 1=On, 0=Off	DKT
	[8] ICM BELL	Dial 2-digit ring tone, 01~15, tone heard.	DKT
	[9] CO BELL	Dial 2-digit ring tone, 01~15, tone heard.	DKT
	[#] PTT GROUP	Not available for ipLDK-60	
[2] COS	[1] COS DOWN	Changes station to COS 7, intercom calls only	All
	[2] COS RESTORE	Dial Authorization code to return to normal COS	All
	[3] WALKING COS	Dial Authorization code to use temporary COS	All
	[4] COS CHANGE	Not used	All
	[5] SMS INBOX	Retrieve SMS messages	DKT w/LCD
	[6] Delete All SMS Message	Deletes all SMS messages	DKT w/LCD
[3] AUTHORIZE/ MOBILE - EXT	[1] AUTH. REGISTER	Dial desired Station Authorization code	All
	[2] AUTH. CHANGE	Dial the present Station Authorization Code	All
	[3] REG MOBILE-EXT	Dial user's Mobile phone number	Configured DKT stations
	[4] ACTIVE MOBILE-EXT	Enable/disable Mobile operation	Configured DKT stations
	[5] REG MOBILE-EXT CLI	Dial CLI for Mobile extension.	Configured DKT stations
	[6] ACTIVE MOBILE HUNT	Enable/disable Mobile as Hunt member	Configured DKT stations

Main Menu	Sub Menu	Entry	Remark
	[7] MW TO MOBILE-EXT	Enable/disable Message wait call to user's registered Mobile phone Dial 1=On or 0=Off	Configured DKT stations
[4] TIME	[1] SET WAKE UP TIME	Set the Alarm Clock Dial 4 digit alarm time, dial '#' for repeating alarm	All
	[2] WAKE UP DISABLE	Disable Alarm Clock	All
	[3] ACTIVE CONF ROOM	Open a Conference Room Dial room number and password	All
	[4] DEACTIVE CONF ROOM	Close a Conference Room Dial room number and password	All
	[5] CALL COVER ATTRIBUTE		
	[5] MESSAGE	[1] SET PRESELECTED MSG	Dial the message (01 ~ 10 and any supplementary data (time or date)
[2] SET CUSTOM MSG		Dial each character of Message, 2-digits for each, max 24 characters	All
[6] ANNOUNCE	[1] RECORD USER GREETING	Record your VMIB greeting Dial '#' and record	All
	[2] LISTEN TIME / DATE	Hear the ipLDK system time and date	All
	[3] LISTEN STA NUMBER	Hear the intercom number of the Station	All
	[4] LISTEN STA STATUS	Hear the status of the Station	All
	[5] RECORD PAGE MSG	Record a Page Message – dial '#' and record	All
	[6] ERASE USER GREETING	Delete the User Greeting	All
	[7] ERASE PAGE MSG	Delete the recorded Page Message	All
	Outbound Notify Menu	Menu for Out call notification of new VM	
	Email Notify Menu	Menu for E-Mail notification of new VM	
	Record User Name	Record the Company Directory User Name	
	Erase User Name	Delete the recorded User Name	
	[7] SUPPLEMENTARY	[1] LCD DISPLAY LANGUAGE	Changes LCD Language from System to English
[2] MBU VERSION DISPLAY		Display the System S/W version	DKT w/LCD
[3] BGM		Dial music source (00 - 08)	DKT
[4] REGISTER STA NAME		Enter the Station LCD Display name, 2-digits for each character	All
[5] SPK/HEADSET		Select On/Off button mode - dial 1 for Speakerphone operation or 0 for headset use	DKT
[6] HEADSET RING MODE		Select ringer Dial 1 for Speaker, 2 for headset or 3 for both	DKT
[7] WTU STA NUM RCVR		Not used	
[8] SERIAL NUMBER		Displays the serial number of the DKT	DKT w/LCD
[9] PC-PHONE LOCK KEY		Displays lock key for PC Phones	DKT w/LCD
[*] USB ALWAYS CALL REC		Not used	
[0] ATTENDANT		Refer to Attendant Services, section 13	Att'd only
[*] SYSTEM	[1] RELOCATION OUT	Station database is stored in the temporary buffer	All

Main Menu	Sub Menu	Entry	Remark
	[2] RELOCATION IN	Data in temporary buffer is written to new station port.	All
	[3] REGISTER BLUETOOTH		DKT w/BTM
	[4] BLUETOOTH USAGE		DKT w/BTM
	[0] HOT DESK LOG IN		DKT w/LCD
	[*] HOT DESK LOG OUT	Logs off of the Hot desk station	DKT w/LCD
	[#] ENTER ADMIN	Access to ipLDK system Admin	Admin only

Operation

To change a User Station Program item,

1. Press the [TRANS/PGM] button or dial '5 6 3'.
2. Confirmation tone is received.
3. Dial the Main and sub-menu item desired.
4. Follow the entry instructions in the chart above.
5. Press the [HOLD/SAVE] button or, for an SLT, flash.
6. Confirmation tone is received and the new value stored

Consideration

- After a menu is programmed, the previous menu list is displayed on the LCD.
- Pressing a Flexible button after the [TRANS/PGM] button activates Flexible button programming (refer to Ref. B).
- Pressing the [TRANS/PGM] button moves to the top of the Main menu. Pressing the [Redial] button, if assigned, moves to the previous menu. Refer to Ref. B to assign a Redial button.
- Accessing a menu item that is not available to the phone will return error tone. Hang-up or press the On/Off button to return to idle.

Reference

- A. Attendant Programming
- B. Flexible buttons

---

## 6 STATION & HUNT GROUPS

---

### 6.1 Hunt Groups

---

#### Description

Calls sent to a Hunt Group are routed to stations in the group based on the type of group and characteristics configured for the group. Five (5) types of groups can be defined:

- **Terminal Hunt Group**  
Calls to a station in the group or a pilot number route to the first available station starting with the called station and proceeding to the last station listed for the group. If all stations are busy, the call can be routed to an Alternate Destination or queue to the last station in the group. The call remains in queue until answered, the caller disconnects or the Overflow timer expires. If the timer expires, the call is routed to the Overflow destination.
- **Circular Hunt Group**  
In Circular Hunt, calls to a station in the group or a pilot number route to the first available station starting with the called station and proceed through the list of all stations in the group. If all stations are busy, the call can be sent to an Alternate Destination or queue at the last station in the circle. The call remains in queue until answered, the caller disconnects or the Overflow timer expires. If the timer expires, the call is routed to the Overflow destination.
- **ACD Hunt Group**  
Calls are sent to the group by dialing the pilot number (UCD Hunt Group number) or assigning CO lines to directly ring to the group. Calls are directed to the station in the group that has been idle for the longest time. If all stations are busy, the call may be routed to the Alternate Destination or may continue to wait (queue) for an available station. The call remains in queue until answered, the caller disconnects or the Overflow timer expires. If the timer expires, the call is routed to the Overflow destination. An ACD Supervisor can be assigned to manage and monitor the group and ACD reports provide group and individual call handling characteristics.
- **Ring Group**  
All the stations in the group receive ring simultaneously for a call to the Ring Group number. The call continues to ring all members until answered, disconnected by the caller or expiration of the Overflow time. The call is routed to the Overflow Destination should the timer expire.
- **Ext. VM Group**  
An External Voice Mail group can be configured employing SLT ports of the system connected to an external VM processor. Calls to the group route using either Terminal

or Circular hunting. The call remains queued until answered, the caller disconnects or the Overflow timer expires. If the timer expires, the call is routed to the Overflow Destination. When ipLDK-60 sends a call to the external VM processor, in-band DTMF signals are provided to identify actions required of the VM as defined in the VM Dialing Table.

The system supports up to ten (10) Station Hunt Groups and each group may have up to 26 member Stations assigned. A station can be a member of multiple groups of the same type. Each group can be assigned a name, up to 12 characters, that is displayed in the LCD of DKTs receiving a call for the Hunt Group.

Terminal, Circular and UCD Hunt and Ring Groups can be configured to play VMIB announcements to the caller. The 1<sup>st</sup> Announcement is played after the 1st Announcement timer expires. The 1<sup>st</sup> Announcement can be configured to play immediately as a guaranteed announcement, meaning it is played in its entirety before the hunt process initiates and the Overflow timer starts. A 2<sup>nd</sup> Announcement can be configured to play when the 2<sup>nd</sup> Announcement timer expires after the 1<sup>st</sup> Announcement. In addition, the 2<sup>nd</sup> Announcement can be configured to repeat at intervals of the 2<sup>nd</sup> Announcement timer until the call is answered or disconnected.

When a station completes a Hunt Group (Terminal, Circular or ACD) the station is held busy to Hunt Group calls for the Wrap-up timer allowing the user to complete paperwork, etc. After the expiration of the timer, the station is returned to idle for Hunt Group calls. In addition, Hunt Group stations may be assigned a Hunt Group DND button. Hunt Group DND allows the station to be Off-duty for Hunt Group calls but available for other call handling.

### Operation

To assign a Hunt Group DND button at a DKT,

1. Press the [TRANS/PGM] button.
2. Press the desired Flexible button.
3. Dial '8 7' and the Hunt Group number.
4. Press the [Hold/Save] button.

To activate Hunt Group DND,

1. Press the Hunt Group DND button.
2. The Station is Off-duty for Hunt Group calls but operates normally for other calls.

### Consideration

- The Overflow Destination may be a station, Group, VMIB Announcement, or System Speed dial. If no destination is assigned, the call is dropped when the timer expires.
- The Announcement Timer is ignored if no VMIB Announcement number is assigned.
- A station in DND or Call Forward will not receive Group calls.
- If all stations in a Hunt group are in DND, Call Forward, or Hunt Group DND, new and queued calls are routed to the Overflow Destination or for UCD Groups, the Alternate Destination.
- Transferred calls to a Station Hunt Group are not subject to recall.

- Only the 1st announcement may be used as a guaranteed announcement by assigning 0 seconds to the first announcement timer.
- While queued to a Hunt Group the call will be provided MOH.
- For ACD, Circular and Terminal Hunt Groups, if the number of queued calls exceeds the Queued Call limit, new incoming CO line calls are disconnected.
- The ipLDK-60 communicates with an external VM using in-band DTMF.

### Reference

- A. VMIB Announcements
- B. Music on Hold
- C. DND
- D. Call Forward
- E. Call Pick-Up
- F. Transfer
- G. ACD Supervisor
- H. ACD Statistic Reports
- I. External Voice Mail

### Admin Programming

Hunt Group Assignment (PGM 190)

Assign Hunt Group and member stations

Hunt Group Attribute (PGM 191)

Assigns attributes for each Hunt Group



### 6.1.1 ACD (Automatic Call Distribution) Group & Supervisor

---

#### Description

A station can be assigned as the ACD Group Supervisor. A station can be the ACD Supervisor for one or multiple ACD groups. The ACD Supervisor receives status information for the group indicating queued calls and time. When a call is queued to a group for longer than a queue time or when a predefined number of calls are queued, the supervisor's LCD will indicate the number of calls in queue, and the queued time for the longest queue.

The ACD Supervisor can control and can change overflow destination and timing as well as the On/Off duty status of ACD group members. In addition, he ACD Supervisor can monitor an agent's call and, if desired, the ACD Supervisor can record an agent's call to the VMIB using Two-way record (refer to Ref. C).

#### Operation

##### ACD Group Agents

ACD Group Agents are logged into each ACD group as a Group Member in the system database. Agents use the Agent Duty code or a Flexible button assigned as a UCD-DND button to go 'On' and 'Off' duty for a specified ACD Group.

To toggle Agent On/Off Duty,

1. Press the [TRANS/PGM] button.
2. Dial '8 7', the Agent Duty (UCD-DND) code.
3. Dial the ACD Group Number.
4. Press the [HOLD/SAVE] button.

Once 'On-duty', the LCD display of the DKT will display the queued call count for the ACD Group. Also, if the station has a DSS button for the ACD Group, the LED will flash to indicate the number of calls queued.

- Off                    0 calls in queue
- 60 ipm                1 to 3 calls in queue
- 120ipm               4 to 6 calls in queue
- 240 ipm               7 or more calls in queue

##### Supervisor

ACD Group Supervisors are logged into an ACD Group as a Group Member and separately assigned as the ACD Group Supervisor. The ACD Supervisor goes 'On' and 'Off' duty as with any ACD Group Agent. In addition, the ACD Group Supervisor must have a Flexible button assigned as an [ACD Group] button for each supervised ACD Group.

An ACD Supervisor can monitor calls of ACD Agents in the supervised ACD Group. The Supervisor enters the conversation in a 'Listen-only' mode (Mic is muted) but can join the conference with the Mute button. A warning tone can be configured to 'beep' when the ACD Supervisor monitors a call.

To monitor an agent's conversation at ACD supervisor,

1. Call the busy agent and receive busy tone.
2. Press [ACD Status] Flexible button.
3. The supervisor can monitor the agent, but will not send audio to the agent.

To Record an agent's conversation at ACD supervisor,

1. Call the busy agent and receive busy tone.
2. Press [ACD Status] Flexible button.
3. The supervisor can monitor the agent, but will not send audio to the agent.
4. Press [Two-Way Record] button.
5. To terminate recording, press the [Two-way Record] button or hang-up. Recording is stored in the ACD Supervisor station VMIB Voice mailbox.

In addition, the Supervisor Menu allows the ACD Supervisor to view group status, view and modify queued call routing, view and modify the status of Agents, and request output of ACD Reports (refer to Ref. A).

To view ACD group Status,

1. Press the idle [ACD Status] button.
2. The Supervisor Menu displays:

```

[1] ACD STATUS
[2] ACD DATABASE
[3] ACD DUTY
[#] ACD PRINT
    
```

3. Dial '1', the ACD Group status displays as below, use the [VOL UP]/[VOL DWN] button to scroll.
  - Total Calls
  - Average Call Time
  - Average Ring Time
  - Busy Count and Time
  - Number of calls in queue,
  - Average and Longest Queued Time
  - Unanswered Call Count

To view and modify queued call re-routing or Wrap-up Time,

1. Press the idle [ACD Status] button.
2. The Supervisor Menu displays.
3. Dial '2', the ACD database menu displays.

```

[1] OVRFL DESTINATION
[2] OVERFLOW TIME
[3] WRAP_UP TIME
    
```

4. Select the database item, scrolling with the [VOL UP]/[VOL DWN] button.
5. Enter new data.
6. Press the [HOLD/SAVE] button.

To view Agent status or modify status,

1. Press the idle [ACD Status] button.
2. The Supervisor Menu displays.
3. Dial '3, for the ACD Agent Duty menu.

```

[1] DUTY STATUS
[2] DUTY ON/OFF
[#] DUTY PRINT
    
```

To view Agent status from the ACD Agent Duty menu,

1. Dial Agent DUTY STATUS code, '1'.

```

AGENT XXXX
TOTAL CALLS
AVE CALL TIME
AVE RING TIME
UNANSWERED CALLS
    
```

2. Use the '\*' key or '#' key for a next/previous ACD Agent.
3. Use the [VOL UP] or [VOL DOWN] to scroll to information for the next Agent
4. Press the [HOLD/SAVE] button to return to the Supervisor menu.

To control Agent duty status from the ACD Agent Duty Menu,

1. Dial the Agent DUTY ON/OFF code, '2'.
2. Use the '\*' key for the next ACD Agent.
3. Dial 0 for Off Duty or, 1 for On Duty.

To output Agent Statistics Report from the ACD Agent Duty menu,

1. Dial the Agent DUTY PRINT code '#'.
2. Press the '\*' key or '#' key for a next ACD Agent.
3. Press the [CLEAR] softkey to clear the database after the print, which eliminates overlap of future outputs.
4. The system outputs the Agent report (refer to Ref. A).

To output the ACD Statistics report from the Supervisor menu,

1. Dial the ACD Group Statistics PRINT code, '#'
2. Press the [CLEAR] softkey to clear the database after the print, which eliminates overlap of future outputs.
3. The system outputs the Agent report (refer to Ref. A).

To re-route queued calls with answer,

1. Press the flashing [ACD Status] button, the oldest call in queue is answered.
2. Transfer the call as normal.

Consideration

- ACD Agents can login and logout using the Hot Desk feature.

Reference

- A. ACD Statistics Reports
- B. Station Hunt Groups
- C. Two-way Record

#### D. Flexible Buttons

##### Admin Programming

Station Hunt Groups (PGM 190)

Assigns group type and member stations

ACD Group Attributes (PGM 191)

Assigns characteristics of the ACD Group

### 6.1.2 ACD (Automatic Call Distribution) Statistic Reports

---

The ipLDK-60 can output ACD Statistics periodically or on-demand from the ACD Group Supervisor. Two reports are available, the ACD Group Report and the ACD Agent Report. When periodic reports are configured, on-demand reports are not supported. The reports are output over the Printer port assigned to receive Traffic Reports.

The on-demand ACD Group Reports include the following statistics for the group:

- Total calls
- Number of unanswered calls
- Average and the longest queued calls
- Number and total time when all agents are busy
- Average ringing time before answer
- Average service time after answer

The format of the report is as shown in the following:

```
===== ACD GROUP INFORMATION (*620 GRP) =====
Date: 05/15/02 Time: 05:40:07
- Total Calls: 2
- Unanswered Calls: 2
- All STA Busy Count: 2
- Average Ringing Time: 00Min 00Sec
- Average Service Time: 00Min 00Sec
- All STA Busy Time: 01Min 17Sec
- Calls in Queue, Now: 0
```

The on-demand ACD Agent Report includes the following statistics for the selected Agent:

- Total calls
- Number of unanswered calls
- Average ringing time before answer
- Average service time after answer

The format of the report is as shown in the following:

```
=====
ACD GROUP INFORMATION (*620)          Date: 05/15/02 Time: 05:42:24
AGENT_NO | TOTAL_CALL | UNANSWERED_CALL | AVE_RING_TIME | AVE_SVC_TIME
=====
      1001 |           0 |                0 | 00Min 00Sec | 00Min 00Sec
=====
```

For the periodic reports, ACD Print should be enabled and the ACD Timer and timer units should be configured to provide the desired interval between reports. The Clear database parameter should be enabled to eliminate overlap in subsequent reports.

The periodic ACD Agent report follows the format described in the following chart.

~	1	=	2	=	3	=	4	=	5	cr	lf	
---	---	---	---	---	---	---	---	---	---	----	----	--

Field(s)	Meaning
~(tilt)	Start of ACD statistics output
=(equal)	Delimiter between data
1	Agent number
2	Total call counter
3	Unanswered call counter
4	Average ringing time in seconds
5	Average service time in seconds
lf	Line Feed (0x0A)
cr	Carriage Return (0x0D)

The periodic ACD Group Statistics Report follows the format described in the following chart.

```
~ 1 = 2 = 3 = 4 = 5 = 6 = 7 = 8 = 9 = 0 cr lf
```

Field(s)	Meaning
~(tilt)	Start of ACD statistics output
=(equal)	Delimiter between data
1	ACD Group number
2	Total call counter
3	Unanswered call counter
4	All busy counter
5	Average ringing time in seconds
6	Average call service time in seconds
7	Total busy time in seconds
8	Number of current queued calls
9	Longest queued time
0	Average queued time
lf	Line Feed (0x0A)
cr	Carriage Return (0x0D)

Consideration

- When periodic reports are configured, on-demand reporting is not supported.

Reference

- A. Traffic Analysis
- B. ACD Group & Supervisor
- C. SMDR

Admin Programming

Station Hunt Groups (PGM 190)

Assigns group type and member stations

ACD Group Attributes (PGM 191)

Assigns characteristics of the ACD Group

ACD Print Enable (PGM 161, Flexible button 9)

Enables automatic printing of ACD statistics

ACD Print Timer (PGM 161, Flexible button 10)

Defines the interval between ACD reports

ACD Clear Database after Print (PGM 161, Flexible button 11)

Enables deleting the database after a print-out

RS232 Port Settings (PGM 174)

Defines characteristics of the system RS-232C port

Print Port Selection (PGM 175)

Assigns the printer port to receive the reports

### 6.1.3 External Voice Mail

---

#### Description

The system provides support for an adjunct Voice Mail system via connection to SLT ports. When a call arrives for the External VM Group, the system will search the group for an idle port and deliver the call employing terminal or circular hunt based on the External VM Group configuration. Signaling information between the system and the external VM system is achieved via in-band DTMF signaling. The system will send DTMF signals informing the external VM of the characteristics of the call. DTMF digit strings are assigned in the VM Dialing Table to various functions allowing the external VM to respond appropriately to the call.

#### Consideration

#### Reference

- A. Hunt Groups

#### Admin Programming

Station Hunt Groups (PGM 190)

Assigns group type and member stations

External VM Group (PGM 191)

Assigns the characteristics of the external VM group

Voice Mail Dialing Table (PGM 234)

Defines the DTMF in-band signaling for the external VM system



## **6.2 Intercom Tenancy Group**

---

### Description

A station can be assigned to one of five Intercom Tenancy Groups. Each Intercom Tenancy Group can be assigned an Attendant for control of Day/Night service and to receive dial '0' calls. Each group is also allowed or denied calling to other groups and access to CO lines.

### Consideration

- When access to another Intercom Tenancy Group is denied, Call Pick-up, transfer, etc are not allowed between groups.
- Each station must have a unique station number regardless of Intercom Tenancy Group assignments.
- The Intercom Tenancy Group Attendant can be any station in the system, and is not required to be a member of the group.
- When the Attendant of an Intercom Tenancy Group sets the Day/Night/Weekend mode, it will affect only their assigned Intercom Tenancy Group.

### Reference

- A. Pick-Up
- B. Call Transfer
- C. Day/Night Service
- D. Attendant

### Admin Programming

Intercom Group Number (PGM 111, Flexible button 13)

Assigns the Intercom Tenancy Group for a range of stations

Intercom Tenancy Group (PGM 120)

Assigns Attendant and allowed access Tenancy groups

CO Line Attributes (PGM 141, Flexible button 10)

Assigns Tenancy groups allowed access to a range of CO lines

## 7 CONFERENCE SERVICES

---

### 7.1 Conference Room

---

#### Description

In addition to ad-hoc conferencing, users may establish a Conference Room. Other internal and external parties are invited to the conference and can join the conference without further action by the user who established the room. DID/DISA callers may access a Conference Room and CO line calls may be transferred to a Conference Room by an Attendant. A Conference Room can be password (5 digits) protected so that only parties that enter the password are allowed to join the Conference Room.

When entering or leaving a Conference Room, a Warning tone can be provided to active Conference Room participants. The entry and exit tone are distinct allowing participants to determine if a party has entered or left.

Once a Conference Room is activated, it remains active until the user or Attendant closes (deletes) the room. There are nine (9) Conference Rooms available and each can support 15 participants.

A Flexible button can be assigned to access a Conference Room and will display the status of the room, On = active, Off = closed, flashing = In use.

#### Operation

To activate a Conference Room,

1. Press the [TRANS/PGM] or dial '5 6 3'.
2. Dial '4 3'.
3. Dial the Conference Room number (1-9).
4. Dial 5-digit password (optional).
5. Press the [HOLD/SAVE] button to activate the room.

To deactivate a conference room,

1. Press the [TRANS/PGM] or dial '5 6 3'.
2. Dial '4 4'.
3. Dial Conference Room number (1-9).
4. Dial 5-digit password, if assigned.
5. Press the [HOLD/SAVE] button.

To join a conference room from a Station or DISA caller,

1. Dial the Conference Room (571 - 579).
2. Dial the password, if assigned.

To transfer a CO call to a Conference Room,

1. Press the [TRANS/PGM] button.
2. Dial the Conference Room number (571 - 579).
3. Dial the password, if assigned.
4. Hang-up to complete the transfer.

To view the Conference Room status from the Attendant Station,

1. Press the [TRANS/PGM] button at attendant.
1. Dial the Room Monitor code '0 4 7'.
2. Dial the Conference Room number (1-9).
3. The Attendant LCD displays the number of participants in the room.

To delete a Conference Room from the Attendant,

1. Press the [TRANS/PGM] button.
2. Dial the Room Delete code '0 4 8'.
3. Dial the Room number.
4. Press [HOLD/SAVE].

#### Consideration

- The maximum number of Conference Rooms is 9 and each Conference Room may have up to 15 participants.
- A Conference Room remains active until deleted by the user or Attendant.
- The Attendant does not require the password to delete a Conference Room.
- An analog CO line must provide disconnect supervision and the Open Loop Detect timer must be properly set for a call to be transferred to a Conference Room or for a DISA caller to join the room.
- To join a Conference Room directly from a DID line, the Conference Room must be defined as the destination in the DID/DISA Destination Table.
- The LED of a Conference Room button will display the status of the Conference Room:
  - On, Conference Room is activate with no participants
  - Off, Conference Room is deactivated.
  - 60 ipm Flash, 1 to 3 participants are in the room
  - 120 ipm Flash, 4 to 6 participants are in the room
  - 240 ipm Flash, 7 or more participants are in the room

#### Reference

- A. DISA
- B. Multi-party Conference
- C. Transfer
- D. Attendant

#### Admin Programming

Conference Warning Tone (PGM 160, Flexible button 11)

Enables warning tone when a participant enters or leaves a room

Open Loop Detect Timer (PGM142, Flexible button 13)

Sets the duration of open loop a CO line will provide for disconnect supervision

DID Flexible Table (PGM 231)

Assigns destinations for DID Type 2 conversion

DID/DISA Destination (PGM 167)

Assigns destination for DID/DISA call that encounter error, busy, DND, etc.

## **7.2 Multi Party Conference**

---

### Description

The system allows multiple internal and external parties to be connected on a call in a conference. A DKT station user can establish a conference with up to 14 other parties that may include any combination of internal and external parties. The user initiating the conference provides supervision for the conference, if the user disconnects, the conference is terminated. The user may, if allowed, temporarily leave a conference establishing an Unsupervised Conference (refer to Ref. A).

### Operation

To establish a Multi-party Conference from a DKT,

1. While on a call, press the [CONF] softkey; the existing call will be put on HOLD and the intercom dial tone will be heard.
2. Place a call to the desired party as normal and announce the conference.
3. When the call is answered, press the [CONF] softkey.
4. To add parties, repeat steps 2 and 3.
5. When all parties have been called, press the [CONF] softkey again; all parties will be connected to the call.

### Consideration

- The [CONF] button, if assigned, remains illuminated at the initiators phone for the duration of the conference.
- If while setting up a conference, system error tone is received, the initiator must press the [CONF] softkey to regain Intercom dial tone.
- A station that is busy, in DND or other non-idle state cannot be added to a conference.
- IP phones can be added to conference up to the number of VOIB channels assigned for IP Phones.

### Reference

- A. Unsupervised Conference

### **7.3 Paging Conference**

---

#### Description

During a page placed over a Conference Page Zone (zones 506 ~ 510) a second station can join the page and simultaneously transmit audio to the Conference Page Zone along with the original paging station.

#### Operation

To join the Page Conference from a DKT,

1. Lift the handset.
2. Press [CONF] softkey.

To join the Page Conference from an SLT,

1. Lift the handset.
2. Momentarily press the hook-switch, intercom dial tone is received.
3. Dial '5 8'.

#### Consideration

- Page Time-out is not applied to a Paging Conference.
- When the second party joins the Page Conference, Meet Me Page is disabled.
- When the Page originator disconnects from the Page, the Page Zone(s) is returned to idle and the second paging party is disconnected.
- Any station can access a Page Conference once initiated, even if the station is denied access to Paging.

#### Admin Programming

System Numbering Plan (PGM 109, Flexible button 5)

Assigns SLT code to join a Page Conference

Paging Warning Tone (PGM 161, Flexible button 4)

Enables warning tone before page announcements

Paging Access (PGM 111, Flexible button 8)

Enables Stations access to paging facilities

Conference Page Zone (PGM 119)

## **7.4 SLT 3-party Conference**

---

### Description

A single line telephone user can initiate a 3-way conference with any combination of CO lines and internal users.

### Operation

To set up a conference from a SLT,

1. Place or receive first call.
2. Momentarily press the hook-switch, the intercom dial tone is received and the active call is placed on exclusive hold.
3. Place the second call and announce the conference.
4. Momentarily press the hook-switch twice within 2 seconds to establish a conference.

### Consideration

- A station that is busy, in DND or other non-idle state cannot be added to a conference.
- When pressing the hook-switch with an active call places the call on exclusive hold and is subject to Hold recall.

### Reference

- A. Multi-party Conference
- B. Unsupervised Conference
- C. SLT Broker's Call
- D. Hold

## **7.5 SLT Broker's Call**

---

### Description

A single line telephone user can alternate between two calls maintaining private conversations with both parties. The parties may be either internal Stations or external CO line calls, and may be incoming or outgoing.

### Operation

To set up a Broker's Call from a SLT, perform the following Steps:

1. Place or receive first call.
2. Momentarily press the hook-switch, the intercom dial tone is received and the active call is placed on exclusive hold.
3. Place the second call.
4. Momentarily press the hook-switch to switch between calls.

### Consideration

- When pressing the hook-switch with an active call places the call on exclusive hold and is subject to Hold recall.

### Reference

- A. Unsupervised Conference
- B. SLT 3-party Conference
- C. Hold



## **7.6 Unsupervised Conference**

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### Description

A DKT Station user may establish a conference with external parties and exit the conference allowing the external parties to converse privately without supervision from the user. An Unsupervised Conference is also established when a DISA caller places a CO line call or a CO line call is sent (transfer or forward) to a CO line.

The system will disconnect an Unsupervised Conference if disconnect is detected with only two parties connected. In addition, because some analog CO lines do not provide a disconnect signal, when an Unsupervised Conference is initiated that involves an analog CO line, the ipLDK will activate the Unsupervised Conference timer. Fifteen seconds prior to expiration of the timer, a Disconnect Warning Tone is provided. At expiration, all parties in the Unsupervised Conference are disconnected.

If enabled, any party in an Unsupervised Conference can request the Unsupervised Conference timer be extended. The party enters the Timer Extend code and a digit 1 to 9 indicating the extension timer multiplier. The system will determine the extension time as the Unsupervised Timer multiplied by the dialed digit. For example, if the Unsupervised Conference timer is 5 minutes and the user dials the digit 4, the timer will extend to 20 minutes (4 multiplied by 5 minutes).

### Operation

To establish an Unsupervised Conference at the conference initiator,

1. During a conference, press the [CONF] softkey.
2. The [CONF] button, if assigned, flashes and the Unsupervised Conference is established.
3. To re-enter the conference, press the [CONF] softkey.

To extend the duration of an Unsupervised Conference,

1. During a conference, dial '#' '#'.  
2. Dial the timer multiplier (1 - 9).  
3. ipLDK extends the Unsupervised Conference timer.

### Consideration

- The Unsupervised Conference Timer applies to all CO line to CO line connections.
- An Unsupervised Conference will be terminated if the system receives a disconnect signal or the Unsupervised Conference Timer expires.
- An idle DTMF receiver is required to support extending the Unsupervised conference. The system has fifteen (15) DTMF receivers for shared use.
- An Unsupervised Conference is not subject to Transfer or Hold recall.
- If the system receives a disconnect signal and no internal parties remain in the conference, the Unsupervised Conference is terminated and all parties are disconnected. If an internal party is still connected when a disconnect signal is received, the connection with the remaining parties is maintained.

- The Unsupervised Conference timer will be reset if the internal party re-enters the conference.
- Call Progress Tone detection must be enabled for the ipLDK to detect busy and error tones when a CO line to CO line connection is attempted.

#### Reference

- A. DISA
- B. Multi-party Conference
- C. Transfer
- D. Hold

#### Admin Programming

System Numbering Plan (PGM 109, Flexible button 6)

Assigns the Unsupervised Conference Timer Extend Code

Unsupervised Conference Timer Extend Enable (PGM 160, Flexible button 18)

Enables extending an Unsupervised Conference

Unsupervised Conference Timer (PGM 182, Flexible button 6)

Sets the Timer for an unsupervised CO-to-CO connection.

Busy/Error CPT Detect (PGM 142, Flexible button 18)

Enables detection of call progress tones for an analog CO line

## 8 PAGING SERVICES

### 8.1 Internal, External Zone and All Call Paging

#### Description

A station, which is permitted to access Page facilities, can connect and transmit voice announcements to any or all of the systems Internal and External Page zones. A DKT Station can be assigned to receive announcements from any combination of zones or not to receive Page announcements.

A Page Warning tone, if assigned, will be provided to the Page Zone(s) prior to the audio connection. The user is allowed to continue the page for the Page Time-out timer after which the user is disconnected and the Page Zone(s) is returned to idle.

Page Zones and the access codes for paging are shown in the following table.

Page ZONE Code	ITEM
501-505	Internal Page Zones
543	Internal All Call Page
545	External Page Zone
549	All Call Page (Internal & External)

Flexible buttons of a DKT may be assigned to access a Page Zone.

#### Operation

To initiate a Page announcement,

1. Lift the handset.
2. Dial the desired Page Zone code.
3. After the Page Warning tone, if assigned, make the desired announcement.
4. Replace the handset to return to idle.

#### Consideration

- When External Page is required, appropriate external equipment should be attached to the External Page port.
- An External Control Contact may be assigned to activate when External Page is accessed to turn on an amplifier or connect voice to the amplifier.
- A station, which is in DND or busy, will not receive paging announcements.
- A station, which is not allowed to Page Access, cannot initiate a Page but may still receive announcements.
- The ipLDK-60 allows only a single internal Page at a time. Attempting to initiate an internal page including All Call Page while another internal or All Call Page is in progress will return error tone. An internal Page is possible at the same time as another External page except for All Call Paging.

- After the Page Timeout, the paging connection is released and busy tone is presented to the paging station.
- When trying to make a page without lifting handset, “LIFT HANDSET TO PAGE” is displayed on the LCD. This eliminates the potential feedback from the station speakerphone.
- If an intercom call is received at a station making a Page announcement, the caller will receive busy tone.
- If a CO line call is received at a station making a Page announcement, the Station receives muted ring.
- A CO Line activating Park and Page is answered using the Park Pick-up procedure and cannot be accessed via Meet-Me-Page.

### Reference

- A. DND
- B. External Control Contacts
- C. Meet-Me-Page
- D. Page Conference
- E. Park and Page

### Admin Programming

Paging Timeout Timer (PGM 181, Flexible button 10)

Sets the time-out for Paging

Page Access (PGM 111, Flexible button 8)

Enables a beep tone prior to page audio

Internal Page Zone (PGM 118)

Assigns stations to Internal Page zones

Page Warning Tone (PGM 161, Flexible button 4)

Enables a beep tone prior to page audio

External Control Contact (PGM 168)

Defines function of the External Control Contacts

## **8.2 Meet-Me-Page**

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Any station may respond to a “Meet Me” Page request over an Internal or External Page Zone. The user answers the page from any station and is connected to the paging party.

A Flexible button of a DKT may be assigned as a Meet Me Page answer button.

### Operation

To respond a Meet-me Page,

1. Lift the handset or press the On/Off button.
2. Dial '5 4 4' (Meet-me Page code)
3. An internal call is established with the paging party and the Page Zone returns to idle.

### Consideration

- The user must respond to the Meet Me request within the Paging Time-out.

### Reference

- A. Internal, External Zone and All Call Paging
- B. DISA

### Admin Programming

System Numbering Plan (PGM 106, Flexible button 4),

Assigns the code for response to Meet Me

Paging Timeout Timer (PGM 181, Flexible button 10)

Sets the time-out for Paging

Page Access (PGM 111, Flexible button 8)

Enables a beep tone prior to page audio

Internal Page Zone (PGM 118)

Assigns stations to Internal Page zones

### **8.3 Push-to-Talk (PTT) Page**

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#### Description

The LG-Ericsson WIT-300HE wireless LAN phone incorporates a Push-to-Talk (PTT) page facility. The button can be assigned to access an internal Page zone to send a PTT page to stations in the page zone.

#### Operation

To assign an internal page zone to the PTT at the WIT-300HE,

1. Press the [TRANS/PGM] button.
2. Press '1#'.
3. Dial the desired Page Zone, or '0' to delete the current assignment.
4. Press the [HOLD/SAVE] button.

To send a PTT Page at the WIT-300HE,

1. Press the PTT button.
2. After the Page warning tone, audio from the WIT-300 is sent to the Internal Zone.
3. Release the button to terminate the PTT Page.

#### Consideration

- Page Time-out is not applied to a Push-to-talk Page.
- Any Internal Page Zone can be assigned for Push-to-talk. Conference Page Zones cannot be assigned for Push-to-talk.
- To access PTT paging, the station must be permitted access to system paging
- A station, which is in DND or busy, will not receive paging announcements.

#### Reference

- A. Internal, External and All-Call Paging
- B. Paging Conference

#### Admin Programming

Page Access (PGM 111, Flexible button 8)

Enables stations access to paging features of the ipLDK

Internal Page Zone (PGM 118)

Assigns stations to Internal Page zones

Page Warning Tone (PGM 161, Flexible button 4)

Enables a beep tone prior to page audio

## **8.4 SOS Page**

---

### Description

The system allows multiple pre-recorded VMIB messages to be used as paging announcements. The user can then quickly send the pre-recorded announcement over the assigned Page Zone. This function is often used to provide an emergency or SOS page.

### Operation

To assign {VMIB SOS Page} at a flexible button,

1. Press the [TRANS/PGM] button.
2. Press the desired Flexible button.
3. Dial the desired Page Zone code,
4. Dial Message number (001-070).
5. Press the [HOLD/SAVE] button.

To activate VMIB SOS paging:

1. Press the assigned {VMIB SOS Page} Flexible button.

### Consideration

- SOS Page is only available to a DKT using a Flexible button assigned for VMIB Page Message.
- The VMIB Page Message for SOS Paging must be recorded by the Attendant before the SOS Page button for proper operation.
- An SOS Page is not restricted by Page Time-out; the entire VMIB Page Message will be heard over the paging facilities even if the Page Time-out timer expires.

## **8.5 VMIB Page Message**

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### Description

A user can record a message in the VMIB that will be played whenever the Station initiates a Page.

### Operation

To record a VMIB Page Message,

1. Press the [TRANS/PGM] button or dial 563.
2. Dial 6 5.
3. The record a message prompt is heard and, if a message has been recorded, it is played.
4. To start recording, dial '#' or, to delete the recording, press the [Speed] button.
5. After the beep tone, record the desired message.
6. Press the [HOLD/SAVE] button.

To activate VMIB Page Message,

1. Lift the handset.
2. Dial the desired Page Zone code.
3. After the Page Warning tone, if assigned, the VMIB Page Message is played over the Page Zone(s).
4. Hang-up the handset to return to idle.

To delete a VMIB Page Message

1. Press the [TRANS/PGM] button.
2. Dial 6 7, the recorded message is deleted.

### Consideration

- When trying to initiate a page without lifting handset, "LIFT HANDSET TO PAGE" is displayed on the LCD. This eliminates the potential feedback from the Station speakerphone.
- A VMIB Page Message is subject to Page Timeout.
- If no Page Message is recorded in the VMIB, the user is connected to the Paging facilities as in a normal Page.
- The station must be allowed VMIB Access to record a Page Message.

### Reference

- A. Internal, External and All-Call Paging

### Admin Programming

System Numbering Plan (PGM 106, Flexible button 4),

Assigns the code for response to Meet Me

Page Access (PGM 111, Flexible button 8)

Enables stations access to paging features of the ipLDK

Internal Page Zone (PGM 118)



Assigns stations to Internal Page zones

Page Warning Tone (PGM 161, Flexible button 4)

Enables a beep tone prior to page audio

Paging Timeout Timer (PGM 181, Flexible button 10)

Sets the time-out for Paging

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

## **8.6 Automatic Park and Page**

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### Description

Outside callers reaching a mailbox are allowed to page the user. The caller is placed in the highest available Park Orbit (Ref. A). The system sends a page to notify the user of the call and the Park Orbit where the call is Parked. The user connects to the call by accessing the Park Orbit. The caller should be instructed to dial the digit '8' to activate the Park and Page feature.

If the Park recall timer expires while the call is parked, the call will recall to the Attendant.

### Operation

To retrieve the Parked call

1. Lift the Handset or press the On/Off button.
2. Dial the announced Park location to retrieve the call.

To record a User Name Greeting,

1. Press the [TRANS/PGM] button or dial '563'.
2. Dial '6 \*' and follow the voice prompts to record a Name greeting
3. Press the [Hold/Save] button to store the Name greeting.

### Consideration

- The User Voice Mailbox greeting should instruct the caller to dial '8' to page the user.
- A User Name Greeting must be recorded.
- The Station must be configured for Park and Page.
- Only one call can use the Park and Page feature at a time, a second Park and Page call will receive the resource limited prompt and returned to the user's Voice Mailbox.

### Reference

- B. Call Park
- C. User Station Programming

### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Enables the Station Voice Mailbox

Park and Page (PGM 113, Flexible button 17)

Enables the station for Park and Page

Call Park Recall Timer

Sets the time a call will remain on Park Hold before recalling the station

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## 9 STATION PAIRS

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### 9.1 Executive/Secretary

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#### Description

Stations in the system can be assigned as an Executive/Secretary pair. By activating DND, the Executive also activates Unconditional Call Forward to the Secretary, which will forward Executive calls to the Secretary. In addition, Intercom and/or CO calls to the Executive can be assigned to forward to the Secretary regardless of the Executive's station status by assigning ICM Call to Secretary and CO Call to Secretary options. Also in the above case,, if the Secretary is in DND, Executive calls sent to the Secretary route back to the Executive if the "Call Exec If Sec in DND" option is enabled.

Each Executive can be assigned a "Grade" (01, highest ~12, lowest). Executives with the same or higher grade can call lower grade Executives overriding the ICM Call to Secretary assignment.

#### Operation

To activate Executive/Secretary Forward from the Executive's DKT,

1. Press the [DND/FOR] button.

#### Consideration

- An Executive may have multiple Secretaries and a Secretary may have multiple Executives. Each forms a separate Executive/Secretary pair.
- If the Secretary is busy when a call is received for the Executive, the caller will receive busy tone.
- The Secretary may override the DND (refer to Ref. A) status of the Executive to Camp-On (refer to Ref. B) and transfer calls to the Executive.
- A chain can be constructed by assigning the Secretary of one pair as an Executive of another. Although a chain may be constructed, a loop back is not allowed.
- If an Executive has multiple Secretaries, calls will automatically route to the Executive's first idle Secretary.
- The Executive may use Call Forward (refer to Ref. D) to send calls to stations other than the Secretary.
- When both Executive and Secretary are busy, camp-on, transfers and messages are sent to the last Secretary Station in the chain.

#### Reference

- A. DND
- B. Camp-On
- C. Call Transfer
- D. Call Forward

Admin Programming

Executive/Secretary Table (PGM 229, Flexible 1)

Assigns stations as Executive and Secretary

CO Call To SEC (PGM 229, Flexible button 2)

Enables all Executive CO calls to forward to the Secretary

Call Exec If Sec DND (PGM 229, Flexible button 3)

Assigns Executive calls to the Secretary in DND to route to the Executive

Executive Grade (PGM 229, Flexible button 4)

Assigns the grade to an Executive

ICM Call To SEC (PGM 229, Flexible button 5)

Enables all Intercom calls to the Executive to forward to the Secretary

## **9.2 Linked-Pair Station**

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### Description

A station can be logically linked to a primary station so that the two stations function as a single station. When linked, the two stations effectively act as a single station with the station attributes of the primary station. The status of one station is reflected in the status of the other and features activated at one are active at the other. All internal or external calls to a linked pair station will ring both stations.

All features available to the primary station are available and controllable by the secondary station, one station may activate Call Forward and the other may cancel the forward. The displays of the linked stations will display the status of the linked pair. If one of the linked stations is busy, the LCD of the other station will display "IN USE AT LINK STA". When a linked station is busy, the other idle linked station will not receive ring for CO lines, transferred ring or intercom calls.

### Consideration

- A station can be linked with only one station. Any combination of DKTs and SLTs may be assigned as Linked pairs.
- A DSS Console, Door Phone Box or Port Blocked Station may not be assigned as a linked pair station.
- Linked pair stations are treated as having a single station number for all features including LCD displays, station programming, ADMIN access, ACD statistics, SMDR, etc.
- Intercom calls to the Linked stations always signal in the Tone ring mode and cannot be changed using the Forced Hands-free feature.
- The station attributes of the secondary station will follow attributes of the primary station, i.e. Day/Night COS, CO Warning Tone, CO Auto Hold, CO Call Drop, DID Call Waiting, Speed Access, Alarm, VMIB Access, DND, FWD, Paging, CO Access, CO Ring, etc.
- An Attendant station can be linked with another station but, the linked station cannot use attendant features (refer to Ref. D).
- Calls can be placed or transferred between the stations of a Linked pair using the primary station number.

### Reference

- A. Call Transfer
- B. DND
- C. Call Forward
- D. Attendant
- E. Intercom Signaling Mode
- F. Forced Hands-free Mode

### Admin Programming

Linked Station Pairs (PGM 179)

Assigns the Linked stations

### **9.3 Mobile Phone Support**

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#### Description

Mobile Phone Support allows a mobile phone to be registered to a station allowing the mobile phone to place and receive calls through the system. ISDN/T1 DID, VoIP CO line and station calls are sent to the user's ipLDK phone and the active registered mobile phone simultaneously. If the mobile phone is paired with a Hunt group station, Hunt group calls can be routed to the station and the active mobile phone.

The mobile phone user can access the facilities of ipLDK to place internal and external calls as well as transfer calls received from the ipLDK. To access the ipLDK to place calls or activate features, the mobile user calls the DID number of the user's ipLDK station. When the call is received, the system matches the CLI to the mobile phone and provides the mobile user with system dial tone.

When SMS is supported, the system can be configured to send an SMS message to notify the mobile user of new Voice Mail messages. The mobile user can access the Voice message through the Voice Mail remote access procedure (refer to Ref. H).

The mobile phone number and CLI may be registered by the user or in the system Admin. The user can activate Mobile Phone Support from the Mobile by calling the user's DID number or DKT station and can assign a Flexible button for one-button activation of Mobile Phone Support.

#### Operation

To register a mobile number from the DKT,

1. Press the [TRANS/PGM] button
2. Dial '3 3'.
3. Dial the mobile phone number that is being registered,
4. Press the [HOLD/SAVE] button.

To activate (deactivate) Mobile Phone Support,

1. Press the [TRANS/PGM] button
2. Dial '3 4'.
3. Dial 1 to activate, or 0 to deactivate.
4. Press the [HOLD/SAVE] button.

To register the mobile extension CLI number from the DKT.

1. Press the [TRANS/PGM]
2. Dial '3 5'.
3. Dial the mobile CLI number
4. Press the [HOLD/SAVE] button.

To use the mobile phone as a Hunt group member,

1. Press the [TRANS/PGM]

2. Dial '3 6'.
3. Dial '1' to activate, or '0' to deactivate
4. Press the [HOLD/SAVE] button.

To enable Voice Message Notification to the mobile,

1. Press the [TRANS/PGM]
2. Dial '3 7'.
3. Dial '1' to activate, or '0' to deactivate
4. Press the [HOLD/SAVE] button.

To Transfer from the mobile back to an ipLDK station,

1. Press the '\*'.
2. Dial the station number to receive the call.
3. Hang-up to complete the transfer

Or, if busy,

4. Press '#' to return to the caller.

To place a call from the mobile using the ipLDK facilities,

1. Dial the user's ipLDK DID number, the ipLDK will return dial tone.
2. Place call using dial codes.

#### Consideration

- If the extension is busy, forwarded, or in the DND state, calls will not route to the mobile extension.
- When the mobile is busy on an ipLDK call, station calls to the associated station will receive busy. In this case, a Message Wait can be left to the station, however Call Waiting is not available.
- When making an external call from a mobile extension through the ipLDK system, the system sends the CLI associated with the station and the mobile extension cannot use the transfer feature
- If the call is not answered in the DID/DISA No-answer time, the ipLDK will reroute the call based on the DID/DISA No Answer Destination or the call may be rerouted by the GSM network depending on which timer is shorter.
- This feature applies only to ISDN DID and IP CO lines, and station-to-station calls including transfers.
- Mobile extensions cannot activate or receive Station Message Wait/Callback features.
- While on an active ipLDK call, the mobile extension is only permitted to transfer the call to another station. If the mobile user attempts to seize a CO line or call a hunt group, error tone will be provided.
- **If the "Transfer recall" timer expires, the call is routed to the extension associated with mobile extension.**
- Outgoing calls from a mobile extension are restricted by the COS of the associated station.
- When receiving a call from ipLDK, the mobile will receive either the original CLI or the CLI of the user's station based on configuration of the ISDN line CLI Transit parameter (ORI/CFW).

- To use the mobile extension as a Hunt group member, enable the Mobile Extension Hunt Calls characteristic in the Mobile Extension Table.

### References

- A. Station Message Wait/Call Back
- B. Call Waiting
- C. User Station Programming
- D. DID Calls
- E. Call Transfer
- F. Mobile Phone Remote Menu
- G. SMS, External
- H. Remote VMIB Mailbox Control

### Admin Programming

ISDN line CLI Transit (PGM 143, Flexible button 7)

Assigns the CLI received by the mobile for ipLDK calls

Mobile Extension Table (PGM 236)

Register, activate and configure characteristics of Mobile Extension Support

SMS Settings (PGM 291)

Configures the SMS Service Center information

SMS CO line Attribute (PGM 292)

Assigns CO lines for SMS and receiving stations



### 9.3.1 Mobile Phone Remote Menu

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#### Description

In many circumstances, the mobile must be able to control call routing, activating and deactivating incoming calls including hunt group calls. The Mobile Phone Remote Menu permits the user to access the system to activate and deactivate Mobile Phone Support as well as forward calls to the user VMIB Voice Mail.

#### Operation

To access the Mobile Phone Remote menu from the Mobile phone,

1. Call DID number of the associated station; the system provides internal dial tone.
2. Dial the Remote Menu code, '# 1'.
3. Dial the desired Control code as below.
  - 1: Activate Mobile extension
  - 2: Deactivate Mobile extension
  - 3: Fwd to VMIB
  - 4: cancel Fwd to VMIB
  - 5: Activate UCD DND (Agent log out)
  - 6: Deactivate UCD DND (Agent log in)
4. Hang-up

#### Consideration

- The mobile phone must be registered prior to attempting to call the system.

#### References

- A. Call Forward
- B. ACD
- C. Mobile Phone Support

#### Admin Programming

Remote Control Menu Code (PGM 109, Flexible button 7)

Assigns the dial code to access the Remote Menu from the mobile

Mobile Extension Table (PGM 236)

Register, activate and configure characteristics of Mobile Extension Support

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## 10 MISCELLANEOUS SERVICES

### 10.1 Alarm

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#### Description

The system can be programmed to recognize the status of an external contact from a relay (open or closed). When activated, the System will signal programmed Stations with a single tone repeated at 1-minute intervals or a continuous tone. This capability is commonly employed to provide remote alarm signals or as a Doorbell signal. When used as an alarm, the assigned stations receive the programmed signal. To stop the signal, the alarm should be deactivated (reset) from a station assigned to receive the alarm signal. Also, the condition causing the alarm should be cleared.

When used as a Doorbell, a single tone is used. Each time the Doorbell contact activates, the assigned stations receive the Doorbell signal. The system can be configured to allow the receiving station to activate a contact to open the door.

#### Operation

When the Alarm activates, the system sends the alarm signal to assigned stations.

To terminate the alarm signal from an idle station assigned to receive the alarm,

1. Dial '5 6 5'.
2. The alarm signal will be terminated at all assigned stations.

#### Consideration

- The external alarm contacts should be dry (no voltage/current source connected) and connected as described in the Installation manual.
- An SLT cannot be assigned to receive alarm signals.
- Alarm Reset can be programmed on a Flexible button.

#### Reference

- A. Doorbell
- B. Station User Programming
- C. Door Open

#### Admin Programming

Alarm Enable (PGM 163, Flexible button 1)

Enables Alarm contact monitoring

Alarm Contact Type (PGM 163, Flexible button 2)

Defines the Alarm contact type, normal open or close

Alarm Mode (PGM163, Flexible button 3)

Sets the contact as an Alarm or Doorbell contact

Alarm Signal Mode (PGM 163, Flexible button 4)

Sets the Alarm signal as one-time or repeating

Station Alarm Attribute (PGM 113, Flexible button 10)

Assigns Alarm notification to a range of phones

## **10.2 BGM (Background Music)**

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### Description

A music source connected to the ipLDK-60 can provide music to the speaker of a DKT speaker as Background Music. BGM is only provided while the Station is idle. Ringing, Paging, or going off-hook will terminate BGM.

### Operation

To turn BGM On/Off,

1. Press the [TRANS/PGM] button.
2. Dial '7 3';
3. Dial the desired BGM source (00 – 08), 00 turns BGM off.
4. Return to idle to receive Background Music from the selected source is connected.

To transmit background music to the External Page from an Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 7 6'.
3. Dial the Background Music channel number.
4. Press the [HOLD/SAVE] button.
5. After returning to idle, the selected background music is sent to the External Page port.

To transmit background music to an Door Phone from an Attendant,

1. Press the [TRANS/PGM] button.
2. Dial 0 7 5.
3. Dial the background music channel number.
4. Press the [HOLD/SAVE] button.
5. After returning to idle, the selected background music is sent to the Door Phone.

### Consideration

- When External music is assigned, the music source should be connected to the system as described in the Installation manual.
- The same music source can be used for BGM and MOH.
- While listening to BGM, use the Volume button to adjust the audio level.
- Only one (1) BGM channel is supported.
- SLT ports connected with an MOHU can be used as the music source.

### Reference

- A. MOH

### Admin Programming

SLT MOH (PGM171, Flexible button 4 ~ 8)

Assigns the SLT ports for Music and tones

### **10.3 Chime Bell**

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#### Description

A station can be configured to activate a Chime at another station. When activated, the assigned station will receive the Chime Bell tone for the Chime Bell timer. In addition, a Chime Bell can be configured to activate the External Loud Bell contact.

#### Operation

To program a Chime Bell Flexible button at a DKT,

1. Press the [TRANS/PGM] button.
2. Press the desired Flexible button.
3. Press the [TRANS/PGM] button
4. Dial '\*' 9.
5. Press the [HOLD/SAVE] button.

To activate the Chime Bell,

1. Press Chime Bell button at a station as to activate a Chime Bell.

#### Consideration

- The ring stops when Chime Bell Timer expires.
- A muted Chime Bell is used if the receiving station is busy.
- Chime Bell is available only to DKTs.

#### Reference

- A. Station User Programming
- B. Flexible Buttons
- C. External Loud Bell

#### Admin Programming

Chime Bell Station Pair (PGM184, Flexible button 1)

Assigns the master and receiving Stations

Chime Bell Relay (PGM184, Flexible button2)

Enable activation of the Loud Bell contact when Chime Bell is activated

Chime Bell Timer (PGM184, Flexible button 3)

Sets the duration of the Chime Bell signal

Chime Bell Frequency (PGM184, Flexible button 4)

Defines the tone frequency to use for the Chime Bell

## 10.4 Door Phone

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### Description

A convenient Door Phone can be connected to the system and stations assigned to receive notification when the Call button on the Door Phone is pressed.

### Operation

To place a call from the Door Phone box,

1. Press the [CALL] button.
2. The assigned stations will ring.

To call an Door Phone,

1. Lift the handset or press the [On/Off] button.
2. Dial station number of Door Phone.
3. After the beep tone, announce the call.

To answer a Door Phone box call from a DKT,

1. Press the Intercom Door Box flexible button at the DKTU

To answer a Door Phone call from an SLT

1. Lift the handset.

### Consideration

- The Door Phone is an intercom device only, it cannot receive calls from a CO line and cannot be a party in a conference (refer to Ref. B) call.
- A Door Phone can be assigned to receive Page announcements (refer to Ref. C).
- To receive a Door Phone call, the station must be assigned ICM Box Signaling.
- After the ICM Box timer expires, an unanswered Door Phone call is terminated.
- In certain regions TELKOM or ISRAEL, for an SLT to receive Door Box signals, a DSS button must be assigned to the SLT as well as ICM Box Signaling. In other regions, the SLT need only be assigned ICM Box Signaling to receive Door Box Signals.
- SLT and WHTU can receive only one call from intercom box.

### Reference

- A. DND
- B. Conference
- C. Paging
- D. Background Music

### Admin Programming

Station ID Assignment (PGM 110)

Assigns the type of Station including ICM Box used for the Door Phone box

ICM Box Signaling (PGM 111, Flexible button 6)

Assigns Stations to receive a Door Phone Call

Flexible Button Assignment (PGM 115)

Assigns Door Box DSS button to stations

ICM Box Music Channel (PGM 171, Flexible button 3)

Assigns the BGM Music source for the Door Phone box

ICM Box Timer (PGM 181, Flexible button 6)

Sets the duration a Door Phone box call will signal assigned Stations

## 10.5 External Control Contacts

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The Basic and Expansion cabinets of the ipLDK-60 each have two (2) contacts (total 4) that can be used to activate a Loud Bell, open a door, or connect external paging equipment.

### 10.5.1 Loud Bell

---

#### Description

External Control Contacts can be connected to an external alerting device (Loud Bell) and configured to activate when an assigned station receives an incoming CO line call, transferred CO line call or intercom call. This is commonly used to notify users of an incoming call and can be used as to activate a Night Bell. With UNA enabled, any station can answer the call.

#### Consideration

- Multiple contacts can be assigned to the station but only the first contact will be activated when the station rings.
- The Loud Bell Contact 1 can be programmed to operate as an External Night ring Contact as well as a LBC contact. In the Night mode, the Loud Bell Contact will follow Night ring assignments (refer to Ref. B).
- An external ringing device should be attached to the contacts.

#### Reference

- A. Universal Night Answer (UNA)
- B. Ring Assignments

#### Admin Programming

CO Ring Assignment (PGM 144)

Assigns the ringing destination for a Normal CO Line

External Control Contacts (PGM 168)

Defines the function of each contact

External Night Ring (PGM 160, Flexible button 7)

Enables Loud Bell operation for calls on UNA enabled CO Lines in Night service mode

Universal Night Answer (PGM 141, Flexible button 8)

### 10.5.2 Door Open

---

#### Description

External Control Contacts can be assigned to activate when a code is dialed. The contact will remain activated for the Door Open time. The contact can be used as a Door Open contact or for other functions requiring the operation of a contact. Each contact has an associated activation code:

Door Open Code	Contact
# * 1	1st Door Open
# * 2	2nd Door Open
# * 3	3rd Door Open
# * 4	4th Door Open

A DKT can be assigned with a Flexible button to activate a Door Open contact

#### Operation

To activate a Door Open contact,

1. Lift the handset or press the On/Off button
2. Dial the Door Open code for the desired contact
3. The contact activates for the Door Open time.

#### Reference

- A. Door Phone Box
- B. Flexible Buttons
- C. Station User Programming

#### Admin Programming

External Control Contact (PGM 168)

Defines the function of each contact

Door Open Timer (PGM 181, Flexible button 5)

Sets the duration a Door Open contact will remain activated

### 10.5.3 External Page Control

---

#### Description

An External Control Contact may be assigned to activate when an External Page Zone is accessed. This is often used to connect external paging equipment to or activate a paging amplifier.

#### Consideration

- External control contacts are rated at 30 volts DC and 1 Amp.

#### Reference

- A. Paging

#### Admin Programming

External Control Contact (PGM 168)

Defines the function of each contact



## **10.6 In-Room Indication**

---

### Description

The Supervisor of an In-Room group can indicate their presence to other members of the group. When activated by the Supervisor, the In-Room button LED at each member station illuminates. Up to ten (10) In-room groups can be configured with up to twenty (20) members excluding the Supervisor in each group. A station can be the Supervisor for up to 10 groups; each group requires a separate activation button.

### Operation

To program an In-Room flexible button at a DKT,

1. Press the [TRANS/PGM] button.
2. Press the desired Flexible button.
3. Press the [TRANS/PGM] button
4. Dial '\* 8' and the In-Room group number (01 ~ 10).
5. Press the [HOLD/SAVE] button.

To activate or deactivate In-Room indication from the Supervisor station

1. Press In-Room Indication button.
2. Press the [HOLD/SAVE] button.

### Consideration

- If Supervisor station is not Idle, In-Room Indication button does not work.
- Pressing the In-Room button at a station other than the Supervisor will return error tone.
- The Supervisor must press the [HOLD/SAVE] button within 5 seconds after In-Room button to activate/deactivate In-Room Indication.

### Reference

- A. Station User Programming
- B. Flexible Buttons

### Admin Programming

In-Room Indication Supervisor (PGM183, Flexible button 1)

Assigns the Supervisor of an In-room group

In-Room Indication Member (PGM183, Flexible button 2)

Assigns members of an In-room group

## **10.7 Station Port Blocking**

---

### Description

A station port can be blocked placing the port out-of-service.

### Consideration

- Attendants and stations allowed Admin access cannot be blocked.
- Both stations of a Linked pair are blocked if either is blocked.
- A station connected to a blocked port is blocked from all use including Emergency calls.

### Reference

- A. Emergency Calling Service
- B. Linked Pairs

### Admin Programming

Station Port Block (PGM113, Flexible button 15)

Assigns stations to be blocked

## 10.8 Wake-up Alarm

---

### Description

The station can be programmed by the user or the Attendant to provide an Alarm clock call for use as a wake-up call or reminder. The Alarm clock can be set as a one-time event or to occur daily. When the alarm activates, the station will ring. Answering the wake-up call delivers stutter dial tone or the VMIB Wake-up announcement.

The system will attempt to alert the user three times with 90 seconds between attempts. During each attempt, the phone will receive a normal ring for 30 seconds. If the user does not answer the alarm, after the third attempt, the alarm is canceled and the Wake-up failure is logged. The System Attendant can be notified of the Wake-up failure by the display of the 'Wake-Up Fail' event message. The notification can be timed (1 to 98 minutes) after which the failure notification is terminated or the notification can be set to '0', requiring the Attendant to call the station to cancel the notification.

### Operation

To set the Alarm time at the station,

1. Press the [TRANS/PGM] button.
2. Dial '4 1'.
3. Dial the alarm time as four (4) digits in 24 hour format (00:00 to 23:59).
4. Dial # to have the alarm repeat on a daily basis.
5. Press the [HOLD/SAVE] button.

To erase an Alarm time at the station,

1. Press the [TRANS/PGM] button.
2. Dial '4 2'.
3. Press the [HOLD/SAVE] button.

To respond to the Alarm,

1. Lift the handset or press the [On/Off] button

To register a wake-up time alarm from the Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 4 2'.
3. Dial the station range to be alerted, for single station, enter ' \* ' as the second station number.
4. Dial the desired time the alert should display (2-digit hour and minute, 24-hour mode).
5. Dial '#' to have the alarm repeat on a daily basis.
6. Press the [HOLD/SAVE] button.

To cancel a wake-up alarm from the Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 4 3'.

3. Dial the station range that was to be alerted, for a single station, enter ' \* ' as the second station.
4. Press the [HOLD/SAVE] button.

To erase the Wake up Fail message from the Attendant Station

1. Call the non-responding Station

#### Consideration

- If a Wake-up alarm is registered at a station, a ' \* ' is displayed in front of the time of day in the LCD.
- If the system is equipped with a VMIB, the Wake-up greeting is played three times followed by MOH.
- A timed notification is terminated when the Wake-Up Ring Fail timer expires or the Attendant calls the station.

#### Reference

- A. Attendant Positions
- B. VMIB System Greetings

#### Admin Programming

Wake-up Fail Ring Timer (PGM 182, Flexible buttons 7)

Sets the duration of the notification (0 = Call station, 1 to 98 minutes, or 99 = no notification)

## 11 VOICE SERVICES

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### 11.1 Company Directory

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#### Description

A CCR Table can include the Company Directory as a destination for a caller dialed digit. In the directory, the caller enters characters from the dial pad and the system matches entries with programmed user and virtual mailbox names. When completed, the Name greeting for each match plays and the caller may select the appropriate party to call.

#### Operation

To Access the directory

1. Caller dials Company Directory index from CCR
2. System plays First/Last name prompt
3. Caller selects Name
4. System plays 3 character prompt
5. User enters characters
6. System plays Name Greeting for each match (maximum 9)
7. Caller selects desired match and call is placed.

To record a User Name Greeting

1. Press the [TRANS/PGM] button or dial the SLT Program Access code '563'.
2. Dial '6\*' and follow the voice instructions to record the Name Greeting.
3. Press the [Hold/Save] to store the greeting.

#### Consideration

- The first and last name of each system user or virtual mailbox must be defined and a Name greeting recorded.
- The station or virtual mailbox must be provided with VMIB Mailbox Access to enable a mailbox and Name recording.

#### Reference

- A. CCR
- B. System Greetings and Prompts

#### Admin Programming

Station First Name (PGM 110, Flexible button 3)

Assigns the First Name for the user in the Company Directory

Station Last Name (PGM 110, Flexible button 4)

Assigns the Last Name for the user in the Company Directory

Station VMIB Access (PGM 113, Flexible button 2)

Enables the Station Mailbox

Virtual Mailbox Attributes (PGM 129)

Enables and assigns attributes to a virtual mailbox

Virtual Mailbox First Name (PGM 129, Flexible button 22)

Assigns the Virtual Mailbox first name in the Company Directory

Virtual Mailbox Last Name (PGM 129, Flexible button 23)

Assigns the Virtual Mailbox last name in the Company Directory

CCR (PGM 228)

Defines the destination for a dialed digit as the Company Directory

## 11.2 Two-way Recording

---

### Description

This feature allows a user to record a CO line call to the station's mailbox. Two-way Recording can be enabled only for DKT stations and the station must have a Flexible button assigned as a {Record} button or, for the 6200 series, the Record softkey may be used. When a call is recorded, the system activates a Message Wait to the station.

### Operation

To assign a {Record} Flexible button at the DKT,

1. Press the [TRANS/PGM] button.
2. Press the desired Flexible button.
3. Press the [TRANS/PGM] button.
4. Dial '5 4'.
5. Press the [HOLD/SAVE] button.

To record a conversation while on a CO line call,

1. Press the {Record} button or softkey.
2. The conversation is record

To stop recording,

1. Press the {Record} or Flash button or hang-up.
2. Conversation recording stops and the message is saved.

### Consideration

- Users should announce the recording, no recording tone is provided.
- During a recording, the {RECORD} button flashes to indicate a recording in progress.
- If the external VM requires DTMF for control signaling or there is no external Voice mail, the conversation is recorded to the user's VMIB Mailbox.

### Reference

- A. Station User Programming
- B. External Voice Mail
- C. Station Hunt Groups
- D. VMIB Voice Mail

### Admin Programming

Two Way Recording (PGM 112, Flexible button 10)

Enables a station to use Two-way Recording

### **11.3 VMIB Auto Attendant Announcement**

---

#### Description

An announcement can be recorded with a corresponding CCR table and assigned as a back-up Attendant. Should the Attendant not answer an incoming CO line call in the **DID/DISA** No-answer time, the call is sent to the Auto Attendant announcement. The call is then routed based on the caller input and the CCR Table.

#### Consideration

- Only new incoming calls are routed to the Auto Attendant announcement. Recalls and transferred calls are not sent to the Auto Attendant.
- The CO line must be assigned for DISA service for proper operation otherwise, the system will not recognize incoming digits and will disconnect after playing the announcement.
- If the CO line is assigned to ring at multiple stations, the call is not sent to the Auto Attendant.

#### Reference

- A. VMIB System Greetings and Prompts
- B. CCR
- C. DISA

#### Admin Programming

Auto Attendant VMIB Announce Number (PGM 165)

Assigns announcement number for Auto Attendant announcement

DID/DISA No Answer timer (PGM 181, Flexible button 2)

Sets no-answer timer for DID and DISA calls

CCR (PGM 228)

Defines the destination for a dialed digit as the Company Directory



## 11.4 VMIB System Greetings and Prompts

### Description

The VMIB incorporates memory for System Greetings and various voice prompts. System Greetings, which are recorded from the System Attendant, are used for Hunt Groups, Auto Attendant, etc. The various voice prompts have been pre-recorded and stored in the VMIB. The System Attendant can record System Prompts to be used in place of the pre-recorded prompts. The pre-recorded prompt remains in memory but the prompt recorded by the Attendant is used. Should the Attendant recorded prompt be erased, the pre-recorded prompt is used..

Announcements 001 to 070 are employed for System Greetings and other announcements are employed in place of the pre-recorded prompts as shown in the chart below.

Prompt No	Pre-recorded Voice Messages
073	Invalid entry
074	The time has expired
075	Please try again
076	Your call will be transferred to the Attendant
078	To record your greeting press the pound key.
079	Record your greeting
080	Enter your password followed by the pound key
081	Station is busy
083	Off-net Call Forwarded station
084	Do-Not-Disturb
085	The station is not answering
088	Main Menu To listen to your messages, press one. To record your greeting, press two. To send a message, press three. For personal options, press 4. To access administrative options, press 9. To access another mailbox, press pound.
089	To hear the current message, press one, To hear the next message, press two, To delete the current message, press three, To reply to the current message, press five, To forward the current message, press six, To call the sender, press seven, For the main menu, press star.
096	At the tone, please leave a message, to stop recording press the pound key.
097	No pre-recorded prompt, replaces SLT stutter dial tone for message waiting indication
098	Used as the default User Greeting, no pre-recorded prompt

Other pre-recorded prompts stored in the VMIB memory are fixed and cannot be changed by the Attendant.

A greeting can be recorded locally by the Attendant or, when defined as a CCR destination, from a remote location. In the latter case, the CCR Table must include the Remote Record destination. The caller may then use the remote access procedure to record a greeting.

### Operation

To record System Greetings from Attendant;

1. Press the [TRANS/PGM] button and dial '0 6'.
2. Dial the desired announcement number (001 ~ 070). The recording instruction "Press the # key to record." plays followed by any previous recording for the announcement.
3. Dial '#' and start recording after the beep, or to record external music dial '\*' to hear the external source then '#' to record external music
4. Press the [HOLD/SAVE] button to stop and store the recording.

To delete a System Greetings from Attendant,

1. Press the [TRANS/PGM] button.
2. Dial the code '0 6'.
3. Dial the appropriate message number, the recording is played.
4. Press the [SPEED] button while the message is playing.

To record a greeting from remote CCR access;

1. Dial the digit assigned for Remote record access in the CCR Table.
2. Dial the System Attendant Authorization code
3. Dial the System Greeting number (00~70) and follow the recording instructions
4. Dial '#' to start recording.
5. Press the '#' to save the recording

### Consideration

- Pressing the [On/Off] button while recording, will stop and store recording the recording.
- System Greetings and prompts can be recorded at the System Attendant or can be downloaded to the system in the maintenance mode.
- There is no time limit for an individual System Greeting or prompt.
- To record or delete a System Greeting or prompt, all the VMIB ports must be idle state.
- When a call is transferred to an Attendant, the 'Transfer to Attendant' prompt followed by ring-back tone is presented to the caller.
- If a System Greeting or prompt is not recorded, tone will be heard.
- The VMIB can maintain 800 messages including user greetings, voice messages, and two-way recordings.
- If a recording exceeds the VMIB memory capacity, recording is terminated and the previous recording, if any, is stored.
- The System Attendant must have VMIB Access enabled to record a greeting or prompt.

Reference

- A. Station Hunt Groups
- B. Auto Attendant Message
- C. DISA
- D. DID
- E. VMIB Voice Mail

Admin Programming

VMIB Message Retrieve Date/Time (PGM 113, Flexible button 9)

Enables playing the Time & Date stamp for VMIB voice messages

VMIB Access (PGM 113, Flexible button 2)

Permits the station access to the VMIB and a Voice mail-box

CCR Table (PGM 228)

Configures destinations for a CCR Table

## 11.5 VMIB Voice Mailbox

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### 11.5.1 Incoming VMIB Voice Mail

---

#### Description

Calls to a station may be sent to the user's VMIB mailbox. The caller receives the user's greeting and can leave a voice message. A message must be longer than the Valid User Message time and cannot exceed the User Record time. After a message is left, the DKT station Call Back button flashes and the DKT LCD displays the message waiting queue count. For SLTs, stutter dial tone is received when lifting the handset.

Calls are sent to a mailbox by

- the user activating Call Forward (refer to Ref. D) or DND with forward assigned,
- Auto Forward to the VMIB after the VMIB Forward No Answer timer
- assigned as the Flexible DID Destination
- Call Transfer to a user's Mailbox
- CCR Destination

#### Operation

To Transfer a call to a user's Voice Mailbox,

1. Press the [TRANS/PGM] button.
2. Press the [CALL BK] button.
3. Dial the Station to receive the voice message
4. Hang Up to complete the Transfer.
5. The transferred party receives the User Greeting for the station.

To leave a voice message for a station:

1. After the User greeting and beep tone, record your message.
2. Hang-up to send the message.

#### Consideration

- For proper operation, the Voice Mailbox user should record a User Greeting.
- Direct transfer to the mailbox of a station in a networked system is not allowed.
- When the recorded message is shorter than Valid User Message time, the message is not saved. If the User Record Timer expires, confirmation tone is heard and the message is saved in the mailbox.
- The VMIB can maintain 800 messages including user greetings, voice messages, and two-way recordings. System Greetings and prompts are stored separately
- If all the VMIB ports are busy, ring-back tone will be provided and the Auto VMIB Forward timer is initiated to retry answer.
- The VMIB recordings including System Greetings, User Greetings, two-way recordings, and Voice Messages are not erased if the system database is returned to default using PGM 450.
- If the database is returned to default using the 4<sup>th</sup> switch position of DIP switch 1, only System greetings are not erased, other VMIB recordings are erased.

- Each Voice Mailbox is assigned a COS that will affect operations including message retention.

### Reference

- A. VMIB User Greeting
- B. VMIB System Greetings
- C. Managing VMIB Voice Mail
- D. Call Forward
- E. Preset Call Forward
- F. CLI Message Wait
- G. Station Message Wait/Call Back
- H. DID
- I. DND
- J. DISA
- K. Private Networking

### Admin Programming

Auto Forward to VMIB (PGM 113, Flexible button 14)

Configures Stations to forward calls automatically to the user's VMIB mail-box

Auto Forward to VMIB timer (PGM 181, Flexible button 20)

Sets the time calls will forward to the user's VMIB voice mailbox

Flexible DID Table (PGM 231)

Assigns destination for DID type 2 conversion

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

Station Voice Mailbox Attributes (PGM 127)

Defines characteristics of the Voice mailbox

VMIB User Record Timer (PGM 181, Flexible button 3)

Defines the maximum duration for a voice message

VMIB Valid User Message Timer (PGM 181, Flexible button 4)

Defines the minimum duration for a valid voice message

CCR Table (PGM 228)

Configures destinations for a CCR Table

Voice Mailbox COS (PGM 238)

Assigns characteristics of the Class of Service for voice mailboxes

### 11.5.2 Announcement only Mailbox

---

#### Description

A mailbox can be configured to provide an announcement only and not provide incoming message storage. After the announcement is played, the call is disconnected or returned to the previous announcement.

#### Consideration

- An announcement Only Mailbox must be assigned with VMIB access.

#### Reference

- A. Park and Page
- B. User Greeting

#### Admin Programming

Station Mailbox Attributes, Announce Only (PGM 127, Flexible button 2)

Assigns code for Announce Only operation

Station Mailbox, Action After Announce (PGM 127, Flexible button 3)

Assigns destination after announcement

### 11.5.3 Administrative Mailbox

---

#### Description

A Mailbox may be assigned as an Administrator allowing the user access to other mailboxes and the administration menu to add, delete, or reset a mailbox password to default (station number). The administrator can record a Greeting and a Name for a mailbox and record and send a broadcast message to all mailboxes.

#### Operation

From the Main Menu, the Administrator can access the Administration Menu and the Administration menu is played.

“To add a mailbox, press 1  
To delete a mailbox, press 2  
To reset a mailbox password, press 3  
To record a mailbox greeting, press 4  
To record a broadcast message, press 5  
To record a mailbox name, press 6”

To Add a mailbox,

1. Press 1
2. Enter the Mailbox
3. Enter the COS (1 ~ 5)
4. Press 1 to confirm or # to cancel

To Delete a mailbox

1. Dial 2
2. Dial the Mailbox
3. Dial 1 to confirm or # to cancel

To Reset the password to default for a mailbox (the station number)

1. Dial 3
2. Dial the Mailbox
3. Dial 1 to confirm or # to cancel

To record a mailbox greeting

1. Dial 4
2. Enter the Mailbox
  - Dial 1 to listen
  - Dial 2 to record a new greeting
  - Dial 3 to delete the existing greeting
3. Dial # to cancel

**To record and send a Broadcast Message**

1. Dial 5
2. Enter the Broadcast Message number (01 ~ 10)
  - Dial 1 to listen
  - Dial 2 to record a new greeting
  - Dial 3 to delete the existing greeting
3. Dial # to cancel

**To record a Mailbox Name**

1. Dial 6
2. Enter the Mailbox
  - Dial 1 to listen
  - Dial 2 to record a new greeting
  - Dial 3 to delete the existing greeting
3. Dial # to cancel

**Consideration**

- An Administrator cannot access Private messages in another mailbox.
- When deleted, all recordings associated with a mailbox are erased as well as any Names assigned. The mailbox settings are returned to their default with VMIB access disabled.

**Reference**

- A. Virtual Mailbox
- B. Incoming VMIB Voice Mail
- C. Recording User Greeting
- D. Company Directory

**Admin Programming**

Station Voice Mailbox Attributes (PGM 127, Flexible button 1)

Assigns Voice mailbox as an Administrative Mailbox

Virtual Voice Mailbox Attributes (PGM129, Flexible button 1)

Assigns Virtual Voice Mailbox as an Administrative Mailbox



#### 11.5.4 Managing VMIB Voice Mail

---

##### Description

After a message is left, the DKT station [Call Back] button flashes and the LCD displays the message waiting queue count. For SLTs, stutter dial tone is received when lifting the handset. The ipLDK-60 can be further configured to notify the user with a telephone call announcing the new message or an e-mail citing the new messages. The station may review, save, delete, reply or transfer the message to another station. In addition, DKT users can reply and forward messages to other users and rewind the message to playback that portion of the message. While in the Mailbox, pressing the '\*' button will return to the previous menu and cancel the existing operation.

##### Operation

To access the VMIB to manage voice messages,

1. Press the [Call Back] button or dial '5 5 7'.
2. If requested, enter the Voice Mail password and '#'.
3. The user hears the Main Menu  
"To Listen to your Message, press one  
To record your greeting, press two  
To send a message, press three  
For personal options, press four  
To access another mailbox, press pound."
4. Dial 1 and follow Voice instructions to manage your Voice Mails.  
Play messages - 1  
Save Message - 2  
Delete messages - 3  
Reply - 5  
Rewind - 6  
Fast Forward - 7  
Skip - 8  
Pause/Start - 9
5. At the end of the message, the Manage Mail prompt is played  
To hear the current message, press one,  
To hear the next message, press two,  
To delete the current message, press three,  
To reply to the current message, press five,  
To forward the current message, press six,  
To call the sender, press seven,  
For the main menu, press star

##### Consideration

- ipLDK supports several types of messages. When multiple types of messages are available to the user, the order of retrieving messages is:  
Station Message Wait

VMIB Message Wait

CLI Message Wait

External VM Group Message Wait

- A VMIB message may also include a CLI Message Wait. In this case, deleting the VMIB message also deletes CLI Message Wait.
- When sending a copy of a message to another station, the receiving station must be allowed VMIB access.
- As a default, the Time & Date stamp, indicating when the message was received, is played followed by the message. If desired, the Time & Date stamp can be removed and only the message is played when retrieving messages.
- For pulse-type SLTs, when the mailbox is accessed, the first message is played then deleted and warning tone is received indicating additional messages are waiting. To retrieve the next message, hang-up and access the VMIB mailbox again.
- Messages are retrieved in the assigned order (VMIB Message Type), either First-in-First-out or Last-in-First-out.
- The Rewind softkey will rewind the message based on the VMIB Rewind timer.
- When a message is sent to another station, a copy of the message is maintained in the sending user's mailbox.

### Reference

- A. CLI Message Wait
- B. Station Message Wait/Call Back
- C. External VM Group

### Admin Programming

System Numbering Plan (PGM 104, Flexible button 17)

Assigns code for Message Wait response from SLTs

Station Type (PGM 110, Flexible button 1)

Defines the type of station equipment for the Station

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

VMIB Message Type (PGM 111, Flexible button 17)

Defines the playback order for voice messages, FIFO or LIFO

VMIB MSG Retrieve Password (PGM 113, Flexible button 8)

Forces the use of the Station Authorization code to access the VMIB mailbox

VMIB Message Retrieve Date/Time (PGM 113, Flexible button 9)

Enables playing the Time & Date stamp for VMIB voice messages

Station Mailbox Attributes (PGM 127)

Assigns characteristics of the mailbox including notification and COS

Virtual Mailbox Attributes (PGM 129)

Assigns characteristics of the Virtual Mailbox including notification and COS

VMIB User Record Timer (PGM 181, Flexible button 3)

Defines the maximum duration for a voice message

VMIB Rewind Message Timer (PGM 181, Flexible button 17)

Sets the VMIB message rewind time

Voice Mailbox COS (PGM 238)

Class of Service settings for Voice mailboxes, including Msg retention and Msg marking (Confirm, Private, or Urgent)

### 11.5.5 New Message Notification

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#### Description

The Voice Mailbox services can be configured to notify a user of new messages. Each mailbox is configured to place an outbound call to a telephone number or send an e-mail to the user indicating new messages.

Further, by employing Cascade Mailboxes, messages marked for Urgent delivery can activate the Message cascade sending the message to another mailbox that can provide different notification than a normal message.

#### Consideration

- For outbound call notification, the number and interval between notification attempts as well as the telephone number and CO Line group for the call are configurable.
- For proper operation, the CO Line used for outbound call notification should provide answer and disconnect supervision.
- For e-mail notification, an SMTP e-mail account must be established for the system to send mails. The SMTP e-mail account information must also be configured in the ipLDK-60.

#### Reference

- A. Incoming VMIB Voice Mail

#### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

Station Mailbox Attributes (PGM 127)

Assigns characteristics of the mailbox including notification and COS

Virtual Mailbox Attributes (PGM 129)

Assigns characteristics of the Virtual Mailbox including notification and COS

VMIB User Record Timer (PGM 181, Flexible button 3)

Defines the maximum duration for a voice message

Voice Mailbox COS (PGM 238)

Class of Service settings for Voice mailboxes, including Msg retention and Msg marking (Confirm, Private or Urgent)

### 11.5.6 Message Reply

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#### Description

A user can send a reply to messages received from other internal system users. After listening to a message, the user is given the option to reply to the message. If the user selects to reply, option voice prompts are provided to instruct the user on recording and sending the reply.

#### Operation

To reply to a message, while listening to or after the message is played,

1. Press '5' to reply to the current message.
2. Follow the voice prompt instructions to record your reply and press # when done

#### Consideration

- Reply is only available for messages from other internal Voice Mail users.
- The user to receive the reply must have VMIB access.
- A user-recorded message (comment) can be added to the message before replying to or forwarding a message.

#### Reference

- A. Incoming VMIB Voice Mail
- B. Add a comment message

#### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

### 11.5.7 Forward A Message

---

#### Description

A user can forward a copy of a Voice Message to another user's VMIB (Voice Mail In-box). A comment can be recorded and sent with the copy of the message.

#### Operation

After hearing a message,

1. Dial 6, mailbox number request plays
2. Follow the voice prompt instructions to forward the message.

#### Consideration

- The receiving station must have a Voice Mailbox.
- Messages can be copied only to another Mailbox.
- A user-recorded message can be added to the message before forwarding a message.
- The user must listen to the entire message before forwarding.

#### Reference

- A. Incoming VMIB Voice Mail
- B. Add a comment message

#### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

### 11.5.8 Add a Comment Message

---

#### Description

A user can add a comment to the current message prior to replying or forwarding a copy of the message to another user.

#### Operation

While after hearing a message,

1. Press the Add softkey, the record prompt plays
2. Follow the voice prompt instructions record a message that is added to the beginning of the current message.
3. Listen to the current message with the comment message added.
4. Forward the message.

#### Consideration

- The receiving station must have a Voice Mailbox.

#### Reference

- A. Incoming VMIB Voice Mail
- B. Forward A Message
- C. Message Reply

#### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

### 11.5.9 Send A Message

---

#### Description

A user may record and send a message to another user's mailbox or to a distribution list of mailboxes. The message can be marked as normal or private and for normal or urgent delivery.

#### Operation

1. Press the [Call Back] button or dial '5 5 7'.
2. If requested, enter the Station Authorization code.
3. The user hears the Main Menu  
"To Listen to your Message, press one  
To record your greeting, press two  
To send a message, press three  
For personal options, press four  
To access administrative options, press nine  
To access another mailbox, press pound."
4. Dial 3
5. Follow the voice prompt instructions to record and send the message.

#### Consideration

- The message may be sent to an individual mailbox or a distribution list of mailboxes. Before selection, a distribution list must be created under the Personal Options of the Voice Mailbox.
- The user may search the Company Directory to select a mailbox to receive the message.

#### Reference

- A. New Message Notification
- B. Managing Personal Options
- C. Dial-by-Name

#### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

Station Mailbox Attributes (PGM 127)

Assigns characteristics of the mailbox including notification and COS

Virtual Mailbox Attributes (PGM 129)

Assigns characteristics of the Virtual Mailbox including notification and COS

Voice Mailbox COS (PGM 238)

Class of Service settings for Voice mailboxes, including Msg retention and Msg marking (Confirm, Private, etc.)



### 11.5.10 Managing Personal Options

---

#### Description

The Personal Options selection from the Mailbox main menu permits the user to define up to five (5) distribution lists that may be used in place of a mailbox number when sending or forwarding a message.

#### Operation

To manage your Personal Options, from the Main menu

- “To Listen to your Message, press one
  - To record your greeting, press two
  - To send a message, press three
  - For personal options, press four
  - To access administrative options, press nine
  - To access another mailbox, press pound”.
1. Dial 4
  2. Follow the voice prompt instructions to edit distribution lists
  3. Dial 1 to edit distribution lists.
  4. Enter the Distribution List number (1\_5)
  5. To add press 1 then the mailbox number  
To delete press 2 then the mailbox number  
To listen to the mailboxes in a list press 3.

#### Consideration

- A user must have VMIB Access.

#### Reference

- A. Forward a Message
- B. Sending a Message
- C. Managing VMIB Voice Mail

#### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

Station Mailbox Attributes (PGM 127)

Assigns characteristics of the mailbox including notification and COS

Virtual Mailbox Attributes (PGM 129)

Assigns characteristics of the Virtual Mailbox including notification and COS

Voice Mailbox COS (PGM 238)

Class of Service settings for Voice mailboxes, including Msg retention and Msg marking (Confirm, Private, etc.)

### 11.5.11 Virtual Mailbox

---

#### Description

The ipLDK-60 incorporates Virtual mailboxes (200 ~ 249), which operate in the same manner as a Station Voice Mailbox except that no physical station is associated with the mailbox. Calls may be transferred to the Virtual Mailbox or routed automatically through the DID/DISA or CCR features.

#### Consideration

- VMIB Access must be enabled for the Virtual Mailbox to receive calls.

#### Reference

- A. Direct Inward Dialing (DID)
- B. Direct Inward System Access (DISA)
- C. Caller Controlled Routing (CCR)

#### Admin Programming

Virtual Mailbox Attributes (PGM 129)

Assigns characteristics of the Virtual Mailbox including notification and COS

Voice Mailbox COS (PGM 238)

Class of Service settings for Voice mailboxes, including Msg retention and Msg marking (Confirm, Private, etc.)

### 11.5.12 Remote VMIB Mailbox Control

---

#### Description

A user can access the VMIB mailbox from a remote location through the ipLDK DID/DISA (refer to Ref. A) facilities. Once accessed, the user can manage voice messages, cancel user activated Call Forward, or listen to and change the User Greeting.

#### Operation

To access a VMIB mailbox from a remote location,

1. Place a DID call to the desired Station or call to a system DISA line and dial the station.
2. The User Greeting is played.
3. While listening to the User Greeting dial '\*' to receive the password request.
4. Enter the Mailbox password code followed by '#'.  
5. The Main Menu is heard.
6. Use the appropriate code for the desired function.

#### Consideration

- Dialing '\*' while in a sub-menu, will return the caller to the main control menu.
- If the user does not dial a digit in the Inter-digit time, the connection is dropped.
- The station must have VMIB access.

#### Reference

- A. DID
- B. DISA
- C. Authorization Codes
- D. Call Forward

#### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Permits the Station access to the VMIB and a voice mail-box

Inter-digit Timer (PGM 181, Flexible button 8)

Sets the allowable inter-digit time

### 11.5.13 Recording User Greetings

---

#### Description

Station users with VMIB access can record a Primary and Temporary User Greeting. Normally, when a call is sent to the user's mailbox, the primary greeting is played. When a Temporary Greeting is recorded, the caller receives the temporary greeting in place of the primary greeting. The primary greeting can thus be recorded once and the temporary greeting recorded for unusual absence such as an extended business trip.

#### Operation

To record a User Greeting at a station,

1. Press the [Callback] button or dial '5 7 1'.
2. Dial '2' during or after the Main Menu.
3. Follow Voice prompts to
  - Play a Greeting
  - Record a Greeting
  - Delete a Greeting
4. After the record greeting prompt, any existing greeting is played.
5. Dial '#' and start recording after the beep.

#### Consideration

- There is no time limit on an individual User Greeting.
- When recorded, the Temporary Greeting is played. To revert to the primary greeting, the Temporary greeting must be deleted.
- Greetings may also be recorded from an Administrative Mailbox.

#### Admin Programming

VMIB Access (PGM 113, Flexible button 2)

Permits the station access to the VMIB and a voice mail-box

Administrative Mailbox (PGM 127, Flexible button 2)

Enables access to Admin features including recording greetings

## **12 CALL STATISTICS & TRAFFIC ANALYSIS**

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### **12.1 Account Code**

---

#### Description

Station users may allow tracking of specific calls by entering a non-verified variable length identifier (up to 12 digits) for a call. The identifier or "Account Code" is output as part of the Station Message Detail Record (SMDR) for the call. A Flexible button may be assigned to enter an Account Code and may include the Account Code, (refer to Ref. B).

#### Operation

To enter an Account Code while connected to a CO line,

1. Press the [TRANS/PGM] button.
2. Dial '5 5 0' (refer to Ref. A).
3. Dial the code (maximum 12 digits).
4. Dial '\*' to complete the entry and return to the call.

To enter an Account Code with an Account Code button while connected to a CO line,

1. Press the Account Code button.
2. Dial the code (max. 12 digits).
3. Dial '\*' to complete the entry.

To enter an Account Code prior to placing a call,

1. Press the Account Code button.
2. Dial the code (max. 12 digits).
3. Dial '\*' to complete the entry, Intercom dial tone is received and a CO line may now be accessed.

#### Consideration

- While entering the Account Code, the conversation is muted so the connected party does not hear the Account Code digits as they are dialed.
- The user may enter an Account Code on an incoming or outgoing call. For an outgoing call, the Account Code may be entered before dialing the telephone number.

#### Reference

- A. SMDR
- B. Station User Programming

#### ADMIN Programming

System Numbering Plan (PGM 106, Flexible button 7)  
Assigns the code to enter an Account Code

## 12.2 Lost Call Print-out

---

### Description

ipLDK can provide information for calls that are terminated by the caller before answered or while on hold. The system will output records for these 'Lost calls' in real-time and will maintain a lost call summary count for on-demand reporting. Records indicate the CO line, called station, duration of ring, time and date as well as the type of call. The format of the individual call record is illustrated below with an explanation of the record.

```
NO  STA  CO  TIME  START  DIALED
-----
0001 EXT 031 00:00:10 24/05/99 11:55 R RING 00:05
```

Normal incoming call is received at an assigned station for CO 031, rings for 5 seconds and disconnects.

```
NO  STA  CO  TIME  START  DIALED
-----
0002 101 003 00:01:20 25/05/99 16:23 R RNG 00:09
```

DID call is forwarded to Attendant 101, rings for 9 seconds and is disconnected.

```
NO  STA  CO  TIME  START  DIALED
-----
0003 100 001 00:00:20 25/05/08 18:11 R100 RING 00:04
```

DID call is received at Station 100, rings for 4 seconds and is disconnected.

```
NO  STA  CO  TIME  START  DIALED
-----
0004 102 002 00:01:20 25/05/08 18:37 R103 RING 00:04
```

DID call is received at Station 102 via unconditional call forward to STA 103, rings for 4 seconds and is disconnected.

```
NO  STA  CO  TIME  START  DIALED
-----
0005 621 008 00:00:20 26/05/08 13:02 G620 RING 00:06
```

DID call is received at Ring Group 621, rings for 6 seconds and is disconnected.

```
NO  STA  CO  TIME  START  DIALED
-----
0006 100 001 00:00:04 06/05/08 16:04 H100 RING 00:02
```

DID call is disconnected while held by Station 100.

```
NO  STA  CO  TIME  START  DIALED
-----
0007 102 001 00:00:07 06/05/08 17:04 H100 RING 00:02
```

DID call is disconnected while being transferred from Station 100 to Station 102.

The Dialed number field is preceded by a character indicating the type of call as follows:

- G - Incoming call to Station Hunt group (refer to Ref. A),
- H – Caller disconnected during transfer or while on Hold
- R – DID or forwarded call disconnected before answer.

### Operation

To print the Lost Call count from the System Attendant,

1. Connect the device to receive call records to the system serial port
1. Press the [TRANS/PGM] button.
2. Dial '0 1 1 7' at Attendant station.
3. The lost call count of record is printed in the PC connected in the system.

To clear the Lost Call count from the System Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 1 8', "The lost call count is cleared" is printed and the count reset.

#### Consideration

- Detailed Lost Call records are not stored, only summary statistics are stored for use with Traffic Analysis.
- Intercom calls are not monitored for the Lost Call report.

#### Reference

- A. Hunt Group: 2.6
- B. DID
- C. SMDR (Station Message Detail Recording)

#### Admin Programming

SMDR Print Enable (PGM 177, Flexible button 2)

Enables real-time output of SMDR

Print Lost Call (PGM 177, Flexible button 6)

Enables real-time output of Lost call records

### 12.3 SMDR (Station Message Detail Recording)

---

Description

SMDR (Station Message Detail Recording) provides detailed information on incoming and outgoing calls. Information detailed in the SMDR record includes outgoing CO Line, dialed number, time, date, station that placed or answered the call, duration of call, etc. ipLDK SMDR assignable options permit recording of all external calls, all external outgoing calls or long distance toll calls, calls that exceed a fixed duration and intercom calls including Page and SMS activity.

Detailed call records either are output upon completion of the call (real-time) or stored in system memory. When stored, an output of the stored information can be requested by the System Attendant. Each SMDR record contains the following:

- 5-digit station call originator (terminating for incoming) field
- 3-digit used CO line field
- 8-digit call duration field (HH:MM:SS)
- 8-digit year, month, and day (YY/MM/DD)
- 5-digit time-of-day call originator field
- 1-digit call identification digit-first digit in digit dial field
- 18-digit collected dial digit field
- 2-digit account group number field
- 5-digit pulse metering count field
- 10-digit call cost field
- 12-digit account code field

NO	STA	CO	TIME	START	DIALED	ACT	CNT	COST	ACCOUNT CODE
1211	100	001	HH:MM:SS	YY/MM/DD	hh:mm x123456789012345678	AA	12345	\$\$\$\$\$\$\$\$\$\$	123456789012

Operation

To print SMDR, from the System Attendant,

1. Connect the device to receive call records to the system serial port
2. Press the [TRANS/PGM] button.
3. Dial '0 1 1 1' (Station Base) or '0 1 1 3' (Group Base).
4. Enter the Station or Group range.
5. SMDR records are sent to the connected device.

To delete SMDR records, from the System Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 1 2' (Station Base) or '0 1 1 4' (Group Base).
3. Enter the Station or Account Group range.
4. Press the [HOLD/SAVE] button

To abort SMDR printing from the Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 1 6', printing is cancelled.



Consideration

- IpLDK determines a long distance call based on the long distance code (digits dialed as the first digit to access toll facilities) or the number of dialed digits. If the user-dialed digits match an SMDR LD code, the number of digits exceeds SMDR LD digit counter or the digits do not match a digit sequence in the Local Codes table, the call is considered an LD call.
- When the SMDR storage buffer approaches full, the system sends a “Buffer full” message to the LCD of the System Attendant. The System Attendant may then delete records as necessary.
- For ease of use, Stations can be organized into SMDR Account Groups.
- ‘SLT DTMF RLS TMR’ should be adjusted to reasonable value in order to print all digits that SLT dialed.
- Internal call records include a character at the beginning of the dialed number identifying the type of call
  - E = station to station call
  - P = Page call
  - S = SMS.
- The dialed digits shown in the SMDR record can be the user-dialed digits or the digits dialed by the system after LCR conversion.
- Intercom calls, which will include Hunt group calls, SMS and Paging activity, must be separately enabled for storage and real-time output.
- Call Park will generate two SMDR records, one to park the call and one when the parked call is picked up or recalls. The duration the call remains parked is not separately provided.

Admin Programming

SMDR Account Group (PGM 124)

Assigns Stations to an account group

SMDR Attributes (PGM 177)

Defines the attributes, including call type, for SMDR

SMDR Print (PGM 142, Flexible button 17)

Enables SMDR for specific CO lines

Metering Unit (PGM 142, Flexible button 3)

Assigns the type of Metering used for a CO line

SLT DTMF RLS Timer (PGM 181, Flexible button 13)

Sets the duration that a DTMF receiver remains connected to an SLT

Local Code Table (PGM 204)

Assigns local dialing codes

Print LCR Digits (PGM 160, Flexible button 10)

Defines dialed number as user or LCR dialed digits

Off-Line SMDR (PGM 175, Flexible button 1)

Sets the port type for Off-line output of SMDR records

On-Line SMDR (PGM 175, Flexible button 6)

Sets the port type for real-time output of SMDR records

## 12.4 Traffic Analysis

---

### Description

The ipLDK can monitor, store and, periodically or upon request, output various traffic statistics covering system resources. The output from the system can be used to:

- Monitor and evaluate system performance
- Observe usage trends and recommend possible corrective actions,
- Determine possible trunk problems, i.e. blocking level too high, and/or
- Recommend system upgrades.

Attendants enable Periodic Reporting. Once enabled, the system continues to monitor and output the requested report until Periodic Reporting is disabled. On-demand reports selected by the System Attendant are output only upon request. The Traffic Report is sent to the defined system RS-232 port or Modem port.

System resources covered by Traffic Reports are:

- Attendant Traffic Report
- Call Summary Report
- Hourly Call Report
- H/W Unit Usage Summary Report
- CO Summary Report
- Hourly CO Report

Reports for each report can be printed or the All Summary Traffic Reports can be selected to print the Call, H/W and CO Summary reports. Summary Traffic Reports cover one of five Analysis periods selected at time of print:

1. Today's peak activity hour (within 24 hours)
2. Yesterday's peak activity hour (24 hours prior to Today's activity)
3. Last hour activity
4. Today's total activity
5. Yesterday's total activity

### Operation

To print the All Summary Traffic Report from the System or a Main Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 2 2', the All Summary report code.
3. Dial the hour and minute for the print
4. Using the Volume Up/Down button select the Analysis Period
5. Select Analysis Period ('1'-'5').
6. When the Analysis period displays, press the [HOLD/SAVE] button.

To cancel the periodic All Summary Report from the System or a Main Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 2 3', the cancel All Summary report code.

To print a traffic report from the System or a Main Attendant,

1. Press the [TRANS/PGM] button.
2. Dial the report code, '0 1 2 1' or '0 1 2 4' – '0 1 2 9'.
  - 0121 All Summary Traffic Reports
  - 0124 Attendant Traffic Report
  - 0125 Call Summary Report
  - 0126 Hourly Call Report
  - 0127 Hardware Usage Summary Report
  - 0128 CO/IP Summary Report
  - 0129 Hourly CO/IP Report
3. Select the Analysis Period or CO Group using the Volume controls.
4. Press the [HOLD/SAVE] button.

#### Consideration

- The output of All Summary Reports generates Attendant Traffic, Call Summary, H/W Unit Usage Summary, and CO Traffic Summary.

#### Reference

- A. Attendant Position
- B. SMDR

#### Admin Programming

RS232 Port Settings (PGM 174)

Defines characteristics of the system RS-232C port

Print Port Selection (PGM 175)

Assigns the printer port to receive the reports

### 12.4.1 Attendant Reports

#### Description

The Attendant Traffic Report covers operational statistics for the Attendants. The Traffic Report is printed periodically or on-demand when requested by the System or a Main Attendant. The following is a sample report with descriptions of the report fields.

```
Site Name : abc co
Report Type : Attendant Traffic Report- Today Peak
Date : 06/05/02 08:34
```

```
=====
ATD Meas ----- Calls ----- Time ----- Time Speed ATD
No  Hour  Total  Ans  Abdn  H-Abd  Held  Avail  Talk   Held  Noans  Ans  Type
100 13:00  104  82   22   3    18  0:12  14:21  01:23  00:52  00:23  Sys
101 13:00   90  72   15   1    11  12:43  30:12  00:54  00:23  00:21  Main
```

- | Field         | Description   |
|---------------|---|
| ▪ ATD No      | Attendant Station Number  |
| ▪ Meas Hour   | (Measurement Hour) Hour data accumulation began   |
| ▪ Calls Total | Total number of calls routed to the Attendant, not including recalls and calls that ring multiple stations unless answered by the Attendant |
| ▪ Calls Ans   | (Calls Answered) Calls answered during the Analysis period  |
| ▪ Calls Abdn  | (Calls Abandoned) Calls abandoned before answer by the Attendant, does not include calls abandoned while on hold.                           |
| ▪ Call H-Abdn | (Calls Abandoned from Hold) Calls abandoned while on hold   |
| ▪ Calls Held  | Number of calls placed on hold by the Attendant   |
| ▪ Time Avail  | (Time Available) Time attendant was available to handle new calls   |
| ▪ Time Talk   | Total time the Attendant was active on calls  |
| ▪ Time Held   | Time Attendant had calls on hold  |
| ▪ Time NoAns  | (Time No Answer) Average time calls were ringing or in queue for the Attendant before abandoned   |
| ▪ Speed Ans   | (Speed of Answer) Average time calls rang before answer by Attendant  |
| ▪ ATD type    | (Attendant Type) System, Main, or Intercom Tenancy Group Attendant  |

#### Operation

To print the Attendant Traffic Report at the System or a Main Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 2 4'.
3. Use the Volume Up/Down button to select the Analysis period.
4. Press the [HOLD/SAVE] button.

#### Consideration

- The Peak Hour is the hour when the system has the highest total call volume.

#### Reference

- A. Attendant Position
- B. SMDR

Admin Programming

RS232 Port Settings (PGM 174)

Defines characteristics of the system RS-232C port

Print Port Selection (PGM 175)

Assigns the printer port to receive the reports

### 12.4.2 Call Reports

---

Call activity statistics are provided in the Hourly Call and Call Summary Reports.

#### Hourly Call Report

The Hourly Call Report covers hourly-completed call activity for the selected Analysis period. The report indicates the number of completed calls for each hour during the Analysis period as shown below.

```

=====
Site Name : abc co
Report Type : Call Summary Report
Date      : 02/12/04 13:14
=====

```

	Analysis Start Hour	Number of Calls Completed
Last Hour	13:00	14
Today Peak	10:00	141
Yesterday Peak	10:00	119
Today Total	--:--	413
Yesterday Total	--:--	970

#### Operation

To print the Call Summary Report from the System or a Main Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 2 5'.

To print the Call Hourly Report from the System or a Main Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 2 6'.

#### Consideration

- The Peak Hour is the hour when the system has the highest total call volume.

#### Reference

- A. Attendant Position
- B. SMDR

#### Admin Programming

RS232 Port Settings (PGM 174)

Defines characteristics of the system RS-232C port

Print Port Selection (PGM 175)

Assigns the printer port to receive the reports

### 12.4.3 CO Reports

#### Description

The CO/IP Traffic Summary and Hourly reports provide statistics on a summary or hourly basis for CO/IP Group activity. The following provides a sample report and description of the major fields in the report.

```
Site Name : abc co
Report Type : CO Group Summary Report- Today Peak
Date : 09/08/02 08:34
```

```
=====
Peak Hour for All CO 11:00
Grp Num  Anal   Total  Total  Inc    Out    Grp  %    %
No  COs  Hour   Usage  Seize  Seize  Seize  Ov1  ACB  FAO
1   16  10:00  9      6      4      2      0   0   0
20  1   10:00  1      1      0      1      0   0   0
```

- | Field         | Description  |
|---------------|--|
| ▪ Grp No.     | CO/IP Group number   |
| ▪ Anal Hour   | (Analysis Start hour) hour in 24-hour mode when the analysis will start.   |
| ▪ Total Usage | Total number of call attempts on CO/IP lines in the Group  |
| ▪ Total Seize | Total number of times CO/IP lines in the group were used for any call  |
| ▪ Inc Seize   | (Incoming Seizures) Total number of incoming calls answered for CO/IP lines in the group.  |
| ▪ Out Seize   | (Outgoing Seizures) Total number of outgoing calls attempted on CO/IP lines in the group.  |
| ▪ ACB         | (All COs Busy) Percentage of the time that all CO/IP lines in the group were simultaneously busy.  |
| ▪ FAO         | (Failed Attempts Outgoing) Percentage of outgoing calls offered to the CO/IP lines in the group that were denied due to All Trunks Busy condition. |

#### Operation

To print the CO Traffic Summary Report from the System or a Main Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 2 8'.
3. Use the Volume Up/Down button to select the Analysis period.
4. Press the [HOLD/SAVE] button.

To print the CO Traffic Hourly Report from the System or a Main Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 2 9'.
3. Enter the CO group number.
4. Press the [HOLD/SAVE] button.

#### Consideration

- The Peak Hour is the hour when the system has the highest total call volume.

Reference

- A. Attendant Position
- B. SMDR

Admin Programming

RS232 Port Settings (PGM 174)

Defines characteristics of the system RS-232C port

Print Port Selection (PGM 175)

Assigns the printer port to receive the reports



#### 12.4.4 Hardware Unit Reports

---

The Hardware Usage report provides statistics for the system's special Hardware resources such as the VMIB as shown in the following sample report.

Site Name : abc co  
 Report Type : Hardware Usage Summary Report- Today Peak  
 Date : 06/05/02 08:34

```

=====
Unit Num  Anal  Total Total
Type Unit Hour  Req  Denied
VMIB  4    09:00  19   0
  
```

- | Field            | Description   |
|------------------|---|
| ▪ Type           | The type of H/W unit being measured.  |
| ▪ Number of Unit | The total number of installed H/W unit.   |
| ▪ Anal Hour      | The starting time of the last hour or the hour with the highest Peak Req. measurement.                                  |
| ▪ Total Requests | The system-wide total number of requests, by call processing for DTMF, CPTU and VMIB during the listed hour.            |
| ▪ Total Denied   | The system-wide total number of requests that are denied because there is no available H/W unit during the listed hour. |

#### Operation

To print the H/W Unit Usage Summary Report at an Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 1 2 7'.
3. Use the Volume Up/Down button to select the Analysis period.
4. Press the [HOLD/SAVE] button.

#### Consideration

- The Peak Hour is the hour when the system has the highest total call volume.

#### Reference

- A. Attendant Position
- B. SMDR

#### Admin Programming

RS232 Port Settings (PGM 174)

Defines characteristics of the system RS-232C port

Print Port Selection (PGM 175)

Assigns the printer port to receive the reports

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## 13 ATTENDANT SERVICES

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### 13.1 Attendant Date/Time Display Control

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#### Description

The LCD of the DKT and IP Phones display the date and time while the phone is idle. The date may be displayed in either the month-day-year or the day-month-year format. The time is displayed in either the 12 or 24-hour format. The format of the Time and Date display can be controlled by the Attendant or in the ipLDK system database.

#### Operation

To change LCD Date format from the Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 4 4', the date format toggles between month-day-year and day-month-year format.

To change LCD Time format from the Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 4 5', the time format toggles between 12 hour and 24 hour format.

#### Admin Programming

Intercom Tenancy Group Attendant (PGM 120, Flexible 1)

Assigns Attendant Station for the Tenancy group

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

LCD Time Display Mode (PGM 169, Flexible button 1)

Assigns the display format for the time

LCD Date Display Mode (PGM 169, Flexible button 2)

Assigns the display format for the date

## **13.2 Attendant Call & Queuing**

---

### Description

When a users dials the Attendant call code (default '0'), the system will place a call to the Attendant associated with the calling station. If the station is part of an Intercom Tenancy Group, the call is sent to the group Attendant. Otherwise, the call is sent to the first available Attendant, System or Main Attendant.

If the Attendant station is busy when a station call arrives, the call is queued to the Attendant. While queued, the caller receives ringback tone or MOH as defined in the System database. The Hold/Save button at the Attendant will flash to indicate the waiting call.

### Operation

To call an Attendant,

1. Lift handset or press [On/Off] button.
2. Dial '0', the Attendant call code.

### Consideration

- When an Attendant calls another Attendant which is busy, the calling Attendant will hear the busy tone and can Camp-on (refer to Ref. C) to called Attendant.
- If an Attendant activates Unconditional Call forward (refer to Ref. D), calls to the Attendant will follow the call forward process.
- An intercom call can be placed to a specific Attendant station by dialing the station number associated with the Attendant.

### Reference

- A. Intercom Tenancy Group
- B. MOH (Music On Hold)
- C. Camp-on
- D. Call Forward

### Admin Programming

System Numbering Plan (PGM 107, Flexible button 13)

Assigns code for Attendant calls

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

MOH Type (PGM 171, Flexible button 2)

Assigns the MOH source

Intercom Group Attendant Assignment (PGM 120, Flexible button 1)

Assigns an Attendant for an Tenant group

Attendant Call Queuing RBT/MOH (PGM 160, Flexible button 1)

Defines the tone received by a Station that calls a busy Attendant

Intercom Tenancy Group Attendant (PGM 120, Flexible 1)

Assigns Attendant Station for the Tenancy group

### **13.3 Attendant Forward**

---

#### Description

An Attendant can forward calls to other stations in the system. When the Attendant activates Unconditional Call Forward, the receiving station, if a DKT, will act as an Alternate Attendant with access to Attendant features.

#### Operation

To activate call forward,

1. Lift the handset or press the [On/Off] button.
2. Press the [DND/FOR] button.
3. Dial the Call Forward type code, '1' for unconditional Forward.
4. Dial station or Group number, or '#' for the VMIB to receive calls.
5. Return to idle.

To activate Follow-me Call Forward from any station,

1. Lift the Handset or press the On/Off button.
2. Press the [DND/FOR] button.
3. Dial Call Forward Code '0'.
4. Dial forwarding station number.
5. Dial forwarding station Authorization Code.
6. Return to idle.

To deactivate Call Forward,

1. Press the [DND/FWD] button.  
Or
2. Lift the handset or press the [On/Off] button.
3. Dial '5 5 9', the DND/FWD Cancel code.

#### Consideration

- The Considerations associated with Call Forward apply.
- If an Attendant activates Unconditional Call Forward, the receiving station will receive Attendant calls and recalls. In addition, if the receiving station is a DKT, the user will be able to activate features normally reserved for a Main Attendant. If the receiving Station is not a DKT, the station will receive Attendant calls and recalls but will not be able to activate other Attendant features.

#### Reference

- A. Call Forward
- B. Unsupervised Conference
- C. Authorization Codes
- D. DND (Do Not Disturb)

#### Admin Programming

System Numbering Plan (PGM 106, Flexible button 11)

- Defines the Call Forward activation code for SLTs
- System Numbering Plan (PGM 106, Flexible button 16)
  - Defines the code to cancel DND/FWD and Absent Message
- Call Forward (PGM 111, Flexible button 2)
  - Permits stations access to Call Forward
- Call Forward No Answer Timer (PGM 181, Flexible button 1)
  - Sets the timer for No answer Forward
- Allow Off-Net Call Forward (PGM 111, Flexible button 18)
  - Permits stations access to Off-net Call Forward
- Open Loop Detect Timer (PGM 142, Flexible button 13)
  - Sets the duration of an for disconnect supervision for analog lines
- Attendant Assignment (PGM 164)
  - Assigns System and Main Attendants
- Authorization Code Table (PGM 227),
  - Defines Authorization codes which might be required for Off-net Call Forward
- Off-net Call Mode (PGM 112, Flexible button 12)
  - Defines call type that can Forward Off-net, external and/or internal
- Unsupervised Conference Timer (PGM 182, Flexible button 16)
  - Sets the duration an Unsupervised Conference is allowed
- VMIB Access (PGM 113, Flexible button 2)
  - Permits the station access to the VMIB and a voice mailbox
- Auto Forward to VMIB (PGM 113, Flexible button 14)
  - Configures stations to forward calls automatically to the user's VMIB mail-box
- Auto Forward to VMIB timer (PGM 181, Flexible button 20)
  - Sets the time calls will forward to the user's VMIB Voice mailbox
- Intercom Tenancy Group Attendant (PGM 120, Flexible 1)
  - Assigns Attendant station for the Tenancy group

## **13.4 Attendant Intrusion**

---

### Description

When an Attendant calls a busy Station, the attendant can intrude upon the conversation and can hear, and will be heard by both parties in the conversation. A tone is available to notify the parties of the intrusion.

### Operation

To intrude on call from the Attendant while receiving busy on an intercom call,

1. Press Attendant Intrusion Flexible button.
2. After intrusion tone, conversation is possible with the station and connected party.

To assign an Attendant Intrusion Flexible button,

1. Press [TRANS/PGM].
2. Press the Flexible button to be assigned
3. Press the [TRANS/PGM] button
4. Dial '8 6'
5. Press the [HOLD/SAVE] button.

### Consideration

- The Attendant must have an Attendant Intrusion Flexible button assigned to activate intrusion.
- Auto Privacy must be disabled and the Attendant must have Override privilege for proper operation.

### Admin Programming

Override Privilege (PGM 113, Flexible button 4)

Assigns Override privilege to the Station

Auto Privacy (PGM161, Flexible button 5)

Disables Privacy

Privacy Warning Tone (PGM 161, Flexible button 6)

Enables a warning tone to indicate another Station is intruding on the call.

Intercom Tenancy Group Attendant (PGM 120, Flexible button 1)

Assigns Attendant Station for the Tenancy group

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

### **13.5 Attendant Override**

---

#### Description

An Attendant can override the DND state at another station to call or transfer a call to the station.

#### Operation

To override a station in DND from an Attendant while receiving DND tone,

1. Dial '\*’.
2. DND warning tone changes to ringback tone
3. Await answer to speak to the station user or hang-up to complete the transfer.

#### Consideration

- If the Attendant is transferring a call, the receiving station must have an available appearance (CO line or Loop button) for the CO Line to receive the transfer. Otherwise, the transfer will recall to the Attendant immediately.

#### Reference

- A. DND
- B. Basic Outbound Line Access
- C. Call Transfer, Internal

#### Administration

Intercom Tenancy Group Attendant (PGM 120, Flexible 1)

Assigns Attendant Station for the Tenancy group

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

## **13.6 Attendant Positions**

---

### Description

An Attendant receives Attendant calls and provides call coverage of common CO lines, unanswered CO recall coverage, Day/Night service control and other supervisory services. The ipLDK provides for up to five (5) stations to be assigned as a member of the Main Attendant group. The first member is the System Attendant, default is Station 100, which can be changed but not removed. In addition, each Intercom Tenancy Group can be assigned a separate Group Attendant to provide Attendant services for the group. If not assigned an Attendant, the Attendants in the Main Attendant Group provide Attendant service for the Tenancy group.

### Consideration

- Only a DKT Station should be assigned as an Attendant, an IP phone cannot be assigned as an attendant.
- A Tenancy Group Attendant only provides services to and affects the operation of CO lines and stations assigned to the Tenancy group.

### Reference

Intercom Tenancy Group

Day/Night Service

Attendant Recall

Attendant Call & Queuing

### Admin Programming

System Numbering Plan (PGM 107, Flexible button 13)

Assigns code for Attendant calls

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

Intercom Tenancy Group Attendant (PGM 120, Flexible 1)

Assigns Attendant Station for the Tenancy group



## **13.7 Attendant Recall**

---

### Description

If a CO call (transferred or held) is unanswered by the destination station in the I-Hold Recall time, the Attendant will receive the recall signal (refer to Ref. A and B). The Attendant receives the recall for the Attendant Recall Timer. If the station or Attendant does not answer the CO call after the Attendant Recall time, the CO call is disconnected.

### Consideration

- If an Attendant is not assigned for an Intercom Tenancy Group, CO Lines will recall to a Main Attendant.
- When a call on Exclusive Hold recalls the Attendant, the call is placed on System Hold (refer to Ref. A).
- Private CO Line calls do not be recall to attendant (refer to Ref. D).

### Reference

- A. Hold
- B. Call Transfer
- C. Intercom Tenancy Group
- D. Private Line

### Admin Programming

Intercom Tenancy Group Attendant (PGM 120, Flexible 1)

Assigns Attendant station for the Tenancy group

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

Attendant Recall Timer (PGM 180, Flexible button 1)

Sets the time the Attendant receives recall before the call is disconnected

Hold Recall Timer (PGM 180, Flexible button 5)

Sets the duration the holding station receives recall before the Attendant also receives recall

## **13.8 CO Call Forward Off-Net**

---

### Description

An Attendant can direct the system to re-route (forward) incoming CO line calls to a directory (telephone) number outside the system assigned to a Speed Dial. When a call is received on the CO line, the system immediately accesses an outgoing CO line and dials the telephone number from the Speed Dial number. The system then establishes an Unsupervised Conference between the CO lines. The system will automatically disconnect the call if an open loop is detected based on the Open Loop Detect timer or after the Unsupervised Conference Timer has expired.

### Operation

To activate incoming CO Call Forward Off-Net at the Attendant,

1. Lift the handset or press the On/Off button.
2. Press the [DND/FOR] button.
3. Dial Off-Net Call Forward code, 7.
4. Access the desired CO line or Group.
5. Dial the Speed Dial bin number with the desired telephone number.
6. Replace the Handset and go on-hook.

To deactivate CO Call Forward Off-Net from the attendant,

1. Lift the Handset or press the [MON] button.
2. Press the [DND/FOR] button.
3. Dial Call Forward code 7.
4. Access the desired CO line or Group.
5. Dial #.

### Consideration

- Considerations associated with Call Forward and Unsupervised Conference apply.
- If an idle CO Line is not available to place the outgoing CO line call, Off-net Call Forward is not activated.
- DID lines are not subject to CO Call Forward Off-Net.
- When activating CO Call Forward Off-Net, a Speed Dial Flexible button may be used in place of dialing the Speed bin.
- Toll restriction will be based on the COS of outgoing CO Line.

### Reference

- A. Call Forward
- B. Unsupervised Conference
- C. Basic Outbound Access

### Administration

System Numbering Plan (PGM 109, Flexible button 6)

Assigns the Unsupervised Conference Timer Extend Code

Intercom Tenancy Group Attendant (PGM 120, Flexible 1)

Assigns Attendant Station for the Tenancy group

Open Loop Detect Timer (PGM 142, Flexible button 13)

Sets the duration of an open loop for disconnect supervision on analog lines

Unsupervised Conference Timer Extend Enable (PGM 160, Flexible button 18)

Enables extending an Unsupervised Conference

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

Unsupervised Conference Timer (PGM 182, Flexible button 6)

Sets the Timer for an unsupervised CO to CO connection

### 13.9 Day/Night Service

---

Description

The ipLDK provides for five (5) service modes; the active mode determines CO line ring destinations and DISA operation as well as the COS for an outgoing call. CO line ring and DISA are assigned for Day, Night, Lunch, Weekend and On-Demand service. COS assignments are made for Day and Night service; Day COS assignments are employed for Day and Lunch service, Night COS is employed for Night and Weekend service.

Weekly Time Tables can be set allowing the system to determine automatically the service mode. A table is available for the system, and separate tables are available for each Intercom Tenancy Group. In addition to the Automatic Service mode, the Attendant can change the active service mode at any time. The System and Main Attendants control the service mode system wide. An Intercom Tenancy Group Attendant controls the service mode only within the specific Tenant Group. The Attendant can select from any of the five (5) service modes or select the Automatic Service.

Operation

To change Service mode from an Attendant,

1. Press the [DND/FOR] button at the Attendant Station.
2. Dial the desired service mode digit.  
1: Day, 2: Night, 3: Weekend, 4: On-demand, 5: Auto Service Mode, 6: Lunch Mode
3. Press the [HOLD/SAVE] button.

Consideration

- The default Weekly Time Table is shown below.

Day of Week	Day Start Time	Night Start Time	Weekend Start Time	Lunch Start Time	Lunch End Time
Mon	09:00	18:00	-- : --		
Tue	09:00	18:00	-- : --		
Wed	09:00	18:00	-- : --		
Thu	09:00	18:00	-- : --		
Fri	09:00	-- : --	18:00		
Sat	-- : --	-- : --	00:00		
Sun	-- : --	-- : --	00:00		

- If a System or Main Attendant changes the active mode to the Day service mode from another mode, the service mode of all Intercom Tenancy Groups changes to the previously active mode.

Reference

- A. Ring Assignment
- B. COS
- C. External Control
- D. Intercom Tenancy Groups

- E. Attendant Positions
- F. Loud Bell
- G. UNA
- H. DISA
- I. DID

Admin Programming

Intercom Tenancy Group Attendant (PGM 120, Flexible 1)

Assigns Attendant Station for the Tenancy group

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

Weekly Time Table (PGM 233)

Sets the times of day for Automatic Service Mode operation

## 13.10 DSS/DLS Consoles

---

### Description

The Attendants and other DKT stations may be equipped with DSS/DLS Consoles that provide an additional 48 Flexible buttons each. The buttons function as any other Flexible button on the associated DKT. Up to three (3) Consoles may be equipped for each station. Each console requires a separate connection to the ipLDK and reduces the available stations by one for each console.

The buttons on the console are fully programmable and provide the user with one-button access to the assigned CO line, station or feature for more convenient operation. Each button has associated LEDs to display status for the CO line, station or feature. One of three templates or Maps for the Flexible buttons is assigned for the Console providing basic configuration of the buttons. The buttons can then be individually assigned a desired function other than the template function.

### Consideration

- There is no limit to the number of DSS/DLS consoles in the system other than the maximum number of DTK stations.
- The default templates (maps) available for the Console are shown in the chart below.

<u>Map 1</u>	Flex 1 - Intrusion
	Flex 2 - All Call Page
	Flex 3 - Call Park 01
	Flex 4 - Station Group 1
	Flex 5 - Camp-on
	Flex 6 - Internal All Call Page
	Flex 7 - Call Park 02
	Flex 8 - Station Group 2
	Flex 9 - Group Call Pick-up
	Flex 10 - External All Call Page
	Flex 11 - Call Park 03
	Flex 12 - Station Group 3
	Flexible buttons 13 to 48 - Stations 100-135
<u>Map 2</u>	Flexible buttons 1 to 12 – Stations 136 to 147
<u>Map 3</u>	No default assignments

### Reference

- A. Flexible Buttons

### Admin Programming

Station ID Assignment (PGM 110, Flexible button 1),

Assigns the type of terminal

DSS/DLS ID Assignment (PGM 110, Flexible button 2)

Assigns a Console and a Console button template to a Station

### **13.11 Door Phone BGM Control**

---

#### Description

A Door Phone box can play BGM while idle. An Attendant can select the source music channel sent to the Door Phone.

#### Operation

To select the music source from the Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 7 5'.
3. Dial the music source (00-08).
  - Channel 00: Music is not used.
  - Channel 01: internal music
  - Channel 02: External music
  - Channel 03: Not available
  - Channel 04 ~08: SLT MOH (see Ref. B)
4. Press the [HOLD/SAVE] button.

#### Reference

- A. BGM (Background Music)
- B. MOH (Music on Hold)
- C. Door Phone Box

#### Admin Programming

Intercom Tenancy Group Attendant (PGM 120, Flexible 1)

Assigns Attendant Station for the Tenancy group

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

## **13.12 Outgoing Access Control**

---

### Description

An Attendant can take a CO line out-of-service for outgoing calls. When disabled, calls cannot be placed on the CO line however, incoming CO calls are not affected.

### Operation

To set a CO line in/out-of- service for outgoing calls from the Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 7 3'.
3. Press the desired CO line button; confirmation tone is received and the status of then selected CO line toggles between in and out-of-service>
4. Hang-up to return to idle.

### Consideration

- The LED of a disable CO line will flash rapidly at the Attendant station and will be steady 'On' at all other stations with an appearance of the CO line.
- The Attendant can change the outgoing service status of a CO line while the line is busy or idle.

### Reference

- A. Basic Outgoing Line Access

### Administration

Attendant Assignment (PGM 164)

Assigns System and Main Attendants



### **13.13 Station Feature Cancel**

---

#### Description

An Attendant can cancel features - such as DND, Call Forward and Pre-selected messages – active at other stations (refer to Ref. A, B, or C).

#### Operation

To disable active features at a station from an Attendant,

1. Press the [TRANS/PGM] button.
2. Dial '0 7 1'.
3. Dial the desired station range.
4. Press the [HOLD/SAVE] button.

#### Reference

- A. DND
- B. Call Forward
- C. Absent Text Message

#### Administration

Intercom Tenancy Group Attendant (PGM 120, Flexible 1)

Assigns Attendant Station for the Tenancy group

Attendant Assignment (PGM 164)

Assigns System and Main Attendants

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## 14 ISDN SERVICE

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### Description

ISDN (Integrated Services Digital Network) is a digital network service providing digital audio and control. In ISDN, control signaling is provided over a 'D' (data) channel and voice is provided over a separate 'B' (Bearer) channel. Separating the signaling and content channels provides more efficient control and bandwidth.

The ipLDK-60 supports the ISDN Primary Rate Interface (PRI). This interface provides high-speed digital services for multiple bearer channels. In the ETSI standard, 30 bearer channels, 1 framing channel and a signaling channel are provided. In NA standards, 23 bearer channels and 1 signaling channel is provided.

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### 14.1 CLI Message Wait

---

#### Description

When a call is received with Caller or Calling Line ID, the Id is displayed in the DKT LCD as well as the LCD of compatible SLTs. If the call is an ISDN line DID call and disconnects before the called station can answer, the system can activate a CLI Message Wait for the called station. The DKT LCD shows the CLI Message and the [Call Back] button LED flashes. The user may view the CLI Messages, return the call, save the CLI number as a Speed Dial or simply delete the CLI Message.

#### Operation

To view CLI Messages,

1. Press the [Call Back] button to display the first message.

0314504626 DATE TIME CNT:xx
--------------------------------

2. Use the Volume Up/Down buttons to scroll through messages.

To manage CLI Messages,

1. Press the Select softkey to select a CLI Message.
2. To call the number, press the Hold/Save button.  
To delete the current CLI Message, press Del Cur softkey.  
To store the number as a Speed Dial, press the Save softkey or.  
To delete all CLI messages press the DND/FOR button.

To delete all CLI Message Waits from the Attendant,

1. Press the TRANS/PGM button.
2. Dial '0 5 5'.
3. Dial the station range.
4. Press the Hold/Save button.

Consideration

- CLI Message Wait is available for calls to a DKT w/LCD from ISDN or VoIP DID lines only. The station must have CLI Message Wait and CLIP Display enabled.
- The system can store 500 CLI Messages and, should the buffer fill, the system will output the buffer full message on the RS-232 port.
- CLI Messages are saved even if a power failure occurs.
- When directed to a Ring Group the first member of the group receives the CLI Message. For Hunt Groups, the first station that rang for the call receives the CLI Message Wait.
- CLI Message Wait is activated when a station or station of a Group receives ring for the call. If a VMIB announcement is assigned, CLI Message Wait is available after the announcement.
- If the call encounters Call Forward, the station activating Call Forward receives the CLI Message Wait.
- If a Station Speed Dial bin is assigned with the CLI number, the name associated with the Speed Dial can be displayed using the [Call Back] button.
- If the caller leaves a VMIB Message, a CLI Message is still activated. In this case, the user cannot delete all CLI Messages; the VMIB Message must be deleted first.
- Messages waiting are delivered to the user in the following priority:
  - VMIB Message -> CLI Message -> VM Grp Message
- If a station receives multiple CLI Messages from the same caller, the system will save a single record with the number of messages (max 15) and the time of the last message was left by the caller.
- The SMDR output can be configured to include both the CLI and station number.

Reference

- A. Incoming Call ID Service
- B. Station Message Wait/Call Back
- C. VMIB Voice Mail
- D. External VM
- E. SMDR
- F. Station Hunt Groups

Admin Programming

CLIP Display (PGM 114, Flexible button 1)

Enables the Station display of Calling Line Id presented by the ISDN

CLI Name Display (PGM 114, Flexible button 11)

Enables displaying the CLI Name when matched to a Station Speed Dial

CLI Message Wait (PGM 114, Flexible button 4)

Enables the Station to receive CLI Message Waiting

## 14.2 CLI Transit Service

Under certain situations, the ISDN Station Attributes are not appropriate for the ipLDK to determine the outgoing CLI. In particular, calls placed on ISDN CO lines

- by a DISA caller,
- as a result of DID Off-net forward,
- using Walking Class of service,
- requiring an Authorization code (Forced Station Account Code, CO Account Code or LCR Password)

may require special handling of outgoing CLI.

For DISA calls, the system determines the Station ISDN CLI parameters, if required, based on the CLI Transit type assigned for the outgoing line and the type of Authorization code entered as shown in the chart below.

CLI Transit Type	DISA Acct Enabled	Auth Code Type Entered	ISDN Attributes
CFW (Call Fwd)	Yes	Station	Use attributes of station associated with the Authorization code
CFW (Call Fwd)	Yes	System	Use attributes of the System Attendant station
CFW (Call Fwd)	No		Use attributes of the System Attendant station
ORI (Original)			Use CLI of original external caller

For a DID call, which is rerouted Off-net, the system will determine Call ID based on the CLI Transit setting. The CLI used will be either the original CLI received by the system, or using the ISDN attributes associated with the System Attendant station.

For ISDN CO line calls placed using Walking Class of Service or requiring the entry of an Authorization code, the system will use the attributes of the station associated with the Authorization code or, for a system Authorization code, the attributes of the System Attendant station.

If CLI Transit is set to CFW, the system generates the CLI as discussed in section 14.3. The CLI Transit can be configured to employ the Call ID received by the ipLDK or based on the type of Authorization code (System or Station)

### Consideration

- If Calling/Connected Line Identification Restriction (refer to REF A.) is enabled for the station and is supported by the ISDN, the system will send the restriction message in place of the CLI or COL.
- If CLI Transit is set to CFW, CLI is generated as discussed in Reference B.

### Reference

- A. Calling/Connected Line Identification Restriction DISA
- B. Outgoing Call ID Service
- C. DISA
- D. DID
- E. Walking Class of Service
- F. Least Cost Routing (LCR)

Administration

CLI Transit Type (PGM 143, Flexible button 7)

Defines the type of CLI treatment for transit calls

### **14.3 CLIP/COLP (Calling/Connected Line Identification Presentation)**

---

#### Description

ISDN CO lines present Calling Line Identification (CLI) for incoming calls in the Call Set-up message to the ipLDK, identifying the incoming caller. These CO lines also provide Connected Line Identification (COL) for outgoing calls in the Call Connect message to the ipLDK, identifying the answering caller. ipLDK will modify the number based on the system configuration as discussed in sections 14.1.1 and 14.1.2.

#### 14.3.1 Incoming Call ID Service

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##### Description

When CLIP or COLP is received on an ISDN CO line, the system modifies the Call ID based on entries in PGM 200, by removing the assigned International code, My Area Code and My Area Prefix Code from the number. The resulting CID is used for CLI Message Wait and SMDR records if CLI Print is enabled. In addition, the LCD of CLI Display enabled stations (DKT and SLT w/CID) will be sent the modified CLI and/or name if received and configured.

If CLI Name Display is enabled for a station, the system will compare the modified CLI for an incoming call with the Station Speed Dial numbers, if a match is found, the name associated with the Station Speed Dial, if any, is displayed.

The CLI number and/or name display is provided while the line is ringing or displayed during ringing and the duration of the call once answered based on the 'Maintain DID Name' characteristic.

##### Consideration

- A maximum of 12 digits can be displayed for CLI.

##### Reference

- A. CLI Message Wait:
- B. SMDR:

##### Admin Programming

CLIP Display (PGM 114, Flexible button 1)

Enables display of CLI for incoming calls to a station

COLP Display (PGM 114, Flexible button 2)

Enables display of COL for outgoing calls from a station

CLI Name Display (PGM 114, Flexible button 11)

Enables display of matching Speed Dial name for a station

CLI Print (PGM 200, Flexible button 6)

Enables recording of CLI for the ISDN CO line

Incoming Prefix Code Insertion (PGM 146, Flexible button 1)

Assigns the insertion of My Area Prefix code in front of the CID body

CID Name/Number (PGM 147, Flexible button 2)

Determines the displayed CID information (Name and/or Number)

International Access Code (PGM 200, Flexible button 7)

Assigns International Access Code for the system, maximum 4 digits

My Area Code (PGM 200, Flexible button 9)

Assigns the Area Code for the system, maximum 6 digits

My Area Prefix Code (PGM 200, Flexible button 10)

Assigns a prefix code for the system, maximum 4 digits

Maintain DID Name (PGM 200, Flexible button 11)

Enables maintain the DID name on the LCD

### 14.3.2 Outgoing Call ID Service

---

#### Description

When a station of ipLDK system places or answers an ISDN CO line the system will generate the CLI or COL, as appropriate, based on the Station ISDN Attributes, ISDN CO Line Attributes, the System ISDN Attributes and COLP/CLIP table. The generated ID will be placed in the appropriate Call Set-up or Call Connect message back to the ISDN.

Each ISDN CO line is assigned a CLI Type, Normal (discussed later) Long CLI1 or Long CLI2. For Long CLI1 and Long CLI2, the Long CLI1 or Long CLI2 assigned for the station make-up the body of the Call ID. Based on the Call Type assigned to the line, the system will add the assigned International Dial Code and My Area Code to the front of the ID. In addition, if assigned for Outgoing Prefix Code Insertion for the ISDN CO line, My Prefix Code will also be appended to the ID. The prefix code is enabled separately for incoming (COLP) and outgoing (CLIP) calls.

When the CLI Type for the line is 'Normal' and the COLP and CLIP Table Index for the ISDN CO line is unassigned (5 0), the system will determine the ID from the Station ISDN Attributes. The system will generate the CID using the extension (EXT) or CO line Attendant (ATD). When assigned for EXT, the station CLI Type is set for Short or Long CLI. For Short CLI, the assigned ISDN CLI Station number (up to 4-digits) is used and for Long CLI the Long Sta CLI (up to 12 digits) is used. The body of the ID is then determined as the EXT or ATD number. Based on the Call Type assigned to the line, the system will add the assigned International Dial Code and My Area Code to the front of the ID. In addition, if assigned for Outgoing Prefix Code Insertion for the ISDN CO line, My Area Prefix Code will also be appended to the ID. The prefix code is enabled separately for incoming (COLP) and outgoing (CLIP) calls.

#### Consideration

- If Calling/Connected Line Identification Restriction (refer to REF A.) is enabled for the station and is supported by the ISDN, the system will send the restriction message in place of the CLI or COL.

#### Reference

- A. CLIR/COLR

## B. CLI Transit Service

### Admin Programming

EXT/CO ATD (PGM 114, Flexible button 5)

Assigns use of the station (EXT) or Attendant number for the CLI body

Station CLI Type (PGM 114, Flexible button 18)

Selects Short or Long CLI when EXT is assigned

ISDN CLI STA (PGM 114, Flexible button 12)

Assigns the CLI body digits when the station CLI type is assigned as Short

Long STA CLI (PGM 114, Flexible button 19)

Assigns the body of the CLI when the station CLI type is LONG

Long CLI1 (PGM 114, Flexible button 21)

Assigns the CLI body for the station when the line CLI type is Long CLI1

Long CLI2 (PGM 114, Flexible button 22)

Assigns the CLI body for the station when the line CLI type is Long CLI2

Line CLI Type (PGM 143, Flexible button 12)

Defines the type of CLI for the ISDN line

COLP Table Index (PGM 143, Flexible button 1)

Defines an index to the COLP table for use as the CID for the line

CLIP Table Index (PGM 143, Flexible button 2)

Defines an index to the CLIP table for use as the CID for the line

Call Type (PGM 143, Flexible button 3)

Assigns the type of calls (International, National, subscriber) allowed on the ISDN line

Outgoing Prefix Insertion (PGM 146, Flexible button 2)

Assigns the insertion of My Area Prefix code in front of the CID body

International Access Code (PGM 200, Flexible button 7)

Assigns International Access Code for the system, maximum 4 digits

My Area Code (PGM 200, Flexible button 9)

Assigns the Area Code for the system, maximum 6 digits

My Area Prefix Code (PGM 200, Flexible button 10)

Assigns a prefix code for the system, maximum 4 digits

CLIP/COLP Table (PGM 201)

Assigns the body of the CID for the line using the defined CLIP/COLP Table index



## **14.4 CLIR/COLR (Calling/Connected Line Identification Restriction)**

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### Description

The system will send calling and connected party identification to the ISDN in the appropriate messages. Identification messages may be restricted, not reported, to the far-end user. Calling Line Identification Restriction and Connected Line Identification Restriction may be enabled in the system or the user may assign Flexible buttons for restricting CLIP and COLP on a per call basis.

### Operation

To activate CLIR or COLR, before placing or answering an ISDN call

1. Press the {CLIR} or {COLR} Flexible button.

### Consideration

- This feature may not be available in the specific ISDN service area or may be a subscription service.

### Reference

- A. Outgoing Call ID Service
- B. User Station Programming
- C. Flexible Buttons

### Admin Programming

ISDN CLIR (PGM 114, Flexible button 14)

Restricts sending CLI when the station places an ISDN call

ISDN CLOR (PGM 114, Flexible button 15)

Restricts sending COL when the station answers an ISDN call

## **14.5 Incoming Calling Line Id (ICLID) Routing of DID Calls**

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### Description

When a call is received on a line assigned for DID service, the ipLDK will compare the CLI with entries in the ICLID Table. The ICLID Table has two entries for each record, the CLI and an index to the Flexible DID Table for routing. If the received CLI matches an entry in the ICLID Table, the system routes the call using the destination defined for the index in the Flexible DID Table. If no match is found, the system will route the call using the normal DID call procedures (refer to Ref A).

### Consideration

- The ICLID Table is automatically sorted by CLI. When a new CLI and index are saved, the table is resorted. If an existing CLI is entered into the table, it will replace any previous entry of that CLI.
- If a CLI is deleted from the Table, the Flexible DID Table index is also cleared.
- The ICLID Table can be initialized using PGM 450, Flexible button 11 Initialization Other Tables

### Reference

- A. DID
- B. Multiple Subscriber Number (MSN)

### Admin Programming

CO Line Service Type (PGM 140, Flexible button 1)

Assigns the type of service for the line

ICLID Usage (PGM 143, Flexible button 18)

Enables use of ICLID routing of incoming calls

ICLID Table (PGM 237)

Assigns CLI and Flexible DID Table index

Flexible DID Table (PGM 231)

Assigns destination for a call using the table index

## **14.6 ISDN Supplementary Services**

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### Description

In many cases, the ISDN service provider offers enhanced services, to which a user may subscribe. ipLDK-60 allows access to Call Deflection or Call Reroute ISDN “Supplementary Service” implemented under the ETSI regime as described below.

#### 14.6.1 ISDN Call Deflection

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### Description

When the ISDN Supplementary Service “Call Deflection” is supported on an ISDN line, a user can forward incoming calls on the ISDN line directly without the call being offered to the ipLDK user.

When Call Deflection is used, the system sends a call-deflection request message with the telephone number to receive calls to the ISDN. The ISDN then sends all incoming calls unconditionally to the desired telephone number until the user deactivates the forward.

### Operation

To activate ISDN Call Deflection to an external number from the station using a Speed Dial,

1. Lift the handset or press the On/Off button.
2. Press the [DND/FOR] button or dial the Forward code ‘5 5 4’.
3. Dial the Call Forward type code 5 or 6.
4. Seize a CO line, optional.
5. Dial the Speed Dial bin.
6. Return to idle.

To activate ISDN Call Deflection to an external number from the station using a Speed Dial,

1. Lift the handset or press the On/Off button.
2. Press the [DND/FOR] button or dial the Forward code ‘5 5 4’.
3. Dial the Call Forward type code 8 or 9.
4. Seize a CO line, optional.
5. Dial the desired telephone number.
6. Return to idle.

To deactivate ISDN Call Deflection from the station,

1. Press the flashing [DND/FWD] button.

Or

1. Lift the handset or press the [On/Off] button.
2. Dial 5 5 9, the DND/FWD Cancel code.

### Consideration

- The Call Deflection Supplementary Service is defined by the ETS300-202/206/207 standards and must be supported by the ISDN.

- ISDN lines that support Call Deflection must be enabled for ISDN Call Deflection in the ipLDK database.
- The ISDN may limit the number of Call Deflections for a call, if the call has been deflected before being offered to the ipLDK, the ISDN may reject the deflection request. Generally, the ISDN will permit up to five deflections for a call.

#### Reference

- A. Call Forward
- B. Station Speed Dial
- C. System Speed Dial
- D. ISDN Supplementary Services

#### Admin Programming

ISDN-SS CD/CR (PGM 143, Flexible button 9)

Enables Call deflection or Reroute for calls to the station on the ISDN line

### 14.6.2 ISDN Call Reroute, CFNR (Call Forward No-response)

---

#### Description

When the ISDN Supplementary Service “Call Reroute” is supported on an ISDN line, a user can forward incoming calls on the ISDN line to a different line over the ISDN without using ipLDK resources. The ISDN call is offered to the ipLDK user and if no response is received in the ISDN’s no-response timer, the ISDN re-routes the call.

When Call Reroute is used, the system sends a call-reroute request message with the telephone number to receive calls to the ISDN. The ISDN then sends incoming calls to the desired telephone number after the no-response timer. Call Reroute remains active until the user deactivates the forward.

#### Operation

To activate ISDN Call Reroute to an external number from the station,

1. Lift the handset or press the [On/Off] button.
2. Press the [DND/FOR] button or dial the Forward code ‘5 4 4’.
3. Dial the Call Forward type code 9.
4. Seize a CO Line, optional.
5. Dial the Speed Dial bin or telephone number.
6. Return to idle.

To deactivate ISDN Call Reroute at the station,

1. Press the flashing [DND/FWD] button.

Or

1. Lift the handset or press the [On/Off] button.
2. Dial ‘5 5 9’, the DND/FWD Cancel code.

#### Consideration

- The Call Deflection Supplementary Service is defined by the ETS300-201/205 standards and must be supported by the ISDN.
- ISDN lines that support Call Reroute must be enabled for ISDN Call Reroute in the ipLDK database.
- The ISDN may limit the number of Forwards for a call, if the call has been deflected before being offered to the ipLDK, the ISDN may reject the Forward request. Generally, the ISDN will permit up to five deflections for a call.
- The ISDN no-response timer is not controlled by the ipLDK. Other timers such as the Preset Forward and VMIB Forward timers may interfere with the ISDN CFNR. It may be necessary to modify or disable these features for operation of ISDN CNFR.

### Reference

- A. Call Forward
- B. Preset Call Forward
- C. Auto VMIB Forward
- D. Station Speed Dial
- E. System Speed Dial
- F. ISDN Supplementary Services

### Admin Programming

ISDN-SS CD/CR (PGM 143, Flexible button 9)

Enables Call deflection or Reroute for calls to the station on the ISDN line

## 14.7 Keypad Facility

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### Description

The ISDN Keypad Facility Information Element (IE) may enable the user to activate certain ISDN services (e.g. Off-Net Forward). To access this facility, the station must be allowed Off-Net Forward and have a {Keypad Facility} Flexible button. When activated, the digits dialed by a user are sent in the Keypad Facility IE instead of the Called Party Number IE.

### Operation

To assign a Flexible button for ISDN {Keypad Facility} operation,

1. Press the [TRANS/PGM] button.
2. Press the desired Flexible button.
3. Press the [TRANS/PGM] button.
4. Dial '8 9'.
5. Press the [HOLD/SAVE] button.

To activate the ISDN Keypad Facility while on an ISDN line,

1. Press the {Keypad Facility} button.
2. Dial the desired digits. Other actions including pressing the {Keypad Facility} button disable Keypad facility feature.

### Consideration

- This feature can be activated or deactivated only after an ISDN CO line is seized.
- Once activated, the system will continue to send dialed digits as Keypad Facility IE messages regardless of ISDN messages. Thus, in the connected mode, DTMF tones are not sent to the connected party, only the keypad message is sent.
- Selecting another feature button (Hold, Speed Dial, Redial, etc.), accessing another CO line, or receiving a Called Party number Information Element from the ISDN will terminate ISDN Keypad Facility.
- If a Speed Dial bin includes a [FLASH] as the first entry, the remaining digits in the Speed Dial number will be sent in a Keypad Facility IE. The '#' is not included in the SMDR call record.
- This feature may not be available in the specific ISDN or may be a subscription service.
- The system can handle only a single Call Reference; services that require handling of two simultaneous Call References, cannot be supported.

### Reference

- A. User Station Programming
- B. Flexible buttons
- C. Station Speed Dial
- D. System Speed Dial

### Admin Programming

CO line Type

Assign the type of CO line

Keypad Facility (PGM 114, Flexible button 6)

Enables/disables Keypad facilities at stations

## **14.8 Multiple Subscriber Number (MSN)**

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### Description

MSN enables assigning multiple subscriber numbers to one ISDN Interface, allowing Direct-Dial-In like call delivery. In addition, Sub-Address information may be included as additional digits in the Called Party Number Information Element allowing for similar call routing by the ipLDK. When MSN or Sub-address information is received, the system references the MSN Table for an index to the Flexible DID Table, which then determines call routing.

A station may be assigned an MSN button. If a call is received for the assigned MSN, the station receives ring and the MSN button will act as the call appearance. In addition, if assigned, the user will receive call waiting for the MSN call

### Consideration

- If the Called Party Number from the ISDN does not match an MSN Table entry, the received digits are treated as defined for DID calls to determine routing of the call.
- The MSN Table employs the CO Line entry as a quick look-up reference. If the CO line number is programmed in the MSN Table, then only those entries with the CO line number are searched, otherwise the entire MSN Table is searched for a match to the Called Party Number.
- The ipLDK allows the sub-address to be delivered in the Called Party Number (CPN) or in the CPN Sub-address field of the Call Set-up message. The sub-address delivery is defined in the S<sub>0</sub> Sub-address for each station.
- The sub-address cannot exceed a single digit, 0 – 9.
- Up to 250 entries can be made in the MSN Table.
- To use an ISDN CO line with MSN, the CO Service type should be set to DID/MSN.

### Reference

- A. DID
- B. ICLID Routing

### Admin Programming

CO Line Service Type (PGM 140, Flexible button 1)

Assigns the type of service for the line, must be ISDN DID/MSN

MSN Wait (PGM 114, Flexible button 20)

Enables call wait for MSN calls to the station when busy

ISDN Calling Sub-address (PGM 146, Flexible button 4)

Enables sending the sub-address of the ISDN station when placing a call

MSN Table (PGM 202)

Assigns an index to the Flexible DID Table (PGM 231) for call routing



## **14.9 Prefix Code Dialing Service (Ukraine)**

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### Description

When a user dials a number, the system compares the digits with the Enblock Code Table, if a match is found; the call is processed based on the Enblock Code Table instructions.

### Consideration

- Prefix code dialing only applies to ISDN lines.
- In the Enblock Code Table, the Prefix code entry can include any digit '0 ~ 9' or 'D' (DND button) as a wild card allowing any digit.
- If Sending Complete (Enblock mode) is enabled, the system will wait to send the ISDN Call Set-up message until the user dials the maximum allowed digits or the Enblock Inter-digit timer expires.

### Admin Programming

Prefix Table Index (PGM143, Flexible 16)

Enblock Prefix Table (PGM 205)

Assigns Prefix codes and call processing instructions

Enblock Inter-digit Timer (PGM 182, Flexible 10)

Sets the maximum time the system will wait between user dialed digits

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## **15 VOIP SERVICE**

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### Description

The ipLDK-60 can be equipped with up to 8 IP channels to support remote LG-Ericsson IP Phones, Phontage (LG-Ericsson IP softphone) and IP CO lines for networking (refer to Ref A) or trunking. Up to 16 IP Phones, which include Phontage softphones, can be registered with an ipLDK-60. Each active call from/to an IP Phone or IP CO line requires a VOIB IP channel. When all channels are busy, such calls will receive fast-busy tone as an indication of all channels busy.

IP CO lines support both H.323 and SIP (Session Initiation Protocol) to establish, maintain and terminate IP calls. The ipLDK supports both SIP Trunking to provide SIP access as a normal CO line or SIP Register for incoming call routing to specific authenticated users.

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### **15.1 H.323**

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#### Description

H.323 is an IEEE standard for multimedia communications “over an unreliable network”. In particular, H.323 is intended for use over IP networks where delivery of individual packets is not guaranteed. In H.323, terminals can register with a GateKeeper for communication services; the GateKeeper then manages the services to the terminal. H.323 employs several other IEEE standards including H.245 for control signaling and H.225 for Call and RAS channel signaling.

ipLDK operates in two modes employing H.323 with a GateKeeper for H.323 trunking and without the GateKeeper for Private Networking (refer to Ref. A). How to configure both modes of H.323 operation are provided; refer to section 15.1.1 for guidelines to configure operation with a GateKeeper. Private Network configuration using H.323 is provided in section 16.7.2. In addition, ipLDK supports several options to improve efficiency and reduce connection set-up times. Options supported include:

- **Fast Mode**  
ipLDK can establish a media channel using H.245 or the H.323 Fast Mode.
- **Early H.245**  
ipLDK implements the early H.245 option; the H245 channel address is provided in the call Setup message hence speeding call setup.
- **H.245 Tunnel**  
ipLDK implements the H.245 tunneling for control and security by encapsulating H.245 in Q.931 messages

#### Consideration

- The TOS byte of the IP packet can be defined in the ipLDK to adjust precedence of voice communication packets.

- Outgoing H.323 calls must use Enblock signaling; the ipLDK will wait for the user to dial '#' or the Enblock Timer setting before setting up an H.323 call.
- MSN tables, COLP, CLIP, CLI Message Wait, etc. apply to VOIB CO lines.
- VOIB CO lines can be used to support mobile extensions.
- A H.323 line can be shared by multiple ipLDK-60 users as a common ('Normal') line.
- The system can be equipped with eight (8) VoIP channels that are used to support IP Trunks, IP Phones and Phontage IP softphones and may be used to support Private Networking. Assure sufficient channels are available to support the desired channels and expected traffic.
- VOIB CO lines may not be compatible with E911 systems; the user is advised to arrange appropriate emergency services.
- It may be necessary to configure stations for access to the assigned CO line group and Flexible buttons for one-touch access.

### Reference

#### A. Private Networking

### Admin Programming

Normal/Fast Mode (PGM 340, Flexible button 18)

Assigns normal or Fast mode call set-up for H.323

Early H.245 (PGM 340, Flexible button 19)

Enables Early mode for H.245 channel set-up

H.245 Tunneling (PGM 340, Flexible button 20)

Enables tunneling for the h.245 channel

Precedence value(PGM 340, Flexible button 21)

Defines the H.323 TOS byte for precedence

#### 15.1.1 Configuring H.323 GateKeeper Mode

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### Description

The following is a quick guide to configuration for operation with an H.323 GateKeeper,

- Set CO Service Type for 'Normal' or 'DID' service (PGM 140)
- Set the CO line group for the VOIB CO lines (PGM 141)
- Set ISDN line characteristics (PGM 143)  
including CLIP and COLP
- Set ISDN Call Type as desired (PGM 143)
- Set ISDN Enblock Send 'On', all IP calls are Enblock (PGM 143)
- Set DID CONV Type, Digit Mask, etc., if required, (PGM 143)
- Set Ring Assignments, if required (PGM 145)
- Set the Enblock Timer, if required (PGM 182)
- Configure DID Tables, if required, (PGM 167 and 231)
- Set Networking CO Line Type as H.323 (PGM 322).
- Configure VOIB IP settings, as required (PGM 340)  
IP addresses, VOIB, subnet mask, Gateway, DNS  
Set VOIB Mode to H.323

- Enter Firewall WAN address, if required
- Set basic H.323 parameters (Fast mode, Early H.245, Tunneling, H.323 TOS)
- Set GateKeeper parameters (PGM 341)
  - Enable GateKeeper Usage
  - Gatekeeper IP Address
  - GateKeeper Call Mode as GK Route for GateKeeper routing
  - Enable GK Open 245
  - For remainder, check with the GateKeeper provider

Reset after programming to assure channel sync and registration

### Consideration

- The TOS byte of the IP packet can be defined in the ipLDK to adjust precedence of voice communication packets.
- Outgoing H.323 calls must use Enblock signaling; the ipLDK will wait for the user to dial '#' or the Enblock Timer setting before setting up an H.323 call.
- MSN tables, COLP, CLIP, CLI Message Wait, etc. apply to VOIB CO lines.
- VOIB CO lines can be used to support mobile extensions.
- An H.323 line can be shared by multiple ipLDK-60 users as a common ('Normal') line.
- The system can be equipped with eight (8) VoIP channels that are used to support IP Trunks, IP Phones and Phontage IP softphones and may be used to support Private Networking. Assure sufficient channels are available to support the desired channels and expected traffic.
- VOIB CO lines may not be compatible with E911 systems; the user is advised to arrange appropriate emergency services.
- It may be necessary to configure stations for access to the assigned CO line group and Flexible buttons for one-touch access.

### Reference

- A. Ring Assignment
- B. Direct Inward Dialing (DID)
- C. Basic Outbound Line Access
- D. CLIP/COLP
- E. Multiple Subscriber Number (MSN)
- F. Mobile Extension
- G. Private Networking

## **15.2 IP address Dialing**

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### Description

ipLDK users can call place calls over IP lines using the IP address of the called party.

### Operation

To call an IP address,

1. Select an IP CO line.
2. Dial the IP address (use the \* to dial the dots "." in the IP address). Ex., for IP Address: 156.147.3.201, dial 156\*147\*3\*201)
3. Press the # button to place the call. If the called party is an ipLDK user, alerting will follow the CO line Ring Assignment (PGM 144).

### Consideration

- When programming a Speed Dial for IP Address calling, the '#' button must not be included.
- For DISA calls, outgoing access to a VoIP CO line is not possible.

### Admin Programming

CO Line Service Type (PGM 140)

Assigns the type of CO line service (Normal, DID, DISA)

VOIB IP Setting (PGM 340)

Sets the IP parameters of the VOIB

### **15.3 IP Phone Support**

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#### Description

The ipLDK-60 can support up to 16 IP Phones. Each IP Phone registers with the ipLDK employing the MAC address of the IP Phone. The IP Phone sends a registration request to the IP address of the ipLDK assigned in the IP Phone. The ipLDK will compare the MAC address in the registration and, if a match is found in the IP Phone Table, registration is allowed. Once registered, the IP Phone will function as any other phone in the system, with the exception that an IP channel must be available for the IP Phone to place or receive a call.

#### Operation

Operation of an IP Phone is the same as the DKT.

#### Consideration

- An IP channel must be available for an IP Phone to place or receive a call including CO line, station to station, transfer, and conference calls. If no channel is available, the system returns error tone for internal calls and re-routes external calls to the Attendant.
- An IP Phone cannot initiate a conference call to stations in a networked system.
- IP Phones employ the MAC address for registration. Phontage IP Softphone employs a User name and Password for registration to the system.
- If the IP Phone or Phontage is not available to the system (network outage), the system can reroute the call to the user's Voice Mailbox or the registered mobile phone. The station must be assigned VMIB access and Auto Forward to VMIB or the mobile extension must be registered.
- To transfer a call to a station in a networked system, the IP Phone user must await ringback tone before disconnecting from the transfer.
- The VOIB must be Appended to the Stations as a Virtual VOIB code 99, see PGM 103, Logical Slot Assignment.

#### Reference

- A. Private Networking
- B. Call Transfer
- C. Multi-party Conference
- D. VMIB Mailbox

#### Admin Programming

Logical Slot Assignment, STA (PGM 103 Flexible button 2)

Assigns the logical order of the KSU Station slots

Flexible Station Numbering Plan (PGM 105)

Assigns station numbers for system

VMIB Access (PGM 113, Flexible button 2)

Permits the station access to the VMIB for Voice mail

Auto Forward to VMIB (PGM 113, Flexible button 14)

Automatically forwards calls to the VMIB after the Forward to VMIB timer

Forward to VMIB timer (PGM 181, Flexible button 20)

Sets the delay before calls automatically forward to the user's VMIB mailbox

Mobile Extension (PGM 236)

Assigns characteristics for the Mobile Extension feature

VOIB Slot Assignment for RSG/IP Phones (PGM 380)

Assigns the number of IP channels for IP Phones

RSG/IP Phone Port Number Assignments

Assigns the number of IP Phones to be supported

RSG/IP Phone Attributes I (PGM 382)

Assigns characteristics of the IP Phone and system communication

IP Phone Attributes (PGM 382)

Assigns IP Phone parameters for registration

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## 15.4 SIP (Session Initiation Protocol)

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### Description

SIP (Session Initiation Protocol) is an IETF (Internet Engineering Task Force) recognized protocol (RFC-3261) to establish, maintain and terminate multimedia sessions between endpoints over an IP network. SIP, is an application layer control protocol that employs other protocols, such as SDP (Session Description Protocol) to describe the desired connection characteristics and other lower layer protocols.

ipLDK-60 operates in two distinct SIP modes based on the SIP call server routing of incoming calls. The first mode is based on the SIP Register 'method'; in this case, the system registers a SIP User ID to the SIP call server 'Registrar', which identifies the location to receive calls. The second mode is 'SIP Trunking'; the SIP call server routes calls to a designated IP endpoint, in this case, the ipLDK VOIB IP address or a proxy. In either case, the VOIB CO line can provide 'Normal' or 'DID/MSN' service. Once active at the ipLDK-60, incoming and outgoing calls are routed and handled as with the type of service and other attributes assigned for the station and VOIB CO line.

### 15.4.1 Configuring SIP Trunks

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#### Description

The following is a quick guide to SIP Trunk configuration,

- Set the Ext or CO ATD Code CLI (PGM 114)  
CLIP/COLP Table for CO line has priority
- Set CO Service Type for 'Normal' or 'DID' service (PGM 140)
- Set the CO line group for the VOIB CO lines (PGM 141)
- Set ISDN line characteristics (PGM 143)  
including CLIP and COLP, as desired
- Set ISDN Call Type as (PGM 143)
- Set ISDN Enblock Send as 'On', all IP calls are Enblock (PGM 143)
- Set DID CONV Type, Digit Mask, etc., if required, (PGM 143)
- Set Ring Assignments, if required (PGM 145)
- Set the Enblock Timer, if required (PGM 181)
- DID Tables, if required, (PGM 167 and 231)
- Set Networking CO Line Type as SIP (PGM 322)
- Configure VOIB IP settings, as required (PGM 340)  
IP addresses, VOIB, Gateway DNS  
Set VOIB Mode to SIP  
Enter Firewall WAN address, if required
- Set Proxy Server Address as URL or IP address of Proxy or Call Server (PGM 500 PC Admin only)
- Set Domain as URL or IP address for SIP Call Server (PGM 500 PC Admin only)

Reset after programming to assure channel sync



Consideration

- For SIP Trunking, the SIP User Id is the combination of the CLIP/COLP established for the station and lines. Based on CLIP/COLP tables or using PGM 114, the table entry, number of the calling station or the Attendant station for the CO line is used as the SIP user ID.
- Outgoing SIP calls must use Enblock signaling; the ipLDK will wait for the user to dial '#' or the Enblock Timer setting before setting up and sending a SIP Invite.
- MSN tables, COLP, CLIP, CLI Message Wait, etc. apply to VOIB CO lines.
- VOIB CO lines can be used to support mobile extensions.
- A SIP Registered line can be shared by multiple ipLDK-60 users as a common ('Normal') line.
- The system can be equipped with eight (8) VoIP channels that are used to support SIP Trunks, IP Phones and Phontage IP softphones and may be used to support Private Networking. Assure sufficient channels are available to support the desired channels and expected traffic.
- VOIB CO line may not be compatible with E911 systems; the user is advised to arrange appropriate emergency services.
- It may be necessary to configure stations for access to the assigned CO line group and Flexible buttons for one-touch access.

Reference

- A. Ring Assignment
- B. Direct Inward Dialing (DID)
- C. Basic Outbound Line Access
- D. CLIP/COLP
- E. Multiple Subscriber Number (MSN)
- F. Mobile Extension
- G. Private Networking

#### 15.4.2 Configuring SIP Registered Lines

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Description

The following is a quick guide to SIP Register method configuration for VOIB channels,

- Set the Ext or CO ATD Code CLI (PGM 114)  
CLIP/COLP Table for CO line has priority
- Set CO Service Type for 'Normal' or 'DID' service (PGM 140)
- Set the CO line group for the VOIB CO lines (PGM 141)
- Set ISDN line characteristics (PGM 143)  
including CLIP and COLP as desired
- Set ISDN Call Type as required (PGM 143)
- Set ISDN Enblock Send 'On', all IP calls are Enblock (PGM 143)
- Set DID CONV Type, Digit Mask, etc., if required, (PGM 143)
- Set Ring Assignments, if required (PGM 145)
- Set the Enblock Timer, if required (PGM 181)
- DID Tables, if required, (PGM 167 and 231)

- Set Networking CO Line Type as SIP (PGM 322).
- Configure VOIB IP settings, as required (PGM 340)  
IP addresses, VOIB, Gateway DNS  
Set VOIB Mode to SIP  
Enter Firewall WAN address, if required
- Set Proxy Server Address as URL or IP address of Proxy or Call Server (PGM 500 PC Admin only)
- Set Domain as URL or IP address for SIP Call Server (PGM 500 PC Admin only)
- In the SIP User Table, select a User Index (PGM 501)
  - a. Set the User ID as abc @sip.registrar.com or IP address (PGM 501 PC Admin only)
  - b. Set the Contact Number for the appropriate station (PGM 501 PC Admin only)
  - c. Set User ID Register as 'Register' (PGM 501 PC Admin only)
- Set User ID Usage as 'On' (PGM 501 PC Admin only)

Reset after programming to assure channel sync and registration

#### Consideration

- Outgoing SIP calls must use Enblock signaling; the ipLDK will wait for the user to dial '#' or the Enblock Timer setting before setting up and sending a SIP Invite.
- MSN tables, COLP, CLIP, CLI Message Wait, etc. apply to VOIB CO lines.
- VOIB CO lines can be used to support mobile extensions.
- A SIP Registered line can be shared by multiple ipLDK-60 users as a common ('Normal') line.
- The system can be equipped with eight (8) VoIP channels that are used to support SIP Trunks, IP Phones and Phontage IP softphones and may be used to support Private Networking. Assure sufficient channels are available to support the desired channels and expected traffic.
- VOIB CO line may not be compatible with E911 systems; the user is advised to arrange appropriate emergency services.
- It may be necessary to configure stations for access to the assigned CO line group and Flexible buttons for one-touch access.

#### Reference

- A. Ring Assignment
- B. Direct Inward Dialing (DID)
- C. Basic Outbound Line Access
- D. CLIP/COLP
- E. Multiple Subscriber Number (MSN)
- F. Mobile Extension
- G. Private Networking

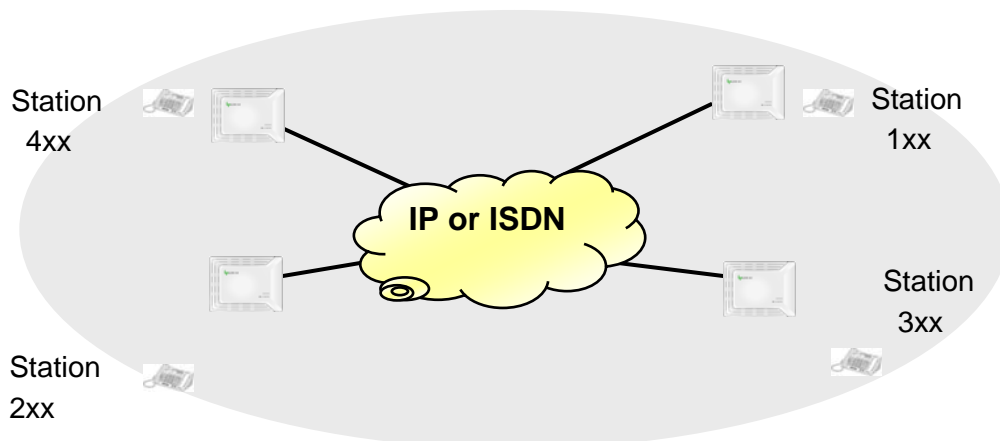
## **16 PRIVATE NETWORKING**

Description

The ipLDK-60 can be linked with other systems over a private network to reduce costs while providing access to common communications features and sharing resources across the network. ipLDK supports two standards based protocols for basic networking functions and LG-Ericsson protocols for advanced network features (e.g., BLF Presence Manager). Protocols employed include:

- QSig protocol via ISDN PRI  
ETS 300 237 / 238 / 256 / 257 / 260 / 261 / 361 / 362 / 363 / 364
- IP standard protocol  
H.323 for Call Control, H.450.1 - H.450.12
- Proprietary protocol for advanced functions

ipLDK can support various interconnected corporate IP and ISDN network configurations with up to 72 systems in the private network. A Unified Numbering Plan is established across the network for transparent operation. Stations of each networked system are assigned a unique 2 to 7 digit station number. Dialing the number from any networked system contacts the station.



**Figure 16 Networking Configuration (UNP-Unified Numbering Plan).**

Consideration

- Special software provided through local distributors activates the Private Networking feature in each ipLDK-60.
- ipLDK supports either Screened or Unscreened call transfer between networked systems.

## **16.1 Busy Lamp Field (BLF) Manager**

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### Description

When systems are networked, the status of stations in one system can be displayed at stations of other systems in the network. The LED of a [Net DSS] button provides the BLF (idle or busy) indication for the assigned station. When only two ipLDK-60 systems are networked, BLF status is provided directly between the two systems with each serving as the others BLF GateKeeper. When more than two systems are in a network, the LG-Ericsson BLF Manager application provides the BLF status across the network. BLF Manager is a Windows® application that communicates with systems on the network as the BLF Gatekeeper. Networked systems send regular updates of station status to the BLF Manager. Periodically, the BLF Manager sends a UDP multi-cast message to all systems on the network. The BLF GateKeeper can reside in any location that has IP access to all systems in the network to receive and send both UDP and TCP packets.

### Consideration

- The BLF Manager GateKeeper is installed as a central function of all ipLDK systems in a network and must be able to communicate using UDP and TCP over an IP network with the ipLDK systems.
- The ipLDK-60 can provide Net DSS BLF status for all stations.
- The BLF Manager provides BLF for stations only, status for CO lines is not provided; CO BLF is not supported.
- The LED of a [Net DSS] button indicates BLF status of busy (LED On), idle (LED Off) or DND (LED flashing). Ring signals at a station, including recalls, are not registered across the network and will display as idle.
- Each ipLDK system updates the station status to the BLF Manager at intervals of the BLF status duration.
- If the BLF manager does not exist, BLF services can be activated between only two Systems. In this case, the GateKeeper address is the IP address of the LAN port of the other system.

### Reference

- A. Flexible Buttons
- B. User Station Programming

### Admin Programming

Networking Supplementary Attributes (PGM 321)

Assigns characteristics of the BLF Manager Gate Keeper for a system in the network

## **16.2 CO Transit - In**

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### Description

When a DID call is received by a networked system, the system will attempt normal DID call routing. The ICLID table is examined, followed by the MSN table, then the defined DID number conversion is completed. The system does a look-up in the Flexible DID Table to identify the destination. If the number is an extension in a networked system, the system routes the call to the destination over the appropriate network connection with CLI provided to the called station.

### Consideration

- CO Transit-In calls are not subject to recall or time restrictions at the originating system.
- If no path is available over the network, the caller will receive busy tone.
- The system, which originally received the call, will generate two SMDR records for the call, the original incoming call and the transit-call to the destination station.

### Reference

- A. DID
- B. ICLID
- C. MSN

### Admin Programming

Networking Routing Table - Numbering Plan (PGM 324)

Assigns the network numbering plan and routing

DID Conversion Type (PGM 143, Flexible button 4)

Defines three type of DID conversion

Flexible DID Table (PGM 231)

Destination table for Type 2 conversion

### **16.3 CO Transit - Out**

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#### Description

CO Transit-Out increases efficiency of CO lines and reduces call costs by routing outgoing CO calls to the nearest appropriate point in a network of systems. A user at one system can dial the appropriate CO line access code for lines in the remote system. Codes are assigned in the Network Numbering Plan at both systems and digit repeating passes the user dialed digits to the remote system.

When the user dials the code, the local system compares the digits with appropriate local Numbering Plan codes then to the Network Numbering Plan codes to determine routing. When the number is found in the Network Numbering Plan table, the call is routed to the designated networked system. Any digits received are sent to the remote system for proper routing.

#### Consideration

- To use CO Transit-Out, the networked system user must dial the CO transit-out code; the CO Transit-Out service is not activated by pressing a CO Line button.
- The Station COS of the networked system is applied for toll restriction.
- For CO Transit-Out, any numbering sequence code can be assigned in the Network Routing Table – Network Numbering Plan (PGM 324). If a conflict exists between the System and Network Numbering Plans, the System Numbering Plan has priority.
- The terminating PSTN lines must be assigned as Networked PSTN CO lines in the terminating system.
- For proper operation of CO Transit-Out, the Network CO Line type and System Usage must be assigned as PSTN.
- While not recommended, a call may be routed to the terminating system through an ancillary system when a direct network connection is not deployed between the originating and terminating systems.
- When enabled in the Network Routing Table – Numbering Plan, the user must enter a valid Station or System Authorization code for access. The associated COS is implemented for the call.
- The system, which terminates the call, will display two SMDR records for the call, the original incoming call from the station number and the transit-out call.

#### Reference

- A. System Numbering Plan
- B. Authorization Codes
- C. Class of Service

#### Admin Programming

Networking CO Line Type (PGM 322, Flexible button 2)

Assigns the PSTN CO line type for the terminating system

Network Routing Table - Numbering Plan (PGM 324)

Assigns the Transit Out code, System Usage and routing information for the code

## **16.4 Centralized Attendant Service (CAS)**

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### Description

Calls can be routed over the network to the Attendants of another system to provide Centralized Attendant Service (CAS). A local System Attendant can route local Attendant calls to the Central Attendants and calls to networked systems can directly ring the Centralized Attendants. These centralized Attendants receive the call under the assigned [Net DSS] button.

### Consideration

- Only DID/DISA calls which are assigned to route to the Attendant destination are routed to the Centralized Attendant.
- The Centralized Attendant can be assigned as a Main Attendant in a networked system. Also, CO lines may be assigned to ring the Centralized Attendant.

### Reference

- A. Flexible Buttons
- B. Flexible CO Ring Assignments
- C. DID
- D. DISA

### Admin Programming

Networking CAS Enable (PGM 320, Flexible button 6)

Enables CAS at the remote systems only

Flexible Button Assignment (PGM 115)

Assigns a [Net DSS] Flexible button at the Central Attendant for remote Attendants

Attendant Assignment (PGM 164)

Assigns System and Main Attendants for the local system

DID/DISA Destination (PGM 167)

Assigns destination of DID/DISA calls to a number that is invalid, busy, etc.

Flexible DID Table (PGM 231)

Destination table for Type 2 conversion

CO Ring Assignment (PGM 144)

Assigns stations to ring for incoming calls on a Normal type Co line

## **16.5 Centralized VM (Voice Mail) Service**

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### Description

An external Voice Mail may be attached to a system in a network to provide centralized Voice Mail services to the stations of all systems in the network. Calls are passed to this central VM over the network to the Network VM Hunt Group assigned of the central ipLDK-60 system. The Station Group number must be assigned in the Network Numbering Plan code of each networked system to deliver calls to the central VM properly.

### Consideration

- The centralized VM must be configured in the central ipLDK-60 system to which it is attached as a Network VM. The Voice Mail Dialing table and other characteristics of the Network VM are established in the central ipLDK-60 system.
- The Destination MBU IP address must be defined as part of the network Numbering Plan configuration for each system that will use the centralized VM except for the master ipLDK-60 system.

### Reference

- A. Station Hunt Groups
- B. Network Call Forward
- C. External Voice Mail

### Admin Programming

Network Destination MBU IP (PGM 324, Flexible button 6)

Assigns the IP address of the Voice Mail terminating ipLDK

Station Hunt Group Assignment – Network VM Group (PGM 190)

Assigns the Hunt Group and characteristics of the Network VM group



## 16.6 Common Network Services

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An array of features is supported over the network employing the same operational procedures as the ipLDK itself. Common features supported include:

- Station calling
- Call Transfer
- Conference
- Camp-on
- Call Back
- CLI Message Wait
- Call Forward

In addition, features such as DND and Absent Text Messaging are supported between networked ipLDK systems.

### 16.6.1 Network Absent Text Message

---

#### Description

Absent Text Message (refer to Ref. A) is available within the networking connection environment. If a user activates Absent Text Message, the message is displayed to callers, networked or local, using a DKT with LCD.

#### Consideration

- Absent Text Messaging is only available to ipLDK systems within the network.

#### Reference

- A. Absent Text Message

### 16.6.2 Network Call Back (Call Completion)

---

#### Description

When calling to a networked station that is busy, the user can leave a Call Back request. When the busy station returns to idle, the station activating the Call Back is called. Upon answer, the ipLDK will place a call to the originally busy station. Network Call Back operates in the same manner as Call Back (refer to Ref. A) within an ipLDK and is subject to the conditions associated with Call Back.

From a network perspective, ipLDK employs H.450 Call Completion, Busy Subscriber (CCBS) protocol to provide call back capabilities. ipLDK supports the Connection Retain mode in the H.450 (CCBS) protocol.

#### Consideration

- Voice Mail is not available to networked intercom calls.

- If the originator does not answer the call back ring within the Call Back ring time (30 seconds), the call is dropped.

### Reference

- A. Station Message Wait/Call Back

### Admin Programming

Networking Basic Attribute (PGM 320)

## 16.6.3 Network Call Forward

---

### Description

A user can forward (send) their calls to another station, Station Group or a user's VMIB mailbox across the private network. The user may forward all calls immediately, if the user does not answer the call or if the station is busy. In addition, the user may select to activate 'Follow-me' forward, forwarding calls from a location other than the station. Calls forward in the same manner as a Call Forward within a system (refer to Ref. A) and is transparent to the user.

From a network perspective, the ipLDK can use either the 'Join' or 'Re-route' signaling method to complete a call forward over the network. The Networking Transfer Mode assigns the method that will be used and should be selected to match other systems connected in the private network.

- Join - an additional connecting path is needed to transfer the call to another station.
- Routing - a new connecting path is used to forward the call.

### Consideration

- To use Call Forward, a station must be permitted to forward calls in the system database.
- Authorization codes are required for 'Follow-me' Call Forward and may be required to activate Network Call Forward 'Follow-Me'
- A station is separately allowed Call Forward and Off-Net Call Forward.
- The ipLDK does not check the status of the receiving destination; once forward is activated, the receiving system is responsible for routing the call.
- Incoming analog CO lines must provide Disconnect Supervision and the Open Loop Detect timer must be set for proper operation.
- Station calls may automatically forward using Preset Call Forward or may forward to the user's VMIB voice mailbox after the VMIB Forward time. If the Preset Call Forward or VMIB Forward timers are shorter than the Call Forward No-Answer timer, calls will not forward over the network.
- Network Call Forward 'Follow-Me' can only be deactivated at the original station.
- Network Call Forward Follow-Me is only available when employing H.323/H.450 networking. When employing QSig Call Forward 'Follow-me' is not available over the network.
- For proper operation of Call Forward Follow-Me, all ipLDK systems in the network must employ the same type of Authorization codes, 5-digit or variable length.

- Call Forward Follow-me is not available between networked systems. Confirmation tone is provided to the user even though Call Forward 'Follow-Me' is not allowed.

### Reference

- A. Call Forward
- B. Preset Call Forward
- C. Authorization Code
- D. VMIB Voice Mailbox

### Admin Programming

Call Forward (PGM 111, Flexible button 2)

Permits stations access to Call Forward

Call Forward No Answer Timer (PGM 181, Flexible button 1)

Sets the timer for No answer Forward

Allow Off-Net Call Forward (PGM 111, Flexible button 18)

Permits Stations access to Off-net Call Forward

Open Loop Detect Timer (PGM 142, Flexible button 13)

Sets the duration of an for disconnect supervision for analog lines

Networking Basic Attributes (PGM 320)

Networking Supplementary Attributes - Transfer Mode (PGM 321)

Networking CO Line Attribute (PGM 322)

Networking Routing Table - Numbering Plan (PGM 324)

## 16.6.4 Network Call Transfer

---

### Description

An active call, internal or CO line, can be transferred to other stations across the Private Network. Calls can be transferred across the network employing screened or unscreened transfer in the same manner as a transfer within a system (refer to Ref. A) and is transparent to the user.

From a network perspective, the ipLDK can use either the 'Join' or 'Re-route' signaling method to complete a call transfer over the network. The Networking Transfer Mode assigns the method that will be used and should be selected to match other systems connected in the private network.

- Join—an additional connecting path is needed to transfer the call to another Station.
- Routing—a new connecting path is used to transfer the call (the old connecting path of transferring station will be cleared).

### Consideration

- The ipLDK automatically determines the type of transfer, local or networked, based on the receiving station number dialed and the Network Routing Table.
- If the user transferring a call presses the [TRANS/PGM] during the transfer, the transfer is cancelled and the active call re-established.
- Network Call Transfer will not recall to the transferring station.

- Networked systems are not designed to be non-blocking. Should no path be available to complete the transfer, error tone is provided to the caller.
- If the receiving station is busy, the transferring user will receive busy tone and may camp the call to the busy station (refer to Ref. B). If the receiving station is in DND, the call is rejected, the caller receives busy tone and cannot camp-on.

#### Reference

- A. Call Transfer
- B. Network Camp-on, Call Offer

#### Admin Programming

Networking Basic Attributes (PGM 320)

Networking Supplementary Attributes - Transfer Mode (PGM 321, Flexible button 1)

Networking CO Line Attributes (PGM 322)

Networking Routing Table - Numbering Plan (PGM 324)

### 16.6.5 Network Camp-on (Call Offer)

---

#### Description

When calling to a networked station that is busy, the user may camp a call on to the called station. The called station receives a camp-on tone and the [Hold/Save] button LED will flash. Network Camp-on operation is the same as system Camp-on (refer to Ref. A) and is subject to the same conditions.

From a network perspective, ipLDK employs H.450 Call Offer protocol to provide camp-on capabilities.

#### Consideration

- Call Offer is only applied when calling a station that is in an active call state.
- Network Camp-on is not available with conference or page.
- The ipLDK-60 System does not support Network Camp-on over ISDN CO lines employing QSig.

#### Reference

- A. Camp-on
- B. Conference
- C. Paging

#### Admin Programming

Networking Basic Attribute (PGM 320)

### 16.6.6 Network Conference

---

#### Description

Stations can join stations from other networked systems to a call with other stations or CO lines.

Reference

- A. Unsupervised Conference

Consideration

- If the Conference initiator returns to idle, the conference will be terminated.
- The maximum number of networked stations in a conference is 3.
- An IP phone cannot initiate a conference with stations in a networked system.

Admin Programming

Networking Basic Attribute (PGM 320)

Network Routing Table - Numbering Plan (PGM 324)

### 16.6.7 Network Identification Service

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Description

When CNIP and CNOP are enabled, the station name is provided as the caller identification received and displayed at the networked station receiving the networked intercom call.

Operation

CNIP and CONP operation is executed whenever a station in a networked system makes a call to a station in a networked system.

Consideration

- To use CNIP and CONP service, the name of the station and CNIP and CONP must be enabled.

Admin Programming

Networking CNIP and CONP Enable (PGM 320)

Enables calling/called station Name display

### 16.6.8 Network MWI, CLI Message Wait

---

Description

When calling to a networked station that does not answer, the user can leave a Message Wait to the called station. The called station will receive a CLI Message Wait (refer to Ref. A). Network MWI operates in the same manner as CLI Message Wait except that the calling party must activate the Message Wait using the Call Back button (refer to Ref. B) and is subject to the conditions associated with CLI Message Wait.

From a network perspective, ipLDK employs H.450 Call Completion, No Reply protocol to provide CLI Message Wait capabilities. ipLDK supports the H.450 Completion of Calls to No Response (CCNR) protocol.

Consideration

- Network MWI is available to DKT stations with an LCD only.
- When the non-responding station replies to the CLI Message Wait, the system will use a CO line from the appropriate Network CO group.

#### Reference

- A. CLI Message Wait
- B. Station Message/Wait Call Back

#### Admin Programming

CLI Message Wait (PGM 114, Flexible button 4)

Enables the Station to receive CLI Message Waiting

Networking Basic Attribute (PGM 320)

Networking CO Line Attribute (PGM 322)

Networking Routing Table - Numbering Plan (PGM 324)

### 16.6.9 Network Station to Station Call

---

#### Description

Users can place calls to stations connected to other networked systems. Station calls across the network are always terminated in the Tone Ring Intercom Signaling mode (refer to Ref. A). Station calls are placed in the normal manner by dialing the desired station number or pressing a [Net-DSS] button.

#### Consideration

- Networked systems are not designed to be non-blocking. Should no path be available to complete a station call, error tone is provided to the caller.
- If a fatal error is detected in the Private Network when placing a call, ipLDK can re-route the call over the public network using the assigned Alternate Route.
- A station-to-station network call is subject to the CO Call Restriction timer (refer to Ref. B).
- If the receiving station is in DND, the call is rejected by the receiving system and the caller receives busy tone.

#### Reference

- A. Intercom Signal Mode
- B. Call Time Restriction

#### Admin Programming

Station Numbering Plan (PGM 105)

Assigns numbers to stations

Alternate Dial Bin (PGM 324, Flexible button 5)

Assigns alternate route to a networked system using a System Speed bin

Network Basic Attributes (PGM 320)

Networking Supplementary Attributes (PGM 321)

Networking CO Line Attributes (PGM 322)

Network Routing Table - Numbering Plan (PGM 324)

## **16.7 Network Configuration**

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### 16.7.1 ISDN QSig

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#### Description

When ipLDK systems are networked using a PRI, QSig protocol is employed. The CO Lines (channels of the PRI) must be correctly assigned for DID/MSN service and appropriate DID configuration (DID Conversion Type and DID Digit Mask) must be assigned. In addition, various Networking parameters must be set based on the functions needed (internal calling, CO Transit Out, Network CO Line Groups and Network Numbering Plan) to support a Uniform Numbering Plan.

Assign a Uniform Numbering Plan at each system,

1. Assign unique station numbers to each system (PGM 105)

Assign the basic CO line parameters

2. Determine the CO line numbers for the PRI used for networking, e.g., 04 to 26 04 to 33
3. Under CO Line Service Type (PGM 140) assign the CO lines for DID/MSN service
4. Assign the CO lines to DID digit Conversion type 1, no conversion

Enable the Networking function, PGM 320

1. Turn on Networking Enable

Assign the attributes for networked CO lines, PGM 322

1. Assign the CO lines to a Network CO line Group.
2. Define the function of the Network CO line group
  - PSTN, use to place a call that terminates to a public network
  - Net, use to access internal stations, paging, etc. in a networked system

Assign the Network Numbering Plan PGM 324

1. Enter Numbering Plan Table code index (00 – 71)
2. Assign function for the Network Numbering code
  - PSTN, use to place a call that terminates to a public network
  - Net, use to access internal resources (stations, paging, etc.) of a networked system
3. Enter Numbering Plan code digits, up to 16.
4. Enter Network CO Group to use for the Numbering Plan Code
5. Repeat for each Network Numbering Plan code.

Each Numbering Plan code is a series of up to 16 digits. If the DID number or station user dialed number matches a Numbering Plan Code, the call is routed using the instructions for the code. The code can be a station number sequence, CO Transit-out code or a Network VM Group code.

The Network Numbering Plan code for local station to station calls is the first digit of the Station Numbering Plan (PGM 105) followed by '#' to indicate local and '\*' as wild card digits, e.g., 1#\*\*

for the remaining station number digits. The Numbering Plan code for networked station to station calls is the first digit of the Station Numbering Plan of the network system followed by '\*' as a wild card for the remaining station number digits, e.g., 2\*\*, 3\*\*, etc. A Network Numbering Plan code is required for each networked system to allow station to station and DID call routing over the network.

Example customer requirements:

Three sites are to be linked in a network using ISDN PRI lines

System A is HQ with station numbers 100 to 133

System B is Branch 1 with station numbers 200 to 214

System C is Branch 2 with station numbers 300 to 328

- Incoming DID numbers are 555-7200 to 555-7450 at HQ only
- HQ has a full 23 channel PRI and 6 analog CO lines, one is a Private line
- Branch 1 has a 6 channel partial PRI for networking and 3 analog CO lines
- Branch 2 has a 12 channel partial PRI for networking and 4 analog CO lines, one is a Private line
- For the main number, 555-7200, a CCR announcement is desired as an Auto Attendant
- Branch 1 and 2 can access local lines in HQ using dial code '4 3', CO Transit-out
- The HQ system will provide CAS and Centralized VM

Station Numbering Plan, PGM 105

HQ	station range 100 ~ 147
Branch 1	station range 200 ~ 247
Branch 2	station range 300 ~ 347

CO line Type and Attributes, PGM 140, 141, 143 and 146

HQ	CO line range 01 ~ 03 analog CO lines Type: Normal, DISA Service On/Off CO Group 01
	CO line range 04 ~ 25 PRI lines Type: DID/MSN DID Conversion Type 2, remove 4 digits, Digit Mask = '* * *' Flexible DID Table Index 200 route to VMIB 001 for Auto Attendant CCR Index 201 to 234 route to stations 100 to 133 respectively Index 300 to 313 route to stations 200 to 213 respectively Index 400 to 428 route to stations 300 to 328 respectively
	CO line range 26 ~ 28 analog lines Type: Normal, 26 & 27 DISA Service On for remote VM access CO line 26 & 27 CO Group 02 CO line 28 CO Group 00 Private line
Branch 1	CO line range 01 ~ 03 analog lines Type: Normal, DISA Service On/Off CO Group 01
	CO line range 04 ~ 9 PRI lines Type: DID/MSN DID Conversion Type 1, use digits from network as received to route call
Branch 2	CO line range 01 ~ 03 analog lines Type: Normal, DISA Service On/Off



CO Group 01  
 CO line range 04 ~ 15 PRI lines  
 Type: DID/MSN  
 DID Conversion Type 1, use digits from network as received to route call  
 CO line range 16 ~ 18 analog lines  
 Type: Normal, DISA Service On/Off  
 CO line 16 CO Group 00, Private line  
 CO lines 17 & 18 CO Group 25, unequipped

DID/DISA Destination, PGM 167

HQ Assign as desired for busy, DND, invalid number destination  
 Branch 1 Assign as desired for busy, DND, invalid number destination  
 Branch 2 Assign as desired for busy, DND, invalid number destination

Station Hunt Group PGM 190 & 191

HQ Station range 140 to 147, SLT ports for Central Voice Mail  
 Members of Group 620  
 Group 620 Type: Network Voice Mail  
 Network VM Group parameters: as required  
 VM Dialing Table as required

Networking, PGM 320

HQ Networking; Enable  
 Networking CAS: Disable, system with Central Attendant must have CAS disabled  
  
 Branch 1 Networking; Enable  
 Networking CAS: Enable, assign a UCD-DND button at Branch 1 System Attendants  
  
 Branch 2 Networking; Enable  
 Networking CAS: Enable, assign a UCD-DND button at Branch 2 System Attendants

Network CO line Attributes

HQ	CO Line range ?? to ??	Networking CO Line Group 01 Network CO Line Type: Net for internal network calling
	CO Line range ?? to ??	Network CO Line Group 02 Network CO Line Type: PSTN for CO Transit-out
Branch 1	CO line range ?? to ??	Network CO Line Group 01 Network CO Line Type: Net for internal network calling
	CO Line range ?? to ??	Network CO Line Group 02 Network CO Line Type: PSTN for CO Transit-out
Branch 2	CO line range ?? to ??	Network CO Line Group 01 Network CO Line Type: Net for internal network calling
	CO Line range ?? to ??	Network CO Line Group 02 Network CO Line Type: PSTN for CO Transit-out

Network Routing Table-Numbering Plan

HQ	Index 001	System Use: Net, internal network call Number Plan Code: 1#**, local station call Network CO Group: 00 Alternate Speed Dial bin: as required Firewall Routing: 'Off'
	Index 002	System Use: Net, internal network call Number Plan Code: 2** , local station call Network CO Group: 01

	Index 003	<p>Alternate Speed Dial bin: as required            Firewall Routing: as required            System Use: Net, internal network call            Number Plan Code: 3**, local station call            Network CO Group: 01</p>
	Index 004	<p>Alternate Speed Dial bin: as required            Firewall Routing: as required            System Use: Net, internal network call to Voice Mail            Number Plan Code: 620, Network Voice Mail            Network CO Group: 01</p>
	Index 005	<p>Alternate Speed Dial bin: as required            Firewall Routing: as required            System Use: PSTN            Number Plan Code: 4 3, CO Transit out code            Network CO Group: 02</p>
		<p>Alternate Speed Dial bin: as required            Firewall Routing: as required            Authorization Code Use: as required</p>
Branch 1	Index 001	<p>System Use: Net, internal network call            Number Plan Code: 2#**, local station call            Network CO Group: 00</p>
	Index 002	<p>Alternate Speed Dial bin: as required            Firewall Routing: 'Off'            System Use: Net, internal network call            Number Plan Code: 1**, local station call            Network CO Group: 01</p>
	Index 003	<p>Alternate Speed Dial bin: as required            Firewall Routing: as required            System Use: Net, internal network call            Number Plan Code: 3**, local station call            Network CO Group: 01</p>
	Index 004	<p>Alternate Speed Dial bin: as required            Firewall Routing: as required            System Use: Net, internal network call to Voice Mail            Number Plan Code: 620, Network Voice Mail            Network CO Group: 01</p>
	Index 005	<p>Alternate Speed Dial bin: as required            Firewall Routing: as required            System Use: PSTN            Number Plan Code: 4 3, CO Transit out code            Network CO Group: 02</p>
		<p>Alternate Speed Dial bin: as required            Digit Repeating: Yes            Firewall Routing: as required            Authorization Code Use: as required</p>
Branch 2	Index 001	<p>System Use: Net, internal network call            Number Plan Code: 3#**, local station call            Network CO Group: 00</p>
	Index 002	<p>Alternate Speed Dial bin: as required            Firewall Routing: 'Off'            System Use: Net, internal network call            Number Plan Code: 1**, local station call            Network CO Group: 01</p>
		<p>Alternate Speed Dial bin: as required</p>

Index 003	Firewall Routing: as required System Use: Net, internal network call Number Plan Code: 2**, local station call Network CO Group: 01 Alternate Speed Dial bin: as required
Index 004	Firewall Routing: as required System Use: Net, internal network call to Voice Mail Number Plan Code: 620, Network Voice Mail Network CO Group: 01 Alternate Speed Dial bin: as required
Index 005	Firewall Routing: as required System Use: PSTN Number Plan Code: 4 3, CO Transit out code Network CO Group: 02 Alternate Speed Dial bin: as required Digit Repeating: Yes Firewall Routing: as required Authorization Code Use: as required

## 16.7.2 VOIP Networking

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### Description

When ipLDK systems are networked using a H.323, additional features are supported employing H.450.01-12. The CO lines (individual VOIB channels) must be correctly assigned for DID/MSN service and appropriate DID configuration (DID Conversion Type and DID Digit Mask) must be assigned. In addition, various Networking parameters must be set based on the functions needed (internal calling, CO Transit Out, Network CO Line Groups and Network Numbering Plan) to support a Uniform Numbering Plan.

Network configuration with VoIP is the similar to ISDN QSig Network configuration with the addition of assigning the 'CPN IP Information' and configuring the VOIB. The CPN IP Information is defined for each Network Numbering Plan code to identify the IP address of the VOIB to receive network call.

Assign a Uniform Numbering Plan at each system,

1. Assign unique station numbers to each system (PGM 105)

Assign the basic CO line parameters

2. Determine the CO line numbers for the VOIB channels for networking, e.g., 07 to 14
3. Under CO Line Service Type (PGM 140) assign the CO lines for DID/MSN service
4. Assign the CO lines to the proper DID digit Conversion type, conversion type can be:
  - 0 – modify the digits with the DID Digit Mask
  - 1 – no conversion required, received digits are a station number
  - 2 – modify as in type 0, use the result as index to Flexible DID Conversion Table, PGM 231
5. Assign the appropriate digit modification
  - DID Remove Digits
  - DID Digit Mask
  - Flexible DID Conversion Table, type '2' only, PGM 231
  - DID/DISA Destination Table, PGM 167

Configure the VOIB IP channels, PGM 340

6. VOIP IP Settings

Enable the Networking function, PGM 320

7. Turn on Networking Enable

Assign the attributes for networked CO lines, PGM 322

8. Assign the CO lines to a Network CO line Group.
9. Define the function of the Network CO line group
  - PSTN, use to place a call that terminates to a public network
  - Net, use to access internal stations, paging, etc. in a networked system

Assign the Network Numbering Plan PGM 324

10. Enter Numbering Plan Table code index (00 – 71)
11. Assign function for the Network Numbering code

- PSTN, use to place a call that terminates to a public network
  - Net, use to access internal resources (stations, paging, etc.) of a networked system
12. Enter Numbering Plan code digits, up to 16.
  13. Enter Network CO Group to use for the Numbering Plan Code
  14. Repeat for each Network Numbering Plan code.

Each Numbering Plan code is a series of up to 16 digits. If the DID number or station user dialed number matches a Numbering Plan Code, the call is routed using the instructions for the code. The code can be a Station number sequence, CO Transit-out code or a Network VM Group code.

The Network Numbering Plan code for local station to station calls is the first digit of the Station Numbering Plan (PGM 105) followed by '#' to indicate local and '\*' as wild card digits, e.g., 1#\*\* for the remaining station number digits. The Numbering Plan code for networked station to station calls is the first digit of the Station Numbering Plan of the network system followed by '\*' as a wild card for the remaining station number digits, e.g., 2\*\*, 3\*\*, etc. A Network Numbering Plan code is required for each networked system to allow station to station and DID call routing over the network.

Example customer requirements:

Three sites are to be linked in a network using ISDN PRI lines

System A is HQ with station numbers 100 to 133

System B is Branch 1 with station numbers 200 to 214

System C is Branch 2 with station numbers 300 to 328

- Incoming DID numbers are 555-7200 to 555-7450 at HQ only
- HQ has a 12 channel partial PRI for DID lines, 3 analog CO lines (one is a Private line) and 8 IP channels for networking
- Branch 1 has 4 IP channels for networking and 3 analog CO lines
- Branch 2 has 8 IP channels for networking and 4 analog CO lines, one is a Private line
- For the main number, 555-7200, a CCR announcement is desired as an Auto Attendant
- Branch 1 and 2 can access local lines in HQ using dial code '4 3', CO Transit-out
- The HQ system will provide CAS and Centralized VM

Station Numbering Plan, PGM 105

HQ	station range 100 ~ 147
Branch 1	station range 200 ~ 247
Branch 2	station range 300 ~ 347

CO line Type and Attributes, PGM 140, 141, 143 and 146

HQ	CO line range 01 ~ 03 analog CO lines
	Type: Normal, 01 & 02 DISA Service On for remote VM access
	CO line 01 & 02 CO Group 01
	CO line 03 CO Group 00 Private line
	CO line range 4 ~ 15 PRI lines for incoming DID and outgoing calls
	Type: DID/MSN
	CO line Group 02
	DID Conversion Type 2, remove 4 digits, Digit Mask = '* * *'
	Flexible DID Table
	Index 200 route to VMIB 001 for Auto Attendant CCR

Index 201 to 234 route to stations 100 to 133 respectively  
 Index 300 to 313 route to stations 200 to 213 respectively  
 Index 400 to 428 route to stations 300 to 328 respectively  
 CO line range 16 ~ 23 VoIP lines  
 Type: DID/MSN  
 DID Conversion: Type 1, use as is  
 ISDN Enblock Send: On

Branch 1 CO line range 01 ~ 03 analog lines  
 Type: Normal, DISA Service On/Off  
 CO Group 01  
 CO line range 04 ~ 07 VoIP lines  
 Type: DID/MSN  
 DID Conversion: Type 1, use digits from network as received to route call  
 ISDN Enblock Send: On

Branch 2 CO line range 01 ~ 03 analog lines  
 Type: Normal, DISA Service On/Off  
 CO Group 01  
 CO line range 04 ~ 06 analog lines  
 Type: Normal, DISA Service On/Off  
 CO line 04 CO Group 00, Private line  
 CO lines 05 & 06 CO Group 25, unequipped  
 CO line range 07 ~ 14 VoIP lines  
 Type: DID/MSN  
 DID Conversion: Type 1, use digits from network as received to route call  
 ISDN Enblock Send: On

DID/DISA Destination, PGM 167

HQ Assign as desired for busy, DND, invalid number destination  
 Branch 1 Assign as desired for busy, DND, invalid number destination  
 Branch 2 Assign as desired for busy, DND, invalid number destination

Station Hunt Group PGM 190 & 191

HQ Station range: 140 to 147, SLT ports for Central Voice Mail  
 Members of Group 620  
 Group 620 Type: Network Voice Mail  
 Network VM Group parameters: as required  
 VM Dialing Table: as required

Customer Call Routing, PGM 228

HQ CCR Table index 01 set to match VMIB announcement 001

Networking, PGM 320

HQ Networking: Enable  
 Networking CAS: Disable, system with Central Attendant must have CAS disabled  
 Branch 1 Networking: Enable  
 Networking CAS: Enable, assign a UCD-DND button at Branch 1 System Attendants  
 Branch 2 Networking: Enable  
 Networking CAS: Enable, assign a UCD-DND button at Branch 2 System Attendants

Network CO line Attributes, PGM 322

HQ CO Line range ?? to ?? Networking CO Line Group 01  
 Network CO Line Type: Net for internal network calling

	CO Line range ?? to ??	Network CO Line Group 02
	VOIB Mode	Network CO Line Type: PSTN for CO Transit-out H.323
Branch 1	CO line range ?? to ??	Network CO Line Group 01
		Network CO Line Type: Net for internal network calling
	CO Line range ?? to ??	Network CO Line Group 02
	VOIB Mode	Network CO Line Type: PSTN for CO Transit-out H.323
Branch 2	CO line range ?? to ??	Network CO Line Group 01
		Network CO Line Type: Net for internal network calling
	CO Line range ?? to ??	Network CO Line Group 02
	VOIB Mode	Network CO Line Type: PSTN for CO Transit-out H.323

## Network Routing Table-Numbering Plan (PGM 324)

HQ	Index 001	System Use: Net, internal network call Number Plan Code: 1#**, local station call Network CO Group: 00 Firewall Routing: 'Off'
	Index 002	System Use: Net, internal network call Number Plan Code: 2**, local station call Network CO Group: 01 CPN IP Information: IP address of Branch 1 VOIB Alternate Speed Dial bin: as required Firewall Routing: as required
	Index 003	System Use: Net, internal network call Number Plan Code: 3**, local station call CPN IP Information: IP address of Branch 2 VOIB Network CO Group: 01 Alternate Speed Dial bin: as required Firewall Routing: as required
	Index 004	System Use: Net, internal network call to Voice Mail Number Plan Code: 620, Network Voice Mail Network CO Group: 01 Firewall Routing: as required
	Index 005	System Use: PSTN Number Plan Code: 4 3, CO Transit out code Network CO Group: 02 Firewall Routing: as required Authorization Code Use: as required
Branch 1	Index 001	System Use: Net, internal network call Number Plan Code: 2#**, local station call Network CO Group: 00 Firewall Routing: 'Off'
	Index 002	System Use: Net, internal network call Number Plan Code: 1**, local station call Network CO Group: 01 CPN IP Information: IP address of HQ VOIB Alternate Speed Dial bin: as required Firewall Routing: as required
	Index 003	System Use: Net, internal network call Number Plan Code: 3**, local station call Network CO Group: 01

		<p>CPN IP Information: IP address of Branch 2 VOIB          Alternate Speed Dial bin: as required          Firewall Routing: as required</p>
	Index 004	<p>System Use: Net, internal network call to Voice Mail          Number Plan Code: 620, Network Voice Mail          Network CO Group: 01</p>
		<p>CPN IP Information: IP address of HQ VOIB          Alternate Speed Dial bin: as required          Firewall Routing: as required</p>
	Index 005	<p>System Use: PSTN          Number Plan Code: 4 3, CO Transit out code          Network CO Group: 02</p>
		<p>CPN IP Information: IP address of HQ VOIB          Alternate Speed Dial bin: as required          Digit Repeating: Yes          Firewall Routing: as required          Authorization Code Use: as required</p>
Branch 2	Index 001	<p>System Use: Net, internal network call          Number Plan Code: 3#**, local station call          Network CO Group: 00          Firewall Routing: 'Off'</p>
	Index 002	<p>System Use: Net, internal network call          Number Plan Code: 1**, local station call          Network CO Group: 01</p>
		<p>CPN IP Information: IP address of HQ VOIB          Alternate Speed Dial bin: as required          Firewall Routing: as required</p>
	Index 003	<p>System Use: Net, internal network call          Number Plan Code: 2**, local station call          Network CO Group: 01</p>
		<p>CPN IP Information: IP address of Branch 1 VOIB          Alternate Speed Dial bin: as required          Firewall Routing: as required</p>
	Index 004	<p>System Use: Net, internal network call to Voice Mail          Number Plan Code: 620, Network Voice Mail          Network CO Group: 01</p>
		<p>CPN IP Information: IP address of HQ VOIB          Alternate Speed Dial bin: as required          Firewall Routing: as required</p>
	Index 005	<p>System Use: PSTN          Number Plan Code: 4 3, CO Transit out code          Network CO Group: 02</p>
		<p>CPN IP Information: IP address of HQ VOIB          Alternate Speed Dial bin: as required          Digit Repeating: Yes          Firewall Routing: as required          Authorization Code Use: as required</p>
VOIB IP Setting, PGM 340		
HQ		<p>IP Address: IP address of HQ VOIB          Gateway: address for HQ VOIB          Subnet mask: mask for HQ VOIB          VOIB mode: H.323</p>
Branch 1		<p>IP Address: IP address of Branch 1 VOIB</p>



	Gateway: address for Branch 1 VOIB Subnet mask: mask for Branch 1 VOIB VOIB mode: H.323
Branch 2	IP Address: IP address of Branch 2 VOIB Gateway: address for Branch 2 VOIB Subnet mask: mask for Branch 2 VOIB VOIB mode: H.323

## **16.8 Network Firewall routing**

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### Description

In order to provide routing through a Firewall, each system in a network can be assigned to use the Firewall IP address allowing communications through the firewall.

### Consideration

- A user can program Firewall Routing feature with each net numbering plan table.

### Admin Programming

Firewall Routing (PGM 324, Flexible button 9)

Enable operation behind a firewall

Firewall IP Address (PGM 340, Flexible button 11)

Assigns the firewall IP address

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## 17 ADMIN & MAINTENANCE

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### 17.1 Digital Keypad Admin

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#### Description

The ipLDK database can be configured employing a designated DKT to access the various parameters of the system. In the programming mode, normal operation of the DKT is terminated and the DKT works as a programming terminal. The dial pad is used to input Program codes and associated data. Flexible buttons are used to access and modify specific parameters and the LCD is employed to view the various parameters and the settings for each. A DKT with the 24-Flexible button array is recommended to allow access to all parameters available.

While in the programming mode, operation of certain buttons is modified to permit easier data entry. The softkey menu generally provides a Back, Delete and Save functions. The Volume Up/Down buttons permit scrolling through menus and entries. The [Conf] button returns to the 1<sup>st</sup> step of the data entry procedure for the Program Code.

Additional details are provided in the ipLDK-60 Admin Programming manual.

#### Operation:

To enter the Programming mode from a DKT,

1. Press the [TRANS/PGM] button.
2. Dial '\* #'.
3. Enter the Admin password, if required.

The general procedure for data entry after entering the Program mode is,

1. Press the [TRANS/PGM] button.
2. Enter the desired Program Code.
3. For station or CO based Program Codes, enter the station or CO range.
4. Select a desired parameter, generally using a Flexible button.
5. Use the dial pad to enter data.
6. Press the [SAVE] softkey.

To exit the Programming Mode,

1. Press the [On/Off] button.

#### Consideration

- As a default, the System Attendant at station 100 (station port 01) can access the database. Any DKT station(s) can be allowed to access and modify the ipLDK-60 database.
- SIP Attributes 1 and 2 are only accessible with the PC Admin and are not available to a DKT for programming.

- The [SAVE] button must be pressed after data entry to store the input.
- Database access is allowed to only a single user, DKT or PC Admin at a time; simultaneous database access is not allowed.

#### Admin Programming

Admin access (PGM 113, Flexible button 1)

Enables station to access the system database.

Admin Password (PGM 162)

Assigns a password for access to Admin from a DKT or PC Admin

## **17.2 Lock-key operation**

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### Description

ipLDK-60 supports a number of software applications including Private Networking, ez-Attendant, ez-Phone, Phontage and TAPI. The system software includes 60-day trial license of the applications. In addition, the software includes the ability to support two Phontage IP softphones. To permanently add a demonstration application or expand the supported Phontage softphones, additional software must be added to the system based on the system serial number. Using the system serial number a key code is generated and, when entered over a telnet or PC Admin session, will add the application to the system software.

### Operations

For detail installation instructions see LG-Ericsson STI-0049 available to LG-Ericsson partners only.

### **17.3 PC Admin, On-Line**

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#### Description

LDK PC Admin is a Windows® application designed for real-time access to the database of an ipLDK. PC Admin presents the ipLDK database in an efficient, easy to use Graphical User Interface roughly following the Program codes of Digital Keypad Admin. PC Admin provides four modes of access to the system database RS-232C serial port connection, Modem connection, ISDN PPP and LAN connection using TCP. The RS-232C port should be connected as outlined in the Description and Installation Manual, and can be configured in the system database to match most terminals.

When using the modem connection for PC Admin access, the system must be equipped with the ipLDK-60 Modem Unit (ModU) as detailed in the Description and Installation manual. The modem automatically answers an incoming call when the configured 'dedicated' CO line or 'associated' station receives a call.

When using the system LAN interface for connection, the system IP address is required. As a default, the ipLDK assigns the private address 192.168.1.1 to permit local PC Admin access. Other IP addressing schemes may provide remote access.

PC Admin User and Level management tools allow controlled database access; one of 99 levels can be defined to a database of users. Each level is configured to allow/deny access to segments of the system database. As a default, the user name is *administrator* and password is *0000* with unlimited Level 1 access.

In addition to access and real-time management of the ipLDK database, PC Admin provides a number of utilities for database storage and back-up (refer to Ref. A), software upgrades (refer to Ref. B.) and lock key entry (refer to Ref. C).

#### Operations

For operations, see the PC Admin User Guide

#### Consideration

- The system and PC must be properly configured for the type of access.
- Database access is allowed to only a single user, DKT or PC Admin at a time; simultaneous database access is not allowed.
- The system may require a reset after the database or software is uploaded.
- When upgrading software, do not interrupt the process. When the system returns to normal operation changes can be made to the database, etc.
- When using the Modem or LAN connection a system reset will disconnect the active session and a new connection must be established for further PC Admin activity.

#### Reference

- A. LDK Utilities, Database Upload/Download
- B. LDK Utilities, Software Upgrade

- C. Lock-key operation
- D. Digital Keypad Admin

#### Admin Program

- IP Setting, Server IP Address (PGM 108, Flexible button 2)
  - Assigns an IP address to the system LAN port
- IP Setting, Gateway IP Address (PGM 108, Flexible button 4)
  - Assigns an IP address for the system LAN port gateway
- IP Setting, Subnet Mask (PGM 108, Flexible 5)
  - Assigns the subnet mask to the system LAN port
- PPP Usage (PGM 108, Flexible button 8)
  - When required, configures the PPP connection
- RS-232C Port Settings (PGM 174)
  - Configures the RS-232 serial port and the Modem Unit
- Modem Assignment (PGM 170)
  - Assigns the Associated CO line or station for the Modem Unit

### 17.3.1 Configuration Text File

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#### Description

A text file, containing the basic configuration of a system, is available in real-time when using PC Admin. The file is generated by the system under the Supplementary Service Program group by selecting 'Save' system information. The file can then be named, saved and reviewed. Information contained in the Configuration file includes:

- Number of stations total and in use;
- Physical port for each station;
- Extension types (digital, analog and IP);
- Number of CO lines;
- Physical port for each CO line;
- Trunk types (R2-MFC, Analog, IP and others);
- Applications in-use and available.

#### Consideration

- Database access is allowed to only a single user, DKT or PC Admin at a time; simultaneous database access is not allowed.
- This feature is not available when using Digital Keypad Admin.

#### Reference

- A. PC Admin, On-Line

#### Admin Program

IP Setting, Server IP Address (PGM 108, Flexible button 2)

Assigns an IP address to the system LAN port

IP Setting, Gateway IP Address (PGM 108, Flexible button 4)

Assigns an IP address for the system LAN port gateway

IP Setting, Subnet Mask (PGM 108, Flexible 5)

Assigns the subnet mask to the system LAN port

PPP Usage (PGM 108, Flexible button 8)

When required, configures the PPP connection

RS-232C Port Settings (PGM 174)

Configures the RS-232 serial port and the Modem Unit

Modem Assignment (PGM 170)

Assigns the Associate CO line or station for the Modem Unit



### 17.3.2 LDK Utilities, Database Upload/Download

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#### Description

Using PC Admin, the ipLDK system database can be stored at the PC and a previously stored database can be sent to the ipLDK. Prior to opening the Upload/Download session, the PC Admin application must be opened. Any open connection to the system must be closed. A session can be established over the RS-232 port, Modem or LAN connection (refer to Ref. A). Databases may be interchanged between systems of the same hardware configuration and can employ the same database as a template. A system database should not be downloaded to systems with different hardware configuration as this may cause difficulty changing logical configurations.

#### Operations

For operations, see the PC Admin User Guide

#### Consideration

- The system may require a reset after the database or software is uploaded.
- When upgrading software, do not interrupt the process. When the system returns to normal operation changes can be made to the database, etc.
- Database access is allowed to only a single user, DKT or PC Admin at a time; simultaneous database access is not allowed.
- When uploading a database to an ipLDK, it is recommended that a database of the same hardware configuration be used.
- Should the connection fail during a session, re-open a session
- When using the Modem or LAN connection a system reset will disconnect the active session and a new connection must be established for further PC Admin activity.

#### Reference

- A. PC Admin, On-Line
- B. LDK Utilities, Software Upgrade

#### Admin Program

IP Setting, Server IP Address (PGM 108, Flexible button 2)

Assigns an IP address to the system LAN port

IP Setting, Gateway IP Address (PGM 108, Flexible button 4)

Assigns an IP address for the system LAN port gateway

IP Setting, Subnet Mask (PGM 108, Flexible 5)

Assigns the subnet mask to the system LAN port

PPP Usage (PGM 108, Flexible button 8)

When required, configures the PPP connection

RS-232C Port Settings (PGM 174)

Configures the RS-232 serial port and the Modem Unit

Modem Assignment (PGM 170)

Assigns the Associate CO line or station for the Modem Unit

### 17.3.3 LDK Utilities, Software Upgrade

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#### Description

PC Admin utilities permit changing the system software to upgrade features and functions. Software can be downloaded to the system using the RS-232 port, Modem or LAN connection (refer to Ref. A).

#### Operations

For operations, see the PC Admin User Guide

#### Consideration

- The system may require a reset after the database or software is uploaded.
- When upgrading software, do not interrupt the process. When the system returns to normal operation changes can be made to the database, etc.
- Database access is allowed to only a single user, DKT or PC Admin at a time; simultaneous database access is not allowed.
- Should the connection fail during a session, re-open a session.
- When using the Modem or LAN connection a system reset will disconnect the active session and a new connection must be established for further PC Admin activity.

#### Reference

- A. PC Admin, On-Line
- B. LDK Utilities, Database Upload/Download

#### Admin Program

IP Setting, Server IP Address (PGM 108, Flexible button 2)

Assigns an IP address to the system LAN port

IP Setting, Gateway IP Address (PGM 108, Flexible button 4)

Assigns an IP address for the system LAN port gateway

IP Setting, Subnet Mask (PGM 108, Flexible 5)

Assigns the subnet mask to the system LAN port

PPP Usage (PGM 108, Flexible button 8)

When required, configures the PPP connection

RS-232C Port Settings (PGM 174)

Configures the RS-232 serial port and the Modem Unit

Modem Assignment (PGM 170)

Assigns the Associate CO line or station for the Modem Unit

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## **18 SHORT MESSAGE SERVICE**

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### **18.1 Public SMS**

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#### Description

ipLDK implements the ETSI ES201 912 V1.1.1 standard SMS protocol 1. The ipLDK sends and receives messages from an SMS Service Center. Messages can be sent from any of the LG-Ericsson softphones (ez-Attendant, ez-phone or Phontage) or specially designed SMS SLTs to the service center.

When an SMS is received, the system routes the message based on the assigned SMS Destination. If the message is for a DKT, it is stored until read otherwise; the message is sent to the designated destination and erased from the ipLDK memory.

#### Operation

For operation with an LG-Ericsson softphone or an SMS SLT, refer to the appropriate User guide.

To retrieve SMS from a DKT when the LCD indicates SMS messages are available,

1. Press the [CALLBK] button, a list of messages is displayed.
2. Use the Volume-up and down buttons to scroll the list.
3. Dial the index for the desire message.
4. Delete the message with the Delete softkey or return to the previous display with the Back softkey.

#### Consideration

- In case of power failure, SMS messages are not saved.
- IpLDK-60 can provide simultaneous service to four (4) SMS channels.
- IpLDK-60 can store a maximum of 100 messages. When the buffer is full, a warning is sent to the System Attendant and new messages are not received.
- For an SMS compatible SLT, ipLDK will respond to a maximum of three regeneration requests after which the message is erased from the ipLDK-60 SMS buffer
- The maximum SMS message size for the LG-Ericsson softphones is 160 characters.

#### Admin Programming

SMS Service Center Number (PGM 291, Flexible button 1)

Assigns telephone number of SMS Service Center

SMS Center CLI (PGM 291 Flexible button 2)

Assigns the CLI for the SMS Service Center

SMS Receive Station Assignment (PGM 292 Flexible button 1)

Assigns stations that can receive SMS messages

SMS Outgoing CO (PGM 292 Flexible button 3)

Assigns the CO line used to call the Service center to deliver an SMS



**ip**LDK-60