



# OfficeServ 7000 [V4.40 S/W] Feature Guide

Document  
Type  
Language  
Code /  
Total page  
Create /  
Exam /  
Approve /  
Registered  
Date  
Item Code

This is official SEC document created, registered, and circulated according to the following SEC corporate-wide technical document management regulations DHQ2-0021K.



## Revision History

<b>Edition No.</b>	<b>Data</b>	<b>Author</b>	<b>Modification Details</b>
00	2008.10.22	HyangNae Park	First Edition
01	2008.10.31	HyangNae Park	Update reviews
02	2008.11.01	Harry Kim	Update reviews
03	2008.11.04	Harry Kim	Update reviews of No MGI
04	2008.11.19	Harry Kim	Update final spec of the MOBEX
05	2008.12.08	Harry Kim	Update CLI transfer option for MOBEX.
06	2009.04.02	HyangNae Park	Second Edition (V4.40)



# Table of Contents

<b>1. Overview</b> .....	<b>5</b>
<b>1.1 Purpose</b> .....	<b>5</b>
<b>1.2 Scope and Compatibility Table</b> .....	<b>5</b>
<b>1.3 Restriction</b> .....	<b>5</b>
<b>2. V4.40 S/W Added/Modification Features</b> .....	<b>6</b>
<b>2.1 MOBEX (Mobile Extension) Feature</b> .....	<b>6</b>
2.1.1 MOBEX Station Setup .....	6
2.1.2 MOBEX Standard Feature.....	10
2.1.3 MOBEX Convenient Feature .....	12
2.1.5 DTMF Detection of MOBEX Phone.....	13
2.1.5 MOBEX Executive Feature .....	15
2.1.6 Detailed Service Features of MOBEX .....	17
<b>2.2 Advanced Media Control: No MGI Feature</b> .....	<b>23</b>
2.2.1 No MGI .....	23
2.2.2 MPS (Media Proxy Service) .....	25
2.2.3 SBC Function .....	26
<b>2.3 Advanced License Control</b> .....	<b>28</b>
3.3.1 Old License.....	28
3.3.2 New License .....	28
2.3.3 Temporary License .....	29
<b>2.4 The Other Features</b> .....	<b>31</b>
2.4.1 SVM Options .....	31
2.4.2 ITP Idle Mode Feature.....	31
2.4.3 SIP-FXO link with Remote ITP .....	33
2.4.4 SIP Peering Feature .....	34
2.4.5 Others.....	35



## List of Figures

Figure 1. MENU 6.3.2. MOBEX STN Card Setup .....	7
Figure 2. MENU 2.8.0. MOBEX Station Number Setup.....	7
Figure 3. MENU 4.1.2. Trunk Group Type Setup.....	8
Figure 4. MENU 2.7.5. MOBEX Station Information Setup .....	8
Figure 5. MENU 4.2.5. Ring Group Member Setup.....	10
Figure 6. MENU 2.8.6. MOBEX Feature ID Setup .....	10
Figure 7. MENU 2.1.5. MOBEX CLI Number Option.....	12
Figure 8. MENU 5.2.12. Incoming CLI transfer to MOBEX via SIP trunk.....	12
Figure 9. MENU 2.2.14. OAS Card Setting .....	12
Figure 10. MENU 4.1.14. MGI Groups .....	12
Figure 11. MENU 3.2.3. MOBEX Feature ID Setup to DDI Ringing.....	16
Figure 12. MENU 2.1.5. MOBEX Automatic Certification Setup.....	16
Figure 13. MENU 2.1.5. MPS Service Setup.....	26
Figure 14. MENU 6.2.3. SIP Phone Status.....	26
Figure 15. MENU 2.1.4. License Key Setup.....	29
Figure 16. MENU 5.9.1. SVMi Automatic Generation Option Setup .....	31
Figure 17. MENU 5.15.14. MMS Service Type.....	29

## List of Tables

<a href="#">Table 1. Version Compatibility Table</a> .....	5
<a href="#">Table 2. Detailed MOBEX Features</a> .....	18
<a href="#">Table 3. MGI/MPS Use by Different Call Types</a> .....	23
<a href="#">Table 4. CID Display Options</a> .....	236

## References

1. OfficeServ 7000 [V4.30 S/W] Requirements Document, 2008.
2. OfficeServ 7000 [V4.30 S/W] Detailed Design Document, 2008.
3. OfficeServ 7000 [V4.40 S/W] Requirements Document, 2008.

# 1. Overview

## 1.1 Purpose

This document is to specify the requirements to be applied when testing the OfficeServ 7000 Families V4.40 S/W. Also, the details presented in the V4.40 S/W Requirements section contain the technical items necessary in the development stage of the V4.40 S/W and have been written from the standpoint of users. The system test units and test items for the V4.40 S/W shall be determined based on the requirements specified by OfficeServ 7000 [V4.40 S/W] Requirements Document and V4.30 S/W Features Guide.

This document is not for the real end-users but is made for installation and education of V4.40 S/W. So it should be used only inside company.

## 1.2 Scope and Compatibility Table

Application Scope and Compatibility Table of V4.40 S/W is as table 1.

Table 1. Version Compatibility Table

	OS7400	OS7200		OS7100			OS7030	OS7070
	MP40	MP20	MCP	MP10	MP10a	MP11		
<b>MP</b>	<b>V4.40</b>	-	<b>V4.40</b>	<b>V4.40</b>	<b>V4.40</b>	<b>V4.40</b>	<b>V4.40</b>	<b>V4.40</b>
<b>LP40</b>	<b>V1.21</b>	N/A	N/A	N/A	N/A	N/A	N/A	N/A
<b>LCP</b>	N/A	-	<b>V4.17</b>	N/A	N/A	N/A	N/A	N/A
<b>SVMi-20E</b>	<b>V5.2.1.1</b>	-	<b>V5.2.1.1</b>	N/A	N/A	N/A	N/A	N/A
<b>TEPRI2/a</b>	<b>V4.26</b>	-	<b>V4.26</b>	<b>V4.26</b>	<b>V4.26</b>	<b>V4.26</b>	N/A	N/A
<b>4BRI</b>	<b>V6.20</b>	-	<b>V6.20</b>	N/A	N/A	N/A	N/A	N/A
<b>MGI16/64</b>	<b>V1.24</b>	<b>V1.24</b>	<b>V1.24</b>	<b>V1.24</b>	<b>V1.24</b>	<b>V1.24</b>	<b>V1.24</b>	<b>V1.24</b>
<b>OAS</b>	<b>V1.00</b>	<b>V1.00</b>	<b>V1.00</b>	N/A	N/A	N/A	N/A	N/A
<b>IT</b>	<b>V1.40</b>	<b>V1.40</b>	<b>V1.40</b>	<b>V1.40</b>	<b>V1.40</b>	<b>V1.40</b>	<b>V1.40</b>	<b>V1.40</b>

Related S/W such as ITP Phone should be upgraded to use V4.40 S/W.

## 1.3 Restriction

V4.40 S/W can be used as following cases:

- Install System for the first time.
- Upgrade from V1.0x S/W of OfficeServ 7030/7070.
- Upgrade from V4.2x S/W of OfficeServ 7400/7200/7100.
- In case of upgrading from lower than V4.2x of OfficeServ 7400/7200/7100. You should upgrade S/W into V4.2x and verify its normal operation, then upgrade to V4.30/V4.40 S/W again.
- The OSM and WebMMC for OfficeServ 7200/7400 can not connect to the V4.30/V4.40 system.

## 2. V4.40 S/W Added/Modification Features

This section describes the detail features to be developed by the V4.40 S/W, which are classified into categories.

### 2.1 MOBEX (Mobile Extension) Feature

The following terms shall be defined to explain the Mobile Extension

- MOBEX Number: The phone number of an external device, such as a mobile phone, to be ringing.
- MOBEX Phone: An external phone that has a MOBEX number.
- MOBEX User: A user of a MOBEX phone.
- MOBEX Station: A virtual station to which a MOBEX number is assigned. Only the Executive MOBEX user can use supplementary services.
- Master Station of the MOBEX: A master station which is assigned to MOBEX station. This station is for certificating the use of the MOBEX supplementary services. When the MOBEX user assigns some special features, these features are assigned to this master station.
- Master Station of the Ring Group: A station that assigns the max 5 stations including MOBEX station as its Ring Group members. When user makes a call to Master Station, all the members (including MOBEX station) are ringing simultaneously.

#### 2.1.1 MOBEX Station Setup

##### 1) MOBEX Setup Procedure

The following procedure describes how to set the MOBEX station.

1. In **MENU 6.3.2 Virtual Card Change**, set MOBEX STN Card.
2. In **MENU 2.8.0 Numbering Plan**, input MOBEX station number to the specific channel whose card is assigned for <MOBEX STN>.
3. In **MENU 4.1.2 Trunk Groups**, assign <Group Type> to each group number and input the appropriate trunk number as a member of its trunk group.
4. In **MENU 2.7.5 Mobile Extension**, input the information to each MOBEX station numbers.
5. In **MENU 4.2.5 Ring Group Pair**, input the MOBEX station number as a member of Ring Group Master station
6. In **MENU 2.8.0 Numbering Plan**, input the MOBEX feature ID.

##### 2) MOBEX STN Card Setup

- Select MENU 6.3.2 Virtual Card Change and set MOBEX STN card.
- MOBEX STN Card can be set by the maximum capacity which the system is allowed. In general some Cabinet-Slot is already assigned for MOBEX STN.
- Each System has some specific slots which MOBEX STN can be assigned to. In other words you can assign MOBEX STN to those specific slots in case MOBEX STN is included in Selectable card types.

6.3.2			
Cabinet	Slot	Previous Card	Current Card
6	1	MOBEX STN	MOBEX STN
	2	MOBEX STN	MOBEX STN
	3	MOBEX STN	MOBEX STN
7	1	MOBEX STN	MOBEX STN
	2	MOBEX STN	MOBEX STN
	3	MOBEX STN	MOBEX STN

Figure 1. MENU 6.3.2. MOBEX STN Card Setup

### 3) MOBEX Station Number Setup

- Select MENU 2.8.0 Numbering Plan and set the MOBEX station number.

2.8.0					Cabinet
Slot	Channel No	Port Number	Device	Tel Number	CABINET-6
1	1	288	MOBEX STN	311	
	2	289	MOBEX STN	811	
	3	290	MOBEX STN	312	
	4	291	MOBEX STN	812	
	5	292	MOBEX STN	313	
	6	293	MOBEX STN	813	
	7	294	MOBEX STN	314	
	8	295	MOBEX STN	814	

Figure 2. MENU 2.8.0. MOBEX Station Number Setup

- MOBEX station number makes a pair with its even channel. So it is recommended the users to input MOBEX station number as Figure 2.
- Odd channel (Number 311~314 of Figure 2):
  - It is virtual station for MOBEX feature.
  - Its operation is similar to that of Normal SLI port.
- Even channel (Number 811~814 of Figure 2):
  - It is virtual station for CTI solution.
  - When MOBEX call is operated, internal and external calls are generated. But CTI solution can't process these two kinds of call in a one port for a same time. In this case, even channel is used for controlling external call.
  - It is not necessary to input station number when CTI solution is not used. Because this even channel is only for an external call not an internal call.

#### 4) Trunk Group Setup

- Select MENU 4.1.2 Trunk Groups.
- Assign Group Type to each Group number and input the trunk number as a member of its Trunk Group.
- Trunk members should accord with its Group Type.  
(ex. ISDN  $\Leftrightarrow$  PRI port 701, SIP  $\Leftrightarrow$  SIP trunk port 8501)
- The reason of setting Group Type is for restricting usable trunk type. Because both MOBEX feature and emergency call of Remote ITP use only specific trunk type.  
(ex. MOBEX  $\Leftrightarrow$  PRI/BRI trunk, Remote ITP  $\Leftrightarrow$  SIP trunk)

4.1.2						
Group Number	801	802	803	804	805	
Group Type	ISDN	Normal	SPNET	H.323	SIP	
Group Mode	Sequential	Sequential	Sequential	Sequential	Sequential	
	1	701	731	8301	8401	8501
	2	702	732	8302	8402	8502
	3	703	733	8303	8403	8503

Figure 3. MENU 4.1.2. Trunk Group Type Setup

#### 5) MOBEX Station Information Setup

- Select MENU 2.7.5 Mobile Extension and set the information to each MOBEX station number.

Tel Number	Dial Number		Incoming CLI Number	Executive		Master	Deactivate
	Trunk	Outgoing Digit		User	License		
311	801	01095304296	01095304296	Yes	1	211	No
312	801	01095304004	01095304004	Yes	2	212	No
313	801	01095304439	01095304439	No			No
314	801	01095304292	01095304292	No			No

Figure 4. MENU 2.7.5. MOBEX Station Information Setup

- Input Dial Number as a speed dial format.(Trunk + Outgoing Digit)  
This number is for calling MOBEX phone.
  - Only PRI/BRI or SIP trunk group is allowed to be entered as Trunk. From V4.40

S/W SIP trunk supports DTMF detection. (In V4.30 S/W, MOBEX Executive services are not operated normally in case of entering SIP trunk group. Because V4.30 S/W is not yet supported DTMF detection in case of SIP trunk.)

- From V4.40 S/W, RFC2833 DTMF detection can be supported only through OAS card or MP-embedded MGI/MPS (OS7100/7070/7030).
- LCR code using PRI/BRI or SIP trunk group also can be entered as Trunk. In actual outgoing call, LCR which doesn't use PRI/BRI or SIP trunk group is treated as Error.
- LCR code using PRI/BRI or SIP trunk group also can be entered as Trunk. In actual outgoing call, LCR which doesn't use PRI/BRI or SIP trunk group is treated as Error.
- Users can input individual trunk numbers. But it is not recommended because those individual trunks have a high possibility to be busy. If you assign trunk group numbers as trunk and make an outgoing call, trunk group will automatically search the free trunk number in its members.
- Input MOBEX phone number to Outgoing Digit. (3~16 digits)
- Input CLI number of incoming call to CLI Number. This incoming call is received from MOBEX phone.
  - Generally MOBEX certification flow is needed when using MOBEX Executive Services. But users can dispense with such certification flow when CLI of incoming call is matched with pre-entered CLI Number.
- Select MOBEX station to use Executive Supplementary services.
  - Select whether MOBEX Executive services are used or not.
  - License Priority displays the number of MOBEX Executive License which is needed for MOBEX station to use supplementary services.
- Input Master Station of MOBEX to Master.
  - Master station is the standard number when using Supplementary services.
  - Users should be entered <Station Number + Password> for MOBEX certification. And this Station Number is called Master station.
  - To use Executive services, users should assign this Master station to Master station of Ring Group.
- Users can disable MOBEX station operation setting Deactivate to <YES>.
  - Setting <Yes> or <No> can decide whether MOBEX station is used or not.
  - Even if MOBEX station operation is disabled, License doesn't change.

## 6) Set MOBEX station as a member of Ring Group

- Select MENU 4.2.5 Ring Group Pair and assign MOBEX station to the Ring Group member of Master station.

- MOBEX station can be assigned to Ring Group of another station which is different from Master station of MENU 2.7.5. But to use MOBEX Executive services, users should assign MOBEX station to Master station of MENU 2.7.5.
- Users can assign one individual station to the Ring Group of many stations.
- When Master station of Ring Group is called, its members are also ringing. In this case if one individual member answers, calls of the other members are cleared.

4.2.5						
Group Number	Member					MOBEX Member Ring
	1	2	3	4	5	
211	311	3301				Enable
212	312	3302				Enable
213	313					Enable
214	314					Enable

Figure 5. MENU 4.2.5. Ring Group Member Setup

#### 7) Set Feature ID which is related to MOBEX feature.

- Select MENU 2.8.0 Numbering Plan and input Feature ID number which is related to MOBEX feature.
- FPICK is the feature ID for intercepting call of Ring Group members. In other words, it is for Forced Call Pickup.
- MOBEX is the feature ID for using MOBEX Executive Services. And it is also used when the user wants MOBEX station not to be ringing.

2.8.0					Cabinet	FEATURES
Slot	Channel No	Port Number	Device	Tel Number		
FPICK	169	535	Features	69		
FWD	12	378	Features	60		
MMPG	28	394	Features	54		
MOBEX	168	534	Features	54		
MSG	30	396	Features	43		

Figure 6. MENU 2.8.6. MOBEX Feature ID Setup

### 2.1.2 MOBEX Standard Feature

MOBEX Standard features are operated without License.

### 1) Calling to MOBEX Station

- MOBEX Station operation is similar to that of Normal SLI phone.
- MOBEX Station is called to dial MOBEX Station Number directly.
- MOBEX Station can be called by assigning it to the secondary of Station Pair.  
: In **MENU 4.2.1 Station Pair**, input MOBEX station to Secondary No.
- MOBEX Station can be called by assigning it to the member of Station Group.  
: In **MENU 4.1.1 Station Groups**, input MOBEX station to Member.
- MOBEX Station can be called by assigning it to the member of Ring Group.  
: In **MENU 4.2.5 Ring Group Pair**, input MOBEX station to Member.

### 2) MOBEX Station Monitoring

- MOBEX Station can be called or monitored by assigning it to DS key.  
: In **MENU 4.9.2 Station Key**, assign “DS + MOBEX station” key.
- MOBEX Station can be called or monitored by assigning it to CC key.  
: In **MENU 4.9.2 Station Key**, assign “CC + MOBEX station” key.
- When SMDR is printed out, MOBEX station number is displayed.
- Users can monitor two numbers of MOBEX station pair in case of CTI solution.

### 3) CLI Transfer

- When transferring CLI to MOBEX phone, it is the principle to send the first caller's CLI.
- When MOBEX station receives a trunk call and is ringing, send CLI as following procedures.
  1. If incoming CLI is existed, send it as outgoing CLI.
  2. If there is no incoming CLI and incoming call is related to Call-Forward, DISA or Transfer, send CLI of station which is the destination of Call-Forward/Transfer/DISA as outgoing CLI.
  3. If there is no CLI to transfer, send CLI of incoming trunk or CO Tel Number as outgoing CLI.
- When the individual station calls MOBEX station, send CLI as following procedures.
  1. Send CLI of outgoing station.
  2. If there is no CLI of outgoing station, send the number of outgoing station.
- **Attention: Some of PSTN operators will not accept unconfirmed calling party number (CLI) that they didn't give to customer. In this case, the PSTN operator will be call reject, or will be modify CLI number that they had given to customer.**
- **For this issue, add some options as [Figure 7] and [Figure 8].**

2.1.5		
Item	Value	
Trunk To MOBEX CLI	ISDN	Received
	SIP	Master

Figure 7. MENU 2.1.5. MOBEX CLI Number Option

- When MOBEX outgoing call, the CLI number can select.
  - : In **MENU 2.1.5 Trunk To MOBEX CLI**, assign used CLI number for PRI/BRI trunk outgoing and SIP trunk outgoing.
    - ◆ Received: Send received CLI number from trunk.
    - ◆ Master: Send the Master station's CLI Number in MENU 2.4.3.
    - ◆ MOBEX: Send the MOBEX station's CLI Number in MENU 2.4.3.
  
- When MOBEX outgoing call via SIP trunk, the incoming CLI number can transfer to MOBEX station.
  - : In **MENU 5.2.12 Received CLI Forward On Alias**, if this option set to enable, than the incoming CLI number attach to ALIAS field when SIP trunk outgoing.

5.2.12		
Item	Item	Value
SIP Trunk Configuration	Default SIP Carrier	1
	iBG Expire Time (sec)	10
	CLIR Flag with Number	Disable
	Incoming Mode	Follow DID Trans
	Peer CLI Table	1
	Received CLI Forward On Alias	Disable

Figure 8. MENU 5.2.12. Incoming CLI transfer to MOBEX via SIP trunk

## 2.1.3 MOBEX Convenient Feature

### 1) Stop Simultaneous Ring of MOBEX Station

- It is possible that only MOBEX station which is a member of Ring Group can not be ringing. But in this case MOBEX phone can be ringing by dialing MOBEX station number directly.
- Simultaneous Ring of MOBEX Station can be stopped by following methods.
  - ◆ MOBEX Feature ID (\*\*)+ ON (1): Activate
  - ◆ MOBEX Feature ID (\*\*)+ OFF(0): Deactivate
  - ◆ Users can activate or deactivate this feature conveniently by using MOBEX key. MOBEX key should be assigned to Master station.
- In MENU 4.2.5 Ring Group Pair, MOBEX Member Ring can be disabled. In this case



only MOBEX station is not ringing. (MOBEX station should be a member of Master station.)

- MOBEX Executive Users can activate or deactivate this feature by making a call to the system.
- LED of MOBEX key displays whether MOBEX station is ringing or not.

## **2) Stop Use of MOBEX Phone**

- In MENU 2.7.5 Mobile Extension, Deactivate field can be set to <YES>. In this case MOBEX station operation is disabled and MOBEX phone can't receive all the incoming calls.

## **3) Call Intercept (Forced Call Pickup)**

- When the individual station is busy, a member station belonging to the same Ring Group or its master station can intercept that call.
- MOBEX call can be transferred to the normal official phone by using this feature. The reverse case is also possible.
- When MOBEX station is busy, master station can intercept the call by dialing following codes. (Master station should have a MOBEX station as its Ring Group member.)
  - ◆ FPICK Feature ID (69) + Ring Group Master Station
  - ◆ FPICK Feature ID (69) + MOBEX Station in Busy
- Another member of Ring Group can intercept the call by dialing following codes.
  - ◆ FPICK Feature ID (69) + Ring Group Master Station
  - ◆ FPICK Feature ID (69) + MOBEX Station in Busy
- SIP Phone can't use this feature.

## **4) Change MOBEX Phone Number**

- In MMC 126, users can change MOBEX phone number and its CLI number. To use this MMC, MOBEX station should be registered as a member of the Ring Group Master station.
- Warning: If users assign a wrong number, MOBEX station could be operated abnormally. So be sure to register correct MOBEX phone number.

## **2.1.5 DTMF Detection of MOBEX Phone**

### **1) MOBEX DTMF Receiver**

- To use MOBEX Executive services, system needs extra receiver which can detect DTMF during call.
- Extra receiver is needed because existing DTMF receiver is only planned for the most suitable capacity to make an outgoing call or use DISA.

### **2) MP Embedded Receiver**

- OfficeServ 7100/7070 system supports 8 channels of receiver without installing

extra H/W.

- OfficeServ 7030 system supports 4 channels of receiver without installing extra H/W.
- 8 channels of receiver can make 8 MOBEX calls for a same time.

### 3) OAS Card Receiver

- OfficeServ 7200/7400 system needs an extra receiver card which is called OAS Card.
- From V4.40 S/W, OAS card can be installed in OfficeServ 7100.
- OAS Card is operated as DTMF receiver with channel numbers such as 64, 32 or 16. It is recommended that OAS card should be installed in 64 channel-slot in case of OfficeServ 7400 and 32 channel-slot in case of OfficeServ 7200.
- There is no restriction how many OAS Card can be installed. But it is recommended that channels should be ensured as much as numbers of MOBEX Executive user.
- It is recommended that minimum channels should be ensured as much as numbers of PRI/BRI trunk.
- If MOBEX Executive user can't use supplementary services due to lack of DTMF receiver, no indication of such circumstance is provided.
- OAS Card setting
  - ◆ OAS card can be used by mixing MFR (for MOBEX) and MGI. In MENU 2.2.14 OAS Card, user can assign the channel of MFR/MGI.
  - ◆ Configuration
    - 1) OAS card is installed in 64 channel-slot of OS7400.
      - 0. MFR 64 CHs – default
      - 1. MFR 48 CHs + MGI 4 CHs
      - 2. MFR 32 CHs + MGI 8 CHs
      - 3. MFR 16 CHs + MGI 12 CHs
    - 2) OAS card is installed in 32 channel-slot of OS7400 or OS7200
      - 0. MFR 32 CHs – default
      - 1. MFR 28 CHs + MGI 4 CHs
      - 2. MFR 24 CHs + MGI 8 CHs
      - 3. MFR 16 CHs + MGI 12 CHs
- MGI is restricted by License but MGI16/64 is operated without License.

2.2.14.	
Item	Value
Service Select	MGI:12+MOBEX:16
Channel Type	MOBEX:64 ONLY
Channel No	MGI:04+MOBEX:48
1	MGI:08+MOBEX:32
2	MGI:12+MOBEX:16
3	MGI:16 ONLY
4	Idle

Figure 9. MENU 2.2.14. OAS Card Setting

- In case of using MGI, user should set MGI port and user configuration in MENU 2.8.0 Numbering Plan and MENU 4.1.4 MGI Groups.

4.1.4		Local ITP	Public IP ITP	VoIP Networking	Public IP Networking	VoIP Trunk	Public IP Trunk	ITP Paged
Group User	Mode	Distribute	Distribute	Distribute	Distribute	Distribute	Distribute	Distribute
	1	3801	3801	3801	3801	3801	3801	3801
	2	3802	3802	3802	3802	3802	3802	3802
	3	3803	3803	3803	3803	3803	3803	3803
	4	3804	3804	3804	3804	3804	3804	3804
	5	3805	3805	3805	3805	3805	3805	3805
	6	3806	3806	3806	3806	3806	3806	3806
	7	3807	3807	3807	3807	3807	3807	3807
	8	3808	3808	3808	3808	3808	3808	3808
	9	3809	3809	3809	3809	3809	3809	3809
	10	3810	3810	3810	3810	3810	3810	3810
	11	3811	3811	3811	3811	3811	3811	3811
	12	3812	3812	3812	3812	3812	3812	3812

Figure 10. MENU 4.1.14. MGI Groups

#### 4) Trunk Type Restriction for MOBEX Executive Service

- MOBEX Executive services are provided by both PRI/BRI trunk and SIP trunk.
- From V4.40 S/W, SIP trunk can support MOBEX Executive Service. But only RFC2833 DTMF can be detected by using OAS card or MP-Embedded MGI/MPS (OS7100/7070/7030). (In V4.30 S/W, SIP trunk can't support MOBEX Executive Services because DTMF signal of MOBEX phone comes to the system mixing with voice signal.(inband) )

### 2.1.5 MOBEX Executive Feature

#### 1) MOBEX Executive User Setup

- Select MENU 2.7.5 and set <Executive User> field to <Yes>.
- Even if MOBEX station is designated as a MOBEX Executive user, all the Executive users can't set supplementary services. Only the users who are not over the numbers of MOBEX Executive License can use supplementary services.
- Basically one MOBEX station is designated as a MOBEX Executive user without License.

#### 2) MOBEX Feature during call

- MOBEX Executive User can be provided supplementary services such as Transfer, Hold, and Conference during call.
- If MOBEX user dials MOBEX Feature ID(\*\*), the opposite party will be hold and the MOBEX user hear transfer tone. In this case MOBEX user can transfer the call to another station by dialing station number.
- If MOBEX user dials MOBEX Feature ID(\*\*), MOBEX phone will have a same operation with Hook Flash of Normal SLI phone.
- MOBEX Features can be set by same methods which is similar to those of Normal SLI Phone.

- For example, Transfer can be set by dialing following codes during call.
  - ◆ MOBEX Feature ID(\*\*) + Station + On Hook: Unscreened transfer
  - ◆ MOBEX Feature ID(\*\*) + Station + Answer/Talking + On Hook: Screened transfer
  - ◆ MOBEX Feature ID(\*\*) + Station + Answer/Talking + MOBEX Feature ID(\*\*): Consultation transfer
  - ◆ MOBEX Feature ID(\*\*) + Station + MOBEX Feature ID(\*\*): Transfer Cancel
  - ◆ If user dials MOBEX Feature ID(\*\*) and hangs up the phone while dialing station number, call on hold will not recall and just be cleared.

3.2.3.								
Entry Number	Incoming digits	Ring Plan						Translate Name
		1	2	3	4	5	6	
1	299	stc	stc	stc	stc	stc	stc	MOBEX
2	2 <sup>stc</sup>	B	B	B	B	B	B	
3	3 <sup>stc</sup>	B	B	B	B	B	B	
4	5 <sup>stc</sup>	B	B	B	B	B	B	

Figure 11. MENU 3.2.3. MOBEX Feature ID Setup to DDI Ringing.

### 3) MOBEX Feature ID Setup to DDI Ringing

- In MENU 3.2.3 DDI Ringing, input MOBEX Feature ID(\*\*) to Ring Plan of specific Incoming digits. (ex. 299)

### 4) MOBEX Incoming Certification

- When the DDI call is incoming and its DDI destination number is same with MOBEX Feature ID, that call is recognized as MOBEX incoming call and MOBEX Certification Flow will be operated. (Refer to Figure 9)
- MOBEX Incoming Certification is operated by inserting "Station + Password".
- Station is Master station of the MOBEX station. In this case set MOBEX station as a member of the Ring Group Master station.
- Password can be set in Key MMC 101. Like DISA password, default password (1234) can't be used in MOBEX Incoming Certification.

2.1.5.		
Item	Value	
MOBEX Executive Option	Auth by CLI	On
	Auth Hold Tone	On
	Tone Source	371
	BLF by CLI	Off

Figure 12. MENU 2.1.5. MOBEX Automatic Certification Setup

### 5) Automatic Certification by CLI

- In MENU 2.1.5 System Options, set <Auth by CLI> to On. In this case MOBEX Incoming call is automatically certified without inserting "Station + Password".
- When Auth by CLI is set to On and DDI call whose destination is same with MOBEX Feature ID(\*\*) is incoming, that call is automatically certified by Master station of the MOBEX station. In this case incoming CLI should be matched with CLI Number of MENU 2.7.5. Otherwise the incoming call is failed to certify MOBEX Incoming.

### 6) MOBEX Incoming Certification Flow

- MOBEX Incoming Certification Flow is as follows:
  1. When DDI call whose destination is MOBEX Feature ID (\*\*) is incoming, user can hear dial tone for the next certification flow.
  2. Dial Master of MOBEX station number. Then confirm tone is heard.
  3. Dial Password of Master station. Then dial tone is heard. Certification is completed.
  4. When Auth by CLI is set to On, Step 2.3 is omitted and dial tone is heard.
  5. After completing certification, user can make a call to other station/trunk or set the supplementary services on its Master station.
- In MENU 2.1.5, set Auth Hold Tone to ON. In this case, dial tone of step 1 is alternated by Tone Source of MENU 2.1.5. Hold Tone or MOH can be selected as Tone Source.
- After completing certification, user can hear dial tone. This state is same with that of Normal SLI phone states off hook.
- After completing certification, MOBEX user can set supplementary services on its Master station or make a call of internal or external.

### 7) MOBEX Station Monitoring

- State of MOBEX station can be monitored by using DS+MOBEX key. In this case only the call which is generated by MOBEX feature is possible to be monitored.
- In MENU 2.1.5, set BLF by CLI to On. In this case, BLF service is provided to Normal DDI incoming call.
  - ◆ When DDI call whose destination is MOBEX Feature ID(\*\*) is incoming:
    - BLF Service is only provided to certified MOBEX station.
    - DS/CC key and CTI solution is possible to use.
  - ◆ When DDI call whose destination is not MOBEX Feature ID(\*\*) is incoming:
    - That call is incoming as normal DDI call and BLF Service is only provided to DS Key of MOBEX station whose CLI is same with DDI call.
    - CC key and CTI solution is not service.
    - Direct pickup by DS key is not operated.

## 2.1.6 Detailed Service Features of MOBEX

Following [TABLE 1] describes the detailed service features of MOBEX and how to use these features. Some features are allowed to use to both Standard and Executive users but most of

features are provided to only Executive users. To certificate as a MOBEX phone, MOBEX user must use ISDN trunk or SIP trunk. SIP trunk is allowed to use from V4.40 S/W.

TABLE 2. Detailed MOBEX Features

NO	FEATURE	STD / EXE	Procedures
1	Call Forward Follow-Me	EXE	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification input the following codes to set Follow-Me FWD on Master station. =&gt; <b>Call Forward Feature ID&lt;60&gt;+Follow-Me&lt;6&gt;+ station&lt;A&gt;</b></li> <li>•Incoming call to station A will be forwarded to Master station.</li> </ul>
2	Call Forward on Busy or No answer	STD	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification input the following codes to set Busy or No answer FWD on Master station. =&gt; <b>Call Forward Feature ID &lt;60&gt;+Forward type &lt;2(Busy)/3(NoAns)/4(B&amp;NoAns)&gt;+station&lt;A&gt;</b></li> </ul>
3	Call Forward on DND	STD	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification input the following codes to set DND FWD on Master station. =&gt; <b>Call Forward Feature ID&lt;60&gt;+DND &lt;5&gt;+ station &lt;A&gt;</b></li> </ul>
4	Call Forward Unconditional	STD	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification input the following codes to set Unconditional FWD on Master station. =&gt; <b>Call Forward Feature ID&lt;60&gt;+All &lt;1&gt;+ station &lt;A&gt;</b></li> </ul>
5	Call Intercept of MOBEX call by FPICK key	STD	<ul style="list-style-type: none"> <li>•When MOBEX user is busy, one of members of Ring Group can intercept the MOBEX call by dialing <b>&lt;FPICK Feature ID + Master station of Ring Group&gt;</b>.</li> <li>Or in the same circumstances, Master station of Ring Group also can intercept the MOBEX call by dialing <b>&lt;FPICK Feature ID + MOBEX station&gt;</b>.</li> </ul>
6	Call pickup of MOBEX ringing	STD	<ul style="list-style-type: none"> <li>•When the MOBEX user receives a call from station A and the MOBEX phone is ringing, input <b>DIRPK Feature ID &lt;65&gt;+ MOBEX station</b>.</li> </ul>
7	Class of Service	STD	<ul style="list-style-type: none"> <li>•Select <b>MENU 4.7.2 COS Contents</b> and set COS of Master station.</li> <li>•When the call is outgoing to MOBEX station or is incoming to System from MOBEX Executive user, that call is operated by following COS of Master station.</li> </ul>
8	Direct Station Select	STD	<ul style="list-style-type: none"> <li>•Select <b>MENU 4.9.2 Station Key</b> and make &lt;DS + MOBEX station&gt; button.</li> </ul>
9	Do not Disturb and Override	STD	<ul style="list-style-type: none"> <li>•Select <b>MENU 4.7.2 COS Contents</b> and set &lt;DND OVRD&gt; of chosen COS to &lt;YES&gt;.</li> <li>•After finishing MOBEX certification, make a call to station A which is set to DND.</li> <li>•When hearing DND Busy tone, input <b>&lt;DND OVRD Feature ID&gt;</b>.</li> </ul>
10	Forced Trunk Release	STD	<ul style="list-style-type: none"> <li>•Select <b>MENU 6.3.5 Port Clear</b> and set &lt;Clear&gt; of MOBEX station to &lt;YES&gt;.</li> </ul>
11	In/Out of Group	STD	<ul style="list-style-type: none"> <li>•Select <b>MENU 4.1.1</b> and register Master station of MOBEX as a member of station group A.</li> <li>•After finishing MOBEX certification input the following codes. =&gt; <b>IG Feature ID&lt;53&gt;+Station Group&lt;A&gt;+IN&lt;1&gt;</b> <b>IG Feature ID&lt;53&gt;+Station Group&lt;A&gt;+OUT&lt;0&gt;</b></li> </ul>
12	Make calls via LCR	STD	<ul style="list-style-type: none"> <li>•Select <b>MENU 3.1.1 LCR Options</b> and set &lt;LCR Enable&gt; to &lt;On&gt;.</li> <li>•Select <b>MENU 3.1.2 Routing Digits</b> and input the information to &lt;LCR Digit&gt;, &lt;Length&gt; and &lt;Routing Table&gt;. (ex. LCR Digit : 27, Length : 2, Routing Table : 1)</li> <li>•Select <b>MENU 3.1.4 Routing Table</b> and input trunk group number to the</li> </ul>

			<p>specific Zone. This zone should be same with &lt;Routing Table&gt; of MENU 3.1.2 (ex. Group : 801) =&gt; You should select PRI/BRI or SIP trunk group to make a normal MOBEX station call. (ex. ISDN trunk group : 801, SIP trunk group : 805) •Select &lt;FEATURES&gt; of <b>MENU 2.8.0</b> and input LCR Feature ID (ex. LCR : 69) •Select &lt;N-LCR&gt; of <b>MENU 2.8.0</b> and input N-LCR number (ex. N-LCR : 27) •Select <b>MENU 3.3.2 Network Dial Translation</b> and input &lt;Translation Digits&gt; of specific N-LCR. This N-LCR number is designated in MENU 2.8.0. Input &lt;Wait Length&gt; and &lt;Max Digit&gt;. Wait Length is the minimum digits to be recognized with N-LCR and Max Digit is for linking with the opposite station by LCR. (ex. Translation digits: 272, WaMENU Length : 2, Max DigMENU : 3). •Input LCR feature ID to &lt;Trunk No&gt; of <b>MENU 2.7.5</b>. Input opposite station number to &lt;Outgoing Digit&gt; for linking by LCR. (ex. Trunk No : 69, Outgoing Digit : 272) =&gt; If user does not assign ISDN/SIP trunk to <b>MENU 3.1.4</b> and calls MOBEX station, error message such as &lt;Not Available&gt; will be displayed.</p>
13	Make calls via OS Operator/Call	STD	•OS Operator or call dials Master station of Ring group number of MOBEX station number.
14	Monitoring on DS or CC key	STD	•Select MENU 4.9.2 Station key, make a DS+MOBEX or CC+MOBEX key. •When MOBEX station is busy, the DS or CC key LED On.
15	Monitoring via OS Operator/Call	STD	•In OS Operator or Call's key, make a STN+MOBEX. •When MOBEX station is busy, the STN key LED On.
16	Originators CLI Transfer	STD	•When external user A makes a trunk call to MOBEX station, (1) If incoming CLI is existed, send it as outgoing CLI. (2) If there is no incoming CLI, send CLI of Master station. => In <b>MENU 2.4.3 CLI Send Number</b> , CLI of master station should be registered to <Send CLI Number>. (3) If there is no CLI master station, send master station Tel number.
17	Preset Forward No Answer	STD	•Select <b>MENU 2.5.4 Preset Forward</b> , and input station A to <Forward Port> of MOBEX station. •When MOBEX station is busy or has no answer, incoming call of MOBEX station is forwarded to station A.
18	Station Group Membership	STD	•Select <b>MENU 4.1.1 Station Groups</b> and input ring group master station of MOBEX station to <Member> of station group A.
19	Ring Group	STD	•Select <b>MENU 4.2.5 Ring Group Pair</b> , and input the max 5 stations (including MOBEX station) to <Member> of Ring Group Master station.
20	Station pairing	STD	•Select <b>MENU 4.2.1 Station Pair</b> , and input ring group master station of MOBEX station to <Primary No> of station A.
21	Station Programmed message	STD	Leave Station Message •After finishing MOBEX certification, make a call to station A. •When station A is busy or has no answer, MOBEX user dials <b>MSG Feature ID &lt;43&gt;</b> to leave station message. In this case, the station has station message from the master station of MOBEX station.
22	Voice Mail Indication	EXE	
23	Account codes voluntary	EXE	•Select <b>MENU 4.5.2 Account Code</b> , and input Account codes to the individual entry.

			<ul style="list-style-type: none"> <li>After finishing MOBEX certification input the following codes. =&gt; <b>ACCT Feature ID&lt;47&gt;+&lt;Entry Number&gt;</b></li> <li>Make an outgoing trunk call when hearing confirm tone.</li> </ul>
24	Conference	EXE	<ul style="list-style-type: none"> <li>MOBEX user dials <b>&lt;MOBEX Feature ID&gt;</b> while talking with station A.</li> <li>Dial <b>CONF Feature ID &lt;46&gt;</b> and then make an outgoing call to station B.</li> <li>Dial <b>&lt;MOBEX Feature ID&gt;+&lt;CONF Feature ID&gt;</b> to make a conference while talking with station B.</li> </ul>
25	Holding Calls	EXE	<ul style="list-style-type: none"> <li>MOBEX user dials <b>&lt;MOBEX Feature ID&gt;+ HOLD Feature ID &lt;11&gt;</b> while talking with station A.</li> </ul>
26	Transfer	EXE	<ul style="list-style-type: none"> <li>MOBEX user dials <b>&lt;MOBEX Feature ID&gt;</b> while talking with station A.</li> </ul>
27	Transfer with Camp-On or Remote Hold	EXE	<ul style="list-style-type: none"> <li>MOBEX user dials <b>&lt;MOBEX Feature ID&gt;+ station &lt;B&gt; in Busy state</b> while talking with station A.</li> <li>Dial <b>CAMP Feature ID &lt;45&gt;</b> when hearing busy tone. Then Camp-on is applied to station B.</li> </ul>
28	Transfer with Consultation	EXE	<ul style="list-style-type: none"> <li>MOBEX user dials <b>&lt;MOBEX Feature ID&gt;</b> while talking with station A.</li> <li>Make an outgoing call to station B. When station B answers the phone, station A is on hold.</li> <li>When MOBEX user dials <b>&lt;MOBEX Feature ID&gt;</b> again station A is on the phone and station B is on hold.</li> <li>=&gt; Whenever MOBEX user dials MOBEX Feature ID, station A and B are transferred by turns.</li> </ul>
29	Activate/Deactivate MOBEX station	EXE	<ul style="list-style-type: none"> <li>Select <b>MENU 2.7.5</b>, and set &lt;Status&gt; of MOBEX station to Deactivate.</li> </ul>
30	Set Call forward	EXE	<ul style="list-style-type: none"> <li>After finishing MOBEX certification input the following codes to set FWD on Master station. =&gt; <b>Call Forward Feature ID &lt;60&gt;+&lt;FWD Type Feature ID&gt;+FWD destination &lt;A&gt;</b> (FWD Type Feature ID :: All (1), Busy (2), NoAns (3), Busy/NoAns (4), DND (5))</li> </ul>
31	Set Call Forward Follow-Me	EXE	<ul style="list-style-type: none"> <li>After finishing MOBEX certification input the following codes to set FWD on Master station. =&gt; <b>Call Forward Feature ID &lt;60&gt;+Follow-Me &lt;6&gt;+Station &lt;A&gt;</b></li> </ul>
32	Set Do Not Disturb	EXE	<ul style="list-style-type: none"> <li>After finishing MOBEX certification input the following codes to set DND on Master station. =&gt; <b>DND Feature ID&lt;40&gt;+ On&lt;1&gt; : activate DND</b> <b>DND Feature ID&lt;40&gt;+ Off&lt;0&gt; : deactivate DND</b></li> </ul>
33	Link/Unlink Ring Group	EXE	<ul style="list-style-type: none"> <li>Dial <b>&lt;MOBEX ID + 0(Off)&gt;</b> on Master station. When call is incoming to Master station, MOBEX station which is one of the member of Ring Group is not ringing.</li> <li>Dial <b>&lt;MOBEX ID + 1(On)&gt;</b> on Master station. In this case simultaneous Ring of MOBEX station will be supported again.</li> <li>When made a <b>&lt;MOBEX&gt;</b> key on Master station of ring group, press MOBEX key then toggle ringing or not ringing to MOBEX.</li> </ul>
34	Set Station Programmed message	EXE	<ul style="list-style-type: none"> <li>Select <b>MENU 4.7.2 COS Contents</b> and set &lt;PGM MSG&gt; of individual COS to YES.</li> <li>After finishing MOBEX certification input the following codes to set Station Programmed message on Master station. =&gt; <b>PMSG Feature ID &lt;48&gt;+PSMG Number &lt;01~20&gt;</b></li> <li>If you want to clear designated PMSG, input the following codes =&gt; <b>PMSG Feature ID &lt;48&gt;+&lt;00&gt;</b></li> <li>PGM MSG is displayed on the LCD when another station makes an outgoing call to Master station.</li> </ul>
35	Station Message	EXE	<ul style="list-style-type: none"> <li>After finishing MOBEX certification, input <b>CANMG (=cancel msg)</b></li> </ul>

	Cancel or Set on Idle		<b>Feature ID &lt;42&gt;+ station &lt;A&gt;</b> then cancelled station A left message from master station of MOBEX.
36	Voice Mail Access	EXE	<ul style="list-style-type: none"> <li>•Station A calls VM and can leave VM message to master station of MOBEX station.</li> <li>•After finishing MOBEX certification, dials to VM then the MOBEX user access to master station's mailbox.</li> </ul>
37	Account codes	EXE	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification input the following codes. =&gt; <b>ACCT Feature ID&lt;47&gt;+&lt;Account Code&gt;+ACCT Confirm digit &lt;*(star key)&gt;</b></li> </ul>
38	Authorization codes	EXE	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification input the following codes. =&gt; <b>&lt;AUTH Feature ID&gt;+&lt;Auth Code&gt;</b></li> </ul>
39	Barge-In on Busy	EXE	<ul style="list-style-type: none"> <li>•Select <b>MENU 5.13.1 Tenant Options</b> and set &lt;Barge In Type&gt; to &lt;With Tone&gt; or &lt;Without Tone&gt;.</li> <li>•Select <b>MENU 4.7.2 COS Contents</b>. Set &lt;OVERRIDE&gt; to YES and &lt;SECURE&gt; to NO.</li> <li>•After finishing MOBEX certification, call station A which is in busy. Then input <b>&lt;Barge Feature ID&gt;</b> while hearing busy tone.</li> </ul>
40	Call Paging to Internal or External	EXE	<ul style="list-style-type: none"> <li>•Select <b>MENU 4.1.3 Page Groups</b> and input Member to Internal Zone(0~4) or External Zone(5~8)</li> <li>•After finishing MOBEX certification input the following codes. =&gt; <b>PAGE Feature ID &lt;55&gt;+Zone No.&lt;0~8&gt;</b></li> </ul>
41	Call Pickup Ringing calls	EXE	<ul style="list-style-type: none"> <li>•Select MENU 2.5.1 and set &lt;Pickup Group&gt;.</li> <li>•After finishing MOBEX certification input the following codes while station A is ringing. =&gt; <b>DIRPK Feature ID &lt;65&gt;+ Station &lt;A&gt;</b> Or <b>GRPK Feature ID &lt;66&gt;+ Pickup Group No. of Station &lt;A&gt;</b></li> </ul>
42	Camp-on Busy	EXE	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification call station A which is in busy.</li> <li>•When hearing busy tone input <b>CAMP Feature ID &lt;45&gt;</b>.</li> </ul>
43	Distinctive Ring on Intercom call	EXE	<ul style="list-style-type: none"> <li>•Select <b>MENU 2.4.2 Port Common Data</b> and set &lt;Distinct DGP Tone&gt; and &lt;Distinct SLT Ring&gt; on each station.</li> </ul>
44	Intercom call	EXE	<ul style="list-style-type: none"> <li>•Select <b>MENU 5.6.3 SMDR Options</b> and set &lt;Intercom Call&gt; to YES.</li> <li>•Select <b>MENU 2.5.3 Customer On/Off</b> and set &lt;Intercom SMDR&gt; to On.</li> <li>•Call log between station A and B is printed out on SMDR.</li> </ul>
45	Make Outside Calls	EXE	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification, make an outgoing trunk call.</li> </ul>
46	Make call by Speed Dial	EXE	<ul style="list-style-type: none"> <li>•Select <b>MENU 5.15.7 Station Speed Dial</b> and set Speed dial of Master station.</li> <li>•After finishing MOBEX certification input the following codes. =&gt; <b>Speed Feature ID &lt;16&gt;+ Speed dial No. of Master station.</b></li> </ul>
47	Meet Me Page and Answer	EXE	<ul style="list-style-type: none"> <li>•Select <b>MENU 4.1.3 Page Groups</b> and input Member to Internal Zone(0~4) or External Zone(5~8).</li> <li>•After finishing MOBEX certification, dial <b>MMPG Feature ID &lt;54&gt;+ Zone No.(0~8)</b></li> </ul>
48	Retrieve Calls on Hold or Parked	EXE	<ul style="list-style-type: none"> <li>•MOBEX user dials <b>&lt;MOBEX Feature ID&gt;</b> to retrieve call on Hold or Parked.</li> </ul>
49	Station Call Back on Busy or No Answer	EXE	<ul style="list-style-type: none"> <li>•When station A call to MOBEX station and hears busy tone or ring back tone, press CBK button then Callback set to the MOBEX station.</li> <li>•When the MOBEX station goes to idle state, the station A receive callback ringing for MOBEX station.</li> </ul>
50	Station Message Reply	EXE	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification input <b>MSG Feature ID &lt;43&gt;</b> while hearing message wait tone. =&gt; In this case MOBEX station makes an outgoing call to the station which is left station message on it.</li> </ul>

51	Station Message Set on Busy/Ringback	EXE	<ul style="list-style-type: none"> <li>•After finishing MOBEX certification call station A.</li> <li>•When station A is busy or has no answer, MOBEX user dials <b>MSG Feature ID &lt;43&gt;</b>.</li> </ul>
52	Toll Restriction	EXE	<ul style="list-style-type: none"> <li>•Select <b>MENU 4.8.5 Trunk Toll Type</b> and set Toll Class of trunk A.</li> <li>•After finishing MOBEX certification, make an outgoing call with trunk A. In this case the call is restricted or allowed by Toll Class of trunk A</li> </ul>
53	Universal Answer	EXE	<ul style="list-style-type: none"> <li>•Select <b>MENU 5.13.1 Tenant Options</b> and input station A to &lt;Universal Answer Device&gt;.</li> <li>=&gt; Call of station A will be picked up by MOBEX user.</li> <li>•After finishing MOBEX certification input <b>UA Feature ID &lt;67&gt;</b>.</li> <li>•When station A is ringing, MOBEX user can pick up the call of station A.</li> </ul>

## 2.2 Advanced Media Control: No MGI Feature

### 2.2.1 No MGI

#### 1) No MGI Overview

- MGI is no more used for a call between IP devices.
- It is the principle of V4.30/V4.40 S/W that calls between IP devices use no MGI.
- MGI can be alternated by MPS in case of VoIP trunk or NAT circumstances which need a device to deliver RTP data.
  - ◆ MGI: RTP -> PCM -> (Time Switch) -> PCM -> RTP  
 There are 2 PCM conversions in this process. Voice quality is worse due to these 2 conversions and it is wasteful to use 2 MGI channels.
  - ◆ MPS: RTP -> (NAT Translation) -> RTP  
 There is no PCM conversion in this process. So voice quality has no deterioration and only 1 MPS call is used.
- If the remote ITP is video phone, video calls as well as voice calls are possible to use MPS (Media Proxy Server).

Table 3. MGI/MPS Use by Different Call Types

		PCM Device		Local IP Device				Remote			Same Network Trunk			Others Network Trunk			
		STN	TRK	ITP	WIP	SIP	UMS	IVR	ITP	WIP	SIP	SIP	H.323	SPNet	SIP	H.323	SPNet
PCM Device	STN	No	No	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI
	TRK	No	No	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI
Local IP Device	ITP	MGI	MGI	No	No	No	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
	WIP	MGI	MGI	No	No	No	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
	SIP	MGI	MGI	No	No	No	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
	UMS	MGI	MGI	MGI	MGI	MGI	N/A	N/A	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI
	IVR	MGI	MGI	MGI	MGI	MGI	N/A	N/A	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI
Remote	ITP	MGI	MGI	MPS	MPS	MPS	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
	WIP	MGI	MGI	MPS	MPS	MPS	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
	SIP	MGI	MGI	MPS	MPS	MPS	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
Same Network Trunk	SIP	MGI	MGI	MPS	MPS	MPS	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
	H.323	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI
	SPNet	MGI	MGI	MPS	MPS	MPS	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
Others Network Trunk	SIP	MGI	MGI	MPS	MPS	MPS	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
	H.323	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI	MGI

	SPNet	MGI	MGI	MPS	MPS	MPS	MGI	MGI	MPS	MPS	MPS	MPS	MGI	MPS	MPS	MGI	MPS
--	-------	-----	-----	-----	-----	-----	-----	-----	-----	-----	-----	-----	-----	-----	-----	-----	-----

## 2) No MGI Mode

- MPS is used for the following two kinds of call instead of MGI.
  - ◆ Calls between Local ITP and Remote ITP
  - ◆ Calls between IP station and IP trunk
- But calls of IP-UMS and IP-IVR should use MGI.
- No MGI Mode has 3 kinds of mode such as No MGI, MPS and MGI mode. Detailed things are described as follows:
  - ◆ No MGI mode: This mode is for calls between IP stations in same IP network.  
Ex) Local ITP – Local ITP, Public ITP – Public ITP
  - ◆ MPS mode: This mode is for calls between IP trunks or ITP devices in NAT circumstance.  
Ex) Local ITP – Remote ITP, Local ITP – SIP/SPNet
  - ◆ MGI mode: This mode is for calls between PCM station/trunk and IP station/trunk.  
Ex) DGP – ITP, DGP – SIP/SPNet, PRI – ITP, PRI – SIP/SPNet

## 3) Difference between No MGI Mode and MPS Mode

- Calls between IP stations have 2 kinds of mode such as No MGI and MPS Mode. These modes are distinguished by environment in which NAT is used or not.
  - ◆ If system has an environment in which NAT is used,
    - Local ITP/WIP/SIP - Local ITP/WIP/SIP : No MGI
    - Local ITP/WIP/SIP - Public ITP/WIP/SIP : MPS
    - Local ITP/WIP/SIP - Public/firewall ITP/WIP : MPS
    - Public ITP/WIP/SIP - Public ITP/WIP/SIP : No MGI
    - Public ITP/WIP/SIP - Public/firewall ITP/WIP : MPS
    - Public/firewall ITP/WIP - Public/firewall ITP/WIP : MPS
    - Public/firewall ITP - Public/firewall ITP (same network) : No MGI
  - ◆ If system has no environment in which NAT is used,
    - Local ITP/WIP/SIP - Public/firewall ITP/WIP : MPS
    - Public/firewall ITP/WIP - Public/firewall ITP/WIP : MPS
    - Public/firewall ITP - Public/firewall ITP (same network) : No MGI
- Calls between IP station and IP trunk always use MPS.
  - ◆ If system has an environment in which NAT is used,
    - Local ITP/WIP/SIP - SIP/SPNet : MPS
    - Public ITP/WIP/SIP - SIP/SPNet : MPS
    - Public/firewall ITP/WIP - SIP/SPNet : MPS
  - ◆ If system has no environment in which NAT is used,
    - Local ITP/WIP/SIP - SIP/SPNet : MPS
    - Public/firewall ITP/WIP - SIP/SPNet : MPS

- Exception
  - ◆ H.323 Trunk always uses MGI.
  - ◆ SIP station of Public/firewall Mode is not supported. In other words SIP station is operated by Public Mode.
  - ◆ WiFi station of Public/Firewall Mode always uses MPS.
  - ◆ ITP of Public/Firewall Mode is operated by No MGI Mode refer to MGI/MPS option of MMC861 when it is in the same NAT circumstance.

## 2.2.2 MPS (Media Proxy Service)

### 1) MPS Overview

- MPS redirects the RTP data for the call between IP Phone of Private IP Network and that of Public IP Network.
- Even if IP station and IP trunk are in the same IP network, it is recommended that the call between them should use MPS. If MPS is not used in this case, call flow will be changed frequently when using Transfer or Hold features and due to this problem the risk of cutting call flow will rise highly.
- Describe MGI/MPS use by different call types to [TABLE 2].

### 2) MPS Capacity

- OfficeServ 7100/7070/7030 provides MPS modules as an embedded-MP type and serves the max 8 calls.
- OfficeServ 7200/7400 serves MGI when MPS should be used.

### 3) Restriction of MPS Use

- If using MPS, DTMF detection which MGI supported can not be used any more.
- Only OfficeServ IP Phone can be provided Outband or Inband(RFC2833) services.
  - ◆ Outband: Method to transfer DTMF signal to Signaling message.
  - ◆ RFC2833: SIP Standard protocol which includes DTMF signal into header of RTP packet.
- IP Phone needs S/W upgrade for using MPS service. Kinds of IP Phone which should be upgraded are as follows:
  - ◆ OfficeServ SoftPhone (Version: )
  - ◆ SMT-W5100 (Version: )
  - ◆ ITP-5100V (Version: )
  - ◆ ITP-51xxD (Version: )

### 4) MPS Setup

- Select MENU 2.1.5 System Options and set MPS Service to On.
- MPS Service: Use MPS in case of On, use MGI in case of Off.
- No MPS->MGI: When MPS channel is not enough and <No MPS->MGI> is set to On, MGI is used for continuing current call. If <No MPS->MGI> is set to Off and system is

short of MPS channel, current call will be cleared.

- DTMF: This option assigns how to send DTMF signal regardless of MPS/MGI use. There are 3 ways to send DTMF signal, Outband, Inband(RFC2833) and Inband(In Voice).

2.1.5		
Item	Value	
VoIP RTP Option	DTMF	Outband
	MPS Service	On
	No MPS -> MGI	On

Figure 13. MENU 2.1.5. MPS Service Setup

- In case of using MGI card, it is recommended that DTMF type of MGI3 should be used.

## 5) MPS SPNet Setup

- In **MENU 3.3.1 System Link ID**, set <No MGI>.
  - : This option indicates that MPS will be used or not when linking with opposite system through SPNet.
- In **MENU 3.3.3 Extra System Link ID**, set <No MGI>.
  - : This option indicates that MPS will be used or not when linking with opposite system through SPNet.
- In **MENU 2.2.15 MPS Card**, set MPS Network environment.
  - ◆ OAS and MGI have a same IP address.
  - ◆ Embedded MPS and system have a same IP address.
  - ◆ Local RTP Port must be different with MGI port.
  - ◆ Public RTP Port must be different with MGI port when the system is under the NAT environment.

## 2.2.3 SBC Function

### 1) SBC

- Registration to OfficeServ and the call service can support when SIP phone is under the NAT environment by Private Hub.
- SIP phone use MPS when it is under the NAT and has a conversation with another station.

### 2) Warning

- SBC can operate normally when MPS Service is set to ON in **MENU 2.1.5 System Options**.

### 3) Setting

- In **MENU 6.2.3 SIP Phone Status**, user can check whether the phone is under the NAT or not.
- If the phone is under the NAT, IP Address and Public IP Address will be different as the above figure. (ex. Tel Number = 3301)

6.2.3							
Tel Number	Current Status	Phone Type	IP Address	Public IP Address	Public Port	Signal Port	Protocol
3301	Registered	Non Samsung SIPP	192.168.0.10	10.254.143.107	896	896	UDP
3302	Not Registered	Disconnected	0.0.0.0	0.0.0.0	0	0	UDP
3303	Registered	Non Samsung SIPP	10.254.143.101	10.254.143.101	8	8	UDP
3304	Not Registered	Disconnected	0.0.0.0	0.0.0.0	0	0	UDP
3305	Not Registered	Disconnected	0.0.0.0	0.0.0.0	0	0	UDP
3306	Not Registered	Disconnected	0.0.0.0	0.0.0.0	0	0	UDP
3307	Not Registered	Disconnected	0.0.0.0	0.0.0.0	0	0	UDP
3308	Not Registered	Disconnected	0.0.0.0	0.0.0.0	0	0	UDP

Figure 14. MENU 6.2.3. SIP Phone Status

## 2.3 Advanced License Control

- V4.30/V4.40 S/W provides two kinds of License Service.
- License Server (OSCL) does not support old and new license type at the same time. So if new License Server (OSCL) is released, system should be operated by only new license type.
- Old license also can be increased by new license type. So user should upgrade the system for V4.30/V4.40 S/W to use new license which is issued by new license server.

### 3.3.1 Old License

- Select MENU 2.1.4 and input PBX (OLD) license key. System is operated by old license.
- If PBX (OLD) license key is entered, table of new license key will be disabled.
- MOBEX Executive: PBX (OLD) license key can't assign MOBEX Executive license. But before releasing new OSCL server, MOBEX Executive license will be issued by old OSCL server temporarily.
- Select MENU 2.1.4 and input DSP (OLD) license key. System is operated by old license.

### 3.3.2 New License

- The license key system of V4.30/V4.40 S/W is changed to increase the license by one user. But new license key system has 7 free values and 5 fixed values as same as old license key system.
- The new license key divides the old license key into 2 categories such as SIP Stack and Service key. Finally the new license key has total 3 categories, SIP Stack, Service and Resource. (Refer to Figure 12)
- SIP Stack License Key
  - ◆ SIP Trunk
  - ◆ Samsung SIP Phone
  - ◆ 3<sup>rd</sup>-Party SIP Phone
- Service License Key
  - ◆ Soft Phone
  - ◆ H.323 Trunk
  - ◆ MOBEX Executive
  - ◆ IP Phone (newly added in V4.40)
  - ◆ WIFI Phone (newly added in V4.40)
  - ◆ SPNet Feature (newly added in V4.40)
  - ◆ CALL Manager (newly added in V4.40)
- DSP License Key
  - ◆ MGI Channel: MP embedded(OS7100,7070,7030)

- ◆ VM Channel
- ◆ AA Channel: OS7070 only
- From V4.40 S/W, 4 license options are added newly.
  - ◆ IP Phone: Registrable numbers of ITP3100/5220/5240/5243
  - ◆ WIFI Phone: Registrable number of New WIFI Phone
  - ◆ SPNet Feature: It indicates whether SPNet(VoIP) is supported or not.
  - ◆ CALL Manager: Registrable numbers of Call Manager Service. This service is provided by Communicator of the following phones.  
=> SOFTPC/SOFTVIDEO/SOFTPDA/ITP3100/5220/5240/5243

2.1.4			
Item	Value		
Temporary	License Type	Disable	
	License Key	NT0XNBML-W3M2TTNJ-JNRJLD90-MU9BUSEX-OJJ5RU97-7Z5XALMQ	
	License Status	OK	
	MGI	8	
	VMS	4	
Resource	License Key	NINNJ2H8-IJJ02Y0X-3YULJCOE-GS382WAD-WJXOJM7U-USECHYMQ	
	License Status	OK	
	SIP Trunk	Max Count	10
	SIP Phone	Connected	20
	3rd SIP Phone	Max Count	30
SIP Stack		Connected	0
		Max Count	30
		Connected	0
		Max Count	0
		Max Count	0
Service	License Key	No Licence Key	
	License Status	No Licence Key	
	H.323	Allowed	0
	Soft Phone	Max Count	0
	Mobex Executive	Max Count	0

Figure 15. MENU 2.1.4. License Key Setup

### 2.3.3 Temporary License

#### 1) Urgent License

- If MP card is changed inevitably and Urgent License is assigned in this case, even if there is no new license, system can provide the related services by inserting the existing license.
- In MENU 2.1.4, set <Temporary License Type> to <Urgent>. If a license key with an old MAC address is entered, even if the MAC address differs from that of new MP card, the old license can be used for two weeks.
- If a normal license key is entered while an urgent license is being operated, the existing license will be disabled automatically and the system will be operated by new license key.
- It can be used only one time on the same H/W.
- Two weeks, the usable time of urgent license, indicates the actual operation time.

## 2) Tutorial License

- Temporary license key for system demo.
- In MENU 2.1.4, set <Temporary License Type> to <Tutorial>. Even if normal license key is not entered, system can be operated by minimum license.
- It is operated normally when there is no normal license key. If a normal license key is entered while a tutorial license is being operated, it will be disabled automatically and the system will be operated by new license key.
- It can be used for two weeks after being enabled.
- It can be used only one time on the same H/W.
- Two weeks, the usable time of tutorial license, indicates the actual operation time.
- When the tutorial license is enabled, the available resources are as follows:
  - ◆ 2 VM channels
  - ◆ 2 MGI channels
  - ◆ 2 MOBEX Executive users
  - ◆ 2 Soft phones
  - ◆ 2 H.323 trunks
  - ◆ 2 SIP trunks
  - ◆ 2 Samsung SIP phones
  - ◆ 2 3<sup>rd</sup>-Party SIP phones
  - ◆ 2 Samsung SIP applications
  - ◆ 2 IP Phone (newly added in V4.40)
  - ◆ 2 WIFI Phone (newly added in V4.40)
  - ◆ 2 CALL Manager (newly added in V4.40)

## 2.4 The Other Features

### 2.4.1 SVM Options

#### 1) SVMi Ring Mode

- In **MENU 2.1.6 SVMi Options**, user can assign SVMi Ring Mode like MP Ring Plan. 1 to 6 can be set to MP Ring Plan and 1 to 99 can be set to SVMi Ring Mode.

#### 2) Holiday Assignment

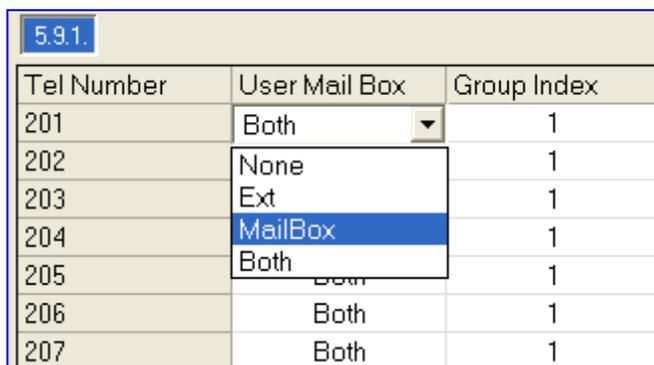
- In **MENU 4.3.2 Holiday List**, user can assign MP Ring Plan and SVMi Ring Mode for each holiday.
- 1 to 6 can be set to MP Ring Plan and 1 to 99 can be set to SVMi Ring Mode.

#### 3) SVMi Download Options

- In **MENU 2.1.6 SVMi Options**, set <Mailbox Download> to Yes. User can download the phone number of each device type when SVMi restart. In this case mailboxes of each device type are generated.
- If <STATION NAME> is set to Yes, user can download the station name when downloading station number. If MBX or EXT name of SVMi is changed, this change will be applied to the station name.

#### 4) SVMi User Options

- In **MENU 5.9.1 SVMi Create Subscriber**, set the SVMi automatic generation option for each station.
  - ◆ None: There are no options to generate automatically.
  - ◆ Ext: Generate station numbers automatically.
  - ◆ MBX: Generate User mailboxes automatically.
  - ◆ Both: Generate both station numbers and user mailboxes automatically.



Tel Number	User Mail Box	Group Index
201	Both	1
202	None	1
203	Ext	1
204	MailBox	1
205	Both	1
206	Both	1
207	Both	1

Figure 16. MENU 5.9.1. SVMi Automatic Generation Option Setup

### 2.4.2 ITP Idle Mode Feature

- S/W of ITP phone will be upgraded to use ITP idle mode.

- Only ITP-51xxD and ITP-5100V can be supported.

### 1) ITP Idle Mode

- If an ITP phone logs out in the state that it is registered on the system, it will go to login standby state and no call service will be performed.
- If an ITP phone logs in, it will be registered on the system and will be able to receive the call service from the system. When logging in, insert User ID and Password.
- If an ITP Phone logs in and there is a phone which has a same User ID, a phone which logged in by existing User ID will log out and the new phone will be able to log in.

### 2) ITP Logout Feature ID

- In MENU 2.8.0, input LOGOUT Feature ID. (ex, 68)
- When ITP phone is registered in the system, dial the LOGOUT feature ID(68). Then the system registration is deactivated and the ITP phone has an idle mode.

### 3) ITP Idle Mode Setup

- ITP phone can select the option to decide whether it is in the login standby state or logs in the system automatically when it restarts.

### 4) V4.40 Additional Features

- Complement of Forced LOGOUT in case of trying LOGIN with the duplicate ID. As following cases, phone doesn't support Forced LOGOUT.
  - ◆ Phone which doesn't support IDLE MODE.
  - ◆ IP LOUT COS is set to NO in **MENU 4.7.2 COS Contents**.
  - ◆ Phone in the login state changes into busy state.
  - ◆ Phone log in the system with the default password (1234).
- Log out all restricted IP phones.
  - ◆ In **MENU 5.2.10 System ITP Options**, set ITP IDLE LOGOUT TYPE to AUTO TIME and input Start Time. At the Start Time, all IP phones in IDLE state will log out.
  - ◆ In **MENU 6.3.14 ITP Idle Logout**, set LOGOUT NOW to YES. Then all IP phones in IDLE state will log out.
  - ◆ As following cases, phone doesn't support Restricted LOGOUT.
    - IP LOUT COS is set to NO in MENU 4.7.2.
    - Phone in the login state changes into busy state.
    - Phone which doesn't support IDLE MODE.
- Forced LOGOUT
  - ◆ In MENU 6.3.15 ITP Forced Logout, set Forced Logout to YES. Then selected IP phone will log out forcedly.

## 2.4.3 SIP-FXO link with Remote ITP

### 1) Definition of SIP-FXO

- It is an external SIP-PSTN gateway device which is installed in STA for providing an emergency phone number to Remote ITP.
- Resource: SIP 1 channel, FXO 1 port
- Interface: Link with PSTN through FXO. Link with OfficeServ through SIP Peering.
- Operation: Incoming local calls from FXO of a remote ITP is routed to OfficeServ through SIP, and incoming calls from SIP of a remote ITP is also routed to PSTN through FXO.
- WIFI Phone can support Remote ITP. (From V4.40)

### 2) Outgoing call to SIP-FXO

- If Remote ITP makes an emergency call (ex, 911), that call is routed to Emergency Center of Remote ITP through SIP FXO.
- It is possible to route the outgoing call to SIP-FXO only when SIP-FXO and Remote ITP are in the same area.
- Outgoing call of Remote ITP is routed to SIP-FXO through SIP Peering and FXO can receive that call.

### 3) Incoming call from SIP-FXO

- If an incoming call is received from SIP-FXO, that call is routed to Remote ITP
- It is possible to route the incoming call (from SIP-FXO) to Remote ITP only when SIP-FXO is distinguished from the Called Party Number.

### 4) Setup

- In **MENU 2.7.1 ITP Information**, input Emergency Code and Emergency Dial. (V4.30 only)
  - ◆ Emergency Code: SIP FXO Identification Code for each Remote ITP
  - ◆ Emergency Dial: Emergency Number
- In **MENU 5.15.15 Emergency Dial**, input Emergency Code and Emergency Dial. (V4.40)
  - ◆ Emergency Code: SIP FXO Identification Code for each Remote ITP
  - ◆ Emergency Dial: Emergency Number
- In **MENU 3.1.2 Routing Digits**, input emergency dial of MENU 2.7.1 to LCR Digit, In case of V4.40, input emergency dial of MENU 5.15.15.
- In **MENU 3.1.4 Routing Table**, input SIP Trunk Group number for LCR Routing.
- In **MENU 5.2.17 SIP Peering**, input IP address of SIP-FXO.
- In **MENU 3.2.3 DID Ringing**, input SIP-FXO Identification Code to incoming digits and Remote ITP number to destination.
- Select **MENU 5.2.3 VoIP Outgoing Digits**.
  - ◆ Access Digit: Input emergency code of MENU 2.7.1 to Access Digit and length of emergency code to Digit Length. In case of V4.40, input emergency code of

MENU 5.15.15.

- ◆ IP Table Number: Input table of MENU 5.2.17.
- ◆ Del Length: Input length of emergency code. This option is for deleting emergency code which is inserted into the outgoing call of SIP-FXO.
- Select **MENU 5.2.4 VoIP Incoming Digits**.
  - ◆ Access Digit: Input emergency code of MENU 2.7.1. In case of V4.40, input emergency code of MENU 5.15.15.
  - ◆ Digit Length: Input length of emergency code.
- If an emergency outgoing call is failed, reassign class to follow the next LCR Table.

#### 5) Call Flow of outgoing call to SIP-FXO

- This flow is only for the case that Remote ITP makes an outgoing call through LCR.
- If the LCR digits is same with a emergency dial in MENU 2.7.1 ITP Information, the emergency code in MENU 2.7.1 ITP Information is inserted during processing LCR digits. In case of V4.40, use emergency code in MENU 5.15.15
- The call is routed to SIP trunk through inserted LCR digit
- The call is routed to SIP FXO through inserted identification code when the call is outgoing to SIP trunk.

#### 5) Call Flow of incoming call from SIP-FXO

- As an incoming call receives from SIP FXO, a SIP FXO identification code is inserted by in-table of MENU 5.2.4 VoIP Incoming Digits.
- The call will be divided by SIP FXO identification code in MENU 3.2.3 DID Ringing and routed to remote ITP.

### 2.4.4 SIP Peering Feature

#### 1) SIP Peering

- It sends OPTIONS message, which checks the state of SIP Peering, to serve Re-routing feature for SIP Peering.
  - ◆ Set Ports of each SIP Peering.
  - ◆ Set protocol type (UDP/TCP) of each SIP Peering/SIP Trunking.

#### 2) Setup

- Select **MENU 5.2.17 SIP Peering**.
  - ◆ Protocol: Choose the protocol of IP Address. (SIP/H.323)
  - ◆ Alive Check: Choose if it needs Link Test or not. If it needs, set to <Option>. (None/Option)
  - ◆ User Information: Input user name to be loaded in To&From header of OPTIONS message. It should be Alphanumeric. System can Not send OPTIONS message without this information.
  - ◆ Remote Port: Input the port for using SIP server.
  - ◆ Check Timer: Input the time interval of Link Test. It will operate while Alive Check

is set to <Option>.

- ◆ SIP Signal Type: Choose Transport Type for sending SIP message. (UDP/TCP)
- Select **MENU 5.2.12 SIP Stack/Ext/Trunk Options**.
  - ◆ CLIR Flag with Number: Set this option to Enable when called party wants not to display the CLI number in case of SIP outgoing call.
  - ◆ Incoming Mode: Choose the display option for an incoming call through VoIP trunk.
    - Follow TRK Ring – Follows the table in MENU 3.2.1 Trunk Ringing.
    - Follow DID Trans – Follows the table in MENU 3.2.3 DID Ringing.
    - Follow Incom Digit – Displays the incoming Called Party Number.
  - ◆ Peer CLI Table: Set the CLIP table number for referring as making a SIP Peering call. User can choose among the table 1~4 in MENU 2.4.3 Send CLI Number. But in case of SIP Trunking, it refers Send CLIP Table in MENU 5.2.13 SIP Carrier Options.
- Select **MENU 5.2.13 SIP Carrier Options**.
  - ◆ Hold Reinvite: Choose if system should send RE-INVITE message as doing HOLD and RESUME, or not. If this option is enabled, system will send RE-INVITE message to inform that the call is changed to HOLD state as user press HOLD button during SIP Session operation.
  - ◆ URI Type: Choose the SIP URI type for SIP messages. It serves SIP-URI and TEL-URI by the SIP standard. (SIP/Tel)
  - ◆ SIP Signal Type: Choose the transport type for SIP messages. (UDP/TCP)

## 2.4.5 Others

### 1) Bath Alarm

- In **MENU 5.13.1 Tenant Options**, set the destination of Bath Alarm.
- If visitor dials the BALARM ID in hotel room, it will make a bath alarm call to the destination phone. BALARM dialing is operated with only one touch.
- Italy only can enable.

### 2) Speaker mode Call Paging

- User can use Station Page in Speaker mode.
- In **MENU 4.7.2 COS Contents**, set SPK PAGE to Yes.

### 3) Station MMC Password

- User can set to prevent to enter the Key MMC 102~199 with password.
- In **MENU 4.7.2 COS Contents**, set MMC PSWD to Yes.

### 4) Macro Key

- User can set feature code in speed dial with MACR key.

- User should input the speed dial number (00~49) to extension of MACR key, and set the dialing number in **MENU 5.15.7 Station Speed Dial**.
- As sending the speed dial number, all digits will be sent at one time in normal case. But if MACR key has been set, system will analyze the each digit.
- In **MENU 4.9.2 Station Key**, set MACR key and extension number.
- In **MENU 2.8.0 Numbering Plan**, set feature ID of MACR.

### 5) All Group Pickup

- User can pick up the call of individual station by dialing feature ID of MYGRPK. In this case the individual station should be a member of Pickup Group.
- If All Ringing Pickup option is set to On in **MENU 5.14.1 Transfer/Recall/Pickup Options**, user can pick up any call by dialing feature ID of MYGRPK even if the phone is not a member of Pickup Group.

### 6) CID Display Option

- Some selectable options for displaying CID during ringing a phone are added.
- User can choose an option for displaying CID during a conversation.
- In **MENU 5.15.3 Station Display**, set CID Display options.  
: No display, Number First, Name First, Number+DID, Name+DID
- In **MENU 5.15.3 Station Display**, set CONV Display options.  
: CLI Number First, CLI Name First, DID Number First, DID Name First, DID/CLI Number
- From V4.40 S/W, CID display options is changed as follows.

TABLE 4. CID Display Options

Option Name	Display Option	Description
RINGING DISPLAY ON LINE1/ LINE2	NO DISPLAY	701: ringing
	CLI NUMBER FIRST	Shown in order of CLI Number, Name. If both number and name are null, display 701: ringing.
	CLI NAME FIRST	Shown in order of CLI Name, Number. If both number and name are null, display 701: ringing.
	DDI NUMBER FIRST	Shown in order of DDI Number, Name. If both number and name are null, follow CLI NAME FIRST.
	DDI NAME FIRST	Shown in order of DDI Name, Number. If both number and name are null, follow CLI NAME FIRST.
	GROUP NAME FIRST	In case of group call, display "Call <group name>" on LINE1 and "Call For 5000" on LINE2. If group is not called, follow CLI NAME FIRST.

	DDI NUMBER/NAME	Display DDI Number/Name by 16 digits. If both number and name are null, follow CLI NAME FIRST.
	DDI NAME/NUMBER	Display DDI Name/Number by 16digits. If both number and name are null, follow CLI NAME FIRST.
CONV DISPLAY FOR INCOMING CALL	CLI NUMBER FIRST	Shown in order of CLI Number, Name. If both number and name are null, display "701: ".
	CLI NAME FIRST	Shown in order of CLI Name, Number. If both number and name are null, display 701: ".
	DDI NUMBER FIRST	Shown in order of DDI Number, Name. If both number and name are null, follow CLI NAME FIRST.
	DDI NAME FIRST	Shown in order of DDI Name, Number. If both number and name are null, follow CLI NAME FIRST.
	CLI NUMBER/DDI	Display CLI Number + DDI Number by 16 digits. If both number and name are null, display "701: ".
	CLI NAME/DDI	Display CLI Name + DDI Number by 16 digits. If both number and name are null, display "701: ".
	DDI NUMBER/NAME	Display DDI Number/Name by 16 digits. If both number and name are null, follow CLI NAME FIRST.
	DDI NAME/NUMBER	Display DDI Name/Number by 16 digits. If both number and name are null, follow CLI NAME FIRST.
	TRUNK NUMBER	701:
TRUNK DISPLAY	TRUNK:DIGIT	701:1234567 (ex. DIGIT: 1234567)
	DIGIT ONLY	1234567 (ex. DIGIT: 1234567)
TIME/COST DISPLAY	NO DISPLAY	Do not display Time and Cost.
	TIME DISPLAY	Display Time (MM:DD).
	COST DISPLAY	Display Cost (\$2.00).

- ◆ LINE1 and LINE2 of Ringing Display INFO can be set separately.
- ◆ If LINE1 and LINE2 have a same INFO and LINE1 is set to NUMBER+NAME, LINE2 will follow CLI NAME First. Or if LINE1 is set to NUMER or NAME, INFO which is not duplicate will display on LINE2.

- ◆ In case of Transfer/Forward/Recall, “TRSF/FWRD/RECL From xxxx” is displayed on LINE2.
- ◆ If there is no CLI, Restricted/Invalid/Private and other related message will be displayed on LINE1 or LINE2. This message can't be displayed on both LINE1 and LINE2.

#### 7) Trunk MOH

- Analog Trunk can be used in External MOH port.
- In **MENU 2.6.5 Loop Trunk Data**, set Use MOH to Yes.

#### 8) SPNet Transfer Auto Answer

- If a trunk call is transferred to SPNet and the call is answered automatically, it will operate in the same way with the normal system.
- In this state, if a station transferring the call hangs on, it will operate as follows:
  - ◆ The held trunk hears Ringback tone.
  - ◆ The auto-answered phone gives a ring.

#### 9) MGI Local RTP Port

- User can change MGI RTP Port to other values. (It was fixed with 30000.)
- In **MENU 2.2.2 MGI Card**, input a value to Local RTP Port.

#### 10) Analog CID for UK

- UK can also receive CID through Analog Trunk.
- Trunk line has to be connected to authorize CO line.
- In **MENU 2.6.5 Loop Trunk Data**, set CLI Trunk to Yes.

#### 11) OS7100 Cabinet with OS7200

- OfficeServ 7200 can connect OfficeServ 7100 Cabinet for expansion rack.
- LCP card, not MP1x card, has to be installed in OfficeServ 7100 cabinet.
- In **MENU 2.1.5 System Options**, set Cabinet Type to OfficeServ 7100.

#### 12) ISDN Message Options

- The message flows of Supplementary Service for PRI or BRI module(or cards) are different. It depends on systems or versions. So user should set the option.
- In **MENU 2.1.5 System Options**, set BRI or PRI ISDN SS Service option.
- User can choose to release the call at once with Progress Indicator or to keep the B channel when the opposite hang on during a call through ISDN trunk.
- In **MENU 2.1.5 System Options**, if ISDN Progress Indicator Call Clear is Enabled, it will keep the B channel until receiving the release message. But if the phone hangs on, the call will be cleared.

### 13) ISDN Dial Sending

- When making an outgoing call via ISDN trunk, the DIGIT sent through D-channel before the call is connected.
- But some applications wait the dial message from B channel.  
(For example, when user inputs some additional digits after answering of IVR)
- In **MENU 5.14.5 ISDN/RT/Trunk Options**, if <ISDN Outgoing Connect when Progress> is set to On, the digits will be sent through B-channel. In this case, it is considered as call connection and reported as call duration in SMDR.
- In **V4.30 S/W**, even if the option is Off, the digit will be sent through B-channel.

### 14) ISDN Supplementary Service (V4.40)

- From V4.40 OfficeServ supports following ISDN supplementary services.
- ISDN Partial Rerouting
  - ◆ In **MENU 2.6.7 or 2.6.8 BRI/PRI Trunk Data**, set <EFWD Mode> is newly added to send FACILITY message for Unconditional/Busy/No Answer Call Forward.  
=> Set <EFWD Mode> to ON.
- ISDN Diverting Information
  - ◆ In **MENU 2.6.7 or 2.6.8 BRI/PRI Trunk Data**, <Diverting Information> is newly added to send facility IE of SETUP message in case of MOBEX call. Facility IE includes Diverting information and Original Called number.  
=> If <Diverting Information> is set to <ON>, both Diverting information and Original Called number are sent.  
=> If <Diverting Information> is set to <Except CD>, only Diverting information is sent except Original Called number.
- Italian 3rd-Party Conference
  - ◆ System transform each key such as iHOLD, iRET and iCONF into the appropriate INFORMATION (KEYPAD FACILITY IE: \*75#, \*76#, \*77#) message and send that message to ISDN.  
=> But INFORMATION message can be sent as following cases.
    - ITALY only
    - In **MENU 2.6.7 BRI Trunk Data**, set Incoming Mode to PTP mode.
    - In **MENU 5.14.5 ISDN/R2/Trunk Options**, set ISDN Key Pad Facility to ON.  
(Default value is ON in ITALY.)
- PRI CLI Blocking
  - ◆ System can send CLIR flag such as 'Allowed' and 'Restricted' to ISDN.  
=> In **MENU 2.5.1 Station Data**, set <CLI Send>.  
If <CLI Send> is set to YES, CLIR flag is designated 'Allowed'.  
If <CLI Send> is set to NO, CLIR flag is designated 'Restricted'.
  - ◆ This feature is provided only in STA.
- PRI Date/Time
  - ◆ System updates its time if CONNECT message contains Date/Time information.  
=> In **MENU 2.1.3 System Time**, set <Auto Update ISDN Time> to ON.

=> In MENU 2.6.8, set <Time Sync> to ON.

### 15) New IP Phone

- From V4.40, New IP phone can be registered to OfficeServ system.
  - ◆ New IP Phone: SMT-i3100, SMT-i5220, SMT-i5243
  - ◆ Add new **MENU 5.6.4 Phone I/O Parameter**.  
=> In this menu, user can save the information of LDAP, XML, SNMP.
  - ◆ Add new option 'Use Station Ring' in **MENU 5.15.4 Keypad On/Off**.  
=> User can select ring type of IP phone. If this option is set to ON, phone will use system ring. If set to OFF, phone will use its self ring.

### 16) VCS and MP20 support

- MP40 and MP20 can support VCS from V4.40.
- VCS setting
  - ◆ In **MENU 6.3.2 Virtual Card Change**, set Virtual Card Type to SIP App. (Refer to programming manual to find selectable Cabinet and Slot information.)
  - ◆ In **MENU 6.3.2 SIP Application**, set Service Type to VCS.
  - ◆ In **MENU 2.1.4 License Key**, input SIP Stack License including SIP application license.
  - ◆ In **MENU 4.1.1 Station Group**, set Group Type to VCS and input channel number in VCS Station Group. In this case user should use the channel which is set to VCS in MENU 6.3.2.
  - ◆ In MENU 5.13.1 Tenant Options, input VCS group number of MENU 4.1.1 in VCS Station Group.
  - ◆ In MENU 5.15.14 MMS Service, user can enable each service options of registered Video Phone.

5.15.14				
Tel Number	MMS Ringback	MMS Ring	MMS Busy	MMS MOH
3203	No	No	No	No
3204	Yes	Yes	Yes	Yes
3206	No	No	No	No
3210	Yes	Yes	Yes	Yes
3217	No	No	No	No
3219	No	No	No	No

Figure 17. MENU 5.15.14. MMS Service Type

### 17) The Others

- VoIP trunk CLI Transfer: When the incoming trunk call routes to SIP or H.323 trunk, the received CLI will be sent through the trunk.
- WiFi phone protocol: SMT-W5100 phone supports both SIP mode and OfficeServ



mode. From V4.40, 'Located' options of **MENU 6.2.5 WIP Status** are changed. In case of registering as SIP mode or Non SIP mode, 'Attach-SIP Mode' or 'Attach-OSPP Mode' will be shown.

- Samsung SIP phone: V4,30 S/W supports the SMT-i2100 phone, economic SIP Phone.
- SoftPhone Multi-Server: OfficeServ SoftPhone supports multi-server.
- Syria Project:  
In **MENU 2.1.5 System Options** in Korea mode, if SYRIA R2/RD Select is enabled, system can connect to PSTN of Syria with R2.  
And then the normal phone, not Large LCD phone, can make a Group Conference.
- SIP Connect: OfficeServ System interworks the BroadWorks SoftSwitch with Samsung Networks specifications.
- OS7200 NMS support: The MP20 supports OfficeServ NMS.
- PRI Booking
  - ◆ From V4.40, not only Analog trunk but also ISDN trunk can support Booking feature. But this feature is usable only in KOREA.
- VM RING BACK TONE DELAY
  - ◆ From V4.40, VM Ring Back Tone can be heard in case of making outgoing trunk call. But there is a delay so tone will be heard after a few seconds.
  - ◆ In **MENU 5.5.0 System Control**, set VM Ringback Delay Time.
- Change IP Options
  - ◆ Following modifications are applied from V4.40.
  - ◆ Delete [Public] IP option  
=> Use only 2 options. ([Private] or [Private with Public])
  - ◆ Delete IP Type of IP phone. System will check IP Type automatically.
  - ◆ IP phones under the NAT always use MPS or MGI.  
=> Modified MMC: **MENU 2.1.2:** SYSTEM IP TYPE => delete PUBLIC IP ONLY.  
**MENU 2.2.2:** MGI IP TYPE => delete PUBLIC IP ONLY.  
**MENU 2.2.15:** MPS IP TYPE => delete PUBLIC IP ONLY.  
**MENU 3.3.1:** delete SPNET IP TYPE.  
**MENU 2.7.1:** delete IPP IP TYPE and PUB TO PUB.
- Change TIME SYNC default value
  - ◆ In ITALY, default value of MENU 2.6.7 BRI Trunk Data and 2.6.8 PRI Trunk Data changes into OFF.



- MP20 can support NMS from V4.40.
- Maintenance of Registration Information
  - ◆ After system reset, user can make a call to SIP phone which doesn't use 5060 port.
- 423 Interval Too Brief Message
  - ◆ If OfficeServ registers to SIP server and receives 423 Interval Too Brief response, system sets Min-Expire of received message to Expire.
  - ◆ SIP phone sends 423 Interval Too Brief response if Expire of registered message is smaller than SIP Expire Time (sec) of MENU 5.2.12 SIP Stack/Ext/Trunk Options.