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OfficeServ V4.6x Feature Guide

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Glossary and Acronyms

AES	Advanced Encryption Standard
AP	Access Point
ARIA	Academy, Research Institute, Agency
BRI	Basic Rate Interface
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CNF24	24 channel Conference Bridge Card
DGP	DiGital Phone
DLI	Digital Line Interface
ECC	Elliptic Curve Cryptography
IP	Internet Protocol
IP Phone	Internet Protocol Interface Phone
LAN	Local Area Network
MGI	Media Gateway Interface Module
MMC	Multi Media Card
NAT/PAT	Network/Port Address Translation
NMS	Network Management System
NTP	Network Time Protocol
PRI	Primary Rate Interface
PSTN	Public Subscriber Telecommunication Network
RBT	Ring Back Tone
RTG	Real-Time Tone Generator
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SLI	Single Line Telephone Interface
SP	Signalling Process
sRTP	secure Real Time Protocol
SSL	Secure Sockets Layer
TLS	Transport Layer Security
VoIP	Voice over Internet Protocol
WEVoIP	Wireless Enterprise Voice over Internet Protocol

References

1. [OfficeServ V4.6] Market Requirements Document



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1. Overview

1.1 Purpose

This document is to specify the requirements to be applied when developing the OfficeServ V4.6. Also, the details presented in the System Requirement section contain the technical items necessary in the development stage of the OfficeServ V4.6 and have been written from the standpoint of users. The system test units and test items for the OfficeServ V4.6 will be determined based on the requirements specified by this document.

1.2 Scope

Skipped

1.3 Document Organization

Skipped



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2. Considerations

Before using OfficeServ Main V4.60 and above software, please refer to the below considerations.

2.1 Compatibility

- 1) MP20/MP40 and LCP/LP40 software packages
 - An old versions of MP20/MP40 and LCP/LP40 are compatible with the latest version of each others. But in this case, the features V4.6 supports do not work properly.
 - MP20 & MP40 : V4.60↑
 - LCP : V4.30↑
 - LP40 : V2.00↑
- 2) Newly released cards
 - To use SVM1-20i and 16SLI3/8SLI3/8Combo3 cards, a proper version of cards is required as below.
 - MP20 & MP40 : V4.60↑
 - LCP : V4.30↑
 - LP40 : V2.00↑
- 3) DB compatibility
 - The database of V4.60 is not compatible with that of a previous version.
 - So you download the database of the existing system before upgrading to V4.60↑using either IT or DM.
 - After upgrading to V4.60↑, upload your previous DB by using OfficeServ DM
- 4) Java compatibility

DM(V4.65) is required for its normal operation to install Java6 only.

If you already installed Java7 or JDK on your Windows PC, it should be uninstalled first before Java6 installation.

You can download Java6 through the following link.

http://www.java.com/en/download/manual_v6.jsp

2.2 Package Upgrade

- 1) Main package

There are some changing and restrictions for upgrading MP package.

 - A. A restriction for MP40/MP20 upgrade
 - When an existing system is equipped with an earlier package than V4.60, new V4.60↑ package for MP40 and MP20 will not be upgraded by using either DM or IT tool. Because the file size of new MP20/40 package is about 20M Bytes, V4.60↑ software should be upgraded by copying it to SD card directly.
 - B. A changing for MP20s/MP10a and OS7030 upgrade by DM
 - In case of MP20s, MP10a and 7030, 'osdm.jar' and 'osdmhelp.jar' files has been separately added to the main package as below. Totally 9 files should be copied to SD card manually.



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<Previous>

aplav460.pkg, cs1av440.pkg, dr1av460.pkg, ms1av460.pkg, rd1av460.pkg, rtlav460.pkg, wslav460.pkg

<Current>

aplav460.pkg, cs1av440.pkg, dr1av460.pkg, ms1av460.pkg, rd1av460.pkg, rtlav460.pkg, wslav460.pkg, **osdm.jar, osdmhelp.jar**

- **Using DM to upgrade your system to V4.60↑ is strongly recommended, even though MP20s/MP10a/7030 packages can be upgraded by IT as well.**

Because if you upgrade MP20s/MP10a/7030 systems from under V4.60 by using IT, there is no DM directory and it doesn't allow you to access an OfficeServ system via embedded DM for web access. (If you use a stand-alone DM, you can connect the system directly)

- For the web connection, please upload DM files(**osdm.jar, osdmhelp.jar**) manually which were included in each package provided as *.zip file.

C. Nothing changed for OS7070 upgrade

- In case of 7070, 7 files will be provided for V4.60↑ main software of OS7070 as previous packages.

aplav460.pkg, cs1av440.pkg, dr1av460.pkg, ms1av460.pkg, rd1av460.pkg, rtlav460.pkg, wslav460.pkg

※ **CAUTION**

“SIP Exclusive” feature in MMC837 in V4.60 is improved.

This option was added from V4.42 and you can set it to disable or enable.

By setting it enable (reject a SIP message from unauthorized IP addresses by referring MMC833 and MMC837), you can prevent an unauthorized SIP call from going through OS system via SIP trunk or SIP peering.

In V4.60, the name is changed from “SIP Exclusive” to “Comm Exclusive” and one more option (no response) is added.

In addition, OS system blocks the IP address for specified period in case OS system gets wrong User ID or Password of an SIP phone several times when an SIP phone tries to register to system.

- You can set None/Response for OS7200/7200s/7100/7070/7030.
- You can set None/Response/No response for OS7400.

	V4.42 to V4.5x	V4.60↑	Description
Name	SIP Exclusive	Comm Exclusive	
Value	Disable (Default)	None	OS system will accept all SIP calls without restriction.
	Enable	Response (Default value, except MP40)	OS System will reject a SIP trunking/peering from unauthorized IP address referring MMC833 and send deny message (403 forbidden) to the opposite end point.
	-	No response	OS system will ignore all SIP messages from unauthorized



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	(Default for MP40)	IP address and block the IP address without any response. And also OS system blocks the IP address for specified period in case of a repetitive attempt with wrong User ID or Password for its registration. - This option is available only for MP40
--	--------------------	---

※ Checkpoints for SIP incoming call fail after upgrading from earlier version to V4.60

1) DM 5.2.17 VoIP Peering (MMC833)

- You need to input “IP address” and “User Information”.
- The value of “User information” should be identical with one of remote peering system.

2) DM4.1.2 Trunk Groups (MMC603)

- You need to input trunk group member and set ISP Selection as Peering.

2) LP40

- MP40 should be upgraded to V4.60↑ before upgrading LP40 to V2.00↑because V4.60↑only can recognize new LP40 file name.
- File name has been changed from LP4xxxxx.PGM to SP4xxxxx.PGM.
- The new LP40 package contains both LP40 bootrom and LP40 software file.
When you try to upgrade LP40 to V2.00↑from an earlier version than V2.00 in MMC818, it will take about 13 minutes. Because LP40 bootrom should be upgraded for about 7 minutes at first..

3) IT Tool

- IT tool has been discontinued and is not released for V4.60. It does not support any new features which V4.60 supports.

4) CNF24 and its prompts

- CNF24 upgrade
You can upgrade CNF24 easily by using DM.
 - ① Run DM and connect to the system.
 - ② Select Util > Package Update
 - ③ If you set CNF24 network configuration normally, CNF24 card information will be shown on Package Update.
 - ④ Upload CNF24 package and then select the card that you want to upgrade.
- CNF24 prompts
 - ① From V4.60, the number of prompt language is increased to 17. So you have to upload new prompt files for normal operation.
 - ② Default prompt is set to UK-English except USA, Korea, Russia, Italy and Spain.



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③ Upload directory is '/mnt/nand0/prompt/'. (*FTP ID: admin, PW: samsung)

5) SVMi-20i feature setting

- You can set SVMi-20i configuration via DM.
In order to configure it, an IP address of SVMi-20i should be inserted in DM 2.2.17 at first, Gateway and Subnet Mask as well.
- You can upgrade the SVMi-20i package via DM.
For more detail, please see the 'Package Update' part in the user guide of SVMi-20i (Page 2-10 and 2-11).
- Upgrade the SVMi-20i to the latest version before you start testing.

6) VM/AA Backup in case of OfficeServ 7100/7030/7070 and MP20s

- A backup of VM/AA is not required before upgrading the system to V4.60↑, because normally the database should be migrated with no damage during its upgrading.
- But you are strongly recommended to consider the backup & restoring when the system version is changed from V4.60↓ to V4.60↑, especially in case of initializing VM/AA or copying the DB to another system.
- Using web management of VM/AA, please backup the DB from 'Operating Utilities > Backup and Restore' before upgrading to V4.60↑. (It is not supported by DM)
- After upgrading to V4.60↑, restore the DB of VM/AA at 'Operating Utilities > Backup and Restore' via DM.

7) MP40/MP20 certification files

- You should copy certification files to SD card. These files are needed to support HTTP/HTTPS/TLS. HTTP/HTTPS/TLS are used to provide DM, Remote dial and Click-to-Dial features.
- Just unzip the MP40 and MP20 package file in a PC and then copy the unzipped files and directories to a SD card. MP20/MP40 packages already include all the files and directories in the following description.
- For your reference, the following is the way to place all certification files to SD card when you receive them separately from MP40/MP20 package.

- ① Make "certification" directory in media card.
- ② Make "mvs" and "single" directories in "certification" directory.
- ③ Copy 'caCert.pem', 'myCert.pem' and 'myPrvKey.pem' files to "mvs" and "single".
Be sure that Remote dial files and Single files are different. So you have to copy appropriate files to the appropriate directory.
- ④ Copy three certification files of "single" directory to root directory.

※ Below is the example of MP40/20 media card directory.

<Root>

In this case, we copy 'single' certification files to root.



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[certification]	<DIR>	2011-09-21 21:34
[dm]	<DIR>	2011-09-21 21:34
caCert	pem	1,180 2011-03-10 10:18
Mpe_V460	pgm	7,957,377 2011-09-08 14:16
myCert	pem	1,237 2011-03-10 10:18
myPrvKey	pem	887 2011-03-10 10:18

<Certification>

[mys]	<DIR>	2011-09-21 21:34
[single]	<DIR>	2011-09-21 21:34

<Remote dial>

caCert	pem	1,058 2011-06-01 18:53
myCert	pem	1,107 2011-06-01 18:52
myPrvKey	pem	887 2011-06-01 18:53

<SINGLE>

caCert	pem	1,180 2011-03-10 10:18
myCert	pem	1,237 2011-03-10 10:18
myPrvKey	pem	887 2011-03-10 10:18

- You do not have to upload certification files to 7100, 7200s, 7030 and 7070. These are already included in 7100, 7200s, 7030 and 7070 Packages.

2.3 Quick guide for the Device Manager(DM)

- 1) To Run DM (Stand Alone)
 - Unzip the DM zip file and execute 'osdm.exe'
- 2) To Upload DM Package (Except 7070) for V4.60.

Unzip the DM zip file and then copy unzipped files and directory into a SD card. Packages already include all the DM files in the DM directory.

For your reference, the following is the ways to upload DM files to the system when you receive DM file separately from system packages.

- Using Stand Alone DM
 - ① Execute "osdm.jar"
 - ② Select Util -> Quick File Control
 - ③ Input system IP address
 - ④ **MP40/MP20** : Upload "osdm.jar", "osdmhelp.jar", "osdm.jnlp" and "osdm_public.jnlp"
MP20s/MP10a/MP11/7030 : Upload "osdm.jar" and "osdmhelp.jar"
 - Copying DM files into media card (SD card)

Copy "osdm.jar", "osdmhelp.jar", "osdm.jnlp" and "osdm_public.jnlp" to "/dm" in media card.
- ※ In OS7070 system, all osdm files are already included.



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※ “osdm.jnlp” and “osdm_public.jnlp” is only for MP40/MP20 Packages.

3) To Connect DM to the system with web browser

	V4.60↑	
	Private network	Public network
MP20 MP40	http(s)://System IP/dm/	http(s)://System IP http(s)://System IP/dmp/ http(s)://System IP/dm_public/
MP20s OS7100 OS7070 OS7030	http(s)://System IP	

4) To Connect PWP to the system with web browser

- URL : http(s)://+IP+/pwp/

2.4 DM (V4.65) consideration

1) Java compatibility

DM(V4.65) is required for its normal operation to install Java6 only.

If you already installed Java7 or JDK on your Windows PC, it should be uninstalled first before Java6 installation.

You can download Java6 through the following link.

http://www.java.com/en/download/manual_v6.jsp

- 2) The system will not allow the DM to access, if the admin password is not changed from a default value.
- 3) Encryption option has been added on the DM login page.
 - OfficeServ V4.65 ↓ : Unchecked the encryption option is required for the system login.
 - OfficeServ V4.65 ↑ : Checked the encryption option is required for the system login.
- 4) The secure feature on V4.65 does not allow old DM to access the V4.65 system.

2.5 NAT configuration

NAT configuration is required in following cases.

- When the SMT phones in a remote network want to use ‘system directory service’.
- When you remotely connect the OfficeServ system including VM via DM.
- When you remotely connect OS7200s/7100/7070/7030 via Telnet.

For the port information of OfficeServ environment, please refer to the table below.

Module	Service	Protocol	Port	Remarck
MP	SIP	UDP/TCP	5060	



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		TCP	5061	
	H.323	TCP	1720	
		UDP	1719	
	SPNET	TCP	6100	
	ITP	UDP	6000	Signaling
			9000	RTP
		TCP	5180	Phone book
	WIP	UDP	8000, 8001	
	Remote dial	TCP	9012	
	DM	Config	TCP	5090,5091
		FTP	TCP	21
		Data	TCP	5090
		File Control	TCP	5003
	IT Tool	TCP	5090, 5091	
	VM Control	TCP	6001,6002	
	Telnet	TCP	5030	
MGI16	MGI	UDP	30000~ (2*NumOfChannel -1)	
MGI64	MPS	UDP	40000~ (2*NumOfChannel -1)	
OAS	RTG	UDP	45000 ~ (2*NumOfChannel-1)	
CNFG24	Conference	UDP	30000 ~ (2*NumOfChannel -1)	
	FTP	TCP	21	
	Upgrade Port	TCP	60000	
SVMi-20i	VM Control	TCP	6001,6002	
	VM	UDP	30000 ~ (2*NumOfChannel -1)	
	FTP	TCP	21	
	Upgrade port	TCP	60024	

2.6 Tutorial License

From V4.60, tutorial license specification is changed as below.

- Time Limit

After tutorial license is enabled, it can be used for 60 days.

For reference, it can be used for 14 days before V4.60.

- Usable channels

When the tutorial license is enabled, each resource has max value which is different by system type. Refer to the “OfficeServ Capacity” document for more information.

[MAX] VM channels

[MAX] MGI channels



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- [MAX] AA channels
- [MAX] FAX channels
- [MAX] MOBEX Executive users
- [MAX] Soft phones
- [MAX] H.323 trunks
- [MAX] SIP trunks
- [MAX] Samsung SIP phones
- [MAX] 3rd-Party SIP phones
- [MAX] Samsung SIP applications
- [MAX] WEVoIP SIP phones
- [MAX] Remote dial SIP phones
- [MAX] IP Phones
- [MAX] WiFi Phones
- [MAX] Call Managers
- [MAX] CNF24 channels
- SPNET enable



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3. OfficeServ V4.6 Description

3.1 OfficeServ V4.6 Outline

OfficeServ V4.6 is system upgrade S/W.

Major functions : New VM, SLI3 cards, Mobile client, MGI channel resource improvement, PnP for IP phone installation, Security(TLS, sRTP), SIP Trunking enhancement, H.264 negotiation and DTMF sending for SIP phone.

3.2 OfficeServ V4.6 S/W structure

Skipped

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4. OfficeServ V4.6 Requirements

Detail functions which are implemented in OfficeServ V4.6 will be described in this section.

4.1 R1. New VM

4.1.1 R1.1. IP interface

5) Description

- The CPU and PHY hardware components of SVMi-20E are discontinued, so it needs to develop new voicemail/auto-attendant card.
- SVMi-20E card cannot interface with IP phone or trunk directly and use MGI card. It needs to improve it.
- New VM/AA card does not use MGI card to interface with IP phone or trunk.
- New VM/AA card is developed with Mindspeed platform and the software that is used in OS7100.
- It uses 2 DSP channels for IP connection to support a few functions that need to monitor the sound(Fax CNG, CED tone detection, Silence detection, In-band DTMF detection)
- If there is a problem to connect with SVMi-20i card via LAN port(for example, No connection, No IP setting, IP conflict), MGI is used like the previous SVMi-20E card.

6) Limitation

- This cards can be installed with OS7200(MP20)/OS7400(MP40).
- Only 1 card can be installed with the system.(Same with SVMi-20E)
- It can only support 16 ports if all ports use sRTP.
- MGI card is required for some features, for example VMMOH, VMAME and Overhead paging if it has to operate with IP phone or trunk because it's conference function.

7) Settings

- Configure IP address

: IP address can be set via KMMC 873 or DM(Device Manager) 2.2.17. This is different from previous SVMi-20E card. If SVMi-20i card is moved to other systems, IP address is changed according to the KMMC 873 or DM 2.2.17 configurations.



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The screenshot shows the OfficeServ Device Manager (DM) interface. The left sidebar contains a tree view of configuration options under 'Functional', with '2.2.17.SVMi-20i Card' selected. The main area displays a table of configuration parameters for this card.

Cabinet/Slot	C1-S2
IP Address	165.213.89.132
Gateway	165.213.89.1
Subnet Mask	255.255.255.0
IP Type	Private Only
Local RTP Port(start)	30000
Public IP Address 1	0.0.0.0
Public RTP Port 1	30000
Public IP Address 2	0.0.0.0
Public RTP Port 2	30000
Public IP Address 3	0.0.0.0
Public RTP Port 3	30000
FTP Port	21
Upgrade Port	60024

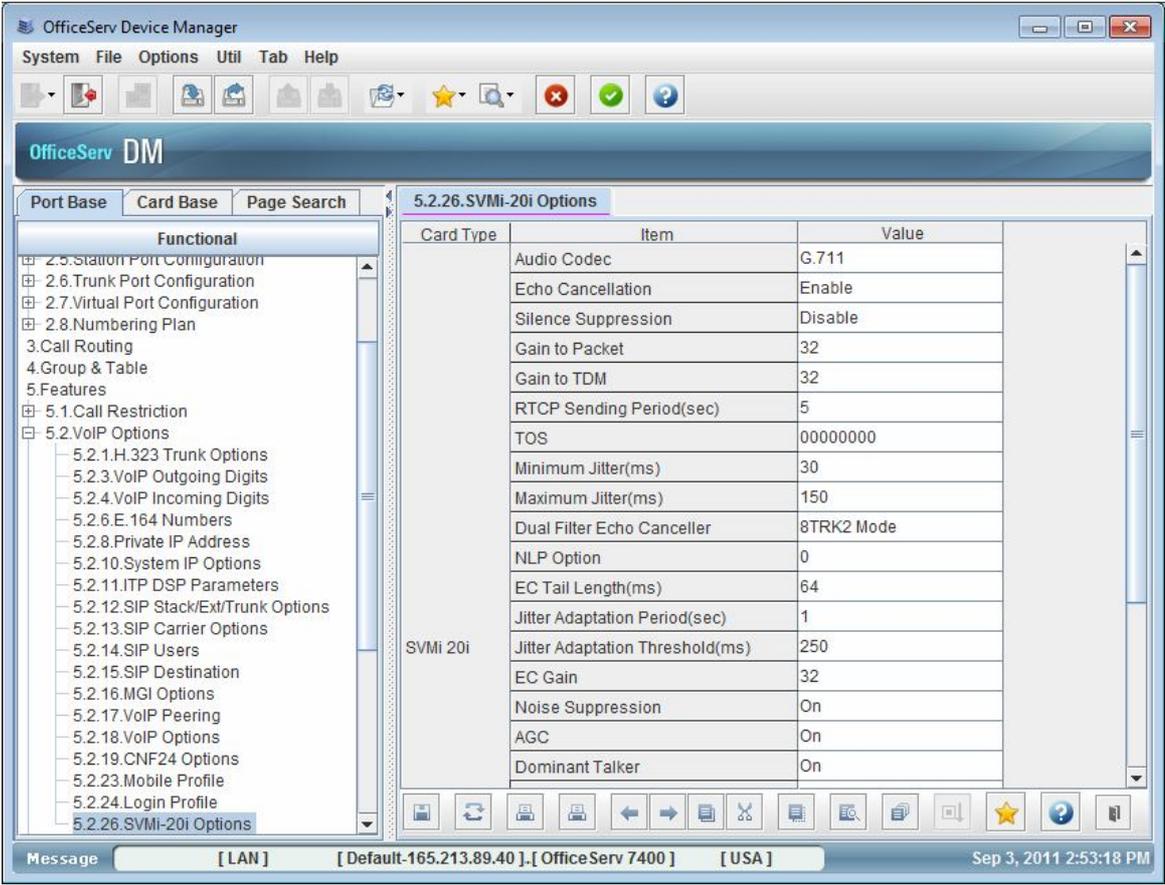
At the bottom of the window, there is a status bar with the following information: Message [LAN] [Default-165.213.89.40].[OfficeServ 7400] [USA] Sep 3, 2011 2:48:31 PM



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- Configure DSP Option

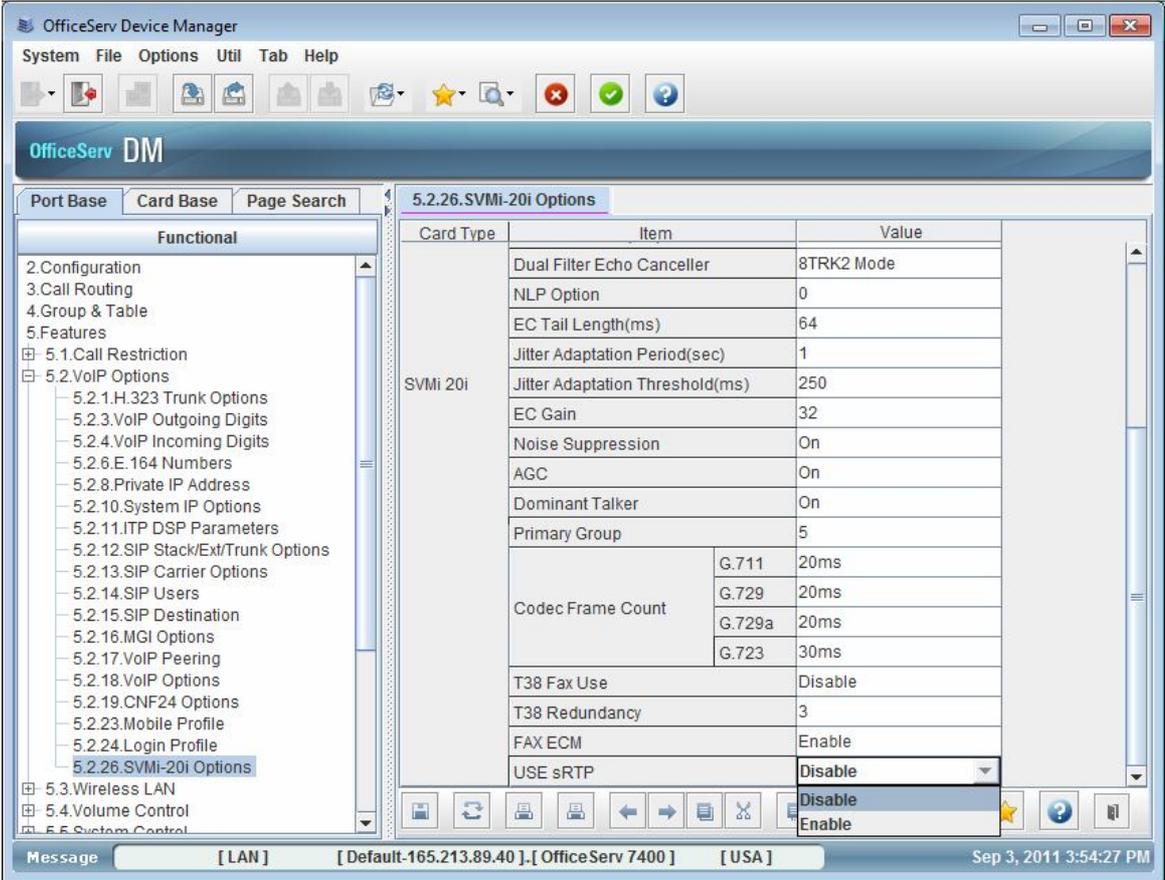
: To interface with IP phone or trunk, DSP option parameters are added like MGI card. It's available via KMMC 835 or DM 5.2.26.





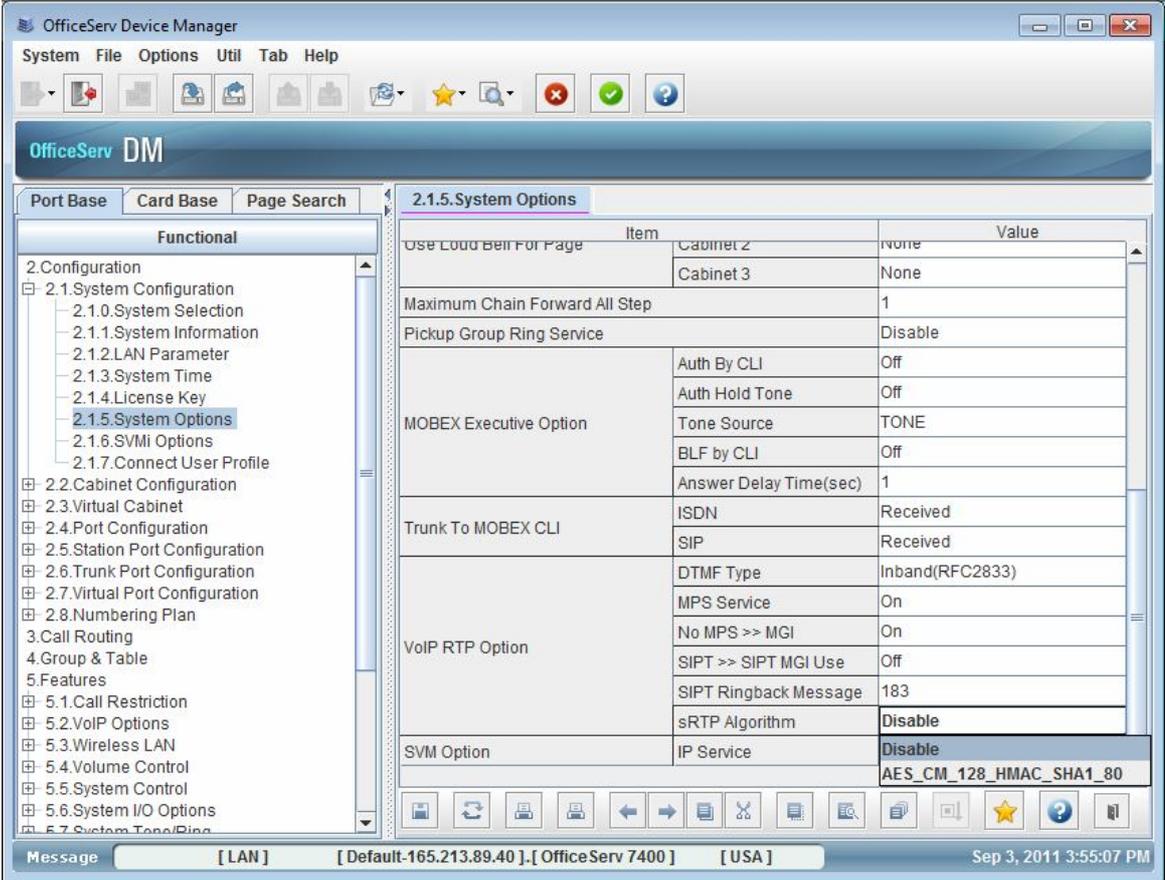
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- sRTP Option
 - : It supports AES algorithm for all countries and also supports ARIA algorithm for Korea.
 - : It can service 16 ports, if all ports uses sRTP.
 - : This can be set via MMC 835 or DM 5.2.26 and you also have to set MMC 861 or DM 2.1.5 to allow this feature for SVMi-20i card.





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4.1.2 R1.2. Fax Mail

- 1) Description
 - New VM/AA card should support Fax Mail/Fax on demand features that SVMi-20E supports.
 - The S/W fax module of IP-UMS is ported to implement this feature.
 - Available fax ports are controlled by s/w license.
 - Fax ports are not dedicated. If there are 20 ports and one port is used for fax, available ports is 19.
- 2) Limitation
 - None
- 3) Settings
 - License
 - : Available fax ports are controlled by license and maximum 4 ports are available.
 - : License page is now developing.(2011.09.17)
 - Support T.38 or Pass-through
 - : Fax mode for IP connection can be set via MMC 835 or DM 5.2.26.



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The screenshot shows the OfficeServ Device Manager interface. The left sidebar contains a tree view of configuration options, with '5.2.26.SVMi-20i Options' selected. The main area displays a table of configuration items for the SVMi 20i card type.

Card Type	Item	Value	
SVMi 20i	Dual Filter Echo Cancellor	8TRK2 Mode	
	NLP Option	0	
	EC Tail Length(ms)	64	
	Jitter Adaptation Period(sec)	1	
	Jitter Adaptation Threshold(ms)	250	
	EC Gain	32	
	Noise Suppression	On	
	AGC	On	
	Dominant Talker	On	
	Primary Group	5	
	Codec Frame Count	G.711	20ms
		G.729	20ms
		G.729a	20ms
		G.723	30ms
T38 Fax Use	Disable		
T38 Redundancy	Disable		
FAX ECM	Enable		
USE sRTP	Disable		



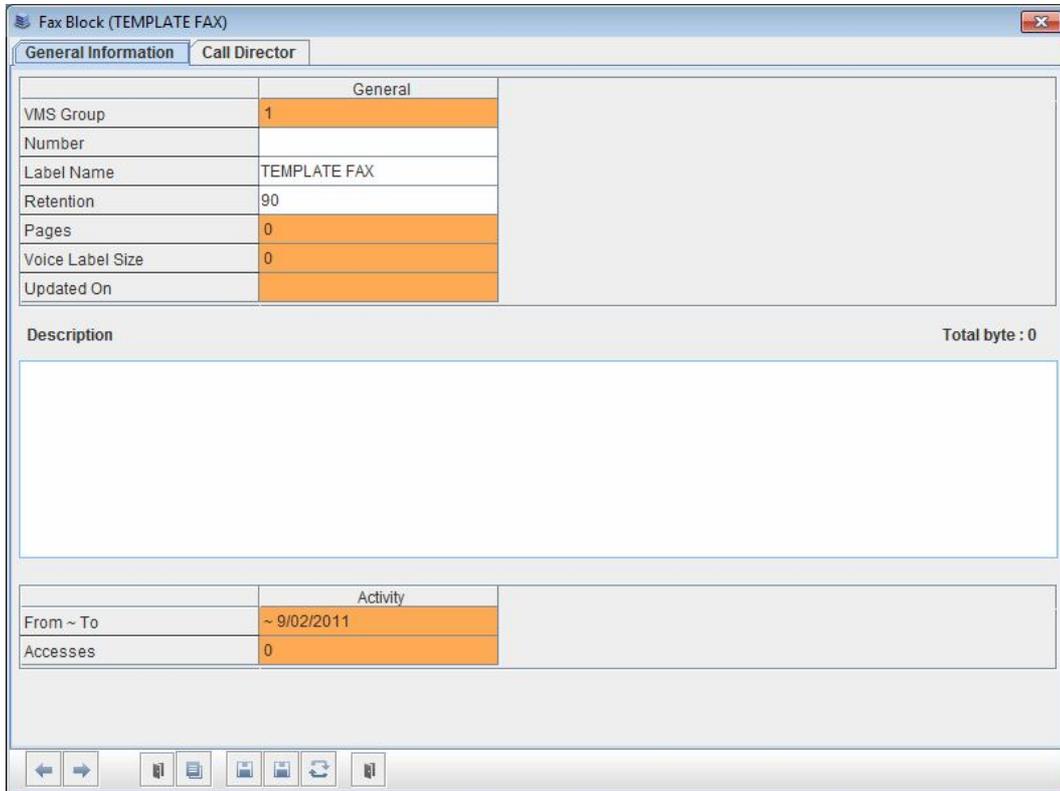
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- Document Lib, Fax block in block table are implemented to support this features. They are same with SVMi-20E.
- Document Lib block

General Information		Interface	Call Director	Prompts
VMS Group	1			
Label Name	Standard			
Max voice label size	60			
Retention days	0			
Maximum accesses	0			
Refresh retention	Yes			
User can create Fax	Yes			
Admin Password	0000			
Cover Page				
Station ID				
Local Area Code				
Ports to use				
	Authorize Y/N	Station Name		
On premise	Yes			
Local call area	Yes			
Long distance	Yes			
Wait for fax tone	45			
Delivery attempts	3			
Retry Interval for Busy	5			
Retry Interval for No-Answer	15			

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- Fax block



General	
VMS Group	1
Number	
Label Name	TEMPLATE FAX
Retention	90
Pages	0
Voice Label Size	0
Updated On	

Description Total byte : 0

Activity	
From ~ To	~ 9/02/2011
Accesses	0

4.1.3 R1.3. Web based management

1) Description

- The web interface of OS7100 or etc is too slow, so it needs to improve.
- Terminal programming interface of SVMi-20E is fast and stable, but the graphic is broken in Non-US OS environment and the trend is web interface.
- The programming interfaces are separated and different for MP, SVMi card. It should be integrated.
- DM(Device Manager) is developed to replace old interface and based on Java technology.
- Programming interface of MP and SVMi are integrated.

2) Limitation

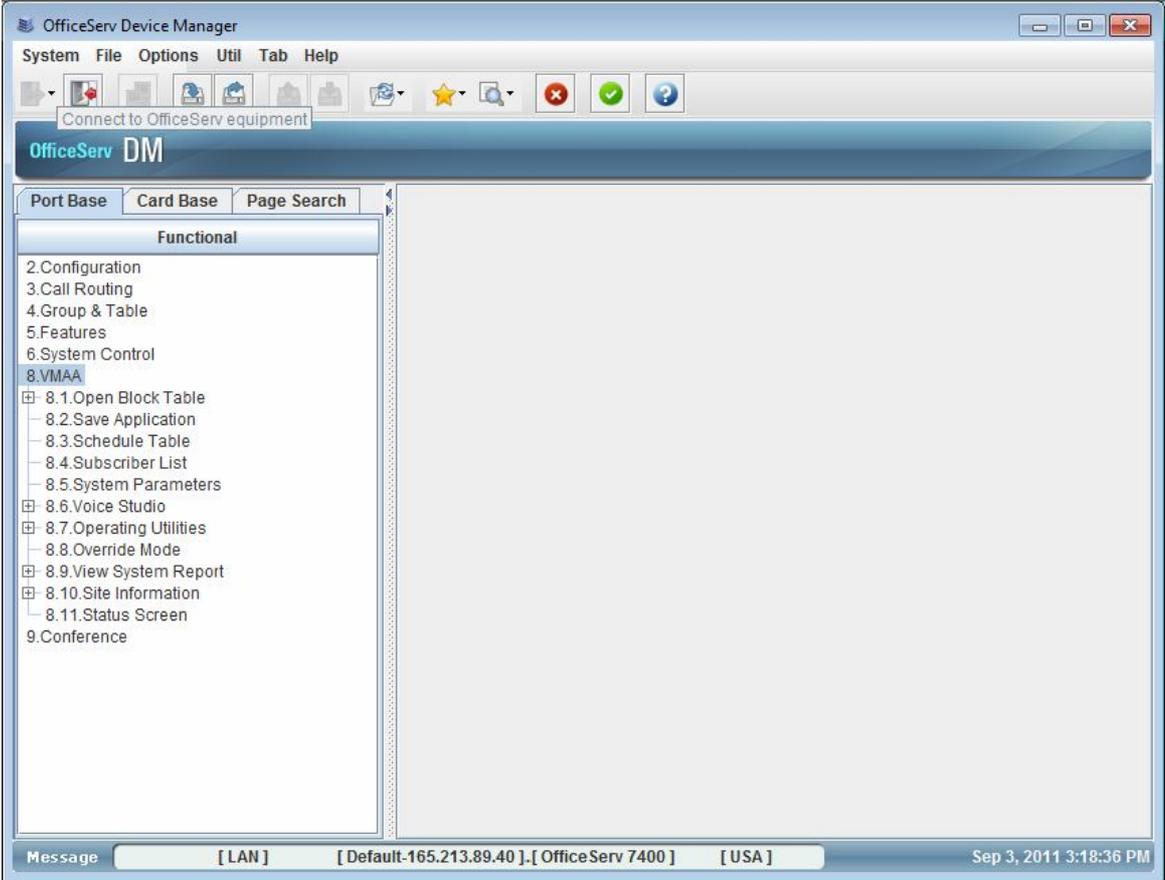
- None.

3) Settings

- The IP address of SVMi-20i card should be set to use DM for VM/AA.
- DM(Device Manager) connects the SVMi-20i via 6001 and 6002 port of SVMi-20i. So, DM has to be able to access that port in any circumstances. Make sure that ports can be accessed in your environments and if it's not, please ask your network administrator what it needs, for example setting port-forwarding in NAT network.



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4.1.4 R1.4. Email Gateway SSL/TLS encryption

- 1) Description
 - SVMi-20E does not support SSL/TLS encryption for E-mail gateway, but nowadays many public mail server requires this for security.
 - SVMi-20i supports SSL/TLS encryption for E-mail gateway.
 - Supported type of encrypted connection is SSL(starts encrypted connection directly) and TLS(Send "STARTTLS" ahead of encrypted connection)
- 2) Limitation
 - None
- 3) Settings
 - Changes of Mclass block : Two fields are added in email gateway tab.



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MClass Block (Standard)

General Public Caller Interface OutCall Prompts E-Mail Gateway Call Director

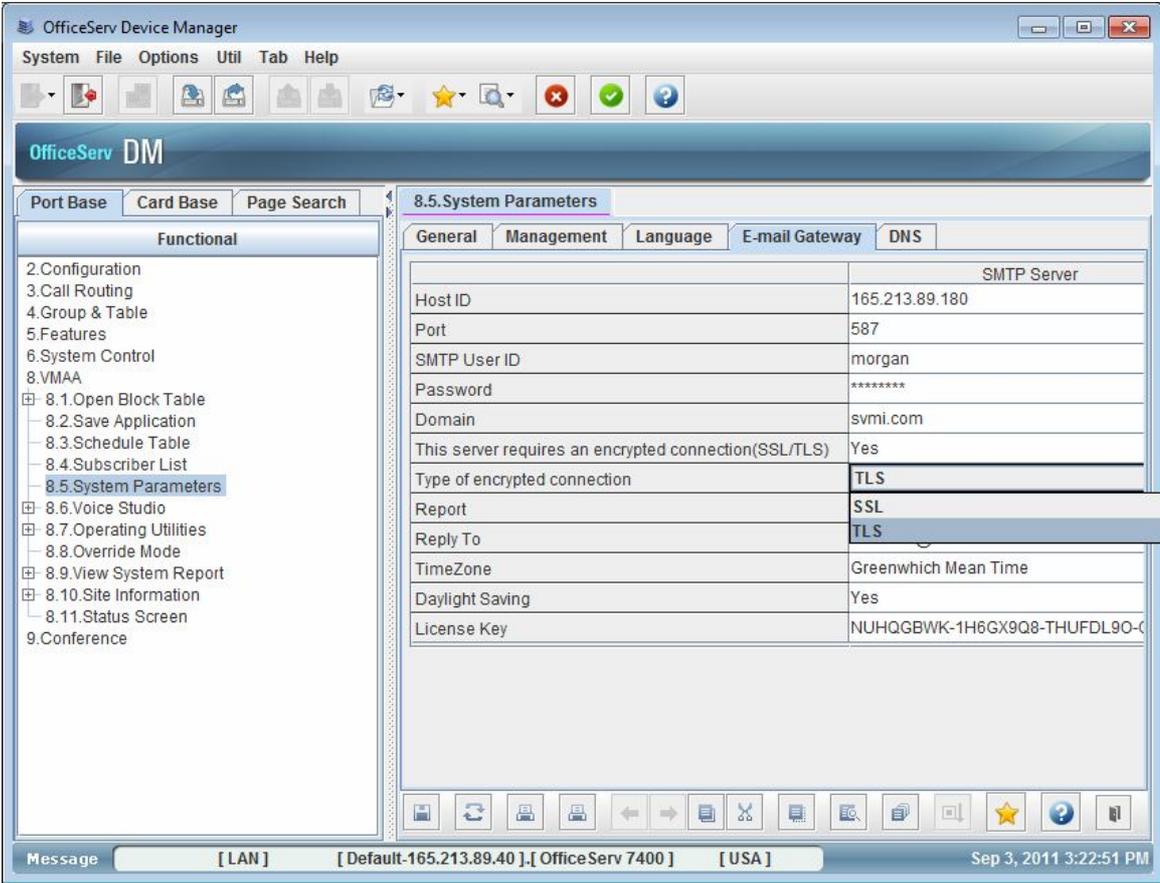
E-Mail Gateway	
Host ID	165.213.89.180
Port	587
SMTP User ID	morgan
Password	*****
Domain	svmi.com
Attempts	1
Retry Interval	5
Adjust message retention	<input type="checkbox"/>
Message retention to use	1
This server requires an encrypted connection(SSL/TLS)	Yes
Type of encrypted connection	TLS

Navigation icons: back, forward, home, print, refresh, stop.



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- Changes of System Parameter



4.1.5 R1.5. Upload wav format prompts

1) Description

- When a user tries to upload their prompt to SVMi-20E card, he has to convert it the format that the system uses.
- If a user upload wav format prompts at Voice Studio, SVMi-20i automatically converts it to the voice format that it uses.
- This feature is also applied to other systems(OS7100, OS7030, OS7070 and OS7200 MP20s)

2) Limitation

- It only supports one wav format(8000Hz, Mono, 16bit signed, 128kbps) when a user upload their prompts at the Voice Studio.

3) Settings



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- Voice Studio : In this page, press upload button, then you can see the pop-up window.

OfficeServ DM

8.6.1.Prompt

English, America

Call Add Delete Upload

1 / 70

Delete	No.	Description	Length(sec)
<input type="checkbox"/>	0001	"Thank you for calling." ...	1
<input type="checkbox"/>	0002	"An operator will be with you in a moment." ...	1
<input type="checkbox"/>	0003	"Our office hours are 8 AM to 5 PM, Monday throu..."	3
<input type="checkbox"/>	0004	"Our office is closed for the holiday." ...	2
<input type="checkbox"/>	0005	"Our office is closed due to emergency conditions..."	7
<input type="checkbox"/>	0006	"If you know the extension of the person you are c..."	3
<input type="checkbox"/>	0007	"To reach the sales department, press 2. For the s..."	4
<input type="checkbox"/>	0008	"To leave a message in our after hours message ..."	3
<input type="checkbox"/>	0009	"Sorry, that is not a valid entry. Please try again." ...	3
<input type="checkbox"/>	0010	"Sorry, that is not a valid entry. Please try again, o..."	4

Message [LAN] [Default-165.213.89.40] ['11.09.02 T4.60] [OfficeServ 7400] [USA] Sep 3, 2011 3:24:26 PM

Upload Prompt

[Supported Voice File Format]
wav : 8000Hz/Mono/16bit signed/128kbps
G.711 : 8000Hz/Mono/mulaw/64kbps

No	File	Select
1		File
2		File
3		File
4		File
5		File
6		File
7		File
8		File
9		File
10		File

Upload Close



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4.1.6 R1.6. Voice storage expansion.

1) Description

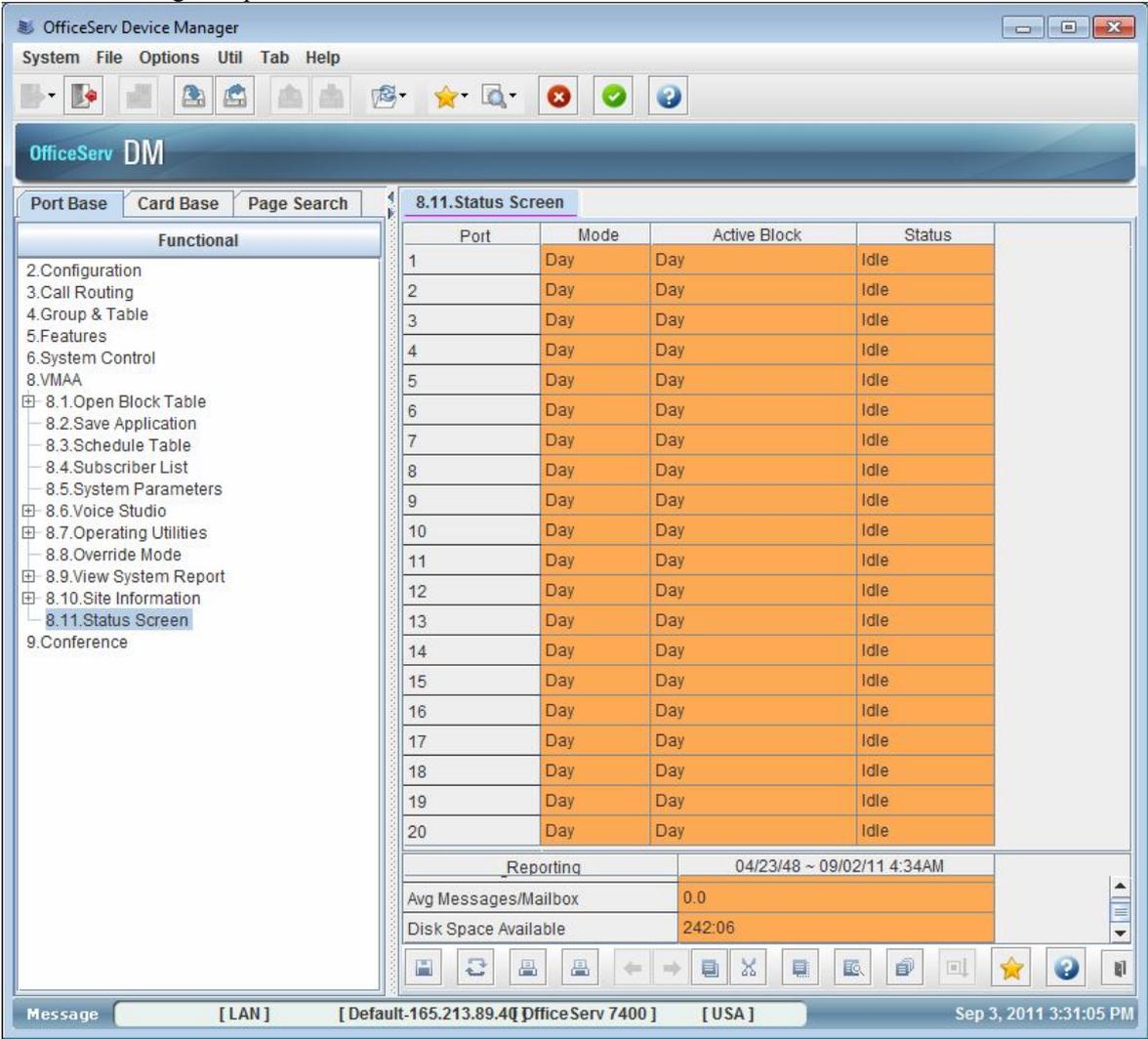
- Total recording time of SVMi-20E is 140 hours and it's not enough storage.(After this VOC, 4GB CF is applied to SVMi-20E and the recording time is increased to 270 hours.)
- 8GB SSD is used for storage device and total available recording time 240 hours.

2) Limitation

- Available recording time can be changed if multiple language prompts are installed.(It supports 240 hours when English, Spanish and French prompts are installed)

3) Settings

- No setting is required.





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4.2 R2. Subscription Card

4.2.1 R2.1 Embedded Sine wave Ringer

SLI3 has a built-in sine wave ringer. External ringer is no more required.

4.2.2 R2.2 Embedded DTMF Receiver

SLI3 has a DTMF receiver per port. It can detect DTMF digit without a common resource (MFR).

4.2.3 R2.3 Embedded CID Transmitter

SLI3 has a CID transmitter per port. It can transmit the FSK CID without a common resource (CID).

4.2.4 Considerations

LP40 package name has been is changed from LP4xxxxx.PGM to SP4xxxxx.PGM.

You can upgrade LP40 through MMC818 as before with MP40 V4.60 or higher.

The new PKG is consisted of two parts: BOOT ROM Upgrade part and LP40 S/W Program part. If the version of LP40 is V1.27 or below, the LP40 BOOT ROM is need to be upgraded for the new S/W. Because the S/W size what could be processed by old BOOT was reached to limit.

If you upgrade LP40 from V1.27(or lower) to V2.00(or higher), OS7400 upgrade LP40 BOOT ROM first and it will takes about 7 minutes. After that OS7400 upgrade LP40 S/W program automatically and it will takes about 6 minutes. Therefore if the upgrade of BOOT is required, it takes about total 13 minutes.

If you upgrade LP40 from V2.00(or higher) to V.2.xx, OS7400 upgrade only LP40 S/W program and it will takes about 6 minutes.

4.2.5 Limitation

- SLI3 cards support blink message waiting indication only. But in Italy, France and Spain, both Continuous MW and Interrupt MW are supported. Because those country use Polarity Reverse Message Waiting.
- Hot swap is only available on OS7400.
- LP40 BOOT ROM needs to be upgraded. But we support Auto-Upgrade feature

4.2.6 Settings

There is one minor change in comparison with SLI2. Because SLI3 cards do not support Continuous Message Waiting, you only have to set Message Waiting On/Off time in DM 2.5.7 SLI Data. But you can set Continuous Message Waiting to 16MWSLI as before.

This change is not applied in Italy, France and Spain.



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The screenshot shows the OfficeServ Device Manager application window. The title bar reads "OfficeServ Device Manager". The menu bar includes "System", "File", "Options", "Util", "Tab", and "Help". Below the menu bar is a toolbar with various icons. The main window is titled "OfficeServ DM" and has three tabs: "Port Base", "Card Base", and "Page Search". The "Page Search" tab is active, showing a tree view of system control options. The "5.5.0. System Control" option is selected and expanded, showing a list of sub-items. The "5.5.0. System Control" sub-item is selected, displaying a table of system control parameters.

Item	Value	
Reference Clock	Priority 1	C1-S1
	Priority 2	C1-S2
	Priority 3	C1-S4
	Priority 4	C1-S5
	Priority 5	C1-S6
	Priority 6	C1-S7
	Priority 7	C1-S8
	Priority 8	C1-S9
	Priority 9	C1-S10
Make/Break Ratio (Make, %)	33	
Pulse Per Second (pps)	10 pps	
16MW Message Waiting Lamp Cadence	Interrupted	
Message Waiting Interrupt Lamp On Time (100msec)	10	
Message Waiting Interrupt Lamp Off Time (100msec)	10	
Hook Off Time (10msec)	10	
Hook On Time (10msec)	100	
Hook Flash Min Time (10msec)	35	
Hook Flash Max Time (10msec)	80	
Power Down Time (100msec)	20	
VM Ringback Delay Time (sec)	0	

At the bottom of the window, there is a status bar with the following information: "Message [LAN] [null-165.213.66.44] ['11.08.18 G3.20] [None] [KOREA] 2011. 9. 19 PM 6:47:27".



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4.3 R3. WE VoIP Client

Samsung WE VoIP is the samsung's own mobile SIP client compatible with OfficeServ system. With WE VoIP, you can make or answer a VoIP call using the default dialer of your smart phone. It supports a normal SIP call feature and has been customized for OfficeServ's call features. For registering it to the OfficeServ system, extra (not SIP client) license key is required

WE VoIP's enhanced features that this client is having are restrictively provided in the following conditions.

Category	Specifications
OS	Android 4.0 or higher
Required space for installation	6 MB or more (10 MB or more recommended)
Resolution	480 × 800 or greater
Supported devices (Refer to the list below)	Galaxy S2/S2 LTE Galaxy S3/S3 LTE Galaxy Note/Note2/Note2 LTE

※ Product code list (Updated : Feb 06, 2013)

WE VoIP F/W version	Country	Product code
V2.4.1.3	Europe Australia South Africa	GT-I9300* GT-I9305 / GT-I9305T / GT-I9300T(Telstra) GT-N7100
	The Americas	D2 VZW : SCH-I535 D2 AT&T : SAMSUNG-SGH-I747 D2 TMO : SGH-T999 D2 SPR : SPH-L710 D2 USC : SCH-R530U T0 VZW : SCH-I605 T0 AT&T : SAMSUNG-SGH-I317 T0 TMO : SGH-T889 T0 SPR : SPH-L900



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		T0 USC : SCH-R950 C Spire : SCH-L710, USC : SCH-R530, Cricket : SCH-R530C MetroPCS : SCH-R530M
--	--	--

4.3.1 WE VoIP's basic configuration

1) Description

The client can register to the OfficeServ system when a user is in the WiFi area.

It supports various PBX features as below.

- A. Integrated dialer & call logs
- B. Hold/Resume
- C. Call Transfer
- D. Call Forward
- E. Direct Pickup
- F. Group Pickup
- G. Multi-Ring Setting (Station Pair)
- H. Missed Call Indication
: Missed call indication is not available in case that SIP client is registered as a ring group member and one of other members receives an incoming call.
- I. Provisioning
: For its registration, there are two necessary configuration options called as Mobile and Login profile.
- J. MOVE
 - a. Desk phone to Mobile phone which is paired with the desk phone.
 - b. Mobile phone to Desk phone which is paired with the mobile phone.
- K. Manual roaming transfer
: Seamless roaming from WLAN to 3G area..
- L. Un-regi Forward
: Customer call to the office number is automatically forwarded to the corresponding mobile number if the WE VoIP client is not registered.
- M. Smart Routing
: When the user makes a call in an office using the public mobile number, the system checks if the dialed number has a corresponding extension number . If it has, the call will be routed to a internal call.



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N. HD Voice (Internal call)

2) Limitation

- SIP and PRI trunk are only supported for WE VoIP's enhanced service such as Move, Manual roaming operating like the MOBEX feature.

3) License

- Input SIP license which includes WE VoIP and Remote dial count in DM 2.1.4.
- If SIP license is saved normally, Mobile phone is restricted by WEVoIP count.
(i.e) When the total count of SIP license is 'SIP (200) & WEVoIP (10)' and two WE VoIP clients among totally 10 are registered to the system, the free license count will be SIP (198) & WE VoIP(8).
- MP40 can support max 224 WE VoIP clients' registration.

1. OfficeServ configuration for WE VoIP

1) License

- Enter the WE VoIP license key in DM 2.1.4(MMC860).
- SIP phone license does not relate to WE VoIP client.

2) Profile

In the DM 5.2.23 and 5.2.24,

① Mobile Profile

- AP SSID

Basically, this should be identical to your local access point. If it differs from the SSID of local AP, WE VoIP client will not recognize that it is in a office where the OfficeServ system is located in and then will try to register with a public IP address of the system.

- Public zone service

Using an unpopular SSID is strongly recommended.

When it is unfortunately same to other one in a public area, the public zone service will not work at that area.



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5.2.23.Mobile Profile		Value
Item		
AP SSID		SMT
Upgrade Server		10.0.1.101
Upgrade URL		tftp
Auto Prefix Code		#
Auto Prefix Exception Number		XXXX,*65!,*66!,*60!,805,9!,804
MVS Local Port		9012
MVS Public Port		9012
MVS Disa Number		07088585570
VMS Public Number		07088585571
Codec Priority	1	SILK
	2	AMR-WB
	3	G.711u
	4	G.711a
SILK Codec	Payload	114
	Sampling Frequency	24000 Hz
	Max Ptime	100 ms
	DTX	Off
	FEC	On
AMR WB	Payload	113
	Bit Rate	23850 bps
	DTX	Off
Direct Mobile Number		02116

- Upgrade Server/URL
These options are needed for the WE VoIP S/W upgrade.
Upgrade server is managed separately and user should fill the server IP address and the directory information in the blank.
- Auto Prefix Code
The client will automatically prefix the code to a dial number.
- Auto Prefix Exception Number
The client will not prefix any code to a dial number, if any exception codes are included in a dial numbe

※ Exception rules

XXXX	4 any digits
XXX	3 any digits
65!	65 + any dialing number
*64	Any dialing number + 64

② Login Profile



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5.2.24.Login Profile															
Tel Number	Mobile Num	Noise Suppres...		Aecm	Echo	Swing Free		Cng	DTMF Ty...	Public Zone...	Scan 5G Only	USE sRTP	Multi Frame	Multicast	TOS
		Rx	Tx			Rx	Tx								
3304		Disable	Disable	Speaker Phone	Enable	Enable	Enable	Enable	RFC2833	Disable	5G Only	Disable	Disable	Disable	11100000
3305		Disable	Disable	Speaker Phone	Enable	Enable	Enable	Enable	RFC2833	Disable	5G Only	Disable	Disable	Disable	11100000
3306		Disable	Disable	Speaker Phone	Enable	Enable	Enable	Enable	RFC2833	Disable	5G Only	Disable	Disable	Disable	11100000
3307		Disable	Disable	Speaker Phone	Enable	Enable	Enable	Enable	RFC2833	Disable	5G Only	Disable	Disable	Disable	11100000
3308		Disable	Disable	Speaker Phone	Enable	Enable	Enable	Enable	RFC2833	Disable	5G Only	Disable	Disable	Disable	11100000

- Mobile Num (Mobile number)
This number will be used for making a unique file name for both mobile and login profile like <sec_mobile_017123456789.xml> , <sec_login_017123456789.xml>.
- Noise Suppression
Noise suppression of RX or TX (default: Disable)
- AECM (Default: Speaker Phone)
Quiet Earpiece of handset / Earpiece / Loud Earpiece / Speaker Phone / Loud Speaker Phone
- Echo: Echo cancellation (default: Enable)
- Swing Free : Swing Free feature of RX or TX (default: Enable)
- CNG : Comport Noise Generation (default: Enable)
- DTMF Type
In Voice / RFC2833
- Public Zone Service (Default: Disable)
When it is enabled, WE VoIP will work in public area/zone.
- Scan 5G Only (Default : Enable)
For reducing a handover delay, a channel scanning can be chosen among the following options.
Auto / 2.4G Only / 5G Only
If your access point doesn't support the 5Ghz bandwidth, it should be set to 'Disable'.
- USE sRTP : sRTP Enable / Disable
- Multi Frame
- Multicast
- TOS
- Jitter
- SIP Signal Type
UDP / TCP / TLS
- Description

- ③ MOBEX station setting
Several MOBEX station settings are required for using 'Move/Manual roaming/Un-registered forward' features, because these setting are related to make an outgoing call to mobile number.
B. Input MOBEX station number in DM 2.8.0

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- C. Configure 'MOBEX Extension' option.
Assign a SIP station number into <Master Station> in the DM 2.7.5 for adjusting CLI number.
- D. Input the MOBEX station number to <Call forward unreachable> in the DM 2.7.2.

④ SIP Expire Time setting

Be sure not to exceed 600 seconds for SIP Expire Time of SIP Extension Configuration in DM 5.2.12. Max expire time of mobile client is 600 seconds and if expire time of client exceeds 600 seconds, mobile client does not try to register.

⑤ MOVE feature code

Make a code for MOVE feature in the DM 2.8.0. This value will be saved at <Mobile Transfer Code> in the client profile and if there is no valid value in this option, MOVE and Manual roaming can't be used.

1. Mobile phone setup for WE VoIP

① WiFi setting

In mobile phone's WiFi setting menu, input AP password and its IP address.

② Provisioning

- Install the client software, after then a provisioning menu will be displayed.

Input the system IP address in <Provisioning Server IP> .



- Your mobile number will be automatically added in the <Cellphone number> field.



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But if no USIM card is installed on the phone, you have to insert the number manually for matching with a 'station number' in the DM 5.2.24.

- Press the menu button and <Request Profile> to get the mobile and login profile from the system.
- Press and dial the hidden code <2580##**> and check the profile information downloaded.

4.3.2 R3.1. Call Move functions

1) Description

A. Desk phone (IPP, DGP) to WE VoIP

When a user selects 'MOVE' key on a desk phone, WE VoIP client will be ringing. After then a user answers the call on WE VoIP client, the call in progress on the desk phone will be automatically disconnected.

B. WE VoIP to a desk phone (IPP, DGP)

When a user selects 'MOVE' key on a desk phone, WE VoIP client will be disconnected from the call in progress and it will be moved to the desk phone with no ringing.

2) Restrictions

Although other type of handsets can be paired with the WE VoIP client, there are two restrictions.

- In case of SLI/WIP phones, 'Call move' is only supported by pressing a pre-assigned number for 'MOVE' feature code.
- SIP phone/client except for WE VoIP client doesn't support the 'Call Move' feature.

3) Settings

- In DM 4.2.1 'Station Pair' option, make WE VoIP client paired with a desk phone (IPP/DGP).
- In DM 4.9.2 'Station Key' option, make 'MOVE' feature key on a desk phone.
- In DM 5.14.2, set the 'Move Wait Time' (Default : 20secs)

4.3.3 R3.2. Remote dial service

This feature has not been officially released and a practical use at a customer site is limited.

Your lab test and trial use will be only allowed.

1) Description

- Mobile phone can connect with OfficeServ by using Remote dial Client even though mobile user is out of the designated WiFi zone and is located in public network. In this case OfficeServ provides only some features
- Remote dial features are provided when mobile phone connects to Remote dial server which is embedded in OfficeServ.
- All types of system provide Remote dial features.



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- Remote dial features are as below.
: Click-to-Dial, DND status display/setting and VM status display/connection.

2) Settings

A. Remote dial certification

In OS7400 and OS7200 certification files should be saved in the SD card before starting the. But in other cases such as OS7100, OS7030, OS7070 and MP20s, these files are already copied in the system software. Certification files are saved in “/certification/mvs/” directory of the SD card.

B. Remote dial License

Remote dial license is assigned to MOBEX station as MOBEX Executive License. In this case Remote dial license should be assigned to MOBEX station which is designated as unregistered forward destination of specific Mobile phone.

- Remote dial license will be used to enable MOBEX Executive license and to allow Remote dial server connection.
- Settings
DM 2.1.4: Input License Key and check Remote dial count.
DM 2.7.5: Assign Remote dial license to each MOBEX station

C. Remote dial port Forward

- When OfficeServ is located in NAT, user should set Port Forwarding.
- Local port of system is fixed as 9012 and assign Router public port to system public port. (default: 9012)
- Settings
DM 5.2.23 Mobile Profile: Remote dial Public Port
➔ Mobile phone should get Remote dial public port by provisioning in advance.

3) Remote dial features

- Click-to-Dial
- DND status and setting

4.3.4 R3.3 VM display and connection

1) Description

OfficeServ sends VM information to mobile phone when mobile user requests. Only VM information is sent to the phone when mobile phone user requests. And the phone requests



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VM info periodically. This period can be changed in MWI option.

If VM message is existed, mail icon is displayed on the upper line Limited Service and drag down this status display to check VM message. You can check VM information as “There are VM messages. x/y”. In this case x is a number of new VM messages and y is a number of old VM messages. If new VM message is not existed, x will not be shown. You can hear the VM message to click a mail icon.

2) Settings

- Auto By CLI to ON in DM 2.1.5.
- AA Goto VM to ON in DM 2.7.5.
- In DM 5.2.23 input DID number which is needed for connecting VMS from the outside. In this case mobile phone should get above DID number by provisioning in advance.
- VMS group as DID destination in DM 7.1.4.

4.3.5 R3.4 Remote dial : DISA

This feature has not been officially released and a practical use at a customer site is limited. Your lab test and trial use will be only allowed.

1) Description

Mobile user who has Remote dial license can use MOBEX Executive features as its executive user. If Remote dial Click-to-Dial is failed, new button will be displayed for connecting to the system. If you click this button, system will be called automatically. In this case, a user hears dial tone and system does not make a real call but just waits for dialing a outgoing number by a user.

2) Settings

- Auth By CLI to ON in DM 2.1.5.
- In DM 5.2.23 input Remote dial disa number which is needed for connecting MOBEX station from the outside. In this case mobile phone should get above Remote dial disa number by provisioning in advance.
- MOBEX feature code as DID destination in DM 7.1.4.

4.3.6 R3.5. Call forward unreachable

1) Description

- Case 1 : Plug-out
When a WE VoIP client is disconnected normally, an incoming call to the client will be transferred to a pre-assigned call number in the ‘Call forward unreachable’ option.
- Case 2 : No Response
If a WE VoIP client does not respond to an INVITE message from the OfficeServ system within a

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timer set in the 'Call forward unreachable Time' option, the call will be transferred to a pre-assigned call number in the 'Call forward unreachable' option.

2) Settings

- In DM 2.7.5 'Mobile Extension' , assign a trunk number and input a mobile number in the 'outgoing digit' option.
- In DM 2.7.2 'SIP Phone Information', enter the Mobex station number to the destination of 'Call forward unreachable' option.
- In DM 5.14.2, set the 'Call forward unreachable Time'. (Default : 5secs)

4.3.7 R3.6. Manual handover

1) Description

When a user presses the button 'To Mobile' on WE VoIP client, the phone will be ringing by another new call with holding current call. If a user answers the call, the initial call will be transferred to without holding the opposite party.

2) Settings

- In DM 2.7.5 'Mobile Extension' , assign a trunk number and input a mobile number in the 'outgoing digit' option.
- In DM 2.7.2 'SIP Phone Information', enter the Mobex station number to the destination of 'Call forward unreachable' option.
- In DM 2.8.0 'Numbering Plan', assign any non-duplicated number to 'MOVE' key.

4.3.8 R3.7. Smart Routing

1) Description

When a user makes an external call via 'Mobile' button on WE VoIP client, the OfficeServ system checks if the dialed number has a corresponding extension number and the station's status is idle.

If exists, the call will be redirected in the system to pass through its own trunk line.

2) Settings

- In DM 2.7.2 'SIP Phone Information', enter the Mobex station number to the destination of 'Call forward unreachable' option.
- In DM 5.2.24 'Login Profile', enter a mobile number that WE VoIP client is having.

4.4 R4. Advanced MGI Channel Allocation Scheme



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4.4.1 R4.1. Advanced RBT Feature (RTG)

1) Functional Requirements

In the version prior to V4.60, Rinback Tone or Hold Tone was supported by MGI at no-MGI solution. RTG service will replace the above case and MGI channel is not used about no MGI solution. RTG module supports G.711,G.729a and G.722. The basic RTG features are the following

- Support Ringback Tone/ Hold Tone Service
- Support NAT traversal feature
- Support RFC 2833 detection feature for MOBEX Service
- Support RTP/sRTP

2) Limitation

The max number channel of RTG is the half number of MPS channels. For example, 32 channels will be supported at OAS cards and 8 channels at OS7100.

3) Setting

RTP port configuration is at DM 2.2.15. If a system is located in NAT network, public port must be configured manually and carefully. RTG's default RTP port is 45000.

2.2.15.MPS/RTG Card	
Cabinet/Slot	C1-S8
Card Type	OAS
IP Address	30.0.0.52
Gateway	30.0.0.2
Subnet Mask	255.255.255.0
IP Type	Private with Public
MPS Local Port	40000
MPS Public IP Address 1	10.254.168.213
MPS Public Port 1	40896
MPS Public IP Address 2	0.0.0.0
MPS Public Port 2	40000
MPS Public IP Address 3	0.0.0.0
MPS Public Port 3	40000
RTG Local Port	46000
RTG Public Port 1	45000
RTG Public Port 2	45000
RTG Public Port 3	45000
RTG Frame Count	20ms

Also, RTG status is DM 6.2.10. Its information includes the remote IP address, remote port, and codec.



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6.2.10.RTG Status							
Cabinet/Slot	Index	Status	Source			Codec	Tone Type
			Tel Number	IP Address	RTP Port		
C1-S8	237						
	238						
	239						
	240						
	241						
	242						
	243						
	244						
	245						
	246						
	247						
	248						
	249						
	250						
	251						
	252						
	253						
	254						
255							
256							

4.4.2 R4.2. Hybrid Voice Paging

1) Functional Requirements

Voice paging service for OSPPs can be supported by unicasting or multicasting, which is called by Hybrid Voice Paging. Using HVP, MGI channel usage is decreased on occasion. In a normal case, OSPP phones co-located with system are serviced by multicasting and it uses one MGI channel. And other phones are serviced by unicasting and each phone uses one MGI channel.

2) Limitation

Hybrid Voice Paging must satisfy the following conditions.

- System : Over v4.60
- MGI16/64: Over v1.27
- OAS: Over v2.03
- OSPP: SMT-i2200/2205/3100/5210/5220/5230/5240/5243/7130

3) Setting

Hybrid voice paging is configured at DM 4.1.3, DM 2.7.1, and DM 5.2.25.



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DM 4.1.3 Paging Groups includes paging members and multicast address for HVP and multicast address ranges from 224.0.1.0 to 239.255.255.254.

4.1.3. Page Groups				
Member	Zone 0	Zone 1	Zone 2	
80				
81				
82				
83				
84				
85				
86				
87				
88				
89				
90				
91				
92				
93				
94				
95				
96				
97				
98				
99				
Multicast Addr	239.0.0.1	239.0.0.2	239.0.0.3	239.0.0

DM 2.7.1 is ITP information and Multicast Page option is appended. 'Auto' option means that system determines unicasting or multicasting automatically. 'On' option indicates multicast paging and 'Off' indicates unicasting paging.



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2.7.1.ITP Information								
Tel Number	DSP Type	Time Zone	Signal Type	Video Codec	Video Size	QoS Enable	USE sRTP	Multicast Page
3201	G.729a	+00 00	UDP	H.263	CIF	Disable	Enable	Auto
3202	G.729a	+00 00	UDP	H.263	CIF	Disable	Enable	Auto
3203	G.711	+00 00	UDP	H.263	CIF	Disable	Enable	Auto
3204	G.729a	+00 00	UDP	H.263	CIF	Disable	Enable	Auto
3205	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3206	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3207	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3208	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3209	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3210	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3211	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3212	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3213	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3214	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3215	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3216	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3217	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3218	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3219	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3220	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto
3221	G.729a	+00 00	UDP	H.263	CIF	Disable	Disable	Auto

DM 5.2.25 multicast page ip list defines the network zone in which multicast is available.

5.2.25.Multicast Page IP List	
Index	Multicast Page IP List
1	0.0.0.0
2	0.0.0.0
3	0.0.0.0
4	0.0.0.0
5	0.0.0.0
6	0.0.0.0
7	0.0.0.0
8	0.0.0.0
9	0.0.0.0
10	0.0.0.0
11	0.0.0.0
12	0.0.0.0
13	0.0.0.0
14	0.0.0.0
15	0.0.0.0
16	0.0.0.0
17	0.0.0.0
18	0.0.0.0
19	0.0.0.0
20	0.0.0.0
21	0.0.0.0

4.5 R5. Enhanced PNP



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4.5.1 R5.1. PNP mode

1) Description

- OfficeServ provides 3 types of PNP mode as below.
 - A. Pre-Mac address

IP phone which is set to PNP mode can connect to the system when its Mac address is saved as user ID in DM 2.7.1.
 - B. Auto PNP

If IP phone tries to connect to the system by PNP mode, system searches free user ID and password and returns these values to the IP phone. Then IP phone connects automatically with this login information and in this case user does not have to set any options.
 - C. Normal login

If IP phone tries to connect to the system by PNP mode, system returns a specific message to change phone's display to accept user ID and password. Then user enters user ID and password which was received from system manager previously.

※ For reference, this process is named as "Normal login" because it is similar to normal login process. But it is one of PNP mode and does not equal to normal login process.

So do not confuse Normal login of PNP mode with normal login process.
- Default option is "Auto PNP".
- Before V4.60 user should set external DHCP server when using PNP mode. But now from V4.60 user does not have to set external DHCP server because OfficeServ systems except MP40 and MP20 provide embedded DHCP server for easy connection. User can activate embedded DHCP server by setting DHCP server use and start/end IP address in DM 5.2.10.

2) Limitation

- Supported IP Phones : SMT-i2200, SMT-i5210, SMT-i5220, SMT-i5230, SMT-i5240, SMT-i5243, SMT-i3100
- When using MP40 and MP20, user should set external DHCP server same as before. Refer to the below setup for detail setting of external DHCP server.

3) Setup

3-1) Set PNP mode

- Select PNP mode in DM 5.2.10. (Pre-Mac address/Auto PNP/Normal login)

If PNP mode is set to Pre-Mac address, system manager should input IP phone's mac address as User ID in DM 2.7.1.
- Default value: PNP Mode (Auto PNP)



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3-2) Set DHCP server

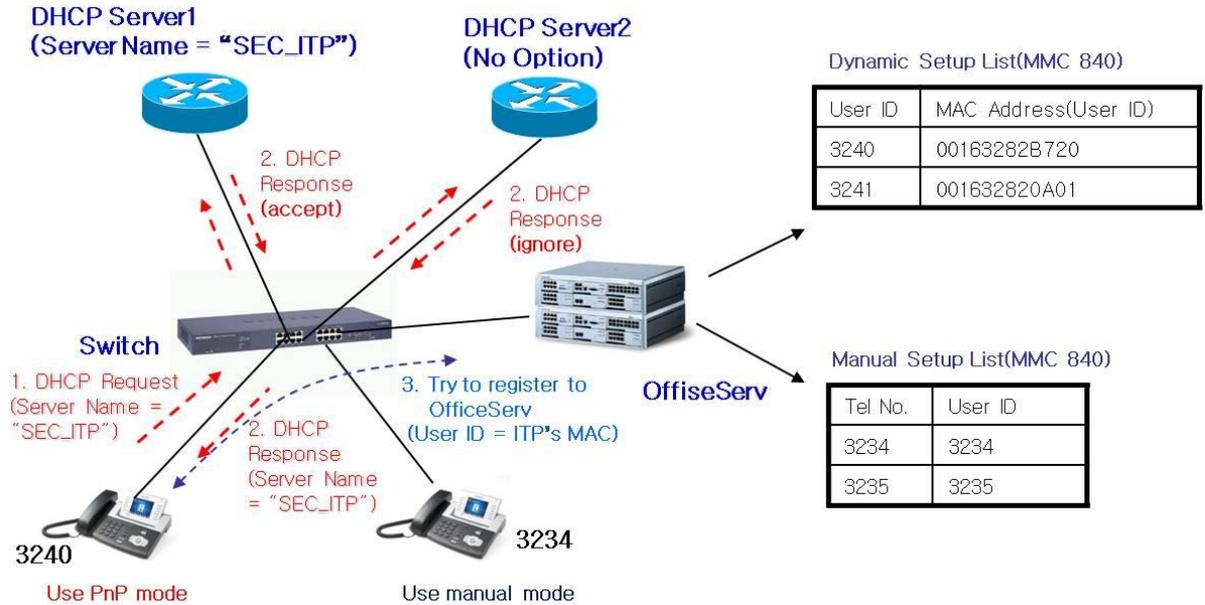
◆ How to set embedded DHCP Server

- In case of using OfficeServ systems except MP40 and MP20.
- Set DHCP Server Use to Enable in DM 5.2.10. If you want to use external DHCP server, select Disable. In case of MP40 and MP20 DHCP Server options are not shown.
- Set DHCP Server Start/End Address in DM 5.2.10.
- Then reboot system after completing above DHCP setting.
Be sure that system must restart to apply changed Start and End Address.
When IP phone tries to connect to the system by PNP mode, DHCP server allocates one of DHCP addresses to IP phone.
- Default value: DHCP Server (Disable), Start/End Address (0.0.0.0)

5.2.10. System IP Options		
Item		Value
ITP Max TX Limit		No
ITP Idle Logout	Type	MMC Command
	Start Time (Hour)	22
	Start Time (Min)	22
WIP DSP Parameter	Frame Count	40ms
	Echo Cancel	Enable
DHCP Server	Use	Enable
	Start Address	0.0.0.0
	End Address	0.0.0.0
PNP Mode		Auto PNP

◆ How to set external DHCP Server

- In case of using MP40 and MP20.
- Following two DHCP server options are used and must be set before using PNP mode.
 - A. TFTP_Server_Name(66) : “SEC_ITP” (66 is DHCP option number).
This option is used to distinguish my DHCP server from the other one under the circumstance one more DHCP servers are running.
 - B. TFTP_Server_IP(128) : OfficeServ IP address (128 is DHCP option number).
Using this IP address, IP phone tries to register to OfficeServ.



※ It is not possible to explain all DHCP options and entire setup process because various types of DHCP server are used. We recommend that you should refer to your DHCP server configuration guide for more information.

4.5.2 R5.2 PKG Upload

4) Description

- In specific MMC menu, system manager can upgrade connected IP phones at the same time.

5) Setup

- Set TFTP server and upload phone package.
- Input TFTP server information in Upgrade Server IP Address of DM 5.2.10.
- In MMC914, select UPGRADE TYPE (TFTP, HTTP) and set AUTO UPGRADE to Enable.



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4.6 R6. OfficeServ Device Manager

DM provides a familiar user interface (UI) for the OfficeServ system management and can be adopted as IT tool substitute.

4.6.1 R6.1 Web management

User can connect the web server on the OfficeServ system with using the DM as new web management tool. Whereas, the installation tool (IT Tool) does not support V4.60 and above.

4.6.2 R6.2 System statistics view

DM shows the statistics of CPU, Memory and Call and provides the usage menu of CPU and Memory.

4.6.3 R6.3 Executable file of DM

Executable file of DM is available and user can use it without any installation on the PC.

4.6.4 Limitation

- Language support : Korean, English, Russian, German, Italian, Spanish.
- DM(V4.65) doesn't support Java7 yet, so that Java6 installation is required.

4.6.5 Settings

Refer to "OfficeServ DM User Guide.doc"



DM User
Guide(Ed_01).doc

4.6.6 Modified Features

Refer to "DM V4.65 (2013.03.18) Release note.doc"



DM V4.64
Release note



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4.7 R7. Security Enhancement

4.7.1 R7.1. TLS for SIP signaling

1) Description

This feature enables OfficeServ system to use secure transport protocol for SIP messages.

2) Limitation

OfficeServ7400 and OfficeServ7200 (MP20) supports SIP TLS. The other models don't support it.

3) Settings

There are three different cases for SIP TLS which are SIP Carrier, SIP Peering and SIP Station. Below is the description for each different case.

A. Using SIP TLS for SIP Carrier

- Configure SIP TLS related parameters in DM 5.2.13 SIP Carrier Options.
- Port parameters such as Outbound Proxy Port : Set remote SIP TLS port for SIP Carrier. It is typically 5061.
- URI Type : SIPS URI Type is added. You can use the added URI type if the SIP Carrier requests SIPS URI type from IP-PBX.
- SIP Signal Type : TLS is added. You must set SIP Signal Type to TLS if you want to use SIP TLS for SIP Carrier.
- SIP Connection Reuse : This is only valid if SIP Signal Type is set to TCP or TLS. If this is set to ENABLE, SIP INVITE reuses the TCP/TLS connection which has been established in SIP REGISTER. If this is set to DISABLE, new TCP/TLS connection will be established whenever making call to SIP Carrier. You can reduce system load by setting this parameter to ENABLE. SIP Carrier side should also support SIP Connection Reuse.
- SIP Mutual TLS Enable : This is only valid if SIP Signal Type is set to TLS. If this is set to ENABLE, OfficeServ system will request certificate from SIP Carrier when SIP Carrier tries to connect to OfficeServ system by TLS (e.g. incoming call case).
- SIP Validate Any TLS Certificate : This is only valid if SIP Signal Type is set to TLS. If this is set to DISABLE, OfficeServ system will abort TLS connection when the certificate from SIP Carrier is unknown certificate. You have to put the certificate of SIP Carrier in SD card in OfficeServ system beforehand if this parameter is set to DISABLE. If this is set to ENABLE, OfficeServ system will omit the verification process for the certificate.



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The screenshot shows the OfficeServ DM configuration interface for SIP Carrier Options. The left sidebar lists various configuration categories, with '5.2.13.SIP Carrier Options' selected. The main area displays a table of parameters for SIP Carrier 2. Several parameters are highlighted with red boxes:

Item	Value
Alternative Outband Proxy	0.0.0.0
Outbound Proxy Port	5061
Proxy Domain Name	ug1.scm.com
Local Domain Name	
DNS Server 1	0.0.0.0
DNS Server 2	0.0.0.0
User Name	2002
Auth Username	2002
Auth Password	0000
Regist Per User	Disable
Session Timer	None
Session Expire Time	1800
Trunk Reg Expire Time	1800
Alive Notify	None
Alive Notify Time	1800
IMS Option	Disable
P Asserted ID Use	None
SIP Peering	Disable
Send CLI Table	1
Supplementary Type	PBX Managed 2
302 Response	Disable
SIP Destination Type	To Header
Codec Auto Nego	Enable
URI Type	SIP
SIP Signal Type	TLS
RACK Support	Disable
Hold Mode	Send Only
Response to Tag	Keep
SIP Connection Reuse	Disable
SIP Mutual TLS Enable	Disable
SIP Validate Any TLS Certificate	Disable

The status bar at the bottom shows: Message [LAN] [OS7100-165.213.80] [OfficeServ 7100] [KOREA] 2011. 9. 5 AM 9:17:28

B. Using SIP TLS for SIP Peering

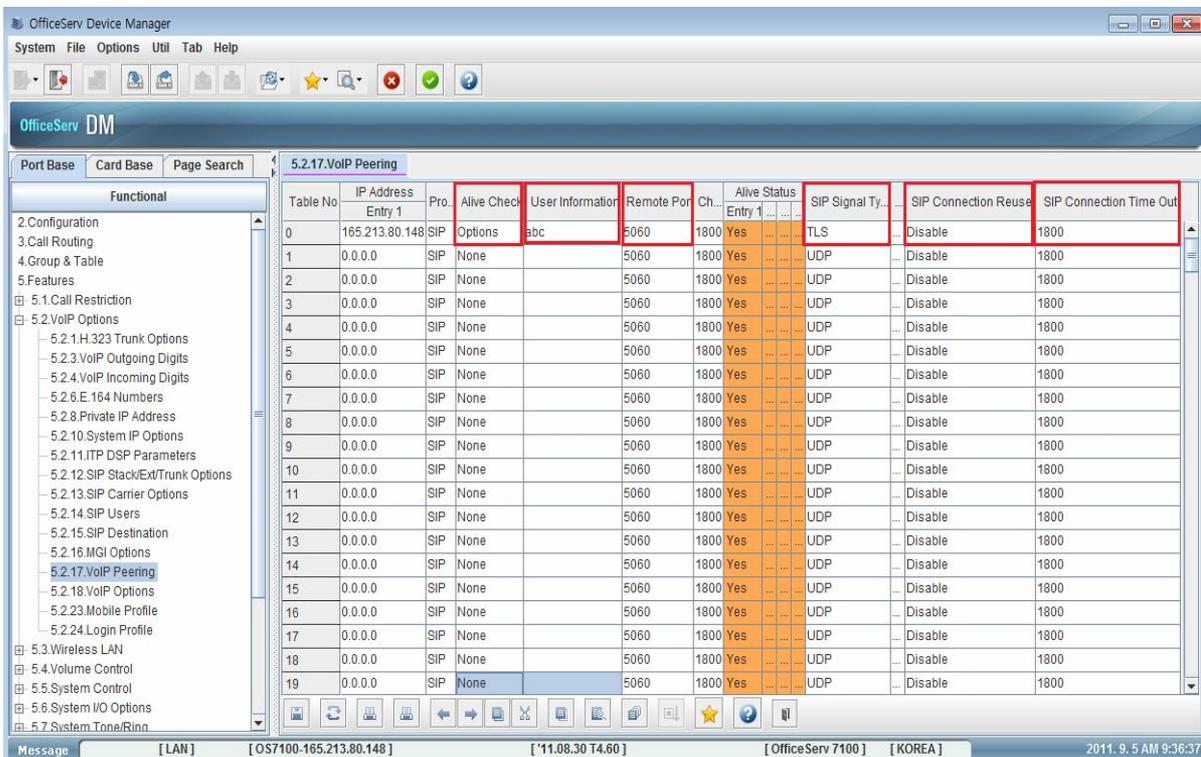
- Configure SIP TLS related parameters in DM 5.2.17 SIP VoIP Peering.
- Alive Check : This parameter should be set to "Option" if Connection Reuse is set to ENABLE.
- User Information : The value of this parameter is mandatory and should be identical with one of remote peering system. It is used for searching Peering Table index and sending successful



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response when receiving keep alive from remote peering system.

- Remote Port : This parameter should be TLS port of remote peering system. It is typically 5061.
- SIP Signal Type : TLS is added. You must set SIP Signal Type to TLS if you want to use SIP TLS for SIP Peering.
- SIP Connection Reuse : This is the same as that of the SIP Carrier configuration. The only difference is that remote system is not SIP Carrier but SIP Peering.
- SIP Connection Timeout : This is only valid if SIP Connection Reuse is set to ENABLE. This is the expiration time for the established connection. If the remote peering system doesn't refresh keep-alive before the expiration, OfficeServ system will tear down the reused established connection. The value of this timer should not be less than "Check Timer" value of peer system.



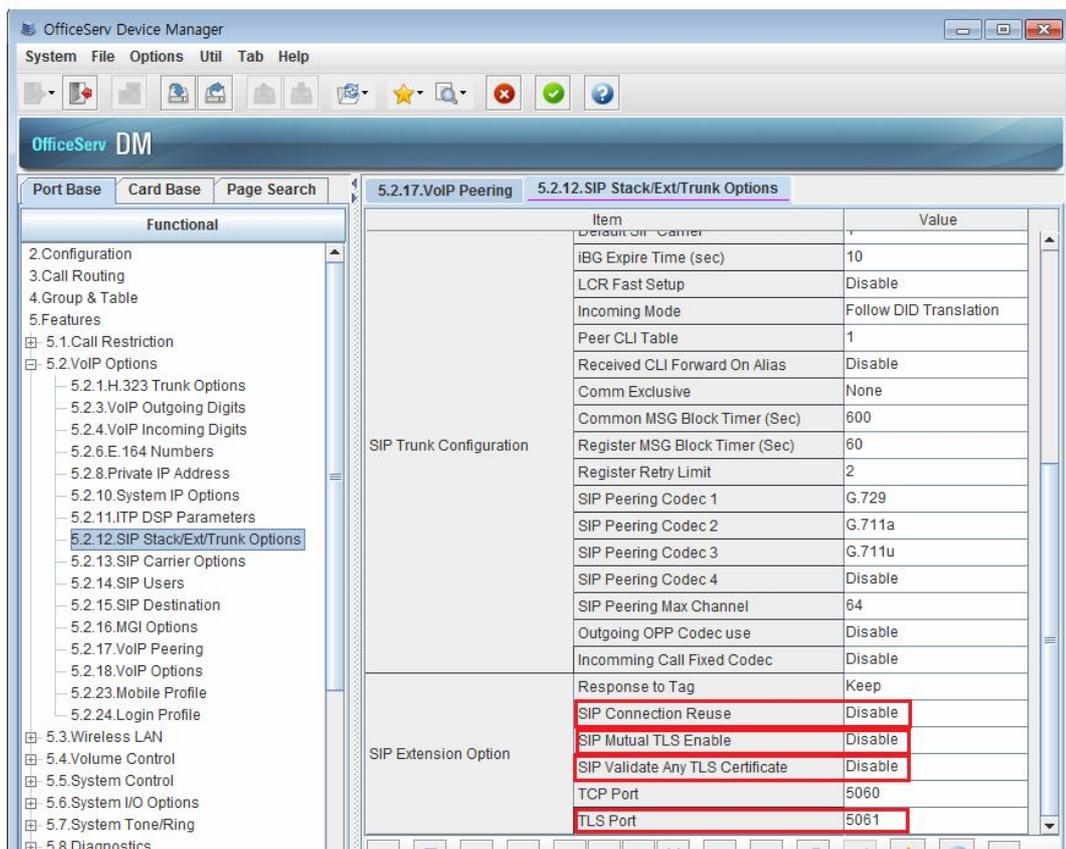
C. Using SIP TLS for SIP Station

- Configure SIP TLS related parameters in DM 5.2.12 SIP Stack/Ext/Trunk Options.
- TLS Port : This is the listening TLS port of OfficeServ system. Its default value is 5061.
- SIP Connection Reuse : This is only valid if the transport protocol with SIP station is TCP or TLS. If this parameter is set to ENABLE, OfficeServ system will reuse the TCP or TLS connection which has been established during SIP REGISTER. If this is set to DISABLE, OfficeServ system will make new TCP or TLS connection whenever making call. You can reduce the load by setting this parameter to ENABLE. But SIP station should support SIP

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Connection Reuse.

- SIP Mutual TLS Enable : If this is set to ENABLE, OfficeServ system will request certificate from SIP Peering or SIP Station.
- SIP Validate Any TLS Certificate : If this is set to DISABLE, OfficeServ system will abort TLS connection when the certificate from SIP Carrier or SIP Station is unknown certificate. You have to put the certificate of SIP Peering or SIP Station in SD card in OfficeServ system beforehand if this parameter is set to DISABLE. If this is set to ENABLE, OfficeServ system will omit the verification process for the certificate.



4.7.2 R7.2. Supporting sRTP

1) Description

- MGI16/64, OAS(MGI,RTG), some OSPP IP Phone, some WiFi Phone, SVMi-20i and UMS support sRTP media channel..
- SIP Trunk and SIP Station negotiate sRTP method and sRTP key by SIP signaling.
- SPNET Trunk negotiate sRTP method and sRTP key by SPNET signaling.
- Supported Encryption method is AES-CM mode and Supported Authentication method is HMAC-SHA1.



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- Master Key size is 128 bit.
- 2) Limitation
- Supported Device.

Device	AES-128	Note
MGI16/64, OAS(MGI,RTG)	Support	Old MGI doesn't Support sRTP
OSPP IP Phone	Support	SMT-i220x, SMT-i5210, SMT-i5220, SMT-i5230, SMT-i5240, SMT-i5243, SMT-i310x support sRTP
OSPP WiFi Phone	Support	SMT-w5120 support sRTP
SVMi-20i	Support	
IP-UMS	Support	
SIP Trunk, Station	Support	SIP Negotiation
SPNET	Support	SPNET Negotiation

- 3) Settings
- First of all, you must DM 2.1.5 system options's sRTP Algorithm for using sRTP. This is a system wide option.
 - After setting this option and you can selectively set each device's 'USE sRTP' option.



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OfficeServ Device Manager

System File Options Util Tab Help

OfficeServ DM

Port Base Card Base Page Search

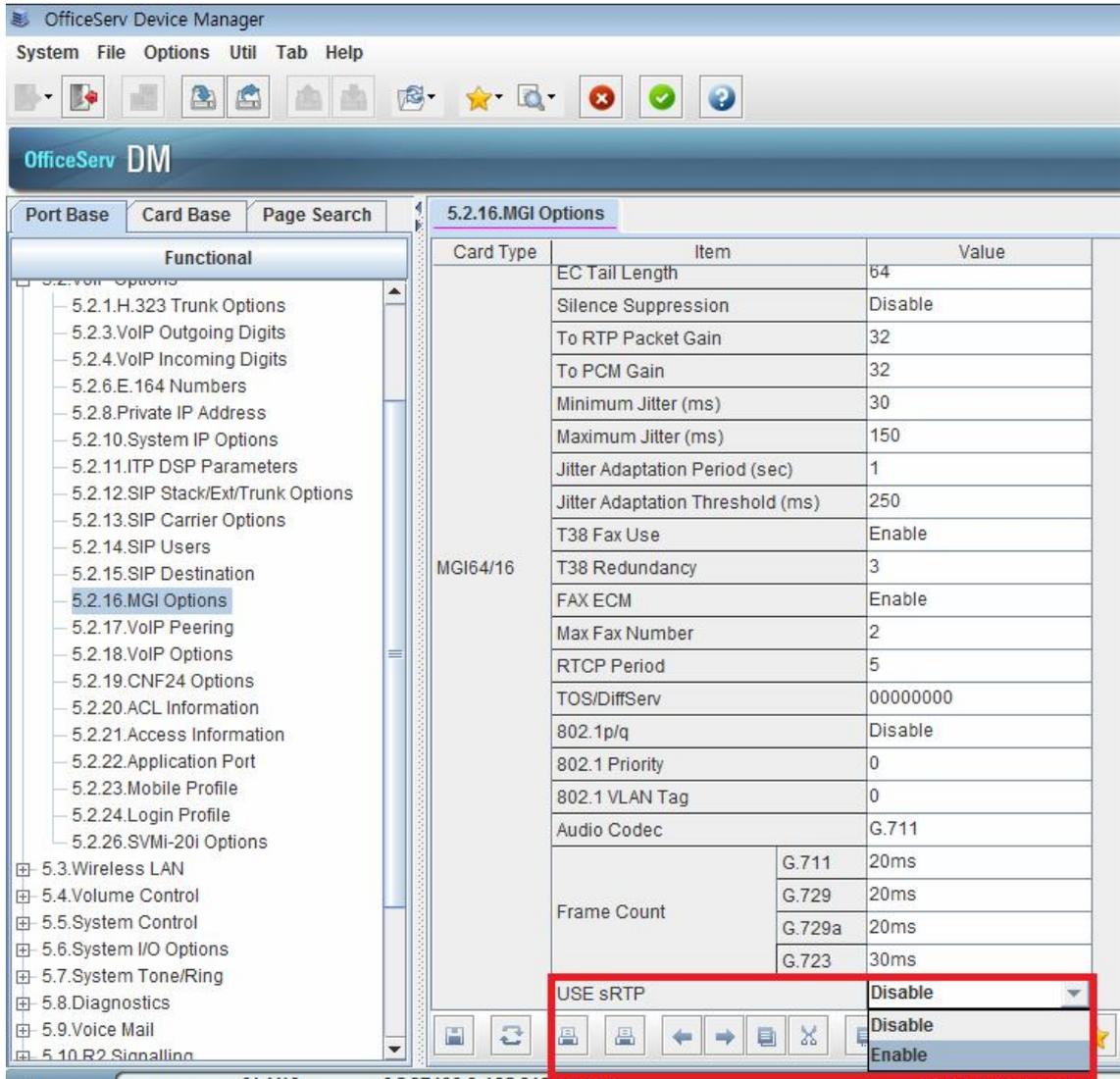
2.1.5. System Options

Item	Value	
Simultaneous Zone External Page	Disable	
Use Loud Bell For Page	Cabinet 1	None
	Cabinet 2	None
	Cabinet 3	None
Loop Trunk Tone Disconnect	Disable	
SLI2 PRS Use	Disable	
Maximum Chain Forward All Step	1	
Pickup Group Ring Service	Disable	
SYRIA R2/RD Select	Disable	
MOBEX Executive Option	Auth By CLI	Off
	Auth Hold Tone	Off
	Tone Source	TONE
	BLF by CLI	Off
	Answer Delay Time(sec)	1
Trunk To MOBEX CLI	ISDN	Received
	SIP	Received
VoIP RTP Option	DTMF Type	Outband
	MPS Service	On
	No MPS >> MGI	Off
	SIPT >> SIPT MGI Use	Off
	SIPT Ringback Message	180
sRTP Algorithm	AES_CM_128_HMAC_SHA1_80	
SVM Option	IP Service	Disable
		AES_CM_128_HMAC_SHA1_80
		ARIA_128_HMAC_SHA1_80
	ARIA_192_HMAC_SHA1_80	

Message [LAN] [OS7400 2-165.213.66.51] [*11]

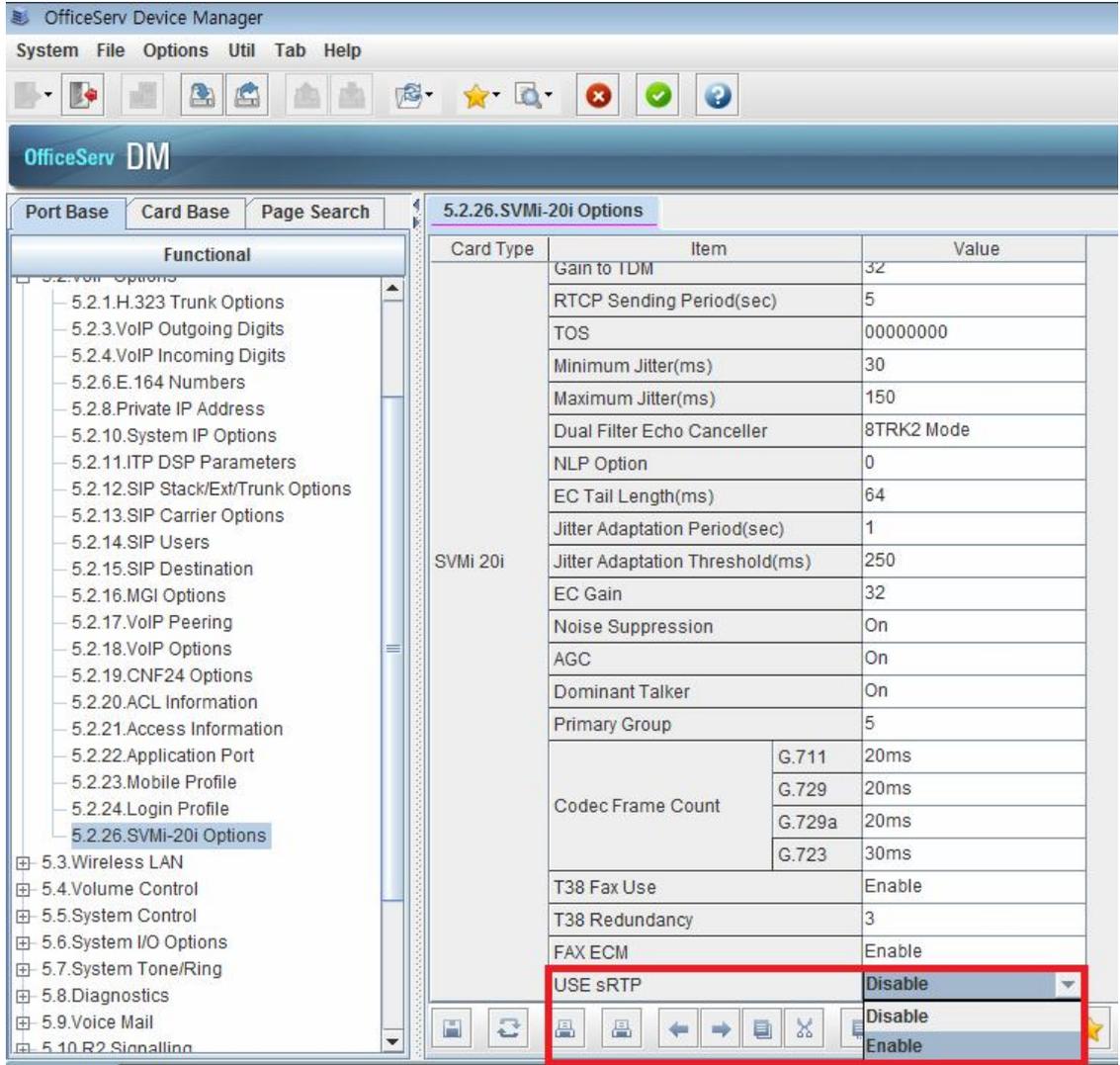
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- You must set DM 5.2.16 MGI Options' USE sRTP for using **MGI**'s sRTP
- This **MGI** means MGI16/64 and OAS(MGI,RTG)
- This option is for IP-UMS too.

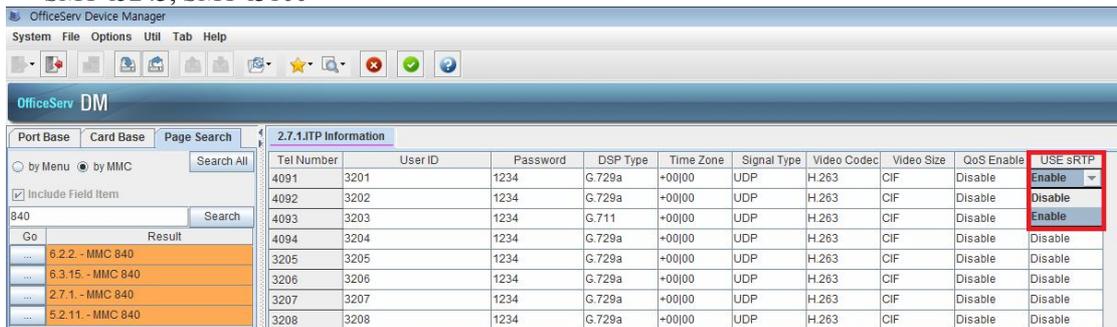


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- You must set DM 5.2.21 SVMi-20i Options' USE sRTP for using SVMi-20i's sRTP.



- You must set DM 2.7.1 ITP Information's USE sRTP for using OSPP IP Phone's sRTP
- Supported OSPP IP Phones : SMT-i2200, SMT-i5210, SMT-i5220, SMT-i5230, SMT-i5240, SMT-i5243, SMT-i3100





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- You must set DM 2.7.3 WIP Information's USE sRTP for using OSPP WiFi Phone's sRTP
- Supported OSPP WiFi Phones : SMT-w5120

Tel Number	User ID	Password	Insert Code	Handover Threshold	Handover Delta Value	Handover Scan Period	USE sRTP
3301	3301	0000		70	5	1	Disable
3302	3302	0000		70	5	1	Disable
3303	3303	0000		70	5	1	Enable
3304	3304	0000		70	5	1	Disable
3305	3305	0000		70	5	1	Disable
3306	3306	0000		70	5	1	Disable
3307	3307	0000		70	5	1	Disable
3308	3308	0000		70	5	1	Disable

- You must set DM 3.3.1 System Link ID's USE sRTP for using SPNET's sRTP

Entry No	System ID	IP Address	System Name	Time Sync	No MGI	Audio Codec	USE sRTP
Self	74	165.213.66.41					Disable
Sys01	75	165.213.66.51		Off	Off	G.729a	Disable
Sys02		0.0.0.0		Off	Off	G.729a	Enable
Sys03		0.0.0.0		Off	Off	G.729a	Disable
Sys04		0.0.0.0		Off	Off	G.729a	Disable
Sys05		0.0.0.0		Off	Off	G.729a	Disable

4.7.3 R7.3. Reinforcement on Security Vulnerability

1) Description

- Target system : OS7100(MP10a) / 7200(MP20s)
- The latest release version of Apache & PHP implementation

2) Limitation and Setting

- None

4.7.4 R7.4. Protection for SIP client registration

1) Description and Setting

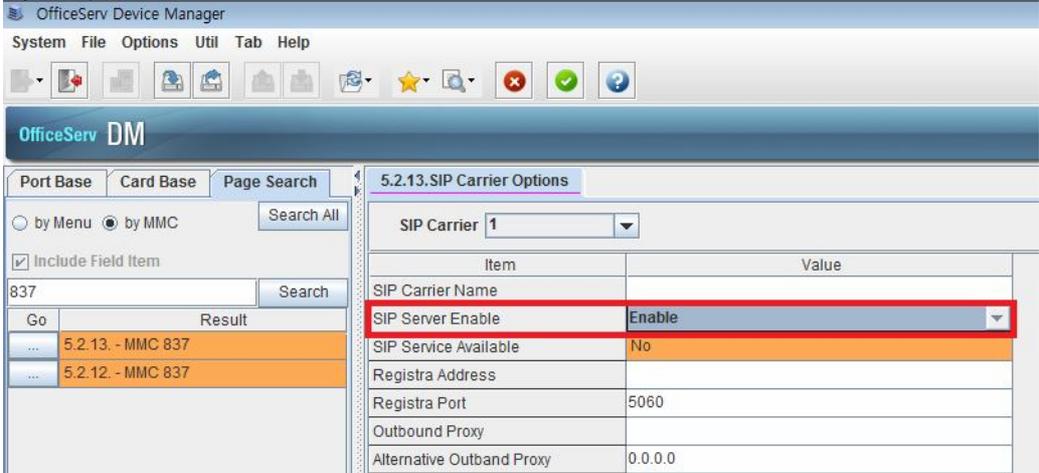
- Entering Password same as User ID and Tel Number is prohibited in DM 2.7.2(MMC 842).
 - Before: Possible to enter Password same as User ID and Tel Number.
 - V4.60: Possible to enter the User ID same as the Tel Number
Not possible to enter the Password same as User ID (Tel. Number)

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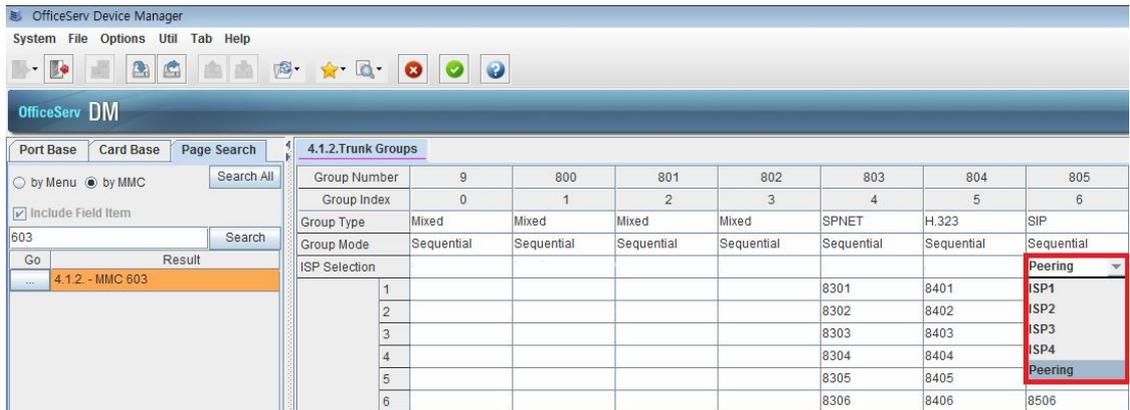
4.8 R8. SIP Trunking Enhancement

4.8.1 R8.1. Supporting Multi SIP carrier's SIP registration.

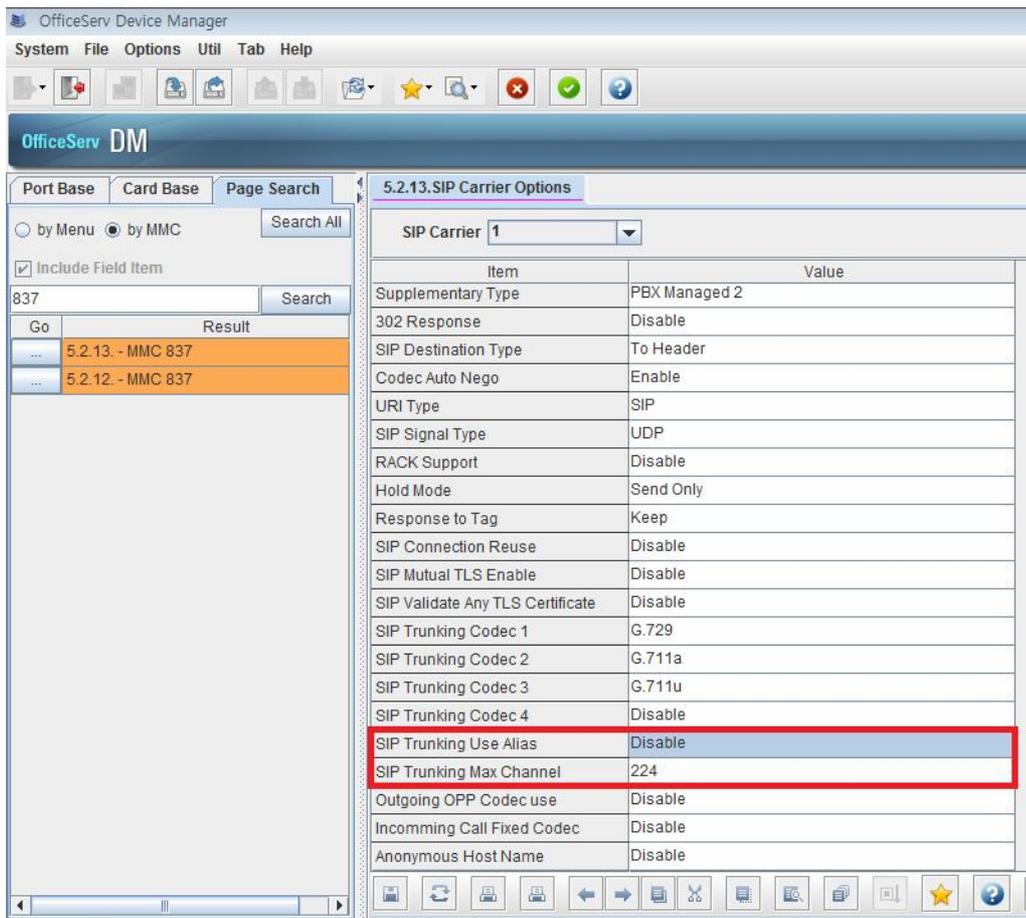
- 1) Description
 - Supporting 4 SIP carrier's SIP registration.
 - Selecting Carrier from dialing trunk group.
 - Sending each CLIP for each SIP Carrier.
 - Supporting auto re-routing from one carrier to another carrier by using LCR
- 2) Limitation
 - None
- 3) Settings
 - From V4.60 you can set all of 4 SIP carrier's SIP Server Enable option. (DM 5.2.13 SIP Carrier Options)



- **DM 4.1.2 Trunk Groups' ISP Selection** is added for selecting SIP trunking (ISP 1 ~ 4) or SIP Peering
- For using SIP Trunking or SIP Peering, you must set DM 4.1.2 Trunk Groups' ISP Selection.
- If you upgrade from bellow V4.60 to V4.60, you must set this option. Otherwise you cannot use SIP Trunking or SIP Peering.



- For each SIP carrier, DM 5.2.13 SIP Carrier Options' SIP Trunking Max Channel is added. Each SIP carrier can use SIP channel until reaching this number.
- For each SIP carrier, DM 5.2.13 SIP Carrier Options' SIP Trunking Use Alias is added. You can select to send SIP Alias or not.



* Example for SIP Multi Carrier.

- *SIP Trunk outgoing case*



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- How to select SIP Peering or ISP 1 ~ 4
 - ◆ In case of accessing SIP Trunk group number
 - Following ISP Selection of SIP Trunk Group
 - ◆ In case of using LCR code
 - Following ISP Selection of LCR Route Table's Trunk Group number
 - ◆ In case of accessing SIP Trunk number directly.
 - Finding out SIP Trunk Group including self trunk port
 - And following the SIP Trunk group's ISP selection
 - ◆ In case of accessing MIXED Trunk group's SIP Trunk
 - Finding out available SIP Trunk Port
 - Finding out SIP Trunk Group including self trunk port
 - And following the SIP Trunk group's ISP selection
 - ◆ By using STN – TRK Use Table and Mixed Trunk Group, you can select different ISP or Peering while dialing one access code for different station.

- *SIP Trunk incoming case*
 - Refer to 'R13.7 Separation SIP Trunking / Peering Port'

*** Call scenario and how to set related DBs**

- *How to Select different ISP from different Trunk Access Code*
[scenario example] -----
 - **Company A - Station 2001**
 - ◆ **Lacal/LD call(0*..) : ISP1**
 - ◆ **International call (00*..) : ISP2's Account a**
 - **Company B – Station 3001**
 - ◆ **Lacal/LD call(0*..) : ISP3**
 - ◆ **International call (00*..) : ISP2's Account b**

- **Company A's local outgoing call : 9 + 031xxxxxxx -> select ISP1**
- **Company A's international outgoing call : 9 + 001xxxxxxxxxxx -> select ISP2**
- **Company B's local outgoing call : 8 + 031xxxxxxx -> select ISP3**
- **Company B's international outgoing call : 8 + 001xxxxxxxxxxx -> select ISP4**

-
- **DM 5.1.12 : ISP1 : SIP SERVER - ENABLE, MAX CH NO. - 4**
ISP2 : SIP SERVER - ENABLE, MAX CH NO. - 4



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ISP3 : SIP SERVER - ENABLE, MAX CH NO. - 4

ISP4 : SIP SERVER - ENABLE, MAX CH NO. - 4

(Same Carrier as ISP2 and different Account (Account b))

- DM 4.1.2 : 805 - 8501, 8502, 8503, 8504 - ISP1
806 - 8505, 8506, 8507, 8508 - ISP2
807 - 8509, 8510, 8511, 8512 - ISP3
808 - 8513, 8514, 8515, 8516 - ISP4
- DM 3.1.1 : LCR ENABLE
- DM 2.8.0 : LCR : 9 (Company A's trunk access code)
E-LCR : 8 (Company B's trunk access code)
- DM 3.1.6 : 0* -> 990* (Adding 99 arbitrarily for using different LCR)
- DM 3.1.2 : 00 - RT 2
01~09 - RT 1
9900 - RT 4
9901~9909 - RT 3
- DM 3.1.4 : RT01 : C1 : G 805
C2 : G 806 (for LCR Re-routing)
RT02 : C1 : G 806
C2 : G 805 (for LCR Re-routing)
RT03 : C1 : G 807 M 001
C2 : G 808 (for LCR Re-routing) M 001
RT04 : C1 : G 808 M 001
C2 : G 807 (for LCR Re-routing) M 001
- DM 3.1.5 : 001 : DEL DGT : 02 (for deleting 99)

- *How to Select different ISP from same Trunk Access Code by using STN-TRK USE table*
[scenario example] -----

- **Company A - Station 2001**
 - ◆ Local/LD call(0*..) : ISP1
 - ◆ International call (00*..) ISP2's Account a
- **Company B - Station 3001**
 - ◆ Local/LD call(0*..) : ISP3
 - ◆ International call (00*..) ISP2's Account b
- **Company A's local outgoing call : 9 + 031xxxxxxx -> select ISP1**



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- **Company A's international outgoing call : 9 + 001xxxxxxxxxx -> select ISP2**
 - **Company B's local outgoing call : 9 + 031xxxxxxxx -> select ISP3**
 - **Company B's international outgoing call : 9 + 001xxxxxxxxxx -> select ISP4**
-

- DM 5.1.12: ISP1 : SIP SERVER - ENABLE, MAX CH NO. - 4
ISP2 : SIP SERVER - ENABLE, MAX CH NO. - 4
ISP3 : SIP SERVER - ENABLE, MAX CH NO. - 4
ISP4 : SIP SERVER - ENABLE, MAX CH NO. - 4
(Same Carrier as ISP2 and different Account)
- DM 4.1.2: 805 - 8501, 8502, 8503, 8504 - ISP1
806 - 8505, 8506, 8507, 8508 - ISP2
807 - 8509, 8510, 8511, 8512 - ISP3
808 - 8513, 8514, 8515, 8516 - ISP4
809 - 8501, 8502, 8503, 8504, 8509, 8510, 8511, 8512 – Mixed Trunk
810 - 8505, 8506, 8507, 8508, 8513, 8514, 8515, 8516 – Mixed Trunk
- DM 5.1.1 : STN GRP 2001(Company A) : 001
STN GRP 3001(Company B) : 002
TRK GRP 8501~8508 : 301
TRK GRP 8509~8516 : 302
- DM 5.1.2 : 001 – 301 : DIAL : YES
001 – 302 : DIAL : NO
002 – 301 : DIAL : NO
002 – 302 : DIAL : YES
- DM 3.1.1 : LCR ENABLE
- DM 2.8.0 : LCR : 9
- DM 3.1.2 : 00 - RT 2
01~09 - RT 1
- DM 3.1.4 : RT01 : C1 : G 809
C2 : G 810 (for LCR Re-routing)
RT02 : C1 : G 810
C2 : G 809 (for LCR Re-routing)



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4.8.2 R8.2. Enhanced SIP display message.

- SIP standard cause value which comes from SIP carrier will be displayed on user's phone in case of SIP trunk call.
- This is applied only for USA.

4.8.3 R8.3. Voice Band Data (VBD) feature

1) Description

Voice Band Data supports fax service without NLP and Jitter Buffer Operation. Its operation will be run when Fax Tone (CNG, CED, ANSam etc) is detected and only supports VoIP Trunk calls. If the voice codec is G.729, voice codec is changed into G.711 through codec negotiation and then voice band operation will be made.

2) Limitation

VBD feature will be run completely when two systems must be configured by VBD. Its feature is the following version

- OAS – v2.03
- MGI16/64 – v1.28
- OS7100/OS7030/OS7070 – v4.60

3) Setting

“T38 Use Enable” is changed into “Fax Option”. Fax Option has three lists.

- PassThrough – Fax Service through G.711 alaw/ulaw
- T.38 – Fax Service through T.38
- VBD. – Fax Service through VBD operation.



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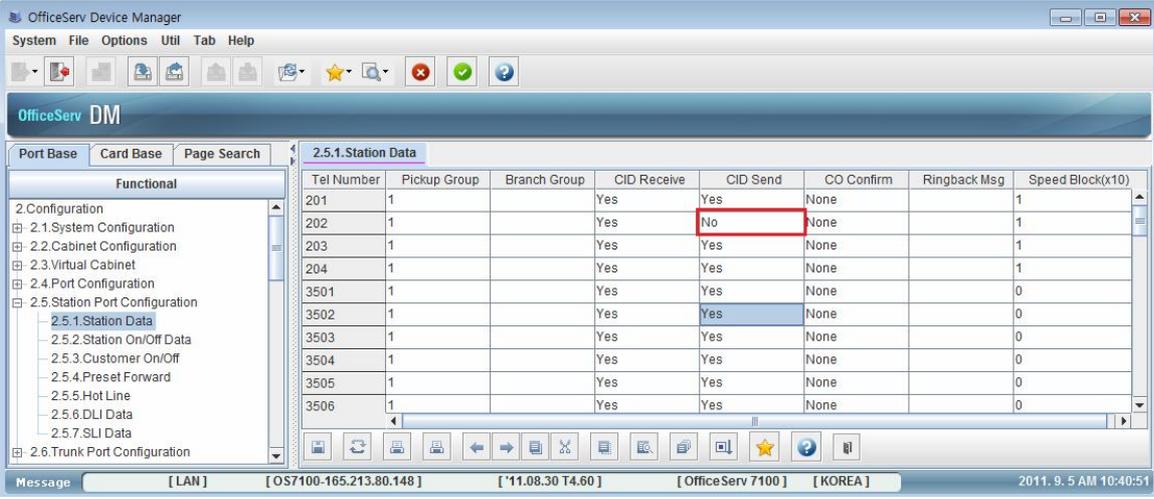
5.2.16.MGI Options		
Card Type	Item	Value
MGI64/16	NLP	0
	EC Gain	32
	EC Tail Length	64
	Silence Suppression	Disable
	To RTP Packet Gain	32
	To PCM Gain	32
	Minimum Jitter (ms)	30
	Maximum Jitter (ms)	150
	Jitter Adaptation Period (sec)	1
	Jitter Adaptation Threshold (ms)	250
	Fax Option	VBD
	T38 Redundancy	3
	FAX ECM	Enable
	Max Fax Number	2
	RTCP Period	5
	TOS/DiffServ	00000000
	802.1p/q	Disable
	802.1 Priority	0
	802.1 VLAN Tag	0
	Audio Codec	G.711
	G.711	20ms

4.8.4 R8.4. Privacy header option for outgoing SIP trunks call

- 1) Description
This feature enables SIP Trunk outgoing call to hide calling line number.
- 2) Description
None
- 3) Description
– Configure “CID Send” parameter for the specified station to “No”.



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- OfficeServ system send INVITE message like below when making SIP trunk outgoing call in the station. OfficeServ system put the anonymous in “From” header and calling line number in “P-Asserted-Identity” header. It also has “Privacy” header with “id;critical”.

```

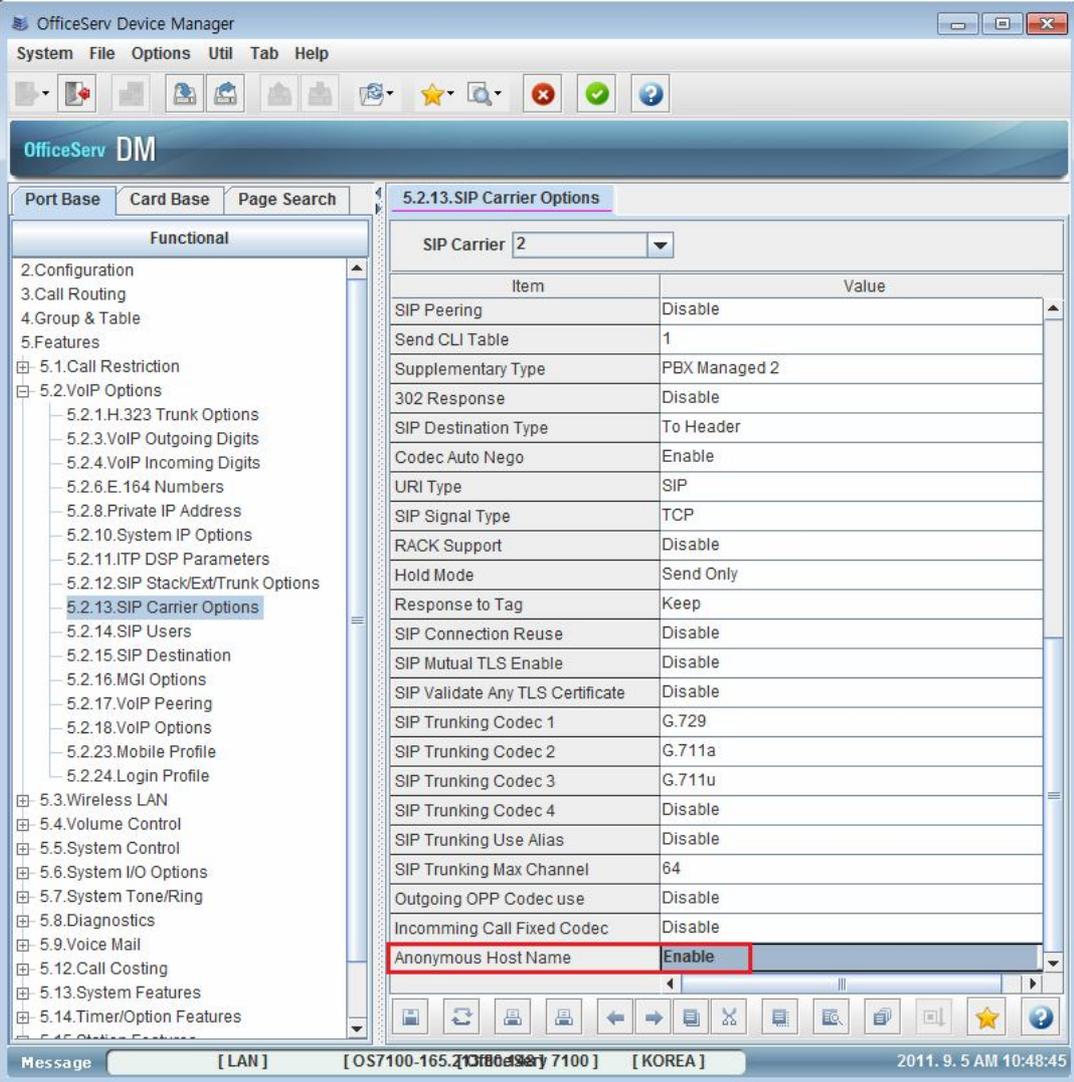
INVITE sip:2009@165.213.80.231 SIP/2.0\r\n
From: <sip:anonymous@165.213.80.231>; tag=1836a08-9450d5a5-13c4-50017-4ec0c739-158a6790-4ec0c739\r\n
To: <sip:2009@165.213.80.231>\r\n
Call-ID: 187e5f0-9450d5a5-13c4-50017-4ec0c739-14167ed2-4ec0c739\r\n
Cseq: 1 INVITE\r\n
Via: SIP/2.0/UDP 165.213.80.148:5060;rport;branch=z9hg4bk-4ec0c739-a10a37de-21bf37a2\r\n
P-Asserted-Identity: "2002" <sip:2002@165.213.80.231:5060>\r\n
Privacy: id;critical\r\n
Max-Forwards: 70\r\n
User-Agent: Samsung OfficeServ\r\n
Contact: <sip:anonymous@165.213.80.148:5060;transport=udp>\r\n
Allow: REGISTER, INVITE, ACK, BYE, REFER, NOTIFY, CANCEL, INFO, OPTIONS, PRACK, SUBSCRIBE, UPDATE\r\n
Content-Type: application/sdp\r\n
Content-Length: 293\r\n

```

- If you want to hide even hostname, you can set Anonymous Host Name in DM5.2.13 SIP Carrier Options to ENABLE.



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- INVITE message which is sent by OfficeServ system is like this.

```

INVITE sip:2009@165.213.80.231 SIP/2.0\r\n
From: <sip:anonymous@anonymous.invalid>; tag=18366c0-9450d5a5-13c4-50017-4ec0cd55-36d3dbbd-4ec0cd55\r\n
To: <sip:2009@165.213.80.231>\r\n
Call-ID: 187e5f0-9450d5a5-13c4-50017-4ec0cd55-a351a2f-4ec0cd55\r\n
CSeq: 1 INVITE\r\n
Via: SIP/2.0/UDP 165.213.80.148:5060;rport; branch=z9hg4bk-4ec0cd55-a1221426-773964cb\r\n
P-Asserted-Identity: "2002" <sip:2002@165.213.80.231:5060>\r\n
Privacy: id;critical\r\n
Max-Forwards: 70\r\n
User-Agent: Samsung OfficeServ\r\n
Contact: <sip:anonymous@165.213.80.148:5060;transport=udp>\r\n
Allow: REGISTER,INVITE,ACK,BYE,REFER,NOTIFY,CANCEL,INFO,OPTIONS,PRACK,SUBSCRIBE,UPDATE\r\n
Content-Type: application/sdp\r\n
Content-Length: 293\r\n

```



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4.8.5 R8.5. VoIP Tandem between SIP trunks

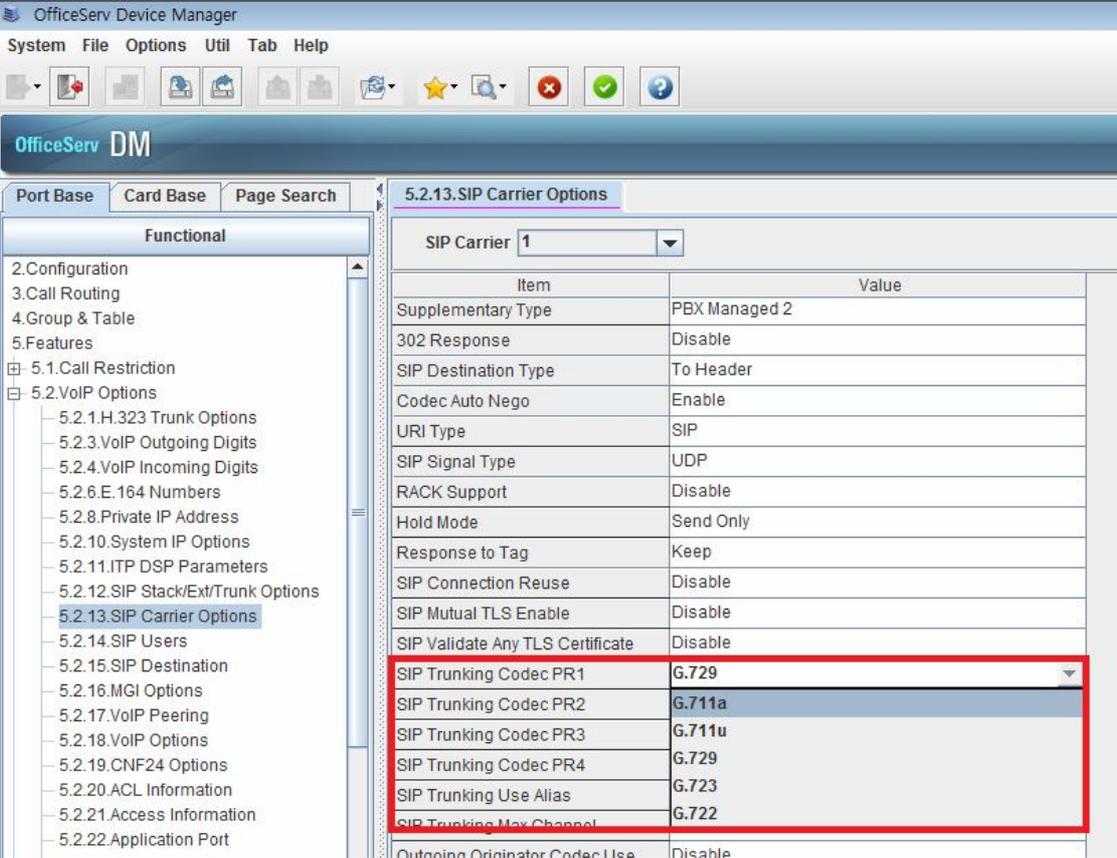
- 1) Description
 - OfficeServ provides VoIP Tandem option to allow tandem between SIP Peering/H.323 trunk.
- 2) Limitation
 - Only SIP peering or H.323 trunk can use this option and the other trunks are not affected.
- 3) Settings
 - Select peering table that you want to restrict tandem. And disable VoIP Tandem option of that peering table. (default: Enable)

4.8.6 R8.6. Selectable codec for SIP trunk

- 1) Description
 - Supporting 4 codecs for SIP trunk.
 - You can differently select 4 codecs for each SIP peering and SIP trunking (each SIP carrier)
- 2) Limitation
 - None
- 3) Settings
 - You can select 4 different codecs per each carrier and SIP peering.
 - You can differently select DM 5.2.13 SIP Carrier Options' Codec PR1~ Codec PR4.



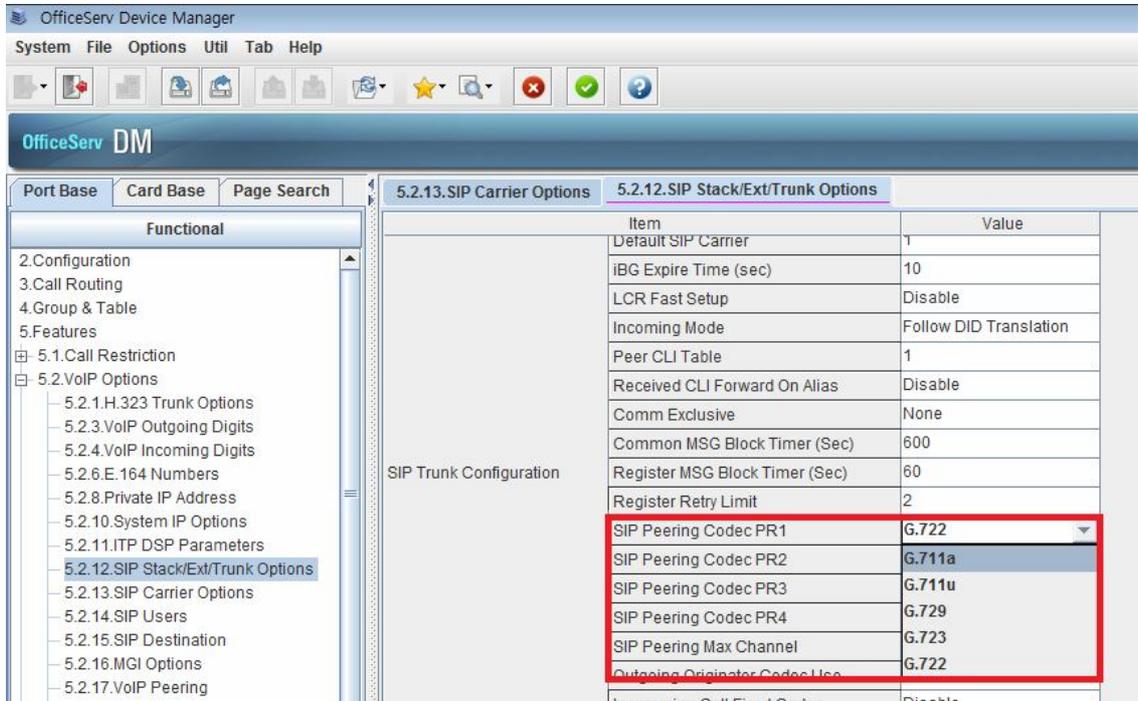
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- Selectable Options
 - A. CODEC PR1 : G.729, G.711A, G.711U, G.723, G722
 - B. CODEC PR2~CODEC PR4 : DISABLE, G.729, G.711A, G.711U, G.723, G722
 - C. Default value : CODEC PR1 (G.729), CODEC PR2 (G.711a), CODEC PR3 (G.711u), CODEC PR4(DISABLE)

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- You can differently select DM 5.2.12 SIP Stack/Ext/Trunk Options ' Codec PR1~ Codec PR4.



- Selectable Options

- D. CODEC PR1 : G.729, G.711A, G.711U, G.723, G722
- E. CODEC PR2~CODEC PR4 : DISABLE, G.729, G.711A, G.711U, G.723, G722
- F. Default value : CODEC PR1 (G.729), CODEC PR2 (G.711a), CODEC PR3 (G.711u), CODEC PR4(DISABLE)

- How to select codecs when making a outgoing SIP trunk call (SIP Invite message)

- G. Basically selecting DM 5.2.13 SIP Carrier Options, DM 5.2.12 SIP Stack/Ext/Trunk Options' Codec PR1 ~ Codec PR4
- H. Deleting duplicated codecs.
- I. Finding out codecs supported by caller device
- J. If you set Outgoing Originator Codec Use to Enable, OfficeServ selects caller device's codec as top priority (same as before V4.60).

- How to select codec when answering a incoming SIP trunk call (SIP 200OK message)

- K. Basically selecting answered device's codec (same as before V4.60).
- L. If you set Incoming Call Fixed Codec, OfficeServ selects available codec among DM 5.2.13 SIP Carrier Options, DM 5.2.12 SIP Stack/Ext/Trunk Options' Codec PR1 ~ Codec PR4 to use by



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answered device

OfficeServ Device Manager

System File Options Util Tab Help

OfficeServ DM

Port Base Card Base Page Search

5.2.12.SIP Stack/Ext/Trunk Options 5.2.13.SIP Carrier Options

SIP Carrier 1

Item	Value
Supplementary Type	PBX Managed 2
302 Response	Disable
SIP Destination Type	To Header
Codec Auto Nego	Enable
URI Type	SIP
SIP Signal Type	UDP
RACK Support	Disable
Hold Mode	Send Only
Response to Tag	Keep
SIP Connection Reuse	Disable
SIP Mutual TLS Enable	Disable
SIP Validate Any TLS Certificate	Disable
SIP Trunking Codec PR1	G.729
SIP Trunking Codec PR2	G.711a
SIP Trunking Codec PR3	G.711u
SIP Trunking Codec PR4	Disable
SIP Trunking Use Alias	Disable
SIP Trunking Max Channel	224
Outgoing Originator Codec Use	Disable
Incomming Call Fixed Codec	Disable
Anonymous Host Name	Disable



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4.9 R9. Mobex Enhancement

4.9.1 R9.1. Call back option for MOBEX calls

1) Description

- MOBEX user requests to system to call back and this feature is good for saving charge.
- MOBEX user dials MOBEX arrival number. (MOBEX CLI certification is needed.)
- Then current call is disconnected automatically
- After mobile callback time (default: 5 seconds), system makes a call to MOBEX phone for MOBEX callback.
- MOBEX user answers callback call and then hears dial tone.
- If callback is failed, retry routine is started. User can change retry count in DM.

2) Settings

- Assign MOBEX Executive License or Remote dial License to specific MOBEX station.
- Set Auth By CLI to ON in DM 2.1.5.
- Set Callback to ON in DM 2.7.5.
- Set Mobile Callback Retry Count (default: 0) and Mobile Callback Time (default: 5) in DM 5.14.3.

4.9.2 R9.2. MOBEX Scheduling

1) Description

- OfficeServ always provided simultaneous ring service for MOBEX station and its master before V4.60. But now this service is provided only in specific time and this time can be changeable.

2) Settings

- Make ring group and assign MOBEX station as its member.
- Set scheduling time in DM 4.10.1 ([MMC129](#)).

If there is no ring group that has MOBEX station as its member, both DM 4.10.1 and MMC129 will not be opened.

4.9.3 R9.3. MOBEX BUSY

1) Description

- If MOBEX station is busy and its master should be also busy. In case of station pair it is OK but in case of ring group it was not OK before V4.60. Now OfficeServ provides ring group busy features.
- In this case ring group master and MOBEX master should be same and MOBEX station should



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be assigned as ring group member.

2) Settings

- Make ring group (master: MOBEX master, member: MOBEX station)
- Set MOBEX BUSY option. If it is ON, MOBEX master will be also busy if MOBEX station is busy.

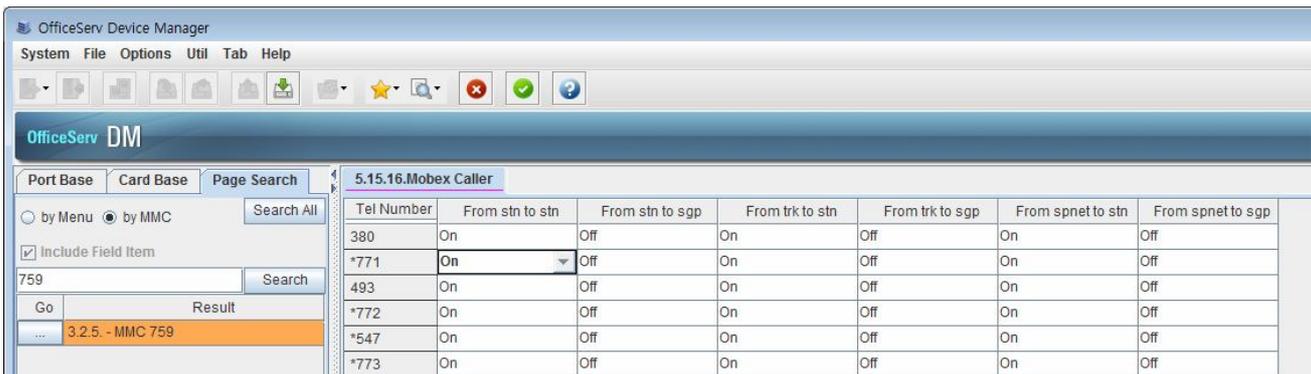
4.9.4 R9.4. Type selection of call which can ring MOBEX

1) Description

- You can have ability to set what type of calls can ring MOBEX station..

2) Settings

- Set On/Off which types of call that user wants to receive to his/her MOBEX station.



No	Option	Description
0	STN->STN	If this option is set to OFF and station user makes a call to MOBEX master, MOBEX station does not ring.
1	STN->SGP	If this option is set to OFF and station user makes a call to station group which has MOBEX master as its group member, MOBEX station does not ring.
2	TRK->STN	If this option is set to OFF and external user makes a call to MOBEX master, MOBEX station does not ring.
3	TRK->SGP	If this option is set to OFF and external user makes a call to station group which has MOBEX master as its group member, MOBEX station does not ring.
4	NET->STN	If this option is set to OFF and SPNet user makes a call to MOBEX master, MOBEX station does not ring.
5	NET->SGP	If this option is set to OFF and SPNet user makes a call to station



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	group which has MOBEX master as its group member, MOBEX station does not ring.
--	--

4.9.5 R9.5. CLI Ringing for MOBEX

1) Description

- Mobex authorization by CLI Ringing table is implemented..

2) Settings

- Set calling party number which can be used for MobEx authorization in DM 3.2.5 (MMC759).
- Enable MOBEX code as destination in DM 3.2.5.

4.9.6 R9.6. MOBEX feature code display

1) Description

- Currently, when dial MOBEX feature code from idle station display reads:

make a selection ON OFF

- We replace “make a selection” with “MOBEX” and also to display which item is currently active. Possible displays are:

MOBEX ON OFF

2) Settings

- In MMC724, set Mobex feature code.
- Press the feature code and check the phone display.



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4.10 R10. 3rd-Party Solution

4.10.1 Functional Requirements

- 1) R10.1 Supporting H.264 to Video SIP Phone
- 2) R10.2 Sending DTMF to SIP Phone
 - When a station or a trunk calls to SIP Phone, system can send DTMF to SIP Phone.
- 3) R10.3. IOT with BroadWorks for SIP
 - We are proceeding the Vendor Validation Process with BroadSoft.
 - All required data is submitted to BroadSoft and waiting for their result and certificate.

4.10.2 Limitation

- 1) R10.2 Sending DTMF to SIP Phone
 - In case of using MPS or not using Media resource, system doesn't follow SIP Phone's DTMF option and follow opposite device's sending DTMF.

4.10.3 Settings

- 1) R10.1 Supporting H.264 to Video SIP Phone
 - OfficeServ system can make a video call to video SIP Phone or answer it from video SIP Phone through H.264.
 - OfficeServ system can negotiate video Codec by SIP signaling
 - Available devices can be video SIP Phone, video OSPP IPP Phone, SIP Trunk, SPNET Trunk.
- 2) R10.2 Sending DTMF to SIP Phone
 - You can select DTMF type when system sends to SIP Phone. (DM 2.7.2 SIP Phone Information)





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4.11 R11. CNF24 phase 2

In phase 2, the existing CNF24 features are enhanced and OfficeServ provides some new features to accept regional requirements such as below.

- 1) The number of prompt language is increased so OfficeServ should provide appropriate prompt and template.
- 2) OfficeServ should provide an easy routine for reserving, attending and checking Meet-Me conference.

Following descriptions let you know the detailed service of CNF24 phase 2 and how to use these features.

4.11.1 R11.1 Outlook 2010 Interworking

1) Description

- CNF24 attaches a reservation schedule when sending an invitation e-mail for Meet-Me conference. If participant clicks this attached file, Outlook2010 schedule will be opened automatically and shows a reservation schedule on its format.

3) Setup

- There is no additional setup so that a reservation schedule is attached on an invitation e-mail every time CNF24 sends an invitation e-mail. And it is saved as “ics” format.

4.11.2 R11.2 Retry routine for entering conference ID and password

1) Description

- OfficeServ provides a retry opportunity when wrong conference ID or password is entered.
- In this case participant can hear announcement for retry. (xx14.snd and xx15.snd prompts are newly added.)
- OfficeServ provides 3 times of retry. If all retries are failed, participant can hear error announcement and the current procedure is ended.

2) Setup

- There is no additional setup.

4.11.3 R11.3 Prompt Language

1) Description

- CNF24 provides the number of prompt language from 3 to 17.

A.	Selectable	Prompt	Language
	English (USA), Korean,	English (UK),	English (Australia)



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German, Greek, Italian, Russian, Spanish, Turkish, Finnish, French, Dutch, Danish, Portuguese, Swedish, Norwegian

2) Limitation

- Even though the kind of language is increased, CNF24 card support only specific prompts which come in within period of V4.6. So USA, UK, Korea, Italy, Russia and Spain have its own language prompts and the other countries use UK prompt basically.

3) Setup

- There is no additional setup. Because system automatically sets prompt language by following its country. (ex. If country is set to Russia, prompt language is set to Russian automatically.)
- If you want to change current prompt language, select another prompt language in DM 9.1.1.

9.1.1.Conference Options	
Item	Conference Options
Password Option	Off
Greeting Option	Off
Whoami Option	Off
SPA Option	Off
Join Alarm Option	Off
Leave Alarm Option	Off
End Alarm Option	Off
Early Ent Time	0
Mail Server Option	Off
Mail Max Retry	3
Mail Retry Interval	5
System Time Zone (GMT)	+09 00 Seoul
Max Rec Time (min)	300
Mail Server Port	
Local Domain	
Mail Server User ID	
Mail Server Password	
Mail Server Domain/IP	
DNS IP	0.0.0.0
Record Alarm Capacity	70
Record Delete Capacity	90
Kick Out Option	Keep
Prompt Language	English(USA) ▼



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- In DM 9.1.7, you can check 16 CNF24 prompts follow current prompt language.
For reference, first two digits of file name represent current prompt language. (ex. 00 is for English(USA))
In this menu, you can upload another prompt that you want to use.

9.1.7.CNF24 Voice Management				
Card	C1-S8	Language Set	English(USA)	
No	Comments		File	
0	Meet-Me Conference Id Request	0000.snd		...
1	Meet-Me Conference Password Request	0001.snd		...
2	Meet-Me Conference Greeting	0002.snd		...
3	Meet-Me Conference SPA Music	0003.snd		...
4	Meet-Me Conference Who am I Request	0004.snd		...
5	Meet-Me Conference Join Alarm Tone	0005.snd		...
6	Meet-Me Conference Leave Alarm Tone	0006.snd		...
7	Meet-Me Conference End Alarm Tone	0007.snd		...
8	Meet-Me Conference : Conf Id Error	0008.snd		...
9	Meet-Me Conference : Password Error	0009.snd		...
10	Meet-Me Conference : Conf Lock	0010.snd		...
11	Meet-Me Conference : Conf Mbr Full	0011.snd		...
12	Meet-Me Conference : Can't Join	0012.snd		...
13	Meet-Me Conference : Abnormal Error	0013.snd		...
14	Meet-Me Conference Id Retry Request	0014.snd		...
15	Meet-Me Conference Pswd Retry Request	0015.snd		...

4.11.4 R11.4 Email Template

1) Description

- When sending an invitation e-mail, appropriate email template is displayed by following selected prompt language.

2) Setup

- There is no additional setup.



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4.11.5 R11.5 Time Zone

1) Description

- Time zone is applied by following Country and its capital. So it can be changed in case of system initialization and change of country.

2) Setup

- User can check current time zone information in DM 9.1.1 and change its value.
- As below picture, time zone is shown on the reservation form. It can be also shown on email template.

Conference Reservation Form

Card: C1-S8

Date: 2011. 9. 1

Subject: [Empty]

Conference ID: [Empty] [Check]

Duration: AM 05:00 ~ AM 05:45

Repeat: Once Daily Weekly

of Attendees: [Empty] Available Channel: 21

Owner Station: [Empty] [Select] GMT +09:00

Attendee List: [Select...]

Send Invitation Email

Password: [Empty]

Options: Early Entrance Recording Paging [Empty]

Greeting Prompt: [Empty] [Upload]

[Submit] [Cancel]

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4.11.6 R11.6 Kick out options

1) Description

- OfficeServ provides Kick out option as “Keep” and “Clear”. (default: Keep)
- If “Keep” is selected, participant who was kicked out of the current conference can’t re-join the conference because its user information is still remained.
- If “Clear” is selected, participant who was kicked out of the current conference can re-join the conference because its user information is cleared.

2) Setup

- User can change this option in DM 9.1.1.

4.11.7 R11.7 Station number searching when reserving conference

1) Description

- When assigning conference participants, user can get information by searching station number on PWP.

2) Setup

- There is no additional setup.

4.11.8 R11.8 Conference Instructions

1) Description

- Before V4.60 conference instructions showed contact number list but now it will show text description that user enters in DM. These instructions will be displayed on invitation e-mail.

2) Setup

- User can enter conference instructions as text format in DM 9.1.8

4.11.9 R11.9 Sender e-mail address

1) Description

- At the end of reserving conference, user can check Meet-Me conference invitation e-mail for confirming. In this step, user can change sender e-mail address as he/she wants. If this information is changed, modified address will be applied to system DB automatically.

2) Setup

- User can change sender e-mail address on Conference Invitation e-mail at confirming step.
- Modified address can be checked in DM 9.1.6.



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Meet Me Conference Invitation X

Conference Subject : 000

Email Subject :

Sender Email Addr... : master@samsung.com

CONFERENCE INVITATION MAIL

SUBJECT : 000
DATE : 2012/Mar/12
TIME : 00:15-01:00 (GMT+09:00)
PASSWORD : NO

CONFERENCE MEMBERS (TOTAL:3) #####
CONF-ID: 000 01 MEMBER: bob (bob@samsung.com)

ADDITIONAL INFORMATION #####
Please join the conference on time.

Name	Phone	Email
bob	01095301234	bob@samsung.com

4.11.10 R11.10 CNF24 port status

- 1) Description
 - Use can check CNF24 port status.
 - Kind of status is as below
 - A. NONE: There is network setting problem.
 - B. IDLE: Card is activated and ready to service.
 - C. CONF: Current port is used as conference channel.



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2) Setup

- User can check CNF24 port status in DM 6.2.9.

6.2.9.CNF24 Status

Cabinet/Slot	Index	Status	OPP			Codec
			Tel Number	IP Address	RTP Port	
C1-S8	1	IDLE				G.711
	2	IDLE				G.711
	3	IDLE				G.711
	4	IDLE				G.711
	5	IDLE				G.711
	6	IDLE				G.711
	7	IDLE				G.711
	8	IDLE				G.711
	9	IDLE				G.711
	10	IDLE				G.711
	11	IDLE				G.711
	12	IDLE				G.711
	13	IDLE				G.711
	14	IDLE				G.711
	15	IDLE				G.711
	16	IDLE				G.711
	17	IDLE				G.711
	18	IDLE				G.711
	19	IDLE				G.711
	20	IDLE				G.711
	21	IDLE				G.711
	22	IDLE				G.711
	23	IDLE				G.711
	24	IDLE				G.711

4.11.11 R11.11 Daylight-saving time

1) Description

- Daylight-saving time is used to calculate e-mail sending time. Before V4.60, it is applied by following CNF24 card option (Mail Day Saving Time) but now it will be applied by following system summer time..
- Previous “Mail Day Saving Time” option is deleted from DM 9.1.1.



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2) Setup

- There is no additional setup because a daylight-saving time of CNF24 follows system value.

4.11.12 R11.12 Daily reserving period

1) Description

- When selecting repeat option as Daily on PWP, user can set reserving period by max 3 months.

2) Setup

- There is no additional setup.

4.11.13 R11.13 Station e-mail address

1) Description

- User can input email address of each station not only in DM but also in KMMC.

2) Setup

- DM 9.1.6 or MMC127 STN EMAIL



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4.12 R12. OfficeServ NMS V1.61

4.12.1 Functional Requirements

- 1) R12.1 Support OS7070 System

- 2) R12.2 Support OS7200L System.

- 3) R12.3 MIB V4.4
Support the MIB which is reflected version 4.4x DB

- 4) R12.4
 - List of Software package on the server can be found.
 - List of Software package on the server can be deleted
 - Booting Information of Software package on the server can be changed.

4.12.2 Limitation

None

4.12.3 Settings

Refer to Install Manual (NMS install manual – ENG.zip)

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4.13 R13. New Features of V4.60

4.13.1 R13.1. Error Message by e-mail

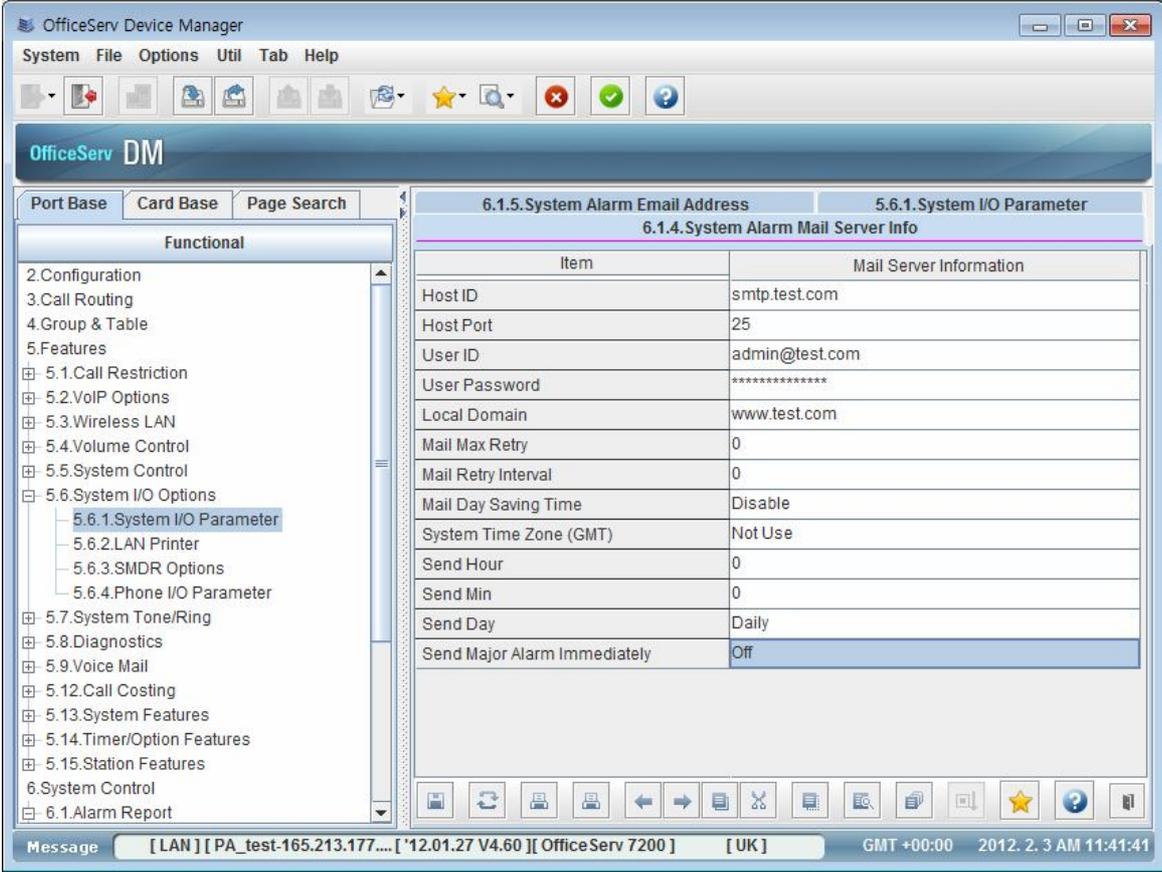
1) Description

- When the error message (system alarm, crash information, etc) created, it will be sent to e-mail account which is already registered on system.

2) Settings

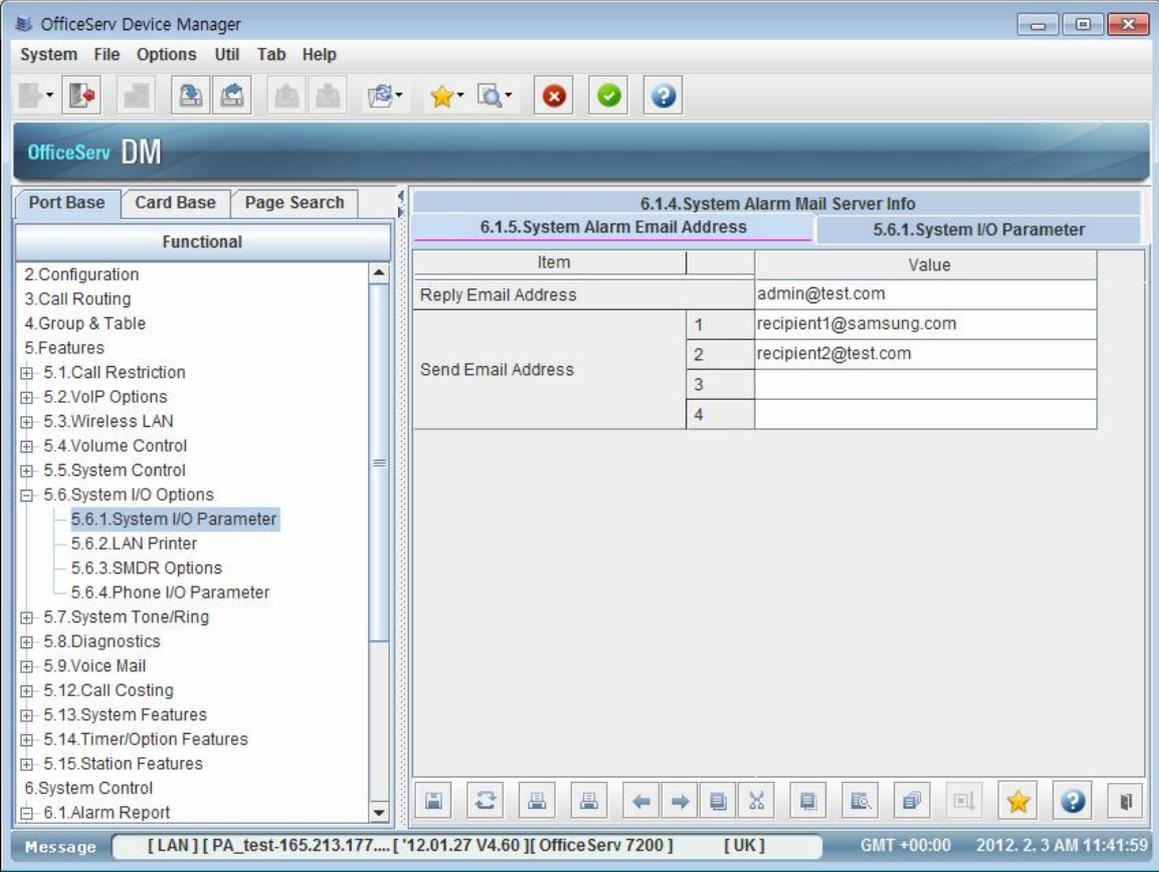
- You can set DM6.1.4 and DM6.1.5 for this feature.
- You need to input your e-mail server information in Host ID, Host Port, User ID, User Password, and Local Domain field at DM6.1.4.
- If you want to get the e-mail in some specific time at every day. Then you can set Send Day as “Daily” and set specific time at Send Hour and Send Min field. If Send Day is “Today” then you will get e-mail just one time when the time is come which you set at Send Hour and Send Min field.

※ If Local Domain field is not IP address, but URL. Then you need to set DNS Server at DM5.6.1.



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- You can put the recipient's e-mail address up to 4 in Send Email Address field.



4.13.2 R13.2. Single Address book

1) Description

- Provide single Address book features for all OfficeServ system.
- If user set the single address book on the system, then SMT-phones can have the common address book.
- User can have a private phone book on the SMT-phones as same as previous.

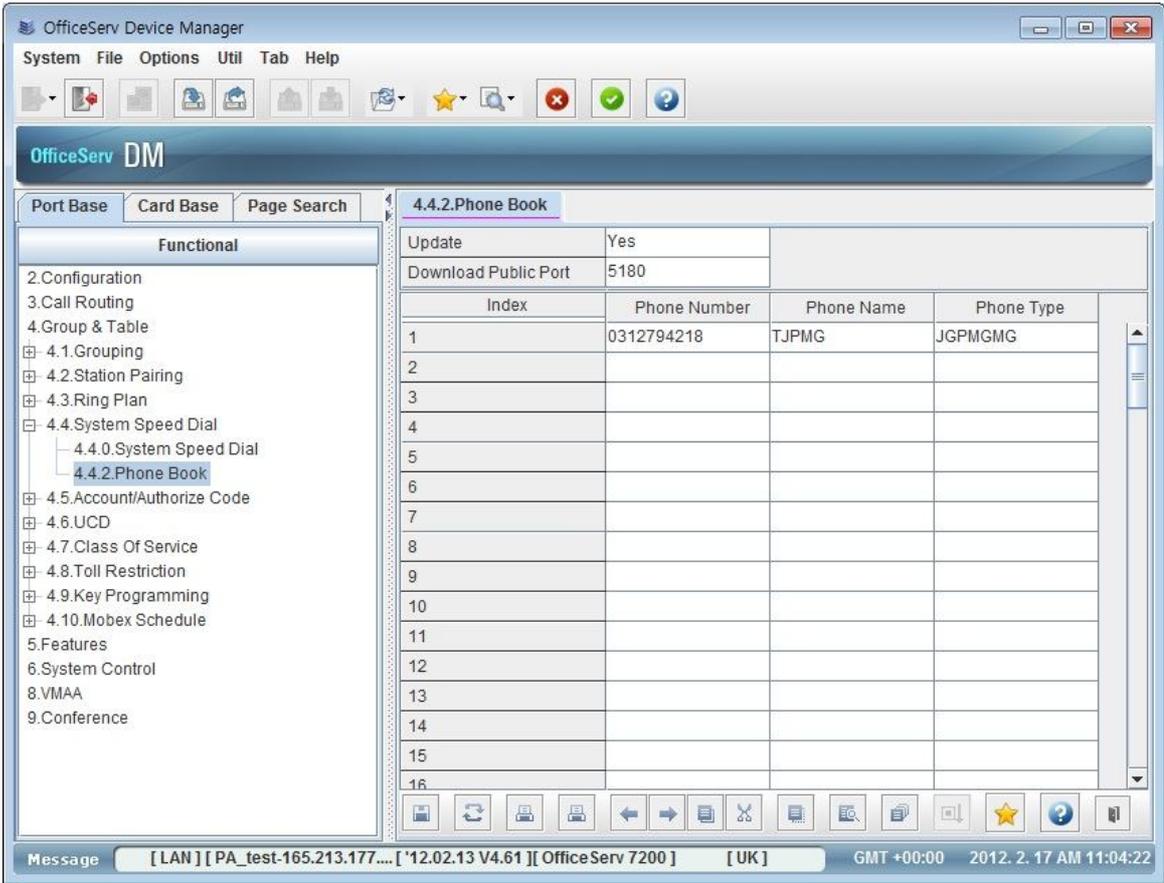
2) Settings

- You can set the single address book on DM 4.4.2 or MMC768.
- After setting address book, if you choose 'Update' option to 'YES' then SMT-phone which is working will get it immediately, and SMT-phone which is turned off will get it when the phone is started.
- For single address book, IP Phone supported single address book feature tries to connect via 80 port of OfficeServ system (or MP). So IP Phone has to be able to access this port in any circumstances. Make sure this port can be accessed in your environments.



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- If OfficeServ system (or MP) is located under NAT network. You surely have to set 'Download Public Port' option in DM 4.4.2 (MMC 768) and set port-forwarding from 'Download Public Port' to 80 port in NAT network (Router).
- 'Download Public Port' option's default value is 5180.



Option	Description
GROUP NAME	Assign category of Phone Book
DIAL NAME	Set Name
DIAL NUMBER	Set phone number
UPDATE YES	If this option is set to Yes then Update Notify will be sent to SMT-phones.

4.13.3 R13.3. SIP Trunk Error Alarm

1) Description

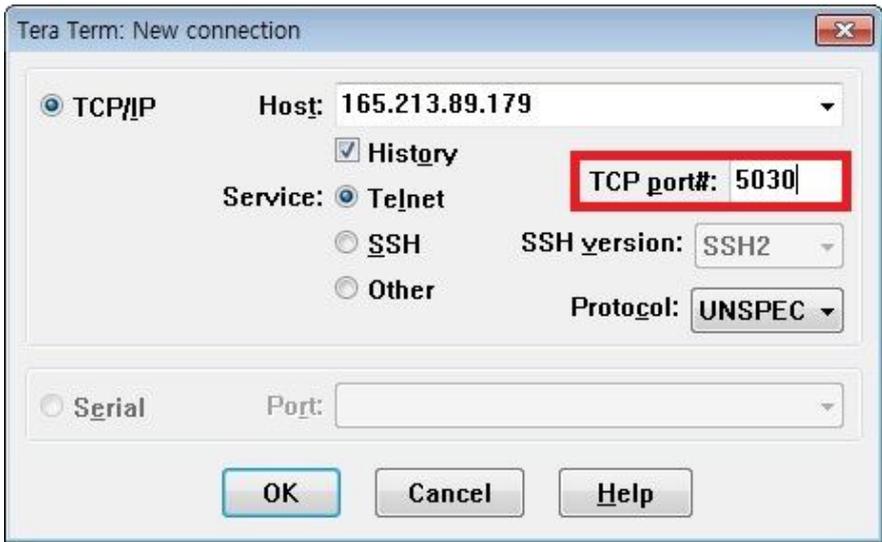
- Registration/Deregistration to SIP server will be saved on Alarm Log.
- Memory and Resource allocation fail will be saved on Alarm Log.

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- You can see this Alarm Log on DM tool and MMC.
- 2) Settings
- You can see Alarm Log on DM 6.1.1 and MMC851.

4.13.4 R13.4. Support Telnet trace

- 1) Description
- You can capture the MP trace through Telnet for OfficeServ 7200s/7100/7070/7030 systems.
 - In this version, there is no ID/Password for telnet connection. This will be more secured.
- 2) Settings
- Telnet port is 5030.



※ in case of OS7200s/7100/7070/7030 system, they already have the TCP port 23 for bash shell interface, therefore, it is not possible to use same port for Telnet trace. So TCP port 5030 is used for Telnet trace.

- You can set the option which you want to capture in MMC993 and MMC995 by using KMMC.

4.13.5 R13.5. Some default values are changed

- 1) Description
- Changing some default values is requested from field.
- 2) Settings



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- DID Busy Route value : ON → OFF in IT5.14.4 from [SEG]
- COS(Class Of Service) value of VMS port : 1 → 30 in MMC301 from [SEI]

DB NAME	OPION	Before	After
TEN_INFO_DB	DID BUSY ROUTE	ON	OFF
SLCB_DB	STATION COS	01	30

4.13.6 R13.6. MP support NTP Client

1) Description

- MP will support NTP(Network Time Protocol).
- If it is plugged into the network then it will update itself by polling the NTS.

2) settings

- Set IP Address or Domain Name on MMC841 NTP SERVER
- Set MMC861 SYS TIME ZONE
- Set MMC840 DNS SERVER1, DNS SERVER2 for Domain Name

4.13.7 R13.7. Separating SIP Trunking Ports and SIP Peering Ports

1) Functional Requirements

- Can use SIP trunking ports and SIP peering ports without separating them.
(same as before V4.60)
- Can use SIP trunking ports and SIP peering ports separately
- Can use ISP 1 ports and ISP 2 ports separately

2) Limitation

- Using one SIP Trunk license key without separating SIP trunking and SIP peering
(same as before V4.60)

3) settings

- You can set DM 5.2.13 SIP Carrier Options' SIP Trunk Max Channe, DM 5.2.12 SIP Stack/Ext/Trunk Options - SIP Trunk Configuration's SIP Trunk Max Channel for separating SIP Trunking and SIP Peering



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OfficeServ Device Manager

System File Options Util Tab Help

OfficeServ DM

Port Base Card Base Page Search

5.2.13.SIP Carrier Options

SIP Carrier 1

Item	Value
Supplementary Type	PBX Managed 2
302 Response	Disable
SIP Destination Type	To Header
Codec Auto Nego	Enable
URI Type	SIP
SIP Signal Type	UDP
RACK Support	Disable
Hold Mode	Send Only
Response to Tag	Keep
SIP Connection Reuse	Disable
SIP Mutual TLS Enable	Disable
SIP Validate Any TLS Certificate	Disable
SIP Trunking Codec PR1	G.729
SIP Trunking Codec PR2	G.711a
SIP Trunking Codec PR3	G.711u
SIP Trunking Codec PR4	Disable
SIP Trunking Use Alias	Disable
SIP Trunking Max Channel	224
Outgoing Originator Codec Use	Disable

OfficeServ Device Manager

System File Options Util Tab Help

OfficeServ DM

Port Base Card Base Page Search

5.2.13.SIP Carrier Options 5.2.12.SIP Stack/Ext/Trunk Options

Item	Value
Default SIP Carrier	1
iBG Expire Time (sec)	10
LCR Fast Setup	Disable
Incoming Mode	Follow DID Translation
Peer CLI Table	1
Received CLI Forward On Alias	Disable
Comm Exclusive	None
Common MSG Block Timer (Sec)	600
Register MSG Block Timer (Sec)	60
Register Retry Limit	2
SIP Peering Codec PR1	G.722
SIP Peering Codec PR2	G.711a
SIP Peering Codec PR3	G.729
SIP Peering Codec PR4	G.729
SIP Peering Max Channel	224
Outgoing Originator Codec Use	Disable

- In case of making a SIP outgoing call
 - A. Basically it's possible to making a SIP outgoing call until reaching SIP Trunk/Peering Max Channel



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- i. E.g.) 805 Trunk Group - Member : 8501, 8502, 8503, 8504, 8505, 8506, 8507, 8508
 - ISP Selection: ISP 1
- 806 Trunk Group - Member : 8501, 8502, 8503, 8504, 8505, 8506, 8507, 8508
 - ISP Selection : ISP 2
 - ISP 1 SIP Trunking Max Channel: 4
 - ISP 2 SIP Trunking Max Channel: 4
 - ➔ ISP1 and ISP2 can use from 8501 to 8508 without separating until reaching SIP Trunking Max Channel.

- B. It's possible to separate SIP trunking and SIP peering, or each ISPs
 - i. E.g.) 805 Trunk Group - Member : 8501, 8502, 8503, 8504
 - ISP Selection: ISP 1
 - 806 Trunk Group - Member : 8505, 8506, 8507, 8508
 - ISP Selection: ISP 2
 - ISP 1 SIP Trunking Max Channel: 4
 - ISP 2 SIP Trunking Max Channel: 4
 - ➔ Separating ISP1 and ISP2

- In case of answering a SIP incoming call
- C. Basically it's possible to answer a SIP incoming call until reaching SIP Trunking / Peering Max Channel.
 - i. E.g.) 805 Trunk Group - Member : 8501, 8502, 8503, 8504, 8505, 8506, 8507, 8508
 - ISP Selection: ISP 1
 - 806 Trunk Group - Member : 8501, 8502, 8503, 8504, 8505, 8506, 8507, 8508
 - ISP Selection: ISP 2
 - 807 Trunk Group - Member : 8509, 8510
 - ISP Selection: ISP 1
 - ISP 1 SIP Trunking Max Channel: 4
 - ISP 2 SIP Trunking Max Channel: 4
 - ➔ In case of incoming a call from ISP 1, OfficeServ searches SIP trunk group set to ISP 1 from 1st trunk group. In upper example, 805 trunk group is selected.
 - ➔ and searching available trunk port from last trunk member. In upper example, 8508 port is selected.
 - ➔ If there is no matched trunk group (E.g. from ISP 4), OfficeServ searches SIP trunk port from last SIP trunk port of all SIP trunk port regardless of SIP Trunk



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group.

D. It's possible to separate SIP trunking and SIP peering, or each ISPs

i. E.g.) 805 Trunk Group - Member : 8501, 8502, 8503, 8504

- ISP Selection: ISP 1

806 Trunk Group - Member : 8505, 8506, 8507, 8508

- ISP Selection: ISP 2

ISP 1 SIP Trunking Max Channel: 4

ISP 2 SIP Trunking Max Channel: 4

➔ In case of incoming a call from ISP 2, OfficeServ searches SIP trunk group set to ISP 2 from 1st trunk group. In upper example, 806 trunk group is selected

➔ and searching available trunk port from last trunk member. In upper example, 8508 port is selected.

4.13.8 R13.8. SMDR Buffering (MP10a, MP2oS)

1) Description

- SMDR data was sent to a billing server in real time. If a billing server is disconnected to a system, we lost the SMDR data.
- Now: Although a billing server is disconnected to a system, SMDR data are buffered in system. Later, when a billing server is connected to a system, we get the buffered SMDR data
- Max buffering capacity : 10,000 data

2) Settings

- None

4.13.9 R13.9. Max call on DDI per Ring Plan

1) Description

- Now: Max call count on DDI is extended to be "per Ring Plan".
- Before: Max call count per incoming digit is applied for all Ring Plan.

2) Settings

Option	MMC	Description
MAX CALL COUNT	DM3.2.3 (MMC714 DID DIGIT)	You can set this per Ring Plan. This means Max count of receivable call



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4.13.10 R13.10. Max call in queue function

1) Description

- Limitation of Max waiting call count for incoming call to UCD group.

2) Settings

- Set Max Call Count which means limitation of waiting call for UCD Group in MMC 607
- Set Max Call Destination in MMC 607. If there is incoming call which is over the limitation of waiting call, then call will be transferred to assigned destination in this option.

Option	MMC	Description
MAX CALL COUNT	DM 4.6.1 (MMC607 UCD OPTIONS)	Acceptable incoming call count
MAX CALL DESTINATION	DM 4.6.1 (MMC607 UCD OPTIONS)	Call Transfer Destination for incoming call when it over the limitation

4.13.11 R13.11. Malicious Call Restriction

1) Description

This feature is able to protect the fraudulent sip call use of OS system. By setting it enable, you can prevent a unauthorized SIP call from going through OS system via SIP trunk or SIP peering. In addition, OS system blocks the IP address for specified period in case OS system gets wrong User ID or Password of an SIP phone several times when an SIP phone tries to register to system.

The allowed IP lists are the following.

- Registered sip station ip address (DM 6.2.3)
- Voip peering ip addresses (DM 5.2.17)
- Carriers ip addresses (DM 5.2.13)

2) Limitation

- You can set **None** / **Response** options for OS7200/7200s/7100/7070/7030.
- You can set **None** / **Response** / **No response** options for OS7400.

3) Settings

- DM 5.2.12 SIP Stack/Ext/Trunk Options – SIP Trunk Configuration
 - Comm Exclusive :
 - None : Disable this function. OS system will allow all SIP calls.



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- ii. Response : OS System will not allow SIP calls from unauthorized IP to go through OS system via SIP trunk/Peering by sending deny message (403 forbidden)
 - iii. No Response : OS system will ignore all the SIP message from unauthorized IP address and block the relevant IP address. Beside, OS system blocks the IP address for specified period in case OS system gets wrong User ID or Password of an SIP phone several times when an SIP phone tries to register to system.
- B. Commom MSG Block Timer : This timer is applied to all sip message except REGISTER message. (1 ~ 84600 second)
- C. Register MSG Block Timer : This timer is applied to REGISTER message. (1 ~ 84600 second)
- D. Register Retry Limit : If SIP phone try to register with wrong User ID or Password over this value, OS system blocks the IP address of SIP phone.

The screenshot shows the OfficeServ Device Manager interface. The left sidebar lists a tree view of configuration options, with '5.2.12.SIP Stack/Ext/Trunk Options' selected. The main area displays a table of configuration items and their values. A red box highlights the 'SIP Trunk Configuration' section at the bottom of the table.

Item	Value
SIP Stack Configuration	
Invite Ring Time (100ms)	50
Provisional Time (100ms)	1800
Invite No Response Time (100ms)	50
General No Response Time (100ms)	50
Request Retry Time (100ms)	80
SIP Extension Configuration	
Signal Port	5060
IPUMS/IVR Signal Port	5070
SIP Expire Time (sec)	600
NAT Reg Expire Time	60
Default SIP Carrier	1
iBG Expire Time (sec)	10
LCR Fast Setup	Disable
Incoming Mode	Follow DID Translation
Peer CLI Table	1
Received CLI Forward On Alias	Disable
Comm Exclusive	Response
Common MSG Block Timer (Sec)	600
Register MSG Block Timer (Sec)	60
Register Retry Limit	2

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4.14 R14. WE VoIP features of V4.65

4.14.1 R14.1. Single CID number for WE VoIP Service

1) Description

- This option enables WEVoIP which is a secondary to have a primary number that a station-paired phone has, so that a primary number & name can be delivered to an opposite user.
- This feature is restricted to an internal call(including station group) only. For an external call, you can make a 'Single CID number' using the <Send CLI Number> option in KMMC323/DM 2.3.4.
- 'Single CID number' is not printed on the SMDR log.

2) Programming

- In the DM4.2.1 Station Pair , <Single CID Number> option has been added.
- Refer to the following example.

4.2.1. Station Pair		
Primary No	Secondary No	Single CID Number
2001	2002	On
2002	2001	Off

- a. 2001 and 2002 are paired with each other.
- b. <Single CID number> of 2001 is enabled, so that 2001 will be designated as a primary number/name.
- c. In this case, enabling <One number Service> of 2002 will be restricted.

4.14.2 R14.2. E.164 support for WE VoIP

1) Description

- This feature enables the WEVoIP to support E.164 number (including '+' prefix) properly.
- For this feature, a digit translation scheme for E.164 number has been newly added to V4.65.

2) Programming (Example for Italy)

- Case1 : Outgoing call through an ISDN trunk.
 - a. Country code : +61
 - b. Dialing number on WEVoIP : +6131234569
 - c. 'Auto Prefix Code' for ISND trunk group : 801
 - d. In the DM 5.2.29 <WE VOIP Outgoing Digits> (MMC877), insert 'Access Digit/Digit



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Length/Delete Length' as below.

5.2.29.FMC Outgoing Digits				
Table No	Access Digit	Insert Digit	Digit Length	Delete Length
0	801+61		6	6
1			0	0

e. OfficeServ system will dial '31234569' through an ISDN trunk.

- Case2 : Incoming call through an ISDN trunk.

a. Calling Number : 04253112342

b. Local area code : 042

c. In the DM 5.2.30 <WE VOIP Incoming Caller Modify>, insert 'Calling Number/Insert Number/Check Length' as below.

5.2.30.FMC Incoming Caller Modify				
Table No	Calling Number	Insert Number	Check Length	Delete Length
0	042	+61	3	0
1			0	0

d. OfficeServ system will modify the phone number of the originating caller and provide the CID '+610425311234' to WEVoIP for direct dialing from the call log.

3) Exception

- If dialed digits are registered in DM 5.2.28 E.164 Deny table, system does not convert dialed digits to E.164 format.

4.14.3 R14.3. WE VoIP Provisioning Enhancement

1) Mobile profile

- In the DM 5.2.23 <Mobile Profile>,

a. Codec Priority

b. SLIK codec

c. AMR-WB codec

d. Direct mobile number : Exception rules for any numbers dialed from the mobile handsets that are to be placed directly to the mobile network, such as Voicemail Access Codes.

1571	Exact same 4 digit-number only
080!	080 and following any number of digits.
077XXXXXXXX	077 and following any 8-digit numbers



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5.2.23.Mobile Profile		
Item	Value	
AP SSID	SMT	
Upgrade Server	10.0.1.101	
Upgrade URL	tftp	
Auto Prefix Code	#	
Auto Prefix Exception Number	XXXX,*65!,*66!,*60!,805,9!,804	
MVS Local Port	9012	
MVS Public Port	9012	
MVS Disa Number	07088585570	
VMS Public Number	07088585571	
Codec Priority	1	SILK
	2	AMR-WB
	3	G.711u
	4	G.711a
SILK Codec	Payload	114
	Sampling Frequency	24000 Hz
	Max Ptime	100 ms
	DTX	Off
	FEC	On
AMR WB	Payload	113
	Bit Rate	23850 bps
	DTX	Off
Direct Mobile Number	02116	

- 2) Login Profile
 - DM 5.2.24 <Login Profile>
 - a. Multi Frame / Multicast / TOS / Jitter / SIP Signal Type / Description
 - b. Roaming Scan List

- 3) WE VOIP Model Parameter
 - In the DM 5.2.27 <WE VOIP Model Parameter>, following options has been moved from DM 5.2.24 <Login Profile>.
 - a. Roaming Trigger / Roaming Delta / Roaming Scan Period



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5.2.27.FMC Model Parameter				
Entry No	Model Name	Roaming Trigger	Roaming Delta	Roaming Scan Period
1	default	-70	10	3
2	SHW-M250S	-70	10	3
3	SHW-M250K	-70	10	3
4	SHW-M250L	-70	10	3
5	SHV-E120S	-70	10	3
6	SHV-E120K	-70	10	3
7	SHV-E120L	-70	10	3
8	SHV-E160S	-70	10	3
9	SHV-E160K	-70	10	3
10	SHV-E160L	-70	10	3

4) WE VOIP Common Option

- In the DM 5.2.31 <WE VOIP Common Option>, set roaming channel information as per roaming scan mode.
- If you select <5G only> in DM 5.2.24 Scan 5G Only, channels of Roaming Scan 5G are saved to Login Profile.
- If you select <2.4G only> in DM 5.2.24 Scan 5G Only, channels of Roaming Scan 2.4G are saved to Login Profile.
- If you select <Auto> in DM 5.2.24 Scan 5G Only, All channels are saved to Login Profile.

5) Codec Limitation

We recommend user should change codec of IPP/SIP/WIP/SIP trunk/SPNet to G.711 in case of using WE VoIP phone. Because WE VoIP phone does not support G.729. So if codec of IPP/SIP/WIP/SIP trunk/SPNet is G.729, there can be a voice problem because of mismatching codec.



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4.15 R15. New security features of V4.65

4.15.1 R15.1. High secure login

1) Password Encryption

- Below passwords are encrypted for enhancing security.
So these passwords are shown as “*****” in each menu and can’t be read from backup DB.

. DM 2.1.7 Admin Password

. DM 2.7.2 SIP Phone Password

. DM 5.2.13 Auth Password

. DM 5.13.8 Community Server

- User should access OfficeServ system (V4.65) with its corresponding s/w version of DM.** If using with old DM, user can’t access to the system because of admin password encryption. But with V4.65 DM, user can access to the old system if he/she deactivates encryption box of DM.

2) Secure DM Login

- The repetitive DM Login attempts in a short time will follow two options in below rule.

a. DM Login Retry Limit / DM Login Prevent Duration

5.14.2. Confirm/Disconnect/NoAction Timer	
Value	Item
Move Wait Time (sec)	20
Unregistered FWD Time (sec)	5
DM Login Retry Limit	10
DM Login Prevent Duration (min)	1

- DM Login password should be changed ‘samsung’ which is a former default into different ones by the DM user.

3) SIP station security

- There are no default ID and password for SIP station. So manager should set these values after upgrading V4.65.
- SIP phone’s password should be more than 6 digits.

4) VM/AA security

- SVMi will make OfficeServ’s subscribers change the default password when using it first.
- For detail information, refer to R15.3. New features of SVMi-20i.



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4.15.2 R15.2. IP White list

1) DM IP White list

- The DM IP address registered in the DM 5.13.9 is only allowed for the system access.
- If there is no IP address inserted as below, the system will fully open the access from any IP address.

※ Be sure that manager should set DM IP White List for system security.

After upgrading V4.65, DM IP White List has no valid IP addresses so system can be vulnerable to hacking.

5.13.9.DM IP White List		
Entry No	IP Address	Description
1	0.0.0.0	
2	0.0.0.0	

- If access is denied, below message will be displayed.

“Current PC is not allowed to access system DB. Please use another PC and check DM Access IP List in DM 5.13.9”

2) Phone IP White List

- The IP address in the DM 5.13.10 (MMC875) is only allowed to register to the system.
- The IP address of the CTI solutions and the services in the DM 5.6.2 <Lan Printer> should be inserted in this list.

※ Be sure that manager should set Phone IP White List.

When you install V4.65, if no valid IP addresses is in the list, all IP/SIP/WIP phones and CTI solutions won't be registered to the system.

5.13.10.Phone IP White List		
Entry No	IP Address	Description
1	165.213.80.50	
2	165.213.80.100	
3	165.213.80.101	
4	165.213.80.102	

- [Exception]

If <Use IP White List> of DM 2.7.1/2.7.2 is set to <Disable>, the designated phone can register to the system although its IP address is not in the DM 5.13.10 Phone IP White List.



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3) Management IP White List

- IP address of Telnet/FTP/SMDR in the DM 5.13.11 (MMC875) are only allowed for the system access.

※ Be sure that manager should set Management IP White List.

When you install V4.65, if no valid IP addresses is in the list, Telnet/FTP/SMDR won't be accessible.

5.13.11.Management IP White List		
Entry No	IP Address	Description
1	165.213.80.177	
2	0.0.0.0	
3	0.0.0.0	
4	0.0.0.0	

- In case of OS7030 Master-Slave system, slave IP address is saved to master list and master IP address is saved to slave list automatically.

4) IP Address range rule

- There are two ways to input DM/Phone/Management IP address. One is inputting one address to each entry. Another is inputting address range as below.

IP address format	IP address range
A.B.C.255	A.B.C.1 ~ A.B.C.254
A.B.255.255	A.B.0.1 ~ A.B.255.254
A.255.255.255	A.255.255.254



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4.15.3 R15.3. New features of SVMi-20i.

SVMi will make OfficeServ' s subscribers change the default password when using it first.

<As is>

Most subscribers are using the VM service with the default password unchanged or simply mixed password. It can cause an illegal use.

<To be>

New security enhancement has been added in V4.65 and V2.01 of SVMi-20i as below.

1. Force Subscriber to Change PSWD" is now available in DM 8.5 <System Parameter>.
2. In the DM 8.5, there are three options.

- None

Same as previous security policy

- Change Default Password (USA: Default)

When a subscriber tries to log on his/her mailbox, a subscriber' s default password will be denied to access and is required to change it into new one

- Change Default Password and Deny Simple One (All other countries : Default)

When a subscriber tries to log on his/her mailbox, if the password is a default value or simple combinations, SVMi will deny these ones to access and make him/her change it into new one.

- Simple ones mean that repeating numbers (like 1111,2222), ascending & descending numbers(like 1234, 4321) or user's own extension number.

- Subscriber PSWD Min Length: 0 -> 4

When SVMi's DB is initialized after upgrading to V2.01(or above), the password length '4' will be applied correctly. In case of uninitialized SVMi DB, this minimum length will be set to '3' temporarily.

※ All changes are related to the subscriber configuration. For an admin password, you can change it in the DM 8.5 <System Parameter>.



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4.15.4 R15.4. Directory service : Name search

1) Description

- V4.65 supports that searching the name on the basis of max 2 characters.

2) How to use

- i.e) ITP-51xx / DS-50xx series

a. Press 'Scroll' button and select the 'CALL' soft key.

```
3100:Admin
CALL OTHER ANS->
```

b. Select the 'DIR soft key.

```
select an option
DIR LOG
```

c. Choose one soft key among 3 directory options.

```
directory dial
PERS SYS STN
```

d. Enter one or two characters to search a name you are looking for.

```
system speed #
Enter 1st letter
```

e. You can press '#' button to enter 2nd letter.

```
Scroll on [AN]
Use volume keys
```

4.15.5 R15.5. Unconditional ringing for SIP stations

1) Description

- V4.65 supports the 'Unconditional ringing' mode for SIP stations as a member of a station group.

2) Limitation

- In case of 'Unconditional Ring' mode, max. members of an UCD group are as below.

System	Number of Member	Number of SIP phone Member
OS7030	16	8
Other systems	32	10

- Interval processing delay
 - OfficeServ system sends 'SIP INVITE' message to 4 group members at once.
 - And every interval delay for the SIP INVITE is 50ms.
 - Logically, total delay will be '400ms' between 1st and 9th SIP INVITE message.

4.15.6 R15.6. SIP cause message display

- SIP messages such as 4xx, 5xx and 6xx can be displayed on the phone.
- To be shown SIP cause message, user should set below options.
 - . In DM 2.1.5, set <SIP Cause Display> to <Disable>.
 - . In DM 5.13.5, set <Language> to <English>.



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4.15.7 R15.7. TOS field of SIP signal packet setting

- In DM5.2.12 user can change TOS field. (MP40 only)

4.15.8 R15.8. SIP Privacy Header

- In DM 5.2.13 SIP privacy header value is able to be changed. (header, session, user, none, critical, token, id ...)

4.15.9 R15.9. Change default value

- MMC110 AUTO HOLD: ON
- MMC120 DIAL MODE: OVERLAP
- MMC501 SIPT CON DELAY TIME: 200 ms
- MMC501 SYS HOLD RECALL TIME: 120 sec
- MMC724 Operator number: 9, Trunk Group number: 0

4.15.10 R15.10. SIP call waiting

- Be sure that when SIP call receives second call during conversation, system does not allow call waiting for second call.

5. Group Development Function MAP

Feature list	7400	7200		7100	7070	7030
		MP20	MP20S			
R1. New VM	Yes	Yes	No	No	No	No
R2. Subscription Card	Yes	Yes	Yes	Yes	No	No
R3. WE VoIP Client	Yes	Yes	Yes	Yes	Yes	Yes
R4. Advanced MGI Channel Allocation Scheme	Yes	Yes	Yes	Yes	Yes	Yes
R5. Enhanced PNP	Yes	Yes	Yes	Yes	Yes	Yes
R6.	Yes	Yes	Yes	Yes	Yes	Yes
R7. Security Enhancement						
R7.1. TLS for SIP signaling	Yes	Yes	No	No	No	No
R7.2. Supporting sRTP	Yes	Yes	No	No	No	No
R7.3. Reinforcement on Security Vulnerability	Yes	Yes	Yes	Yes	Yes	Yes
R8. SIP Trunking Enhancement	Yes	Yes	Yes	Yes	Yes	Yes
R9. Mobex Enhancement	Yes	Yes	Yes	Yes	Yes	Yes



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R10. 3rd-Party Solution	Yes	Yes	Yes	Yes	Yes	Yes
R11. CNF24 phase 2	Yes	Yes	Yes	No	No	No
R12. OfficeServ NMS V1.61	Yes	Yes	Yes	Yes	Yes	Yes
R13. New Features	Yes	Yes	Yes	Yes	Yes	Yes



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6. Compliance Table

#	Qnet	VoCs	Country	요구사항
1	-	new VM	-	R1
2	-	Subscription Card	-	R2
3	VoC	Mobile Client	KOR, MIA, Plant, SEG, STA, SEUK, SEI	R3.1 R3.2
4	VoC	Unregistered forward	KOR	R3.3
5	VoC	Remote dial	KT	R3.4
6	VoC	Saving MGI channel for MOH, page and so on	Plant	R4
7	VoC	Plug and Play for IP phones	MIA,Plant,KOR,SEI	R5.1
8	VoC	embedded DHCP server	SEG,Plant,KOR	R5.2
9	-	System Management Enhancement	-	R6.1~ R6.3
10	N091120002	Statistics of OfficeServ7000 CPU performance and Memory usage	KOR,SEUK	R6.4
11	N100331002	Need to add TLS and SSL Encryption support to all Platforms with the EMG feature.	STA,Plant,SEG	R7.1 R7.2
12	N091215009	OS7400 ARIA Security Protocol implementation	KOR	
13	VoC	Security improvement of PHP and Apache	STA	R7.3
14	N100122001	Need to support multiple SIP trunk accounts at the same time	STA,Plant,SEG,SEI, SERC	R8.1
15	N090625009	Change the 480 Sip Message	SEUK	R8.2
16	-	[Voice Band Data Mode through SIP trunk]	-	R8.3
17	N100525002	Privacy header option for outgoing SIP trunks calls.	Plant	R8.4
18	VoC	Option for Tandem between SIP Trunk	STA	R8.5
19	VoC	Selectable codec for SIP trunk calls using MPS	Plant,SERC	R8.6
20	N101116006	Call Back option for MobEx calls	SEI, KOR	R9.1
21	N090625013	STA needs MOBEX feature to support scheduling	STA	R9.2
22	VoC	Busy status at MOBEX extension. Paired fixed line should be also busy.	SEG	R9.3
23	N090625012	STA needs ability to set what type of calls can ring MOBEX stn.	STA, SEG	R9.4
24	N101115002	Authorize MOBEX by CLI Ringing Table	SEI, SERC	R9.5
25	N090625008	Recommend to change MOBEX feature code displays	STA, SEG	R9.6
26	VoC	IOT of Video Doorphone	All(SEI,SEUK,KOR, Plant,STA,SERC,SEI)	R10.1 R10.2
27	N100802005	Broadsoft Certification for OS7000	Plant	R10.3
28	VoC	CNF24 & Outlook	STA, SEG	R11.1



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29	-	CNF24 enhancement	-	R11.2~ R11.11
30	-	OfficeServ NMS 1.61	-	R12
31	VoC	Error Messages by email	STA,SEUK,SERC,SEG,KOR	R13.1
32	VoC	single Address book and calendar	Plant, SEUK	R13.2
33	-	SIP Trunk Error Alarm implementation	SEUK	R13.3
34	-	mp telnet trace	SEUK	R13.4
35	N101116008	CoS of VMS ports	SEI, SERC, KOR	R13.5
36	VoC	Set Menu 5.14.4 DID Busy Route value default = off	SEG	R13.5
37	VoC	MP SUPPORT NTP	STA	R13.6
38	VoC	Add an option to allocate the SIP trunking ports for SIP peering	STA, SEG	R13.7
39	N101116002	SMDR Buffering (MP10a-MP20S)	SEI	R13.8
40	VoC	Max call on DDI is improved and extended to be "per Ring Plan"	SEUK, SERC	R13.9
41	VoC	Max call in queue function	SEUK, SERC	R13.10