

To : ALL	Doc. No.	SECYMM-XX-X
C.C. :	Country	ALL
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Subject : OfficeServ7000 S/W V4.99 (2016.09.30) release		

**Overview :**

## OfficeServ 7000 series MP S/W release

System	Package name	Description
OS7200 MP20S	MP20s_V499_20160930.zip	MP S/W for MP20S card
OS7100 MP10A	MP10a_V499_20160930.ZIP	MP S/W for MP10A card
OS7070	MP07_V499_20160930.ZIP	MP S/W for OS7070
OS7030	MP03_V499_20160930.ZIP	MP S/W for OS7030

**S/W Compatibility Table:**

## (1) OfficeServ 7030 MP03, OfficeServ 7070 MP07 Module version table

System	OS7030 (MP03)	OS7070 (MP07)
System	V4.99 '16.09.30	V4.99 '16.09.30
MP	V4.99 '16.09.30	V4.99 '16.09.30
SP	V2.63 '15.03.27	V4.36 '15.03.27
VM	V2.86 '16.08.19	V2.86 '16.08.19
MGI	V2.06 '11.12.09	V2.06 '11.12.09
BRM	V4.22h '13.11.20	V4.22h '13.11.20
PRM	-	V4.28 '10.09.07 V4.29 '11.05.03(STA only)
WEB	V4.12h '10.04.13	V4.12h '10.04.13
MPS	V2.01 '11.12.09	V2.01 '11.12.09
SNMP	V1.61 '11.09.01	V1.61 '11.09.01-
Boot	V4.40 '09.04.21	V4.30 '08.09.22
DM	V4.99 '16.09.02	V4.99 '16.09.02
RTG	V1.00 '11.12.09	V1.00 '11.12.09

## (2) OfficeServ 7100 MP10a/MP11, OfficeServ7200 MP20S Module version table

System	OS7100 (MP10a)	OS7200 (MP20S)
System	V4.99 '16.09.30	V4.99 '16.09.30
MP	V4.99 '16.09.30	V4.99 '16.09.30
SP	V2.65e '16.09.08	V2.65e '16.09.08

<b>VM</b>	<b>V2.86 '16.08.19</b>	<b>V2.86 '16.08.19</b>
<b>MGI</b>	<b>V2.06 '11.12.09</b>	<b>V2.06 '11.12.09</b>
<b>BRM</b>	<b>V4.22h '13.11.20</b>	<b>V4.22h '13.11.20</b>
<b>WEB</b>	<b>V4.12h '10.04.13</b>	<b>V4.12h '10.04.13</b>
<b>MPS</b>	<b>V2.01 '11.12.09</b>	<b>V2.01 '11.12.09</b>
<b>SNMP</b>	<b>V1.61 '11.09.01</b>	<b>V1.61 '11.09.01-</b>
<b>Router</b>	<b>-</b>	<b>-</b>
<b>Boot</b>	<b>V1.07 '09.02.24</b>	<b>V0.30 '09.09.22</b>
<b>DM</b>	<b>V4.99 '16.09.02</b>	<b>V4.99 '16.09.02</b>
<b>PWP</b>	<b>-</b>	<b>V4.60 '11.10.24</b>
<b>RTG</b>	<b>V1.00 '11.12.09</b>	<b>V1.00 '11.12.09</b>

**(3 SMT phone series version table**

<b>SMT-i5243</b>	<b>SMT-i5230</b>	<b>SMT-i5220</b>	<b>SMT-i5220S</b>	<b>SMT-i5210</b>	<b>SMT-i5210S</b>
<b>V1.97 (2013.06.03)</b>	<b>V1.38 (2015.11.15)</b>	<b>V2.65 (2015.11.15)</b>	<b>V2.65 (2015.11.15)</b>	<b>V1.54 (2015.11.16)</b>	<b>V1.08 (2015.11.15)</b>
<b>SMT-i3100</b>	<b>SMT-i5264</b>	<b>SMT-i5343</b>	<b>SMT-i6011</b>	<b>SMT-i6021</b>	
<b>V1.73 (2015.11.11)</b>	<b>V1.35 (2014.08.29)</b>	<b>V1.46 (2015.11.12)</b>	<b>V1.30 (2016.04.27)</b>	<b>V1.30 (2016.04.27)</b>	

## New Version Changes

### '16.09.30 V4.99

#### ※ DB Compatibility

- The database of V4.99 is not compatible with that of a previous version including the V4.95, V4.94, V4.93
- So you download the database of the existing system before upgrading to V4.99 using DM.
- After upgrading to V4.99, upload the downloaded DB by DM.

#### ※ All of changes are accumulatively applied to last version.

#### [New features]

- 1. [-/ALL/ ALL] system log files is deleted periodically.**  
<Description>  
System log files including /var/log/wtmp, /var/run/utmp are deleted every 1 minute to prevent full of nand flash
- 2. [-/ALL/ALL] KMMC 900 CLEAR PASSCODE - ADM and KMMC 905 DEVICE TEST - COLSET DM ACL options are deleted for security**  
<Description>  
KMMC 900 CLEAR PASSCODE - ADM and KMMC 905 DEVICE TEST - COLSET DM ACL options are deleted for security  
<Alternative method>  
**If you want to know how to initialize DM password of Admin or DM ACL IP Lists, contact HQ**
- 3. [-/ALL/ALL] telnet/web to MP is disabled for security**  
<Description>  
Basically telnet/web to MP is disabled from V4.98/V4.99. But you can enable them in DM 5.2.10 System IP options – Protocol Enable – HTTP/TELNET  
<Related Features>  
If you want to use FMC Client, OSPP phonebook, DM download from MP features, you have to enable this option  
<Caution>  
**In case of upload old version DB by using OfficeServ DM for upgrade to V4.99, this option is enabled.**
- 4. [#TE16702L /ALL/ALL] add “Anonymous” display text in the From Header (Reported by USA)**  
<Description>  
When making an anonymous call, there was no “Anonymous” display text in the from header. “Anonymous” display text is added in this version.
- 5. [-/ALL/ALL] Upload embedded DM file by using DM.**  
<Description>  
You weren't able to upload embedded DM file by using DM at V4.98, This is changed to be able to upload it from V4.99.

#### [Bug Fix]

- 1. [#TE16706U /ALL/ALL] send SIP 200 OK including 'sendonly'. (Reported by UK)**  
<Description>  
When receiving reINVITE including session level 'sendonly', OfficeServ sent 200 OK including 'sendrecv' instead of 'sendonly'. This is fixed.

2. **[-/ALL/ALL] Email gateway delivery fails when domain name is set for SMTP server. It's fixed. (Reported by: Chile)**  
<Description>  
Email gateway delivery often failed when domain name is set for SMTP server because it does not verify the IP address returned by DNS server.  
This bug is included since 2015. Previous version software does not have this issue.
  
3. **[#TE16603C/ALL/ALL] DTMF administrator will not be able to create more than 6 digit subscriber. (Reported by: Australia)**  
<Description>  
If a hacker can access DTMF administrator, he can make a long distance call although the outcall authorization is off.  
The hacker can make long digits extension and he makes a call to transfer to that extension, SVMi will make a long distance call.  
So, It's changed that DTMF administrator will not be able to create subscriber that has more than 6 digits. To create long digits subscribers, you have to use DM
  
4. **[#TE16600T/ FPGA version 16DLI2 Card on MP20S,MP10A/ALL] DGP speaker open issue is fixed. (Reported by USA)**  
<Description>  
If you use FPGA version 16DLI2 Card, it's possible to send speaker on message to DGP during conversion. This issue is fixed.
  
5. **[-/ALL/ALL] The time of going to idle state after user disconnects a call and keep hook-off is changed**  
<Description>  
The time of going to idle state after user disconnects a call and keep hook-off is changed from 1 minute to 3 minutes. If you answer a new call by using button after this time, both handset and speaker are opened at the same time.  
\* This time was changed from 3 minutes to 1 minute at V4.91
  
6. **[#TE16803H, #TE16404F/ALL/ALL] Incorrect call duration time of SMDR is fixed. (Reported by UK and Holland)**  
<Description>  
Incorrect call duration time of SMDR is fixed at 2 cases as below  
- After setting absent message by using jog shuttle of IP phone  
- After UCD group in/out by UCD supervisor
  
7. **[#TE16902I/ ALL/ALL] Issue retrieving calls from Park is fixed. (Reported by UK)**  
<Description>  
if you have parked an incoming call when the DS phones have another incoming call you cannot retrieve the parked call while the incoming call is ringing. This issue is fixed.
  
8. **[-/OS7070/ALL] Management IP White List Problem is fixed.**  
<Description>  
Intermittently Management IP White List feature didn't work correctly on OS7070. This is fixed.
  
9. **[#TE16803P/MP40,MP20/ALL] 'SIP Register Per User fails' Issue is fixed. (Reported by Australia)**

<Description>

In case of using SIP Register Per User feature (DM 5.2.14) at V4.98, SIP registrations are failed. This issue is fixed.

**16.08.01 V4.98**※ **DB Compatibility**

- ※ - **The database of V4.98 is not compatible with that of a previous version including the V4.92**
- ※ - So you download the database of the existing system before upgrading to V4.98 using DM.
- ※ - After upgrading to V4.98, upload the downloaded DB by DM.

**[New features]****10. [-/ALL/ USA] 911 CID number to PRI trunk can be modified. (Requested by USA)**

## &lt;Description&gt;

You can modify 911 emergency CID number

DM 2.1.5 System Options - Emergency CID Table Number is added and this means table number of 'DM 2.4.3 Send CLI Number'

**11. [-/ALL/ALL] telnet/web to MP is disabled for security**

## &lt;Description&gt;

Basically telnet/web to MP is disabled from V4.98. But you can enable them in DM 5.2.10 System IP options – Protocol Enable – HTTP/TELNET

## &lt;Related Features&gt;

If you want to use FMC Client, OSPP phonebook, DM download from MP features, you have to enable this option

**12. [-/ALL/ALL] telnet/ftp to SVMi-20i is disabled for security**

## &lt;Description&gt;

Basically telnet/web in SVMi-20i is disabled from V4.98. But you can enable them in DM 5.2.10 System IP options – SVMi Enable – TELNET/FTP

## &lt;Related Features&gt;

If you want to use some options as below, you have to enable this option

DM 8.6.1 Voice Studio

DM 8.7.1~3 User, Error, Activity log

DM 8.7.5 Subscriber import

DM 8.7.6 Backup and Restore

**13. [-/ALL/ALL] IP white List for telnet/ftp in SVMi-20i is applied for security**

## &lt;Description&gt;

IP white List for telnet/ftp in SVMi-20i is applied. You can set it in DM 5.13.11 Management IP White List

**14. [-/ALL/ALL] SIP phone user can Set or Cancel Message to extension excluding SIP phone. (Requested by USA)**

## &lt;Description&gt;

SIP phone user can Set or Clear message to extension except for SIP phone or MSG group.

SIP Phone user can dial SETMG or CANMG feature code + extension number.

SIP Phone user can dial SETMG or CANMG feature code + MSG group number.

**15. [#TE16403Z/ ALL/ALL] 'Send 20ms ptime' feature (Requested by UK)**

## &lt;Description&gt;

You can set 'Send 20ms ptime' feature in DM 5.2.12 SIP Stack/Ext/Trunk Options .- SIP Trunk Configuration – Send 20ms ptime

If you set this option, all of frame count in OfficeServ set to 20ms.

And 20ms ptime is added in SIP SDP message.

**16. [-/ALL/ALL] OAS Card Network Error Recovery**

## &lt;Description&gt;

In case that network between MP Card and OAS card is disconnected for 2 minutes, MP reboot OAS Card automatically for network link recovery.

**[Bug Fix]**

1. **[-/ALL/ALL] LCR reroute failure when DNS Query for SIP trunk is failed (Reported by Australia)**  
<Description>  
Set 1<sup>st</sup> LCR trunk to SIP trunk and 2<sup>nd</sup> LCR trunk to PRI trunk or other trunk. Set outbound proxy to URL and set DNS Server to IP address.  
In case of DNS Query for SIP Trunk is failed, LCR call cannot reroute 2<sup>nd</sup> trunk.  
<Cause>  
DNS Query failure was not considered for LCR reroute.  
<Resolution>  
In case of DNS Query for SIP Trunk is failed, LCR call can reroute to 2<sup>nd</sup> trunk.
2. **[#TE16404E/ALL/ALL] There is a one-way speech when uncondition station group call from trunk overflows to sip phone in uncondition station group including 2 sip phones or more. (Reported by UK)**  
<Description>  
There is one-way speech when uncondition station group call from trunk overflows to sip phone in uncondition station group including 2 sip phones or more.  
<Cause>  
In this case, common MGI channel for sip phone in uncondition station group is released.  
<Resolution>  
Fixed releasing common MGI channel for sip phone in uncondition station group
3. **[#TE164004/ALL/ALL] There is a call drop when receiving '480 temporarily unavailable' message from SIP phone in uncondition station group. (Reported by Italy)**  
<Description>  
You make a call to uncondition station group including sip phone and sip phone send '480 temporarily unavailable' message. Then this call is dropped.  
<Cause>  
In case that sip phone in uncondition station group send '480 temporarily unavailable' message. There is a call control error.  
<Resolution>  
You make a call to uncondition station group including sip phone and sip phone send '480 temporarily unavailable' message. Then this call continue and other phones are ringing.
4. **[#TE16603L/ALL/ALL] Allowing RFC2833 payload type 101 (Reported by South Africa)**  
<Description>  
When voice mail receive 96 telephone-event in SIP SDP, OfficeServ only allow rtpevent including payload type 96.  
In case that SIP service provider send 96 telephone-event in SIP SDP and rtpevent including payload type 101, OfficeServ don't detect rtpevent including payload type 101  
<Cause>  
This is modification of V4.92 for particular SIP service provider.  
<Resolution>  
Do not support RFC2833 rtpevent including payload type 96.
5. **[-/ALL/ALL] Intermittent message loss from MP to OAS (Reported by Australia/UK)**  
<Description>  
There is a possibility of intermittent message loss from MP to OAS  
<Cause>  
There is a possibility of intermittent message loss from MP to OAS when MP send message to OAS and OAS send message to OAS at the same time.

<Resolution>  
Re-transmission scheme is added.

**6. [-/ALL/ALL] Distorted ringback tone ((Reported by UK)**

<Description>  
A SIP Trunk incoming call go to SIP trunk (tandem). In this condition, OfficeServ receives 183 and 180 at once. And call originator can listen distorted ringback tone

<Cause>  
A SIP Trunk incoming call go to SIP trunk (tandem). When OfficeServ receives 183 and 180 at once. OfficeServ send 183 and 180 to SIP trunk.and send 2 RTP stream from MPS and RTG. This can make distorted ringback tone at call originate side.

<Resolution>  
In this case, OfficeServ don't handle 180 w/o SDP after 183 w/ SDP or 180 w/. This means OfficeServ only send 1 RTP stream from MPS to SIP Trunk (call originator).

**7. [-/ALL/Russia] DS button display of SDM is wrong in Russian (Reported by Russia)**

<Description>  
DS button display of SDM is wrong in Russian.

<Cause>  
DS button display of SDM is wrong in Russian.

<Resolution>  
Change Russian character. **Вн.Тлф -> Вн.**

**8. [-/ALL/ALL] DSS key for 3rd party SIP extn locks up the keyset. (Reported by Australia)**

<Description>  
Incoming ISDN call goes to SIP extension and other phone pick up the call using DSS key. After that, SIP extension locks up.

<Cause>  
In this case, there is an error of call control.

<Resolution>  
An error of call control is fixed.

**9. [-/ALL/ALL] No transfer tone after hunt time timed out. (Reported by Australia)**

<Description>  
Set DM 5.14.1 Transfer Ring Back MOH to On  
Make a SIP Trunk call and talk to each other. Transfer the call to hunt group.  
There is no transfer tone after hunt time timed out.

<Cause>  
In this case, RTP is released for transfer MOH

<Resolution>  
In this case, RTP is used for transfer MOH continuously

**10. [#TE16701K/ALL/ALL] Absent message in SMDR isn't printed out when setting it from UC dial on SMT phone. (Reported by UK)**

<Description>  
Absent message in SMDR isn't printed out when setting it from UC dial on SMT phone

<Cause>  
When setting absent message using UC dial on SMT phone, OfficeServ don't send SMDR message.

<Resolution>  
When setting absent message using UC dial on SMT phone, OfficeServ sends SMDR message.

- 11. [-/ALL/ALL] Intermittent DM connection lockup after receiving SNMP Get Bulk Message. (Reported by Korea)**

<Description>  
After receiving SNMP Get Bulk Message DM connection is failed.

<Cause>  
There is an error of memory management when receiving SNMP Get Bulk Message..

<Resolution>  
An error of memory management is fixed.
  
- 12. [#TE16703S/ALL/ALL] There is a one-way speech when using headset and speaker key on IP phone. (Reported by Australia)**

<Description>  
DM 2.1.5 System Options – VoIP RTP Option – SIPT Ringback Message set to 180. Incoming SIP Trunk call goes to IP phone which is set to headset mode. And IP phone answers the call by using speaker key. In this case, IP phone user gets one-way speech.

<Cause>  
In this case, there is an error of t-switch control

<Resolution>  
An error of t-switch control is fixed
  
- 13. [#TE164021/ALL/ALL] MPS don't send packet to opponent. ((Reported by UK)**

<Description>  
MPS don't send packet to opponent in a specific call scenario as below. Incoming SIP trunk Call goes to uncondition station group including SIP phones. One SIP phone rejects the call and other phone picks up the call.

<Cause>  
In this case, there is an error of MPS control

<Resolution>  
An error of MPS control is fixed

16.03.08 V4.95

※ **DB Compatibility**

- ※ **The old s/w cannot recognize the new nand. If the old package is upgrade to the new system, the booting is failed. But The V4.95 s/w has a compatibility on the old and new system.**

**[Major Changes]**

**1. [-/OS7100/OS7200s/ALL] The New Nand Device id was added.**

<Description>

NAND Die was changed. (D die → E die). The old version and the new system doesn't become a compatible. If the old package is upgrade to the new system, the booting is failed.

**[Bug Fix]**

**14. [-/Linux/ALL] System restart after set DM 6.1.4 System Alarm Mail Server Info**

<Description>

System restart after set Send Day and Send Major Alarm Immediately on DM 6.1.4 System Alarm Mail Server Info because of failure of socket control

<Cause>

In this case, there is a failure of tcp socket control

<Resolution>

checking tcp socket validation.

**15. [-/ALL/ALL] Crosstalk issue or one way speech problem when transferred call is recalled to IP phone.**

<Description>

IP phone which is a member of uncondition group member answers incoming uncondition group call and does a blind transfer to another phone. IP phone answers new incoming uncondition group call. Nobody answers the transferred call and the transferred call is recall. IP phone user finishes current call and answers the recalled call. In this case one way speech problem occurs. A crosstalk can be generated, because a used mgi channel can be allocated by the other port,.

<Cause>

In this case there is a MGI channel control error

<Resolution>

MGI channel control error is fixed.

**16. [-/ALL/ALL] Average Call per week in 8.11 Status Screen is always 0.**

<Description>

Average calls per week in DM 8.11 Status Screen is always 0.

<Cause>

The start time for this statistics is fixed to 2006. So total calls divided with a big number.

<Resolution>

Start time of reporting will be assigned with current time when SVMi-20i starts if the configuration(block table) is default.

<Ref>

We recommend to clear VM/AA report count with DM 8.7.7 when you install the system.

**17. [-/ALL/ALL] If Registrar Address is set, Proxy Domain Name is not used at host name of REGISTER**

<Description>

If Registrar Address is set, Proxy Domain Name is not used at host name of REGISTER (in DM 5.2.13)

<Cause>

A host name of REGISTER is filled with Registrar Address always although Proxy Domain Name is set

<Resolution>

If Proxy Domain Name is set, a host name of REGISTER is filled with Proxy Domain Name

**18. [-/ALL/ALL] The DNS query was not working when setting the DNS server in DM5.2.13**

**SIP Carrier Options.**

<Description>

In case DM 2.1.5 'Multi DNS Server' option was enabled and the DNS server was changed, DNS Query was not working.

< Resolution >

If changed the DNS Server Address in DM 5.2.13, OfficeServ is rebooted automatically. The DNS query operation is working after system restart.

**19. [-/MP20s/ALL] Not supporting the function detected the IP collision.**

<Description>

We revise so that the related function doesn't be supported in OS7200s

Because the issue which the capacity is related compared with MP10a can happen, MP20s deletes a function

< Resolution >

In order to confirm the IP collision by using the ping or the data equipment.

15.06.30 V4.91 / 15.05.15 V4.90

※ **DB Compatibility**

- The database of V4.91 is not compatible with that of a previous version including the V4.90 because of System Call Log feature.
- So you download the database of the existing system before upgrading to V4.91 using DM.
- After upgrading to V4.91, upload the downloaded DB by DM.

**[New Feature]**

2. **[-/ALL/ALL] Supporting New IP Phones, SMT-i6021/6011**

SMT-i6021 and SMT-i6011 phones can be connected to V4.90 system and this version supports below new feature.

- BT (Bluetooth) Feature for SMT-i5343 and SMT-i60XX
- Syslog for SMT-i60XX

Syslog options are added to MMC840/841 or DM 2.7.1/5.2.10 for phone trace.

If you enable syslog options, these options are sent to the phone and phone will save its trace data to the local directory or send it to the remote server.

For more details, refer to the below description.

MMC841 Syslog	Control	System Syslog (All phones will follow MMC841 Syslog Mode & Level when controlling its trace data.) Phone Syslog (Each phone will follow its MMC840 Syslog Mode & Level when controlling its trace data.)
	Mode	Disable (do not save phone trace) Enable (save phone trace or send it to the remote server)
	Type	Local Store (save phone trace inside the phone) Remote Send (send phone trace to the remote server)
	Server Address	Input Remote Server IP or Domain.
	Port	Input Remote Server Port
	Level	Select phone trace level (Emergency/Alert/Critical/Error/Warning/Notice/Info/Debug)
MMC840 Syslog	Mode	Disable (do not save phone trace) Enable (save phone trace)
	Level	Select phone trace level (Emergency/Alert/Critical/Error/Warning/Notice/Info/Debug)

- Hook Off/On Time for SMT-i60XX

IP phone gets Hook Off/On time by provisioning to prevent a handset hook off/on decision error. As new phones' handset is easy to lift, a call is disconnected frequently whenever user touches the handset by mistake.

- New options for SMT-i5343 and SMT-i60XX

. DM 5.15.17 ITP Options: Use Hotspot, Use Bluetooth, Use USB, Use NFC

. DM 5.2.11 DSP Parameters: Roaming Trigger, Roaming Delta, Roaming scan period

. DM 5.2.31 AP Common Option

: Roaming Scan Channel option is enabled for SMT-i5343 and SMT-i6021/6011.

AP searches the roaming scan channel that is set to on in DM 5.2.31. So set DM 5.2.31 by following your AP's usable channel setting.

- Support New Codecs (AMR, AMR-WB) for SMT-i5343 and SMT-i60XX

. SMT-i5343: It supports both AMR and AMR-WB.

. SMT-i60XX: It supports only AMR-WB.

3. **[-/ALL/ALL] MP Trace Tool**

User can check and save trace data from remote with MPT Tool.

If you want to connect over public IP address, you should open telnet port 23 and TCP port 5030. And in case of OS7400 and OS7200, you should add mptTool.jnlp, mptTool\_public.jnlp and mptTool.jar files to /DM directory in SD card.

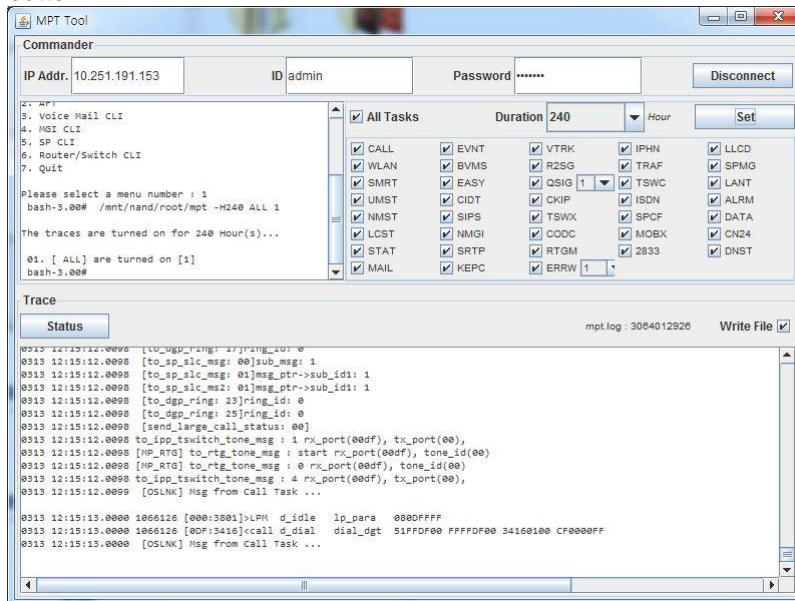
For more details, refer to the below description.

- Input PC IP Address to Management IP White List in DM 5.13.11.

- Input below URL on the web browser.

<Linux System – OS7100, 7070, 7030, MP20s>

- . <http://system> ip address/mptTool.php (regardless of private or public ip)
- <VxWorks System – OS7400, OS7200>
- . <http://system> ip address/card/mptTool.jnlp (private ip)
- . <http://system> ip address/card/mptTool\_public.jnlp (publid ip)
- When MPT Tool is shown, input password and click [Connect] button.
- <Linux System>
- . ID/PSWD: admin/default telnet password
- <VxWorks System>
- . ID/PSWD: kpsw/default telnet password
- Select Task and Duration that you want to trace and click [Set] button.
- You can see trace data is printed out. If you want to save these data to file, click [Write File] button.



#### 4. [-/ALL/ITALY] CID Display without Name

<Description>

When system receives CID Information without Name, “CID NO Name” message is always shown on LCD. SEI requests that user can select option to determine this message is shown or not.

<Resolution>

Add CID No Name Send option to DM 2.1.5 (or MMC861).

If you enable this option, “CID NO Name” message will not be displayed on LCD.

#### 5. [18099/ALL/ALL] SIP Station Call Waiting Feature

<Description>

SEUK requests to add SIP station call waiting feature.

<Resolution>

System will allow user to answer 2nd call if Call Waiting of DM 2.7.2 is enabled.

※ WE VoIP can't use Call Waiting feature.

#### 6. [-/Linux System/USA] Hotel feature implementation

<Description>

SEA requests to add Hotel feature to Linux systems.

<Resolution>

Hotel feature is enabled in Linux systems and related MMC and DM menu are also shown as MP40 and MP20.

#### 7. [-/ALL/AUSTRALIA] MCID during Barge-in or Conference

<Description>

PC requests to add MCID feature during Barge-in or Conference.

<Resolution>

MCID key is activated during Barge-in or Conference.

#### 8. [-/ALL/ALL] IP Conflicion Alarm

When system IP is conflicted, the administrator is able to check the Mac address of related devices in DM 6.1.1 Alarm History.

(Ex) DM 6.1.1 Alarm History for IP Confliction

ID	Date	Time	Alarm Name	Position
MJA40	04/29/2015	15:00:00	00001632F0FA	IP Conflict

DM 6.1.1 Alarm History for IP Confliction Recovery

ID	Date	Time	Alarm Name	Position
MJA41	04/29/2015	15:30:00	IP Conflict Rec	IP Conflict

**9. [-/ALL/ALL] Support Payload 96 of RFC2833**

In UK, some SIP service providers use payload 96 instead of payload 101.

Modified VM can detect DTMF with payload 96.

**10. [-/ALL/ALL] Support TCP connection of SMT-iXXXX phones**

<Description>

In some sites, IP phone messages can be reversed due to UDP connection. For preventing such a case, system needs to support TCP connection.

<Resolution>

System supports TCP connection.

Set MMC840(or DM 2.7.1) SIG TYPE to TCP if you want to use TCP mode.

※ It is not recommended as possible because system load will be high when selecting TCP mode. So you have to consider system load when you select TCP mode.

**11. [-/ALL/ALL] Support FTP Passive Mode**

<Description>

Some sites request to add FTP Passive Mode due to below reason.

When PC is under NAT circumstance, user can't connect to the system with DM. As DM tries to connect to the system by active mode, firewall blocks this connection. To solve this, DM should try to connect to the system by passive mode.

<Resolution>

Add FTP Passive Mode for DM user under NAT to be able to connect to the system.

You can check this modification by setting your PC under NAT and trying to connect to the system with DM. There's no additional setting for this feature.

**12. [-/ALL/ALL] Upgrade Vsftpd version to 3.0.2**

<Description>

We upgrade Vsftpd version from 2.X.X to 3.0.2 because previous Vsftpd version 2.X.X has PCI Compliance problems.

**13. [-/ALL/ALL] System Call Log**

<Description>

System saves essential call log in real time and user can check this log in DM.

This log will be cleared when system restarts.

There's no setting for this feature but for normal operation use DM V4.91 or later.

- Boot the system.

- Make several calls and check call log in DM 6.7.1.

- Make SIP calls and check SIP call log in DM 6.7.2.

(SIP log is saved only when there are some errors. Due to this reason SIP log might not be saved in normal case.)

**14. [-/MP20s,MP10a/ALL] Package upload specification was changed**

<Description>

The software uploading method to SD card is changed because of insufficient memory.

As Is: The package is written to dram memory first and then copied the files to SD card.

To Be: The package is written to SD card directly without copying to dram memory.

Like before, please do not remove the SD card while DM uploading the package to SD card.

**[Bug Fix]**

**20. [-/MP20s/ALL] CNF24 Silence on answer**

<Description>

Approximately every 1-2 months CNF24 card fails with No Voice when call is answered.

<Resolution>

We can't find root cause so we suggest workaround that reset CNF24 card every Sunday 4:15.

In order to reset CNF24 card periodically, you have to set “CNF24 Reset for Maintenance” in DM 2.1.5.

**21. [-/ALL/UK,USA] System stops sending SMDR traffic out when enabling Ring Time in DM 5.6.3.**

<Description>

When you enable “Ring Time” in DM 5.6.3, system will stop sending SMDR traffic out.

<Cause>

If you enable “Ring Time”, total size of SMDR data will be increased and it is bigger than system’s max SMDR buffer size. Because of this system restricts sending SMDR data out.

<Resolution>

Increase system’s max SMDR buffer size and modify not to check total size of SMDR data when sending SMDR data out.

**22. [-/MP20s/USA] The scheduled time and the time on the email are off by two hours.**

<Description>

With V4.75 software, when we schedule a meet-me conference with CNF24 and receive an email notification, the scheduled time and the time on the email are off by two hours.

<Cause>

In case of countries like USA that use –X:00 time zone, system sends an email notification including wrong time zone offset. Because of this outlook calculates and displays the start and end time wrong.

<Resolution>

System sends right time zone offset range when current time zone is –X:00.

**23. [-/MP20s/ALL] System restart on CNF subscriber name search**

<Description>

Create a subscriber “andrew.truong” and a new meet-me conference. When “Select Conference Member” window is displayed, enter “andrew.truong” and select search button. Then system restarts.

<Cause>

System restarts when user enters the subscriber name that its length is over 11.

-> When searching subscriber name, system searches DM 9.1.5 Meet-Me Members List first and if it can’t find matching name, then it searches DM 2.4.2 Station Name List.

But even though system finds the name in Meet-Me Members List, it searches the name in Station Name List unnecessarily. And during searching Station Name List, system restarts due to buffer overflow as max length of station name is 11.

<Resolution>

If system finds the name in Meet-Me Members List it will not search Station Name List. And if system searches the name in Station Name List and length of entered name is over 11, it will search only 11 lengths of name to prevent system restart.

**24. [-/ALL/ALL] Inquiry for improper MGI channel assignment**

<Description>

MGI channel is used improperly during the incoming call to a station group with a mix of SMT-1 and 3rd party SIP phones.

<Cause>

When several IP and SIP phones are assigned as a member of Unconditional Group, system needs at least 2 MGI channels. Because one is assigned to SIP phone before answering and another is assigned to IP phone for preparing IP phone answer. But if system only has 1 MGI channel and call is incoming to unconditional group, IP phone can’t answer the incoming call as there is no available MGI channel.

<Resolution>

System will use only 1 MGI channel if call is incoming to unconditional group that has both IP and SIP phones.

**25. [-/ALL/ALL] Incoming FAX call failure using T.38**

<Description>

User can’t receive incoming Fax call when connecting to particular SIP Server

<Cause>

Some SIP Servers send T.38 reINVITE without receiving T.38 reINVITE that OfficeServ sends. So set T.38 Reinvite option to Not Send in DM 5.2.13 to solve this issue. But there is a media control error when receiving T.38 reINVITE from SIP Server and for this reason incoming FAX call fails.

<Resolution>

Media control error is fixed when T.38 Reinvite option is set to Not Send in DM 5.2.13.

**26. [-/ALL/ALL] Abnormal MGI status**

<Description>

User of SIP phone called 2N Helios makes the 1st call and makes the 2nd call before the opposite party answers the 1st call. Then both MGI and SIP extension are blocked in the Connection Status Table.

<Cause>

System does not consider below exception case.

<Resolution>

System will cancel the 2nd call if user of SIP phone makes the 2nd call before the opposite party answers the 1st call.

**27. [-/ALL/ALL] WE VoIP Display issue**

<Description>

Corrupted Name displays on the mobile when extension/Group numbers have 11 characters in the name.

<Cause>

System sends wrong size of station name.

<Resolution>

System will send correct size of station name for normal display.

**28. [88842/ALL/ALL] Conference on SIP phone using feature code does not work**

<Description>

There is no voice issue when making a conference with feature code on SIP phone.

<Cause>

System has a media control error when making a conference with feature code on SIP phone.

<Resolution>

Media control error is fixed.

**29. [-/ALL/ALL] PNP Mac addresses are incorrectly assigned after system upgrade**

<Description>

After upgrading the system from V4.60b to V4.80, incorrect Mac addresses are shown in DM 2.7.1.

<Cause>

When upgrading to V4.80, system DB is cleared and PNP mode of DM 5.2.10 is also changed to Auto PNP. But in this case IP phone whose mode is PNP is registered to the system and user ID is changed to the received IP phone Mac address..

If PNP Mode is changed, system will reboot.

<Resolution>

System will clear internal flag before saving DM 2.7.1 DB to show normal user ID.

**30. [-/ALL/RUSSIA] SMT-I phone book**

<Description>

Phone book name is not displayed normally when setting system code page to WIN-1251.

<Cause>

When saving phone book name to XML file, system sets display charset to euc-kr regardless of language type and encodes the name with system code page.

<Resolution>

System will encodes xml file with koi8-r and also set file tag with koi8-r.

**31. [-/ALL/ALL] Change Remote Port of SIP Peering incoming call**

<Description>

User can't hold or resume the call when DM 5.2.17 remote port and remote port of SIP peering incoming call are different.

<Cause>

System sends request message (ex. re-INVITE or BYE) to DM 5.2.17 remote port but the opposite system expects those messages to be received into its own remote port that is different from DM 5.2.17 remote port.

<Resolution>

If SIP peering call is incoming from the port that is different from DM 5.2.17 remote port, system will send request message to that port until current call is disconnected.

**32. [-/ALL/ALL] Change VM/AA Announcement**

<Description>

If VM admin password is 0000 or blank, system does not allow user to login and in this case he/she hears VM/AA announcement such as "Invalid entry, try again". But this announcement is not appropriate for such a case.

<Resolution>

User will hear "Current password not set" if VM admin password is set to default.

33. **[-/ALL/ALL] Hotel Billing to Printer**

<Description>

Intermittently the printer prints over 30 pages for the 1 x room bill repeating the bill details.

<Cause>

Sometimes billing information is used as it is not initialized.

<Resolution>

System will delete billing information and not print such an invalid data if it is out of normal range.

34. **[-/ALL/ALL] Ringback tone is not heard when making ISDN trunk call on SIP extension**

<Description>

From a sip extension if I dial a geographic telephone number, my sip handset does not play ringback.

<Cause>

System can't send 180 message to SIP phone because ISDN network does not send Alert message.

<Resolution>

System will send 180 message to SIP phone if it succeeds to seizure ISDN trunk even though ISDN network does not send Alert message.

35. **[91544/ALL/ALL] Easyset Programmed Message not working with 4.60 or 4.75**

<Description>

Easyset Programmed Message is not working with 4.60 or 4.75.

<Cause>

Programmed message flag is used as it is not initialized.

<Resolution>

Initial programmed message flag.

36. **[-/ALL/ALL] SIP Extensions locking up**

<Description>

SIP phone makes a Loop trunk call and then hold current call. In this case if loop trunk call is disconnected due to busy tone detection, SIP phone is blocked.

<Cause>

When disconnecting loop trunk call, system does not send clear message to SIP phone.

<Resolution>

System will send clear message to SIP phone if loop trunk call on hold is disconnected.

37. **[P140814001/ALL/ALL] Call is discontinued when SVMi-20E answers the call**

<Description>

Call is discontinued when SVMi-20E answers the call.

- OS7100 SLT makes a call to DGP.

- No Answer Forward is set on DGP. So if DGP receives a call, the call is forwarded to OS7400 VM through SPNET.

- In this case call is disconnected.

<Cause>

If Power Down is set on SLT, system disconnects the call that SLT makes due to power down feature.

<Resolution>

Power down feature will not be activated if SLT makes a call through SPNET.

38. **[21545/ALL/ALL] Problem with SIP extension clear down.**

<Description>

DGP and SIP phone are members of Unconditional Group. If unconditional group receives an incoming call and DGP answers the call, SIP phone stops ringing after a period of time.

<Cause>

If DGP answers the call before sending INVITE to SIP phone, system can't send clear message to SIP phone.

<Resolution>

System will not send INVITE to stop ringing when it has to send BYE to SIP phone before sending INVITE.

**39. [95519/ALL/ALL] OS7100 SYSTEM LOCKUP**

<Description>

System locks up if the number of SPNET incoming calls is bigger than total number of MGI channels.

<Cause>

System repeats searching the operator when there is no available MGI channel and due to this operation system can't control the other calls.

<Resolution>

System will disconnect SPNET calls if there is no available MGI channel.

**40. [100399/ALL/ALL] TCLIP does not work over SPNET with Tandem Trunking**

<Description>

TCLIP does not work over SPNET with Tandem Trunking.

<Cause>

TCLIP feature is not considered when making a SPNET call.

<Resolution>

Add TCLIP feature in case of SPNET call.

**41. [P141028001/ALL/ALL] LNR can't be assigned to IP Set key**

<Description>

LNR key can't be assigned to IP Set key.

<Cause>

LNR and MOVE fixed keys are added to the previous Fixed Key Table for SMT-i5343.

Because of this system restricts setting LNR key to all IP phones.

<Resolution>

Separate Fixed Key Table into two tables. One is for previous IP phones except SMT-i5343 and another is for only SMT-i5343 that has LNR fixed key.

**42. [-/ALL/ALL] MP20s Slow Operation**

<Description>

Operation of IP phones repeats stop and start several times a day.

<Cause>

If remote site disconnects TCP port 6000, IP phone interface task loops itself as system does not close socket connection normally.

<Resolution>

Modified to close socket connection normally when TCP port 6000 connection is closed.

**43. [-/ALL/ALL] Change the time for returning IDLE state when user hooks off the handset of DGP/IP phone.**

<Description>

It takes 3 minutes to return IDLE state when user hooks off the handset of DGP or IP phones.

It needs to be reset the time because current time is too long.

<Resolution>

Change the return time from 3 minutes to 1 minute.

**44. [-/ALL/ALL] Make up weak point in glibc security (ghost)**

<Description>

A weak point in glibc security is found and it needs to be modified.

<Resolution>

This weak point is found in case of DNS query. So change the function from gethostbyname to getaddrinfo to make up weak point.

**45. [-/ALL/ALL] The Absent message not work with Action MMC115 feature.**

<Description>

In below case forward does not work because forward information is deleted

- In MMC115 set programmed message and forward information.

- Set forward information in MMC102.

- Move jog button to set Forward on SMT-i5210.

- Move jog button to cancel Forward on SMT-i5210.

- Move jog button to set Programmed message on SMT-i5210.

<Cause>

Forward information is deleted when you set programmed message by using Jog button.

<Resolution>

Modified not to delete Forward information when you set programmed message by using Jog button.

**46. [-/ALL/ALL] Support SSL and TLS in case of sending Alarm Email**

<Description>

System needs to support SSL and TSL because some email servers only support SSL or TLS.

<Resolution>

Add SSL and TLS options to DM 6.1.5 to support those connection types.

**47. [-/ALL/ALL] Add Audio Stop Send option for IP phones**

<Description>

System always sends Audio Stop message before sending Audio Start message. But when current network is unstable, message can be reversed and no voice can occur.

<Resolution>

System will not send Audio Stop message before sending Audio Start message if DM 2.1.5 "IP Phone Audio Stop Send" is set to Disable. (default: Enable)

**48. [-/ALL/ALL] Change default value of Use First Codec**

Default value of Use First Codec in DM 5.2.12 and DM 5.2.13 is changed to ENABLE.

**49. [-/ALL/ALL] SIP phone can't be shown in DM 5.11.1 Hotel Room Type**

<Description>

SIP phone can't be shown in DM 5.11.1 but it is shown in MMC221.

<Cause>

SIP phone type is not considered in DM 5.11.1.

<Resolution>

Add SIP/WIP/S0 phone types to DM 5.11.1 to be shown.

※ You can set SIP/WIP/S0 phones as Guest Room but in this case user can't use hotel features with dialing feature code and he/she can only make and receive the call.

**50. [-/ALL/TURKEY] Can't make a PRI Overlap call**

<Description>

User can't make a PRI call when its dial mode is set to Overlap.

<Cause>

When station group receives the incoming trunk call and the system overflows the call to next port after overflow time is expired, system refers the uninitialized parameter instead of next port. Because of this reason, DTMF sending task stops and it can't send DTMF to ISDN network.

<Resolution>

System will refer next port instead of uninitialized parameter.

**51. [-/OS7070/ALL] Multicast Page**

<Description>

In OS7070 multicast page does not work.

<Cause>

System controls wrong MGI channel when using multicast page.

<Resolution>

Modify to send right MGI channel number.

**52. [-/ALL/ALL] VM Ringback Tone improperly uses the VMS license**

<Description>

- In DM 2.1.4 install a License Key that enable 1 VMS channel (It is very important that only one VMS channel is enabled.)

- Configure multimedia ringback tone for a station group.

a. In DM 2.4.2 enable "Distinct VM ringback" for BRI lines 701 and 702

b. In DM 2.6.2 enable "Color Ring As" for BRI lines 701 and 702

c. In DM 4.1.1 check group 507 (AAOnly) and group 509 (BI-VMS) are assigned.

d. Through the DM 5.13.1 assign as "RTB Source" the group 507

e. Through the DM 4.1.1 create the group 501 with members the 201 (the number of a SLT port) set the "RTB Message"= '5000' (5000 is an existing message)

f. Through the 'DM 3.2.1 (MMC 406) assign incoming calls for 701 and 702 lines to destination 501 group.

- Make an incoming call, then caller hears the RBT properly.

- In the meantime the call of point 4 is active, from any extension call the VM group 509 to access to its own mailbox. Then you listen the ringback tone instead of the VM answers.

<Cause>

System assigns 1 VM license to VM channel for ringback service.

<Resolution>

When controlling VM license, system will not assign VM license to VM channel for ringback service.

**53. [-/ALL/ALL] SIP trunk-trunk transfer is failing**

<Description>

When AA receives SIP trunk call, AA transfers the call to virtual SLT and disconnects before SLT answers. In this case incoming call to virtual SLT should be forwarded to another SIP trunk as no answer forward is set but this SIP trunk call fails.

<Cause>

Call fails because system does not send ACK for received 200 OK after sending reINVITE. There's an internal message handling error.

<Resolution>

Modified internal message handling error to send ACK for received 200 OK.

**54. [-/ALL/ALL] I/C SIP over SPnet no RTG**

<Description>

There's a no voice issue for SPNet call when two systems are set as below.

- Set two systems.

- Change IP Type to Private with Public in DM 2.1.2.

- Input Private and Public IP in DM 2.1.0 and 2.1.2.

In this case two systems use different public IP and Subnet.

- Input opposite system's Private IP in DM 5.2.8.

- Input opposite system's Private IP to SPNET IP Address in DM 3.3.1.

<Cause>

There's an error when checking opposite system's Private IP. Because of this error system checks Private IP as Public IP.

<Resolution>

Modified Private and Public IP check routine.

**55. [-/Linux/ALL] VM CLI system freeze issue**

<Description>

System freezes when user connects to the system by telnet, enters VM CLI and input "endvmcli" command.

<Cause>

Start module of VM CLI is stuck in while loop when it receives "endvmcli" command.

<Resolution>

Modified not to be stuck in while loop.

**56. [-/ALL/ALL] Transfer of Video SIP call**

<Description>

There are two errors when system is set to Private with Public and SIP phone & SIP Peering Server are located in Private Network.

- Video is not shown normally when video SIP phone makes a call to SIP Peering Server.

- When two Video SIP phones are connected and user transfers the call to SIP Peering Server, transferred SIP Phone and SIP Peering Server can't receive Video information.

<Cause>

- When sending Video SDP to SIP Peering Server, system sends public information instead of local information.

- System sends wrong Video information of SIP Peering Server to transferred SIP Phone.

<Resolution>

Modified above two errors.

**57. [-/ALL/ALL] SIP with VMS ringback**

<Description>

VM ringback is not heard in the below cases.

- When station group receives SIP trunk call, caller can hear VM ringback after VM Ringback Delay time (DM 5.5.0) but can't hear after Hunt time.

- When IP phone receives SIP trunk call, caller can't hear VM ringback tone.

<Cause>

- System releases MGI channel when Hunt time is expired.
  - System does not play VM Ringback because it misjudges that IP phone uses RTG instead of MGI channel.
- <Resolution>  
Modified above two errors.

**58. [-/ALL/ALL] Multi Transfer Call Drop**

- <Description>  
Intermittent call drop-offs when they made calls between sites
- <Cause>  
- When System receive re-INVITE of "SendReceive" type for transfer or hold. System can't work well it. then System doesn't response next re-INVITE. so It makes call-drop.
- <Resolution>  
Modified above an error.

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**14.08.07 V4.82**

**[Bug Fix]**

**1. [VM/-/ALL] Block table is broken when the disk is full**

- <Description>  
When the disk is full, it fails to save block table and it is broken. Then backup block table becomes broken, too. So, it does not work when it resets.
- <Cause>  
This does not occur in normal situation because SVM is designed to leave a small space of disk(over 100Mbytes). But if it's filled up by abnormal reason, it makes a problem. Block table cannot be saved. It's ok, but original block table is broken. Then backup block table is broken, too. So the system does not work, if it resets.
- <Resolution>  
Saving block table is modified.  
It will create a temporary file and save block table to it.  
If there is no problem, it will replace old block table with new one.

**2. [VM/-/ALL] SVM resets when it repeats to play the prompt does not exist.**

- <Description>  
SVM resets when it repeats hundreds of times to play the prompts does not exist
- <Cause>  
It's because the playback thread resource is lost when the prompt is gone.  
Resource leak is getting bigger and bigger while it repeats to play the prompt does not exist and the system is stopped or reset.
- <Resolution>  
SVM will check the file is really available before it plays and although it tries to play the prompt does not exist, it will retrieve the playback thread resources.

**3. [VM/-/ALL] DM 8.11 Status Screen and DM 8.9.3 Messaging Activity cannot show over 100,000 messages**

- <Description>  
If total message count is over 100,000, wrong number is played in DM 8.11 Status Screen and DM 8.9.3 Messaging Activity.
- <Cause>  
The buffer to save message count is overflowed.  
It's defined as 2 bytes, so it can handle max 65535. If it's overflowed, it displays totally wrong number.
- <Resolution>  
The buffer to save message count is extended to 4 bytes.
- <Consideration>  
The message format between SVM and DM is changed. It affects DM 8.11 Status Screen and DM 8.9.3 Messaging Activity.  
Previous version of DM(lower than V4.80) is not compatible with latest system software in DM 8.11 Status Screen.(It's not updated automatically)  
You have to use latest DM(V4.80 or later) for that pages.

**4. [VM/OS7030/ALL] If the disk is full of messages, OS7030 often resets when it starts**

## &lt;Description&gt;

If the disk is full of messages, OS7030 resets sometimes when it starts.  
So it takes a long time the service to be available.

## &lt;Cause&gt;

OS7030 rebuilds message table when it starts, it accesses the disk a lot. The file system of OS7030 is slower than other system, so the page cache is full, then the application stops until the cache data is processed.

So the system resets because it cannot trigger the watch dog timer.

## &lt;Resolution&gt;

When it rebuilds the message table, it will flush the cache after processing each file.

## &lt;Consideration&gt;

It's applied only for OS7030. It takes a little more time to boot when it has so many messages.

**5. [-/MP20S/ALL] SLI port's None Status on MP20S**

## &lt;Description&gt;

When Booting up with only SLI2, SLI3 or A-TRK cards in all slots, last several SLI ports' port status was changed to 'None'

## &lt;Cause&gt;

In this case, last several SLI ports were not initialized because of internal message queue overflowed.

## &lt;Resolution&gt;

Sending meaningless duplicated messages is eliminated.

**6. [-/ALL/ALL] Cannot delete DM IP White List in DM 5.13.9**

## &lt;Description&gt;

Cannot delete DM IP White List in DM 5.13.9

## &lt;Cause&gt;

IP comparison with IP 0.0.0.0 (means 'delete') had a problem.

## &lt;Resolution&gt;

IP comparison with IP 0.0.0.0 (means 'delete') is fixed.

**7. [-/ALL/ALL] System reset when using 'IP & Port' option in DM 2.7.2's NAT Check option**

## &lt;Description&gt;

After Setting DM 2.7.2's NAT Check option to 'IP & Port', when the SIP extension tried to register, System reset.

## &lt;Cause&gt;

Internal memory access error

## &lt;Resolution&gt;

This internal memory access error is fixed.

**8. [-/ALL/ALL] No Voice when SIP Station Group overflows into SIP Station Group.**

## &lt;Description&gt;

A call went to uncondition type's SIP extension Group and overflowed into uncondition type's SIP extension Group. When one SIP extension answered the call, there was no voice path

## &lt;Cause&gt;

Media resource (MGI or MPS) control error

## &lt;Resolution&gt;

This media resource control error is fixed.

**9. [-/ALL/ALL] No Voice when Auto Call Recording on SIP Extension**

## &lt;Description&gt;

When talk with SIP extension set auto call recording, there was no voice path

## &lt;Cause&gt;

Internal t-switch control error

## &lt;Resolution&gt;

Internal t-switch control error is fixed.

**[ Changes to improve SVM's security]****1. [-/ALL] "Force Subscriber to change PSWD" will be set when it's updated.**

## &lt;Description&gt;

To increase security, "Force Subscriber to change PSWD" option of DM 8.5 System Parameter is included in previous version. But it's not enabled when it is upgraded.

## &lt;Cause&gt;

It was not enabled not to affect current users using SVM system.  
It's enabled when the system is initialized.

<Resolution>

To increase security, it's changed to be enabled when it's upgraded from previous version.

**2. [!-!ALL] "Outcall Authorization" of EClass and MClass will be set to 'No' when it's updated.**

<Description>

To prevent toll fraud, "Outcall Authorization" of EClass and "Callback Authorization" of MClass is set to 'No' if it is initialized. But it's not changed when it is upgraded.

<Cause>

Not to affect current subscribers using SVM system, the outcall authorization setting is not changed when it's upgraded.

<Resolution>

To increase security, it will be set to 'No' when it's updated.

<Consideration>

All outcall feature for example, 'Follow me', 'Find me', 'Call back', 'Message alert', 'Direct call' etc will be blocked if the outcall authorization is set to No.

But it only blocks when the subscriber tries to set the features. If it's already set before it changes the setting, outcall feature will work. 'Direct call' will be blocked because you have to enter the phone number when you run it.

**3. [!-!ALL] "Edit stored telephone number" will be limited by "Outcall Authorization" of EClass.**

<Description>

"Edit stored telephone number" was not limited by "Outcall Authorization". It's used by "Find me".

If the subscriber is allowed to use "Find me" by an administrator(It's not allowed in default settings), the subscriber can set long distance telephone number and SVM will dial that number although "Outcall Authorization" is set to No.

<Cause>

Design error

<Resolution>

"Edit stored telephone number" will be limited by "Outcall Authorization" of EClass.

**4. [!-!ALL] "Operator" extension cannot use "Direct call".**

<Description>

"Operator" extension(500) is included in default configurations.

It's password is often not set, so it can be a security hole if the "No Messaging" EClass is allowed to make a outbound call.

<Cause>

"Operator" extension is quite same with the other extension, but no one changes the password.

<Resolution>

"Operator" extension is blocked to use "Direct call" regardless of "Outcall Authorization" of EClass.

**5. [!-!ALL] DTMF administrator is not allowed to access by pressing "#0000" when the admin password is not changed.**

<Description>

DTMF administrator can create a subscriber, record a prompt and change the operating mode, but it's easy to log on when the password is not changed.

<Cause>

The administrator can change admin password by DM 8.5 System Parameter, but it's not force to change it.

<Resolution>

It will not allow to access as dtmf administrator by "#0000" before it changes the admin password from the default one.

<Consideration>

"VMADM" key made by MMC 722 is allowed to access although the admin password is not changed. But we strongly recommend to change the admin password.

**6. [!-!ALL] Remote caller cannot change his password.**

<Description>

SVMi allows remote caller logs on his mailbox and uses it as if he is on his station. It can be a problem, when SVMi has an extension block that no one uses.

Somebody can log on it and he can make a outbound call if it's allowed.

<Cause>

SVMi downloads the number plan and creates extensions and mailboxes.

In this process, many unused extensions and mailboxes are created.

Somebody can log on because it's password is not changed.

<Resolution>

Remote caller is not allowed to change his password. It's only allowed in station call.

This modification can solve above security problem if it's used with "Force Subscriber to change PSWD".

When a subscriber log on his mailbox at first, he should change the password, but illegal remote caller cannot change it and log on.

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**14.06.05 V4.80****[New Feature]**

1. [-/ALL/ALL] Supporting New IP Phone, SMT-i5343  
Refers to 'OfficeServ system Programming guide for SMT-i5343.pptx' for more details.
2. [-/MP40, MP20, MP20S/ALL] SLI3 Card's Long Loop Check Feature is added.  
<Reason>  
When a port of SLI3 card detect long loop, the port was blocked and. You couldn't disable this feature.  
< Modified >  
SLI3 Card's Long Loop Check Feature can be disabled or enabled.  
If you select DM 2.1.5 SLI3 Long Loop Check option to Disable, SLI3 Card's Long Loop Check Feature is disabled. (Default value: Enable)

**[Bug Fix]**

7. [-/ALL/ALL] pickup Call's SMDR error  
<Issue>  
When picking up the call using VMS ringback tone and hanging up the call, SMDR's I Flag is not printed out.  
<Reason>  
When picking up the call using VMS ringback tone, SMDR information was not saved internally.  
<Modified>  
When picking up the call using VMS ringback tone, SMDR information is saved internally.
8. [-/ALL/Argentina] End of ANI (CID) of E1R2 signal is changed in Argentina  
<Reason>  
Requirement from Argentina.  
<Modified>  
End of ANI (CID) of E1R2 signal is changed from I-15 to I-12 only in Argentina
9. [P140327001/OS7000/ALL] mmc 856 Watchdog timer restart  
<Issue>  
Accessing MMC 856 with several entries but register not filled will cause a watchdog timer restart on the switch. With a keyset, access mmc 856 and scroll down to black line, hit the third soft key and the system will restart.  
<Reason>  
Accessing wrong memory address  
<Modified>  
Fixed accessing wrong memory address.
10. [P130604001/OS7000/ALL] Group Overflow Time  
<Issue>  
Incoming trunk call go to hunt group A and overflow VMS group. After that, the call goes to hunt group B from VMS. But the call is applied wrong hunt time.  
<Reason>  
Internal hunt count was not initialized after transfer to VM,.  
<Modified>

- Internal hunt count is initialized after transfer to VM.
11. [P130529003/OS7000/ALL] Call Wait display does not clear down  
<Issue>  
Station A, B,C are a member of uncondition type's station group. Station A talks to someone through SPNET. Incoming SIP Trunk call goes to this station group and Station A or B answer the call. In this case, station A's display show call waiting information continuously. This problem arises only in case of using SMT IP phone like SMT-i3100, SMT-i5120, SMT-5220  
<Reason>  
In this case, OfficeServ didn't send a message related to clearing call wait to SMT phone.  
<Modified>  
OfficeServ send a message related to clearing call wait to SMT phone..
  12. [-/ALL/ALL] SIP trunk Unsupervised transfer to WIP no speech  
<Issue>  
A station which is talking with incoming SIP Trunk makes a blind transfer to WIP-5120, there was no speech.  
<Reason>  
In this case, OfficeServ didn't send a message related to audio control to WIP phone.  
<Modified>  
OfficeServ send a message related to audio control to WIP phone.
  13. [-/ALL/ALL] Not creating user mailbox for security  
<Reason>  
After factory reset user mailbox is created automatically.  
< Modified >  
After factory reset user mailbox is not created automatically for security.  
DM 5.9.1 User mail box's default value is changed from Both to None.
  14. [-/ALL/ALL] Auto answer from voicemail  
<Issue>  
Although DM 5.15.2 Auto Answer CO option is Off, Auto Answer CO feature worked abnormally  
<Reason>  
Auto answer co feature didn't depend on Auto Answer CO option about trunk incoming call  
< Modified >  
In this case, auto answer co feature work or not as Auto Answer CO option is set or not
  15. [-/ALL/ALL] Wrong DS key's station number.  
<Issue>  
Set relocation between 2 stations and save DM 4.9.2 DS key related to set relocation. You can see wrong DS key's station number.  
<Reason>  
There was a port number error between OfficeServ and DM  
< Modified >  
The port number error is fixed.
  16. [-/ALL/ALL] Problem of receiving T.38 fax through H.323  
<Issue>  
OfficeServ cannot receive T.38 fax through H.323  
<Reason>  
When receiving T.38 fax, message related to T.38 fax was not sent to H.323 task.  
< Modified >  
When receiving T.38 fax, message related to T.38 fax is sent to H.323 task.
  17. [-/ALL/ALL] UCD final destination over SPNet tromboned call fails  
<Issue>  
When Incoming ISDN call is directed over SPNet and returns back on SPNet to UCD group, dialling final UCD destination results in call disconnect. The call must go to final destination.  
<Reason>  
There was a SPNet route optimization during listening UCD prompt. The SPNet route optimization made a wrong call flow  
< Modified >  
SPNet route optimization will be accomplished after answering the call.
  18. [-/ALL/ALL] Loss of RX on outgoing call

- <Issue>  
when a call, internal or external, to a voicemail AA(site A) is transferred by selecting an AA option to an SPNet extension(site B), which has forward all set back to an extension at site A, which is on an outgoing ISDN call then RX will be lost as off hook ringing begins.  
It only occurs when the call is transferred by voicemail AA
- <Reason>  
Wrong handling internal digit buffer made a voice path problem
- < Modified >  
Handing internal digit buffer was fixed.
19. [-/ALL/ALL] Open Mic after VT transfer
- <Issue>  
DID incoming call goes to station A and answer the call. The station A makes a supervised transfer to station B and do not hang up the call Station B do VT transfer to station C. The outside caller can now hear a open mic from an internal station in a idle condition.
- <Reason>  
VT transfer must works only after supervised transfer complete. But VT transfer works before not completing.
- < Modified >  
VT transfer doesn't work and is ignored before supervised transfer complete.
20. [-/ALL/ALL] WB codec of call between WE VoIP and WE VoIP
- <Issue>  
WE VoIP call to some extension except for WE VoIP, but WE VoIP answers the call by reason of Station Pair, Ring Group, Pick-up. The call between WE VoIP and WE VoIP cannot be made by using WB codec.
- <Reason>  
OfficeServ made a decision for codec from first destination
- < Modified >  
OfficeServ makes a decision for codec from final destination
21. [-/ALL/ALL] Consultative transfer call's wrong call duration time of SMDR
- <Issue>  
Consultative transferred party's call duration time is accumulated
- <Reason>  
Consultative transferred party's call duration time is accumulated
- < Modified >  
Consultative transferred party's call duration time is newly calculated after transferred party answers the call.
22. [-/ALL/Italy] WE VoIP's Italian language is added.
- <Issue>  
When set call forward or DND by using feature code from WE VoIP, confirm messages of the setting in WE VoIP show in English instead of Italian.
- <Reason>  
Some messages related to call forward or DND in WE VoIP cannot be displayed in Italian.
- < Modified >  
Some messages related to call forward or DND in WE VoIP can be displayed in Italian.
23. [-/ALL/ALL] SMT-i series' Missed Calls indicator
- <Issue>  
If it is missed calls occurs then message waiting indicator is flashing.  
Then if you visit call log menu and check unanswered calls list and back to the home screen then message waiting indicator continue to flash.
- <Reason>  
In this case, Officeserv didn't send led off message to SMT phone.
- < Modified >  
Officeserv send led off message to SMT phone.
24. [-/ALL/ALL] Internal Conference Ports Not Releasing
- <Issue>  
After the conference ports are being used, they are not releasing intermittently.
- <Reason>  
When OfficeServ receive abnormal message from DGP during conference, the conference resource is not released

< Modified >

When OfficeServ receive abnormal message from DGP during conference, OfficeServ releases the conference resource

25. [-/ALL/ALL] Voicemail audio cuts off with SVM IP Service enabled.

<Issue>

When Incoming call is directed over SPNET and returns back on SPNET to SVMi-20i in IP Service enabled condition, About ½ of the SVMi-20i's greeting plays and then we lose audio

<Reason>

MP does not send the message for voice path change to SVMi-20i when the SVMi-20i voice path is changed from IP to DTM by SPnet Route Optimize Feature

< Modified >

MP sends the message for voice path change to SVMi-20i when the SVMi-20i voice path is changed from IP to DTM by SPnet Route Optimize Feature

26. [-/ALL/ALL] Page from SIP extension

<Issue>

Page initialized from SIP extension seems to work but have no speech.

<Reason>

Page feature from a SIP extension is not supported

< Modified >

When SIP extension dials page feature code, OfficeServ send 404 not found message to SIP extension.

27. [-/ALL/ALL] Unifying some SIP option names in DM and KMMC

<Issue>

Option names of User Name, Auth User Name, Auth Password were different at each menu in DM or KMMC

< Modified >

Unifying option names of User Name, Auth User Name, Auth Password

- DM

DM No.	before	after	note
5.2.13	User Name Auth Username Auth Password	User Name Auth User Name Auth Password	
5.2.14	User Name Auth User Name Auth Password	User Name Auth User Name Auth Password	Not change
2.7.2	Tel Number User ID Password	Tel Number Auth User Name Auth Password	

- KMMC

KMMC No.	before	after	note
837	USER NAME AUTH USER AUTH PSWD	USER NAME AUTH USER AUTH PSWD	Not change
839	USERNAME AUTH UID AUTH PWD	USR NAME AUTH USR AUTH PWD	
842	- USER ID PASSWORD	- AUTH USER AUTH PSWD	

28. [-/MP40,MP20/ALL] MAC Address Recovery

<Issue>

Intermittently MP40 or MP20's MAC address is initialized after boot-up

< Modified >

In case that MAC address is initialized after boot-up, MAC address will be recovered.

29. [-/ALL /ALL] Trunk call's DND forward

<Issue>

Station A set to call forward to station B. Station B is DND and set to DND forward to Station C. When call from outside routes to station A, the call don't go to Station C but Station B.

<Reason>

- Checking DND forward's status have some problem.  
< Modified >  
Checking DND forward's status is fixed.
30. [-/ALL /ALL] wrong SMDR's outgoing dial number through overlap SPNET  
<Issue>  
Set up overlap SPNET and make a call from Large IP Phone like SMT series through SPNET.  
In this case, first 4 digit of outgoing dial number in SMDR is cut.  
<Reason>  
Routine about controlling internal dial buffer has some problem.  
< Modified >  
Fixed routine about controlling internal dial buffer.
31. [-/ALL /ALL] 911 PRI Routing at normal type's room in hotel feature.  
<Issue>  
When hotel feature is enabled and the room is set for normal in MMC 221, the calls will route out the PRI trunk group and work normally. But if someone makes a call from the room which has no deposit by dialing DM 4.8.4 Toll override number, the call doesn't route out.  
<Reason>  
Routine about checking toll override number have some problem.  
< Modified >  
Fixed routine about checking toll override number.
32. [-/MP20 /ALL] No Internal dial tone on 1 x virtual port  
<Issue>  
IP Phone whose port number is 255 cannot hear dial tone.  
<Reason>  
In this case, Internal RTG sent RTP packets to IP Phone whose port number is 255.  
< Modified >  
Internal RTG don't send RTP packets to IP Phone..
33. [-/ALL /ALL] SMT-i5210S phone shows Phone Type as 'SIP Standard' instead of 'SMT-i5210S'  
<Issue>  
SMT-i5210S phone shows Phone Type as 'SIP Standard' instead of 'SMT-i5210S'  
<Reason>  
SMT-i5210S's phone type is not added before version V4.80.  
< Modified >  
SMT-i5210S's phone type is added.

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**14.02.11 V4.75**

1. [P130703002/ALL/ALL] Transfer to WIP5100 stops ringing.  
<Issue>  
A station which receives an external call transfers the current call to WIP5100 and disconnects itself. WIP5100 stops ringing after 15 seconds.  
<Reason>  
About 2 years ago system was modified to make WIP5120 ring external bell when WIP5120 receives a blind transferred external call. This modification should not be applied to WIP5100 because it has only one common bell but it was applied to all WIP phones in that time. (For you reference, WIP5120 has internal and external bells. It selects one of bells depending on call type.)  
<Modified>  
It is fixed not to apply previous modification to WIP5100.
2. [P131011003/ALL/ALL] Unanswered wake-up calls ring to the second Admin improperly.  
<Issue>  
Operator station group has more than two ADMINS and its ring mode is UNCONDITIONAL.  
Set to wake-up alarm to GUEST room.  
If guest does not answer wake-up alarm, system makes a wake-up recall to ADMINS.  
In this case one of ADMINS answers wake-up recall and disconnects a call.  
After few second, a reminder call rings to the other ADMINS improperly.  
<Modified>

- System will not make a wake-up recall to the rest ADMINS if wake-up recall is cleared by one of ADMINS.
3. [-/OS7070/ALL] External Page does not work.  
<Issue>  
OS7070 External Page does not work because External Page port refers to OS7100 port table.  
<Modified>  
It is fixed to refer right port table.
  4. [-/OS7070/ALL] Loud Bell does not work.  
<Issue>  
OS7070 Loud Bell does not work because there is a relay control error.  
<Modified>  
Relay control error is fixed.
  5. [-/MP20s,MP10a,MP07,MP03/ALL] System Halt due to frequent BRI interrupt.  
<Issue>  
When BRI interrupt is happened very frequently because of bad chip, noise or disconnecting cable, system becomes halt state and all digital phones display version information.  
<Modified>  
When too many BRI interrupts are happened so BRI module can't handle all interrupt messages, it will throw away some interrupt messages which are over restricted message pool size to prevent system halt.
  6. [-/MP20s,MP10a,MP07,MP03/ALL] BRI Layer1 Lock up  
<Issue>  
When system receives abnormal signal, BRI chip is stuck in unknown state. Since then it can't process Layer1 Deactivate signal.  
<Modified>  
Even though chip is in unknown state, BRI module will deactivate Layer1 when it receives L1 Deactivate signal.
  7. [-/ALL/ALL] MGI port lockup  
<Issue>  
When system receives SIP trunk call which is routed to unplugged station, MGI port is stuck in unknown state because system tries to make an external call with LCR code 0.  
Issue is happened only when LCR code is set to 0.  
<Modified>  
It is fixed not to make an external call with LCR code in the above case.
  8. [-/ALL/ALL] SMDR IA flag is not printed.  
<Issue>  
Set SIP phone as unconditional station group member. When SIP phone answers the group call, SMDR IA flag is not printed.  
<Modified>  
It is fixed to show IA flag when SIP phone answers the unconditional group call.
  9. [P130604002/ALL/SEUK] Change default UCD Wrap-up time.  
For only SEUK, default UCD wrap-up time is changed from 3 to 0 second.
  10. [P130530001/ALL/ALL] SMDR Duration of IA flag is displayed as 0.  
<Issue>  
When Ring Preference is set to station 2xxx and it answers an external call, SMDR duration of IA flag is displayed as 0.  
<Modified>  
It is fixed not to initiate duration buffer so that right duration is displayed on SMDR.
  11. [P130628006/ALL/ALL] Station name display error in case of Hotel Check-in.  
<Issue>  
Administrator inputs guest name during check-in but this name is not displayed on Large-IP phone (ex. SMT-i5220). Because system does not send guest name to Large-IP phone.  
<Modified>  
System will send guest name to Large-IP phone.
  12. [-/ALL/ALL] Speech issue of SIP trunk call  
<Issue>

System can handle only one codec so when it receives 200 OK that has more than two codecs, it selects one of codecs and sends reINVITE to ISP. But particular ISP re-sends reINVITE that has more than two codecs and system also sends reINVITE again. This situation continues and SIP signaling between OfficeServ and ISP can't complete. Because of this user has a speech issue.

<Modified>

'Use First Codec' option is added in DM5.2.12 SIP Trunk Configuration and DM 5.2.13. You can see this option with DM V4.75 or later.

If your ISP or opposite system sends reINVITE that has more than two codecs over and over again and has a speech issue because of this, you should set 'Use First Codec' to ON.

In this case system will select the first codec of the received codec list and not send reINVITE to ISP or opposite system.

13. [-/ALL/STA] Tolling feature

For only STA, system provides Tolling feature which was already applied to SEUK.

14. [-/ALL/ALL] SIP trunk call disconnection in particular case.

<Issue>

SIP phone user makes a SIP trunk call and does blind transfer to another SIP phone before external user answers the SIP trunk call. In this case if another SIP phone user answers the call first, current call is disconnected.

<Modified>

In previous version, call is disconnected due to SIP message handling error.

With V4.75 or later you can see normal operation.

15. [-/ALL/ALL] IP-UMS is connected to the system even though its IP address is not set to Phone IP White List.

<Issue>

This issue is happened in below case.

- IP-UMS is connected to under V4.70.
- Upgrade the system to V4.70 or later. After completing upgrade, set DM2.1.6 Halt Status from Halt to Restart. (Before upgrading the system, IP-UMS should keep its connection.)
- Do not input any IP address to DM 5.13.10 Phone IP White List.
- In this case IP-UMS does not lose its connection.

<Modified>

In previous version, when Halt Status is changed from Halt to Restart, system does not check Phone IP White List because it misjudges IP-UMS connection state.

It is fixed to check right IP-UMS connection state.

16. [-/ALL/ALL] SIP phone Hold Pickup issue

<Issue>

SIP phone user can pick up a station call on hold with Hold Pickup feature code. But he/she can't pick up a trunk call on hold with same feature code.

<Modified>

This feature has not been implemented from the beginning.

It is fixed to be able to pick up a trunk call on hold by SIP phone user with Hold Pickup feature code.

17. [-/ALL/ALL] Add confirm tone for WE VoIP user.

<Issue>

When WE VoIP user sets call forward with feature code, error tone is always heard. Because of this error tone, user is hard to know current setting succeeds or not.

<Modified>

Add confirm tone to inform WE VoIP user that current feature code setting succeeds. If setting fails, error tone is heard.

※ Make sure that you should use the latest WE VoIP version(V3.5.0.0 or later) for normal operation.

18. [-/ALL/ALL] MP10a email gateway distorts low audio message when converted to WAV.

<Issue>

When you record a very low audio message, email gateway WAV file is distorted, so it's hard to understand. But when you hear a message with a telephone, it has no problem.

<Cause>

Email gateway converts audio message G.726 (32kbps) -> 16-bit signed linear pcm (128kbps) -> 8-bit unsigned linear pcm (64kbps), then quantization noise always happens.

If a message is recorded with very low audio volume, it's hard to hear it because the noise level is similar with the level of voice.

<Modified>

New option "Message format of E-Mail gateway" is available in MClass. You can select 8bit pcm (default) or 16bit pcm. If you select 16bit pcm, the quantization noise does not happen, but the converted WAV size becomes twice.

19. [-/ALL/SEI] 'AM' is wrong in 24 hours form in DM 8.11 Status Screen.

<Issue>

In DM 8.11 Status screen, time is displayed in 24 hours form, but 'AM' is attached.

<Cause>

The time is displayed in 24 hours form for Italy, but 'AM' is not fixed.

<Modified>

'AM' is removed.

20. [-/ALL/ALL] E.164 deny operation error

<Issue>

You have enabled 'SIP TRK E.164 support' and entered '070' in DM 5.2.28 E.164 deny table. Then first '0' is discarded when you make a call with "070XXXXXXX".

<Cause>

There is a logical error. It has to refer the deny table first, but it removes first '0' before it checks deny table.

<Modified>

It's fixed to check deny table before it edits the numbers.

21. [-/OS7070, OS7030/ALL] CTI application Lockup

<Issue>

Input max entries in DM 3.2.3 DID Ringing table. CTI application becomes halt state when it gets DID table data from the system.

<Cause>

When CTI application requests DID table data, system sends all entries by dividing 100 entries at a time. But in case of OS7030 and OS7070 whose max entry number is a multiple of 100, system does not send end of sending data message to CTI after completing to send all DID entries. So CTI continues to wait until it receives end of sending data from the system to finish DID query.

<Modified>

There's a timer control error. It is fixed.

22. [-/MP20s,MP10a,MP07,MP03/ALL] Memory Recovery Reset

<Issue>

In particular customer site, slow operation was happened due to memory leak.

At that time average memory was over approximately 90%.

<Modified>

To prevent slow operation, system will reset when its average memory is over 88%.

It is called memory recovery reset and this feature only works when current circumstance meets the below conditions.

- Only Linux system has Memory Recovery Reset feature.
- 'System Reset for Memory Recovery' option is added in DM 2.1.5. (default: DISABLE)  
To activate the feature, you should set this option to ENABLE.
- When average memory is over 88%, specified flag is set. System always checks this flag is set or not at every 4 am. If it is set at 4 am, system will reset automatically.

You can check memory recovery reset alarm in DM 6.1.1 after system reset.

23. [-/ALL/ALL] MOBEX Executive Auto Record

<Issue>

One way audio issue is happened in below case.

- An inbound call with SIP trunks is made to the MOBEX executive feature code.
- Dial tone is returned to the user.
- They dial extension 201 and are routed to that extension.
- Station 201 has auto record turned on to mailbox 201 with voice mail port 301.
- When the station 201 answers the call, they have one way audio on the call.  
When station 201 is not using auto record feature, the one way audio is not present.

<Modified>

There's a t-switch control error. It is fixed.

24. [-/ALL/ALL] Barge-In LED error  
<Issue>  
Barge-In LED is not turned off in below case.
- Set Trunk Monitor in DM 5.14.5 to ON and set Barge-In related options.
  - Station A is on conversation with SIP/PRI trunk.
  - Station B overrides station A and station A disconnects the call.
  - In this case Barge-In LED is not turned off.
- <Modified>  
It is fixed to turn off Barge-In LED when current station becomes IDLE state.
25. [-/ALL/ALL] H.323 Speech issue  
<Issue>  
Station A has a conversation with external user B with H.323 trunk. Then station A does blind transfer to another external user C with SIP trunk. In this case external user B and C can't hear each other's voice.
- <Modified>  
System sends a codec that is using in H.323 trunk call to opposite party when making a SIP trunk call.
26. [-/ALL/ALL] System reset during MOBEX call  
<Issue>  
Make a MOBEX call with SIP peering. When the opposite party presses the HOLD button repeatedly and disconnects the call, system resets.
- <Modified>  
It is fixed to check that opposite party of MOBEX station is valid. If it is not valid, system will not refer to related buffer to prevent system reset.
27. [P131210001/ALL/ALL] VT TRANSFER REMOTE NODE plays main greeting, not subscriber greeting.  
<Issue>  
VT Transfer in a centralized setup with 1xxxx stations routes to the main menu.  
Software V4.53, 4.65, 4.70 and 4.74a have the same issue.  
We are testing with a SVMI-20E Voice mail.  
Any other stations not starting with a 1xxx work fine.
- Node 001 - 7400 (4.53w, 4.65, 4.70d, 4.74a)  
Node 002 - 7400 (4.53w, 4.65, 4.70d, 4.74a)
- Node 1 has 1xxx extensions and centralized voicemail.  
Node 2 has 2xxx extensions
- Call comes in via PRI to Node 2 extension 2117. Extension 2117 performs a VT transfer to remote voicemail group 5149 in Node 1. Voicemail answers the call with main greeting, not subscriber greeting.
- <Modified>  
This issue is happened due to confliction of Network LCR port and Remote Station port.  
It is fixed but we recommend that you should follow below recommendation SPNet setting rule for normal operation.  
=> You should set SPNet number with 4 digits and these numbers should start with minimum 2. (For example, 2001 is OK but 1001 is not recommended.)
28. [-/ALL/ALL] NS KEYS USING FORWARD ALL AND FORWARD BUSY  
<Issue>  
ALL/BUSY forward does not work when a station of second node makes a SPNet call to a station of first node with NS key. In this case ALL/BUSY forward is set to a station of first node and this station is in busy state.
- <Cause>  
When user selects NS key, system activates PICKUP feature first.
- <Modified>  
It is fixed to work ALL/BUSY forward with NS key.
29. [P131220001/ALL/ALL] PUBLIC CALLER CID  
<Issue>  
Public caller is displayed as Caller ID of mailbox in below case.

- Incoming trunk call is routed to VMAA and VMAA calls station A.
  - Station A has a call forward setting to forward incoming call to VM.
  - Finally incoming trunk call is routed to VM again because of the above forward setting.  
And in this case Public caller is displayed as caller id of mailbox.
- <Modified>  
It is fixed to send caller ID.
30. [-/ALL/ALL] Analog Modem can't be connected to the system.  
<Issue>  
If DM IP White List is set, Analog Modem can't be connected to the system.  
<Modified>  
It is fixed not to apply DM IP White List rule to Analog Modem.
31. [-/ALL/ALL] MOBEX calling number type error  
<Issue>  
In below case mobile phone user can't make a call with incoming call number.
- MMC420 PRI Trunk Data > Calling Number Type : NATIONAL
  - MMC401 Trunk On/Off > Repeat CLI : ON
  - MMC861 System Options > Trunk to Mobex CLI : Received
  - MMC328 Mobile Extension: Set to make a call with ISDN trunk
  - MMC714 DID Ringing: Incoming ISDN trunk call is routed to designated Mobex station
  - When incoming ISDN trunk whose calling number type is INTERNATIONAL is routed to Mobex station, CID of NATIONAL type is displayed on mobile phone. If user tries to make a call with this number, he/she can't make an international call.
- <Modified>  
If Repeat CLI option(MMC401) is set to ON and Trunk to Mobex CLI(MMC861) is set to Received, system will send received calling number type instead of MMC420 calling number type.
32. [Bug Fix/ALL/ALL] DM access failure through modem  
<Issue>  
If DM 5.13.9 DM IP White List was set, DM access was failed through modem.  
<Modified>  
If DM 5.13.9 DM IP White List was se, It's possible to access to DM though modem.
33. [Change/Linux system/ALL] Changed saving location of DM user profile.  
<Modified>  
Utable file for DM user profile is no longer used. DM user profile is only saved in internal memory for security.
34. [P130731001/MP40/ALL] DB backup to SD card  
<Issue>  
If LP40 auto upgrade feature (MP40's dip switch5) was set and LP40 version and **SP40** file's LP40 version in MP40 were same. You were not able to access KMMC 815, 818, 819 because of seeing 'Media is busy' on LCD.  
<Modified>  
You can access KMMC 815, 818, 819 in this case.
35. [Bug Fix/Linux system/ALL] Malfunction of Management IP White List feature.  
<Issue>  
If you enter IP address on DM 5.13.11 Management IP White List and delete all IP address on it. After that, you can access DM from any IP address.  
<Modified>  
There was an internal file control error. This problem is fixed.
36. [P131017002/ALL/ALL] SIPT > SIPT MGI Use option no longer works.  
<Issue>  
After transferring a call with SIP trunk to other SIP Trunk, Officeserv used MSP channel instead of MGI channel in spite of setting 'SIPT > SIPT MGI Use option'.  
<Modified>  
There was a media (MGI or MPS) selection error. This problem is fixed.
37. [Bug Fix/Linux system/ALL] system halt came from memory leakage when using DNS Server.  
<Issue>  
system halt came from memory leakage when using DNS Server.  
<Changed>  
DNS query policy is changed below to prevent system halt.

- a. How to query DNS server is changed.
    - Officeserv changed how to query DNS server
    - Cycle of DNS query during failure of query is changed.
      - .until 1 hour from start of query: every 20 seconds
      - .after 1 hour: 1 minute, 2 minutes, 4 minutes .... 128 minutes (max time)
  - b. DM 2.1.5 'Multi DNS Server' option is added for user using multi ISP. (Default : Disable)  
If 'Multi DNS Server' option is disabled, Officeserv use only DM 5.6.1 DNS Server 1/2 options.  
If 'Multi DNS Server' option is enabled, Officeserv use DM 5.6.1 DNS Server 1/2 options and DM 5.2.13 DNS Server 1/2 options at the same time.
  - c. DM 2.1.5 'System Reset for Memory Recovery' option is added (default : Enable)  
If Average memory usage is over 88% because of memory leakage, OfficeServ is automatically rebooted at 4 AM.
  - d. If DND query is failed, This information is saved in DM 6.1.1 Alarm History by using expression 'Memory Rec(means Recovery) Reset'
  - e. If you change DM 5.6.1 'DNS IP address', OfficeServ is rebooted automatically.
38. [-/ALL/ALL] IP-UMS's fax receiving failure from SIP ISP which don't support T.38  
<Issue>  
If you use IP-UMS, you should set Fax option to T.38 because IP-UMS only supports T.38  
In this case, if SIP ISP doesn't support T.38 there was a fax receiving problem.  
<Modified>  
If you receive Fax call from SIP ISP, You can select each ISP to send reINVITE with T.38 or not respectively.  
In this problem case, you can set DM 5.2.13 'T.38 Reinvite' option to 'Not Send' for not sending reINVITE with T.38 and working with Fax pass through mode.

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**13.10.17 T4.74a**

- 1. [ALL/ALL] DM 4.1.2 Trunk Group is not shown.  
There's an error when disabling some features of DM 4.1.2 which don't have to be shown in T4.74. It is modified in T4.74a.

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**13.10.10 T4.74****[Bug Fix]**

- 1. [ALL/ALL] WE VoIP Smart Routing  
<Issue>  
With before T4.74 package, provisioning and smart routing can't be used at the same time if Mac address is saved as <Mobile Number> in DM 5.2.24.
  - ⇒ In case of using general version or GS3/4, WE VoIP client can't read mobile number from SIM card and Mac address is shown instead of mobile number. In this case <My Phone Number> of WE VoIP client is deactivated, so user can't change its value to mobile number and after all provisioning fails. If you input Mac address to <Mobile Number> in DM 5.2.24, provisioning will be processed successfully but you can't use Smart Routing because there's no valid mobile number.  
<Modified>  
To solve this problem, we separate previous <Mobile Number> into <Authentication Number> and <Mobile Number>. You can check this modification in DM 5.2.24.
  - ⇒ <Authentication Number>: Authentication number is used for making mobile & login profile and provisioning is processed with this number not <Mobile Number>. You should check <Authentication Number> of WE VoIP and input this number to <Authentication Number> of DM 5.2.24. In this case <Authentication Number> of WE VoIP can be mobile number or Mac address and system allows all these two types.
    - ※ WE VoIP changes previous <My Phone Number> to <Authentication Number>.  
To check this modification you should use V3.5.0.0 or later.
    - ※ Make sure that if you upgrade the system from V4.70 to T4.74, <Authentication

- Number> field will be shown as blank. For provisioning, you should copy <Mobile Number> to <Authentication Number>.
- ⇒ <Mobile Number>: This number is used for Smart Routing. For normal operation you should input only mobile number to this field.
2. [ALL/ALL] WE VoIP E.164 registration  
<Issue>  
In case of E.164 number, WE VoIP client will show all digits including <+>. In this case phone can't register to the system because WE VoIP sends Authentication number to the system without excluding <+>.  
<Modified>  
WE VoIP will send Authentication number excluding <+>. Then system will also use this number as it receives. For normal operation you should input digits without <+> in DM 5.2.24. Both WE VoIP and system packages are modified so you have to use WE VoIP whose version is V2.5.2.1T or later.
  3. [ALL/ALL] UCD modification  
<Issue>  
Sometimes call waiting count of Periodic UCD is not correct.  
<Modified>  
Add to check UCD queue status. If garbage value is remained in UCD queue, system will clear related queue information.
  4. [MP20s,MP10a,MP07,MP03/ALL] Voice issue in case of using Multicast Page  
There's a voice path control error when using Multicast Page. It is modified.  
If you use Multicast Page with SMT-i5210 which uses VLAN, you have to use SMT-i5210 whose version is T01.43\_201430927 or later for normal operation.
  5. [ALL/ALL] SIP registration failure  
<Issue>  
When using Alive Notify as SIP Options, SIP trunk registration fails if system can't receive response message after sending SIP INVITE. In this case you can't make an outgoing call until registration is recovered and it takes Alive Notify Time.  
<Modified>  
System will recover SIP trunk registration immediately not to wait for Alive Notify Time.
  6. [ALL/ALL] E.164 Deny  
<Issue>  
E.164 Deny feature does not work due to internal searching error.  
<Modified>  
System will not apply E.164 feature when making a call with Dialed Digits of DM 5.2.28 E.164 Deny Table.
  7. [ALL/ALL] System Booting Failure  
-. If system IP address and gateway have different subnet, system can't boot normally. This issue is occurred from V4.64. It is modified.  
-. Without applying T4.74, you can solve this problem by upgrading system to V4.60 and entering system & gateway IP normally. After that, downgrade the system to the previous version.
  8. [MP20s,MP10a,MP07,MP03/ALL] MP20s Watchdog  
MP20s restarts the below environment.  
-. SIP trunk call is incoming to unconditional station group which has a member pairing with MOBEX station. MOBEX station is set to make a call through SIP trunk.  
-. All group members and MOBEX station are ringing. In this case if MOBEX station user answers the call, then MP20s restarts.
  9. [ALL/STA] One Time DND improvement  
If station group is set as DND forward destination, DND forward setting will be maintained until One Time DND is deactivated.
  10. [ALL/ALL] T38 Fax failure  
-. G.711 will be used when fax receives SIP trunk call and T38 FAX negotiation fails.  
-. In the previous version, MGI64 restarts when T38 fax fails. It is modified.
  11. [ALL/ALL] SIP 200 OK Expire Time  
<Issue>

If OfficeServ tries to register to the server which does not support SIP trunk registration, the server sends 200 OK which has no expire time. As 200 OK does not have expire time, system eventually stops its registration process.

<Modified>

System will consider expire time as the value which server sent at first if the server sends 200 OK with no expire time. So system will send Register message to the server every designated expire time.

12. [ALL/ALL] SIP REG OK/NOK Alarm

<Issue>

System provides SIP alarms such as ISP OK/NOK to show SIP trunk registration status. But these alarms are too simple to show specific information. And sometimes same alarms are displayed in a row.

<Modified>

-. System displays only one alarm when same alarms are generated in a row.

-. System provides detailed alarm information as below and you can see these status in DM 6.1.1.

Status	Alarm	Description
OK	ISP1 OK REGIST	System succeeds in registering to the server with REGISTER message.
	ISP1 OK NO EXP	System succeeds in registering to the server which sends 200 OK with no expire time. In this case system uses expire time which the server offered at first.
	ISP1 OK OPTION	System succeeds in registering to the server by receiving response of OPTION message.
NOK	ISP1 NOK OPUPDT	System fails to register to the server because user changes the current setting when using OPTION message.
	ISP1 NOK UNREGI	System unregisters to the server.
	ISP1 NOK REFAIL	System tries to register to the server but it receives 4xx message.
	ISP1 NOK RE TO	System tries to register to the server but it can't receive response message.
	ISP1 NOK REAUTH	Authentication fails.
	ISP1 NOK OP TO1	System can't receive response message even though it sends OPTION message.
	ISP1 NOK OP TO2	System fails to register to the server even though it sends OPTION message.
	ISP1 NOK INVT01	System can't receive response message even though is sends INVITE message. (Inviting call leg state)
	ISP1 NOK INVT02	System can't receive response message even though it sends INVITE message. (Proceeding call leg state)
	ISP1 NOK NONAME	This alarm is shown in case of Korea.
ISP1 NOK REGRQ	System receives 403 response whose reason includes "Register request" even though it sends INVITE message.	

**13.07.08 V4.71 MP03**

**[Improved Feature]**

1. [MP03/ALL] File system recovery feature is included.

**[Bug Fix]**

1. [MP03/ALL] booting failure after uploading damaged package files is fixed.
2. [MP03/ALL] Intermittent booting failure when voice mail storage is full is fixed.
3. [MP03/Russia] Russian display message error is fixed.
4. [MP03/ALL] System reboot problem after receiving SIP 401 or 407 message is fixed.
5. [MP03/ALL] Multicast paging lock-up problem is fixed.
6. [MP03/ALL] ringback tone problem when blind-transferring to station group is fixed.

**13.05.03 V4.71 MP20****[Bug Fix]**

- [Bug Fix/MP20/ALL] MP20(New ASIC) Tone Cadence Control Issue

**13.04.08 V4.70****[New Feature]**

- [New Feature/ OS7400,OS7200,OS7100/ALL] Applying Transcend 2GB SD Card. Samsung will provide Transcend 2GB SD card (9161BA) instead of Samsung 2GB SD card for OfficeServ system. If you use Transcend 2GB SD card (9161BA), you have to use V4.70 S/W or over.  
If you encounter the situation you need to change SD card from Samsung SD card to Transcend 2G SD card for OfficeServ system, you must upgrade S/W version to V4.70 or over.

SD card	Available S/W
Samsung 2GB	V4.53 or over
Transcend 2GB	V4.70 or over

**13.03.18 V4.65****[New Feature]**

- [New Feature/ALL/ALL] Add SMT-i5210s
- [New Feature/ALL/ALL] INVITE is not sent to SIPP when IP header source port of REGISTER message and Contact header port are different.
- [New Feature/ALL/ALL] Add modification for normal SIP phone registration. SIP phone is blocked and can't re-register to the system when Unsubscribe messages are incoming continuously after it is unregistered.

**[Bug Fix]**

- [Bug Fix/MP20S/ALL] SLOT2 8COMBO SLI CID Tx Issue
- [Bug Fix/ALL/ALL] Time zone for email gateway
- [Bug Fix/ALL/ALL] System freezing issue
- [Bug Fix/MP40/ITALY] System Restart (Megawatt site)
- [Bug Fix/ALL/ALL] Provision Enhancement for We VoIP SIP Signaling Port
- [Bug Fix/ALL/ALL] IP White List of DM 6.2.2 can't be disabled.
- [Bug Fix/MP20S,7100,7070,7030/ALL] Display error in case of initializing ADMIN password by MMC900.
- [Bug Fix/ALL/ALL] PERIUCD Unanswered call count error when UCD Group is designated as Operator Group.
- [Bug Fix/ALL/ALL] Opposite party can't hear ringback tone when receiving H.323 trunk call.
- [Bug Fix/OS7400,OS7200/ALL] SVMi-20i Public port issue when IP Service is set to ON.
- [Bug Fix/ALL/ALL] Ringback tone is not heard when incoming SPNet call is transferred to MOBEX(SIP trunk) and in this case system receives 180 message.
- [P130122001/ALL/ALL] Conf call with MOBEX when using mobile phone
- [Bug Fix/ALL/ALL] E.164 Issue in case of interworking with Vodacom SIP trunk (It is occurred in V4.64)
- [Bug Fix/ALL/ALL] SVMi-20i VMAME Issue
- [Bug Fix/ALL/ALL] Delete "NONE" sub-option from Comm Exclusive in DM5.2.12 and MMC837.
- [Bug Fix/OS7400,OS7200,MP20s/ALL] Install Task is suspended when searching name of CNF24 conference member in DM 9.2.1.
- [Bug Fix/ALL/ALL] SIP Task is suspended when handling with IPUMS Message.
- [P130223005/ALL/ALL] Ticket 16158 - IP Phone license
- [P130223009/ALL/ITALY] Some strings are still displayed in English instead of Italian
- [P130223007/ALL/ITALY] Access level in DM 5.13.5 have some issues

59. [P130223006/ALL/ITALY] DM 8.6.1 does not allow uploading voice file from a shared dir named by IP address
60. [P130223004/ALL/ITALY] The DM repetitively downloads DB to local disk.
61. [P130223003/ALL/ITALY] Change string translation DM5.13.10
62. [P130223002/ALL/ITALY] Some DM's tables are not sorted properly
63. [P130223001/ALL/ITALY] The extension assigned to a function key cannot be deleted in DM 4.9.2
64. [P130222007/ALL/ITALY] Copy&Paste doesn't work for multiple choice option of the drop-down menus.
65. [Bug Fix/ALL/ITALY] Change FMC related terms.
66. [Bug Fix/OS7400,OS7200,MP20s/ALL] PWP-DM Lockout
67. [Bug Fix /ALL/ALL] Modified not to save default value as admin password.
68. [P120831007/OS7400,OS7200,MP20s/ALL] CM indication for SPNet conf. members
69. [Bug Fix/ALL/ALL] Co-Co disconnect for Spnet Calls
70. [Bug Fix/ALL/ALL] Call transfer Fail when WEVOIP transfers current call to PRI trunk. (Enbloc mode)
71. [Bug Fix/ALL/ALL] Call transfer Fail when WEVoIP transfers current call to PRI Trunk. (Overlap mode)
72. [ALL/MP20,MP40/ALL] Allowed IP range for TELNET access
73. [Bug Fix/ALL/ALL] Rearrange IP White List
74. [Bug Fix/ALL/Russia] OfficeServ7000 V4.64 Russian Code page
75. [Bug Fix/ALL/STA] ITP phone Password

#### 13.02.06 V4.64

1. **V4.64 S/W isn't compatible with previous version.**
2. **V4.64 system security is enhanced so without registering DM, Phone, FTP/Telnet IP addresses, user can't access to the system or register IP/SIP/WIP phones. Before upgrading V4.64, please read "OfficeServ V4.64\_FeatureGuide" for detail information of V4.64 security.**
3. **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.

#### [New Feature]

4. [-/ALL/ALL] Unconditional ring group supports SIP phone.
5. [-/ALL/ALL] Directory search by 2 digits.
6. [-/ALL/ALL] Add trunk access group number for dialing via SIP phone's log.
7. [-/ALL/ALL] Change default value as below.
  - (1) MMC110 AUTO HOLD: ON
  - (2) MMMC120 DIAL MODE: OVERLAP
  - (3) MMC501 SIPT CON DELAY TIME: 200 MS
  - (4) MMC501 SYS HOLD RECALL TIME: 120 SEC
8. [-/OS7400/ALL] TOS field of SIP signal packet setting  
User can set TOS field for SIP signal packet in DM 5.2.12.
9. [-/ALL/ALL] SIP Privacy Header  
DM 5.2.13 Privacy header value is able to be changed. (header, session, user, none, critical token, id ...)
10. [-/OS7400,OS7200,OS7100/ALL] MGI ID/PSWD Security  
ID and password of MGI can be changed in DM 2.2.2.
11. [-/ALL/ALL] Password security
  - Below passwords are encrypted for enhancing security.
  - . 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

**※ Due to Encryption, V4.64 system can't be connected by old DM. You should use V4.64 DM.**

- Default DM admin password should be changed on the DM Login menu. (mandatory)
  - DM Login Retry Limit and DM Login Prevent Duration are added in DM 5.14.2.
  - VMS subscriber password should be changed. (mandatory)
  - “Force Subscriber to Change PSWD” is added in DM 8.5 System Parameter.
12. [-/ALL/ALL] No Response of SIP Common Exclusive option.  
If ‘No Response’ is selected as common exclusive option, system will ignore all SIP messages from unauthorized IP address.
  13. [-/ALL/ALL] SIP phone ID and Password.  
There are no default ID and password so user should set these values more than 6 digits.
  14. [-/ALL/ALL] DM IP White List  
If specific IP address is registered in DM 5.13.9 IP White list, DM which has this IP can access the system. If there are no IP addresses in this list, there's no limitation to access the system via DM.
  15. [-/ALL/ALL] Phone IP White List  
If user wants to use IP phones and any other CTI solutions, user should set their IP address or address range.
  16. [-/ALL/ALL] Management IP White List  
If user wants to access the system via FTP, Telnet or SMDR, user should set PC IP address or address range.
  17. [-/ALL/ALL] Enhanced One Number Service  
In case of internal call, FMC Client and Desk phone which are paired can send the same CID by option. Regarding external call, one number service already has been serviced.
  18. [-/ALL/ALL] Support Wideband Codec for FMC client  
SILK and AMR-WB are added for HD voice call between FMC clients.
  19. [-/ALL/ALL] Auto 3G Number Dial List  
Auto 3G Number option is added in DM 5.2.23. If specific number is set in this field, this number is sent via 3G dial not FMC dial.
  20. [-/ALL/ALL] Support E164 Format of FMC call  
DM 5.2.29 FMC Outgoing Digits and DM 5.2.30 FMC Incoming Caller Modify menus are added to support E164 format of FMC call.

**[Bug Fix]**

21. [P120227006/OS7100/ALL] OS7100 S0-BUS calls failing
22. [-/ALL/ALL] Long call dropoff on SIP trunks  
Below problem is fixed.
  - ➔ a problem with long duration calls on SIP trunks: SIP trunk calls disconnect after just under 40 minutes. A SIP trace shows that the OfficeServ is dropping the call to the SIP carrier.
23. [P120320001/ALL/ALL] 911 Call Alarm incorrectly displayed
24. [P120320003/ALL/ALL] Tutorial Licence incorrectly displayed
25. [P120320004/ALL/ALL] Clock incorrectly set when using NTP
26. [-/ALL/ALL] SIP cause message display  
SIP messages such as 4xx, 5xx and 6xx are displayed when MMC861 SIP CAUSE DISP is ON and station language of MMC121 is English.
27. [P120329003/ALL/ALL] PGM Message fwd via CTI fails
28. [P120330003/ALL/ALL] MMC842 USER ID Limitation  
User can create user ID larger than 4 digits.
29. [-/ALL/ALL] SIP Hold re-invite in MMC837  
Modified to show Hold Re-Invite only when MPS Service is disabled.
30. [-/ALL/ALL] Page All  
Modified not to use MGI channels in case of Page All.
31. [-/ALL/ALL] SMT -i52xx Call Log redial of International numbers
32. [P120419002/OS7100/ALL] OS7100 Resetting due to WatchDog alarm and Broadcast traffic
33. [-/ALL/ALL] OfficeServ sending two “Proxy-Authorization” headers in the Registration message.
34. [P110824001/ALL/ALL] OS7000 SLT NAME PROBLEM

35. [P120425001/ALL/ALL] Pressing # to send caller to voicemail wont work using DIDs direct to a keyset.
36. [P120427001/ALL/ALL] Caller id to SLT port only show number
37. [-/ALL/ALL] Digits starting with \* in DM5.2.3 does not work
38. [-/ALL/South Africa] Default CO access code/Operator code change  
Change default value of operator number and trunk group.  
(Operator number: 0 -> 9 / Trunk group: 9 -> 0)
39. [P120509001/OS7100,7070,7030,MP20s/ALL] Embedded voicemail time stamp incorrect using NTP server
40. [P120521006/ALL/ALL] CR (call record) key not working correctly on 7030 with BRI
41. [P120521007/ALL/ALL] UCD Group busy takes 5 seconds to overflow - Ticket 14146
42. [P120522002/ALL/ALL] Tutorial License IP Phones - Ticket 14044
43. [P120529002/ALL/ALL] Problem with svmi20i and IP phones
44. [-/ALL/Dubai] REVW key  
Add STN CALL LOG in MMC110. (Dubai only)  
If this option is set, user can check call log of station calls by using REVW key.
45. [P120623002/ALL/ALL] SMDR Directory Name is overwritten when check-in/out a Room.
46. [-/ALL/ALL] R2 DID does not work
47. [P120713001/ALL/ALL] VT TRANSFER REMOTE NODE
48. [-/ALL/ALL] MP20s Watchdog reset
49. [P120731002/ALL/ALL] Ticket 14732 - Barge in ISDN2
50. [P120807002/ALL/ALL] Mobex does not work with UCD groups
51. [-/ALL/ALL] OfficeServ sends G.729 codec which is not included in INVITE SDP
52. [-/ALL/ALL] T.38 Fax outgoing failure
53. [-/ALL/ALL] Problem when the Operator does a blind transfer of DID calls to busy station with forward on busy to voicemail set.
54. [-/OS7100,7030,7070,MP20s/ALL] No NewZealand in the time zone
55. [P120830003/ALL/ALL] Station Pair SMT Phones in Station Group
56. [P120830005/ALL/ALL] Station Pairing
57. [P120830006/OS7100,MP20s/ALL] Tepri-a Upgrade failure with v4.60b
58. [P120831004/ALL/ALL] Mobex SIP Trg (peering) CLI Table
59. [P120831005/OS7400,7200/ALL] CNF 24 E-Mail Russian code page
60. [P120831007/OS7400,7200/ALL] CM indication for SPNet conf. members
61. [-/ALL/ALL] DND and Incoming call LED flashing is same.
62. [-/ALL/ALL] SIP Diversion header not work correctly in SIP trunk in 4.60b sotware  
Modified to be able to change DM 5.2.13 Privacy header value.
63. [P120913001/OS7100,7030,7070,MP20s/ALL] os7100 svmi remote hold and page feature.
64. [-/ALL/ALL] System reset
65. [-/ALL/ALL] System reset by SNMP attack
66. [P121024001/ALL/ALL] System does not send the SIP message 180 / 183.
67. [P121024005/ALL/ALL] OfficeServ does not send the right CLI to SIP peering
68. [P121024006/ OS7100,7030,7070,MP20s/ALL] An undefined error occurs when you save VM data
69. [P121025001/ALL/ALL] SMT-i5230 string is not translated
70. [P121025002/ALL/ALL] A paired DGP doesn't ring when primary/secondary is in a unconditioned station group.
71. [P121025003/ALL/ALL] Call from PSTN - MOBEX associated with SIP phone does not have voice
72. [P121025004/ALL/ALL] Call from PSTN - SLT paired with SIP does not have voice
73. [P121025005/ALL/ALL] INVITE has not media description when make a call to MOBEX via SIP peering.
74. [P121106001/ALL/ALL] MOBEX SIP CONNECTION
75. [P121128001/OS7200/ALL] SVMI-20I os7200 AUDIO ISSUE
76. [P121128002/OS7200/ALL] DM asks for 6 digit password for ITP password
77. [P121218001/OS7200/ALL] Ticket 15857 - TAPI and PGM Message
78. [P121221001/OS7200/ALL] Security risk with FTP access



**12.09.04 V4.63****[New Feature]**

- 1.
- 2.

**[Bug Fix]**

3. [VM/ALL/ALL] Pointer file(FILE.PTR) is not included when it runs 'Backup and Restore'.
4. [VM/ALL/South Africa] Trunk group number changed for South Africa
5. [P120425001/ALL/ALL] Pressing # to send caller to voicemail wont work using DID's direct to a keyset.
6. [VM/ALL/ALL] port scan made a memory leakage and system down.
7. [VM/ALL/ALL] UCD Prompts stopped when pressing dtmf digits except for exit code.
8. [-/7070,7100,MP20a/ALL] DuSLIC which is SLI chip of OS7070, 4SL2 card is changed from V1.3 to V1.4
9. [-/7100,MP20s/ Australia] applied the approved value from Australia of SLI3 card
10. [-/ALL/ALL] Cannot find HOLD REINVITE option.
11. [-/ALL/Germany] MOBEX CLIR
12. [-/ALL/South Africa] When receiving SIP 4xx, 5xx, 6xx message, display message is changed in detail
13. [-/ALL/ALL] Multicast PAGE \* fail
14. [P120529002/ALL/ALL] Problem with svmi20i and IP phones
15. [P120521007/ALL/ALL] Ticket 14146 - UCD Group busy takes 5 seconds to overflow
16. [-/ALL/Dubai] revitalize MMC110 STN CALL LOG option in Dubai
17. [P120713001/ALL/ALL] VT TRANSFER REMOTE NODE
18. [P120731002/ALL/ALL] Ticket 14732 - Barge in ISDN2
19. [-/ALL/ALL] 2205x phone's Mute button allocation is failed by kmmc or DM.
20. [-/ALL/ALL] When you receive a R2 call, the call is not follow MMC714 did digit table.
21. [-/-/SEG] sending T.38 FAX is failed
22. [-/ALL/ALL] When receiving SIP Trunk call, OfficeServ send 183 message including wrong codec information
23. [P120830006/MP20S,MP10A/ALL] Tepri-a Upgrade failure with v4.60b
24. [P120509001/LINUX system/ALL] Embedded voicemail time stamp incorrect using NTP server
25. [P120905001/ALL/ALL] Device Manager 5.2.3 Voip Outgoing Setting
26. [P120522002/ALL/ALL] Ticket 14044 - Tutorial License IP Phones



OS7000\_Changes  
\_V4.63\_20120904.xl

**12.04.02 V4.60b****[New Feature]**

1. [-/-/] DM's default Password is changed  
→ This will be applied after initializing DB
2. [-/-/]DM 5.2.13 SIP Carrier Options' Auth Password appears asterisks for security  
→ This happen in case that you use DM V4.60b or after V4.60b  
→ If you use IT or MD bellow V4.60b, you cannot see Auth Password option
3. **911 conference feature's restriction**
  - a. *Setting*
    - MMC709 TOL.PASS COD's OVRD USE TRK GRP
    - MMC769 911 DESTINATION
  - b. *In case of using 911 conference feature with analog trunk, there is*

approximately 4 seconds dialing delay

- c. In case of using 911 conference feature with analog trunk, there can be wrong digit sending because of noise.
- d. In case of using SIP station, 911 conference feature don't work but can make a 911 outgoing call.

4.

**[Bug Fix]**

- 5. [P120321001/-/-] Executive mobex - No audio when paging from SIP trunk
- 6. [P120320003/-/-] Tutorial License incorrectly displayed
- 7. [-/-/-] Tutorial License is expired in 14 days not in 60 days.
- 8. [P120313001/OS7000/-] Dialing 911 requires entry in toll pass table. (MMC 709)
- 9. [P120322001/OS7000/-] 911 calls wont send caller id out from MMC 321 on v4.60 software
- 10. [P120320001/OS7000/-] 911 Call Alarm incorrectly displayed
- 11. [P120320004/OS7000/-] Clock incorrectly set when using NTP
- 12. [-/-/Plant] Long call dropoff on SIP trunk
- 13. [P120330002/-/-] STA SIP call drops
- 14. [P120227006/MP20s,MP10a,MP11,7070,7030/SEUK] OS7100 S0-BUS calls failing
- 15. [-/-/-] SIP Hold Re-Invite option issue
- 16. [-/-/-] Video calls from SIP phones get early video from a called SMT phone or OSC
- 17. [P120329003/-/-] PGM Message fwd via CTI fails



OS7000\_Changes  
\_V4.60b\_20120402.:

**12.03.12 V4.60a**

- 18. [-/OS7400,OS7200(MP20,MP20S)/SEI] ITALY CNF24 default license. 4 channels are provided
- 19. [-/OS7400,OS7200(MP20,MP20S)/-] Display of DGP/SMT not refreshed in case of terminating Meet-Me conference.
- 20. [-/-/-] Supporting OfficeServ Communicator's H.264 codec and sRTP feature
- 21. [-/-/-] Changing how to check FMC phone's license
- 22. [-/MP40,MP20,MP20S/-] Changing CNF24's Daily Reservation period from 30 days to 3 months
- 23. [-/-/-] Long call dropoff on SIP trunk
- 24. [-/MP20S, MP10A, 7030, 7070/-] When uploading DM DB, DHCP Server Option is not applied
- 25. [-/-/-] Supporting SIP Phone's auth code dialing
- 26. [-/-/-] Mobex caller XXX to SGP Problem in case of uncondition type's station [group](#)
- 27. [-/-/-] spnet to spnet Video call
- 28. [-/-/-] video call between OSPP IPP and SPNET's hold/resume



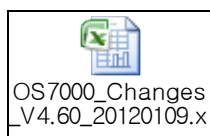
OS7000\_Changes  
\_V4.60a\_20120312.:

**12.02.16 V4.60**

- 1. **V4.60 S/W isn't compatible with below V4.60 DB**
- 2. [-/OS7400,OS7200(MP20)/-] New SVMi-20i card are now available.
- 3. [-/OS7400,OS7200,OS7100/-] New SLI3 cards(16SLI3, 16SLI3s, 8SLI3, 8COMBO3) are now available.
- 4. [-/MP20s,MP10a,MP11,OS7030,OS7070/-] MP20s restart problem when a station send DTMF repeatedly was fixed.

5. [-/ALL/-] Mobile Client  
Provide Mobile Enterprise service as below.  
-> SIP basic call feature, move, one-step transfer, unregistered forward, smart routing
6. [-/-] Support Real-Time Tone Generation service (R4.1)
7. [-/-] Plug and Play for IP Phones.  
Provide 3 types of PNP mode. (Pre-Macaddr, Auto PNP, Normal Login)
8. [-/-] IPP Auto Upgrade.  
Add MMC914 to upgrade all IP phones at the same time. This option is for only manager.
9. [-/MP40, MP20/-] SIP TLS
10. [-/OS7400, OS7200 MP20/-] Supporting sRTP
11. [-/ALL/-] Supporting Multi Carrier SIP registration  
After uploading DM DB to V4.60, Set specific option related to SIP trunk. Or you can't use sip trunk.
12. [N100430002/-/Denmark] Change the 480 Sip Message.
13. [eMGI] Support VBD(Voice Band Data) service
14. [-/ALL/-] Privacy header option for outgoing SIP trunks calls.
15. [-/-] SIP Peering Option.  
Tandem(SIP/H.323 call => SIP/H.323 call) is allowed by MMC833 VoIP Tandem option.
16. [-/ALL/-] Selectable codec for SIP trunk calls
17. [N101116006/ALL /-] MOBEX Callback
18. [N090625013/-/-] MOBEX Scheduling  
In the previous version, MOBEX station and its master are ringing at the same time regardless of current time. But in V4.60, simultaneous ringing is allowed by MOBEX scheduling time.
19. [-/-] MOBEX Ring Group Busy  
When MOBEX station is busy, its master will be also busy.
20. [-/-] Ability to set what type of calls can ring MOBEX stn is added.
21. [-/-] MOBEX authorization by CLI Ringing Table is added.
22. [N090625008/-/STA] Recommend to change MOBEX feature code displays.
23. [-/ALL/-] Supporting H.264 codec to Video SIP Phone
24. [-/ALL/-] Sending DTMF to SIP Phone
25. [-/-] CNF24 Enhancements.
26. [-/-] Single Address book and calendar features are added.
27. [-/-] Add SIP Trunk Error Alarm.
28. [-/-] Modify default value for each country.
29. [-/ALL/-] Separating SIP Trunking / Peering port
30. [-/-] DID Limit Count per Ring Plan is added.
31. [-/-] UCD Queue Limit Count feature is added.
32. [-/ALL/-] NTP Server
33. [P110204001/ALL/-] Ticket 10313 - Virtual port Mailbox delete
34. [P110825001/ALL/-] PWP Scheduling Issue
35. [P110715004/ALL/-] Wrapup timer
36. [P110413004/ALL/-] LCR code on Disply
37. [P110614001/ALL/-] Multiple connections via IT Tool
38. [P110816001/ALL/-] MMC 741 options will not
39. [P110816002/ALL/-] Unsupervised Conference Bridge
40. [P110716002/ALL/-] SPNET TOLL RESTRICTION
41. [N100227012/ALL/STA] NO CLIP BUTTON
42. [N091116002/ALL/-] Change priority between Alias or CLIP Translation table

43. [N101123004/ALL/-] Trunk number information at LCD should be changeable to given Names.
44. [P110824002/ALL/-] 7100 voicemail language key code
45. [P110817001/ALL/-] Pre configured data in MMC712
46. [P110810001/ALL/-] GPLIM V1.41 NTP CLOCKING ISSUE
47. [P110727001/ALL/-] MP20S RESTART
48. [P110707002/ALL/-] MWI (PRS mode) feature does not work on 16SLI2 card
49. [P110707001/ALL/-] OfficeServ does not detect the congestion tone to disconnect the cleared call automatically
50. [P110704005/ALL/-] CID option cannot be set for CRM by means IT2.2.1
51. [P110704003/ALL/-] Rings before answer option (VMAA->PORT BLOCK->SAMSUNG IN-SKIN) does not work properly
52. [P110625002/ALL/-] Wrong translation, LCD display text and still some English words
53. [P110623001/ALL/-] Ticket 11529 - UCD from AA on auto answer
54. [P110527002/ALL/-] Find Me Feature Callout Problem
55. [P110527001/ALL/-] SVM I ALERT FEATURE OUTCALL FAILURE
56. [P110429009/ALL/-] IDCS28 BTN SET INTERACTION
57. [P110421001/ALL/-] SPECIFIC REDIAL FEATURE ISSUE
58. [P110418002/ALL/-] Programming od ANS/RLS Key to keysets
59. [P110414001/ALL/-] Call Forward No-Ans
60. [P110413003/ALL/-] WIFI Dect
61. [P110413001/ALL/-] ANI CLID E&M Trunking not Displaying
62. [P110402001/ALL/-] MMC 430 DMS100 Setting No Name
63. [P110707002/MP20/SEI] MWI (PRS mode) feature does not work on 16SLI2 card.
64. [-/ALL/-] SMDR buffering (All system, 10000ea)
65. [-/All/-] Support Hybrid Voice Paging feature (R4.2)
66. [-/OS7100, OS7030, OS7070/-] change MSP firmware for voice encryption and fixing VM bug.
67. [eMGI/OS700,OS7030,OS7070/-] Support RTCP-XR.
68. [P110915-0012] RTG Version Display Error
69. [P111004-0002] Call Drop at the long-run test about SIP peering.
70. [P111004-0003] MPS Task Suspend at the long-run test about SIP peering.
71. [P110707001/ALL/SEI] OfficeServ does not detect the congestion tone to disconnect the cleared call automatically
72. [P110824003/ALL/STA] Group display Station Group data.
73. [P111103-0003/ALL/-] RTG Higher Gain and RTG sound difference between G.729 and G.711.
74. Samsung OfficeServ 7100(MP10a) and 7030 version V4.54a confirmed as Vendor Validated with BroadWorks R17SP3 for SIP trunking interoperability.
75. [N090625013/ALL/ALL] Move MOBEX Scheduling menu.
76. [-/ALL/ALL] Move MOBEX Group Busy option.
77. [-/ALL/ALL] After initializing or restoring VM, the system will reboot automatically.
78. [-/ALL/ALL] MSP firmware is updated to fix silence detection bug(Message cutoff issue is related.)
79. [-/ALL/ALL] Email address was not saved in a mailbox because it exceeded the Email gateway license but there was no notice. It's fixed to show a popup window.
80. [-/MP10a, MP20s/ALL] Email gateway did not work with Google mail server. It's because the certificates are not copied to the system while upgrading to V4.60. It's fixed.



**11.03.18 V4.53c**

1. [-/MP11/-] NAT options are initialized after reboot on OS7100 MP11 system.
2. [-/-/] Improving SIP Trunk exclusive function.
3. [P110309002/-/-] Networking issue os7100 and os500 switch
4. [-/-/] WiFi pause issue
5. [P110222003/-/-] It is fixed that the clip translation name does not display on an SLT handset.
6. [-/-/] SIP Hold Pick up does not work
7. [-/-/] Consultation transfer does not work through Spnet on the 3rd party SIP station
8. [-/-/] Changing Station name in Common Port Control (ITool 2.4.2) using IT makes SMT phones losing connection to MP10a
9. [P101123007/-/-] DS5012L incorrect time
10. [-/OS7070/-] SLT port plug-out issue
11. [-/OS7100 MP10a/-] Fail to connect VM web on MP10a.
12. [P110322002/-/-] System Reset Sequence
13. [P110328001/OS7100/-] OAS card restart in OS7100



OS7000\_changes\_  
V453c\_20110318.xl:

**11.01.25 V4.53b**

14. [-/-/] SIP OPTIONS problem
15. [-/-/] Adding SIP Station Diversion Header
16. [-/-/Turkey] Trunk Disconnect Time is updated when transferring a call



OS7000\_changes\_  
V453b\_20110125.xl:

**10.12.31 V4.53a**

1. [-/OS7400,OS7200 MP20, MP20s/-] PWP last week problem.



OS7000\_changes\_  
V453a\_20101231.xl:

**10.12.14 V4.53**

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1. **V4.53 S/W is compatible with V4.52 DB, but isn't compatible with below V4.52 DB**
2. **When you upgrade OS7400 MP40 S/W or OS7200 MP20 S/W by using Installation Tool from V4.4x to V4.53, you have to upgrade twice (V4.4x -> V4.51 -> V4.53) because of MP S/W package size.**
3. **[-/OS7400,OS7200,OS7100/-] Samsung 2GB SD Card applied**
4. [P100803006/OS7030/-] Second cabinet trunk ports do not show in trunk group
5. [P100908002/OS7100 MP10/-] MP10 v/mail message forward
6. [P100915002/OS7200 MP20/-] Supervisor key dial tone
7. [P100910003/OS7100/-] IT Tool MMC 831 public rtp port update
8. [P100819002/-/Australia, New Zealand] ITool
9. [P100824001/-/-] ABW CTI DATAVIEW ISSUE
10. [P100813001/-/-] MMC 119 GROUP DISPLAY
11. [P100914001/-/Ireland, Turkey] OS7030 CLI Incorrect on Analog trunks in Ireland
12. ~~[P100909002/-/-] SVMi20 Call Recording after a Barge In~~  
-> This will be applied in V4.60.
13. [-/OS7400, OS7200 MP20, MP20S] Add self join alarm
14. [-/OS7100 MP10/-] Language key code of system parameter is not saved.
15. [-/-/] WiFi Phone Registration Problem after power off/on
16. [-/-/] WiFi Phone Registration Problem after temp License is expired
17. [-/Except for OS7200 MCP/-] Diversion Header Support in Mobex Call
18. [-/-/Finland] E164 number support for Finland
19. [-/Linux System/-] Add MMC516 (Tone Cadence set)
20. [-/-/] 52XX station's one-way speech
21. [-/OS7200 MP20s/-] voice prompt distortion on the MP20S VM ports
22. [-/OS7070/-] SLI Port plug-out
23. [-/OS7200/-] OS7200 SPNET FAX Problem.
24. Apply new msp file to CNF24 card.
25. [DM] Change FTP Port for NAT
26. [DM] DM Package files can upload with File Control Menu.
27. [DM] Add the Sort Function.
28. [DM] Input password 3 times without closing login screen.
29. [CNF24] STA "Who am I Request" prompt
30. [-/-/Central and South America] Busy Tone Frequency Option is Added
31. [-/-/] SVM Max Message
32. Change invalid conference ID prompt
33. [P101112003/OS7200 MP20S/-] MP20S prints out the room status and the wake up call set up for 40 stations only
34. [P101030001/-/-] STATION PAIR and Call Coverage
35. [P100915001/-/-] NND feature while doing auto record cause lost calls
36. [-/Linux System/Russia, New Zealand] Called Number Plan & Type option of PRI, BRI is shown in OfficeServ Linux System.
37. [-/-/] MCONF Unsupervised Conference Problem

38. [-/OS7200/-] Caller didn't hear ringback tone.

**10.08.18 V4.51**

1. [P100526001/OS7000/-] Barge In Failure  
When 3rd party tries to barge-in and then routes to voice mail, original call is dropped.  
Modified not to disconnect current call in the above case.
2. [P100507004/OS7000/-] (SEUK) Communicator shows LCR code  
Communicator show the LCR code when call is connected.  
Modified to show original dialed number.
3. [P100522004/OS7000/-] (STA) VMAME Feature Broke  
IP phone with VMAME key doesn't hear the message being left by the calling user so they don't know if to answer or not.  
Modified to send audio message during ringing mode.
4. [P100602001/OS7000/-] Virtual SLT ports ring Normal SLT ports  
When you have a pot phone connected to the 8 combo card and ring a virtual extension that is in the same slot and port then the pot phone will also ring.  
Modified not to send ring message to virtual SLT.
5. [-/OS7000/-] Group pickup info remains on screen.  
Group pickup information remains on screen when call is received on analog trunks.  
Modified to delete pickup group information in the above case.
6. [-/OS7000/-] Add IP-AOM in MMC841.
7. [-/OS7000/-] Speech comes over the speaker when in headset mode.  
Under certain configuration and call scenario you can hear speech coming over the speaker when in headset mode.  
Modified not to open speaker path by following SECURE OHVA option.
8. [N091006001/OS7000/-] External Call Forward does not follow Station/Trunk use.  
Add Enhanced FWD to MMC210.  
If this option and EXT FWD are ON, trunk call can be outgoing or not by following strn/trk use.
9. [P100615003/OS7000/-] WIP cannot answer calls  
Cannot answer call to WIP5000 handset.
10. [P100617006/OS7000/-] Auto Answer intermittently not working  
Under certain scenario Auto Answer is intermittently not working and instead and Announce ring mode is being used even though the extension is set to use Auto answer.
11. [P100523001/MP20/-] 16-TRK FLASH ISSUE  
When you flash the port 5 thru 16 of the 16TRK/8TRK2 card, you will not get dial tone and disconnect the first call.
12. [P100527001/OS7000/USA] Page Timeout Broke  
Because of request, MMC 501 page timer will affect during internal paging

13. [P100615001/MP20S/-] MP20S MAIN GRETTING CLICKING  
Clicking on the MP20S voice mail main greeting when the main greeting is redone.
14. [P100617004/OS7000/-] UCD Prompts Not clipped  
Prompts on the UCD group were not being clipped when an agent becomes free.
15. [P100622003/OS7000/-] P100208005 SIP Hold Reinvite does not work correctly for Transferred calls
16. [P100622004/OS7000/UK,EU] Incorrect SMDR code for UCD to UCD abandoned call. On v4.46d  
if a call to UCD Group and go to final destination which is UCD Group, and the caller hangs up when the agent phone is ringing, the system sends a 'C' code and not an 'A' code to SMDR.
17. [P100629004/OS7000/] SMT IP Phone delayed speech problem.  
call into a system on alog trunk, answered by an SMTi5243, There is a delay of about 3 seconds
18. [P100706002/OS7000/-] SIP Calls being rejected with a SIP 513 message too big  
On an OS7100 when a call to a SIP extension is routed out over an SIP Trunk the call is being rejected with a 513 message too big error.
19. [P100708003/OS7000/-] Peri UCD stats showing 65xxx unanswered calls  
If there are calls in the queue when performing either a manual Clear Stats or Automatically then the unanswered calls show 65535 unanswered calls.
20. [P100715002/OS7000/Germany] All SMT-i52xx/ Umlauts are missing after hooked off  
Umlauts are not indicated when hooked off at source side
21. [P100724001/MP20/-] MP20 CONFERENCE SQUELL
22. [P100728001/OS7000/-] OS Operator not displaying CLIP Translation Properly  
Incoming calls are not showing the callers CLIP Translation name when programmed in MMC728 on OS Operator. Instead of showing the name there are some strange characters
23. [P100729003/OS7000/-] FOLLOW ME FORWARDING ISSUE  
In case of follow me feature, MMC 210 Chain Forward option didn't affect.
24. [P100729004/MP20/-] PAGE SUPERVISOR FEATURE INTERACTION
25. [P100807002/OS7000/-] CBK with station Pair issue  
Paired port did not assign a call button when call back ring occurred. To assigned a call button is missed.



OS7000\_changes\_  
V451\_20100818.xls;

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**10.07.12 V4.50**

1. [-/-] SEG V450 Issue
2. [-/-]The dialed PSTN no. is flickering during redialing with the redial function.
3. [-/OS7400,OS7200 MP20/Australia] Australia Radio Station  
- Add Function 1: Radio Station Link On/Off with DT button.  
- Add Function 2: Conference call is also connected with Radio Station.
4. [-/OS7030/Germany] OS7030 Germany License

5. [-/OS7400,OS7200 MP20/Algeri] SMDR Buffering
6. [-/-/] SEG T450 Issue
  - Bug No.4 : Internal Dial tone must be German styl
  - Bug No.5 : Mismatch Device error
7. [P100615003/OS7400,OS7200 MP20/-] WIP cannot answer calls
8. [-/OS7400,OS7200 MP20/Australia] Australia Radio Station
9. [-/OS7030/Germany] OS7030 Germany License
10. [-/OS7400,OS7200 MP20/Algeri] SMDR Buffering

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**10.06.14 V4.48**

1. [-/OS7400/USA] Remote Station Function
  - : Remote Station Function and NS function are same.

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**10.05.03 V4.46d**

1. [-/OS7030/ALL] OS7030 slave cabinet lock-up
  - : Link check error between master cabinet and slave cabinet is fixed.
2. [-/ALL/ALL] Version display of IP-AOM was wrong on KMMC 840
  - : Version display of IP-AOM was fixed on KMMC 840
3. [-/ALL/ALL] Incoming calls to a UCD group with Sip extensions get stuck on second UCD message.
  - : In this case, new internal call flow is added.
4. [-/ALL/ALL] Desi-less phone's 10 character name error
  - : Last 10th character will be checked normally.
5. [-/ALL/ALL] a call from SIP trunk to AA makes a one-way speech.
  - : If a call from SIP trunk route to AA, SIP module sends update response message to MP.
6. [-/OS7200 MCP(Only SmartMedia Card Type)/ALL] BRI Clicking Noise.
  - : If OS7200 MCP's SmartMedia Card was controlled, BRI/PRI reference clock was changed.
  - This problem is fixed.

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**10.05.31 V4.46e**

1. [-/OS7200(MP20s),OS7030,OS7070,OS7100/ALL]
  - VM language is set to English although the system is set to other country(for example, Korea, Italy etc.)
2. [-/ALL/ALL]Voice Cross talk after paging
3. [P100525004,P100522003/ALL/ALL] RTO Key problem

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**10.04.15 V4.46c**

1. [P100303001/ALL/ALL]Missed call notification on SMT -ixxx
  - Missed call notification follows a setting in MMC 110. Phone & MP S/W modified.
2. [P100215001/ALL/ALL] UCD PERI data is incorrect for calls waiting
  - Ringing call is not counted at Missed call

3. [P100208005/ALL/ALL] Cannot disable HOLD REINVITE  
If MPS off, HOLD REINVITE can be disabled.
4. [P100310012/ALL/ALL] UCD Peri Stats Incorrect
5. [P100310013/ALL/UK,EU] SMDR data is incorrect for ABANDONED call at UCD group  
Spec change
6. [-/ALL/ALL] Speed dial bins 00-09  
You can enter 00~09 speed dial bin number.
7. [P100128004/ALL/ALL] CENTRALIZED VOICE MAIL DND ISSUE <sup>□</sup>.  
When DND forward, Call is forwarded to Centralized VM correctly.
8. [P100105003/ALL/ALL] Voice Announce over SPNet  
When SPNet, CO call will meet voice announce.
9. [-/ALL/ALL] VPN with SPNet  
It is changed to check private setting from class to range.
10. [-/ALL/Turkey] Turkish translation
11. [-/ALL/ALL] Prompts is installed after rebooting, although user deletes the installed prompts.
12. [-/OS7030/Turkey] Improve voice quality of Turkish prompts
13. [P100122003/ALL/ALL] Account Code Response Invalid
14. [P100128006/ALL/ALL] RING BACK HEARD OVER SPEAKER

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**10.02.26 V4.46a**

1. [-/ALL/ALL] Set relocation restriction between IP phones.  
It is requested to restrict set relocation between IP phones because it is more convenient to use Extension Login rather than Set Relocation in case of using IP phone.
2. [-/ALL/ALL] Restriction of using Headset key during conversation.  
When headset key is used during conversation, there are a lot of considerations and side effects. So headset key should not be used during conversation.
3. [-/ALL/ALL] Default language is not changed.  
After changing the default language from English to French, system rebooted. The default language is reverted to English.
4. [-/ALL/ALL] If station names contain a space and the first part is the same the second mailbox is not imported.  
=> Delimiter is changed to be at least tab character or more than 2 blank spaces.

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**10.02.22 V4.46a**

1. [-/ALL/ALL] NET DSS key assignment  
User can assign DSS key from SMT-i5230 but can't assign NET DSS key.
2. [-/ALL/ALL] MMC120 AOM CURSOR  
New Large IP Phones don't have to use AOM CURSOR in MMC120, because these new IP phones always display the previous position not following MMC 120 AOM CURSOR.  
=> Modified not to display AOM CURSOR in case of new large IP phones
3. [-/ALL/ALL] Delete the previous follow me destination number

Even though user deactivates Follow ME by inputting "600", the previous destination number is still remained in system DB.

To make it clear for deactivating, destination number should be deleted.

4. [-/ALL/ALL] Disable Boss and Secretary Function menu in case of SMT-i3105  
SMT-i3105 doesn't support Boss and Secretary function.  
=> Modified that SMT-i3105 can't be set as Boss or Secretary in MMC303
5. [-/ALL/ALL] Group Call to Paired Station  
Secondary member of station pair doesn't ring in below case.
  - (1) In MMC601, create group 501 with member 2001.
  - (2) In MMC217, set 2001 as primary with 2002 as secondary.
  - (3) Make a call to group 501.
  - (4) 2001 will ring but 2002 does not.
6. [-/ALL/ALL] OS7030 VM License Issue  
7030 has two VM licenses basically. But in below case, user can't make a call to VM.
  - (1) In MMC601, create VM group 519 with two VM members 301, 302.
  - (2) Make a call to VM group 519. It works OK.  
User can hear "enter your password".
  - (3) Make another call to VM group 519.  
User can't hear "enter your password" but hear ring back tone only.  
(If user makes a call to 301 and 302 individually, there is no problem.)  
=> Correct usable license checking process.
7. [-/ALL/ALL] Delayed RP Key Operation  
RP key LED switching time is delayed about 1 minute.
  - (1) Set Ring Plan 1 as 00:00 to 23:59.
  - (2) Set Ring Plan 2 as 07:00 to 16:00.
  - (3) Make two RP key for two ring plans.
  - (4) RP key 2 turns on about 7:01 and RP key 1 turns on 16:01.  
=> Modified that next RP key LED turns on when current time and current ring plan end time are same.
8. [P091224006/ALL/ALL] Announce Transfer Headset Operation  
Headset is deactivated in below case.
  - (1) Answer trunk call on extension 201 (digital keyset)
  - (2) Press TRSF, dial 3201 (SMT-i3100)
  - (3) 3201 press SPKR to answer with Headset.
  - (4) 201 announce's call (Speech OK)
  - (5) 201 hangs up.
  - (6) Issue 3201 can't hear caller. Keyset Microphone is activated.
  - (7) if 3201 places call on hold, then retrieves call from hold.  
In this case Tx and RX speech is OK but microphone is deactivated.  
=> Modified not to send beep tone in case of using IP phone and headset mode.

1. [P091031014/ALL/ALL] KMMC cannot set LNR key for SMT-i5243
2. [P091031004/ALL/ALL] OS 7000 - All models - Cannot KMMC with iDCS 8 button phone
3. [P091031017/ALL/ALL] OfficeServ Call - GLISTEN SPEAKER problem
4. [P091031012/ALL/ALL] OS 7000 - All models - UCD Final Destination problem
5. [P091106001/ALL/ALL] ALL OS Systems - Caller hang up during transfer can cause problems
6. [P091102005/ALL/ALL] OS 7000 - All Models - CID button not working properly.
7. [-/ALL/ALL] Tutorial License's SPNET is Enabled
8. [-/OS7070/Russia] OS7070 H323 Application
9. [-/ALL/ALL] MMC430 PRI ANY CHANNEL set issue for Network mode  
: It's changed that you can set PRI Any Cannel for Network Mode on MMC430.
10. [P091118002/ALL/ALL] OS 7000 - All models - KMMC 326 show incorrect data
11. [-/ALL/STA] MMC210 area delete option issue for Phonebook dialing  
: When you dial using keyset's phonebook, if digit size is 10 digits and MMC 210 cid code insert option is on, MMC210 area delete option was not applied. System will handle area delete option before handling cid code insert option
12. [-/ALL/ALL] Desi-less feature  
: From V4.46, New IPP(SMT-i5230) user can change its key info and edit its key name in MMC766.
13. [-/ALL/ALL] From keyset you cannot change MMC 104 (Station Name).  
: System supported Korean name only in Korea. After V4.46a, System will support Korean name in USA country code.
14. [-/ALL/ALL] Modify the formula to calculate the REGISTER expire time.  
There was a problem to calculate the REGISTER expire time. Because of this problem, the registration of OfficeServ for SIP server is expired in case of the SIP server doesn't send a 180 Ringing message for the INVITE message from OfficeServ. I fixed a bug of the formula.
15. [-/OS7100,7030,7070/ALL] The default volume is changed.
  - Problem : The volume of attached file sent via email gateway was too loud.
  - Reason : There was a mismatch between prompt volume level and user recorded message volume. The user recorded message was so quiet, compared with prompt volume. So the recording gain(of DSP) is set to 6dB. It was a US prompt.  
But the Italian prompt volume is a little low than US prompt. So, the recorded message is too loud compared with the prompt.
  - Modification
    1. The recording gain is changed 2dB for Italy.
    2. The default playback volume is set to normal(all countries)  
: It's because the many countries reported that the default playback volume was too loud.  
This won't be applied when upgrading the product because it will not change the configuration that a user sets.
16. [-/OS7100,7030,7070/Russia] The toll restriction is applied for Russia.
  - Problem : [Russia] The outcall authorization is set to yes, so illegal use can be a problem.

- Reason : The outcall is allowed in default configuration. So SVMi can dial any number and it can be a security hole for illegal use.
  - Modification : The outcall authorization is set to NO in default configuration and message alert feature is changed to follow the authorization.
17. [-/OS7100,7030,7070/-] The maximum mailboxes is limited to 256.  
The maximum mailboxes was 1,000, But the web management can not display the all mailboxes. The buffer used to exchange the data between VM and Web module can not handle 1,000 mailboxes. The maximum mailbox number is changed to 256.
18. [P091215006/ALL/ALL] V4.42b Hotel mode does not delete/create voicemail mailboxes on Checkout, Checkin.
19. [-/ALL/ALL] SIP Trunk to SIP Trunk call's mgi use option is added
- In case of Specific SIP Server, if the server don't receive rtp packet, the server don't send rtp packet. MPS can forward received rtp packet to opposite side and cannot make rtp packet.
  - In case of this Specific SIP Server, OfficeServ system cannot use MPS channel for SIP trunk to SIP trunk call. If added SIP2SIP MGI option (MMC861 - VOIP RTP OPTION)set to on, OfficeServ system use MGI channel for SIP trunk to SIP trunk call.
20. [-/ALL/ALL] Add Secretary Answer Mode
- As boss answer option is newly added to New Large IPP (such as 5230, 5243 etc.), not only BOSS but also SECRETARY should be able to set its own boss answer mode. So system needs to add boss answer mode for secretary.  
(In previous version, only boss can have its own boss answer mode. And this boss answer mode is applied when secretary makes a call to his/her boss using BOSS key.)
  - Modified to be able to set secretary's boss answer mode in MMC125.  
Be aware that secretary can set only boss answer mode. The other options(EXEC STATE, STATE(IN/OUT)) can't be set in case of secretary.
21. [-/ALL/STA,Russia] Disconnection problem of WIFI 5120 in case of deleting all license Keys.
- In USA or Russia, 5120 lost its connection when all license keys are deleted in IT 2.1.4.
  - Modified not to disconnect 5120 when all license keys are deleted. And this modification is only applied in USA and Russia.
22. [-/ALL/ALL] IT 4.9.2 data upload problem
- System can send the limited data to Installation Tool. And as key name is newly added in IT 4.9.2, the total data size is increased. But this increase doesn't consider 2 or 3 cabinet system but only 1 cabinet system.
  - Increase the total data size(system → IT) considering the max capacity.  
(System can have max capacity in case of having 3 cabinets and installing many 16 DLI cards or assigning many virtual DLI/ITP slots.)
23. [-/ALL/ALL] SIP Trunk 183/180 message option added.
- When OfficeServ system receive SIP trunk call, it always send 183 message.
  - You can select 183/180 message for SIP trunk.  
(MMC 861 - VOIP RTP OPTION - SIP-T RBACK)
24. [P091206002/ALL/ALL] Incoming SIP Trunk DDI calls routed to an extension do not have

ringback

25. [P091206003/ALL/ALL] OS7200 v4.42 after 10-15 minutes loses registration
26. [P091120003/ALL/ALL] Trunk to Trunk Transfer on Sip Trunks does not work properly
27. [P091207003/ALL/ALL] SIP calls failing due to CANCEL message
28. [P091031011/ALL/ALL] OS 7000 - All models - MGI codec problem
29. [-/ALL/ALL] Call waited station's voice path error during conference
  - Call waited IPP station's voice path error during conference as follows:
    1. One IPP station makes a conference call using a handset.
    2. At that time, incoming calls go to the IPP
    3. The call is in a call-waiting state.
    4. After around 5~7 seconds, automatically the IPP's speaker is opened.
  - this problem is fixed.
30. [P100118004/OS7200 MP20/UK] MP20 Caller ID on analogue trunks not working
31. [-/Linux system/-] Linux system's MSP file is changed,
  1. MP10/MP10a, OS7070, OS7200 MP20S
    - R2 error fixed
    - Fax error fixed(Compatible with Audio Codes MGI)
    - After G.711 extended playback, Silence Detection error fixed
  2. MP11, OS7030
    - NU Tone Detection error fixed
    - UD Trace error fixed
    - Fax error fixed(Compatible with Audio Codes MGI)
32. [-/ALL/ALL] Mobex extension's CID issue.
  - Problem : Any call that is transferred to a phone with Mobex set, does not send the calling party's Caller ID to the Mobex extension. The caller ID does not show on the desktop phone (digital or IP) but the outside phone that is set up in Mobex ( i.e.: Cell phone or Home phone) only receives the CLI information (set in the "Send CLI Number" program in MMC 321) from the port that transferred the call.
  - Modification: MMC 210 'HOLD ID SEND' option is added. If you set this option to on, you can get the transferred CID on MOBEX extension.
33. [-/ALL/STA] Large LCD Phone's default dial type is changed from enbloc to overlap in only USA.
34. [-/ALL/ALL] New Database option entry for Prack Support

**Problem** : There is a reported problem that certain types of SIP Servers do not handle PRACK messages, which cause sudden call drops while listening to ringback tones. Therefore, customers whose OfficeServ systems are connected to that kind of SIP Servers request that OfficeServ can choose to send PRACK or not by users' choice.

According to the SIP standard, it is absolutely legal to send PRACK requests reacting upon 183 responses from the SIP server side. Therefore, it is not OfficeServ's fault, but the ISP's.

However, in terms of customer satisfaction through maximizing interoperability with other SIP entities, OfficeServ decided to accept the customer's request.

**Reason :** SIP Servers interacting with OfficeServ do not handle PRACK message properly.

**Modification :** Made a new DB entry called Prack Support, which enables OfficeServ administrator choose whether to use PRACK or not.

35. [-/ALL/ALL] In case that MPS Service option is enable, one way speech problem of spnet call over V4.14k or V2.75b S/W is fixed.
36. [P091224006/OS7030/ALL] OS7030 slave cabinet causes DB problems
37. [P100109001/OS7100,OS7200/ALL] Failure to update LEDs status of any DGP connected to the first port of a DLI card.
38. [P100130001/OS7100,OS7200/ALL] DT LAMP OPERATION
39. [P091119001/OS7200/ALL] All DTMF receivers lockup every week
40. [P091231002/ALL/USA] Name Transfer Voice Mail to Switch
41. [P091215004/ALL/ALL] Macro Keys
42. [P100122008/ALL/ALL] ITP5121 Ringing Speaker Operation Issue
43. [-/VxWorks System/ALL] The contents applied to VxWorks stack
  - 1) Mbuf leakage is observed with some of the Wind River's after many TCP connection requests.
  - 2) An IPv4 fragment received with MF and an invalid payload length is leaked
  - 3) Intermittent TCP checksum error, consistently greater by one, on PowerPC, ARM/XScale, and SH targets

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**09.11.05 V4.42b**

1. [N090815007/ALL/ALL] TEMP LICENSE display screens in MMC 860 should be changed
2. [P090815002/ALL/ALL] Headset mode cannot be set or unset during conversation
3. [P090815001/ALL/ALL] ABW key cannot be used during phone call
4. [P090813001] KMMC 406 cannot be used set extension 3999 modem as trunk ring.
5. [P090811001] OS7200 MCP Set Relocation
6. [P080424014/ALL/ALL] ITP5112L OVERLAP MODE OPERATION
7. [-/ALL/ALL] transferred call's CID from AA didn't display

When incoming call with CID is answered by the AA and then transfer to a station (eg; 203) the display of station 203 will not show [CID name and number]. Display show: "TRANS FM 701" on the second line. This is fixed
8. [-/ALL/ALL] can't use IT tool to assign VT key on button 13 of 14 button set

When using software load 4.41 and IT tool version 1.40, to IT tool 4.9.2 (MMC 722) cannot be used to program an VT key on any digital or ITP phone. It's fixed. When using KMMC programming the VT key on a key phone is not a problem.
9. [P090822001/ALL/ALL] Missed Calls display causes date lockup on keyset
10. [P090901004] MMC119 Ringing Display on Line 2 is set to None when i5243 connected
11. [P090728007/ALL/ALL] DND FORWARD FUNCTIONALITY OS7400
12. [P090808001/ALL/ALL ] Network forward 2nd Call across network
13. [P090902001/ALL/ALL] Set Relocate does not work between SLT extension and Virtual SLT extension
14. [P090811001/ALL/ALL] OS7200 MCP Set Relocation

15. [P090818016/ALL/ALL] E-LCR & Standard SIP phones
16. [-/ALL/USA,RUSSIA,KOR] Changed new Keypad's License policy  
Changed new Keypad's License policy  
These new keypads are IP2200, IP3100, IP5220, IP5240, IP5243, WIP5120  
In case of USA and Russia, you can use these keypads without license.
17. [-/ALL/ALL] DTMF sending through SIP trunk for dialing  
If DTMF type is outband, It's possible to send DTMF through SIP trunk for dialing.
18. [-/ALL/ALL] SIP stack failure on the Plant Office system  
Reason : The failure stopped all outgoing SIP messages from the OfficeServ, no responses were sent to REGISTER messages from WIP Phones, or from the IP-UMS. Outgoing SIP trunk calls also failed with the message "No User Response" displayed on the calling phone. The OfficeServ did not send any INVITE messages for the outgoing trunk calls.  
  
The SIP functionality was restored by performing a reset of the system in IT Tool menu item: 6.3.1  
  
This problem occurred twice, and appears to have been caused by changing the Peer CLI table parameter in the 5.2.12 SIP Stack/Ext/Trunk Options screen. The value of the Peer CLI table parameter was changed from 1 to 2. Changing it back to 1 did not fix the problem, only a reset solved the problem.  
  
Modification : Whenever trying to send any SIP message for UMS interop, SIP module checks whether the callleg exists or not. When not found, simply ignores the send request from Call module.
19. [-/ALL/ALL] SIP Registration CALL ID bug  
Reason : REGISTER messages for multiple users must have unique Call-ID but OfficeServ has same Call-ID for multiple users.  
  
Modification :
  - 1.If Registration status is UnReg, each regClient detach from regClientMgr, so it can be possible for each regClient to use unique Call-ID and CSeq.
  2. If Registration status is Reg, Call-ID and CSeq are stored and managed for each user.
20. [-/ALL/ALL] New IPP(5220,5243) LED Color Control  
Modified to send both led state and color information when connecting with New IPP(5220,5243).
21. [-/ALL/ALL] Sometimes, call log doesn't save caller name  
The caller name was showed on the incoming call but sometimes it showed " Unknown" in the call log.  
  
Modified to send call log information again after receiving FACILITY msg.
22. [P090826001/ALL/ALL] No light indication for DT key that refer to SIP trunk

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**09.09.09 V4.42**

1. Nursing System (Emergency Message)  
Emergency messages received from CTI are displayed at the designated devices. The available target phones are limited to two-line DGP/IPP and WIP.  
  
The configuration and operation for Nursing System are described as follows
  - 1) For target phones, assign EMERGENCY KEY using IT 4.9.2.(MMC722/723)
  - 2) If a target group is needed, make EMERG GROUP and assign available members to the group with IT 4.1.1 (MMC601). Only two-line DGP/IPP and WIP is available.

- 3) At IT 5.14.8(MMC501), you can set Emergency Clear Time and Emergency Retry Time. Emergency Clear Time is used to remove the message from the queue. For WIP, message re-send is supported by Emergency Retry Time.
- 4) When the emergency message is received from CTI, the target phones display the message with lamp and beeping.
- 5) By confirming with the Emergency Key, lamp gets OFF. In the case of DGP/IPP, the emergency message is remained until the state of phone is changed.

**09.07.13 V4.41**

1. SMDR Buffering

SMDR buffer required because if for some time PC or Printer is off due to any reason we can not get the call details of that period.

Responding to PC/PRINTER, SMDR buffering is modified as follows.

- (1) SMDR~PMS which provide PC/PRINTER service can buffer.
- (2) Buffering for PC is 300 and all services can use these buffer.
- (3) Buffering for PRINTER is 126 and all services can use these buffer.
- (4) If buffer is full, the oldest data will be erased.
- (5) Data is saved as files for the case that PC or Printer is off. And the storing period is 15 seconds. (Only modified data will be stored.) So be sure that data which is stored during 15 seconds might be loss.
- (6) Buffer is managed by Linked List, error is detected and corrected every 5 minutes .

2. Change Default Key value. (5014D)

Default Key value 5014D

```
/*-----*  
* DT701 DT702 DS203 DS204 DS205 DS206 VMMSG *  
* DT703 DT704 DS207 DS208 DS209 DS210 CALL1 *  
*-----*/
```

3. After initializing DB for VM in Web, the system sometimes doesn't restart automatically. There was a bug in initializing VM. The initializing module could not check the acknowledge of VM sometimes.

4. [Only STA] Extending the VM recording time

To extend the VM recording time, OS7030 has 3 VM prompt(USE,USF,USS) in USA. So OS7030 uses 14 hours as the VM recording time.

(Other countries still has 16 VM prompt and about 8 hours as the VM recoding time.)

We'll apply to this when S/W and VM prompt are saved in the NAND Flash Memory in Factory. So now you can't check it. If you want to check it, please execute the following.

1. Restart system after reserving the following S/W and VM prompt using Web/IT.

Other unnecessary package and VM prompt must be deleted.

- 1 package in each CS/MS/AP/DR/RD/RT/WS package
- 3 VM prompts(VM\_L\_USE.tgz, VM\_L\_USF.tgz ,VM\_L\_USS.tgz)

2. Check the VM recording time in VM Web menu.

- Disk Space Available in Status Screen page.

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**09.07.03 V4.40**

1. [-/ALL/SEI] Wrong phone LCD display is fixed.  
When the call was connected and there was COLP from network side, Called party number was displayed instead of COLP.  
Now, if there is COLP then COLP will be displayed. And if it is longer than 8 digits then first 8 digits will be displayed on the phone LCD.  
If no COLP then CPN(Called Party Number) will be displayed. And if CPN is longer than 10 digits then last 10 digits will be displayed.
2. [-/ALL/SEI] H.323 suspend is prevented by using Automated Defect Prevention Tool.
3. [-/ALL/SEUK] Call Transfer Fail through SIP trunk.  
Now, when the incoming call is transferred, MP will send the UPDATE\_RQ. Then there will be REINVITE message to network side.

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**09.06.16 V4.40****[Qnet code/System/Country]**

1. [P090605003/ALL/ALL] Busy Tone detection Issue on Analog Trunk  
- System often decodes women's voice as a busy tone. There is a calculation error when checking busy tone cadence. We modify busy tone calculation scheme.
2. [-/ALL/ALL] VM/AA sometimes cannot download station name, if it is max length. It's fixed.
3. [-/ALL/ALL] New license key is applied for email gateway license.
4. [-/OS7070/ALL] If AA(391~394) hears fax tone, VM/AA hangs. It's fixed.
5. [-/OS7100,OS7030,OS7070] Max subscriber(mailboxes) is limited to 256 instead of 1000.
6. [-/OS7100,OS7030,OS7070/ALL] Email gateway failure causes VM/AA stops to operate while telnet VM CLI is logged on. It's fixed.
7. [-/ALL/ALL] When a user log on his mailbox with some ip phone, he hears "Nter your password" instead of "Enter your password". It's fixed to insert delay before playing first prompt for ip phones.
8. [-/ALL/INDIA] Disable confirm/Barge tone in case of supervised transfer.
9. [-/ALL/INDIA] Caller ID Feature Name/Number Display  
- DTMF CID doesn't have valid time and name information. So if user sets cid type to DTMF and inserts cid name in MMC728, system should transfer null data to cid name of MMC728.
10. [-/ALL/INDIA] Abandon Call List  
- DTMF CID doesn't have valid time information. So system should transfer null data to system's own time information.
11. [-/ALL/ALL] One-way issue in case of Executive Barge-In (=Override).
12. [-/OS7030, OS7070/ALL] H.323 COUNT option is displayed in MMC860 even though OS7030,7070 doesn't support H.323.
13. [P090606002/ALL/Russia] WLI cards and Combo AP support. OS7000 V4.21c did not support the WLI cards and Combo base stations.
14. [-/ALL/ALL] ITP-i2200 and SMT-W5120 are supported
15. [N090505007/ALL/ALL] SIP Trunks & H.323 Trunks Security – A feature that OfficeServ system rejects a trunk call from an invalid source node. If an IP address of the source node is not set in MMC833, MMC 836, or MMC837, OfficeServ will reject the call. This feature operates only if "ALLOW GW CHECK" in MMC 834(in case of H.323 trunk call) or "EXCLUSIVE" in MMC 837 > EXT (in case of SIP trunk call) is set to ENABLE.

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**09.05.26 V4.40**

1. The problems which were reported during field test are fixed. Please refer to issue list.

- A. OS7x00\_V440\_SEI\_Issue\_List\_(English).xls



V440\_SEI\_Issue\_List.xls

- B. Milestone of launching New ITPs\_090522.xls



Milestone of launching ITPs\_09

#### 09.05.09 V4.40

##### [Qnet code/System/Country]

1. [-/ALL except MCP/ALL] R1. MOBEX\_P2. Mobex Executive Feature by SIP TRUNK
  - In case of OS7200 MCP, this feature is allowed in EU, UK and SEG only.
  - System supports MOBEX Executive Feature in case of SIP Trunk Call.
  - RFC2833 DTMF Detection
2. [-/ALL/ALL] R6.1. User License : Service License Key includes below options.
  - SPNET License : To use VOIP SPNET
  - IPP License : To register new IP phone (ITP3100/ITP5220/ITP5240/ITP5243)
  - WIFI License : To register new WiFi phone (New WiFi phone is under development)
  - CMG License : To use call manager feature in communicator. This feature is only support in below IP phones  
(Soft Phone, ITP3100/ITP5220/ITP5240/ITP5243)
3. [N090124005/7200,7400/ALL] R6.5.4 Introduction of "Answer Delay Time for Ring Back Tone" feature
  - For an incoming trunk call, the VM ring back tone will be heard after VM RBK DELY TM (MMC 501).
4. [N090204003,N090204004/ALL/ITALY] Change Default Values
  - MMC861 AUTO UPDATE TIME Default Value => DISABLE
  - MMC861 ISDN PROG IND Default Value => DISABLE
  - MMC419/420 TIME SYNC => OFF
5. [-/ALL/ALL] IPP IDLE MODE FEATURE IMPROVEMENT
  - In case of Forced Logout, it is limited by station status or Class Of Service.
  - In case of Forced Logout, phone displays the reason. (logout by other user/logout by admin)
  - ALL IP phones can logout by setting logout time or KMMC (or IT).
6. [-/ALL/ALL] SIPFXO LOCAL Emergency Routing Function Improvement
  - Emergency call can be routed by Emergency code (also emergency dial number).
  - IP phone determines LCR routing table by Emergency code (in MMC330). DGP can determine that by Emergency dial number (ex. 911).
  - WIP phone supports this feature.
7. [N081008003/ALL/ALL] CLIP DISPLAY options changed. (IT5.15.3/KMMC119)
  - RING LINE1 : When incoming call is ringing, the specific information is displayed in the upper line of the phone display.  
(NO DISPLAY, CLI NUMBER FIRST, CLI NAME FIRST, DDI NUMBER FIRST, DDI NAME FIRST, GROUP NAME FIRST, DDI NUMBER/NAME, DDI NAME/NUMBER)
  - RING LINE2 : When incoming call is ringing, the specific information is displayed in the lower line of the phone display. RING LINE2 has the same options as RING LINE1.
  - CONV INCOM: When incoming call is answered, the specific information is displayed in the upper line of the phone display.  
(CLI NUMBER FIRST, CLI NAME FIRST, DDI NUMBER FIRST, DDI NAME FIRST, CLI NUMBER/DDI, CLI NAME/DDI, DDI NUMBER/NAME, DDI NAME/NUMBER, TRUNK NUMBER)
  - TRUNK DISP: When call is answered, trunk number can be displayed or not. This option was moved from KMMC110.  
(TRUNK:DIGIT, DIGIT ONLY)

- TIME/COST : When call is answered, duration time or cost can be displayed or not. This option was moved from KMMC110.

(NO DISPLAY, TIME DISPLAY, COST DISPLAY)

8. [N080211010/ALL/ALL] IP options changed.
  - 1) [Public] IP options deleted => use 2 option: [Private] or [Private with Public].
  - 2) IP type option of IP phone is deleted and automatic checked by system.
  - 3) MPS or MGI always used when the call between IP phone and IP phone which are all under NAT.

Changed MMC: MMC830 [IT2.1.2] : SYSTEM IP TYPE => PUBLIC IP ONLY deleted  
MMC831 [IT2.2.2] : MGI IP TYPE => PUBLIC IP ONLY deleted  
MMC843 [IT2.2.15] : MPS IP TYPE => PUBLIC IP ONLY deleted  
MMC820 [IT3.3.1] : SPNET IP TYPE option deleted  
MMC840 [IT2.7.1] : IPP IP TYPE & PUB TO PUB options deleted

% System which exists in public zone should set to PRIVATE ONLY including MGI and MPS.

9. [-/ALL/ALL] WIP Mode Display Added at MMC 846 LOCATED option.  
"DETACH", "SIP MODE ATTACH", "OSPP MODE ATTACH"
10. [N081114009/ALL] New IP Phones like SMT-i5220, i5243, 5264 can be connected to OfficeServ system  
Changed MMC : MMC110 USE STN RING  
MMC844 UC IPP INFO (XML,, LDAP, SNMP information)
11. [N080211006/except 7030/STA] PRI outbound CLID blocking
12. [N080211007/except 7030/STA] PRI CID Date and Time Stamp from Telco
13. [N080103005/ALL/SEUK] Redirection number (PCR150)
  - In case of diverting call via MOBEX extension, the CLI number of MOBEX extension should be the designated number by CO.
14. [N080128007/ALL/SER] "Redirection From" field in ISDN signaling adding
15. [N080122022/ALL/SEI] Add 3PTY conference Telecom Italia standard service
16. [N071110002/except 7030/KOR] Support PRI Booking function
17. [-/ALL/ALL] SIP early media handling Improvement; we support SIP early media services to following stations; DG phone / IP phone / SIP phone when MPS is either enabled or disabled.
18. [-/ALL/ALL] Maintenance of information about registration of standard SIP phones: The new feature makes signaling port number remain in system DB.
19. [-/ALL/ALL] Feature to handling the "423 Interval Too Brief" message : When the OfficeServ system tries to send REGISTER message to SIP server with very short expire time, most of the SIP servers will usually send 423 message. This feature handles this message and normally registers with proper expire time.
20. [-/ALL/ALL][eMGI] RFC 2833 Detection Feature in SIP Trunk supporting MOBEX
21. [-/ALL/ALL][eMPS] RFC 2833 Detection Feature in SIP Trunk supporting MOBEX
22. [-/ALL/ALL][eMGI] Modified the routine related to G.711 codec type due to codec mismatch when MPS is enabled.
23. [-/OS7100 except MP11, 7070/ALL]MSP firmware Upgrade (v6.14 → v7.17)
24. [-/ALL except MCP/ALL] Feature to make it possible to use SIP phones under NAT in OfficeServ System : Owing to ability of SIP Stack getting the IP address and port number the access point send from, the signaling and RTP communication with SIP phones under NAT become possible. This feature work only when MPS is set to enable.
25. [-/ALL/ALL]Video Phones is applied using MPS
  - \* A kind of Video Phone
    - : SIP video Phone, ITP Video Phone, and Call Communicator
  - \* Video stream's MPS using
    - NO MPS : Video call between Private Network's video phones.  
(directly send & receive video stream between video phones)
    - MPS : All video calls excluding between Private Network's video phones.
  - \* Limitation
    - Video call is only possible between video phones.
    - In case of OS7400, OS7200 System, you must have OAS card for MPS.
    - If you make a video call using remote video phone (not same network to system's network), you must have MPS.

- Make a call from video phone to non video phone, and transfer to video phone. Then video stream is not available.
- 26. [-/7100/ALL] System default IP address is changed to "10.0.2.1" from "165.213.176.100"
- 27. [-/MP20/KOREA] OS7200 (only MP20) supports VCS functions.
- 28. [-/OS7070/ALL] BRI feature is available.
  - S/T modes are supported.
  - Install 2BM D-b'd on B8S/E8S LOC1,2,3 slot.
  - Connect the specific 2 port-cable to B8S/E8S port to use BRI service
- 29. [-/OS7030 Boot/ALL] System sets default IP address by setting DIP switch 6 to ON.
  - Default IP address : 10.0.2.10(Standalone, Master), 10.0.2.12(Slave)
  - Server IP address : 10.0.2.11
- 30. [-/OS7100, 7070, 7030/ALL] System accepts only one telnet access.
  - Prevent multiple telnet access.
  - System denying telnet access which is port number 23 but one.
- 31. [-/ALL except MP11/ALL] Implemented PCR: Multiple Public Address/ WAN Links for different services.
- 32. [-/ALL/INDIA] Change INDIA default values
  - Default operator code digit: 9
  - Default 1st trunk group: 0
  - Default Ring Mode of station group 500: Unconditional.
  - Default DSS KEY DPU in MMC 210: ON
- 33. [-/ALL/ALL] Add IR Flag in MMC 725.
- 34. [-/OS7070/INDIA] Add DTMF CID Type in MMC 809.
- 35. [-/OS7070/ALL] SMDR Buffering.
- 36. [-/ALL/ALL] Change the display in case of using ISPY key.
- 37. [-/ALL/ALL] Change the display in case of using CCBS/CCNR.
- 38. [-/OS7030 Boot/ALL] When system upgrades CSP/MSP/IRD packages using Web/IT, CSP/MSP/IRD packages are not upgraded according to booting side.
  - Boot program sets the start address of CSP/MSP/IRD according to booting side. But old boot program always sets side #1 as start address of CSP/MSP/IRD.
  - If system upgrades CSP/MSP/IRD in side #2 and starts using side #1, old CSP/MSP/IRD package runs.
  - In case TFTP upgrade in boot mode, this problem doesn't happened.
  - **If you use old Boot(under V4.40) program, you must upgrade V4.40 CSP/MSP/IRD twice manually.**
  - If you use V4.40 Boot program, you upgrade V4.40 CSP/MSP/IRD package once.
  - [Remark] After upgrading boot program, system must upgrade CSP/MSP/IRD package using Web/IT.
- 39. [N090126007/ALL/ALL] OfficeServ Multiple WAN interface when using SIP, ITP's and SPNET
  - There are three PUBLIC IP SETs and user can select PUBLIC IP CATEGORY.
- 40. [-/ALL/ALL] Email gateway in OS7100 fails to contact SMTP server too many times it causes SVM application crash and cannot restore without rebooting switch. It's fixed.
- 41. [OS7100, OS7030, OS7070] [Web] Web management interface will show changes to recorded greetings.
- 42. [Web] Backup DB of VM/AA has failed in VM/AA->Operating Utilities->DB backup page. It's fixed.
- 43. [Web] Subscriber import feature has failed in VM/AA->Operating Utilities->Subscriber import page. It's fixed.
- 44. [Web] <OS7070 AA Timer, MMC501>  
Add options in 5.14.9 Auto Attendant Options.
  - AA Inter Digit Time  
Range = 1~25
  - AA No Act Time  
Range = 1~25
  - AA Trans Time  
Range = 0~25
  - \* OS7070 Only for all options
- 45. [Web] <OS7070 AA Password, MMC202>  
Add options in 5.13.1 Tenant Options.

- AA Record Password  
Default = 4321
- 46. [Web] <SMDR Format, MMC725>  
Add options in 5.6.3. SMDR Options.
  - Call Index  
Range = No/Yes
- 47. [Web] <ISDN Trunk, MMC419/420>  
Add options in 2.6.7 BRI Trunk Options and 2.6.8 PRI Trunk Options.
  - Diverting Information  
Range = Off (0), On (1), Except Called Number (2)
  - Centrex Service  
Range = Off(0), On(1)
- 48. [Web] <Use Station Ring, MMC110>  
Add new options in 5.15.4 Keypad On/Off.
  - Use Station Ring  
Range = Off (0), On (1)
- 49. [Web] <MPS Status>  
Add new options in 6.2.7 MPS Status.
  - Add range for Codec  
Org : G.711, G.723, G.729A, G.729, T.38, G.722, G.711U, G.711A  
Add : H.263, MPEG4, H.264  
Read Only
- 50. [Web] <License, MMC860>  
Add new options in 1.1.2 Quick/License.
  - IP Phone Max
  - WiFi Phone Max
  - SPNet Feature (Disable/Enable)
  - Call Manager (Disable/Enable)  
Read Only
- 51. [Web] <License, MMC860>  
Add new options in 2.1.4 License Key.
  - IP Phone Max
  - WiFi Phone Max
  - SPNet Feature (Disable/Enable)
  - Call Manager (Disable/Enable)  
Read Only
- 52. [Web] <Emergency Code, MMC330>  
Move Emergency Code feature from 2.7.1 IP Phone Information to new 5.15.15 Emergency Code.
  - Emergency Code  
Range = 0~9, \*, #, P(Pause)
  - Emergency Dial  
Range = 0~9, \*, #
- 53. [Web] <SIP Options, MMC837>  
Add new options in 5.2.13 SIP Carrier Options.
  - NAT Registration Expire Time  
Range = 0~999999
- 54. [Web] <Class of Service, MMC701>  
Add new options in 4.7.2 COS Contents.
  - ITP Forced Log-out  
Range = No/Yes
  - Call Manager Use  
Range = No/Yes
- 55. [Web] <ITP Idle Mode, MMC841>  
Add new options in 5.2.10 System ITP Options.
  - ITP All Log-out > Type  
Range = MMC Command (0), Auto Time (1)
  - ITP All Log-out > Time  
Range = HH:MM

- 56. [Web] <ITP Idle Mode, MMC841>  
 Add a new 6.3.14 ITP Idle Log-out.  
 - Log-out Now  
   Range = No(0), Yes(1)  
 \* ITP Phone should be operated in idle mode.  
 \* COS of ITP Forced Log-out should be enabled.
- 57. [Web] <ITP Idle Mode, MMC840>  
 Add a new 6.3.15 ITP Forced Clear.  
 - Current Status  
   Range = No, Registered  
 - Phone Version  
 - Clear/Upgrade  
   Range = No, Clear Registration, SW Upgrade  
   Moved from 6.2.2 ITP Status  
 - Forced Log-out  
   Range = No, Yes  
 For forced log-out:  
 \* ITP Phone should be operated in idle mode.  
 \* COS of ITP Forced Log-out should be enabled.
- 58. [Web] <WIP Status, MMC846>  
 Add new options in 6.2.5 WIP Status.  
 - Add range for Located  
   Old : Detach, Attach  
   New : Detach, SIP Mode Attach, OSPP Mode Attach  
   Read Only  
 - Add range for Phone Type  
   Old : WIP-500M, WIP-5100M  
   New : WIP-500M, SMT-W5100, SMT-W5120  
   Read Only
- 59. [-/OS7030/ALL]  
 When switching OS7030 off/on consecutively, sometimes some module is not working properly. So the system fails to boot properly. It's fixed.
- 60. [-/OS7030/ALL]  
 When switching OS7030 off/on many times continuously, system sometimes restarts by watchdog. It's fixed.
- 61. [-/OS7400 Boot/ALL]  
 Old boot can load MP when the media card (MMCplus, SD card) has files below 20.  
 As media card capacity increases, we modified that max number of files is 100.

**Table A. QNET N-issue**

QNET ID	Title	System	Country	Remarks
N080211006	ISDN PRI Outbound CLI Blocking	OS7400	STA	Applied to V4.40
N080211007	PRI Date and Time stamp from Telco	OS7200	STA	Applied to V4.40
N080211007	PRI CID Date and Time Stamp from Telco	OS7200	STA	Applied to V4.40
N080211010	ITP auto relocate	OS7200	STA	Applied to V4.40
N080103005	Redirection number (PCR150)	OS7200	SEUK	Applied to V4.40
N081115001	CID on Atrk (allow unrecognised characters)	OS7100	SEUK	Applied to V4.40
N081008003	IMPROVED FUNCTION OF SVMi20 - TRANSFERRED CALLS TO SHOW DDI NUMBER AND CLI	OS7400	SEUK	Applied to V4.40
N080122022	Add 3PTY conference Telecom Italia standard service.	OS7200	SEI	Applied to V4.40
N090204003	Change default value of Auto Update ISDN Time to DISABLE	OS7100	SEI	Applied to V4.40

N090204004	Change default value of Progress Indicator Call Clear to DISABLE	OS7200	SEI	Applied to V4.40
N090124005	Introduce an answer time delay for Ring Back Tone feature like for DISA ANSWER TIME	OS7100	SEI	Applied to V4.40
N080128007	"Redirection From" field in ISDN signaling	OS7200	SER	Applied to V4.40
N081114009	SMT-i5220, i5243, i5264 phones are applied	-	South Africa	Applied to V4.40
N081114009	New IP phones (SMT-i5220, i5243, i5264) are add to OS 7000 systems	OS7000 systems	South Korea	Applied to V4.40
N090126007	OfficeServ Multiple WAN interface when using SIP and ITP's/SPNET	OS7000 systems	STUK	Applied to V4.40
N090505007	SIP Trunks & H.323 Trunks Security	OS7000 systems	SEUK	Applied to V4.40

**Table B. QNET P-issue : V4.40 contains the issues that fixed in T4.30h**

No	Issue	Country	Priority
1	Hotel/Motel feature : when check-in keyset that is station paired to another keyset, the paired set does not have restriction set	STA	Major
2	Hotel/Motel feature : Checked in keyset that has restriction set does not error when attempting to dial external number	STA	Major
3	RP keys behave incorrectly when MMC 507 has no entry	STA	Major
4	(P090413010 ) Cannot upgrade TEPRIa/2 software via MMC818 when TEPRIa/2 card installed in OS7100 cabinet	STA	Major
5	There is no way of converting the 12 Ports with older software to the 20 Ports with newer software on the SVMI-20E when 8VPM card installed in Expansion cabinet.	STA	Major
6	(P090408001)SIP Refer message not setting FROM Field correctly Using an OfficeServ configured to use Centrix in MMC837 If you transfer an external call from a SIP extension to another external number, the "FROM" field in the "Refer" message is missing the last digit of the initially dialed number. In the attached trace packet 16 & 20 show the call initially being setup to "07879405420" (TO Field) however when the refer message is sent in packet 23 to complete the transfer the "FROM" field is incorrect and is missing the last digit of the number "0787940542"	SEUK	Major
7	(P090326005) Phones continue to ring when Using OS connect and UCD groups If call is rejected by mobex ext the other keysets in the UCD group continue to ring and if you try to pickup the call by dialing 65 and the grp number you receive "no ringing call can't pickup. If you change the UCD group to normal grp it works ok. Tested on OS7030 and OS7400 V4.30d in UK and Sweden	SEUK	Major
8	(SEUK) some of CID digit will not be displayed in Analogue Trunk (OS7030)	SEUK	Major
9	(P09032004)BLF not working correctly on 64b AOM in OS7200 cabinet when OS7200 cabinet is used as the expansion cabinet of a 7400	SEUK	Minor
10	(P090408008) RP key's led doesn't match with the current ring plan	SEI	Minor
11	(P090323003) VM MoH (KMMC756) and VM In/Out (KMMC757) features are not available by I/T 2.2.3	SEI	Minor
12	(P090303004) Trunk Auto Answer MoH feature does not work properly for incoming DDI calls to stations	SEI	Minor
13	(P090219003) MG116/64's IP address and subnet mask cannot be set up	SEI	Major

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	properly		
14	(P081204002) The system inserts an improper "0" to the analog CID received from Mobile phone number	SEI	Minor
15	(P081204002) The system inserts an improper "0" to the analog CID received from Mobile phone number	SEI	Minor
16	Kirk IP-DECT issue	PC	Major
17	(P090415002) AUTH Code does not change COS	PC	Major
18	(P090409001) SIP DDI routing (SITP TRK DDI Number/Name error)	GTL	Minor
19	(P090408004) RP Keys issue	GTL	Major
20	(P090408003) Hotel BD-PMS Not working on OS7100/7030	GTL	Major
21	Key programming change in IT 4.9.1 (MMC723) will update all the types of phones	GTL	Minor
22	OS7200 v4.30a the MMC503 MF/DP INT timer not applies	SER	Minor
23	User can hear Dial Pulse on SLT. It should be muted	SER	Minor
24	Call drop issue during the conversation when busy tone detection is set for Analog TRK	SER	Major
25	(P090605003) Call drop on Analog Trunk	LSP	Minor
26	(P090606002) WLI cards and Combo AP support	SER	Major

**Previous Version Changes****09.03.02 V4.30d**

1. [7400] IT 2.1.1 Says MODEM Exist: NO, but I have MODEM card installed and working for OS7400.
2. [-/ALL/ALL]System failed to send MWI message to SIP station
3. [P081222006/ALL] Blind Transfer from Sip Ext to sip ext with Call forward
4. [P081008006/ALL] Chain Fwd cannot be turned off for remote networked site.
5. [P080923002/ALL] CALL Waiting CLID Issue
6. [P081029007/7400] OS7200 3RD EXPANSION
7. [VMWEB/7100,7030]Web Management Voice Studio NEXT and PREVIOUS button not working.
8. [VMWEB/7100,7030]Voice studio SAVE or SAVE & EXIT function not working
9. [VMPMT/STA]The English prompts are changed.
10. [NEWZEALAND] In case of unconditional group ring by trunk call, the DTS led of the extension which is ringing becomes red on or green on (green flash is normal)
11. [N090204001/STA]Clear and default the SIP profile contents in MMC 837
12. [P081113015/ALL] SIP stations receive but cannot make any call
13. [P081024004/7100] MGI Public IP Address does not update after upgrade .
14. [P081023003/ALL] If the message is deleted without reading it, it remains in WIP memory.
15. [P080701010/ALL] VM Notify/alert notification feature will always use any outbound port to dial out.
16. [P080923005/7200] Incoming SIP calls fail on OS7200 SIP trunks with Telia
17. [P081023002/ALL] DND ON Soft Menu doesn't work properly when disabled by mean feature code "400"
18. [P081029006/ALL] H.323 call does not work when Gatekeeper RAS type is set to Manual in MMC836
19. [P081115002/7100] Problem with 7100 and Fax call routing of subsequent voice call..
20. [P081021001/ALL] CALL ROUTING IPUMS ISSUE
21. [P08092500/7100] 7100 VMS Call progress detection does not detect answer on some cell phones.
22. [P081202003/7400] DS VOICE MAIL LAMP
23. [P081025003/ALL] UCD REPORT LONGEST QUEUE TIME
24. [P081206002/ALL] IT Tool User level restrictions not restricting access.
25. [SEG/7400]Send Busy cause when using ISDN trunk. (SEG MP40 Request)
26. [P081125005/ALL]In OS Call and OS Operator if you set DND status, the keyset light does not flash on iDCS keysets.
27. [7030/SEI]In case 2BM card is installed in Slave cabinet, if a user tries to send/receive the fax through that BRM card, the fax does not work. (SEI reported)
  - Reason : The 2BM Card in installed in the Slave cabinet, the reference clock is not working. By this reason, the fax does not work in the Slave cabinet.
  - Result : The reference clock is working in the Slave cabinet.
28. [STA] STA TEST REPORT ANSWER -> Refer to the attached excel document.
29. [P090108004/ALL]Can't perform announced transfers directly to a sip ext when it is paired with another ext
30. [P081216001/ALL] Ticket 4136 – OS Operator Not Updating Virtual ports



STA\_Test\_Report  
\_Ans\_090212.xls

**Previous Version Changes****'09.01.30 OS7030 V4.30c**

1. [7400] IT 2.1.1 Says MODEM Exist: NO, but I have MODEM card installed and working for OS7400.
2. [-/ALL/ALL]System failed to send MWI message to SIP station
3. [P081222006/ALL] Blind Transfer from Sip Ext to sip ext with Call forward
4. [P081008006/ALL] Chain Fwd cannot be turned off for remote networked site.
5. [P080923002/ALL] CALL Waiting CLID Issue
6. [P081029007/7400] OS7200 3RD EXPANSION
7. [VMWEB/7100,7030]Web Management Voice Studio NEXT and PREVIOUS button not working.
8. [VMWEB/7100,7030]Voice studio SAVE or SAVE & EXIT function not working
9. [VMPMT/STA]The English prompts are changed.
10. [NEWZEALAND] In case of unconditional group ring by trunk call, the dts led of the extension which is ringing becomes red on or green on (green flash is normal)
11. [N090204001/STA]Clear and default the SIP profile contents in MMC 837
12. [P081113015/ALL] SIP stations receive but cannot make any call
13. [P081024004/7100] MGI Public IP Address does not update after upgrade .
14. [P081023003/ALL] If the message is deleted without reading it, it remains in WIP memory.
15. [P080701010/ALL] VM Notify/alert notification feature will always use any outbound port to dial out.
16. [P080923005/7200] Incoming SIP calls fail on OS7200 SIP trunks with Telia
17. [P081023002/ALL] DND ON Soft Menu doesn't work properly when disabled by mean feature code "400"
18. [P081029006/ALL] H.323 call does not work when Gatekeeper RAS type is set to Manual in MMC836
19. [P081115002/7100] Problem with 7100 and Fax call routing of subsequent voice call..
20. [P081021001/ALL] CALL ROUTING IPUMS ISSUE
21. [P08092500/7100] 7100 VMS Call progress detection does not detect answer on some cell phones.
22. [P081202003/7400] DS VOICE MAIL LAMP
23. [P081025003/ALL] UCD REPORT LONGEST QUEUE TIME
24. [P081206002/ALL] IT Tool User level restrictions not restricting access.
25. [SEG/7400]Send Busy cause when using ISDN trunk. (SEG MP40 Request)
26. [P081125005/ALL]In OS Call and OS Operator if you set DND status, the keyset light does not flash on iDCS keysets.
27. [7030/SEI]In case 2BM card is installed in Slave cabinet, if a user tries to send/receive the fax through that BRM card, the fax does not work. (SEI reported)
  - Reason : The 2BM Card in installed in the Slave cabinet, the reference clock is not working. By this reason, the fax does not work in the Slave cabinet.
  - Result : The reference clock is working in the Slave cabinet.
28. [STA] STA TEST REPORT ANSWER -> Refer to the attached excel document.
29. [P090108004/ALL]Can't perform announced transfers directly to a sip ext when it is paired with another ext
30. [P081216001/ALL] Ticket 4136 – OS Operator Not Updating Virtual ports



STA\_Test\_Report  
\_Ans\_090212.xls

## Previous Version Changes

'09.01.30 OS7030 V4.30c

### (1) MCP S/W changed items

1. RUN LED is turned off for memory clear time (about 1 to 3 seconds) when the memory clear by pressing the reset button over 7 seconds in order to let the user or technician recognize the memory clear. It takes about 1 to 3 seconds because the memory clear time varies per DB size. **(Improvement)**
2. 2BM cannot be served after restart from IT 6.3.1 **(SEI reported)**
  - Reason : MP does not send the card reset message to 2BM card when user restart it by IT tool.
  - Result : MP sends the card reset message to 2BM card.
3. In case 2BM card is installed in Slave cabinet, if a user tries to send/receive the fax through that BRM card, the fax does not work. **(SEI reported)**
  - Reason : The 2BM Card is installed in the Slave cabinet, the reference clock is not working. By this reason, the fax does not work in the Slave cabinet.
  - Result : The reference clock is working in the Slave cabinet.
4. When outgoing calls are made via SIP Trunk from either ITP phone or SIP phone and MPS is activated, there is no speech. When OS7030 receives '183 Session Progress Message' from the opposite through SIP Trunk, there is an error that the SIP task transmits "Port" and "Codec" to Call task. **(Plant Communication reported)**
  - Reason :
    - 1) A port number was transmitted incorrectly because the SIP Task didn't consider "Little Endian".
    - 2) SIP Task does not transmit Codecs of "183 message".
  - Result : We have added a part that processes correctly for both cases.
5. After MP DB is cleared, the system does not display the version information. **(SEC Outgoing inspection team reported)**
  - Reason : During clearing MP DB, OS7030 system uses "sync" function. But if system reset when "Sync" process is not finished, some files can be deleted.
  - Result : It was fixed by finishing the "sync" process before system performs reset process by removing unnecessary and time consuming routine while clearing MP DB
6. When system restarts by power recycle, sometimes the booting fails because few task running abnormally. **(SEC manufacturing team reported)**
  - Reason : When tasks starts sleep condition during reading/writing NAND, it uses a kernel timer. However, the relevant tasks sometimes runs abnormally in case this kernel timer does not wake up on time.
  - Result : The system monitors that timer regularly to prevent this case.
7. The margin for reorder tone (congestion tone) for Italy was changed from +/- 10ms to +/- 30ms in order for the system to detect this tone correctly. **(SEI reported)**
8. VM Prompt noise is removed when Australian VM language file is used. **(GTL reported)**

In order to remove the noise, VM language file (VM\_L\_LSP.tgz) should be replaced by new one attached.

9. After testing t-switch in AFT mode, engine test fails. **(for the Manufacturing team)**
  - Reason : After testing t-switch in AFT mode, system does not recover register value
  - Result : It fixed that system recovers the register value after testing t-switch in AFT mode.
  
10. In AFT mode, it can't hear internal MOH when system country is USA. **(for the Manufacturing team)**
  - Reason : When system country is USA, default BGM source is external MOH.
  - Result : System sets internal MOH source to BGM for AFT IMOH.

## Previous Version Changes

08.12.18 OS7400 MP40 V4.30b

### OfficeServ 7400 MP40 S/W release

System	Package name	Description
OS7400	MPEV430B.PGM	MP S/W for MP40 card

#### Warning

1. The previous DB will be erased when upgrading system S/W to V4.30.
2. Only the system whose version is V4.2x or higher than V4.2x can be upgraded to V4.30. If the system version is lower than V4.2x, that system should be upgraded to V4.2x before being upgraded to V4.30.
3. In case of OS7100, it recommends that all previous files existing in SD/MMC card remove before upgrading to V4.30, especially 'sysver.inf' file. Because this file is including backup information. And this moved into the ramdisk file. If 'sysver.inf' file still remain in SD card, it can occur some problems.
4. SVMi20E software should be V5.2.1.1 (2008.11.06) or higher version in order to use MP V4.30

## S/W Compatibility Table:

### (1) OfficeServ 7400 S/W Version Compatibility table

System	OS7400 (MP40)
MP40	V4.30b (2008.12.18)
LP40	V1.20 (2008.12.08)
LCP	V4.16 (2008.12.05)
TEPRI2/TEPRIa	V4.24 (2008.06.05) V4.25 (2008.11.27)
MGI16/64	V1.24 (2008.11.27)
SVMi-20E	V5.2.1.1 (2008.11.06)
OAS	V1.00 (2008.11.27)
IT	V1.30a (2008.12.08)
OS Link	V3.0.0.4
IP-UMS	V1.3.2.14
Bootrom	V1.07 (2007.08.20)

## Previous Version Changes

08.12.18 V4.30b

## Fixed Bugs

31. System crashes down when SIP Station tries to Register. It's fixed.

## Previous Version Changes

'08.12.08 OS7000 series V4.30a

### OfficeServ 7000 series MP S/W release

System	Package name	Description
OS7400	MPEV430A.PGM	MP40 S/W V4.30a (2008.12.08)
	LP40V120.PGM	LP40 S/W V1.20 (2008.12.08)
	OAS1v100_20081127_pkg.rar	OAS card S/W V1.00 (2008.11.27)
	MGI16v124_20081127_pkg.rar	MGI16 card S/W V1.24 (2008.11.27)
	MGI64v124_20081127_pkg.rar	MGI64 card S/W V1.24 (2008.11.27)
OS7200	svmi20e_5211_xxx.zip	SVMi-20E card S/W V5.2.1.1 (2008.11.06) (xxx indicates customer name such as STA.)
	MPPS430A.ZPG	MCP S/W V4.30a (2008.12.08)
OS7100	LPPSV416.pgm	LCP S/W V4.16 (2008.12.05)
	MP10_V430a_20081208.zip	MP10 S/W V4.30a (2008.12.08)
	MP10A_V430a_20081208.zip	MP10a S/W V4.30a (2008.12.08)
OS7030	bt1av106.bin	MP10a Bootrom V1.06 (2008.11.24)
	7030_V430a_20081208.zip	OS7030 S/W V4.30a (2008.12.08)
IT	IT(081208)_V1.30a.zip	Installation Tool V1.30a (2008.12.08)

#### Warning

1. The previous DB will be erased when upgrading system S/W to V4.30.
2. Only the system whose version is V4.2x or higher than V4.2x can be upgraded to V4.30. If the system version is lower than V4.2x, that system should be upgraded to V4.2x before being upgraded to V4.30.
3. In case of OS7100, it recommends that all previous files existing in SD/MMC card remove before upgrading to V4.30, especially 'sysver.inf' file. Because this file is including backup information. And this moved into the ramdisk file. If 'sysver.inf' file still remain in SD card, it can occur some problems.
4. SVMi20E software should be V5.2.1.1 (2008.11.06) or higher version in order to use MP V4.30

## S/W Compatibility Table:

### (1) OfficeServ 7000 S/W Version Compatibility table

System	OS7400 (MP40)	OS7200 (MCP)	OS7100	OS7030
MP	V4.30a (2008.12.08)	V4.30a (2008.12.08)	V4.30a (2008.12.08)	V4.30a (2008.12.08)
LP40	V1.20 (2008.12.08)	N/A	N/A	N/A
LCP	N/A	V4.16 (2008.12.05)	N/A	N/A
TEPRI2/TEPRIa	V4.24	V4.24	V4.24	N/A

	(2008.06.05) V4.25 (2008.11.27)	(2008.06.05) V4.25 (2008.11.27)	(2008.06.05) V4.25 (2008.11.27)	
<b>MGI16/64</b>	V1.24 (2008.11.27)	V1.24 (2008.11.27)	V1.24 (2008.11.27)	N/A
<b>SVMi-20E</b>	V5.2.1.1 (2008.11.06)	V5.2.1.1 (2008.11.06)	N/A	N/A
<b>OAS</b>	V1.00 (2008.11.27)	V1.00 (2008.11.27)	N/A	N/A
<b>IT</b>	V1.30a (2008.12.08)	V1.30a (2008.12.08)	V1.30a (2008.12.08)	V1.30a (2008.12.08)
<b>OS Link</b>	V3.0.0.4	V3.0.0.4	V3.0.0.4	V3.0.0.4
<b>IP-UMS</b>	V1.3.2.14	V1.3.2.14	N/A	N/A
<b>Bootrom</b>	V1.07 (2007.08.20)	V1.07 (2007.05.08)	MP10: V1.01 (2007.05.11) MP10a: V1.06 (2008.11.24)	V4.30 (2008.09.22)

**(2) OfficeServ 7100/7030 Module version table**

<b>System</b>	<b>OS7100 (Except MP11)</b>	<b>OS7030</b>
<b>System</b>	V4.30a (2008.12.08)	V4.30a (2008.12.08)
<b>MP</b>	V4.30a (2008.12.08)	V4.30a (2008.12.08)
<b>SP</b>	V2.25 (2008.11.25)	V2.24 (2008.10.27)
<b>VM</b>	V2.78 (2008.11.27)	V2.78 (2008.11.27)
<b>MGI</b>	V2.01 (2008.11.27)	V2.01 (2008.11.27)
<b>BRI</b>	V4.21b (2008.10.14)	V4.21b (2008.10.14)
<b>WEB</b>	V4.10a (2008.12.05)	V4.10a (2008.12.02)
<b>MPS</b>	V1.00 (2008.11.04)	V1.00 (2008.11.04)
<b>Router(MP11)</b>	V1.09 (2008.11.27)	N/A

**Previous Version Changes****08.12.08 OS7000 series V4.30a**

32. [SEUK] New option added for MOBEX CLI issue. When incoming call routed to MOBEX, the MOBEX phone via SIP trunk didn't ringing in Sweden because of CLI TRANSFER feature.
  - In MMC861, TRUNK->MOBEX CLIP option added.
    - TO ISDN (RECEIVED, MASTER, MOBEX)
    - TO SIP (RECEIVED, MASTER, MOBEX)
  - In MMC837, RCV CLI FWD option added in TRK option.
    - ENABLE: The RECEIVED CLI will be inserted to SIP ALIAS field.
33. [SEUK] The MOBEX phone can use barge-in feature, with tone only. (not without tone)
34. [SEUK] Bug fixed. The VM holiday mode didn't work automatically issue.
35. [SEUK] Default value changed. If Country is UK, the default value of CID DSP ALLOC TM in KMMC501, change form 500 ms to 0 ms.

- 36. [SEUK] Default data changed. In KMMC751, the default value of ITP, WIP and SIP change from NONE to BOTH, because of the ITP's mailbox deleted issue when package upgrade.
- 37. [SEUK] Bug fixed. The BASIC WBS didn't work issue when package upgrade.
- 38. [STA] OS7100, Enhanced VM TUI slow issue.
- 39. [Plant Com.] Bug fixed. In OS7400, the LAN Lock up issue.(In case of DNS query fail )
- 40. [GTL] Bug fixed. In OS7400, the system restart issue when Fidelio PMS solution send bulk data.
- 41. [SEI] Bug fixed. In MP10a, the Fax call via BRI trunk failed issue.
- 42. [MP10a/MP11] Boot image changed. Because of system didn't upgrade CS/MS/RD package by KMMC 818 or IT/Web issue.
- 43. MP11only. The Router software upgrade to V1.09
  - GRE Tunnel over IPSec is provided.
    - Please refer to the GRE over IPSec configuration Guide Ed02 and GRE tunnel handbook.
    - Changed ZebOS protocol and Web GUI in order to support GRE Tunnel over IPSec in the transport mode.
    - Changed Routing Information to recognize GRE in the RIP and OSPF.
  - So MP11 supports following functions
    - Single GRE tunnel.
    - GRE Tunnel over IPSec.
    - GRE Tunnel over IPSec with OSPF
  - Support Dynamic DNS feature.
  - Added new features of the enhanced SIP-ALG feature.
    - Enhanced SIP-ALG can support SIP station in the public Area.
  - Fixed problem of the switch port status in the 4SWM
- 44. [P081023002] Bug fixed. SMT-W5100 DND improper work Because Soft Menu DND is not properly updating internal DND status field.
- 45. [P081029006] Bug fixed. H.323 call does not work when Gatekeeper RAS type is set to Manual in MMC836
- 46. 7100/7200/7400 1GB SD card compatibility issue => fully applied to V4.30

**Previous Version Changes**

**08.11.19 OS7000 series V4.30**

- 47. Added MOBEX(Mobile Extension) feature.  
Refer to the attached 'V4.30 Feature Guide.doc'.
- 48. Advanced IP call's media path control  
Refer to the attached 'V4.30 Feature Guide.doc'.
- 49. Advanced License control  
Refer to the attached 'V4.30 Feature Guide.doc'.
- 50. Advanced several minority features.  
Refer to the attached 'V4.30 Feature Guide.doc'.
- 51. WebMMC is not supported any more. (From V4.30 user can't access WebMMC.)  
Refer to the attached 'V4.30 Feature Guide.doc'.
- 52. Applied QNET new requirements.  
Refer to the following table A.
- 53. Applied patch issues form V4.21x, V4.22x, V4.23x and QNET P-issues..  
Detailed description skipped (see the following table B).

**Table A. QNET N-issue**

QNET ID	Title	System	Country	Remarks
N070720003	SIP EXTENSIONS	OS7400	GTL	Already supported from V4.2x
N071110002	booking feature with PRI	OS7200	Korea	Applied to V4.30
N081114006	Mobile extension	OS7200	Korea	See the MOBEX feature
N081114012	Hot Desk feature	OS7200	Korea	See the ITP idle mode

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N071206002	Support for external voicemail	OS7100	LSP	Already supported from V4.2x
N080312005	Best IP settings	OS7400	LSP	Applied to V4.30
N080512004	MOH feature added to 8TRK-16TRK card	OS7400	LSP	Applied to V4.30
N070215008	PMS interface	OS7200	MIA	Already supported from V4.2x
N070330011	Door phone	OS7200	MIA	Make use of the MOBEX feature
N061017034	01_SEG_FEATURE_REQUEST_CALL_PIC KUP_GROUP	OS7200	SEG	Already supported from V4.2x
N061105002	08_SEG_FEATURE_REQUEST_other ring tones type are needed	OS7200	SEG	Already supported from V4.2x
N061105003	035_SEG_BUG_Report_system_speeddial_more_letters_for_search_option	OS7200	SEG	Already supported from V4.2x
N080808004	SAZ requirement(2): Fixed Mobile Convergence	OS7400	SEG	See the MOBEX feature
N080612002	OS system does not send full information of Calling Party Number through SIP and H.323 trunk	OS7100	SEI	Applied to V4.30
N081023010	Make available the external call forward feature from door phone calls like for station/PSTN calls.	OS7100	SEI	Make use of the MOBEX feature
N071229002	Add a feature code for Bath Alarm call	OS7200	SEI	Applied to V4.30
N080428013	Introduce a new Pick Up feature	OS7200	SEI	Applied to V4.30
N080428014	Change default value for Access Point	OS7200	SEI	Already supported from V4.2x
N071024008	MMC 723: Keypad Type: DS-5021D SET	OS7100	STA	Already supported from V4.2x
N080211005	ITP Idle Mode Login	OS7200	STA	See the ITP idle mode
N080211015	Mobile Extension AnyCall Feature(Phase I)	OS7200	STA	See the MOBEX feature
N080602001	Remote Trunking for 911 and Local Routing via SIP FXO	OS7200	STA	See the SIP FXO interworking
N080602002	Transfer Auto Answer CO Call in Single & Network Node	OS7200	STA	Applied to V4.30
N080602003	MMC 744 Ring Plan Enhancement	OS7200	STA	See the SVMi options
N080602004	Tenant Service Clean Up	OS7200	STA	See the SVMi options
N080602005	MMC 512 Holiday Assignment Enhancement	OS7200	STA	See the SVMi options
N080602007	Clean Up MMC 740 and 741 Functionality and Consistency	OS7200	STA	See the SVMi options
N080602008	Name Field Download SVMi	OS7200	STA	See the SVMi options
N080602009	Mobile Extension AnyCall Feature_2nd Phase	OS7200	STA	See the MOBEX feature
N080808001	Softphone License policy	OS7200	STA	Already supported from V4.2x
N080816001	TRAFFIC REPORT PRINTOUT	OS7400	STA	Applied to V4.30
N071102002	Need to be able to use different MGI ports for speech path.	OS7100	STUK	Applied to V4.30
N080610004	Support UK CID via analog trunk	OS7100	STUK	Applied to V4.30
N081115001	CID on Atrk (allow unrecognised characters)	OS7100	STUK	Applied to V4.30
N061127015	PCR 144 - Mobile Extension (New)	OS7200	STUK	See the MOBEX feature
N071121008	A required improved feature - Page without lifting the handset	OS7200	STUK	Applied to V4.30
N080314004	Mobex software	OS7200	STUK	See the MOBEX feature
N080314005	Greek display	OS7200	STUK	Applied to V4.30
N080427001	Default 24hr clock for Sweden market on LCD (on all OS7000 systems)	OS7200	STUK	Applied to V4.30
N080427002	Following our recent EU tech forum in London, Sweden require this	OS7200	STUK	Applied to V4.30
N080427003	Enhancements to Softphone to match ITP51xx phone	OS7200	STUK	Applied to V4.30

N080427004	LCD display enhancement on incoming group calls	OS7200	STUK	Applied to V4.30
N080429006	Improved group overflow feature for external destination.	OS7200	STUK	Make use of the MOBEX feature
N080429008	Ability to allow DDI info to stay on LCD during call	OS7200	STUK	Applied to V4.30
N080429009	The need for a STRING key to be allowed	OS7200	STUK	Applied to V4.30
N080430004	Ability to search Speed Dial Alpha name better like Mobile phones	OS7200	STUK	Already supported from V4.2x
N080430006	New integral Mobex feature set	OS7200	STUK	See the MOBEX feature

**Table B. QNET P-issue**

QNET ID	Title	System	Version	Country	Remark
-	UNI(DLM) slot will not work properly, when slot is changed in order UNI(DLM), DLI, UNI(DLM)	MP10a	V4.23c	LSP	V4.30a
-	IT displays wrong DIP switch (MP) information, when you select 2.2.0 cabinet information.	MP10a	V4.23c	LSP	V4.30a
-	When we changed TEPRI2 into 16DLI card, DGP is not working. =>It's fixed. DLI Interrupt is registered at intervals of 30 second.	MP10a	V4.23c	LSP	V4.30a
-	System doesn't upgrade CS/MS/RD package designated in startup.ini. (Startup.ini is created when user select each package in Web/Installer/MMC818) * Reason: Boot compares CS/MS/RD package filename using ASCII code and MP10a/11 package uses lower case. But boot has bug that compare with upper case.	MP10a	V4.23c	PA	V4.30a
-	Fax over ISDN is not working. => Reference clock setting routine is deleted in SP S/W.	MP10a	V4.23c	SEI	V4.30a
-	The Timer smaller than 1 sec didn't work and it's fixed. It affects the response time of VM when waiting for the two digits input.	MP10a	V4.23c	STA	V4.30a
P080319006 P080904005	Date and Time change	OS7100	v4.14	LSP	
P080505001	Programming internal modem number in DID table for OS7100 with IT Tool	OS7100	v4.14k	LSP	
P080512001	Incoming calls without CLI from SIP trunks to WIP phones do not connect properly	OS7400	V4.14K	LSP	
P080520005	TEPRIa (old) and TEPRI not operational with V4.14K	OS7100	V4.14k	LSP	
P080904006	DLI port changes to SLI port	OS7100	V4.14	LSP	
P081024004	MGI Public IP Address does not update after upgrade	OS7100	V4.12C	LSP	
P081204004	IT Tool Fwrd destination	OS7400	V4.21c	LSP	
P080606004	IT-Tool 3.2.3 DID Ringing - Error Message	OS7200	V4.21	SEG	
P080606005	IT-Tool 5.2.13 SIP Carrier Options without registration	OS7200	V4.21	SEG	
P080606006	WIM Card Web Interface for WAN over PPPoE	OS7200	V4.21	SEG	
P081003002	Retry does not work properly when LCR Allow is set ON	OS7100	V4.21c	SEI	
P081003003	A SIP paired to DGP continuously rings after the DGP answers the call	OS7100	V4.21c	SEI	
P081003004	There is an improper MGI channels retention after an internal call transfer	OS7100	V4.21c	SEI	
P081003005	After the import of VM/AA DB and restart of the System VM personal prompts are lost	OS7100	V4.21c	SEI	
P081023002	DND ON Soft Menu doesn't work properly when disabled by mean feature code "400"	OS7200	4.21c	SEI	
P081023003	If the message is deleted without reading it, it remains in WIP memory.	OS7200	4.21c	SEI	

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P081023004	When a W5100 is registered in Non SIP mode, than Clear Registration as Normal doesn't clear data	OS7200	4.21c	SEI	
P081112008	OS 7030 does not send "Samsung OfficeServ" User-Agent field in the 200 OK Message	OS7100	2.04	SEI	
P081112009	MoH port is not available anymore after you assign it in MMC756 with SVM-400E as DLI connection	OS7200	4.21c	SEI	
P081113015	SIP stations receive but cannot make any call	OS7400	T4.21g	SEI	
P080609005	CLI Table in MMC837	OS7200	4.21	SER	
P081027005	Dial Pulse does not work on 16TRK in OS 7100 cabinet used as OS 7200 cabinet	OS7200	4.14	SER	
P081029006	H.323 call does not work when Gatekeeper RAS type is set to Manual in MMC836	OS7400	4.21g	SER	
P081113016	MCP Watchdog Reset	OS7200	V4.21c	SER	
P080501002	SVMi Using Network MailBox. 2nd Time Forward Fails	SVMI	ALL	STA	
P080501005	Double Beep issue when using Auto Record	SVMI	ALL	STA	
P080514001	MP40 MESSAGING LOCKUP ISSUE	OS7400	4.14	STA	
P080701010	VM Notify/alert notification feature will always use any outbound port to dial out.	OS7100	T4.22	STA	
P080716012	IPSec via PPPoE connection	OS7200	v1.33t	STA	
P080829001	CID Failure when WIP phone is paired.	OS7100	4.14K	STA	
P080923002	CALL Waiting CLID Issue	OS7400	4.22	STA	
P080925001	7100 VMS Call progress detection does not detect answer on some cell phones.	OS7100	4.22	STA	
P081021001	CALL ROUTING IPUMS ISSUE	OS7400	4.22	STA	
P081025003	UCD REPORT LONGEST QUEUE TIME	OS7400	v4.22	STA	
P081029007	OS7200 3RD EXPANSION	OS7400	v4.22	STA	
P081115002	Problem with 7100 and Fax call routing of subsequent voice call..	OS7100	4.22	STA	
P080319004	PRS Problem in Sweden similar to Spain - Analog trunks not clearing down	OS7100	v4.14	STUK	
P080731020	AoC in Greek Market on Ericsson PTT	OS7100	test pkg	STUK	
P080903019	OS Bug 359 - UCD Report Longest Q fault	OS7400	V4.2X	STUK	
P080923005	Incoming SIP calls fail on OS7200 SIP trunks with Telia	OS7200	T4.22	STUK	
P081006005	Unable to assign SIP trunks using IT Tool or Webmmc	OS7100	V2.03	STUK	
P081007001	Password option for a menu shows some error messages after saving.	OS7100	V2.03	STUK	
P081007004	Only first 5 digits of cli showing on display phone.	OS7100	V1.00	STUK	
P081015001	Fax to desk email gateway sending blank pages intermittently.	OS7400	4.14	STUK	
P081025001	LCR carrier code showing on display phone when call is answered.	OS7100	V2.04	STUK	
P081027004	Key extenders are not saved when using the IT Tool	OS7100	V4.21c	STUK	
P081027006	IT Tool Incorrectly Setting MMC821 Q-SIG TRUNK	OS7200	v4.21c	STUK	
P081105006	Display speed dial name is not working on analogue trunks when using LCR	OS7100	V2.04	STUK	
P081126009	After upgrading to V4.30 and restoring DB AP IP address is removed and cannot be reentered.	OS7200	V4.30	STUK	
P080521003	In case of SLOVENIA, the default language must be ENGLISH.	OS7100		TURKEY	

**Previous Version Changes**

**08.08.20 V4.21c**

- The database in v4.21c is not compatible with the database in v4.1X or older versions. Database address was changed because of adding new features. So if the system is upgraded to new version of V4.21c, then the system will be initialized with default database.

But you can upload old version's database by Installation tool in V4.21c

- Procedure uploading old database.
  1. Connect Installation tool to system of old version.
  2. Download the old version's database.
  3. Please make a backup file of old version's database. Because when uploading at new version software, the old version database would be converted to new version database format. So we recommend that you make a backup file of the original database.
  4. Upgrade in new software(V4.21c)
  5. Connect Installation tool to system of new version.
  6. Upload the old version's database.

- V4.21c includes V4.22 items which are STA requirements (V4.22).

**(1) MCP S/W changed items**

**Main items**

- This package includes SIP Trunking function. And Excel IOT test was also finished with this package. Refer to the attached document. (Excel - SAMSUNG Final ver 2 IP PBX Interop testing.doc)
- This package includes all changes from result of ACD field test and IP-UMS Field test.

**Added Features & Modified items**

1	[SEG][N061017034] CALL PICKUP GROUP WITH PINGRING
2	[SEG][N061105003] SPEED DIAL WITH MORE LETTERS
3	[STA][N071024008/N071025013] MMC723 KEYSSET TYPE : DS-5021D in STA
4	[LSP][N071206002] SUPPORT FOR EXTERNAL VOICEMAIL
5	[SEG][N061017035] MISSED CALLS WITH DATE/TIME AND SET SIGNALLING

6	[SEG][N061017037] ANSWER GROUP
7	[SEG][N061018004] ALPHANUMERIC SUPPORT FOR MMC837 PRIMARY ID
8	[SEG][N061018005] ALPHANUMERIC SUPPORT FOR MMC839 USER ID
9	[SEG][N061102005] REMOVE TRUNK INFORMATION FOR EXTENDED LCD DISPLAY
10	[SEG][N070627006] CHAINED CALL FORWARDING
11	[SEI][N071229001] SPD DIAL/LOG/CLI BLOCK ARE NOT ENOUGH TO SATISFY THE AMOUNT OF OS7400 USERS
12	[LSP][N070418007] CHANGE DEFAULT TO MCID Setting
13	Group Listening feature has to be applied to the all DGPs.
14	[STA] Prefixed profiles for 3 SIP providers
15	[STA] MMC837 Field Name Change
16	[STA] E.164 '+1' incoming call problem
17	[STA] Register enable/disable' option
18	[STA] Support Alternate Proxy when DNS failed
19	[STA] OS7100 reboot when SIP restart
20	[STA] Add REGISTRAR address field in MMC 837
21	[STA] Send 302 Response
22	[STA] SIP Trunk MWI (Message Waiting Indication)
23	[ALL] Alphanumeric Routing
24	[STA] Primary ID Authorization (INVITE)
25	[STA] Support cname in DNS Query
26	[SEUK]MMC821 Q-SIG TRK types are added and the previous names are also changed : QSIG BASIC 1 and QSIG BASIC 2 are added at MMC 821 for QSIG with Notel/Simens PBX throught UK MoD. 1. NORMAL TRNK : this type is same as NORMAL 2. QSIG TRUNK : this type is same as Q-Signalling option 3. QSIG BASIC 1 : this type don't send facility message 4. QSIG BASIC 2 : this type only send caller name including QSIG BASIC 1.
27	[SEUK]Remote STN DGT translation is added for Q-SIG call with other kind of PBX system : REMOTE STN NUMB is added at MMC 724 and REMOTE STN is added at

	<p>MMC 868. REMOTE STN NUMB at MMC 724 is translated to REMOTE STN TRNS DGT at MMC 868.</p> <p>Homeworking with WIP function is added. And some parameters related voice quality are changed.</p> <p>: Homeworking with WIP function is only available in case that AP type is DUAL AP and IP mode of WIP is STATIC IP MODE.</p> <p>1) WLAN : DELETE SIP is added at MMC 845</p> <p>    ENABLE : Homeworking with WIP function is available.</p> <p>    DISABLE : Homeworking with WIP function is not available.</p> <p>2) Some parameters related voice quality are changed.</p> <p>- MMC114 STN VOLUME is applied to WIP</p> <p>    HANDSET VOL : 01 ~ 14</p> <p>    SPEAKER VOL : 01 ~ 05</p> <p>    HEADSET VOL : 01 ~ 14</p> <p>    MMC807 DGP VOL.CTRL is applied to WIP</p> <p>    SIDETONE VOL : 1 ~ 7</p> <p>    HANDSET TX : 1 ~ 8</p> <p>    MIC TX LEVEL : 1 ~ 8</p> <p>    HEADSET TX : 1 ~ 8</p> <p>    WIP DSP PARA option is added at MMC841</p> <p>    M-FRAME : 20 ~ 60 MSEC</p> <p>    ECHO CANCEL : DISABLE/ENABLE</p> <p>- MMC846 WIP INFO options are added</p> <p>    HO THRS : 00 ~ 99</p> <p>    HO DELTA : 00 ~ 99</p> <p>    HO SCAN : 00 ~ 99</p>
28	Regarding NMS function, it provide partial functions for previous OSNMSv1.2x Packages.
29	With an existing OSNMSv1.2x package, it supports only Query Functions. It could not set any functions, and it could not add NEs. If OSNMS package's neither installed nor used,there is no need to check it out.
30	The default value of DELETE SIP was changed in MMC845 (ENABLE → DISABLE)
31	If feature keys are entered and include free stack, all free sip stacks will be assigned to SIP trunk.
32	Only one of the SIP Servers can be enabled.
33	If Country is Australia, the gain value of SLI card must be different to other country.
34	[SIP] ISP interconnection with SIPconnect recommendation
35	[SIP] Diversion header in INVITE message
36	Codec update feature between SCMv2 and OS7000 series was added.
37	In case of OS7200, the default BIVMS group number has to be changed (529,5029 → 539,5039)
38	Allow to change wip codec to 711

**Fixed Bugs**

1	[P071121002/P071128002]7100 voicemail does not stop recording on silence
2	[P071218001]Voicemail will not send digits to pager after answering.
3	[P071121003]OS7100 VM WEB MENU SEARCH ERROR : Web page bug. This bug occurs because there is no checking routine of string to search.
4	[P080221002] Wrong Frame Count problem of H.245 Terminal Capability in H.323 call
5	[P080307001] Softphone Licence Issue : The Softphone license key will be checked by using user id. So every softphone has to have unique user id respectively.
6	[P080313004]System does not notify to SIP station when red message
7	[P080325006]SIP Extensions - BLF on DLI phones : When SIP Station makes outgoing call, the LED handling of DGP DS Key is omitted. So If SIP Station reaches inviting state, system turn on light.
8	[P080401017] T.38 is not working at H.323 call in OS7200
9	[P080428016] UCD SYSTEM LOCKUP
10	[UK]7400 Set Relocation Fail : If 7400 has 2 expansion racks, the extension of MP40 was set relocated with the extension of LP40 expansion. In this case DSS led color do not changed correctly.
11	[P080503001] SMT-W5100 cannot talk to SIP phone because wrong G.711 codec management
12	[P080505001] The internal modem number can not be programmed into the DID table
13	[GTL] Samsung 3rd party SIP License - system reset
14	[P080319006] OS 7100 Date and Day display issue
15	[P080512001]Incoming calls without CLI from SIP trunks to WIP phones do not connect properly
16	[P080517001] OS system does not consider 183 Session Progress sent by SIP server as 180 Ringing
17	[P080430009] WIP-5000M cannot talk to SIP phone because of Invalid Codec
18	[P080402002]Final destination of UCD group to NONE does not disconnect the call after retry counter exceeded
19	[P080402001]Call drops when an incoming through ISDN is forwarded to A-TRK line
20	[P080317009]On ITP, the dialed number disappears when called party answer the call
21	[P080201001] Network Paging Disconnect

22	[P080521003] If country is SLOVENIA, the default language must be ENGLISH.
23	[P080611009/P080603003]ITP5107 LEDs don't work at all
24	[P080611008] Carrier code inserted in MMC 713 appears on LCD when a call is made
25	[P080603011] SPNET station name lost on answer
26	[P080603004] Set relocate FWD All not working
27	[P080609005] CLI Table in MMC837
28	[P080609006] The UCD call wait queue gets stuck with calls waiting in queue when all calls have cleared down.
29	[P080612014] Setting public IP in 2.1.2 doesn't automatically set 2.2.2 public IP field
30	[P080521005] There is no way to program a default key template for an 7 button IP phon
31	[PA] After Set Relocation, DLI card could not be initial.
32	[IIFRA00033262] TSP dead issue - V4.14k & V4.21
33	[IIFRA00033262] SPNET DOWN PRI INTERFACE ROUTING
34	[P080424005]CLI ring destination function doesn't work.
35	[IIFRA00033269]When country is not defined, DS24D/S phone is not work
36	[IIFRA00033273]When you use OS7400 system with OS7200 cabinet. 16MWSLI, 16SLI2, 8TRK located at OS7200 cabinet doesn't work.
37	[P080529008]CR key with Group extender
38	[P080418010] [OS7100] UCD call queued to delay announcement followed by ringback instead of MOH.
39	[P080501005][SVMi] Double Beep issue when using Auto Record
40	[P080612001]MMC 607 MOH function doesn't work in incoming call
41	MCL code dial error for SIP provider
42	[P080612017]System assign AUTH and ACCT code to wrong station randomly

43	[P080606005]IT-Tool 5.2.13 SIP Carrier Options without registration - it is important for SIP Trunking not to register, means without User Name, but no SIP message are send.
44	[P080611010]SIP trunking service is temporarily unavailable when ISP doesn't send "180" or "183"
45	MMC 722 "Station Key" were lost
46	[P070912002]OS7400 Networked with QSIG or IP networking will not pass the incoming name feature.
47	[BRM]BRI Incoming call fail : When 3.1K AUDIO in Bearer capability, Progress_indicator, and TELEPHONY in High layer compatibility are in ACU_CONN_IN message. Incoming call is failed.
48	[P080630005] RETRY does not work when LCR feature is enabled
49	In MMC837, "Incoming Mode" field has to be moved to the Trk Category from Ext Category.
50	[MP] In case the expansion cabinet of OS7400 is OS7200, if set relocated between OS7400 and OS7200, the LED of dgp in OS7200 does not operate correctly.
51	[MP] IPP and SIP Phone have station pair relation. If ipp receive the transferred call from dgp and the caller of this call is wip phone, SIP phone will ring also. But if sip phone answer this call, one way occurs.
52	[MP] large IP Phone does not support PINGRING service.
53	[MP] The 3rd cabinet does not work correctly in OS7400.
54	[MP]How to display version of OS7100 has to be same as the other system's (YY.MM.DD V4.XX.a → 'YY.MM.DD V4.XXa)
55	[MP]A HOLD RE-INV option is added for MOH operation in MMC837
56	[VM] VM stops when fax call is incoming while using UCD.
57	[P080503002] 7100 Web Management tool List Block cannot clear List member
58	[P080507001] Cannot clear Extension Block Station Type
59	[P080507002] OfficeServ Operator cannot do VT to "phantom" mailbox
60	[P080506009] Voicemail will not send digits to pager after answering.
61	[P080307001] Softphone Licence Issue
62	[P080611011] Some labels are missing from "VM/AA" menu of Web Management when language is Italian
63	[P080707003] An improper Holiday is set on the OS 7100 VM/AA Schedule Table

64	[P080709015] OS 7100 VM/AA is out of service after HALT/PROCESS command by IT
65	[P080710001] Parse Error occurs when you try OS 7100 Voice Studio upload feature
66	[P080804001] Cannot pick up ringing UCD call with DP key
67	Dial Pulse doesn't work.
68	[P080704010] When used os7400 system with os7200 cabinet, LCP cabinet doesn't work.
69	[P080731020] AoC in Greek Market on Ericsson PTT
70	[P080725001] Voicemail prompt "The message storage unit is full".
71	[P080621002]IS-TOOL User Accounts: Level 2 or 3 user can access levels restricted.
72	[P080805001]Cannot set DNS server under system parameters.
73	[P080730005] Easysset Send Msg not working when Info Display is turned on.
74	[P080730006] Cannot set different User Access Parameters for IT Tool
75	[P080730007] IT Tool, delete entries in Toll Deny all entries after are also deleted.

**QNet Problem List**



D:\WPAW7000W  
20080811\_V422 파일

**S/W Compatibility Table :**

**(1) OfficeServ 7000 MCP S/W**

MODEL	MCP	LP40	IT	LCP	ETC
OS7400	V4.21c	V1.17	V1.21c	-	
OS7200	V4.21c	-	V1.21c	V4.12	
OS7100	V4.21c	-	V1.21c	-	
DATE	08.08.20	07.11.12	08.08.20	07.11.09	ALL

**(2) OfficeServ 7100 Modules**

MODULE	Version	Date
MP	V4.21c	08.08.20

<b>SP</b>	<b>V2.12</b>	<b>08.07.15</b>
<b>VM</b>	<b>V2.75</b>	<b>08.08.30</b>
<b>MGI</b>	<b>V2.00</b>	<b>08.07.28</b>
<b>BRI</b>	<b>V4.21a</b>	<b>08.08.07</b>
<b>WEB</b>	<b>V4.07h</b>	<b>08.07.28</b>

## Previous Version Changes

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08.02.25 V4.14k

### (1) MCP S/W changed items

#### Added Features & Modified items

- Hunt Mode of STN Group(Sequential mode) is changed
- OS7100 cabinet can be used for OS7200's 2<sup>nd</sup> cabinet.
- Refer to the attached QNet Problem List

#### Fixed Bugs

- [P080122002] ITP does not forward the call when it is disconnected
- [P080122003] ITP as secondary station does not ring when paired to a SIP station.
- [N070720003] SIP EXTENSIONS
- [P070925003] Daylight Savings
- [P071114014] Installation Tool Fails with Modem
- [P071122011] Missing info in SIP Invite Message
- [P071126002] OS7100 V4.14 SIP DNS disconnect issue
- [P071126007] OS Bug 353 - BRI in 7400 when placed in a 7200 as an expansion
- Refer to the attached QNet Problem List

#### QNet Problem List



OS7000\_changes\_  
V414k\_20080225.xls

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## S/W Compatibility Table :

(1) OfficeServ 7000 MCP S/W

MODEL	MCP	LP40	WebMMC	IT	
OS7400 MP40	V4.14k	-	-	V1.1.4	
OS7200 MCP	V4.14k	-	-	V1.1.4	
OS7100 MP10	V4.14k	-	-	V1.1.4	
DATE	08.02.25	-	-	08.02.25	

07.11.15 V4.14

**(1) MP40 S/W changed items**

**Added Features & Modified items**

- ATRK -> UMS TSW gain value is changed
- Refer to the attached 1Q Problem List (check 1Q number started with 'N')

**Fixed Bugs**

- [P071010005] V4.11 Software Set relocate fails after reboot is fixed
- [P070519003] OS7400 System deletes the MoH msg assigned to the VM port in MMC756 after a OS system restart
- soft phone's max count check error at license key
- VG programmable key is not available using IT
- large phone's speed block is delete after DB downloading and DB uploading using IT
- When you download/upload all of DB using WebMMC, specific DB is wrong value. (MMC400, 502, 503)
- Refer to the attached 1Q Problem List (check 1Q number started with 'P' or 'D')

**1Q Problem List**



C:\Documents and Settings\WAdr

**S/W Compatibility Table :**

**(1) OfficeServ 7400 MP40 S/W**

<b>MODEL</b>	<b>MP40</b>	<b>LP40</b>	<b>WebMMC</b>	<b>Installer</b>	
<b>VERSION</b>	<b>V4.14</b>	-	<b>V1.15.0.0</b>	<b>V1.1.1</b>	
<b>DATE</b>	<b>07.11.15</b>	-	<b>07.09.20</b>	<b>07.10.26</b>	

## Previous Version Changes

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07.10.17 V4.12

### (1) MP40 S/W changed items

#### Added Features & Modified items

- **Hotel/Motel function added**

- **Set Authorization code**

The Phone in the room remain blocked outbound dialing, but only the Authorization code that is given to the guest be given a class of service to enable him to dial outbound. The authorization code can be set using BD-PSM

- **Set phone credit**

The purpose of this feature is to allow third party PMS systems to set the posted phone credit to a particular amount. The reason for this is to adjust to mis-calculations between the phone system costing and an external call accounting system costing

- **Set DND**

In a hospitality installation, the front desk attendant can set to place the telephone of individual room guest into DND, and clear DND at the request of the room guest. Setting DND will prevent the room guest from receiving calls (internal and xternal) with the exception of DNDO (DND override) calls from the operator or attendant, while allowing the guest to make outbound calls. Upon request of the guest, the attendant must also be able to clear or cancel the DND setting from the PMS console

- **Phone Restriction at Check-in**

When no deposit is posted at check-in the guest phone is unrestricted. Currently, administrators are having to post a minimum deposit (eg: \$1.00) to the guest room. When the amount of the phone deposit is exhausted by the guest, then the phone restrictions are enabled. An option will need to be added to MMC 210 to allow the telephone technician to program the system to have the following options

- \* CHKIN RESTRC = OFF (No dialing restrictions at check-in)  
CHKIN RESTRC = ON (Restrict outbound dialing at check-in)

- **ACD and IP-IVR with SVMi-20E**  
IP-IVR has to be used with SVMi-20E at the same time.
- **Changed MMC default values in German.**
- **Refer to the attached 1Q Problem List (check 1Q number started with 'N')**

### Fixed Bugs

- Refer to the attached 1Q Problem List (check 1Q number started with 'P' or 'D')

### 1Q Problem List



C:\WreleaseW  
OS\_V412\_Release'

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### S/W Compatibility Table :

#### (1) OfficeServ 7400 MP40 S/W

MODEL	MP40	LP40	WebMMC	Installer	
VERSION	V4.12	-	V1.15.0.0	V1.1.0	
DATE	07.10.17	-	07.09.20	07.10.19	

### Previous Version Changes

**07.08.01 V4.11**

#### (1) MP40 S/W changed items

- Wireless IP phone license was deleted by request.
- According to the VoIP License Policy, VoIP Stack License was added in OS7200 MCP V4.10.  
But wireless IP phone license was deleted in this version by request.
- WIP PHONE can not assign in MMC841 SIP STACK ALLOW option.

- **Maximum 32 wireless IP phones are available without license.**

### **Added Features & Modified items**

- **Added some features and Changed some default values for IP phone's voice quality.**
  - MMC835 "DUAL FILTER EC" option was added.
    - This option is only available in MGI16/MGI64 card.
    - Dual filter echo canceller's parameter of MGI16/MGI64 card will be downloaded differently according to the trunk's type.
    - Default – 8TRK2 MODE
    - Range – DISABLE, 8TRK MODE, 8TRK2 MODE, DTRK MODE
  - MMC841 "ITP MAX TX LIMIT" option was added
    - If this option is set to YES, maximum TX level of IP phone can be limited.
    - Default – NO
    - Range – YES / NO
  - SIDETONE VOL's default data for ITP was changed to 4 in MMC807.
  - MIC TX LEVEL's default data for ITP was changed to 3 in MMC807.
  - Some T-SWITCH GAIN default data were changed in MMC809.
    - ITP <-> ATRK : 0
    - ITP <-> DTRK : 0
- **QSIG Time synchronization feature was added.**

When several systems are connected in network with the using of QSIG in TEPRI or VoIP, time synchronization will work.

  - MMC820 "TIME SYNC" option was added.
    - (Default – OFF / Range – OFF/ON)
  - This option does not need in SELF, only can assign from SYS(01) to SYS(99).
  - If the user want to use SYS(01) system to time master, TIME SYNC option of SYS(01) have to change ON. And then when other system's station connected in network make a call to SYS(01), time will be synchronized by SYS(01)'s time.
- **C-M&A Feature was added in RUSSIA**

## Fixed Bugs

- Refer to the attached 1Q Problem List (check 1Q number started with 'P' or 'D')

## 1Q Problem List



1Q\_Problem\_List\_  
OS7400\_V411.xls

## S/W Compatibility Table :

### (1) OfficeServ 7400 MP40 S/W

MODEL	MP40	LP40	WebMMC	Installer	
VERSION	V4.10	-	V1.14.0.0	V1.0.2	
DATE	07.08.01	-	07.08.06	07.08.01	

### 07.07.01 V4.10

#### (1) MP40 S/W changed items.

- System database is not compatible with old version's database in V4.10. Database address was changed because of adding new features and merging source file of OS7000 series. So if the system is upgraded to new version of V4.10, then the system will be initialized with default database. But you can upload old version's database by WebMMC in V4.10
- Procedure uploading old database.
  7. Connect WebMMC to system of old version.
  8. Download the old version's database.
  9. Please make a backup file of old version's database. Because when uploading at new version software, the old version database would be conversed to new version database format. So we recommend that you make a backup file of the original database.
  10. Upgrade in new software(V4.10)
  11. Connect WebMMC to system of new version.

12. Upload the old version's database.

- If you downgrade MP40 from V4.10 to V3.34, set backup switch OFF for deleting Nand Flash and start system because V3.34 has a problem of System IP Address set

**Added Features & Modified items**

- **Virtual Cabinet configurations were changed.**
  - Number of Virtual cabinet : 2 (Cabinet 4 ~ Cabinet 5)
  - Number of Slot per 1 Cabinet : 12 (12slots / 1cabinet)
  - Number of Port per 1 Slot : 32 (32ports / 1slot)
  - The user can specify the type of virtual cabinet card in MMC857. The type of virtual cabinet card is as follows.

Type	Range	Max. Port	Type	Range	Max. Port
SLT	C4S01~ C4S08 (8)	256	BRI STN	C5S02~ C5S06 (5)	160
DGP	C4S01~ C4S08 (8)	256	GCONF STN	C5S03~ C5S06 (4)	128
WIRED ITP	C4S03~C5S05 (15)	480	SPNET TRK	C5S06~ C5S12 (7)	224
WLAN ITP	C4S10~ C5S01 (4)	128	SIP TRK	C5S09~ C5S12 (4)	128
SIP STN	C4S10~ C5S01 (4)	128	H323 TRK	C5S11~ C5S12 (2)	64
IP UMS	C4S10~ C5S01 (4)	128			

- Virtual cabinet card type's default data is as below.

Cabinet	Slot	Card Type	Cabinet	Slot	Card Type
4	1	SLT	5	1	WLAN ITP
	2	SLT		2	BRI STN
	3	DGP		3	BRI STN
	4	DGP		4	BRI STN
	5	WIRED ITP		5	GCONF STN
	6	WIRED ITP		6	GCONF STN
	7	WIRED ITP		7	SPNET TRK
	8	WIRED ITP		8	SPNET TRK
	9	WIRED ITP		9	SIP TRK
	10	WIRED ITP		10	SIP TRK

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	11	WLAN ITP		11	H323 TRK
	12	WLAN ITP		12	H323 TRK

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If virtual cabinet card type is changed, PCMMC can not use anymore with changed virtual card type. That is, some MMCs about changed virtual card type do not work correctly.

So, the user have to use INSTALLATION TOOL when MMC857's virtual cabinet card type is changed.

- **VoIP Stack License was added.**
  - License key have to be assigned in MMC841 FEAT LICENSE KEY for using VoIP features
  - H.323 stack and SIP stack license was added.
  - H.323 stack license is used for H.323 trunk. So, available number of H.323 trunk depend on the number of H.323 stack license.
  - SIP stack license is used for SIP Trunk, SIP Station, IP-UMS/IVR, WIP Phone. Features using SIP stack can be specified available number of SIP stack each other in MMC841 SIP STACK ALLOW. Detail items of MMC841 SIP STACK ALLOW are as follow.
    - ◆ MAX COUNT : The number of SIP Stack License (Read only)
    - ◆ NON SEC SIP : The number of Non Samsung SIP phone (Read only)
    - ◆ FREE COUNT : The number of free SIP Stack License (Read only)
    - ◆ SIP TRUNK : The number of available SIP Trunk (Can be assigned in Free Count)
    - ◆ SIP PHONE : The number of available Samsung SIP Phone (Can be assigned in Free Count)
    - ◆ WIP PHONE : The number of available WIP Phone (Can be assigned in Free Count)
    - ◆ IP-UMS/IVR : The number of available IP-UMS/IVR (Can be assigned in Free Count)
  
- **New Card (8TRK2, 16TRK, 8COMBO2, 16SLI2)**
  
- **Installation Tool was added.**

- Refer to attached Installation Tool’s release notes



- **Simple IP phone was added.**
  - Operation is same with Large IP phone.
  
- **Standard SIP Phone interface was added.**
  - Registration Procedure
    - ◆ License key assign in MMC841 FEAT LICENSE KEY for using SIP STN. (Non Samsung SIP Station or Samsung SIP Station)
    - ◆ Assign SIP STN virtual card type in MMC857
    - ◆ The user can see the USER ID / PASSWORD in MMC842. SIP Station can be registered with USER ID/PASSWORD in MMC842. (Default USER ID: equal with Tel Number / PASSWORD: 0000)
  
  - SIP Station Feature List

FEATURE	DESCRIPTION																					
BASIC CALL	<p>The number of MGI by opposite side is as below.</p> <table border="1" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th colspan="2" style="background-color: #FFFF00;">Connection Status</th> <th style="background-color: #FFFF00;">Using MGI Number</th> </tr> </thead> <tbody> <tr> <td>SIP Station</td> <td>Digital/SLT Phone</td> <td style="text-align: center;">1</td> </tr> <tr> <td>SIP Station</td> <td>Trunk (except VoIP trunk)</td> <td style="text-align: center;">1</td> </tr> <tr> <td>SIP Station</td> <td>VoIP Trunk</td> <td style="text-align: center;">2</td> </tr> <tr> <td>SIP Station</td> <td>IP Phone</td> <td style="text-align: center;">X</td> </tr> <tr> <td>SIP Station</td> <td>SIP Station</td> <td style="text-align: center;">X</td> </tr> <tr> <td>SIP Station</td> <td>WIP Phone</td> <td style="text-align: center;">X</td> </tr> </tbody> </table>	Connection Status		Using MGI Number	SIP Station	Digital/SLT Phone	1	SIP Station	Trunk (except VoIP trunk)	1	SIP Station	VoIP Trunk	2	SIP Station	IP Phone	X	SIP Station	SIP Station	X	SIP Station	WIP Phone	X
Connection Status		Using MGI Number																				
SIP Station	Digital/SLT Phone	1																				
SIP Station	Trunk (except VoIP trunk)	1																				
SIP Station	VoIP Trunk	2																				
SIP Station	IP Phone	X																				
SIP Station	SIP Station	X																				
SIP Station	WIP Phone	X																				
CALL HOLD / RESUME	Use SIP station’s HOLD/RESUME KEY																					
CONSULTATION CALL	SIP station can have maximum 2 calls by using HOLD key																					
TRANSFER (BLIND / CONSULTATION)	Use SIP station’s BLIND / CONSULTATION Transfer menu																					
CALL FORWARD	Use “FWD” System Feature Code of MMC724																					
CONFERENCE	Use “CONF” System Feature Code of MMC724																					

CALL PARK	SIP station's call park feature is same with system hold feature. Use "HOLD/HLDPK" System Feature Code of MMC724
CALL PICKUP	Use "GRP/DIRPK/MYGRP" System Feature Code of MMC724
DND	Use "DND" System Feature Code of MMC724
CALLBACK	Use "CBK" System Feature Code of MMC724
VOICE MAIL INDICATION	Voice Mail number would be sent to SIP station when new voice mail message is left.

**- Added New Features for GERMANY**

(Refer to attached 1Q Problem List N061105005, N061105004, N061108015, N061017036)

**Fixed Bugs**

- Refer to the attached 1Q Problem List (check 1Q number started with 'P' or 'D')

**1Q Problem List**



D:\User\Wjskim\release\W2007\_03M

**S/W Compatibility Table :**

**(1) OfficeServ 7400 MP40 S/W**

MODEL	MP40	LP40	WebMMC	Installer	
VERSION	V4.10	V1.15	V1.13.0.0	V1.0.0	
DATE	07.07.01		07.07.01	07.07.01	

**07.02.02 V3.34**

**(1) MP40 S/W changed items**

**Added Features & Modified items**

- Refer to the attached 1Q Problem List (check 1Q number started with 'N')

**Fixed Bugs**

- Refer to the attached 1Q Problem List (check 1Q number started with 'P')

or 'D')

### 1Q Problem List



D:\release\W  
OS7400\WV334\WO€

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06.07.14 V3.32

### (1) MCP S/W changed items

#### Added Features & Modified items

- Refer to the attached 1Q Problem List (check 1Q number started with 'N')

#### Fixed Bugs

- Refer to the attached 1Q Problem List (check 1Q number started with 'P' or 'D')

### 1Q Problem List



D:\User\Wjskim\W  
release\W6월의뢰\W