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SPECIAL APPLICATIONS SECTION

PART 1. OVERVIEW

This part of the technical manual is titled “Special Applications” because it provides information about interfacing with customer-provided equipment or using a feature in a different way than it was intended. Perhaps an application may require a combination of CPE, creative programming, unusual feature operation or all of the above.

Because these applications require installation instructions and a combination of programming sequences, there is no obvious place to put this information; therefore, we created this part of the manual. As additional special applications are reported from the field, we will include them in this section.

PART 2. APPLICATIONS

2.1 THIRD PARTY VOICE MAIL/AUTO ATTENDANT INTEGRATION

Because of the increased popularity of voice mail and auto attendant use, the OfficeServ 7200-S system includes many programmable options to address this demand. Obviously the degree of integration that can be achieved depends on the abilities of the voice mail/auto attendant (VM/AA) system as well as the telephone system.

This list details the capabilities provided by the OfficeServ 7200-S system for voice mail integration.

HARDWARE PROVISIONS

a. The VM/AA system must be connected to single line circuits on any SLI type of card.

b. Each port on a 4 port SLI is equipped with a dedicated DTMF receiver for detecting DTMF signaling from the VM/AA.

c. VMAA ports will also provide an instant break in loop current when the calling party hangs up. This is called a disconnect signal.

SOFTWARE PROVISIONS

a. SCREENED OR UNSCREENED TRANSFER

There are no special codes needed to transfer a call. Simply hookflash, receive transfer dial tone and dial the destination.

b. DIRECT IN LINES

Any C.O. call can be assigned to ring at an individual station or a station hunt group assigned to the VM/AA.

c. CALLS OR RECALLS TO THE OPERATOR

Dialing 0 will always result in a ringback signal. If the operator is busy, the call continues to ring in queue to the operator. This prevents a caller from dialing 0 and reaching another mailbox because the operator is busy.

Note: If the group busy option is ON in MMC 601 for the VMAA group then calls to the operator group will receive a busy signal if all ports are busy.

d. MESSAGE WAITING

A VM/AA port can leave a message at any station or group of stations. The message waiting indication can be set or canceled at any station or station group with or without the stations ringing.

e. IN BAND SIGNALING

The OfficeServ 7200-S system can be programmed to send the calling station's extension number after the voice mail system answers. These DTMF signals may include a leading digit to indicate the type of call and additional information about the original caller. DTMF signals may also be substituted for call progress tones to speed up voice mail call processing. This program allows call forwarding to a mailbox and bypassing of the main greeting for automatic message retrieval. Blind transfers may be performed because the recall will be correctly identified. NOTE: The effectiveness of this program depends on the ability of the voice mail system to make use of this information.

f. STATION HUNT GROUP WITH OVERFLOW

Each station group can have an individual overflow destination with an individual overflow timer. The overflow destination will ring whenever a call to the group is not answered. If the voice mail system becomes inoperative, calls are automatically routed to the overflow destination.

Note: If the group busy option is turned on in MMC 601 then calls will not overflow.

g. INTERNAL CALL FORWARDING TO VOICE MAIL

This option in MMC 300 will allow or deny intercom calls from following call forward to voice mail. This feature conserves disk drive space by only storing calls originating outside the OfficeServ 7200-S system.

h. ONE TOUCH VOICE MAIL ACCESS

One Touch speed dial keys can be programmed to automatically dial, log into and retrieve messages from voice mail.

i. CALL PROGRESS TONES

The only tones sent to a VM/AA port are dial tone, busy and ringback. To eliminate confusion, busy tone is substituted for DND or error tones on voice mail ports only.

2.2 STAND-ALONE ADD-ON MODULE

To make a DCS 32 button add-on module operate as a stand-alone unit, perform the following steps in the order they are listed.

MMC 103 With the technician or customer passcode, assign answer mode as Voice Announce, Auto Answer or ring.

MMC 105 Assign speed dial numbers for the AOM.

MMC 606 Advance to the extension number of the AOM you want to use as stand-alone. Assign blocks of speed dial numbers to the AOM.

NOTES:

1. Transferred calls cannot be camped-on to a busy DCS 32 Button AOM. If a station attempts to do so, the transferred call will ring back to the station immediately.

2. Busy station camp-on will not work when calling a busy DCS 32 Button AOM.

2.3 INDIVIDUAL STATION PAGE

The system was not designed to permit page announcements to individual keysets. However, a forced auto answer key (FAUTO) can be used to accomplish this objective.

1. Program a keyset for RING in MMC 103.

2. Assign an FAUTO key to each keyset that is allowed to page individual keysets.

3. Call another station. When you hear ringback tone, press the FAUTO key. The ringing will stop and an Auto Answer call is set up.

NOTE: To prevent the use of this feature from getting out of control, only assign FAUTO keys to those keysets needing to page individual keysets.

2.4 CALLER ID

The OfficeServ 7200-S is compatible with both types of Caller ID as defined by BELLCORE. These are the single message format or "Number Only" sometimes referred to as standard Caller ID and the multiple message format or "Name and Number" sometimes referred to as Deluxe Caller ID. In the case of Number Only delivery, there is a translation table available that may be used to add names to the delivered number.

HARDWARE PROVISIONS

In order to install Caller ID on an OfficeServ 7200-S system you must have the following equipment available:

- MP20S Processor Card
- An 8TRK2, 8TRK, 16TRK Card, or TRM Module on a UNI Card.

SOFTWARE PROVISIONS

The MMCs related to Caller ID are listed below with a short description of their use. They are listed in the recommended order in which they should be programmed. This sequence is suggested so that the installer gets a better understanding of how the feature works. There is no technical reason to follow this sequence.

- **MMC 414 CALLER ID TRUNKS**
  This MMC is used by the technician to determine which trunks will receive Caller ID data.

- **MMC 312 ALLOW CALLER ID**
  This MMC is used by the technician to determine which keysets are allowed to receive Caller ID displays.

- **MMCs 722 and 723**
  These MMCs have keys related to Caller ID features added to them. It is strongly recommended that all keysets allowed Caller ID in MMC 312 are programmed with a CID key.

- **MMC 728 CID TRANSLATION**
  This MMC allows the technician to create a list of names that correspond to numbers received from the Central Office. These names will be displayed when a call rings in that has NUMBER ONLY data provided by the CO.

- **MMC 725 SMDR OPTIONS**
  The ability to print Caller ID data and abandoned calls is determined by this MMC.

- **MMC 119 CALLER ID DISPLAY**
  This MMC is used by the end user to determine which piece of Caller ID data is displayed when a call rings at the user's station.

- **MMC 501 SYSTEM TIMERS**
  This MMC has two new timers related to Caller ID. The only timer that may need adjustment is the CID DISPLAY TIME. This is the length of time that CID data is displayed after the CID key is pressed.

- **MMC 415 TRK. ABANDON**
  This MMC is used by the technician to determine which trunks will record data in the Call Abandon list and print with an Abandon "A" flag on SMDR.

- **MMC 608 ASSIGN REV BLOCK**
  This MMC is used by the technician to assign CID Review blocks to keysets to allow the user to review CID data for previous calls.

- **MMC 701 ASSIGN COS**
  All of the Caller ID features have been added to this MMC to enable the technician to allow or deny them.

- **MMC 724 NUMBER PLAN**
  The Caller ID features have been added to this MMC to allow a technician to assign an access code where necessary.

In addition to the above MMCs, it is necessary to have LCR programmed on the system to enable certain features with a DIAL/REDIAL option to be used. This is because the number format provided by the Central Office contains the area code. This area code must be stripped off in the LCR modified digits section to allow a local number to be correctly dialed.

For example, if the system is located in the 305 area code, the LCR digit table points the entry 1305 to a modified digits entry that deletes the first four digits of the CID number. Of course, this is a much simplified LCR scheme. As there are long distance calls to be made within the home area code, additional entries are required to identify these.

For example, if 1305-426 is a local call, the area code has to be stripped, but if 1305-858 is long distance, the area code has to remain to allow the number to be dialed. There are two ways of doing this. You can either enter all of the local office codes and tell the system to strip the area code from them or you can enter the long distance codes and tell them to ignore the modify digits entry.

A list of all of the local office codes can be found at the front of the local telephone directory.

2.5 USING LCR TO INSERT LONG DISTANCE PIC CODE

One of the more common uses for LCR is to use this feature to automatically insert the long distance access code for long distance calls within your own area code. This will allow these calls to be processed by the selected long distance carrier instead of the local telephone company. The following example is based on an area where all long distance calls must be preceded by 1 + area code as this is the most common scenario.

In MMC 710, program the following entries:

MMC 710	**LCR DIGIT TABLE**		
INDEX	LCR DIGIT STRING	LENGTH	ROUTE
001	1	11	1
002	2	7	1
003	3	7	1
004	4	7	1
005	5	7	1
006	6	7	1
007	7	7	1
008	8	7	1
009	9	7	1
010	411	3	1
011	911	3	1
012	0	1	1
013	1AAA	11	2

NOTE: AAA is your home area code.

In MMC 711, program the following entries:

MMC 711	**LCR TIME TABLE**							
TIME CHANGE BANDS								
	A		B		C		D	
	HHMM	LCRT	HHMM	LCRT	HHMM	LCRT	HHMM	LCRT
DAY								
SUN	0001	1						
MON	0001	1						
TUE	0001	1						
WED	0001	1						
THU	0001	1						
FRI	0001	1						
SAT	0001	1						

In MMC 712, program the following entries:

MMC 712	**LCR ROUTE TABLE**			
LCR ROUTE	TIME CHANGE	LCRCOS	TRK GROUP	MOD DIGITS
1	1	1	800	
2	1	1	800	001

In MMC 713, program the following:

MMC 713	**LCR MODIFY DIGIT TABLE**		
INDEX	NO. OF DELETE DIGITS (15)	NO. OF INSERT DIGITS (14)	NO. OF APPEND DIGITS (14)
001		1010XXX	

NOTE: 1010XXX is the access code for the long distance carrier of your choice.

- In MMC 603, move all of the C.O. lines from trunk group 9 to trunk group 800. You will have to delete the line numbers from trunk group 9 as trunks can be in more than one group.
- In MMC 724, assign 9 as the LCR access code. This will delete 9 from the first trunk group automatically.
- In MMC 210, turn on LCR.

Setting LCR up like this will not prevent C.O. lines from being accessed by Direct Trunk (DT) keys but it does mean that speed dial numbers will have to be reprogrammed to allow them to access LCR.

2.6 USING LCR WITH CALLER ID

It is necessary to have LCR programmed on systems with Caller ID to enable certain features with a DIAL/REDIAL option to be used. This is because the 10 digit CID number format provided by the Central Office always contains the area code. This area code must be stripped off in the LCR modified digits section to allow a local number to be correctly dialed. To make this task easier, the system inserts a 1 in front of the received digits. This makes it look like a normal 11 digit telephone number so LCR can process the number and modify the digits.

For example, if the system is located in the 305 area code, the LCR digit table points the entry 1305 to a modified digits entry that deletes the first four digits of the CID number. Of course, this is a much simplified LCR scheme. As there are long distance calls to be made within the home area code, additional entries are required to identify these calls. This is illustrated in the sample table for MMC 710 below where entries 013 to XXX are the local area code (305 in the previous example) followed by the local CO prefixes. This will tell the system which calls need to have the first four didits stripped off using the modifiy digits table.

In MMC 710, program the following entries:

MMC 710	**LCR DIGIT TABLE**		
INDEX	LCR DIGIT STRING	LENGTH	ROUTE
001	1	11	1
002	2	7	1
003	3	7	1
004	4	7	1
005	5	7	1
006	6	7	1
007	7	7	1
008	8	7	1
009	9	7	1

MMC 710	LCR DIGIT TABLE		
010	411	3	1
011	911	3	1
012	0	1	1
013	1AAALLL	11	2
↓	↓	11	2
XXX	1AAALLL	11	2

NOTE: AAA is your home area code and LLL is a local prefix. For example, at STA, AAA= 305 and one LLL = 426. To operate correctly, all of the local prefixes must be entered in this table. These prefixes can be found at the front of the local telephone directory.

In MMC 711, program the following entries:

MMC 711	LCR TIME TABLE							
TIME CHANGE BANDS								
	A		B		C		D	
	HHMM	LCRT	HHMM	LCRT	HHMM	LCRT	HHMM	LCRT
DAY								
SUN	0001	1						
MON	0001	1						
TUE	0001	1						
WED	0001	1						
THU	0001	1						
FRI	0001	1						
SAT	0001	1						

In MMC 712, program the following entries:

MMC 712	**LCR ROUTE TABLE**			
LCR ROUTE	TIME CHANGE	LCRCOS	TRK GROUP	MOD DIGITS
1	1	1	800	
2	1	1	800	001

In MMC 713, program the following:

MMC 713	**LCR MODIFY DIGIT TABLE**		
INDEX	NO. OF DELETE DIGITS (15)	NO. OF INSERT DIGITS (14)	NO. OF APPEND DIGITS (14)
001	4		

NOTE: The deleted digits are the 1 + AAA from the LCR digit table in MMC 710.

- In MMC 603, move all of the C.O. lines from trunk group 9 to trunk group 800. You will have to delete the line numbers from trunk group 9 as trunks can be in more than one group.
- In MMC 724, assign 9 as the LCR access code. This will delete 9 from the first trunk group automatically.
- In MMC 210, turn on LCR.

Setting LCR up like this will not prevent C.O. lines being accessed by Direct Trunk (DT) keys but it does mean that speed dial numbers will have to be reprogrammed to allow them to access LCR.

2.7 UNIFORM CALL DISTRIBUTION

UCD is used to distribute calls to a group of agents. If the group members (agents) are all busy, UCD controls queue patterns and information messages. Callers are held in queue for an available agent. First and second announcements reassure the caller until an agent becomes free. UCD announcements are provided by the Samsung Voicemail system.

The following step by step example demonstrated how to create an UCD application.

A SAMPLE UCD APPLICATION (STEP BY STEP)

1. In MMC 601 create a group of type VMSUCD and place the voicemail ports in it as members. In this example we call this group 505. The ring type should be distributed. Do not use the overflow, GRP TRANSFER or NEXT.

2. In MMC 601, create a group (we use 525) defined as an UCD group and containing the station numbers for the UCD agents. Set the group options as follows:

   a. Set the ring type to DISTRIBUTED.
   b. Set the overflow to ten seconds (creates a ten second ringback “grace period” in case all agents are busy before the call overflows to the UCD module).
   c. Set NEXT to 505.
   d. Set the wrapup timer. This is an optional timer to allow UCD agents to finish the work associated with a call before receiving the next call.

3. Decide what C.O. lines are to go to the UCD group and assign them to ring 525 in MMC 406.

4. In MMC 607, program the UCD options. These consist of the following:

   Message 1: After the caller has overflowed from the UCD group, the first message will immediately play. The default value of ‘5061’ plays a default recording of “I’m sorry, all stations are presently busy”. This message number can be assigned from a valid range of 0001 to 9999 (corresponding to voicemail prompts). Please refer to voicemail documentation for customizing prompt recording instructions.

   Message 2: If no agent has become free after the UCD recall time, the caller will be played the Message 2. The default value of ‘5062’ plays a default recording of “I’m sorry, all stations are still busy’. This message number can be assigned from a valid range of 0001 to 9999. Please refer to Part 4 or Part 6 of the manual for customizing prompt recording instructions. This message will repeat at the UCD recall interval.

   Exit Code: This is an option that allows the caller to exit UCD by dialing this digit.

Retry Count: This is the maximum number of times that a caller will be recalled to the UCD message before being transferred to the final destination.

Final Destination: The destination the caller will reach if the exit digit is dialed or the retry count is exceeded.

Ring Next: This timer determines how long an unanswered call will ring a UCD agent before that station is logged out and the next station in the UCD group rings.

UCD Recall: This timer determines the interval between UCD messages played to the caller.

MOH: This determines the MOH source that the caller will be connected to while the UCD recall timer is running.

AUTO LOG OUT: Determines if the ringing station will be logged out if the ring next timer expires.

ALL OUT→FINAL: Determines if calls will go directly to final destination if all agents are logged out.

AGENT PIN NUMBER: Agents have to enter a PIN to log into the group.

GBUSY NEXT: If all agents are busy go directly to next destination.

You have now completed the UCD programming.

2.8 ISDN OVERVIEW

INTRODUCTION

Programming Primary Rate Interface (PRI) circuits into the OfficeServ 7200-S is different from working with analog facilities but does not have to be overly frightening or complicated to do. To reduce the amount of document study and reading necessary to install ISDN in the OfficeServ 7200-S system, this overview section includes flow chart diagrams that show the sequence of operations, with examples, necessary to program PRI service. Hopefully, this should make working with ISDN in the OfficeServ 7200-S easier for the average installation technician.

PRI PROGRAMMING

1. ISDN PRI uses the OfficeServ 7200-S T1 capability as the basic transport media. However, unlike a normal T1, signaling for each call is handled on a Common Channel (i.e., the ISDN "D" channel) basis. The PRI uses the 24th T1 channel as the "D" channel with the other 23 channels serving as the "B" or Bearer channels. Then, each ISDN PRI can have from one (1) up to a maximum of twenty-three (23) "B" channels.

2. In ISDN PRI, there is no relationship between telephone numbers and actual "B" channels. Thus, calls to any pre-subscribed PRI telephone number can arrive on any free "B" channel. From this standpoint, PRI acts very similar to T1 Direct Inward Dialing (DID) service. As with DID service, incoming call routing is programmed relative to the received "Called Party" telephone number rather than the "B" channel that the incoming call arrived on. Like DID trunks, the PRI uses MMC 714 to find the ring destination for the dialed telephone number. Therefore each telephone number must be programmed into MMC 714. However, MMC 714 has been modified to accept the longer digit string provided with a PRI call (i.e., the Called Party telephone number) as opposed to the shorter number of DID digits (i.e., typically three or four digits).

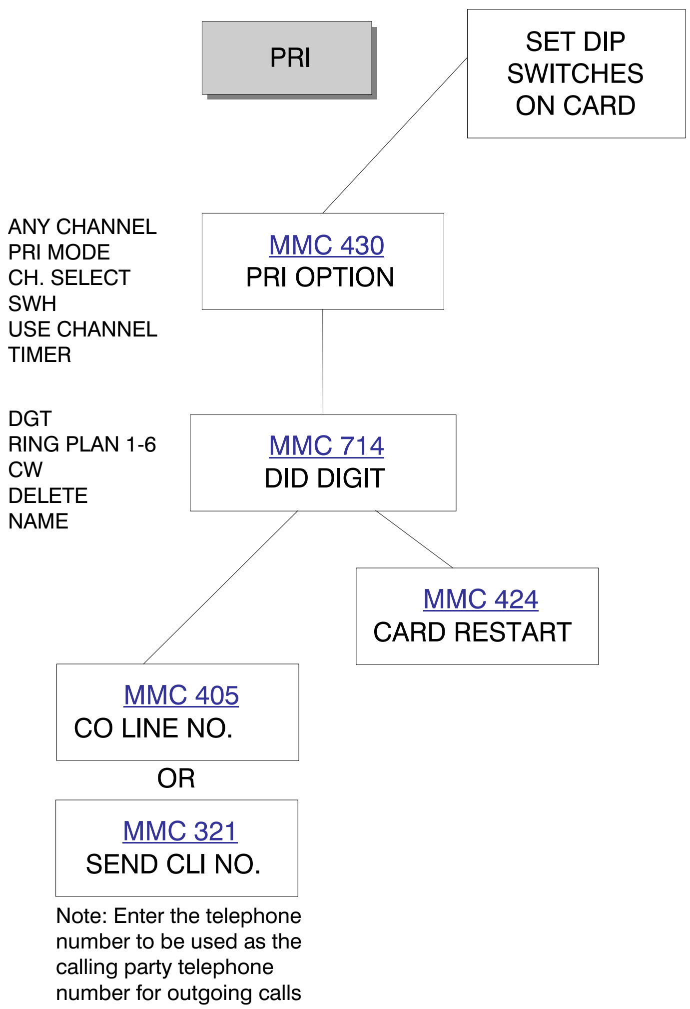
3. When making outgoing PRI calls, it is possible to send a "Calling Party" telephone number other than the main billing number for the PRI circuit. The OfficeServ 7200-S has two methods to determine the number that is sent.

   METHOD 1: Per Channel Programming The first method is by using MMC 405. MMC 405 is used to program the "Calling Party" telephone number that will be sent on each call on a specific "B" channel of the PRI. Normally, all PRI channels would have the same calling party telephone number (e.g., the facility Billing number) assigned. However, any valid telephone number assigned to this PRI can be used as the calling party number. These telephone numbers are purchased from the Telephone Company. Only numbers purchased for a specific PRI can be used as a calling party number for that PRI (i.e., you can't use numbers from other facilities). The Telephone Company screens the calling party telephone number on all outgoing PRI calls and checks for valid numbers. If an invalid number is found, the Telephone Company will generally send the facility Billing number instead on this call (note: this policy may vary on a state by state basis). It is possible to send a blocked Caller ID to mask the call from the caller party. This is done by setting SND to NO in MMC 312 for the station.

METHOD 2: Per Station Programming The second method is to use the CLI tables in MMC 321. These tables allow up to 4 calling party numbers to be assigned to each station on the system. Each table is then assigned to a PRI circuit in MMC 430 and when an outbound call is made over a PRI then the number assigned to that station for that PRI will be sent. Any valid telephone number assigned to this PRI can be used as the calling party number. These telephone numbers are purchased from the Telephone Company. Only numbers purchased for a specific PRI can be used as a calling party number for that PRI (i.e., you can't use numbers from other facilities). The Telephone Company screens the calling party telephone number on all outgoing PRI calls and checks for valid numbers. If an invalid number is found, the Telephone Company will generally send the facility Billing number instead on this call (note: this policy may vary on a state by state basis).

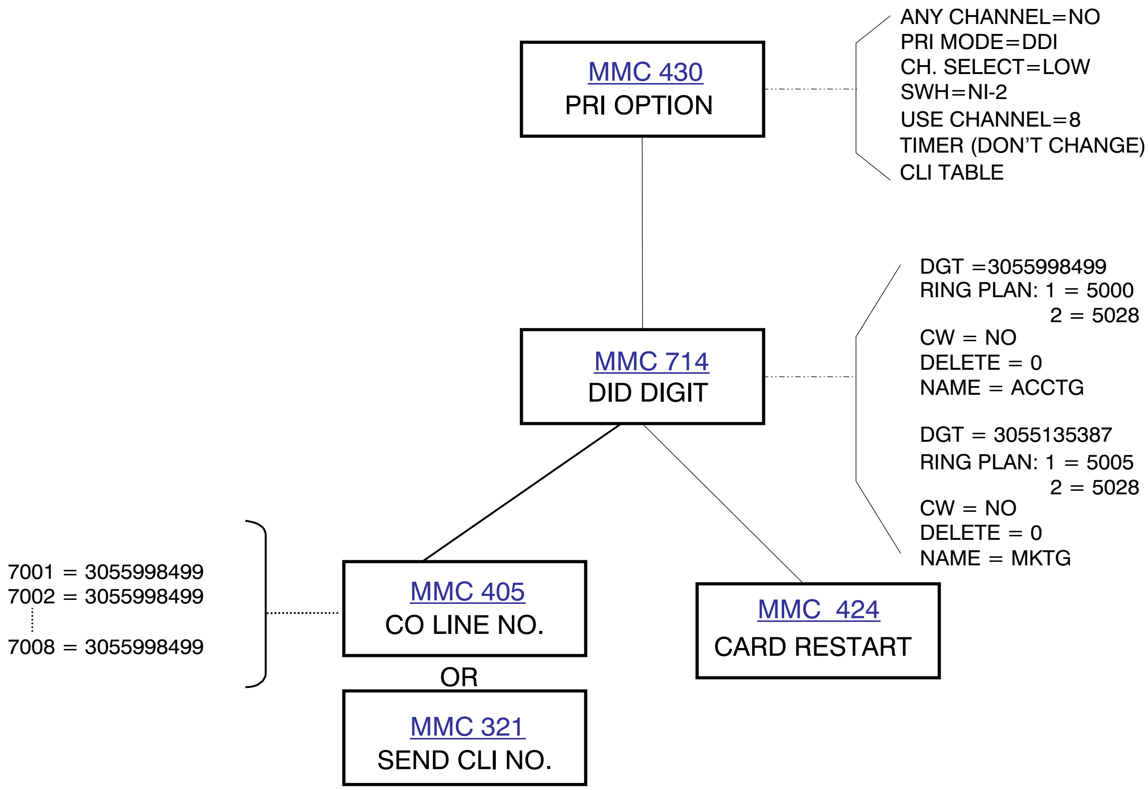
TEPRI SETUP METHOD – FIGURE 1

The following shows a step by step approach to programming a TEPRI board as either a T1 or as a PRI circuit. Please consult the individual MMCs for specific details on the various parameters used.



• EXAMPLE 1: PROGRAMMING A PRI TRUNK

The following shows a typical step by step approach to programming a ISDN PRI.



2.9 ORDERING AN ISDN PRIMARY RATE INTERFACE (PRI) TRUNK FACILITY

Ask the Telephone Company Service Order Representative or Customer Account Manager questions to determine the following:

1. What are the various types of PRIs that the Telephone Company can provide service to the desired site with. If you have a choice, tell the Telephone Company that you prefer a National ISDN 2 (NI-2) PRI. If this is not available, the OfficeServ 7200-S will support either the #5ESS Custom or DMS-100 PRI protocols. However, the NI-2 PRI is most desirable because it is generally the easiest one to work with. In addition to the normal calling party number delivery, the OfficeServ 7200-S will support calling party name delivery on a #5ESS Custom or on a NI2 PRI. Name delivery is supported on a DMS 100 type circuit.

2. The Telephone Company will usually ask about High/Low Selection. Typically, to avoid glare conditions, the Telephone Company and the OfficeServ 7200-S need to start searching from opposite ends of the “B” channel list when each attempts to select a “B” channel for call handling purposes. For example, let’s say that you have a PRI with eight (8) “B” channels. The channels are numbered 1 through 8. The Telephone Company suggests that in completing incoming calls to the PRI that they will start hunting from the highest numbered “B” channel (i.e., 8) to the lowest numbered “B” channel (i.e., 1) while attempting to find an idle “B” channel to use for a specific call. This means that the system, when looking for an idle “B” channel to service an outgoing call attempt, should start hunting from the lowest “B” channel (i.e., 1) to the highest “B” channel (i.e., 8). This agreement needs to be established between the Telephone Company equipment and the OfficeServ 7200-S customer premises equipment for service to be provided.

3. The “B” channels should be requested optioned for switched voice/data service. Thus, they will be able to handle both voice (e.g., speech) and data (e.g., Internet access) traffic.

4. The OfficeServ 7200-S presently supports a single “D” channel for from 1 to 23 “B” channels. The OfficeServ 7200-S does not presently support NFAS (i.e., Non-Facility Associated Signaling which allows a single “D” channel to control “B” channels on more that one PRI T1 digital span) nor does it presently support a “backup D-Channel”.

5. The Telephone Company will ask the number of “B” channels and telephone numbers you want to have provided. There is no relationship between telephone numbers and “B” channels. The number of “B” channels desired is proportional to the facility traffic handling capability similar to a DID trunk group on a T1. The PRI can support any number of telephone numbers since the telephone number simply allows a specific terminating destination in the OfficeServ 7200-S when a call is placed to that specific telephone number. Again, this is similar to DID service on a T1. Typically, the Telephone Company

charges for both the number of "B" channels provided and for each telephone number provided.

6. As indicated in item 5, incoming calls on a PRI act in a manner very similar to DID service over a T1. Now, the Telephone Company can send you a 3, 4, 7, or 10 digit telephone number as the "Called Party Number" when an incoming call is received. This information must be known and agreed upon by both the Central Office (CO) and Customer Premises equipment (CPE) since the "Called Party Number" must be programmed in MMC714 for routing calls for this number to a specific station, station group, etc. Further, the Telephone Company can transmit a 7 or 10 digit "Calling Party Number" to the premises equipment. This is another point that must be agreed upon by both the CO and CPE so that CPE users can identify where the call was placed from. Typically, a 10 digit "Calling Party Number" is the normal choice of most people.

7. By definition, U.S. ISDN PRI uses B8ZS line code and ESF (i.e., extended superframe) format. This is provided automatically when a PRI is setup and is not a selectable option. The CSU (customer service unit) used to terminate the T1 span supplying PRI service should accommodate this.

8. Prior to cutting the PRI into service, the Telephone Company may ask you to "loop" the CSU (i.e., place the CSU in loopback mode) for test purposes. Only do this if the Telephone Company asks you to do this. You should never loop a PRI with an active "D" channel unless instructed to do so by the Telephone Company since this can cause a problem condition (i.e., the ISDN "D" channel will try to communicate with itself in a "looped" PRI situation).

2.10 OfficeServ 7200-S Voice over IP (VoIP)

Introduction

This OfficeServ 7200-S Technical Manual Special Applications section is intended to introduce qualified personnel to the following OfficeServ 7200-S VoIP features:

- Supporting the VoIP trunking gateway features via the OAS card.
- Supporting Samsung proprietary IP keyset. OfficeServ 7200-S can support up to 64 IP stations, which can be combination of any of the following: WIP 5000 Wireless Handset, ITP 5100 Series, or Softphone.
- IP Networking: OfficeServ 7200-S has the ability to network up to 99 systems together with a high level of feature integration. The networked systems may be any combination of OfficeServ 100, OfficeServ 500, and OfficeServ 7200-S systems running V2.46 or higher software.

It is highly suggested that a basic knowledge of IP network topology and networking equipment functions be known prior to applying these services. Infrastructures of private Intranet and public Internet applications should also be familiar subjects.

Overview

The OAS VoIP series card is a standard and proprietary based Voice over Internet Protocol option card that provides toll quality voice and seamless integration with the OfficeServ 7200-S systems via an IP network.

The OAS and the MP20S cards also support Samsung's proprietary ITP 5100 series IP keyphone, WIP 5000 Wireless Handset, and Softphone. VoIP converts voice and signaling into IP packets, which can be transmitted over any TCP/IP network so that voice looks like data. VoIP provides the solution for desktop accessibility in the enterprise business environment. The OfficeServ 7200-S OAS card(s) can provide up to sixteen simultaneous voice conversations per card over an IP network. The easy addition of OAS boards in the OfficeServ 7200-S systems requires no integration into a server or desktop personal computer. VoIP calls can be established from desktop telephone instruments without complex integration of software or hardware. VoIP eliminates the cost of maintaining separate networks for voice communications in the Intranet environment.

VoIP Overview

VoIP is transported by the OfficeServ 7200-S OAS card utilizing the standards based Internet Protocol. This standard addresses the means of transferring voice, data, and images through IP (Internet Protocol) networks. IP is the accepted protocol standard for transporting data. With VoIP certain compression standards have also been adopted to represent each second of voice with an amount of bandwidth. The OfficeServ 7200-S OAS cards can utilize G.711, G.729A, G729 or G.723 standards voice compression codec's (card dependant). This allows for a selectable 64kbps, 8Kbps or 6.3Kbps bandwidth use when preparing voice compression for IP transport on a particular network. Compression is used to reduce the digitized voice into a smaller bandwidth that can be carried in smaller packets. The OAS card can determine the compression method for each individual call setup. There is also a certain amount of frame/packet overhead in each compression channel. 64Kbps of bandwidth can support 4~5 calls simultaneously. This can vary depending on efficiency features like codec selection, Silence Suppression and Multiframe counts. Unlike switched networks, VoIP connections use the packet switching method and consist of a sequence of numbered of data packets. Since voice conversation is usually considered “real time” these packets need to be delivered in a consistent manner with minimal delay. The OfficeServ 7200-S OAS is Gatekeeper compliant.

In any Ethernet environment, packet transfers are subject to delays and/or loss. If these delays are great the voice quality will deteriorate severely. The Ethernet data traffic and network topology should be a consideration when applying VoIP transport. Network congestion does affect call quality in any VoIP application.

ITP 5100 Series Keyphone Overview

The Samsung ITP 5100 series IP keyphones provide full iDCS keyphone functionality locally or remotely on an IP network. The unique ITP keyphone instruments communicate with the OfficeServ 7200-S system in a proprietary IP message format to emulate full keyphone functionality. Working in conjunction with the MP20S and an OAS card the ITP keyphone can access all station features and trunk facilities on the OfficeServ 7200-S system. The ITP keyphone can function on private networks (Intranet) or public IP networks (Internet). ITP keyphones support **DHCP** services and can also function behind **NAT (Network Address Translation) routers**. There are two types of ITP 5100 series keyphones:

ITP 5100 Series Keyphone Overview

The Samsung ITP series IP keyphones provide full iDCS keyphone functionality locally or remotely on an IP network. The unique ITP keyphone instruments communicate with the OfficeServ system in a proprietary IP message format to emulate full keyphone functionality. Working in conjunction with the MP20S and an OAS card the ITP keyphone can access all station features and trunk facilities on the OfficeServ system. The ITP keyphone can function on private networks (Intranet) or public IP networks (Internet). ITP keyphones support **DHCP** services and can also function behind **NAT (Network Address Translation) routers**. There are three types of ITP 5100 series keyphones:

- **ITP-5121D and ITP-5107S**

  The **ITP-5121D** and **ITP-5107S** have 21 and 7programmable buttons respectively and full traditional iDCS keyset functionality. The LCD display is a 2 line 36 character display that informs the user as to call status and call states. The LCD also indicates incoming call parameters. Traditional features such as Station Camp-on, Station Paging, Offhook Voice Announce are seamless with the ITP series keyphones. Other traditional buttons such as Transfer, Hold, Speaker, Volume control, Redial, Conference and the Samsung intuitive Softkeys make them truly functional. In addition to the standard keyphone keys there is a navigation button that allows scrolling though various keyphone menu parameters (navigation key is not available on 5107S model). See the ITP 5100 Series Keyphone User Guides for more details.

Figure 1. ITP-5121D Keyphone and ITP-5107S Keyphone

- **ITP-5112L**

The **ITP-5112L** is a large LCD color display ITP keyphone. The large display features interactive softkeys to navigate through system keyphone features. The large display supports a visual status bar, title bar, Icons associated with 12 selection buttons and call aid graphics. The LCD also indicates incoming call parameters via Icons. Other traditional buttons such as Transfer, Hold, Speaker, Volume control, Redial, Conference make the ITP-5112L truly functional. In addition to the standard keyphone keys the navigation button also allows scrolling though various keyphone menu parameters. See the ITP 5100 Series Keyphone User Guides for more details.

Figure 2. ITP-5112L Keyphone

OAS Hardware Overview (OAS) Version 2

The OAS (Optional Application Services) card running version 2.x software or higher supports a diverse range of features and applications. The selectable features are H.323 VoIP trunking, SIP VoIP trunking, OfficeServ 7200-S System Networking via IP and the Samsung ITP 5100 Series keyphones. The OAS card also supports MOBEX services with a selectable range of Media Gateway ports and MOBEX DTMF services. Each OAS card has an embedded RJ45 10/100 BaseT Ethernet female connector for IP network connections. LED's on the front of the card provide operational status. The OAS cards automatically detect 10BaseT or 100 BaseT Ethernet networks. There are no selectable options on the OAS card.

The OAS card has a recessed RESET button that will initialize the MGI card manually if required. To load the flash memory of the OAS to the DRAM after power up or after resetting the system takes approximately 3 minutes.

The SIO connector on the face of the OAS card allows trace monitoring of the OAS functions. This is used for engineering purposes only.

OAS CONFIGURATION OPTIONS

MOBEX	MGI
0	16
16	12
24	8
28	4
32	0

The OAS card has the following set of options set in MMC 858. Here are the options per OAS card.

1. MOBEX 32 ONLY
2. MGI-4 MOBEX 28
3. MGI-8 MOBEX 24
4. MGI-12 MOBEX 16
5. MGI-16 ONLY

The MGI functions of the OAS card support the following functions.

- IP Phones
- IP Networking (Network multiple systems over an IP Network)
- G729 (8K) CODEC, G.723.1, G.711, G.729A CODECs
- SIP and H.323 IP Trunking
- T.38 Fax CODEC
- Inband or out-of-band signaling of DTMF tones

OAS Card Installation (OAS)

An OAS card can mount in slot 3, 4 and 5 of the OfficeServ 7200-S cabinet running an MP20S processor card. There is a processor on the OAS cards, so it is not recommended to insert this card with system power on.

Figure 3. OAS Card

System OAS Card Capacity

For the OfficeServ 7200-S running MP20S processor, a maximum of 3 OAS cards can be installed in each cabinet.

OAS Power Requirements

The OAS card does not use any power units against the maximum power supply requirements.

Card Insertion

Power OFF the OfficeServ 7200-S system. Insert the required quantity of OAS cards into the appropriate card slots. Push firmly in the middle of both card ejectors on each card to ensure that it is fully inserted into the backplane connector. Connect IP network cables. Power ON the OfficeServ 7200-S system.

OAS and MP20S IP Network Connection Cable

The cable needed to connect the OAS card to an IP network is a standard “straight” 8 conductor data cable. This cable is sometimes referred to as a LAN UTP Category 5, 4 pair RJ45 male to male data patch cable. Connection to the network can be via a patch panel or data hub type device. The data patch cable is not provided with the OAS card and must be purchased separately through a data equipment supplier.

NOTE: The Ethernet network cables on the OAS and MP20S cards MUST be connected to the data network upon power up to detect the Ethernet connection. The OAS or MP20S cards CAN be connected after power up but must be reset to detect the network connection.

OAS Card Labeling

Ports & LEDs	Function Description
LAN	Port that connects the Ethernet.
SIO	UART port (for tests).
RST	Button for resetting the MGI.
PWR LED	This LED indicates the power supply status. - Off : Power is not being supplied. - On : Power is being supplied properly.
RUN LED	This LED indicates MCP/MP20 status. - Off : Power is not being supplied. - On : Booting. - Blink : The RAM program is operating.
LAN Tx LED	This LED indicates the status of the Ethernet data transmission. - Off : Data does not exist. - On or blink : Data is being transmitted.
LAN Rx LED	This LED indicates the reception status of the link and Ethernet data. - Off : Data does not exist or the link is not connected. - On or blink : Data is being received.
SVC LED	This LED indicates if the service is being offered. - This LED blinks when the software task can be serviced.
DSP LED	This LED indicates if the VoIP DSP is operating. - This LED blinks when the VoIP DSP operates.
RTPT LED	This LED indicates if the voice packets are being forwarded. - This LED turns on when the voice packets are forwarded.
RTPR LED	This LED indicates if the voice packets are being received. - This LED turns on when the voice packets are received.

OAS Operation Overview

The OAS cards have the voice CODEC's for VoIP and an RJ45 10/100 10BaseT connector for the network connection. The MP20S supports the controls in relation to the OAS card(s), i.e. setup, CODEC selection, availability, CODEC usage type etc. The MP20S also has an RJ45 10/100 BaseT connector. BOTH the OAS and MP20S cards must be connected to the same network together. This is required to reduce delay of information if traversing routers and other data network equipment.

Unique programming allows the access to the private Intranet and Public Internet services simultaneously. This is accomplished via programming static IP addresses for specific uses.

The OAS card use is distributed via channels. Port reference is used to accommodate a more static perception of the OAS channel identity. For simplicity the word port will be used to reference OAS card CODEC channels.

The OAS cards can be used for several functions simultaneously or individually. Each OAS port will auto negotiate the service of each port for the type of usage required. This means that the same port can use SIP protocol for one call and H.323 for the next call without any programming changes. With multiple OAS cards installed, service grading of the MGI services can be programmed so that specific OAS cards can be selected for different uses. This grading is similar to standard trunk grading where the number of users do not match the number of available lines. If a one to one ratio is required for total non-blocking service this can be accomplished via programming and selection of the number of simultaneous uses in proportion to the number of OAS ports.

OAS card(s) use of services can also be shared. MMC 615: OAS Group, allows the segregation of services to specific cards. MMC 724: Dial Numbering Plan, allows addressing of the default numbers and trunk assignments to customize the application. MMC 831: OAS Parameters provides the OAS card(s) with IP address information. The OAS Services will be explained in detail in the OAS Functional Overview section found later on in this Application Section. The OAS services will be defined in MMC 858 separating the media gateway and DTMF services for MOBEX support.

OAS Services (OAS):

- H.323 Trunking Gateway
- SIP Trunking Gateway
- ITP keyphone CODEC support
- VoIP Networking Gateway
- MGI Requests (facsimile over IP)
- MOBEX (DTMF Services)

MP20S (Main Control Processor Release)

The MP20S provides a robust IP platform on the OfficeServ 7200-S. The MP20S supports the default LAN connection via the RJ 45 10/100 BaseT port on the face of the card. This connects to the same IP network that the OAS card(s) are connected to. (ITP keyphones can be on the same or a different network). The OfficeServ 7200-S system IP address is registered on the MP20S card. All IP connectivity is directed to the MP20S IP address. MMC 830: Ethernet Parameters, provides the selections for the system MP20S IP address.

OAS and MP20S Card Relationships

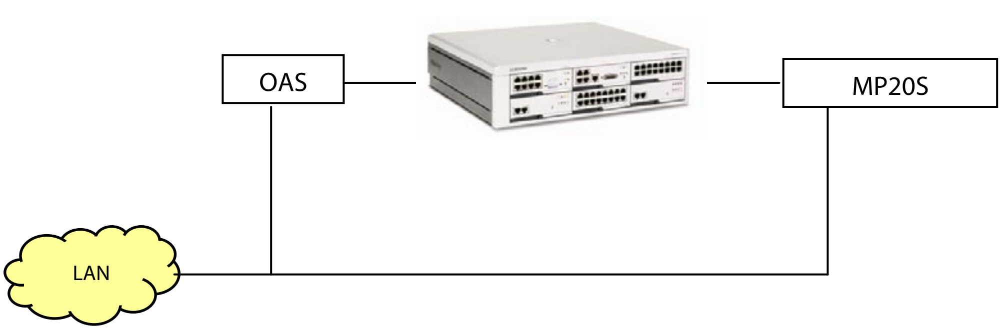
Theory of Operation

When an IP call is directed to the OfficeServ 7200-S system the MP20S performs the call control and setup to the remote calling or called IP endpoint. This endpoint can be a legacy Samsung SMG series H.323 gateway, Samsung ITM series card or other manufacturer standards based VoIP gateway.

Once the MP20S negotiates the setup a determination is made if the IP call is intended to be received by another IP device on the IP side or a TDM port on the circuit switched side of the OfficeServ 7200-S. When the request is for a circuit switched side connection, the MP20S controls the OAS card and sets up a channel for the IP call to use a codec on the OAS as a gateway to the circuit switched environment. The OAS supports the voice codec to the trunks or stations inside the OfficeServ 7200-S system. All other controls are managed by the MP20S. i.e. Setup, teardown, LED status updates in the case of the ITP series keyphones.

The same theory of operation applies on an incoming IP call. The called **IP address** is the MP20S card. The MP20S informs the calling IP party that the voice portion of the call will be managed by the OAS cards **IP address**. See Figure 3.

Figure 3. OAS and MP20S Relationship



ITP Keyphone, OAS and MP20S Relationships

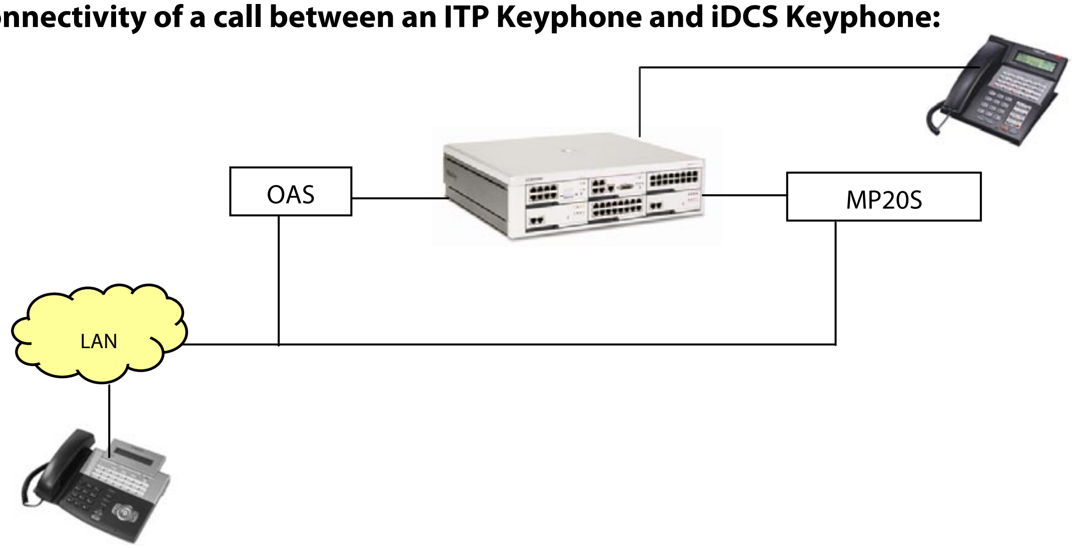
ITP Keyphone and iDCS Keyphone Operation

The ITP 5100 series keyphones use the OAS as the voice gateway when connected to the OfficeServ 7200 system stations or trunks. The MP20S is involved to provide call setup and tear down, programmable button functions, displays and system feature access.

ITP 5100 series keyphones provide Auto Codec Negotiation and follows the VoIP codec settings of the OAS card(s) used. There can be more ITP keyphones than available OAS channels. This is due to voice traffic possibilities. If it is not anticipated that all ITP keyphones will be in use at the same time the probability factor of not having a voice channel can be calculated based on the station activity or use. If an ITP keyphone attempts to use an OAS card that does not have a voice channel available the ITP keyphone display will show “No OAS Channels Available”

The ITP 5100 series keyphones must be programmed to provide the keyphone IP address, subnet mask, router gateway (if used) and MP20S (IP Server) address. Each ITP keyphone must also have a User ID and Password. This User ID and Password must match the User ID and Password in the OfficeServ 7200-S system (MMC 840: IP SET INFO). The ID and Password are used to register the ITP keyphone to the OfficeServ 7200-S and retain the number associated with that particular ITP keyphone. The ITP keyphones will register with the MP20S and be provided a default extension number from the OfficeServ 7200-S system provided the ID and password match. The assigned default numbers can easily be changed to match the iDCS keyphone station numbering plan. New ITP keyphone registrations can be enabled or disabled via MMC 840 and MMC 841: IP SEN INFO, SYS IP OPTN. See Figure 4 for the relationship of ITP keyphone and iDCS keyphones.

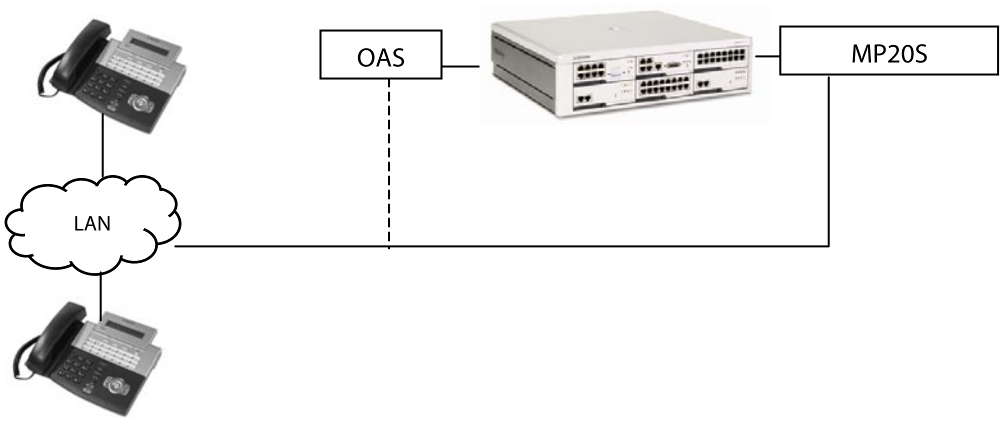
Figure 4. Connectivity of a call between an ITP Keyphone and iDCS Keyphone:



ITP Keyphone to ITP Keyphone Operation

When an ITP keyphone is connected and in conversation to another ITP keyphone the OAS card audio codec is not involved when called on the private network side. The MP20S is involved to provide call setup and tear down, programmable button functions, displays and system feature accesses. However, an OAS port is used for system features such as conference and transfer or internal paging as needed. See Figure 5.

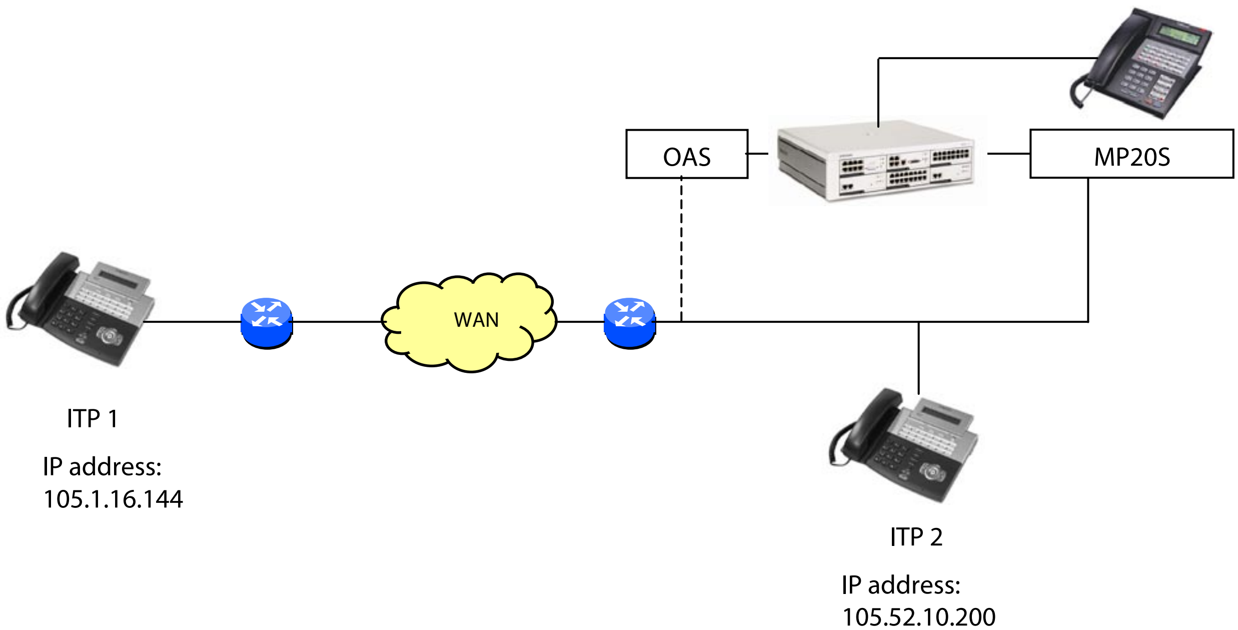
Figure 5. Connectivity between ITP Keyphones:



Remote ITP Keyphones

The ITP 5100 series keyphones are unique in the fact that they function beyond the confines of the existing LAN and will work normally on a WAN (Wide Area Network) infrastructure and traverse routers and switches when connected remotely. The ITP keyphone simply needs to know the MP20S server IP address and have the matching ID and Password. The ITP Server is the address of the MP20S OfficeServ 7200-S system supporting that ITP keyphone. The ID and Password are used to register the ITP keyphone to the OfficeServ 7200-S and retain that number associated with the registered ITP keyphone. The data infrastructure (IE routers and switched hubs) allows connectivity via IP between the ITP keyphone and the MP20S. Note that in Figure 6 that ITP 1 is on a different network than ITP 2. Public IP and Private IP relationships will be covered further on in this Special Application Guide. See Figure 6.

Figure 6. Remote ITP Keyset (shown with iDCS keyphone connectivity also)



OfficeServ 7200-S OAS Functional Overview

OAS Operational Concepts

The OAS cards support several VoIP services. In supporting multiple services the OAS ports are capable of using different protocols (IE Proprietary, SIP or H.323) automatically on a per port basis as needed. This allows the mixing of different standards on the same OAS card(s).

In certain applications there may be a need to separate or segregate the card(s) for specific uses. An example of this is where there are two OAS cards mounted and one card is used exclusively as a VoIP trunk gateway and the second card is used to support ITP keyphones. In this case each card can be defined as to what service it will provide. IE. VoIP trunking gateway use or ITP keyphone use. In another case, where the OAS has 16 VoIP channels, it is possible to segregate or partition some of these channels for ITP keyphones and use the remaining channels for trunking gateway use on the same card.

In order to understand the task of segregating OAS cards for traffic or grading purposes the following is a brief description of the multiple uses of an OAS card. MMC 615: OAS Group allows separation of these services. By default all OAS ports are available for all services.

OAS Services

Each of the MGI services provide for private network and public network exposure. This allows connectivity to the public Internet without exposing the private network. This allows **router** traversal by remote ITP keysets and far end gateway services and systems. Public network and Private network relationships will be reviewed further on in this Special Applications section.

H.323 Trunking Gateway

The OAS card(s) when used as a VoIP trunking gateway allows VoIP call connectivity to opposite MGI's, ITM products, SMG products or other H.323 standards based VoIP gateways. Calls placed in this mode are treated like IP tie line trunk calls. As a tie line type trunk, incoming digits are expected. Programming allows routing of incoming calls based on incoming digits, digit translation or trunk ring destinations. Outgoing calls are made by selecting the trunk or trunk group and dialing the appropriate digits to reach a predetermined destination. The flexible programming allows outgoing digit manipulation with LCR like routing.

SIP Trunking Gateway

The OAS card(s) when used as a SIP (Session Initiation Protocol) trunking gateway allows VoIP call connectivity to opposite MGI's, or other SIP standards based VoIP gateways or devices. Calls placed in this mode are treated like IP tie line trunk calls. As a tie line type trunk, incoming digits are expected. Programming allows routing of incoming calls based on incoming digits, digit translation or trunk ring destinations. Outgoing calls are made by selecting the trunk or trunk group and dialing the appropriate digits to reach a predetermined destination. The flexible programming allows outgoing digit manipulation with LCR like routing.

SIP trunking gateway programming follows the same programming as the H.323 programming but has an additional MMC to address the SIP attributes. MMC 837: SIP Options, includes general SIP inquiries, response times, UDP ports, SIP server addresses and other related SIP options.

ITP Keyphone Support

The ITP keyphones use the CODEC's on the OAS card when calling a circuit switched digital keyphone or an outside trunk in the OfficeServ 7200-S system. The MP20S controls the setup messages and determines if the destination (IE iDCS keyphones or outside trunks) require the use of the OAS card. The ITP keyphones only use the OAS card when accessing the OfficeServ 7200-S circuit switched system side. ITP to ITP calls do not use the OAS card(s) when called on a private network. This aspect of use can determine how many OAS cards need to be installed based on ITP station traffic considerations. The number of ITP keyphones does not need to match the number of OAS channels unless a complete non-blocking application is required.

VoIP Networking Gateway

The OAS supports the voice when connections between OfficeServ 7200-S systems are set up to use the IP Networking Feature. IP Networking Features are the same as the PRI Networking features but use IP as the transport method. Point to point calls are originated and terminated as VoIP trunks but the informational data is exchanged via IP packets between MP20S cards.

Facsimile over IP

This option assigns the number of ports used to support IP facsimiles. The OAS card supports the T.38 facsimile over IP standard.

MGI Group Numbering Plan

When using MMC 615: OAS Group, an understanding of the multi-use aspect of the OAS should be understood. There is a correlation between the Numbering Plan and Trunk Group Members. To provide a perspective of the port assignment on the OAS card(s) Table 1 should be used as an example to understand default assignments. For brevity the first 8 port assignments are shown in the example.

MMC 615: OAS Group Assignments: Selects which OAS ports will be accessed based on the call type. Default assignments allow all ports to connect to all services.

Local ITP: This refers to ITP keyphone stations on a private network.

Public IP ITP: This selects ports to be used by ITP keyphone by the public or exposed side of the network.

VoIP Network: This refers to ports used when using the networking features on a private network. These ports will transport the voice side of an OfficeServ 7200-S system networking call.

Public IP Network: This selects ports that will be used when using the networking features via a public or exposed IP networking. These ports will transport the voice side of an OfficeServ 7200-S system networking call.

VoIP Trunk: This refers to the OAS ports that will be used for VoIP trunking gateway calls on a private network.

Public VoIP Trunk: This refers to the OAS ports that will be used for VoIP trunking gateway calls via a public or exposed network.

Home Page

Table 1: MGI Group default Numbering Plan Assignments:

MGI PORT/ MEMBER	MGI NUMBER PLAN	LOCAL ITP GROUP MEMBER MMC 615	PUBLIC ITP GROUP MEMBER MMC 615	ITP STATION NUMBER PLAN IDX MMC 724	IP NETWORK GROUP MEMBER MMC 615	PUBLIC IP NETWORK GROUP MEMBER MMC 615	VOIP NETWORK NUMBER PLAN MMC 724	VOIP TRUNK MEMBER MMC 615	PUBLIC VOIP TRUNK MEMBER MMC 615	VOIP TRUNK NUMBER PLAN MMC 724	VOIP SIP TRUNK NUMBER PLAN MMC 724	ITP PAGED
3801	3801	3801	3801	3201	3801	3801	8301	3801	3801	8401	8501	8501
3802	3802	3802	3802	3202	3802	3802	8302	3802	3802	8402	8502	8502
3803	3803	3803	3803	3203	3803	3803	8303	3803	3803	8403	8503	8503
3804	3804	3804	3804	3204	3804	3804	8304	3804	3804	8404	8504	8504
3805	3805	3805	3805	3205	3805	3805	8305	3805	3805	8405	8505	8505
3806	3806	3806	3806	3206	3806	3806	8306	3806	3806	8406	8506	8506
3807	3807	3807	3807	3207	3807	3807	8307	3807	3807	8407	8507	8507
3808	3808	3808	3808	3208	3808	3808	8308	3808	3808	8408	8508	8508

*SIP trunks are VoIP Trunk members in MMC 615: MGI Group.

For traffic considerations, (IE CODEC availability at any given time) specific ports may be selected to be accessed for defined uses. The other services are not used so they are removed for clarity.

Table 2: OAS programmed assignments

MGI PORT/ MEMBER	MGI NUMBER PLAN	LOCAL ITP GROUP MEMBER MMC 615	PUBLIC ITP GROUP MEMBER MMC 615	ITP STATION NUMBER PLAN IDX MMC 724	VOIP NETWORK GROUP MEMBER MMC 615	PUBLIC IP NETWRK GROUP MEMBER MMC 615	VOIP NETWORK NUMBER PLAN IDX MMC 724	VOIP TRUNK MEMBER MMC 615	PUBLIC VOIP TRUNK MEMBER MMC 615	VOIP TRUNK NUMBER PLAN IDX MMC 724	SIP TRUNK NUMBER PLAN IDX MMC 724
3801	3801	3801		3201							
3802	3802	3802		3202							
3803	3803	3803		3203							
3804	3804	3804		3204							
3805	3805			3205				3805		8405	
3806	3806			3206				3806		8406	
3807	3807			3207				3807		8407	
3808	3808			3208				3808		8408	

*SIP trunks are VoIP Trunk members in MMC 615: MGI Group.

In the above numbering plan example there is a one to one relationship of available MGI ports and ITP keyphones.

Further refinement to the numbering plan assignments can be made by adding the VoIP trunks to a specific trunk group. (IE. 802, 803 etc.)

H.323 or SIP Trunking Gateway

The OfficeServ 7200-S MGI functions as a VoIP trunking gateway. The OfficeServ 7200-S software considers the MGI ports as trunk ports. As a trunking gateway the OAS card support the H.323 or SIP standards on a per port basis as needed. The OfficeServ 7200-S OAS card programming is similar to installing or adding other cards in the OfficeServ 7200-S system. Each OAS card has ports assigned that are comparable to the system trunk ports. In a default system the OfficeServ 7200-S will automatically identify that an OAS card is present and assign trunk numbers to the available ports. The trunk numbers will be in the 84XX range. If an OAS card is to be added to an existing system it is recommended to power off the cabinet that the card is going to be installed in.

Programming MGI trunks are similar to programming traditional trunks. The MGI VoIP trunks can be in trunk groups (MMC 603 Assign Trunk Groups) or MGI VoIP trunks can be individually assigned as Direct Trunk Keys (MMC 722, MMC 723 Key Assignments). MGI VoIP trunks can also be included in LCR programming to provide alternative routing of outgoing calls. Station calls between OfficeServ 7200-S systems via VoIP can also be accomplished via MGI trunks. An MGI or other compatible VoIP gateway equipment must be present on the LAN or WAN to place VoIP calls to and from the OAS card.

VoIP Trunk Call Routing Concept

VoIP calls via the MGI use routing tables to determine where to route the call based on digits dialed. Digits can be added or removed as part of the call and be transparent to the calling station. The digits dialed then reference a table that has the destination IP address then repeats the digits needed at the far end.

Programming of the MGI follows an LCR type of programming where the tables are used to reference the dialed number and send the call to the right IP destination. It is not necessary to have System feature LCR programmed to use the MGI VoIP facilities. Incoming calls via the MGI can be directed three different ways, Follow the DID Translation table, Follow Incoming Digits or Follow Trunk ring assignments. These settings are system wide and affect all OAS cards used as trunking gateways. The outbound routing tables are referenced by OAS cards system wide.

A basic example of the direct station to station dialing string in an OfficeServ 7200-S directing a call over an OAS card to another OfficeServ 7200-S OAS card is as follows:

80 0 201

- 80 is the trunk group access code to the MGI trunks.
- 0 is the access code that references an IP address table.
- 201 is a station in the distant OfficeServ 7200-S system.

In the above example the station caller will hear a second dial tone when 80 is dialed. After dialing 0 and station number 201 the MGI program looks up the corresponding IP address with the access code 0. IP communications are established between the MP20S of the originating system and the far end MP20S. The access code 0 is deleted then the digits 201 are sent in an IP packet and repeated by the far end MGI. The caller will hear station ringback tone and is able to converse when the called party answers. The access code can also be imbedded in the sent digit string information allowing more transparent dialing.

This concept is used only when using the MGI as a trunking gateway. IP Networking uses the communications between MP20S cards to determine the MGI use so trunking gateway inputs are not required.

Internet Protocol (IP) Addressing

To program the MGI VoIP it must be known where to route IP based calls. A basic knowledge of Internet Protocol (IP) networking should be understood to program IP addresses and IP gateway addressing. The OfficeServ 7200-S must have an IP address associated with the MP20S and each MGI card installed. The OAS cards can be programmed only after installed in the system. The IP address can match the existing LAN (Local Area Network) addressing scheme or it can be a totally different IP address level. It is suggested to match the existing LAN IP address plan to permit ease of administration. Also required will be an IP subnet mask and a gateway address.

Subnet addresses allow connectivity in the same network and determines if the data needs be forwarded to a router.

Gateway is a term used as a junctor or the meeting point of two networks. This meeting point can be a router or an OAS card. When a request is made by a PC or other IP device (MP20S or OAS) and the address is not in the local network the gateway IP address is where the next search is performed. The gateway then looks outside the local network for a response to the request. This all happens within milliseconds. Once a response is obtained, the LAN gateway router acts as the bridge between the two networks. The information can be strictly data or it can be voice data. In the OfficeServ 7200-S MP20S and OAS, programming the gateway is equivalent to the router address to leave the local LAN and access another network.

MGI Programming Responses

The MGI will not update while programming. The update of MGI data changes take effect after programming has been exited and closed. This can be observed by the IPC message activity on the MP20S card LED’s.

If the IP address of the OAS card is changed the MGI must be restarted to update the new IP address information. The update takes about 3 minutes to write to and then retrieve from flash memory.

ITP Keyphones and IP Network Connectivity

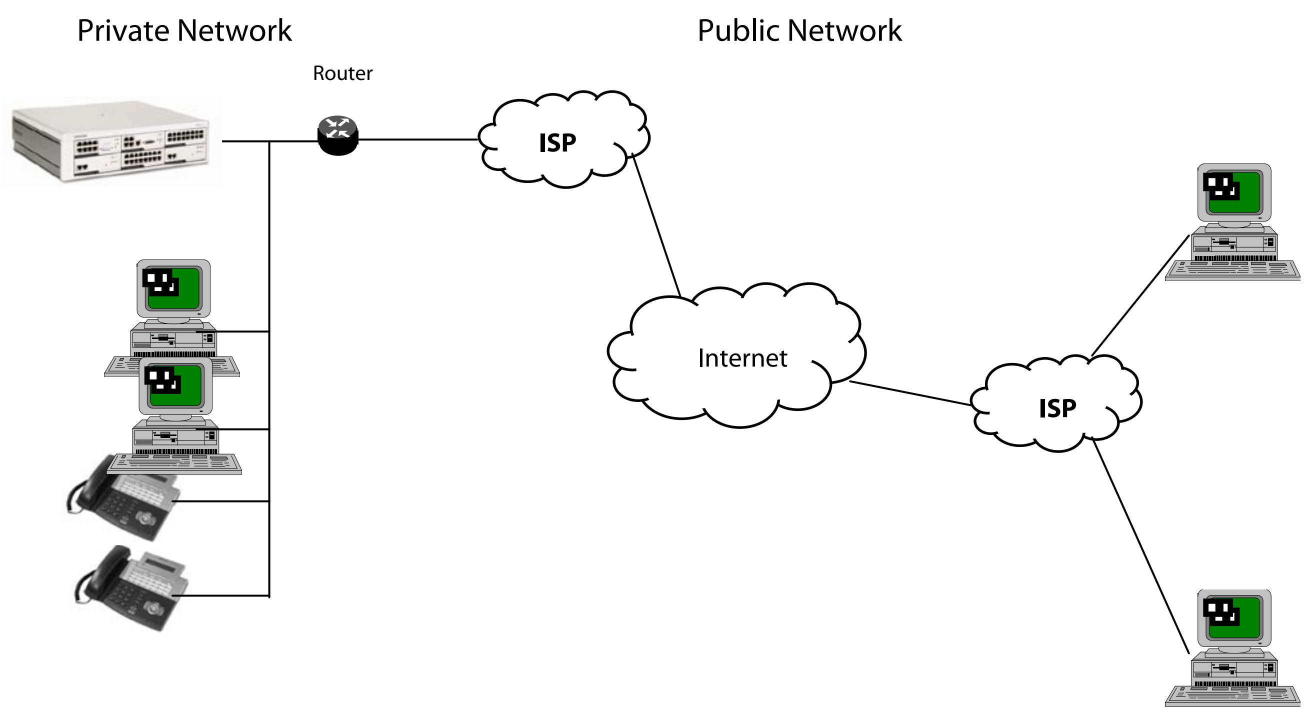
The OfficeServ 7200-S OAS cards provide a proprietary gateway between the ITP keyphones and the system features and services. The connections between networks should be understood before requesting or installing the services for local or remote location ITP keyphones.

Public and Private IP Connectivity

The MP20S, OAS and ITP keyphones can be used on the private Intranet LAN/WAN and also have a public Internet presence. This is accomplished by defining the public address on the MP20S and OAS cards. ITP keyphones can also be registered as public IP addresses but still communicate with the private network stations and system. Figure 7 shows the relationship between public and private networks.

In any case with Public network exposure there must be a public address for the MP20S and each MGI that will connect to the Public network. The remote ITP must have at least one public IP address available. When using a NAT router at the remote ITP end there only needs to be one public IP address.

Figure 7. Public and Private Networks



“Public” networks can be defined as a network that is open to the public Internet. Anyone with access to the Internet can access the endpoint(s) via an IP address. This endpoint can be a server, email server, PC or router.

"Private" network is the network that has an IP addressing scheme that is used behind a router to protect the user endpoints from being accessed by the public. In some cases the router has a "NAT" (Network Address Translation) setting. This allows the user(s) behind the NAT router to look as a single address to the public network.

In the OfficeServ 7200-S "public" and "private" do not necessarily actually mean public and private. These are relational terms to determine if the MP20S must look at source IP addresses and convert to a destination address. This is done when there are two different networks. As an example, if all the IP addresses are exposed to the "public" unprotected Internet network. The programming for the MP20S, OAS, and ITP keyphones would be programmed as private. They are all on the same network.

If the MP20S, OAS and several ITP keyphones were on a LAN based "private", protected network, behind a router and the Internet IP address was in front of the router and "exposed" to the public Internet the programming for the MP20S, OAS and local ITP keyphones would be private. In this case the MP20S, OAS and remote ITP keyphone would also have public IP entries to show the external Internet addresses for translation to the internal Intranet. This external public entry address allows the MP20S to convert the external IP address to the internal private addresses.

Customer Provided Routers

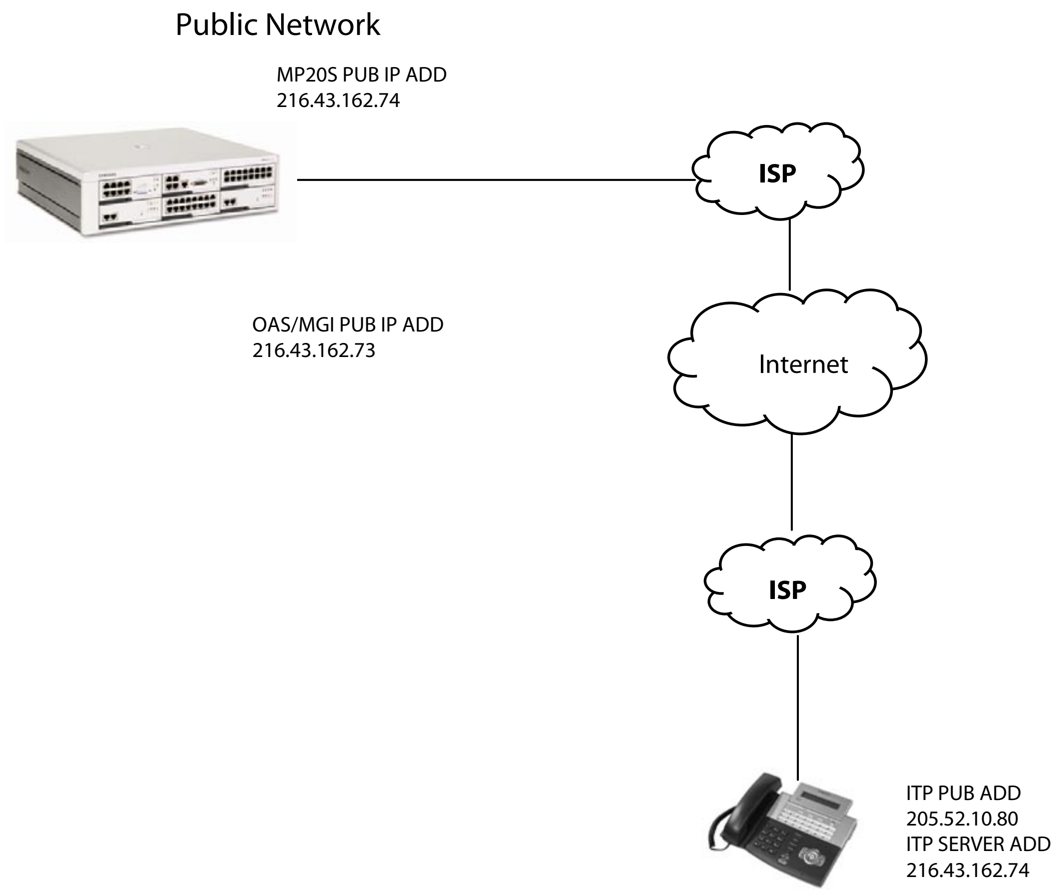
Exposure of the private network to the public can compromise the security of the private network. To use the Private with Public relationship the networks are usually separated by a router and/or firewall. The routers used MUST be able to support "reservations" or "port forwarding" feature. The Reservations feature in routers allows the router to "reserve" an internal IP address to be permanently connected or forwarded to an external IP address. This permits packets sent to and from the external IP address to be sent to the internal IP address. Different router manufacturer's support varying numbers of reservations or port forwarding. Without this feature in the customer supplied router the OfficeServ 7200-S Private with Public IP addressing will not function and all address must then be exposed to the public network. A router supporting this feature permits the VoIP features while also allowing the customer to access the public Internet from their LAN based PC's (separate IP address required).

In most applications the equipment supporting reservations would be located at the OfficeServ 7200-S system side. The remote location can have a either a public static IP address for each ITP keyphone, a NAT supporting router with a single public IP address with multiple ITP keyphones.

Remote ITP Keyphone with Public Static IP Addressing

Figure 8 shows the relationship between the MP20S, OAS and remote ITP keyphone. With the entire network “exposed” connectivity is permitted but there is no way to put ITP keyphones on a private LAN network without “exposing” the private data network to the public domain. In this topology the programming in the OfficeServ 7200-S for the MP20S, OAS and ITP keyphone can be considered Private because they are all on the same network.

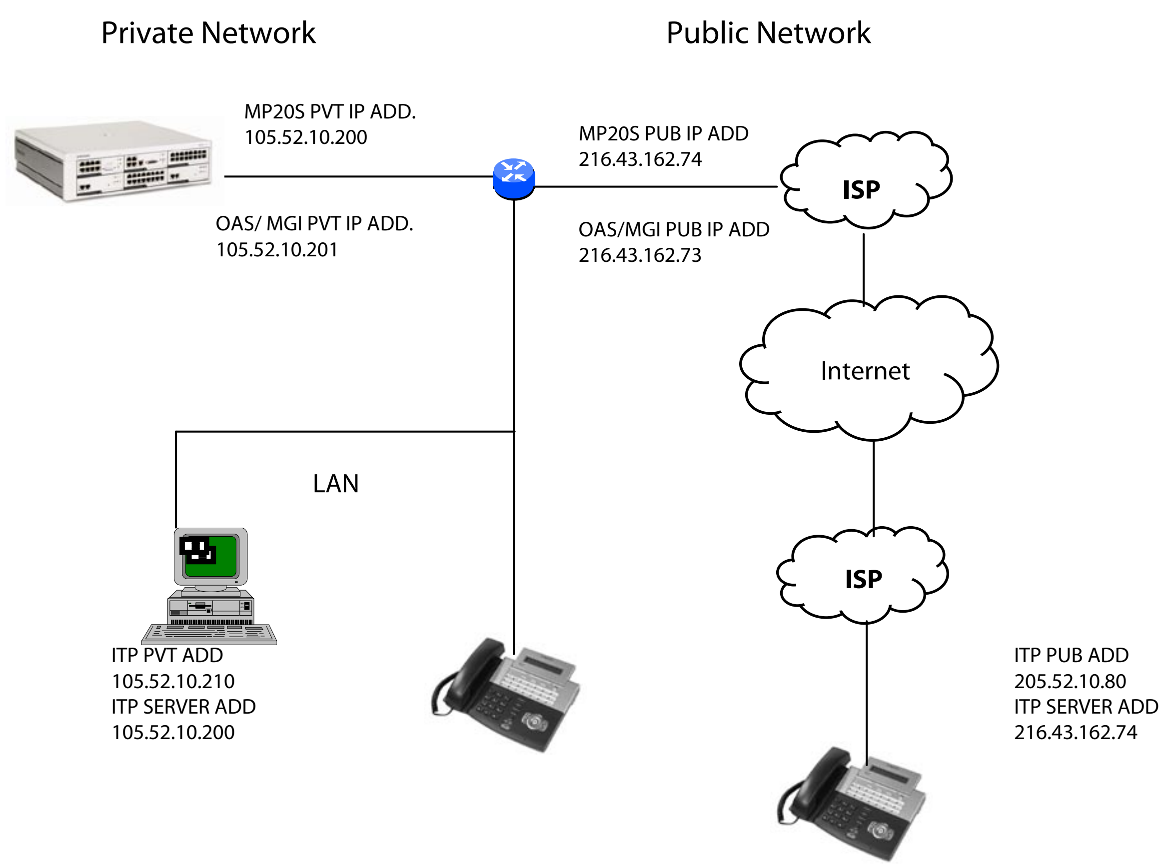
Figure 8. Remote ITP with Public Static IP Addresses



Reservations / Port Forwarding Router Addressing with Static IP Remote ITP Keyphone

Figure 9 shows the relationship between the MP20S, OAS and a remote ITP keyphone using a router to provide a separation between the Private and Public networks. With the MP20S on the private network this also permits ITP keyphones to be connected on the private network side and the public network side. Note in Figure 9 that the Server Address in the ITP keyphone is the “exposed” public MP20S Public IP address. The customer provided router in this case must “reserve” or forward the MP20S and MGI public IP addresses to the private MP20S and MGI IP addresses. Note that the remote ITP Server IP address is that of the public MCP.

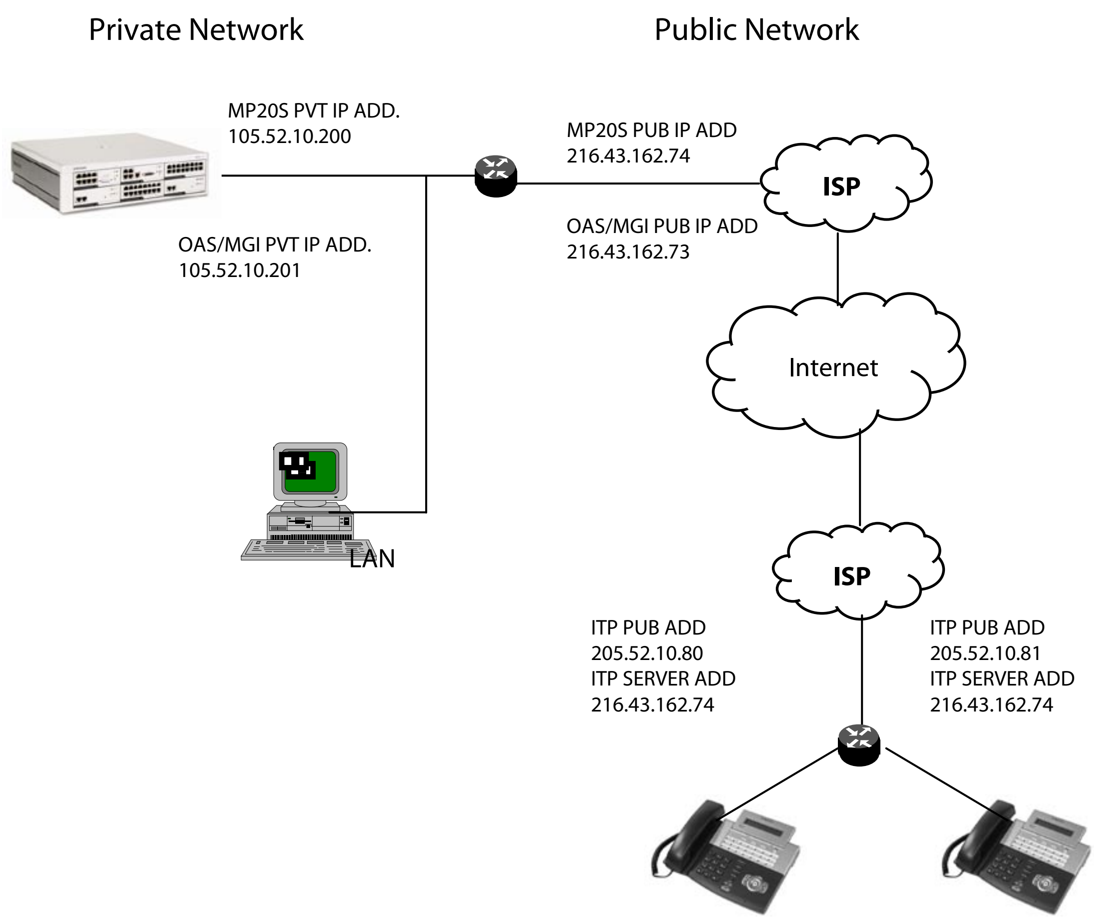
Figure 9. Private Network with Public Network ITP Keyphones



Remote ITP Keyphones with Reservations Router Addressing

Figure 10 shows Private with Public connectivity using router reservations or port forwarding at the OfficeServ 7200-S end the remote ITP endpoint(s). In all cases, on a public application, a public IP addresses must be obtained for the MP20S and the OAS card(s). A public IP address must also be obtained for each remote ITP keyphone that will be addressed when using router reservations and a publicly exposed remote end. Note in this example that both ends (MP20S, OAS and ITPs) are using port forwarding or reservation to expose the MP20S, MGI and ITP's to the public network and still allowing a private network that is not exposed.

Figure 10. Private Network with Public ITP Keyphones using Router Reservations



ITP Keyphone with Remote NAT Router Addressing

Figure 11a shows Private with Public connectivity using a router supporting reservations or port forwarding at the OfficeServ 7200-S end and a NAT supporting router at the remote ITP endpoint. In public network cases a public IP addresses must be obtained for the MP20S and the OAS card(s). A static public IP address OR a DHCP address can be obtained for the remote ITP keyphone(s) location(s). The ITP keyphones can also acquire an IP address from the NAT/DHCP router so that they may match the IP addressing scheme of the network where they are located. When using the routers' NAT feature multiple ITP keyphones can register to the MP20S.

Figure 11b also shows remote ITP keyphones using a NAT router. This approach is extremely cost effective due to the ITP and LAN PC's using the same public address. The ITP's use the public address to reach the OfficeServ 7200-S system and the LAN PC's use the public address to access the Internet

Figure 11a. Private Network with Remote ITP Keyphone NAT Router Addressing

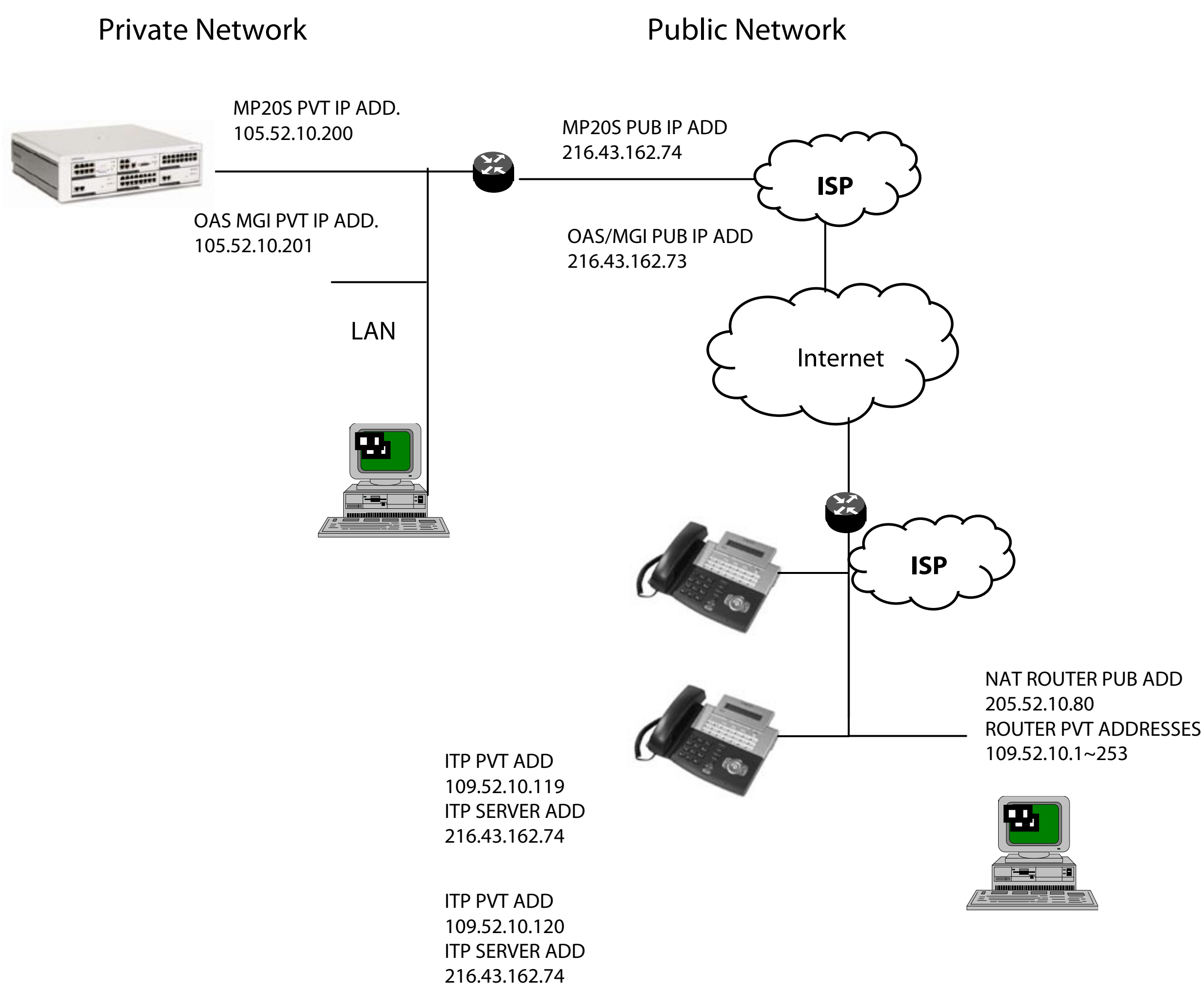
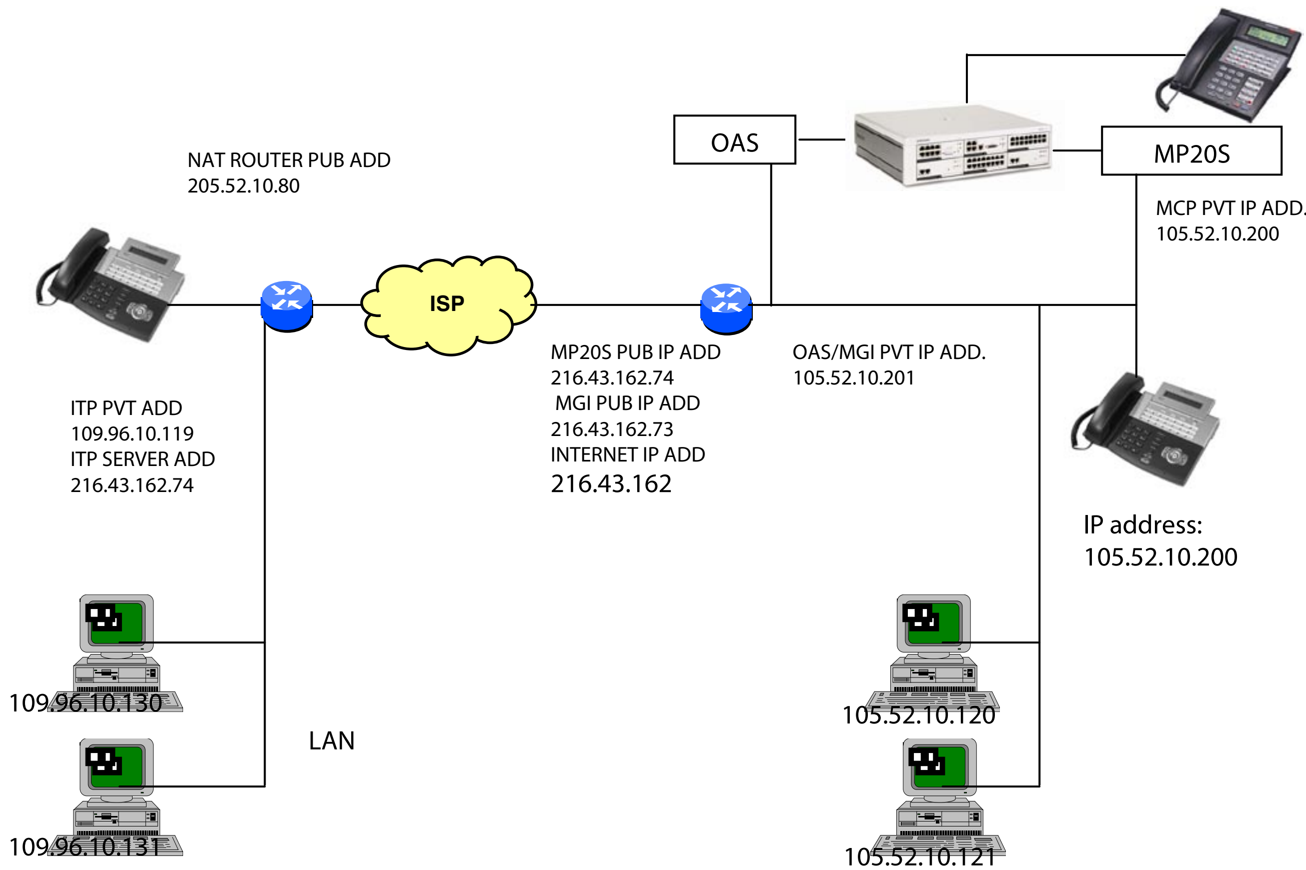


Figure 11b Private with Public Remote NAT Router Connections



Connecting to the Public Network

When connecting to the public network there are several considerations for IP addressing and connectivity.

1. Sufficient bandwidth to support ITP voice and data.
2. Minimum two static IP addresses at the OfficeServ 7200-S system side. One to expose the OAS card(s) to the public Internet and a second one to be used for MP20S communications.
3. A router with the ability to allow IP Reservations or IP Forwarding.
4. Use one of the 3 ways to connect a remote public ITP keyphone:
   - A static IP at remote ITP end OR
   - An ITP keyphone standalone using DHCP services from the ISP OR
   - An ITP keyphone behind a NAT router.

NOTE: If remote ITP has to traverse a firewall or NAT Router to communicate with the MP20S, then set:

- MMC 830 and MMC 831: IP type to “Public with Private” set up private and public IP addresses.
- MMC 840: For each remote ITP, set IP type to “Public with Firewall”.

Quality of Service

Quality of Service when deploying VoIP is always a consideration. In the IP network environment there are several tools to assist in providing a voice quality that can sometimes be comparable to circuit switched voice. When working via an IP network there are issues that are beyond the control of the communications system.

The main concerns for QoS are Lost Packets, Delay (fixed or variable) and Jitter. To use these terms it must be understood that the quality of voice via IP is based or measured by a Mean Opinion Score (MOS). MOS is a numerical measure of the quality of human speech when using VoIP. Subjective test (opinionated) provide a numerical score to determine voice quality. These scores are based on compression methods and provide a mathematical equivalent to speech quality where 5 is the best and 1 is the worst.

Table 3. MOS Guidelines

Compression	MOS Score	Byte/ms	Bit Rate (Kbps)
G.711	4.5	80/5	64
G.729	3.9	10/10	8
G.729A	3.65	10/10	8
G.723	3.8	24/30	5.5~6.5

Table 4. MOS Quality Assessments

Score	Listening Quality	Listening Effort
5	Excellent	Complete relaxation possible, no effort required
4	Good	Attention necessary, no appreciable effort required
3	Fair	Moderate effort required
2	Poor	Considerable effort required
1	Bad	No meaning understood with any feasible effort

Before applying QoS sufficient bandwidth must be available. A G.711 call over a 64Kbps link will result in a 20% loss of data and provide poor quality. IE. G.711 = 64 kbps payload +16 kbps header = 80 kbps. So, then an 80 kbps packet on a 64 kbps circuit results in 20% loss. Provisioning of bandwidth is an important factor in all VoIP applications.

Network congestion is another source for packet loss and delay. There are tools available in many router operating systems that provide for equal queuing to voice and data packets. Since data packets are smaller and sampled more often, a large data package can consume the time needed to send the voice packets and cause lost or delayed packets. The most common methods used by router manufactures are Low Latency Queuing, Fair Queuing, Weighted Fair Queuing, RSVP Queuing and Fragmentation. Insuring these options are available will assist in providing a good quality of service.

In all, QoS can be a complicated endeavor when transporting voice over an unmanaged public Internet environment. The OfficeServ 7200-S has many tools embedded to allow adjustments for VoIP QoS but they are not the catch-all solution to total network quality which should be governed by the network equipment.

TCP/UDP Ports

In the IP environment ports are used for specific connections. These ports need to be known to travese **routers**, firewalls and to provide prioritization in certain environments. The IP ports and port types utilized by the OfficeServ 7200-S are listed in Table 5.

System TCP/UDP Port numbers

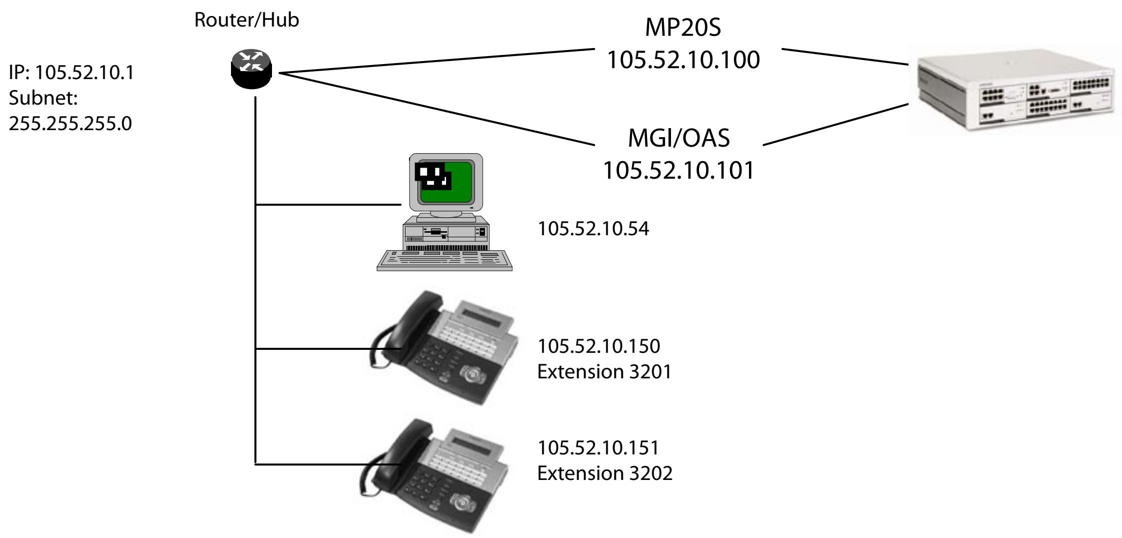
Table 5. OfficeServ 7200 TCP/RTP/UDP Port

Service Type	Port Number	Protocol	Remarks
VoIP Networking	6100	TCP	For connection setup
IP Phone interface (system side)	6000	TCP/UDP	For connection setup
H.323 Gateway	1719 1720 1024 ~ 4999	UDP TCP TCP	For connection with Gatekeeper For connection setup For maintaining TCP Connection
SIP Gateway	5060	UDP	For connection setup
OAS	30000~30031	RTP RTCP (also seen as UDP)	Even port: RTP for each MGI channel Odd port : RTCP for each MGI channel
IP Phone (phone side)	6000 9000, 9001	UDP RTP,RTCP	Signaling for MCP/MP20 Voice data for MGI or ITP
PCMMC/OSM	5000, 5003, 5200	TCP	PCMMC/OSM connection
CTI	5002	TCP	CTI Connection
Program Upload	5003	TCP	Program upload to MMC+/Smart Media
Installation Tool	5090	TCP	Connection port for the Installation Tool system programming application.
SMDR Print	5100	TCP	SMDR printout to IP connection

Service Type	Port Number	Protocol	Remarks
UCD Print	5101	TCP	UCD printout to IP connection
Traffic Report	5102	TCP	Traffic Report to IP connection
Alarm Report	5103	TCP	System Alarm Report to IP connection
UCD View	5104	TCP	UCD View printout to IP connection
Periodic UCD	5105	TCP	Periodic UCD printout to IP connection
Hotel Report	5106	TCP	Hotel Report to IP connection

ITP Application Programming Examples

In the following example shows the private corporate LAN application on the enterprise network. The network is considered "private" because all of the IP addresses are on the same network. In the following example the IP addresses are "static" and were provided by the IT person that manages the IP network.



Programming this application on the OfficeServ 7200-S system is as follows:

1. MMC 830 ETHERNET PARAMETERS

- SYSTEM IP ADDR: 105.52.10.100 (MP20S IP address)
- SYSTEM GATEWAY: 105.52.10.1
- SYSTEM NET MASK: 255.255.255.0
- SYSTEM RESTART: Yes
- SYSTEM IP TYPE: PRIVATE (default)

2. MMC 831 MGI PARAMETERS

- IP ADDRESS: 105.52.10.101
- GATEWAY: 105.52.10.1
- SUB MASK: 255.255.255.0
- PUBLIC IP: Not Applicable (default 1.1.1.1)

3. MMC 840: IP PHONE SETTINGS FOR EACH ITP ON THE PRIVATE NETWORK

- IP TYPE: Private

4. ITP Keysets

The following needs to be entered into each of the ITP keyphones accordingly. For detailed instructions of configuring the ITP keyphone, see the user's guide.

Network – Mode – Manual IP
Network – IP – 105.52.10.150 or 151 (accordingly)
Network – Netmask – 255.255.255.0
Network – Input Def. Gateway – 105.52.10.1
Server – Server IP – 105.52.10.100
Server – Input ID – 3201 or 3202 (accordingly)
Server – Password – 1234 (default)
Exit

MGI Programming MMCs

There are 14 MMCs directly related to the use and operation of the OAS card.

MMC 615 MGI GROUP

Assigns permissions of use for individual MGI ports. Separates range of ports to be allowed for MGI Services

MMC 616 MGI USER

Assigns permissions of use for individual MGI ports. Separates individual ports to be allowed for MGI use. Allow specific stations or trunks to only use the specified MGI port.

MMC 830 ETHERNET PARAMETERS

Assigns MP20S system IP address, subnet, gateway and MP20S Public address (if required).

MMC 831 MGI PARAMETERS

Assigns MGI IP address(es), subnet, gateway and Public address (if required).

MMC 832 VoIP OUT DIGITS

Assigns VoIP access codes, access code length, the number of digits to delete or insert, remote end trunk access code and IP selection tables

MMC 833 VoIP IP ADDRESS

Allows assignment of IP addresses in specific tables to route calls to remote destinations

MMC 834 H.323 OPTIONS

This MMC programs incoming call destination type, CLIP table, CID type, auto codec negotiation, call set up and signaling types and other parameters associated with VoIP signaling when using as a trunking gateway.

MMC 835 MGI DSP OPTIONS

Assigns the individual parameters associated with the VoIP DSP operation. Codec type, filtering, input gain, voice volume and RTP (real time transport protocol) parameters.

MMC 836 H.323 GATEKEEPER OPTIONS

This MMC allow the setup parameter to allow the MGI to operate with a network supported by VoIP Gatekeeper(s).

MMC 837 SIP OPTIONS

This MMC programs the SIP protocol parameters.

MMC 838 PRIVATE IP

This MMC programs the IP addresses for devices that are using the H.323 standard within the network.

MMC 840 IP SET INFO

This MMC programs and stores the station number, user ID and password for ITP keyphones. This MMC also shows ITP MAC and IP addresses and is where the ITP keyphones are registered with the OfficeServ 7200.

MMC 841 SYSTEM IP OPTION

This MMC allow new ITP registrations to be disabled and also allows UCD and SMDR to be sent to the CTI link.

MMC 842 SIP STN INFO

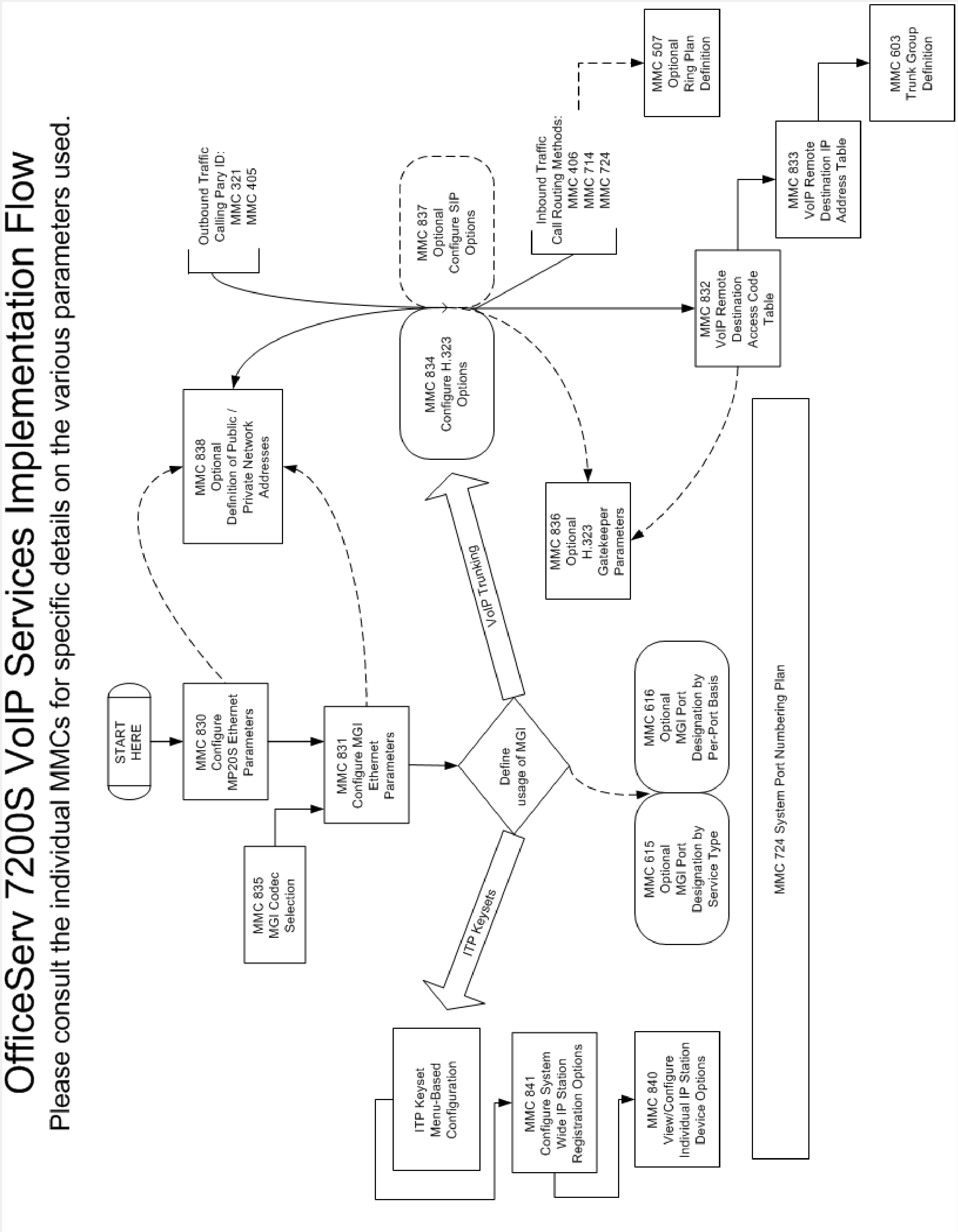
This MMC provides various proprietary Samsung SIP integration options of non-Samsung SIP terminals. The options set in this MMC are system-wide.

MMC 843 MPS OPTION

Location of OAS card and IP address of MPS. Port 40,000

MMC 858 OAS CARD SRV

Physical locations of MOBEX/MGI configured up to 16MGI and MOBEX port.


OfficeServ 7200S VoIP Services Implementation Flow
Please consult the individual MMCs for specific details on the various parameters used.
START HERE
MMC 830 Configure MP20S Ethernet Parameters
MMC 831 Configure MGI Ethernet Parameters
MMC 835 MGI Codec Selection
MMC 838 Optional Definition of Public / Private Network Addresses
Define usage of MGI
VoIP Trunking
ITP Keysets
MMC 834 Configure H.323 Options
MMC 837 Optional Configure SIP Options
Outbound Traffic Calling Pary ID: MMC 321 MMC 405
Inbound Traffic Call Routing Methods: MMC 406 MMC 714 MMC 724
MMC 507 Optional Ring Plan Definition
MMC 836 Optional H.323 Gatekeeper Parameters
MMC 832 VoIP Remote Destination Access Code Table
MMC 833 VoIP Remote Destination IP Address Table
MMC 603 Trunk Group Definition
MMC 616 Optional MGI Port Designation by Per-Port Basis
MMC 615 Optional MGI Port Designation by Service Type
MMC 724 System Port Numbering Plan
ITP Keyset Menu-Based Configuration
MMC 841 Configure System Wide IP Station Registration Options
MMC 840 View/Configure Individual IP Station Device Options

2.11 NETWORKING OVER PRI

The following procedures are intended to show how to set up networking between systems only. It is assumed that all the systems to be networked together have already been set up as stand alone systems with LCR and inbound trunks already set up. If these items have not already been set up then some networking features such as Tandem Dialing and DID pass through cannot be made to work.

The first thing that must be done if networking is to work is to determine the numbering scheme for all ports in the total combined network including room for expansion at each of the networked systems. If the total number of stations is going to be less than 150 then a three-digit station numbering scheme can be easily used (extension numbers 201 to 349). However if more than 150 stations are required then it is strongly recommended that a four-digit numbering scheme be used. Doing this will allow all the systems to have the same feature access codes and will reduce the amount of system programming.

Hardware Setup

Connection Type

The type of circuit used to connect two systems together is a "point to point" T1 configured with ESF and B8ZS signaling. The TEPRI card will convert this T1 into a proprietary type of PRI signaling. If Drop and Insert type CSU/DSUs are to be used on the circuit so that the span can be used for data as well then the system must have control of channel 24 on the span and voice channels from channel 1 down. For example, if a customer wants to use 8 channels for data and 15 channels for voice, then the system has to have control of channels 1 through 15 for voice and channel 24 for signaling.

TEPRI Card DIP Switch Setup

On the DIP switch bank on the TEPRI card switches 1, 2 and 8 must be set to the ON position. If this card is to be the PRI D channel controller card for the span then DIP switch number 4 should also be in the ON position. There can only be one D channel controller on each span.

Software Setup

1. Each switch must be given a unique multi-digit ID number that does not conflict with the North American Dialing Plan. e.g. one switch is ID 001, a second is ID 002, a third is 003, etc. The digits star (*) and pound (#) cannot be used in switch ID numbers, This information is entered in MMC 820.

2. Determine the extension number range(s) for each switch:

   e.g. Switch 001 is 2001 → 2199
   Switch 002 is 2200 → 2299
   Switch 003 is 2300 → 2399

3. Determine the group number ranges for each switch:

   e.g. 001 is 5000 → 5049
   002 is 5100 → 5149
   003 is 5200 → 5249

4. Determine the network LCR access codes from the information in items 2 and 3 above. These are the common leading digits for the station and group numbers in the other switches.

   e.g. for switch 001 this would be 22 (so 22XX extensions can be called), 23 (so 23XX extensions can be called), These are for stations in the other systems, 51 (so 51XX groups can be called) and 52 (so 52XX groups can be called), These are for station groups in other systems.

   This information is entered in MMC 724 in the network LCR area (NTWK LCR NUMPLAN).

5. Combine the switch ID and the LCR code to create the entries for MMC 824.

   e.g. for switch 001 those would be: XXXYY Where
   00222 XXX = Switch ID
   00251 Y = LCR Access Code
   00323
   00352

   As we are using 4 digit extension and group numbers the size field for each entry will be 4.

   As we are using 3 digits switch IDs and 4 digit station and group numbers the max digits field for all entries will be 3 + 4 = 7

If the networked systems are to use a centralized voicemail system then the switch that has the VMAA will have all the MB entry = Y, all others will be = N.

In our example switch 001 will host the centralized voice mail so the MMC 824 entries for switch 001 will now look like this:

```
[01:22  →  00222]
[SZ:2 MAX:07 MB:Y]

[02:23  →  00323]
[SZ:2 MAX:07 MB:Y]
```

And so on for all four entries.

The trunk side of the network intercom dial plan must now be set up to allow LCR to send intercom calls across to the other switch.

6. Put the trunks for each network link into their own trunk group.

   e.g. in switch 001 put the trunks to switch 002 in group 802 and the trunks to switch 003 in 803.

7. Using MMC 821 change each network link from “Normal” to “Q-SIG”.

8. Set up LCR routes in MMC 712 to allow LCR to access the network links.

   e.g. for switch 001 the route could be route 2 to access switch 002 set up as 02:1 C:1 G:802 M:_ _ _
   Route 3 to access switch 003 set up as 03:1 C:1 G:803 M:_ _ _

9. Set up the LCR digit tables in MMC 710 to allow access to the network link routes.

   e.g. for switch 001 the entries would be:

Digits	Length	Route
002	7	2
003	7	3

   In order to receive incoming calls over the network the DID table must be set up.

10. Make entries in the DID digit table MMC 714 for the digits to be received over the network links.

   e.g. for switch 001:

Digits	Destination	Delete	Reason
20**	RP1~RP6 = B	0	Intercom Calls to
21**	RP1~RP6 = B	0	Switch 001
22**	RP1~RP6 = 22	2	ICM to 002 from 003
23**	RP1~RP6 = 23	2	ICM to 003 from 002
50**	RP1~RP6 = B	0	Intercom to 001 groups
51**	RP1~RP6 = 51	2	Calls to 002 groups
52**	RP1~RP6 = 52	2	Calls to 003 groups
561	RP1~RP6 = 9	1	tandem local calls

When the 10 steps described above have been completed intercom networking is completed.

Centralized Voice Mail Setup

This setup takes place at each of the remote locations and allows these locations to use the SVM system in the central location. When messages are left at the central location the VMMSG key at the remote location will illuminate to indicate a message. The “new message” counter will not be shown in the keyset displays at the remote location. When the message is returned to the SVM by pressing the VMMSG all navigation must be done by following the audio prompts and dialing digits as the interactive soft keys will not be shown at the remote location.

1. In MMC 825 set the “USE REMOTE VM” option to YES
2. In MMC 825 program the station group for the SVM in the main location under “REMOTE VM NUMBER” e.g. 50-49
3. Confirm that at the main location that the entries in MMC 824 that require voice mail boxes have the MB: option set to Y so that when a VM download is performed these mailboxes and extension id’s will be automatically created.

Centralized Operator Setup

This setup allows all dial ”0” calls in the remote system to call the operator in the main system. This feature only works with dial “0” calls. Recalls etc. will ring the designated local operator.

1. In MMC 724 under NTWK LCR NUMPLAN make an entry “0”, for example:

```
[NTWK LCR NUMPLAN]
[LCR-03:NONE→0    ]
```

2. In MMC 824 under for the “0” entry fill out the display as 0010,1,04,N for example:

```
[03:0    →0010    ]
[SZ:1 MAX:04 MB:N]
```

3. In MMC 710 make an entry for digits “0010”, length 04, and the route to access the main location. For example:

```
[LCR DIGIT (0002)]
[DIGIT:0010      ]

[LCR DIGIT(0002)]
[LENGTH:04 RT:02]
```

DID Pass Through

This will allow DID calls that come in on circuits on the main location to ring at the remote locations.

1. In MMC 724 under NTWK LCR NUMPLAN make an entry for an unused station number block for example 30:

```
[NTWK LCR NUMPLAN]
[LCR-05:NONE→30  ]
```

2. In MMC 824 for entry “30” fill out the display to route the calls to switch 002 indicating the total number of DID digits plus the 3 digit access code as the MAX entry as follows:

```
[05:30  →002     ]
[SZ:2 MAX:07 MB:N]
```

3. In MMC 714 tell the incoming DID digits to “ring” 30 for all ring plans

4. Translate the digits on the receiving side in MMC 714 as normal

NOTE: Repeat steps 1 through 4 for each remote switch using a different NTWK access code for each switch.

Tandem Trunking

If the networked switches are in different area codes, e.g.

Switch 001 is 561
Switch 002 is 954
Switch 003 is 305

then calls from one switch can be routed over the network link and sent over as local calls. For example if a user in Switch 001 wants to call a number in the 305 area code the call could be routed over the network link to switch 003 and sent out as a local call. This is called Tandem Trunking.

To achieve this, LCR entries must be created to route the calls as described by modifying the digits and routing the call across the network link.

1. Create LCR entries for each networked Switch area codes.

   E.g. for Switch 001 these would be:

Digits	LengthRT	
1954	11	05
1305	11	06

2. Create the LCR routes with the digits modified to reflect 7 digit dialing in 954 area code and 10 digit dialing in 305 area code.

   Eg. for Switch 001 these would be:

Route	Class	Group	Modify
05:1	1	802	002
06:1	1	803	003

3. Create the modifying digits entries as described in step 2.

   Eg. for Switch 001 these would be:

Entry	Del	I	A
002	01	0029	_ _ _
003	01	0039	_ _ _

These modify digit entries convert the digits dialed by the user into a digit string that will let the destination switch route the call through the DID translation table (MMC 714) to the appropriate local trunk group. This is the group already set up in LCR before networking was applied to allow local calls to be sent out of that switch. For example in switch 001 if local calls go out over trunk group 800 there would be an entry in MMC 714 for digits 561 (switch 001's home area code) to "ring" 800 for each of the ring plans.

Home Page

NETWORKING DATA COLLECTION FORM

Switch ID: (MMC 820)		Area Code:	
Extension Range: (MMC 724)		to	
		to	
		to	
Station Group Range: (MMC 724)		to	
Central Operator Host:	☐ Yes ☐ No		
Central VMAA Host:	☐ Yes ☐ No		
LCR Access Code: (MMC 724)			

Network LCR Access Code
(MMC 724)

Network LCR Table Entries
(MMC 824)

Entry	Size	Max	MB

Home Page

Network Trunk Group

(MMC 603)

Switch	Group

Networking Link Access

(MMC 710)

Digits	Length	Route

Network Routes

(MMC 712)

Switch ID	Route	Class	Group	Modify

MODIFY DIGITS

(MMC 713)

Entry	DEL	I	A

DID TABLE (MMC 714)

Digits	RP1	RP2	RP3	RP4	RP5	RP6	Delete

NETWORKING PROGRAMMING EXAMPLE

A company has three locations, its headquarters is in 561 area code, a location in 954 area code and a location in 305 area code. 561 and 305 areas are ten digit local dialing and 954 is 7 digit local dialing. There are 150 stations in the headquarters and this system will also host the SVMi-8 for the company. The 954 location has 50 sets as does the 305 location. All calls from area code to area code are long distance so calls from any of the locations to the area code of another location must be re routed across the network links and sent out as local calls. The data sheets attached show how the system would be programmed.

NETWORKING DATA COLLECTION FORM-EXAMPLE 1

Switch ID: 001 Area Code: 561
(MMC 820)

Extension Range:
(MMC 724)

2001	to	2099
2100	to	2199
	to	

Station Group Range: 5000 to 5049
(MMC 724)

Central Operator Host: ☒ Yes ☐ No

Central VMAA Host: ☒ Yes ☐ No

LCR Access Code: 9
(MMC 724)

Network LCR Access Code
(MMC 724)

22	23	
52	53	

Network LCR Table Entries
(MMC 824)

Entry	Size	Max	MB
00222	4	7	Y
00252	4	7	Y
00323	4	7	Y
00353	4	7	Y

Network Trunk Group

(MMC 603)

Switch	Group
002	802
003	803

Networking Link Access

(MMC 710)

Digits	Length	Route
002	7	3
003	7	4
1305	11	5
1954	11	6

Network Routes

(MMC 712)

Switch ID	Route	Class	Group	Modify
002	03:1	1	802	
003	04:1	1	803	
TANDEM	05:1	1	802	002
TANDEM	06:1	1	803	003

MODIFY DIGITS

(MMC 713)

Entry	DEL	I	A
002	01	002	
003	01	003	

DID TABLE (MMC 714)

Digits	RP1	RP2	RP3	RP4	RP5	RP6	Delete
20✱✱	B	B	B	B	B	B	0
21✱✱	B	B	B	B	B	B	0
22✱✱	22	22	22	22	22	22	2
23✱✱	23	23	23	23	23	23	2
50✱✱	B	B	B	B	B	B	0
52✱✱	52	52	52	52	52	52	2
53✱✱	52	52	52	52	52	52	2
561	800	800	800	800	800	800	0

NETWORKING DATA COLLECTION FORM-EXAMPLE 2

Switch ID: (MMC 820)	002	Area Code:	954
Extension Range: (MMC 724)	2201	to	2299
		to	
		to	
Station Group Range: (MMC 724)	5200	to	5249

Central Operator Host: ☐ Yes ☒ No

Central VMAA Host: ☐ Yes ☒ No

LCR Access Code: (MMC 724) 9

Network LCR Access Code (MMC 724)

20	21	23
50	53	

Network LCR Table Entries (MMC 824)

Entry	Size	Max	MB
00120	4	7	N
00121	4	7	N
00150	4	7	N
00323	4	7	N
00353	4	7	N

Network Trunk Group

(MMC 603)

Switch	Group
001	801
003	803

Networking Link Access

(MMC 710)

Digits	Length	Route
001	7	2
003	7	3
1305	11	4
1561	11	5

Network Routes

(MMC 712)

Switch ID	Route	Class	Group	Modify
001	02:1	1	801	
003	03:1	1	803	
TANDEM	04:1	1	803	002
TANDEM	05:1	1	801	003

MODIFY DIGITS

(MMC 713)

Entry	DEL	I	A
002	01	003	
003	01	001	

DID TABLE (MMC 714)

Digits	RP1	RP2	RP3	RP4	RP5	RP6	Delete
20**	20	20	20	20	20	20	2
21**	21	21	21	21	21	21	2
22**	B	B	B	B	B	B	0
23**	23	23	23	23	23	23	2
50**	50	50	50	50	50	50	2
52**	B	B	B	B	B	B	0
53**	53	53	53	53	53	53	2
954	800	800	800	800	800	800	3

Home Page

NETWORKING DATA COLLECTION FORM-EXAMPLE 3

Switch ID: (MMC 820) 003

Area Code: 305

Extension Range: (MMC 724)

2301	to	2399
	to	
	to	

Station Group Range: (MMC 724) 5300 to 5349

Central Operator Host: ☐ Yes ☒ No

Central VMAA Host: ☐ Yes ☒ No

LCR Access Code: (MMC 724) 9

Network LCR Access Code (MMC 724)

20	21	22
50	52	

Network LCR Table Entries (MMC 824)

Entry	Size	Max	MB
00120	4	7	N
00121	4	7	N
00150	4	7	N
00222	4	7	N
00252	4	7	N

Network Trunk Group

(MMC 603)

Switch	Group
001	801
002	802

Networking Link Access

(MMC 710)

Digits	Length	Route
001	7	2
002	7	3
1954	11	4
1561	11	5

Network Routes

(MMC 712)

Switch ID	Route	Class	Group	Modify
001	02:1	1	801	
002	03:1	1	802	
TANDEM	04:1	1	801	001
TANDEM	05:1	1	802	002

MODIFY DIGITS

(MMC 713)

Entry	DEL	I	A
001	01	002	
002	01	001	

DID TABLE (MMC 714)

Digits	RP1	RP2	RP3	RP4	RP5	RP6	Delete
20**	20	20	20	20	20	20	2
21**	21	21	21	21	21	21	2
22**	22	22	22	22	22	22	2
23**	B	B	B	B	B	B	0
50**	50	50	50	50	50	50	2
52**	52	52	52	52	52	52	2
53**	B	B	B	B	B	B	0
305	800	800	800	800	800	800	0

2.12 NETWORKING OVER IP

The following procedures are intended to show how to set up IP networking between systems only. This method will allow multiple OfficeServ 7200-S systems to be networked together over an IP network. It is assumed that all the systems to be networked together have already been set up as stand alone systems with LCR and inbound trunks already set up. If these items have not already been set up then some networking features such as Tandem Dialing and DID pass through will not work.

This document assumes that the user is knowledgeable in TCP/IP concepts and IP network configuration. It is also assumed that a functioning IP network exists between nodes. The procedures involved in building or setting up this IP network connectivity is beyond the scope of this document. See your data network administrator for details.

Hardware

Each node in the network should have at least one OAS installed. In each system, the OAS cards and the MP20S cards must be programmed with a unique IP addresses on the network. The nodes can reside on the same IP network or different IP networks as long as each MP20S and OAS in each system can communicate with the MP20S' and OAS' in all the other nodes and with each other over an IP network. Ping tests are the best way to verify this. From a computer attached to the same network segment that one of the nodes is on, ping the IP addresses of the other node's MCP/MP20 and MGI's as well as the IP addresses of the OAS and MP20S on the local node.

If the pings are successful, then it verifies that the communication path is good between devices. However, this does not guarantee other performance related requirements (ie. sufficient bandwidth to handle voice traffic, delay, jitter control, etc…). Some routers and firewalls may have disabled the ability to reply to a ping. Verify this with the data administrator.

The first thing that must be done is to determine the numbering scheme for all ports in the total combined network including room for expansion at each of the networked systems. If the total number of stations is going to be less than 150 then a three-digit station numbering scheme can be easily used (extension numbers 201 to 349). However if more than 150 stations are required then it is strongly recommended that a four-digit numbering scheme be used. Doing this will allow all the systems to have the same feature access codes and will reduce the amount of system programming.

Software Setup

The following MMC's must be programmed for each node in the network:

1. The following MMC's related to IP and MGI set up must be programmed **MMC 830. System IP Address for MP20S.**

   Configure the MP20S with the relevant IP information:

   - Assign IP Address, Default Gateway, and Subnet Mask, <u>then restart the system for the changes to take effect.</u>
   - Select IP type (private, public, or private with public) and assign public IP if applicable.

   MMC 831. MGI PARAMETERS

   Configure the OAS card with the relevant IP information:

   - Assign IP Address, Default Gateway, Subnet Mask for each OAS card, then restart the OAS card/s for the changes to take effect. Card can be reset using the reset option in MMC 831.
   - Select IP type (private, public, or private with public) and assign public IP if applicable.

   MMC 615. MGI Use Group

   Add the IP trunk members (MGI Channels) for the VoIP Networking function.

   1. USER field should be set to "VoIP Networking". Select the IP trunk members for this function. Each IP trunk represents an MGI channel.

   MMC 616. MGI Fixed User

   If needed you can use this MMC to permanently dedicate MGI channels for a specific function. These settings will over-ride the functional assignments in MMC 615. For example, you can assign channels 3801 and 3802 to always be used only for the "VoIP Networking" function.

2. Each switch must be given a unique multi-digit ID number that does not conflict with the North American Dialing Plan. e.g. one switch is ID 001, a second is ID 002, a third is 003, etc. The digits star (*) and pound (#) cannot be used in switch ID numbers, This information is entered in MMC 820.

MMC 820. System Link ID

- Link ID : System node ID to distinguish systems in the network. It can not be duplicated. Assign a unique node ID for each system in the network including one for the node you are on ("SELF").
- Signal G/W: For VoIP networking, all IP addresses for the connected systems should be assigned. The IP address for the current node should also be assigned for SELF.
- IP Type : IP type connected through VoIP networking should be assigned.
- Assign the id for all other nodes
- When a call is placed on the networking trunk, the system usually regards the number as node number + station number. If the node number is identified with the self node number, the system removes the node number and places the call according to MMC 714. If not, forwards the call to another system according to MMC 710 (Tandem Call). If there is no identified number, the system regards it as the station number and places the call according to MMC 714.

3. Determine the extension number range(s) for each switch:

   e.g. Switch 001 is 2001 → 2199
   Switch 002 is 2200 → 2299
   Switch 003 is 2300 → 2399

   Determine the group number ranges for each switch:

   e.g. 001 is 5000 → 5049
   002 is 5100 → 5149
   003 is 5200 → 5249

4. Determine the network LCR access codes from the information in items 2 and 3 above. These are the common leading digits for the station and group numbers in the other switches.

   e.g. for switch 001 this would be 22 (so 22XX extensions can be called), 23 (so 23XX extensions can be called), These are for stations in the other systems, 51 (so 51XX groups can be called) and 52 (so 52XX groups can be called), These are for station groups in other systems.

   This information is entered in MMC 724 in the network LCR area (NTWK LCR NUMPLAN).

MMC 724. Numbering Plan

- STN DIAL NO: A non-conflicting range of station numbers should be assigned across all the nodes in the network.
- STNG DIAL NUMBER: A non-conflicting range of station group numbers should be assigned across all the nodes in the network.
- NTWK LCR DIAL NO: Numbers used to call a station in another system are assigned. This number is converted to node number + station number in MMC 824, and placed in the destination.
- VOIP NET DIAL NO: For VoIP networking, these are virtual trunks for signaling. They should be assigned to trunk groups in MMC 603.
- MGI DIAL NO: These are the channel numbers of the OAS card for voice. They do not need to be set for networking, but should be assigned in the VoIP network group in MMC 615.

5. **MMC 824. Networking LCR Translation**

For this program, networking dial number in MMC 724 should be assigned. When index 01 is assigned “22” in networking, dial number in MMC 724, ❶ is displayed.

Enter like ❷ for the conversion to node number + station number.

❶
```
01:22 →
SZ:0 MAX:00 MB:N
```

❷
```
01:22 →09122
SZ:4 MAX:07 MB:N
```

In ❷, the meaning of each field is as follows.

- 01: Index for NTWK LCR DIAL NO in MMC 724.
- 22: NTWK LCR DIAL NO in MMC 724.
- 09122: Node number(091) + station number(22). The system whose node id is "091" has the station numbers 2200 ~ 2299. Added node number should be assigned to “DIGIT” in MMC 710.
- SZ:4 means the user dials 4 digits when he/she calls node number "091" + station number starting "22".
- MAX:07: Total number of digits in node number + station number. This is very important in "ENBLOCK" dial. When the system receives 7 digits, it does not wait for more digits and makes the call using LCR.
- MB:N: When using Centralized Voice Mail using SVMi-4E/8E/16E/20E cards, other systems need to manually create mail boxes for each system. If this option is "Y", the system creates up to 100 mail boxes for the phone numbers which starts with the numbers assigned in MMC 724, with a length specified by the value of “SZ”. That is, it makes 100 mail boxes of 2200~2299 automatically.

Combine the switch ID and the LCR code to create the entries for MMC 824.
e.g. for switch 001 those would be: XXXYY Where

00222	XXX = Switch ID
00251	Y = LCR Access Code
00323	
00352	

Since we are using 4 digit extension and group numbers, the size field for each entry will be 4.

Since we are using 3 digit switch IDs and 4 digit extensions and group numbers, the max digits field for all entries will be 3 + 4 = 7.

If the networked systems are to use a centralized voicemail system then the switch that has the VMAA will have all the MB entry = Y, all others will be = N.

In our example switch 001 will host the centralized voice mail so the MMC 824 entries for switch 001 will now look like this:

[01:22 → 00222]
[SZ:2 MAX:07 MB:Y]

[02:23 → 00323]
[SZ:2 MAX:07 MB:Y]

And so on for all four entries.

The trunk side of the network intercom dial plan must now be set up to allow LCR to send intercom calls across to the other switch.

6. **MMC 210:** Tenant On and Off**:** LCR ENABLE: Set to ON to enable Least Cost Routing.
7. **MMC 310:** LCR CLASS: Assigns the LCR class. If there is only one path connected to another system, it is set to 1(default).
8. **MMC 603:** TRUNK GROUPS: For VoIP networking, all VoIP signaling trunks are assigned to one group. VoIP signaling trunks are the virtual trunks specified in “VOIP NET NUM PLAN” in MMC 724.
9. Put the trunks for each network link into their own trunk group.

   e.g. in switch 001 put the trunks to switch 002 in group 802 and the trunks to switch 003 in 803.
10. Set up LCR routes in MMC 712 to allow LCR to access the network links.

MMC 712. LCR Route Table

Route table to select trunks for LCR.

The meaning of each field is as follows.

- (01~32) : Index numbers of route tables.
- (1~4): Time table number assigned in MMC 711.
- C: LCR class assigned in MMC 310.
- G : Trunk group number.
- M : The modified table number in MMC 713 that is in need of modification.
- Assuming, the user usually uses VoIP networking, but uses PRI networking only when VoIP signaling trunks are not available, LCR class in MMC 310 should be 2 or higher, and "C:1 G:801", "C:2 G:802". Here, 801 is a trunk group consisting of VoIP signaling trunks and 802 is trunk group consisting of networking PRI trunks.

e.g. for switch 001 the route could be route 2 to access switch 002 set up as 02:1 C:1 G:802 M:_ _ _
Route 3 to access switch 003 set up as 03:1 C:1 G:803 M:_ _ _

MMC 713. LCR Modify Table

Assigned when conversion is needed for outgoing call.

The meaning of each field is as follows.

- (001~200): Index numbers of translation tables.
- NOF DEL DGT: Number of digits which will be deleted. Only beginning of digits can be deleted.
- I: Digits inserted in front of the outgoing digits, if needed.
- A: Digits appended behind the outgoing digits, if needed.

11. Set up the LCR digit tables in MMC 710 to allow access to the network link routes.

MMC 710. LCR Digit Table

Assigns route table according to the numbers entered.

The meaning of each field is as follows.

- (0001~2000): Index numbers.
- DIGIT: Beginning of digits for outgoing call. Because a networking call should have a node number, it can be the starting point. It should be included in the numbers assigned in MMC 824.
- LENGTH: The system does not send digits to the trunk, until “LENGTH” of digits are collected.

RT: Route table number assigned in MMC 712.
e.g. for switch 001 the entries would be:

Digits	Length	Route
002	7	2
003	7	3

In order to receive incoming calls over the network the DID table must be set up.

MMC 711. LCR Time Table: ***IF YOU ARE NOT USING THIS TABLE, YOU MUST STILL ENTER 0000 for HHMM and set LCRT=1.***

Assigns 4 time zones for each date of the week for LCR.

The meaning of each field is as follows.

- SUN~SAT: Sunday to Saturday.
- A~D: 4 time zones.
- HHMM: Time in 24 hour mode.
- LCRT: Time table number used in MMC 712.

12. Make entries in the DID digit table MMC 714 for the digits to be received over the network links.

MMC 714. DID Destination

Assigns an incoming DID call to a specific ring plan destination. The destination includes a station, station group, trunk, trunk group, LCR number, and networking dial number.

The meaning of each field is as follows.

- (001~999): Index numbers.
- DGT: Numbers which will be received. '*' means any digit between '0'~'9'.
- MOH SOURCE: Assigns MOH which will be used according to the received number. If None, it works according to MMC 409.
- 1~6: Assigns a specified station or group which will receive the call according to the ring mode 1~6. B means a station, station group, trunk or trunk group which is the same number with the received one.
- CW: For Call Waiting. This is not used for networking.
- DELETE: The number of digits to delete.
- NAME: The name of the DID table.

e.g. for switch 001:

Digits	Destination	Delete	Reason
20**	RP1~RP6 = B	0	Intercom Calls to
21**	RP1~RP6 = B	0	Switch 001
22**	RP1~RP6 = 22	2	ICM to 002 from 003
23**	RP1~RP6 = 23	2	ICM to 003 from 002
50**	RP1~RP6 = B	0	Intercom to 001 groups
51**	RP1~RP6 = 51	2	Calls to 002 groups
52**	RP1~RP6 = 52	2	Calls to 003 groups
561	RP1~RP6 = 9	1	tandem local calls

When the 10 steps described above have been completed intercom networking is completed.

Centralized Voice Mail Setup

This setup takes place at each of the remote locations and allows these locations to use the SVM system in the central location. When messages are left at the central location the VMMSG key at the remote location will illuminate to indicate a message. The "new message" counter will not be shown in the keyset displays at the remote location, unless all nodes are running 2.3x software or higher. When the message is returned to the SVM by pressing the VMMSG all navigation must be done by following the audio prompts and dialing digits as the interactive soft keys will not be shown at the remote location, unless all nodes are running 2.3x software or higher.

- In MMC 825 set the "USE REMOTE VM" option to YES
- In MMC 825 program the station group for the SVM in the main location under "REMOTE VM NUMBER" e.g. 50-49
- Confirm that at the main location that the entries in MMC 824 that require voice mail boxes have the MB: option set to Y so that when a VM download is performed these mailboxes and extension id's will be automatically created.
- Make sure you have created mailboxes for the remote node stations on the SVMi using MMC 740 (or SVMi Mailbox Administration). Create these mailboxes on the system where the SVMi resides.

Centralized Operator Setup

This setup allows all dial "0" calls in the remote system to call the operator in the main system. This feature only works with dial "0" calls. Recalls etc. will ring the designated local operator.

- In MMC 724 under NTWK LCR NUMPLAN make an entry "0", for example:

```
[NTWK LCR NUMPLAN]
[LCR-03:NONE→0  ]
```

- In MMC 824 for the "0" entry fill out the display as 0010,1,04,N for example:

```
[03:0   →0010   ]
[SZ:1 MAX:04 MB:N]
```

- In MMC 710 make an entry for digits "0010" , length 04, and the route to access the main location. For example:

```
[LCR DIGIT (0002)]
[DIGIT:0010    ]

[LCR DIGIT(0002)]
[LENGTH:04 RT:02]
```

DID Pass Through

This will allow DID calls that come in on circuits on the main location to ring at the remote locations.

- In MMC 724 under NTWK LCR NUMPLAN make an entry for an unused station number block for example 30:

```
[NTWK LCR NUMPLAN]
[LCR-05:NONE→30  ]
```

- In MMC 824 for entry "30" fill out the display to route the calls to switch 002 indicating the total number of DID digits plus the 3 digit access code as the MAX entry as follows:

```
[05:30 →002   ]
[SZ:2 MAX:07 MB:N]
```

- In MMC 714 tell the incoming DID digits to "ring" 30 for all ring plans

- Translate the digits on the receiving side in MMC 714 as normal

NOTE: Repeat steps 1 through 4 for each remote switch using a different NTWK access code for each switch.

Tandem Trunking

If the networked switches are in different area codes, e.g.

Switch 001 is 561
Switch 002 is 954
Switch 003 is 305

then calls from one switch can be routed over the network link and sent over as local calls. For example if a user in Switch 001 wants to call a number in the 305 area code the call could be routed over the network link to switch 003 and sent out as a local call. This is called Tandem Trunking.

To achieve this, LCR entries must be created to route the calls as described by modifying the digits and routing the call across the network link.

- Create LCR entries for each networked Switch area codes.

  E.g. for Switch 001 these would be:

Digits	Length	RT
1954	11	05
1305	11	06

- Create the LCR routes with the digits modified to reflect 7 digit dialing in 954 area code and 10 digit dialing in 305 area code.

  Eg. for Switch 001 these would be:

Route	Class	Group	Modify
05:1	1	802	002
06:1	1	803	003

- Create the modifying digits entries as described in step 2.

  Eg. for Switch 001 these would be:

Entry	Del	I	A
002	01	0029	_ _ _
003	01	0039	_ _ _

These modify digit entries convert the digits dialed by the user into a digit string that will let the destination switch route the call through the DID translation table (MMC 714) to the appropriate local trunk group. This is the group already set up in LCR before networking was applied to allow local calls to be sent out of that switch. For example in switch 001 if local calls go out over trunk group 800 there would be an entry in MMC 714 for digits 561 (switch 001's home area code) to "ring" 800 for each of the ring plans.

PROGRAMMING EXAMPLE:

This example demonstrates the steps involved in networking 2 nodes (Nodes A and B) together over IP. It is assumed that a functional IP network exists between the nodes. This physical network segment can be as simple as a single hub or switch, or can be as complex as a network involving routers, switches, firewalls, etc. For simplicity, it will be assumed that both nodes reside on a local network segment 192.168.1.0.

****See your Data Network Administrator to ensure that the MP20S' and OAS cards can communicate with each other over TCP/IP.*

NODE A:

Node id: 091
Station numbering plan: 2000~2099
VoIP Network trunk group: 803
MP20S IP: 192.168.1.2
OAS: 192.168.1.3

NODE B:

Node id: 092
Station numbering plan: 2100~2399
VoIP Network trunk group: 803
MP20S IP: 192.168.1.4
OAS: 192.168.1.5

Hardware: Install an OAS in each node. Attach the Ethernet cables of the MP20S' and OAS' to the local network segment 192.168.1.0.

NODE A Programming:

- **MMC 830:** Program the IP information for MP20S.
  IP address: 192.168.1.2
  Gateway: 192.168.1.1
  Subnet mask: 255.255.255.0
  <u>*** Now select the reset option to reset the MP20S</u>

- **MMC 831:** Program the IP information for the MGI/OAS
  IP address: 192.168.1.3
  Gateway: 192.168.1.1
  Subnet mask: 255.255.255.0
  <u>*** Now select the reset option to reset the OAS card</u>

- **MMC 615:** Program the MGI Use Groups.
  Set USER=VoIP Networking and choose the IP trunk members (MGI channels) to be used for this "Networking" use. The default values should start with member 1 = 3801, etc.

- **MMC 820:** Program the following values for LINK ID, SIGNAL G/W (enter the MCP/MP20 IP address for each node), and IP type for each node as follows (SELF refers to the current node you are programming):

Index	Link ID	Signal G/W	IP Type
SELF	091	192.168.1.2	Private
SYS01	092	192.168.1.4	Private

- **MMC 724:** Set up the numbering plans. Trunk group 803 will contain all the IP networking trunks.

STN NUM PLAN	Station numbers (2000 ~ 2099)	
TRKG NUMBER PLAN	803 (Add other trunk groups here if other trunk cards are installed. For simplicity, only 803 will be added here.)	
FEAT Dial Number	LCR 9	For C.O. outgoing calls, 9 + phone number
NTWK LCR NUMPLAN	IDX-01 : 21	Converted to 2100~2199 in MMC 824
	IDX-02 : 22	
	IDX-03 : 23	

- **MMC 210:** Tenant On and Off: For LCR ENABLE: Set to ON to enable Least Cost Routing. Set ICM EXT FWD to ON.

- **MMC 824:** Set up the extension dialing plan for the networked system (node B). 092 is the node id for Node B. **<u>Be sure to set DISP=Y</u>.**

Index	Station	Converted	SZ	Max	MB	DISP	
01	21	09221	4	7	N	Y	Stations in Node B
02	22	09222	4	7	N	Y	
03	23	09223	4	7	N	Y	

- **MMC 714:**

Entry	Digit	MOH	1~6	CW	Delete	Name
001	20**	-	B	-	0	-
002	21**	-	B	-	0	-
003	9	-	9	-	1	-

- **MMC 310:** LCR class of all stations are set to 1 (default) since there is only one path to the other Node.

- **MMC 603:**

  803 VoIP networking signaling trunks

- **MMC 710:**

Index	Digit	Length	LCRT	
0001	092	7	1	stations, station groups in system B
0002	1	3	4	special numbers
0003	2	7	4	local calls need at least 7 digits (if needed)

  Note: this example does not have local CO trunking.

- **MMC 711:** You MUST Set HHMM= 0000, LCRT=1 for each day of the week <u>unless</u> specific times and days are required to be set.

- **MMC 712:**

Table	Time	Class	Route	Modify
01	1	1	803	Calls group 803

- **MMC 825:**

ADD Number To Name	Yes
Use Remote Voice Mail	No
Remote CID Number	Yes

- **MMC 823:** Set CCNR and CCFB to Yes for certain features to work across the network.

NODE B Programming:

- **MMC 830:** Program the IP information for MCP/MP20.
  IP address: 192.168.1.4
  Gateway: 192.168.1.1
  Subnet mask: 255.255.255.0
  <u>*** Now select the reset option to reset the MCP/MP20</u>

- **MMC 831:** Program the IP information for the MGI2
  IP address: 192.168.1.5
  Gateway: 192.168.1.1
  Subnet mask: 255.255.255.0
  <u>*** Now select the reset option to reset the MGI2</u>

- **MMC 615:** Program the MGI Use Groups.
  Set USER=VoIP Networking and choose the IP trunk members (MGI channels) to be used for this "Networking" use. The default values should start with member 1 = 3801, etc…

- **MMC 820:** Program the following values for LINK ID, SIGNAL G/W (enter the MCP/MP20 IP address for each node), and IP type for each node as follows (SELF refers to the current node you are programming):

Index	Link ID	Signal G/W	IP Type
SELF	092	192.168.1.4	Private
SYS01	091	192.168.1.2	Private

- **MMC 724:** Set up the numbering plans. Trunk group 803 will contain all the IP networking trunks.

STN NUM PLAN	Station numbers (2100 ~ 2399)	
TRKG NUMBER PLAN	803 (Add other trunk groups here if other trunk cards are installed. For simplicity, only 803 will be added here.)	
FEAT Dial Number	LCR 9	For C.O. outgoing calls, 9+phone number
NTWK LCR NUMPLAN	IDX-01:20	converted to 2000~2099 in MMC 824

- **MMC 210:** Tenant On and Off: For LCR ENABLE: Set to ON to enable Least Cost Routing. Set ICM EXT FWD to ON.

- **MMC 824:** Set up the extension dialing plan for the networked system (node A). 091 is the node id for Node A. **<u>Be sure to set DISP=Y</u>**.

Index	Station	Converted	SZ	Max	MB	DISP	
01	20	09120	4	7	N	Y	Stations in Node A

- **MMC 714:**

Entry	Digit	MOH	1~6	CW	Delete	Name
001	20**	-	B	-	0	-
002	21**	-	B	-	0	-
003	9	-	9	-	1	-

- **MMC 310:** LCR class of all stations are set to 1 (default) since there is only one path to the other Node.

- **MMC 603:**

  803 VoIP networking signaling trunks

- **MMC 710:**

Index	Digit	Length	LCRT	
0001	091	7	1	stations, station groups in Node A
0002	1	3	4	special numbers
0003	2	7	4	local calls need at least 7 digits (if needed)

Note: this example does not have local CO trunking.

- **MMC 711:** Set HHMM= 0000, LCRT=1 for each day of the week <u>unless</u> specific times and days are required to be set.

- **MMC 712:**

Table	Time	Class	Route	Modify
01	1	1	803	—

- **MMC 825:**
  ADD Number To Name Yes
  Use Remote Voice Mail No
  Remote CID Number Yes

- **MMC 823:** Set CCNR and CCFB to Yes for certain features to work across the network.

- **MMC 835:** You must set AUDIO CODEC to G.729A for IP Networking. DTMF TYPE should be set to the same setting on each node (INBAND or OUT of BAND)

Now you should be able to dial a 4 digit extension from Node A to a station in Node B (and vice-versa).

2.13 NETWORKING

The networking feature package allows up to 99 OfficeServ 7200-S systems to be connected together with a high level of feature transparency. The physical connection between the systems can be set up via an IP network or a proprietary PRI connection (based on Q-SIG specification). The following feature enhancements are supported between networked systems.

- **Auto Answer Accross Network:** This setting will allow station to station calls across the network to follow the auto answer setting of the called keyset.
- **Call Completion, Busy Station (CCBS)** also known as Callback or Busy Station Callback. When a station in one system calls a station in another system across the network link and the destination station is busy the calling station can set a Callback to the busy station. When the busy station becomes idle the system will notify the callback originating station by ringing that station and when the originating station answers, the system will call the destination station. Not available on QSIG over PRI.
- **Call Completion, No Response (CCNR)** also known as Callback or No Answer Callback. When a station in one system calls a station in another system across the network link and the destination station does not answer the calling station can set a Callback to the called station. When that station indicates the user is present by becoming busy then idle the system will notify the callback originating station by ringing that station and when the originating station answers, the system will call the destination station.
- **Forward External:** This feature operates in the same manner as a non networked system with the exception that, because calls across a network link are trunk calls, network calls do not follow the ICM FWD EXT ON/OFF setting in MMC 210. It is therefore suggested that this setting be set to ON in a networked switch to avoid confusion in operation between networked and non networked calls.
- **Call Intrusion (Barge In):** Calls across the network link can be barged in on however the barging station will not be muted.
- **Call Offer/Call Waiting (Camp On):** This feature operates in the same manner as in a non networked switch. When a called station is busy the caller can press a camp on key and appear as a ringing call on the second call button. The Auto camp on feature will not work on calls across a network link if set to ON in MMC 110.
- **Call Transfer:** Calls answered in one network node can be transferred to a station or station group in another network node.
- **Transfer Retrieve:** Calls on Transfer Hold during a screened transfer can be retrieved by pressing the call button for that call.
- **Transfer Recall:** Calls transferred across a network link will recall to the transferring station after the originating system transfer recall timer expires. After recalling, if not

answered prior to that systems attendant recall timer expiring, the call will recall to that systems designated operator group. Attendant recalls will not recall to a ‘Centralized Attendant”.

- **DID with Pass Through:** Incoming DID, DNIS or DDI calls can be routed through one switch across a network link to be processed by the DID table of the destination switch.
- **Do Not Disturb (DND):** This feature operates in the same manner as in a non networked switch.
- **Caller ID:** Caller ID in its various forms that are currently available (Analog CID Name and Number, ANI Number, PRI Name and Number and BRI number) will be transported across the network link with the original call.
- **Centralized Attendant:** This feature basically allows a user in any switch to dial “0” and ring at the designated Central attendant group. Each system on the network requires its own designated attendant group for local usage, recalls and the like.
- **Intercom Calling/Uniform Dialing Plan:** Station to station and station to group calls can be made across the network link without having to dial an access code for a call within the network. LCR can also be programmed to route calls across a network link and to access local trunks in another networked system.
- **Centralized Voice Mail with Message Waiting Lights:** This feature will only operate with SVMi voicemail systems only. Users in one node can call forward to the SVMi group in a different switch and messages left in that switch will be indicated on the VMSG key in the origination switch. Messages can be returned to the SVMi by pressing the VMSG key.

NOTES

1. The OfficeServ 7200-S does not support the SVMi-20E card. However, the OfficeServ 7200-S can be a remote system that used the Centralized Voicemail off of another node. All references to SVMi and Centralized Voicemail in this document refers to SVMi/Centralized Voicemail installed in other nodes such that the OfficeServ 7200-S is a remote node using the Centralized Voicemail resources remotely via the SPNET.
2. It is recommended that the embedded OfficeServ 7200-S Voicemail NOT be used as the Centralized Voicemail.

Enhanced Networking

OfficeServ 7200-S also supports the following Enhanced Networking features:

NETWORK TRUNK RING DESTINATION

This feature permits for analog trunks to be assigned to ring at any station or group destinations anywhere in the network. Trunk facilities terminating at the main office location can be assigned to ring directly at stations, station groups, or single line ports in the branch offices. This feature provides greater flexibility for networking customers to share incoming trunk facilities across the corporate network.

Notes:

- The ring destination for local and remote stations or station groups must be assigned in the system where the trunks facilities are terminated.
- Remote stations and group numbers from anywhere in the network can be assigned in MMC 406 of the source node of the connecting trunks.
- Any trunks not assigned a specified ring destination in MMC 406 will default ring to the operator group.
- When CID in present, the CID information is passed across the network.
- This feature is supported over Q-sig and IP networking.
- MMC 406 is primarily used to terminate analog lines. Digital trunks will not follow the setting of MMC 406, but will following the setting of MMC 714.

MMC INFORMATION: MMC 406 (set trunk ring destination)

CENTRALIZED ATTENDANT: This feature permits for an Attendant position to be assigned to handle second level recalls (hold/e-hold/transfer/campon/park) from local operators of other nodes within the network. The way this feature operates is transferred and held calls will first recall to the station that performed the transfer or placed the call on hold. After the preset timer expires (transfer recall or hold recall timer) the transfer/held calls will then recall to central attendant instead of recalling to the local operator.

An additional option is available for centralized attendant that allows the remote attendant recall destination to be determined by time of day. This is accomplished by selecting the remote attendant destination and which ring plan the attendant recalls will follow. An example would be during normal working hours, transfer and hold recalls from stations in the branch office will recall to the main attendant at the main office location. When the attendant at the main office goes home at 5:00pm, the recalls from the branch office can be automatically rerouted to a local attendant or local station within the branch office that is still on duty after 5:00pm. This is accomplished by changing the attendant destination on a per ring plan basis in MMC 825.

Notes:

- Centralized attendant handles second level recalls across the network. This is not to be confused with centralized operator which handles calls from remote stations dialing 0.

- Centralized attendants can only be assigned on a per node basis. Each node in the network can have a local attendant or a centralized attendant, but not both.
- Attendant recall destination can be changed by time of day by selecting ring plans and recall destinations in MMC 825.
- This feature is supported over Q-sig and IP networking.

MMC INFORMATION:

MMC 825: USE CENT ATT: RING1-6, YES/NO). This is used to set each node to use attendant of a remote system.
MMC 825: (REMOTE ATTN NUM:RING1-6, XXXX). This is used to assign the station number or attendant group of the remote attendant that second level recalls will be routed.
MMC 501:
(E-HOLD RECALL TM) This timer controls the duration of time a call is held exclusively at a station before recalling.
(SYS HOLD RECALL) This timer determines the time calls can be left on hold before recalling back to the holding station. This is a system-wide timer. Setting timer to 000 will defeat this feature and no recalling will take place.
(TRANSFER RECALL) This timer determines the time transferred calls ring before recalling. This is a system-wide timer.
(PARK RECALL TIME) This timer controls the duration of time a call is parked before recalling to the call park originator.
(CAMP ON RECALL) This timer controls the duration of time a camped-on call will stay at a destination before recalling to the transferring station.
(ATT. RECALL TIME) This is the length of time a transfer recall will ring at a station before recalling the local operator or Centralized Attendant.
(RECALL WAIT TIME) This is the time any recall (hold or transfer) continues to recall at your station before it recalls to the local operator or Centralized Attendant.

UCD FINAL/INVALID DESTINATION NETWORK ROUTING

This feature has been improved to allow UCD groups to overflow unanswered calls to other stations or station groups anywhere in the network. For example, call arriving at the branch office can ring to a UCD; If the call goes unanswered for a predetermined time period, the call will then overflow, and ring to the final destination. The final destination can now be programmed as a station number or group number in the main office or any other station or station group in the network.

Notes:

- Station groups must include local station only. Network groups are not supported. Station groups cannot include members from different nodes
- Only the final destination or invalid destination setting can include network station or station groups.
- When CID in present, the CID information is passed across the network.
- This feature is supported over Q-sig and IP networking.

MMC INFORMATION:

MMC 607 (set UCD FINAL DEST)
MMC 733 (set INVLID DEST)

GROUP OVERFLOW ACROSS THE NETWORK

This feature allows the overflow destination for a station group to exist in the same node or a remote node within the network. An example of the how this works is this feature will allow calls to arrive and ring at the branch office. If the call is not answered by any member of the branch office group within a preset time period, the calls will then leave the branch office and overflow to a station group in the main office.

Notes:

- Station groups must include local stations only. Network groups are not supported.
- Station groups cannot include members from different nodes.
- Only the next port, final destination or invalid destination setting can include network station groups.
- When calls ringing at the originating group in one node overflow to a backup station group in another node, the calls will stop ringing at the originating node and will only ring at the overflow destination. The calls will not ring at the originating station group and the overflowed group simultaneously.
- The Group overflow feature only supports overflowing to other station group. Individual extensions or virtuals from another node cannot be assigned as a network overflow destination.

- Calls are allowed to ping-pong from one network node to another. For example a call can ring a station group in the originating node, overflow to a station group in another node, and then overflow back to the originating node or station group.
- Calls to station groups of the originating node can be overflowed to a centralized voicemail group within another node on the network.
- When incoming calls with caller ID are presented to the originating group, the caller ID information is passed on the overflow destination.
- This feature is supported over Q-sig and IP networking.

MMC INFORMATION:

MMC 601 set “OVERFLOW timer” and assign “NEXT PORT” destination.

CENTRALIZED VOICEMAIL WITH SOFT KEY FUNCTIONALITY

This enhancement allows all users within the network that are using the SVMi() centralized voicemail to use the softkey options to navigate through the voicemail functions. For example, when a user calls voicemail to retrieve messages, softkey options such as FAST FORWARD, REVIEW, PLAY, SAVE, DELETE are displayed to the keyset users. This offers a new level of flexibility to centralized voicemail. Stations in branch offices that do not have an SVMI() card installed can share most of the voicemail functions of the main office. Station users in remote office using centralized voicemail will not require any additional training because voicemail operation is the same for local and remote users.

Notes:

- The functionality is only available when using the SVMi() card for Centralized Voicemail.
- Softkey options are only available to the Samsung digital stations.
- This feature is supported over Q-sig and IP networking.

MMC INFORMATION:

MMC 825 (USE REMOTE VM=YES/NO) set to YES for remote nodes using centralized voicemail.
MMC 825 (REMOTE VM NUMBER= XXXX) in remote nodes, enter voicemail station group number of centralized voicemail.

VOICEMAIL TRANSFER (VT) ACROSS THE NETWORK

The VT key function has been modified to allow a remote station user to program a VT key with the group number of the centralized voicemail (which resides in a different node). This enhancement will allow stations in the main or branch offices to transfer calls across the network directly into personal mailboxes of users anywhere in the network, regardless or where the centralized SVMi voicemail card is. The operation will remain the same as a standalone node. The user with a call in progress will press the network VT key, and then dial a mailbox number. For example, the user with a call in progress will press the network VT key, and then dial a specific user's extension number. The caller is then transferred directly to the specified user's personal greeting.

Notes:

- This Network VT key functionality requires the use of the SVMi centralized voicemail feature.
- MMC 825 USE REMOTE VM must be turned on in the nodes remote to the centralized voicemail hardware before the VT key in the remote nodes can use the station group number of the node with the SVMi card.
- MMC 722/723 (program VT key with centrailized voicemail station group extender, for remote nodes).
- Any keyset user can have multiple VT keys with different extenders.
- This feature is supported over Q-sig and IP networking.

MMC INFORMATION:

- MMC 825 (set to “USE REMOTE VM=ON”, in remote nodes)
- MMC 825 (set to “REMOTE VM NUMBER XX-XX”, in remote nodes)

NETWORKING MESSAGE KEY WITH LED INDICATIONS

This feature permits station users to set a message wait indication to remote stations in the network. When calling a remote station and receiving busy or no answer condition, the caller can press the MSG softkey in the display (or dial the MSG access code) and leave an indication that a message is waiting. The message key will flash red at the remote station receiving the message notification. The remote station can then press the message key to see which station left the message. The remote station user can then press reply to return the call to the station that set the message.

Notes:

- Single line telephones will receive a distinctive message waiting dial tone.
- A maximum of five messages can be sent to a network station MW key.
- In order for this feature to work properly, the feature access code for message waiting must be the same for every node in the network.
- This feature is supported over Q-sig and IP networking.

MMC INFORMATION:

- MMC 722 (assign MSG key)
- MMC 724 (Feature access code, MSG=XX)

NETWORK PAGING

The feature allows station users to assign and page other page zones in other nodes in the network. The main requirement for this feature to work is each node must be programmed to have a unique network page access code or a network page (NP) key. An example would be to program a Network page key as NP023. When a user presses this key, a internal zone page will be made to all member of zone 3 in node 02. The step for accessing a network page group is:

Network paging using access code: Go offhook, dial XX+NN+Z= Paging across the network

OR

Network paging using NP key: Press the selected NP(NNZ) key = Paging across the network.

XX = (two digit network Page access code)
NN = (two digit item entry assigned in MMC 820)
Z = (One digit page access code, see chart below)

Internal Zone Paging	External Zones
= All Internal Zones	5 = External Zone
1 = Internal Zone	6 = External Zone
2 = Internal Zone	7 = External Zone
3 = Internal Zone	8 = External Zone
4 = Internal Zone	9 = All External Zones

* = All Internal/External Zones

Notes:

- Multiple NP keys can be programmed per keyset in MMC 722.
- Pages can be made to page groups in other nodes; however, stations from other nodes in the network cannot exist in a local page group.
- NP keys can be programmed in MMC 722/723 with or without the extenders.
- This feature is supported over Q-sig and IP networking.

MMC INFORMATION:

- MMC 722 (program NPAGE key to keysets)
- MMC 724 (program NPAGE feature access code)

REMOTE HOLD ACROSS THE NETWORK

The purpose of the remote Hold feature is to allow a user to place a trunk call on exclusive hold at another station in another node. This is called Remote Hold because it is on hold at a station other than yours. To do this, a user (ext. 2001) talking on a CO line, initiates a transfer to another station in another node. Press TRSF, receive dial tone, then dial the extension number (3001). When you hear ring back tone press the HOLD key. This places the trunk call on hold at the remote station (3001). The user then can use the designated Network Page key to make an announcement that the caller is holding on station 3001.

At this point the user at ext 3001 simply takes the call off hold and begins speaking. When the user is not near his station when he hears the page announcement he can go to another station and dial the Hold Pick up code 12 + 3001.

Note: A CO call cannot be put on Remote Hold at a virtual extension in another node. It can be put on Remote Hold at a virtual extension within the same node.

MMC INFORMATION: NONE

CALL PICKUP ACROSS THE NETWORK

This feature enhancement allows ringing calls, recalls and held calls to be picked up by other stations across the network. A station user in the main office can use the directed pick-up, hold pickup or page/park pickup codes to answer calls from the main or branch offices. An example would be a call is answered at the branch office. The call is then parked. The station user performs an internal page to all members in the main office and announces that the call is parked for pickup. Any user can respond to the page by going to any station in the main office and dialing the page pickup access code to remove the caller from hold in the branch office location.

Notes:

- Directed call pickups, hold pickups, and park pickups are all supported over the network.
- This feature is supported over Q-sig and IP networking.
- When CID in present, the CID information is passed across the network.

MMC INFORMATION:

MMC 722 (set DIRPK, HPK, and PAGPK keys)
MMC 724 (set directed pickup, hold pickup, and page/park pickup access codes)

NETWORK SELECTION AND BUSY LAMP FIELD INDICATION ACROSS THE NETWORK

Programmable keys can be assigned as NS/BLF keys that will function across the network. A station with NS/BLF keys in can programmed with station numbers from other nodes within the network. The NS/BLF keys will provide a visual status indication to the associated station. The NS/BLF key will be off when that associated station is idle, will light red when the station is in use, and will flash when the station is in DND. The station user can press a programmed Network Selection key to call or transfer to a station user in other nodes in the network. The steps for accessing an NS/BLF key are:

NSS/BLF NS key: Press the selected NS(XXXX) key = Place call to selected station across the network.
XXXX = (3 or 4 digit network station number)

Notes:

- NS/BLF keys can be programmed on any keyset or add-on module.
- This feature is not supported over PRI Q-sig. This feature requires a LAN connection.
- Network stations are supported on NS keys, but virtuals and station group numbers cannot be associated with NS Keys.
- Any keyset user can have multiple NS keys with different extenders.
- The NSS keys can be used to answer ringing calls from the associated station.
- A visual indication is provided when the associated station is ringing, but no audible indication is presented.
- The network selection keys can be used for directed call pickup across the network if this feature option is activated in MMC 210, DSS KEY DPU set to ON in the NODE where that ringing station is physically connected.

MMC INFORMATION:

- MMC 722/723 (set NS key to keysets)
- MMC 210 (DSS KEY DPU)

ADDITIONAL ENHANCEMENTS

Call Coverage Key

The call coverage key (CC key) is a feature where one station can visually and/or audibly monitor the call status of another, or several other stations, and serve as a backup answering position of other stations users. The secretary can monitor (and answer) the call status of several executives. For example, an incoming call would begin to ring at the executive's telephone. After a pre-programmed time period, the call would continue to ring the executive, but would also delay ring to a programmed Call Coverage key on the secretary's station. The secretary can press the associated Call Coverage key to answer the call for the executive._The steps to accessing a CC key are as follows: Call coverage is a single node feature. The feature is not supported across the network.

CC key: Press the selected CC(XXXX) key = Answer delayed ringing call for another station.

XXXX = (3 or 4 digit station)

Covered Station Status	Covering Station LED Indication
Ringing on CO	Fast Flash Green
Ringing on intercom call	Fast Flash Green
Transfer/Hold Recall	Fast Flash Amber
Busy	Steady Red
In DND	Flashing Red
Idle	LED OFF

Notes:

- Call Coverage only works for stations in a standalone node. Call coverage will not work for virtuals extensions, or station groups. Call Coverage also does not support stations, virtuals, or station groups in other nodes.
- A call cannot be transferred using the call coverage key
- The call coverage key cannot be used as a direct station selection (DSS) key.
- When programming MMC 722/723, the extension of the call coverage key can only be programmed using station numbers from the local node. Stations, station groups or virtual numbers from remote nodes cannot be used.
- If the CC delay ring timer is set to 000 (MMC 502) the CC key will ring immediately.
- The CC delay timer is programmable on a per station basis.
- If the station user is busy, there will not be any offhook ringing audible for a CC key.
- The call coverage key can be used to perform directed call pickups.
- Calls ringing on the CC key will not follow the covering station's call forwarding, but will follow the forwarding of the original or covered station.
- Hold recalls and transfer recalls to the covered stations will also recall to the covering station. The CC key will flash amber during transfer/hold recalls.
- This feature is not supported over the network.

Home Page

MMC INFORMATION:

- MMC 722/722 (set CC keys to keysets)
- MMC 502 (set CC RNG DLY timer, set on a per station basis)

PROIRITY CALL ROUTING

This new feature allows certain calls queuing to station groups to be given a higher priority over other calls already in the queue. This is a very useful enhancement to customer service organizations. Business owners can offer improve customer service and reduced hold time for their most preferred customers. An example of this is: Six calls are holding in a queue. The seventh call arrives into queue. The DID information from the telco identifies this caller as a preferred customer. This preferred customer has been designated a higher priority. This customer will be removed from the seventh spot in the queue and move to the first position in queue. Another example of how priority call routing can be used is a call to the branch office is queued to a busy station group. After a predetermined time, the call is overflowed from the branch office station group to the main office. When the call arrives at the station group in the main office, it is designated a higher priority because it has already been in queue at the branch office for a long time. The higher priority overflow from the branch office is now move up in the queue and answered immediately. The next available agent will receive the priority call from the preferred customer. Priority routing can be assigned by the incoming DID number, the Caller ID, or on a per trunk basis.

Notes:

- Priority call routing is supported on analog and digital trunks.
- Priority routing can be set on each DID number or each Caller ID line or trunk port.
- From 1 to 9 priority levels can be set per call with 1 being the highest priority.
- Call priority can be set to station or station groups within the local node only.
- Priority calls can be set to normal or UCD groups, but not to VM/AA/MSG groups.
- The settings for priority call routing in MMC 759 will take precedence over distinctive ringing settings in MMC 419.

DISTINCTIVE RINGING

This existing feature has been enhanced to allow each stations or trunk to assign and send a distinct ring tone to any station receiving the incoming call. When specified trunks or stations are assigned to distinctive ring, a different audible tone and cadence will be heard at the station receiving the incoming call. This will allow the receiving station users to identify one incoming caller from another by the audible sound. An example of how this may be used is the receptionist is answering many incoming calls and transferring the calls to other stations. She is expecting a call from the boss. When the boss calls in using his mobile phone, the system will recognize the bosses caller ID and provide a unique audible tone and ring pattern. Now the receptionist can positively identify the call of the boss and answer the call immediately with a personalized greeting. This feature can be set on a per node basis.

Notes:

- Distinctive ringing is assigned on a per station, per trunk basis.
- There are eight different ring tones available for distinctive ringing.
- There are 5 cadences (ring patterns) available for digital and single line telephones.
- This feature is not supported over the network.

MMC INFORMATION:

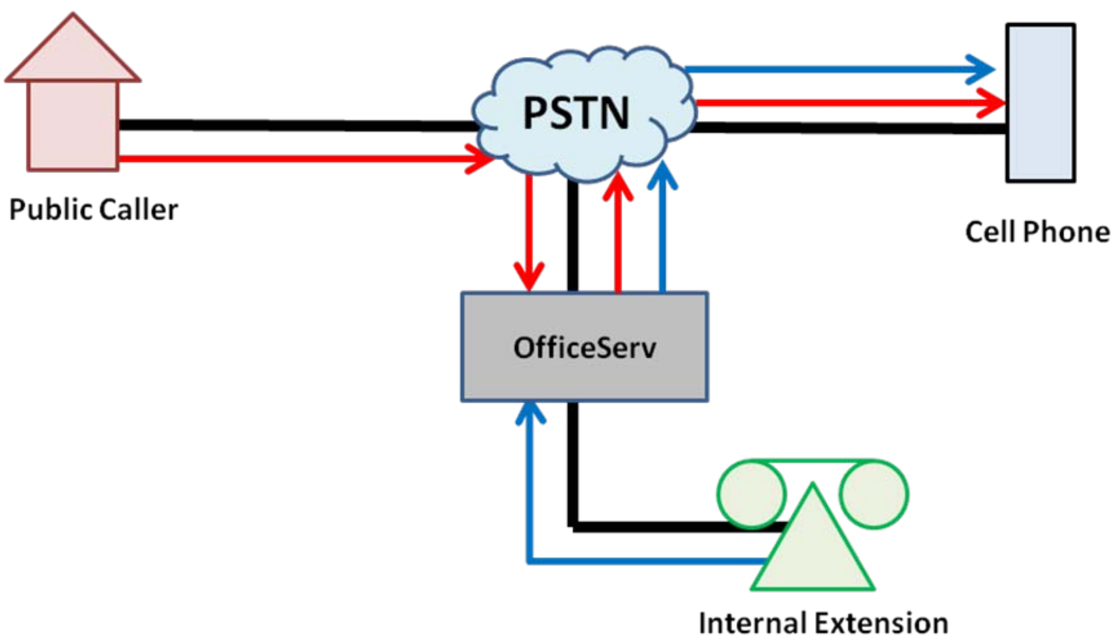
MMC 759 (set CID Priority)
MMC 714 (set DID Priority)
MMC 419 (set TRUNK Priority)

2.14 OFFICESERV CONNECT AND MOBEX

The OfficeServ Connect, MOBEX, and Executive MOBEX features are a part of the Samsung Fixed Mobile Convergence solution. These 3 distinct features can be used separately, or combined together. This provides maximum flexibility and customization.

MOBEX

MOBEX, short for Mobile Extension, is a feature that allows you to assign a local 2, 3, or 4 digit extension to a telephone that is not a part of the OfficeServ system. This can be a cell phone, home phone, or any other phone number. This is very similar to a speed dial, except that MOBEX ports can be assigned to DSS keys, placed in Station Groups, and even have a DID routed to them. The diagram below show the basics of MOBEX:



Call to Mobile Extension

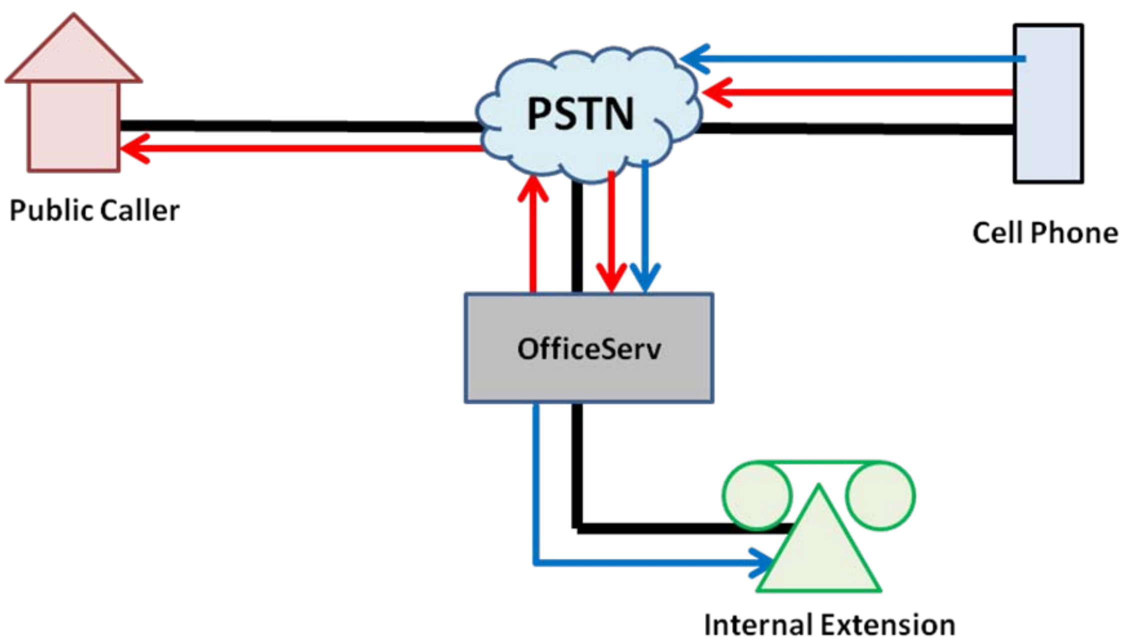
1. CO Call to Mobile Extension displays CID of Public Caller
2. Intercom call to Mobile Extension displays CID of Extension

The red arrows show the call flow of a CO call. The public caller dials a DID number which has a destination of a MOBEX port (in this case a cell phone). When the call reaches the cell phone the call will display the caller ID of the public caller.**

The blue arrows show the call flow of an intercom call. The internal extension dials the 2, 3, or 4 digit MOBEX port number and the call is routed automatically to the cell phone. When the call reaches the cell phone the call will display the extension number of the internal extension user.**

Executive MOBEX

Executive MOBEX is an optional, licensed enhancement to the MOBEX feature. Executive MOBEX is used to assign MOBEX DSP to a selected MOBEX port. These MOBEX DSPs are a special type of DTMF Receiver that will connect to any calls made to or from the MOBEX port. MOBEX DSPs are provided by the OAS card. These MOBEX DSPs allow the Executive MOBEX user to dial a code during a call on their Mobile Extension and receive system dial tone back so they can perform call control activities such as transferring the call, setting up a conference, or placing the caller on hold at another keyset. In addition, Executive MOBEX allows a special DID to be set up that allows Executive MOBEX users to call in to the system from their Mobile Extension and receive system dial tone in order to place calls through the system. This is simply a variation of the DISA feature. The diagram below shows the call flow for an Executive MOBEX call:



Call from Executive Mobile Extension

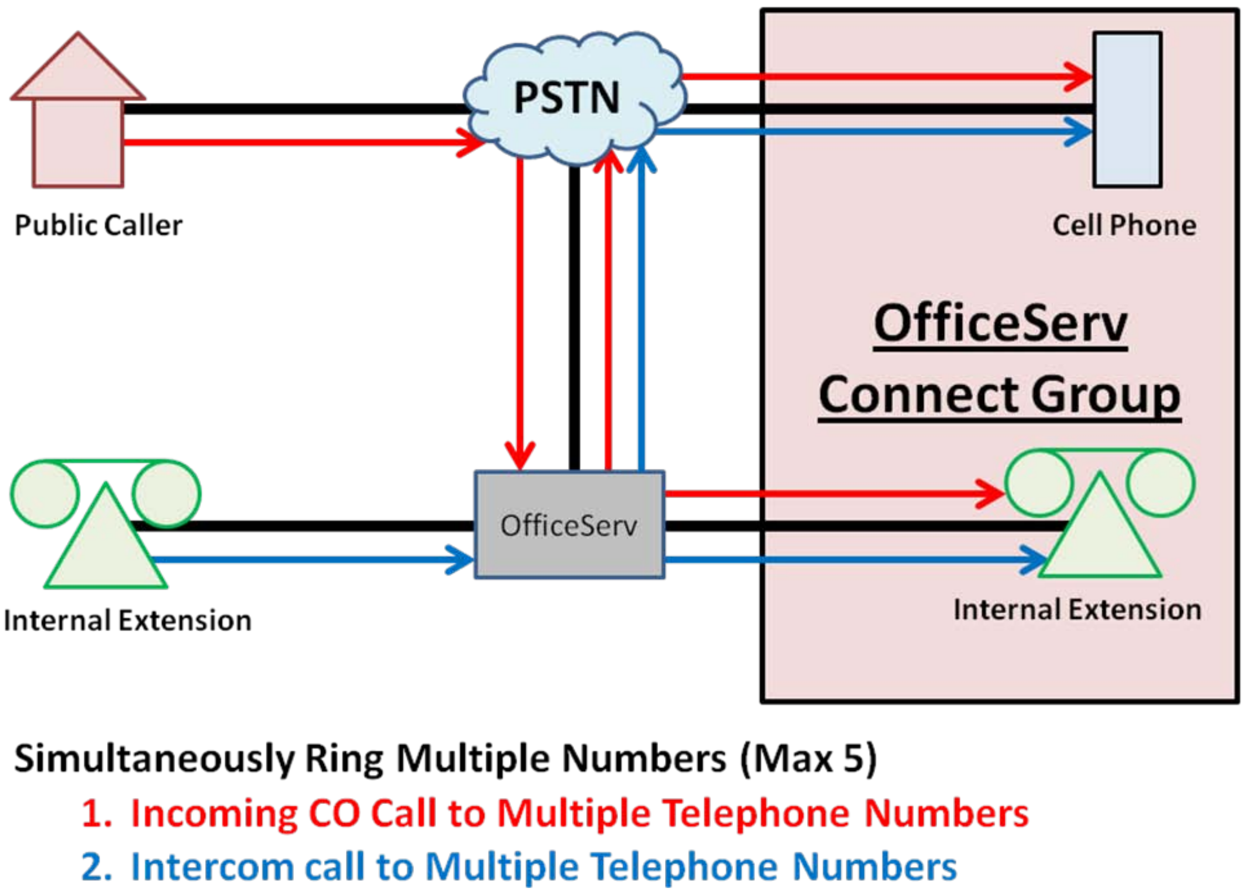
1. Call to CO will display CID of MOBEX port instead of Cell Phone
2. Call to Internal Extension will show MOBEX Extension as CID

The red arrows show a call from the Executive MOBEX user to an external party by calling into the OfficeServ system. This allows the user to make a call from the Mobile Extension without the called party seeing their cell phone's CID. This is because the caller will instead see the CID of the MOBEX port itself, which can be set in MMC 321.

The blue arrows show a call from the Executive MOBEX user to an Internal Extension. By calling in to the OfficeServ system the user can dial the 2, 3, or 4 digit Extension number and reach the Internal Extension without displaying their cell phone number. Instead the system will show the Internal Extension user the 2, 3, or 4 digit MOBEX port number as CID instead.

OfficeServ Connect

The OfficeServ Connect feature allows simultaneous ringing of a Master keyset and up to 5 other devices. These devices can be internal keysets (SLT, DLI, WIP, ITP, SIP Stations, etc) as well as MOBEX ports. This means that when combined with the MOBEX feature, OfficeServ Connect allows calls to ring at a keyset and a cell phone (or other external device) simultaneously. When one device answers the call, the other devices stop ringing. If no device answers the call within the Master keyset's No Answer Forward time the call is routed to the Master keyset's voicemail. The diagram below shows the call flow for OfficeServ Connect:



The red box shows that the Internal Extension and MOBEX cell phone are set up in an OfficeServ Connect group. This means that when an internal or external call is made to the

Internal Extension (the Master keyset of the group) it will ring at both the keyset and the cell phone simultaneously. The call can be answered at either location.

NOTE: MOBEX requires the use of PRI or SIP trunking. Executive MOBEX requires a valid Executive MOBEX license as well as an OAS card configured to allow MOBEX DSPs in MMC 858.

****:** The public caller or internal extension caller ID can only be sent to the MOBEX port if your PRI/SIP service provider allows it. Some PRI/SIP providers only allow outbound calls to send the caller ID of a number than bills to them. In these cases you must choose a number that will be sent for all MOBEX calls.

A SAMPLE APPLICATION (STEP BY STEP)

A customer site of yours has 4 sales people who work 1 day a week in the office and spend the rest of their time travelling. When potential clients call in they are routed to a group that contains 4 keysets belonging to these sales agents. When calls aren't answered they go to a general delivery mailbox that is checked hourly by the on-call agent. The customer would like you to provide a more efficient solution to this scenario.

PLANNING:

There are a few aspects of this scenario to consider:

- Callers come in to a group
  - This means that setting all call forwarding on each agent's phone isn't a good solution because no group calls would ever get to an agent (group calls can't follow station forwarding).
- Each agent spends more time out of the office than in
  - This means the solution has to center around the agent being out of the office instead of in the office.
- They need to process calls in and out of the office easily
  - This means the solution has to transition from being local to being away easily. IP phone / Softphone use isn't a good fit here because the user would have to reprogram the server IP address each time they came into or left the office.
- Each agent needs access to the group delivery mailbox
  - This means all agents will need some kind of alerting when a message is left.

This allows us to flush out which features and/or programs we need to set up:

- Callers come in to a group
  - We need to set up a DID in MMC 714 that rings to a station group in MMC 601 that contains all agent phones (2011 – 2014)
- Each agent spends more time out of the office than in

- A MOBEX port will allow them to process calls remotely, so each agent will need a MOBEX port
- They need to process calls in and out of the office easily
  - They will have a local extension, so they will need OfficeServ Connect turned on to allow calls to ring to their keyset and their MOBEX port simultaneously
  - They will need to be set as Executive MOBEX users to allow them to make calls from their Mobile Extension through the system.
    - In order to use Executive MOBEX you will need a license for 4 Executive MOBEX users and an OAS card.
- Each agent needs access to the group delivery mailbox
  - The Samsung Voicemail will need the general delivery mailbox to be a LIST box that distributes messages to the 4 agents' mailboxes and each mailbox will need Message Alert enabled to call their MOBEX extension when messages are left.

PROGRAMMING:

Armed with the above knowledge we can now begin programming. For the purposes of this sample we will skip the voicemail programming. Consult your voicemail documentation for details on how to set up a general delivery mailbox.

1. Configure the DID in **MMC 714**: set a digit of 1234 to ring group 5005 for all 6 ring plans.

2. Configure group 5005 in **MMC 601**: set the type to Normal, the ring type to Distributed, the Next Hunt timer to 20 seconds, the Next Port to voicemail, and an overflow of 120 seconds. This will give agents ample time to answer the call before sending them to the group mailbox.

3. In **MMC 724** set up 4 MOBEX ports: Port 1 = 3011, port 3 = 3012, port 5 = 3013, and port 7 = 3014. Note that only odd-numbered MOBEX ports get directory numbers. This is because each MOBEX call uses up two MOBEX ports, so the even-numbered ports do not require directory numbers as they are not user-accessible. Also set the MOBEX Feature Code. For this example we will use *8 as the MOBEX feature code.

4. In **MMC 329** set a MOBEX port to member 1 of each station's OfficeServ Connect group: 2011 -> 3011, 2012 -> 3012, etc. This will allow calls to ring the desk phone and cell phone simultaneously.

5. In **MMC 328** configure the MOBEX ports to dial the cell phones: set the TEL NUMBER field to dial trunk group 801 (the default PRI trunk group) and the 7 or

10 digit cell phone number. This will allow the cell phones to be called when the MOBEX directory number is dialed.

6. In MMC 860 enter the Executive MOBEX license

7. In MMC 328 set each of the 4 MOBEX ports to be Executive users. In the CLI field enter the 7 or 10 digit cell phone number. Set the Master station to the corresponding station from MMC 329 (2011 for 3011, etc). This will assign the MOBEX ports as Executive MOBEX users.

8. In MMC 714 create a back door DID number that the agents can call in to in order to make calls through the switch. Set the Destination of this DID to *8. For this example we will use the DID 972-555-1234

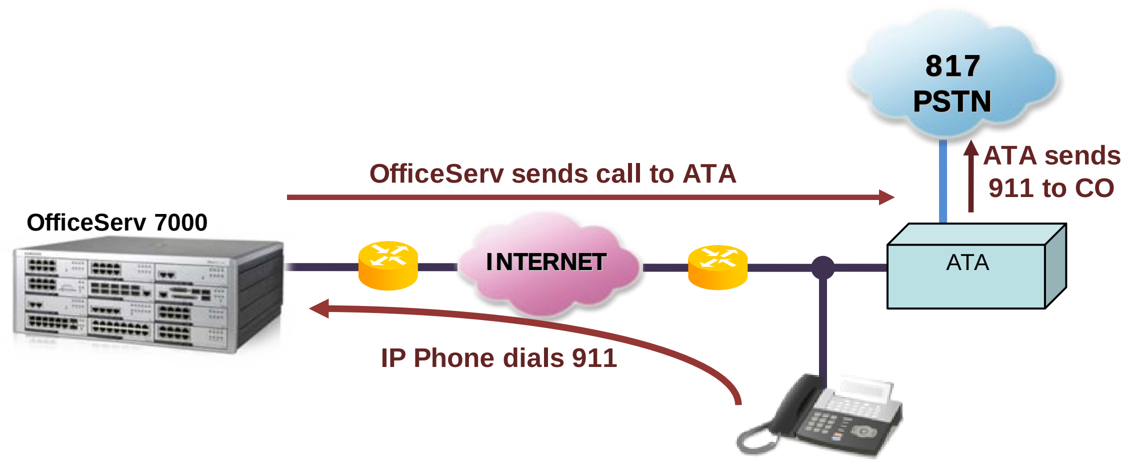
9. In **MMC 722** place a MOBEX button on each of the 4 stations (2011 – 2014). This will allow the agents to turn their OfficeServ Connect ringing on and off with a one-touch key.

That is all the programming that is necessary. Calls that ring to the group will ring to an agent's keyset and cell phone for 20 seconds and then ring to the next agent. If no agent answers within 2 minutes the caller will go to the general delivery voicemail box. When an agent is in the office they can press their MOBEX button to disable the ringing at their cell phone without having to adjust group parameters or affect their voicemail forwarding. When the agent answers the call on their Mobile Extension they can press *8 to receive system dial tone and transfer the caller back to the office. In addition, they can dial 972-555-1234 from their cell phones to receive system dial tone that will allow them to make calls as if they were in the office.

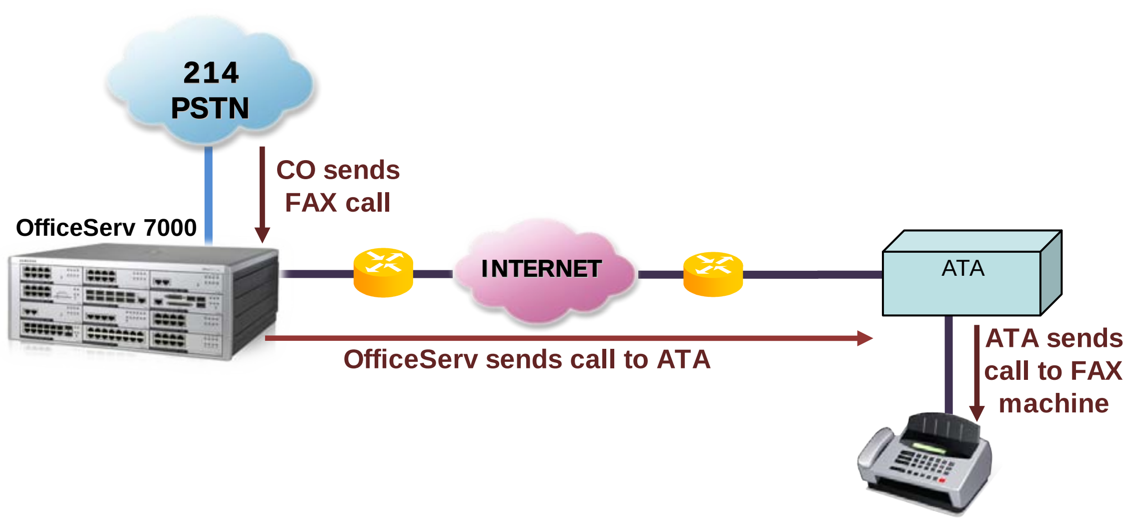
2.15 SIP CONNECTIVITY TO 3rd PARTY ATA DEVICES

An Analog Terminal Adaptor (ATA) is a SIP device that allows you to connect an analog device, such as a loop start trunk or a fax machine, to a SIP-compatible phone system. ATAs are used to circumvent the cable distance limits on analog devices at a remote location that need to connect back to the main office or headquarters. The OfficeServ 7000 series can be connected to 3rd party ATA for special applications such as remote IP Phones requiring local 911 emergency services, local 411 directory assistance, remote fax machines and/or analog phone deployment. This document outlines deployment of the following specialized applications:

1. Routing 911 calls from a remote IP Phone over SIP peering to a 3rd party ATA device at the same remote location.



2. Routing incoming fax calls from the OfficeServ over SIP Peering to a remote Fax station or Analog station using a 3rd party ATA via SIP.



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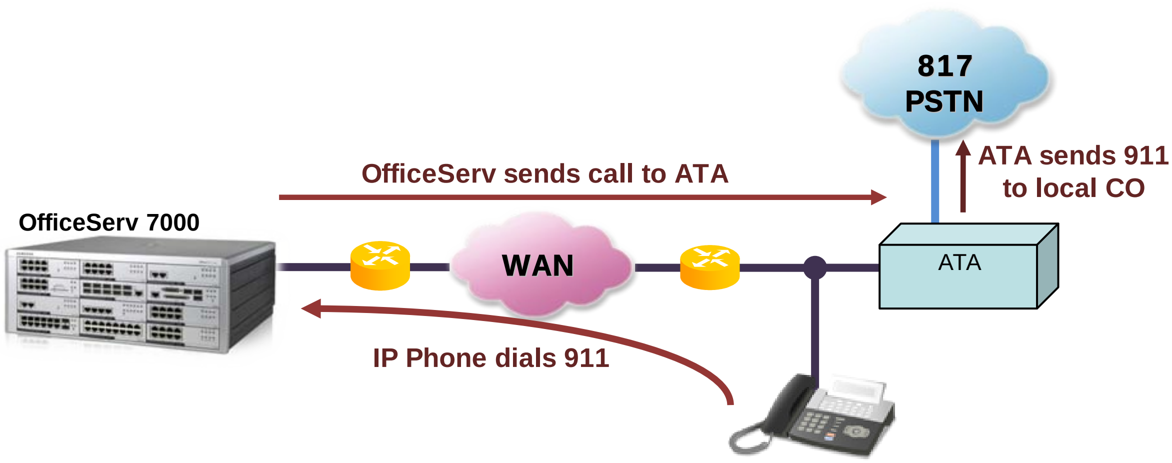
1. 911 Routing from remote IP Phone to remote 3rd Party SIP FXO Device

The first sample application deals with remote office IP phones. By default remote IP phones will dial out on trunks located at the main office. But when these remote workers dial 911 it is critical that emergency services be routed to the remote location. This challenge can be resolved in a few different ways:

1. If you have SIP trunking, many SIP service providers offer this as an additional service that can route 911 calls from the SIP server to the appropriate local exchange of the IP phone. This is the preferred solution. Contact your VoIP/SIP service provider to see if this service is available and get more details.
2. If SIP trunking is not in use, an additional OfficeServ 7000 Series system can be connected at the remote site, and can support the 911 routing to the local exchange via SPNet networking.
3. When the above options are not practical, a 3rd party ATA can be used as a SIP peer to the OfficeServ 7000 Series system to route 911 emergency calls to the local PSTN.



It is this third option that this application will focus on. To set up routing to the ATA we must proceed through 4 basic steps:

1. Program the OfficeServ to support SIP services.
2. Program the OfficeServ to support remote IP phones.
3. Program the ATA to register to the switch as a SIP Peer, and/or a Station device in some cases.*
4. Program the OfficeServ to route emergency calls from the remote IP phone to the ATA.

Due to the wide variety of ATAs in the market and the variety of programming options they have this document will describe, in very general terms, how to connect the ATA via SIP protocol to any of the OfficeServ 7000 series platforms. Consult the ATA manufacturer for specific programming needs.

1.1 OfficeServ Programming for SIP Services:

1. First, go to **MMC 727 (IT/DM 2.1.1)** and verify that the main software version of your OfficeServ 7000 Series system is **V4.30 or higher**.
2. Enter the **SIP LICENSE** key you have purchased for the system using **MMC 860 (IT/DM 2.1.4)**.
3. In **MMC 857 (IT/DM 6.3.2)** you will need to configure at least **1** virtual cabinet for **SIP Trunks** and **1** virtual cabinet for **SIP Stations**.
4. In **MMC 724 (IT/DM 2.8.0)** you will need to enter directory numbers for the SIP trunks and stations. For this example we will number SIP trunks **85XX** and SIP stations as **33XX**.

1.2 OfficeServ Programming for Remote IP Phones:*

1. Set up the private and public IP addresses for the OfficeServ 7000 Series system processor in **MMC 830 (IT/DM 2.1.0 and 2.1.2)**. For this example we will use **12.240.8.209** as the public IP.
2. Set up the private and public IP addresses for any **OAS**/**MGI16**/**MGI64** cards in **MMC 831 (IT/DM 2.2.2)**. ***(skip this step for OfficeServ 7030 systems)***
3. Set up the private and public port ranges for **MPS** channels (both embedded and those on **OAS** cards) in **MMC 843 (IT/DM 2.2.15)**.
4. In **MMC 840 (IT/DM 2.7.1)** set up user IDs and passcodes for each remote IP phone. For this example we will number IP phones as **89XX**.

****For more information on the interworkings of IP Phones, MGI channels, and MPS channels see section 2.10***

1.3 3rd Party ATA Setup:

The programming methods for ATAs will vary by make and model, but typically ATAs will be equipped with embedded web servers which allow you to use a web browser to connect and program the device.

There are 3 basic things you will need to set in the ATA:

1. The IP addressing (public and/or private) of the ATA

   For this example we will use the following settings:

   IP Address: **215.27.88.191**

   Gateway: **215.27.88.1**

   Subnet: **255.255.255.0**

2. SIP Station settings for the ATA (used to register the ATA to the OfficeServ 7000 Series system)

   For this example we will use the following settings:

   SIP Port: **5061**

   Proxy Registration: **12.240.8.209** (*the public IP of the OfficeServ MP*)

   Display Name: **3302**

   User ID: **3302**

   Password: **0000**

   Use Auth ID: **YES**

3. PSTN Line settings for the ATA (used to register the ATA's analog port to the OfficeServ)

   For this example we will use the following settings:

   SIP Port: **5060** (*cannot use the same port from step 2*)

   Proxy: **12.240.8.209**

   Display Name: **3303**

   User ID: **3303**

   Password: **0000**

   Use Auth ID: **YES**

   VoIP to PSTN Gateway Enable: **YES**

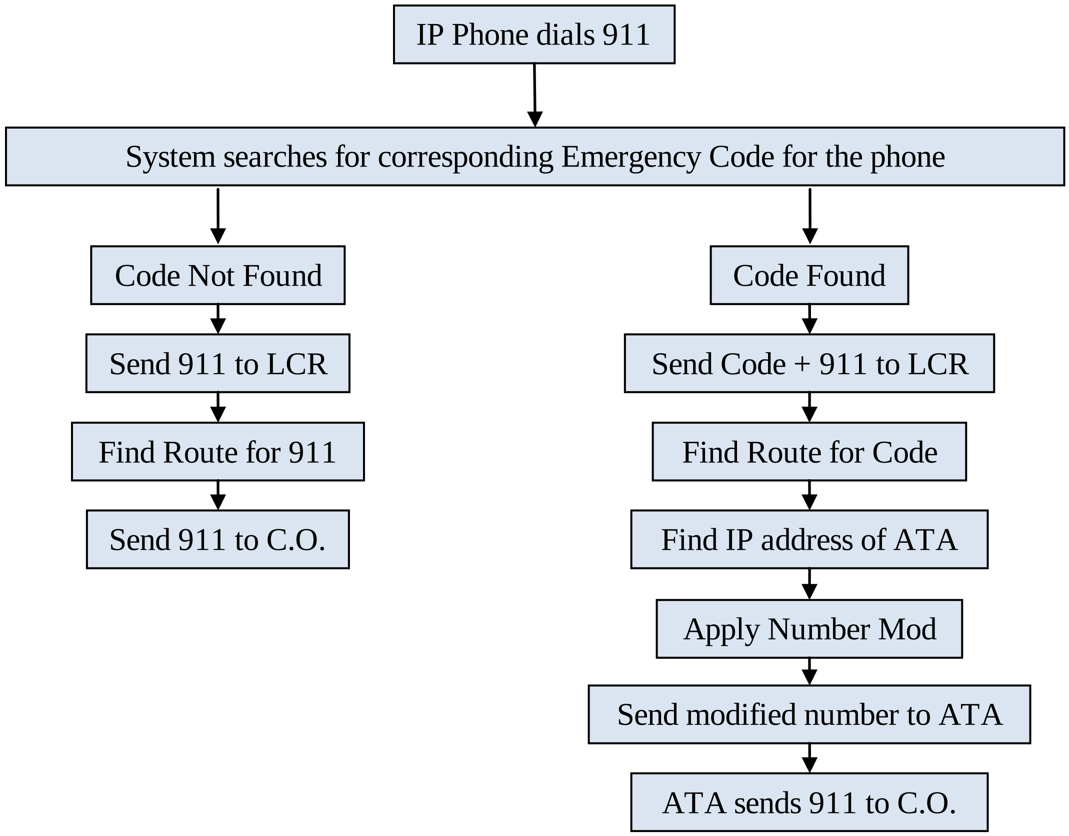
   One Stage Dialing: **YES**

After programming the applicable ATA settings check MMC 842 to verify that the device has registered to the system. You should see something similar to the following:

[3302]REGISTERED
YES

1.4 Programming Emergency Routing in the OfficeServ

Before diving in to the MMC programming take a moment to consider the flow of the call through the OfficeServ processor:



This call flow is a bit deviated from the standard LCR dialing you may be used to in the OfficeServ systems. A dedicated **MMC 330 (IT/DM 5.15.15)** exists that allows you to inject an Emergency Access Code when an IP phone dials a specific number, such as 911. When one of these dedicated numbers is dialed the system will insert the access code and send that to the LCR routing system instead, effectively allowing you to specify per-station LCR routing. This allows you to specify remote IP phone or phones that can access your ATAs analog trunk.

The programming steps are as follows:

1. In **MMC 210 (IT/DM 3.1.1)** set **LCR ENABLE** to **ON**
2. In **MMC 724 (IT/DM 2.8.0)** set the **LCR** Feature Code to **9**
3. In **MMC 603 (IT/DM 4.1.2)** set trunk group **805** to a **TYPE** of **SIP TRUNK** with members **8501 ~ 8504**
4. In **MMC 710 (IT/DM 3.1.2)** create **LCR DIGIT** of **4444** with a **LENGTH** of **07** and a **RT** of **01**
5. In **MMC 712 (IT/DM 3.1.4)** set the Group (**G**) for **LCR ROUTE (01:1)** to **805**

6. In **MMC 330 (IT/DM 5.15.15)** set **EMGY CODE** to **4444** for IP phone **8955** and set **EMGY DIAL** number **1** to **411** (***we use 411 here to avoid accidentally dialing 911!***)
7. In **MMC 832 (IT/DM 5.2.3)** select Access Code **O:000** and set **ACCESS DGT** to **4444**, **DGT LENGTH** to **7**, **DEL LENGTH** to **4**, and **SERVER USE** to **YES**
8. Finally, in **MMC 833 (IT/DM 5.2.17)** set the IP Address of **TB(000)IP ADDR 1** to **215.27.88.191** (***the ATA's IP address***)

Let's explore some of the programming settings you may not be familiar with, as they can be a bit overwhelming until you get some practice:

- In **step 4** notice that we are setting an LCR digit string of **4444**. This is the Emergency Access Code that we are assigning to the IP phone in **step 6**. It has a Length of **7** to account for the length of the Access Code plus whatever emergency number you want to dial (in this case, **4444**+**411** = **7** digits).
- **Step 6** is also where you set which dialed numbers you want the IP phone to route to the ATA. In this case we are just using **411**, but you can set up to **4** numbers or number prefixes that will use the ATA. This is useful not only for emergency dialing, but for making local calls as well (though the LCR table entries would need to be adjusted to compensate for the longer dialing length).
- In **step 7** you are using the Inbound/Outbound VoIP tables to set up the ATA connection. The leading **O** denotes that you are going to set up an Outbound access code. There are **251** outbound access code entries available. Typically you do not want to send the access code to the ATA, you simply want to dialed number sent. That is why we have set the **Digit Length** to **7** and the **Delete Length** to **4**.
- **Step 8** simply identifies the IP address of the device you want to send the calls to.

When extension **8955** dials **9-411** the Emergency Dial string **411** is recognized and the Emergency Access Code **4444** is inserted. **4444411** is then sent to the LCR system. A **7** digit number beginning with **4444** is sent to route **01**, which grabs trunk group **805**, our SIP trunk group.

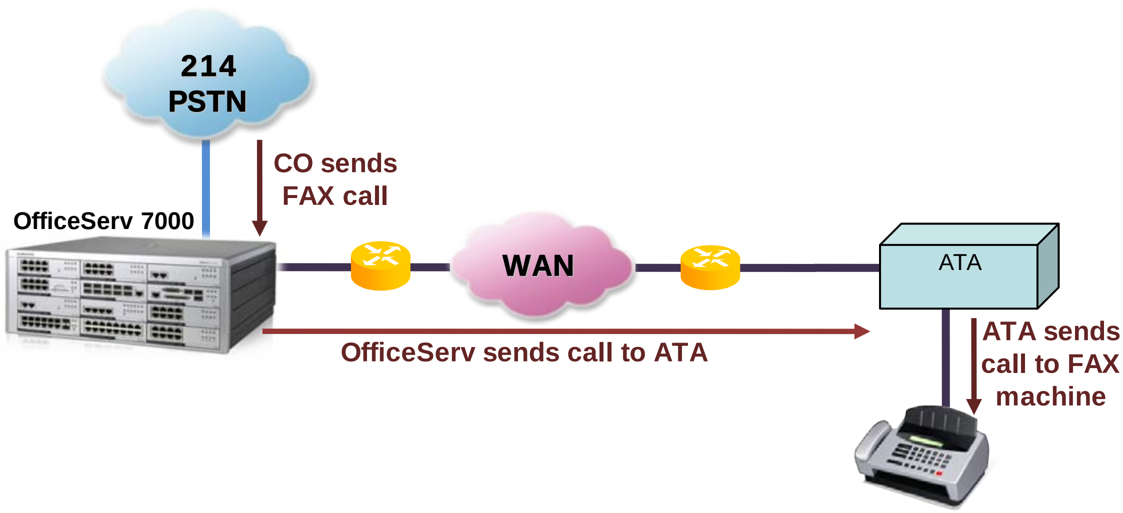
In order to properly route the call the OfficeServ will scan the Outbound Access Digits in **MMC 832** for a match. **4444** with a length of **7** is recognized, and the **4444** is stripped off. Digit string **411** is then passed on to **MMC 833** to find the IP address at table 000, entry 00 (**TB(000)IP ADDR 1**). The IP address of the ATA is returned and a SIP packet is generated requesting that the ATA dial **411** from its analog trunk port.

If the ATA is properly configured to accept inbound routing you may now go to IP phone **8955** and dial **9** to access LCR and then **411** to dial from the analog trunk of the ATA. Quite often it is necessary to perform some further adjustments in the ATA to get calls routing properly. In these cases you should consult the documentation that came with the device or the manufacturer of the device.

2. Routing incoming fax calls to a 3rd Party SIP FXS Device

The second sample application deals with remote FXS devices, such as analog phones or fax machines. When a fax comes in to the main office it is very straightforward to send the call to a fax machine connected to the OfficeServ 7000 Series system. But what do you do when you need to route the call to a remote office? This challenge can be resolved in a few different ways:

1. You could order a dedicated fax line at the remote office
2. You could install a second OfficeServ 7000 Series system at the remote office and use SPNet networking to route the call to a fax machine connected to the second system
3. When the above options are not cost effective, a 3rd party ATA can be used as a SIP peer to the OfficeServ 7000 Series system, allowing you to route calls to a fax machine connected to the ATA.



Again it is the third option that this application will focus on. To set up routing to the ATA we must proceed through 3 basic steps:

1. Program the OfficeServ to support SIP services.
2. Program the ATA to register to the switch as a SIP Station device.*
3. Program the OfficeServ to route fax calls from to the ATA.

Due to the wide variety of ATAs in the market and the variety of programming options they have this document will describe, in very general terms, how to connect the ATA via SIP protocol to any of the OfficeServ 7000 series platforms. Consult the ATA manufacturer for specific programming needs.

2.1 OfficeServ Programming for SIP Services:

1. First, go to **MMC 727 (IT/DM 2.1.1)** and verify that the main software version of your OfficeServ 7000 Series system is **V4.30 or higher**.
2. Enter the **SIP LICENSE** key you have purchased for the system using **MMC 860 (IT/DM 2.1.4)**.
3. In **MMC 857 (IT/DM 6.3.2)** you will need to configure at least **1** virtual cabinet for **SIP Trunks** and **1** virtual cabinet for **SIP Stations**.
4. In **MMC 724 (IT/DM 2.8.0)** you will need to enter directory numbers for the SIP trunks and stations. For this example we will number SIP trunks **85XX** and SIP stations as **33XX**.

2.2 3rd Party ATA Setup:

The programming methods for ATAs will vary by make and model, but typically ATAs will be equipped with embedded web servers which allow you to use a web browser to connect and program the device.

There are 3 basic things you will need to set in the ATA:

1. The IP addressing (public and/or private) of the ATA

   For this example we will use the following settings:

   IP Address: 215.27.88.191

   Gateway: 215.27.88.1

   Subnet: 255.255.255.0

2. SIP Station settings for the ATA (used to register the ATA to the OfficeServ 7000 Series system)

   For this example we will use the following settings:

   SIP Port: 5061

   Proxy Registration: 12.240.8.209 (*the public IP of the OfficeServ MP*)

   Display Name: 3302

   User ID: 3302

   Password: 0000

   Use Auth ID: YES

After programming the applicable ATA settings check MMC 842 to verify that the device has registered to the system. You should see something similar to the following:

[3302]REGISTERED

YES

2.3 OfficeServ Call Routing

The final step in setting up this sample application is to route the OfficeServ calls to the ATA. This is a simple matter of routing the specific trunk or DID to the ATA's extension number, **3302**, for the desired Ring Plan(s). In the case of SIP or PRI trunks this is done in **MMC 714 (IT/DM 3.2.3)**. In the case of T1 or Loop Start Analog trunks this is done in **MMC 406 (IT/DM 3.2.1)**.

In cases where faxes aren't necessarily coming in on a dedicated number or line you can use the Samsung Voicemail to answer calls and selectively route just the fax calls to the ATA. For more information on this functionality see the programming manual for your specific Samsung Voicemail product.

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2.16 CONFERENCING and CNF24 CARD

CNF24 (CONFERENCE CARD)

This optional application card provides 24 conferences channels that can be individually assigned to either **Meet-Me** or **Ad Hoc** conferences, but not both. The application program and related database are stored in memory on the card. The CNF24 can be installed in any universal card slot that has 24 channels on the slot. Note: If the CNF24 card is installed on a slot that supports more than 24 channels, all 24 conference channels are used. If installed on a 16 channel slot, only 16 conference channels are supported. The OfficeServ 7200-S system can have a maximum of one CNF24 card for a total of 24 conference channels. A conference cannot be split between cards. Only outside callers on PRI or SIP trunks and internal stations can access the conferences on the CNF24. System must be running software version V4.53 or higher to use the CNF24 features.

Meet Me Conference

Using the optional CNF24 card users can host a meet –me conference of up to 24 members maximum or multiple smaller conferences with less attendees. System software version V4.53 or higher comes with an embedded web server. Users log in to the OfficeServ Conference Scheduler to schedule and manage their personal conferences. There are options to set the conference ID, select the attendees, either internal users or external people, schedule for once, daily or weekly, set for early entrance, deliver invitations by email, include instructions and comments and page internal users to remind them of a conference that is about to start. The conference can be recorded and saved as .wav file and then moved to your PC or server like any other file for later review or archive.

During the meet-me conference the Host screen shows who is In, Not In or has Exited, Caller ID, and member ID if entered. Host has options to *Remove* or *Mute* any attendee as well as start or stop recording and terminate the conference. Internal attendees can join the conference using the MJOIN button on their telephone as an alternative to using outside telephone lines. The conference can be locked to prevent additional users from joining.

Ad-Hoc

Using the optional CNF24 card, users can set up an Ad-Hoc conference with up to 24 parties (yourself and 23 others). The maximum number is determined by the number of channels dedicated to the Ad-Hoc conference feature. The parties can be internal stations or outside calls. The Ad-Hoc conference works similar to the OfficeServ Add-on conference but is not limited to 5 parties. Users must have the MCONF button to initiate an Ad-Hoc conference.

For more information on how to install and program the CNF24 card on a Samsung OfficeServ 7200, 7200-S, or 7400 please refer to the CNF24 Technical Manual.

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OfficeServ™ 7000 Series

SIP Trunking

Applications & Best Practices

V1.0

SAMSUNG

Publication Information

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1. Introduction

This document describes the required steps to configure the OfficeServ 7000 series system for the SIP trunking service.

SIP trunking is a real-time IP communications service delivered by an Internet Telephony Service Provider (ITSP) or SIP trunking provider. The SIP trunking service routes inbound, outbound voice, and fax traffic over a broadband data connection by using Session Initiation Protocol (SIP) for signaling and Real-Time Transport Protocol (RTP) for traffic.

SIP services can offer many benefits, SIP trunking services typically allow flexible provisioning, customers can adjust network the capacity on an as-needed basis, changes can be implemented remotely eliminating the need to schedule a site visit (talk to your SIP service provider for details).

2. Deployment Considerations

There are three components necessary to successfully deploy SIP trunks: a SIP-enabled PBX, an SIP-enabled edge route, and an ITSP.

Deploying a SIP trunking service for the first time may be challenging. The following areas need to be considered carefully.

2.1. OfficeServ 7000

Only OfficeServ 7000 series systems with latest the software version can support SIP trunking service. OfficeServ 100 and 500 systems do **not** support SIP trunking service.

2.1.1. Components

The OfficeServ 7000 system must have the following components:

- SIP trunking license
- Sufficient MGI and MPS channels
- Latest software version on OAS or MGI16/64 card
- MP software version v4.22 and up
- If IP phones are used, make sure they are loaded with the latest software version

2.1.2. Technical Knowledge

The following technical certifications provide the basic knowledge to configure the system and to obtain the technical support from Samsung.

- OfficeServ 7000 certification
- ITP/SMT-i phone certification
- SIP trunk certification

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2.1.3. Media Gateway Interface (MGI)/Media Proxy Service (MPS) Resource Planning

The following tables shows the resources requirement for MGI/MPS channels. Please provide sufficient resources for the system to operate properly.

MPS/ MGI Channel Usage on Call Conversation

	SIP Trunk	PRI/Analog Trunk	SPNET[1]/ SIP Peering (one system)	H323 Trunk	Remote IP Phone	Local IP Phone	Voice Mail
Local IP Phone (ITP or SMT-i or SMT-W, or 3rd Party)	2 MPS[2] or 2 MGI channels	1 MGI channel	2 MPS or 2 MGI channels	2 MGI channels	2 MPS or 2 MGI channels	0	1 MGI channel
Remote IP Phone (ITP or SMT-i or SMT-W, or 3rd Party)	2 MPS or 2 MGI channels	1 MGI channel	2 MPS or 2 MGI channels	2 MGI channels	2 MPS or 2 MGI channels	2 MPS or 2 MGI channels	1 MGI channel
Non-IP Phone (TDM phone, or Analog Phone, Fax Machine, or SVMi)	1 MGI channel	0	1 MGI channel	1 MGI channel	1 MGI channel	1 MGI channel	0

Notes:

1. For SPNET to use MPS channels, one need to enable MPS on both MMC 861 (MPS SERVICE: ON) and MMC 820 (NO MGI:ON).
2. **<u>Two MPS channels equivalent to one MPS call</u>**. Every MPS connection always requires two MPS channels or one MPS call.

MGI Channel Usage on Trunk Ringing State

	Local IP Phone	Remote IP Phone	Non-IP IP hone
IP Trunk (SIP, SPNet[1], H.323)	2 MGI chs	2 MGI chs	1 MGI ch
PSTN Trunk (Analog & PRI)	1 MGI ch	1 MGI ch	0 MGI ch
Local IP Phone	0 MGI ch	0 MGI ch	1 MGI ch
Remote IP Phone	0 MGI ch	0 MGI ch	1 MGI ch
Non-IP Phone (TDM, Analog, FAX)	1 MGI ch	1 MGI ch	0 MGI ch

Note: 1. If MPS is enabled for SPNET, it will take only one MGI channel in the ringing state.

If there is no MGI channels to answer the call, SIP trunk caller will receive the busy tone and intercom phone caller will display “No MGI chs are available”.

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MGI Channel Usage on Background Music State

Local IP Phone	Remote IP Phone	Non-IP IP Phone
1 MGI ch	1 MGI ch	0 MGI ch

MGI Channel Usage on Music-on-Hold State

	Local IP Phone	Remote IP Phone	Non-IP IP Phone
IP Trunk (SIP, SPNet, H.323)	1 MGI ch	1 MGI ch	1 MGI ch
PSTN Trunk (Analog & PRI)	0 MGI ch	0 MGI ch	0 MGI ch

Paging

Local IP Phone	Remote IP Phone	Non-IP IP Phone
1 MGI ch	1 MGI ch	0 MGI ch

Access to Voice Mail

Local IP Phone	Remote IP Phone	Non-IP IP Phone	IP Trunk	PSTN Trunk
1 MGI ch	1 MGI ch	0 MGI ch	1 MGI ch	0 MGI ch

Access to IP-UMS

Local IP Phone	Remote IP Phone	Non-IP IP Phone	IP Trunk	PSTN Trunk
2 MGI chs	2 MGI chs	1 MGI ch	2 MGI chs	1 MGI ch

2.1.3.1. MGI/MPS Usage Guidelines

MGI channel is used to convert the digital bit streams to IP packets and vice versa. MGI channel can also be used to convert two different format of IP packets. MPS channels are only used to convert different format of IP packets. Normally, MPS channels are used on communications of IP phones to external SIP trunks, or IP networking, or remote IP phones.

- SIP Trunking/Peering
  - Outgoing calls
    - From IP phone
      - Hold/Transfer/Conference always use MGI channel whenever switching channel from MPS to MGI and vice versa.
      - If there is no MGI at Hold, it goes to Hold state but no music.
    - From non-IP phone
      - Use one MGI channel and stays the same channel till the end of the call.
  - Incoming calls
    - To IP phone
      - In ringing state, two MGI channels are assigned by default. In conversation state, MGI channels are released and MPS channels are used. If there is no MPS at this point, OfficeServ will continue using the MGI channels.
    - To non-IP phone
      - Use one MGI channel and stays the same channel till the end of the call.
- H.323
  - H.323 does NOT use MPS channel.
- Intercom calls
  - IP phone to non-IP phone: one MGI is needed.
  - IP phone to remote IP phone (IP phone behind NAT): use two MPS channels or two MGI channels.
  - IP phone to IP phone: no MPS or MGI is needed.
- Paging
  - Paging always uses MGI channel.
  - Call from IP phone: 1 MGI is needed.
  - Call from non-IP phone: no MGI is needed.
  - To non-IP phone: no MGI is needed.
  - To IP phone: 1 MGI per phone is needed.
- If there is no MGI or MPS resource available in the following cases,
  - OfficeServ will reject the SIP trunk inbound call. ITSP will provide busy tone to the caller.
  - If not enough MGIs for all paging members, the bottom of the members will get dropped.
  - There will be no music when the call is put on hold (from MPS to MGI)
  - Can't resume the held call

2.2. Broadband Internet and Quality of Service

SIP trunking service requires a broadband data connection that is provided by an Internet Service Provider (ISP). ISP and ITSP can be the same company or different companies. Because SIP trunking is delivered over the same IP connection for voice, data and fax communications, it can yield significant cost savings. On the other hand, it can yield significant challenges too.

The bursty nature of the data traffic will affect voice communications. Planning for adequate data bandwidth for voice is necessary and required. Voice over IP communications requires equal bandwidth for both downstream and upstream. The numbers of simultaneous voice calls are depended on customer's Internet connection. If an asymmetric connection is used (such as DSL), the upstream speed will be the limiting factor.

One of the most important factors to consider for the VoIP application is proper capacity planning. Bandwidth calculation is an important factor to consider for capacity planning. Different codecs require different bandwidth for packet voice calls.

For example, G.711 codec with 20 ms frame count will require approximately 87.2 Kbps while G.729a will only require 31.2 Kbps. At the same time, G.711 provides better voice quality than G.729a. MMC 835 is used to select the codec type.

Adequate network "Quality of Service" (QoS) is also required for the voice over IP service. Check with your ISP/ITSP for their QoS policy. Some ITSPs own their private network for the SIP trunking service. Some use the public Internet for the SIP trunking service.
In general, VoIP network requires to exceed the following parameters:

Packet loss:

- Requires less than 1% loss rate
- Latency (delay)
    - Targeted for 150 ms or less
- Jitter
    - Requires less than 1 ms
- MOS (Mean Opinion Score)
    - This is subjective measurement. MOS score can be improved by using different CODEC. For example, G.711 has MOS score of 4.4 and G.729A has 3.7. Therefore, G.711 offers better audio quality than G.729. However, G.711 requires more bandwidth than G.729.

Mean Opinion Score (MOS)		
MOS	**Quality**	**Impairment**
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

2.3. ITSP Interoperability

There are hundreds of ITSP/SIP trunking providers in North America. Some offer services in a local region and some offer services to national level. Different vendors implement different SIP components and extensions. Both SIP trunking providers and equipment providers are constantly adding new features to respond to the market requirements. Conducting certification or interoperability tests can be costly and time consuming.

The SIP Forum is the industry association that generates the industry-wide technical recommendations related to SIP interoperability. For more detail info, see SIP Forum's website, http://www.sipforum.org/.

Samsung is a member of the SIP Forum and OfficeServ 7000 systems comply with SIP Forum's standard for interoperability. Any member of the SIP Forum will comply to the same standard. Also, OfficeServ 7000 systems contain rich SIP trunking options to meet most ITSPs' services. Samsung has not encountered any issue with any ITSP for the basic voice services.
Samsung is also continuously adding new features to enhance the SIP trunking service. Please watch for product bulletins and technical bulletins for update. For a list of certified ITSPs for OfficeServ systems, please refer to Samsung GSBN website.

2.4. Fax Over IP

OfficeServ 7000 supports Fax transmission via G.711 or T.38 protocol over SIP trunking. However, due to inhomogeneous hardware standards among fax machines and the quality of service over the end-to-end IP network, fax is inherently unreliable. The reliability and functionality of Fax cannot be guaranteed. With the use of lower rate Fax machines and T.38 protocol, the chance of getting fax service will be increased. **It is not recommended to use Fax over IP for mission critical application.**

2.5. The Edge Router/Firewall

SIP is an application layer protocol that sends network addresses and port allocations within the application data. It usually uses many network ports to set up and maintain the voice communications. For example, SIP uses port 5060 to set up the call and use different ports (MGI/MPS) for RTP audio stream.

Most of enterprise networks also maintain an internal network of private addresses and map them to a single public address. Network Address Translation (NAT) translate between the IP addresses in the private network and the public addresses on the Internet. MMC 830 is used to set up the public and private address to the OfficeServ system.

The network edge device or router/firewall will need to be configured to allow passing packets to the OfficeServ system.

The following ports are the default settings for OfficeServ systems.

Service Type	Port Number	Protocol	Remarks
SIP	5060	TCP, UDP	MMC 837
SPNet	6100	TCP	
IP Phone	6000	TCP, UDP	
IP Phone	9000	UDP	

Service Type	Port Number	Protocol	Remarks
H.323	1720 (tunneling mode) 1024 to 4999 (non tunneling mode)	TCP TCP	If it uses tunneling mode, it only uses port 1720. If not, it can use additional port randomly chosen between 1024 to 4999.
MGI 16/64	30000 ~ 30xxx	UDP (RTP)	MMC 831
MPS	40000 ~ 400xx	UDP (RTP)	MMC 843 If external OAS cards are used, different IP addresses and port numbers need to be assigned for each card.
SMT-W Wi-Fi phone Remote Location	8000	TCP	
ITP or SMT-i phone Remote Location	6000 and 9000	UDP	MMC 840

3. Configuration

3.1. Overview

Steps to set up SIP trunking:

- Enter SIP license keys in MMC 860.
- Enter SIP account data and enable the "SIP SERVER" in MMC 837
  - Verify basic incoming call
  - Verify outgoing call
- Ensure caller ID is displayed properly.
- Use MMC 832 to strip out unwanted digits such as "+" sign.
- Enter the DID to the "send CLI number" corresponding to the stations.
  - Verify DTMF setting in MMC 861.
- Set up supplementary services in MMC 837 if offers.

3.2. SIP License

SIP trunking requires to have the SIP trunk license keys to operate.
The license key is resided in MMC 860.

3.2.1. Verification

The default SIP trunk group is 805 for OfficeServ 7100/7200/7400 and 835 for OfficeServ 7030. If there is dial tone when dialing the SIP trunk group, the SIP trunk license is good.

3.3. Inbound Call

Incoming call is the first item to verify after setting up the SIP account in MMC 837. If one can receive the incoming call the SIP account is in good standing. Outbound call requires several more configuration.

3.4. Domain Name System (DNS) Record

Service records (SRV) are a form of DNS (Domain Name System) record. They contain information about where to send requests for a particular service offered at a specific domain. All OfficeServ 7000 systems except 7200 MCP support DNS.

3.4.1. SIP Trunking Account Configuration

The OfficeServ 7000 series offers four ITSP (Internet Telephone Service Provider) profiles. The current software allows one profile to be activated at one time.

Typically, ITSP uses two types of authentication methods to authenticate the SIP trunk account: static type or registration based type. Static type authenticates an account based on the static IP address of the OfficeServ system. Registration based type authenticates an account based on the username and password. It doesn't check the IP address of the OfficeServ system.

The SIP account information is given by the ITSP. The following fields need to be filled based on the information given by the ITSP.

To activate the service, the ISP must be set to "Enable" in MMC 837.

Field with RED letter are the must have areas. Others can be left as default.

IT/DM 5.2.13 [MMC 837]	**Registration Base Account**	**Static IP Based Account**	**Remark**
Item	**Value**		
SIP Carrier Name [SIP CARRIER]	(user defined)	(user defined)	As a reminder, information is not used.
SIP Server Enable [SIP SERVER]	Enable	Enable	Need to enable the selected profile
SIP Service Available [SVR AVAIL]	Yes	Yes	Status indication
Registra Address [REGIST ADDR]	(if supplied by ITSP)	N/A	Optional field
Registra Port [REGIST PORT]	5060 (05060)	5060 (05060)	Default is 5060. No need to change it unless instructed by ITSP.
Outbound Proxy [OUT PROXY]	Must Have	Must Have	Can be domain name or IP address
Alternative Outbound Proxy [ALTER PROXY]	0.0.0.0	0.0.0.0	Used only if instructed by ITSP
Outbound Proxy Port (PROXY PORT)	5060 (05060)	5060 (05060)	Default is 5060. No need to change it unless instructed by ITSP.
Proxy Domain Name [PROXY NAME]			Used only if instructed by ITSP. Sometimes called Realm. See 3.9 for detail.
Local Domain Name [LOCAL NAME]			Not used. See 3.9 for detail.
DNS Server 1 [DNS SERVER1]	Must Have (if uses domain name in proxy)	Must Have (if uses domain name in proxy)	Either provided by ITSP or your ISP

IT/DM 5.2.13 [MMC 837]	Registration Base Account	Static IP Based Account	Remark
Item	**Value**		
DNS Server 2 [DNS SERVER2]	Optional	Optional	
User Name [USER NAME]	Must Have (phone number or user name)	N/A	Usually same as auth username
Auth Username [AUTH USER]	Must Have (phone number or user name)	N/A	Sometimes called username or user name
Auth Password [AUTH PSWD]	Must Have (password)	N/A	Supplied by ITSP
Regist. Per Use [REG PER USR]	Disable	Disable	Not used①.
Session Timer [SESSION TMR]	NONE	NONE	No need to change it unless instructed by ITSP.
Session Expire Time [SESSION EXP]	1800 (001800)	1800 (001800)	No need to change it unless instructed by ITSP.
Trunk Reg Expire Time [TRK REG EXP]	1800 (001800)	1800 (001800)	No need to change it unless instructed by ITSP.
Alive Notify [ALIVE NOTI.]	NONE	OPTIONS	Options message is used to check the connectivity if required.
Alive Notify Time [NOTIFY TIME]	1800 (001800)	1800 (001800)	Interval to send OPTIONS message. Unit is in second.
IMS Option [IMS OPTION]	DISABLE	DISABLE	Enable only if need to send IMS header information
P Asserted ID Use [ASSERTED ID]	NONE	NONE	See 3.8 for detail
Privacy [PRIVACY]	DISABLE	DISABLE	See 3.8 for detail
SIP Peering [SIP PEERING]	DISABLE	DISABLE	Set the FROM field's address as the OS IP address (ENABLE) or the OUT PROXY address (DISABLE).
Send CLI Table [CLI TABLE]	1	1	Set which of the CLI table from MMC 321 to use for outbound calls.
Supplementary Type [SS TYPE]	SERVER MANAGED	SERVER MANAGED	See 3.6 for detail
302 Response [302 RESP]	DISABLE	DISABLE	See 3.7 for detail
SIP Destination Type	TO HEADER	TO HEADER	Set the field to use as the inbound number for incoming call

IT/DM 5.2.13 [MMC 837]	Registration Base Account	Static IP Based Account	Remark
Item	**Value**		
[DEST TYPE]			routing. TO HEADER use the address in the TO header. REQ URI uses the request URI address.
Codec Auto Nego [CODEC NEGO]	ENABLE	ENABLE	Should always ENABLE for auto CODEC negotiation.
HOLD Reinvite [HOLD RE-INV]	ENABLE	ENABLE	Enable to send re-invite message when on hold. This option is available only when MPS channel is disabled.
URI Type [URI TYPE]	SIP	SIP	Determines the Universal Resource Identifier Type for the connection.
SIP Signal Type [SIGNAL TYPE]	UDP	UDP	Set the TCP/IP signaling type to use when communicating with ITSP.
E.164 Support [E.164 ENABL]	DISABLE	DISABLE	If enable, prefix '+' for outbound call.
PRACK Support [PRACK]	DISABLE	DISABLE	Set to send Provisional acknowledge (PRACK) messages if required by ITSP.
Hold Mode [HOLD MODE]	Send Only	Send Only	See 3.6 for detail

① Please refer to "SIP Trunking & Peering Operation Guide" and "OfficeServ Programming Manual".

3.4.2. Alive Notification

For ITSPs that require login registration (i.e. requires user name and password), the OfficeServ 7000 system will re-register itself periodically based on the timer given by ITSPs to ensure the service is available. Do not enable Alive Notify feature for this case.

If the ITSP does not require login registration (i.e. use static IP address for authentication), the administrator needs to set up the Alive Notification option for the OfficeServ 7000 system to check the availability of SIP service periodically.

The OfficeServ 7000 series uses the **SVC AVAIL** field in MMC 837 to indicate the availability of the SIP service. If SVC AVAIL shows NO, it indicates that the SIP services is not available.

- MMC 837 is used to set up Alive Notification.
- Alive Notify: Options

When OPTIONS is set, the OfficeServ 7000 system will periodically send a message to ITSP to query the available options. If the OfficeServ 7000 system receives a response, the SIP service is alive.

3.4.3. Verification

If operator station can receive the incoming call, the SIP account is good. The call will go to operator group unless MMC 714 is programmed to direct the DID to a specific station.

3.4.4. Caller ID

The OfficeServ 7000 series supports both standard and E.164 international format.

3.4.5. Inbound E.164 Format

E.164 is the recognized international standard format for inbound and outbound calling. It is characterized by a “+” followed by a country code (i.e. “1” U.S. and Canada) and phone number.

OfficeServ 7000 systems support both inbound E.164 format and outbound E.164 (v4.30 and up).

If ITSP uses this format, the MMC 832 needs to be used to strip out the “+” sign for inbound call. Otherwise, no caller ID will be displayed on the phone.

For example:

Inbound caller ID:

Local & long distance: +19728896700
International: +4402074941234

OfficeServ System:

To delete “+1” from the inbound digits in MMC 832:

(I:00) ACCESS DGT

+1

(I:00) DGT LENGTH

2

(I:00) DEL.LENGTH

2

3.4.6. Direct DID to a Specific Station

Incoming call can be directed to a specific station. MMC 714 is used to match the DID number and redirect it to a station based on the ring plan.

For example, one needs to direct the DID 972-555-1234 to extension 3201.

MMC 714
DID DIGIT (001) DGT:9725551234
1:3201 2:3201
3:3201 4:3201
5:3201 6:3201

3.4.7. Inbound SIP Trunk Fails Over to PSTN Trunk

This is ITSP feature. Most of the ITSP allows calls to be forwarded to other PSTN or cellular phone number in case of failure on the Internet connection.

3.5. Outbound Call

Most of the ITSPs require to send caller ID for outbound call. The caller ID needs to be set for each individual station. It can be set from MMC 321. The MMC 321 must use the same CLIP table index as set in the MMC 837.

If LCR is used in MMC 832, the “SERVER USE” field must be set to “YES” for SIP trunking.

3.5.1. Verification

Dial 8501 (default SIP trunk access) and the out digits, the outbound call should be able to go through it.

3.5.2. Outbound E.164 Format

If ITSP requires the outbound call ID comply with E.164 format, i.e. prefix with “+”, the “E.164” item in MMC 837 must be enabled.

Home Page

3.5.3. DTMF

The MGI card of OfficeServ 7000 series supports three types of methods to handle DTMF in MMC 861 for SIP trunking. The type of DTMF setting must be matched between the OfficeServ system and ITSP. The DTMF type will be provided by the ITSP.

- **Inband (RFC2833)**: most popular type
- Inband (in voice)
- **Outband**: use SIP info message, DTMF is generated by the ITSP.

3.5.4. Outbound SIP Trunk Fails Over to PSTN Trunk

If the SIP trunk and PSTN trunk are both available in the OfficeServ 7000 system, the administrator can program LCR to use the PSTN trunk if the SIP server or Internet is out of service.

The following provides a quick example for programming LCR for SIP trunk fails over to PSTN trunk, if one analog trunk 701 is available for back up:

Set trunk group 801 as the analog trunk in **MMC 603**

[801] TRK GROUP, MEMBER 01 -> 701

Trunk group 805 is the default SIP trunk group

Set LCR digit in **MMC 724**

FEAT NUMBER PLAN, LCR: 9 (to use digit 9 for LCR)

Set dialed digits for each prefix in **MMC 710**

LCR DIGIT (0001), DIGIT:1, LENGTH:11 (digits), RT (route table):1

LCR DIGIT (0001), DIGIT:9, LENGTH:10 (digits), RT (route table):1
Set LCR trunk group order in **MMC 712**

LCR ROUTE (01:1), C:1 (class), G:805 (SIP trunk first), M:--

LCR ROUTE (01:1), C:2 (class), G:801 (analog trunk 2nd), M:--

Set all stations to have two LCR class in **MMC 310**

[ALL] LCR CLASS, LCR CLASS:2

Enable LCR class in **MMC 210**

TEN. ON AND OFF, LCR ENABLE: ON

3.6. External Call Transfer and Hold Re-Invite

The OfficeServ 7000 series supports two types of methods to handle SIP semi-blind transfer and consultation transfer feature. It is important to select the correct type to match services from the ITSP. The selection of SS (Supplementary Service) Type in MMC 837 will determine how the external call transfer is handled.

SS Type	Usage	Comments	SIP Protocol	
			Transfer	Hold/Resume
SERVER Managed	ITSP handles connection on trunk-to-trunk call transfer. Preferred method if ITSP can support REFER message.	If user hangs up the call after transferring an inbound call to an outbound call, ITSP maintains the connection on two external calls. OfficeServ releases all SIP trunk associated resources.	OfficeServ sends REFER message to ITSP.	OfficeServ sends Re-Invite message to ITSP. OfficeServ uses one MGI for music on hold.
PBX Managed 2	OfficeServ handles connection on trunk-to-trunk call transfer. Normal application.	If user hangs up the call after transferring an inbound call to an outbound call, OfficeServ maintains the connection on two external calls by using two SIP trunks resources.	OfficeServ sends Re-Invite message to ITSP. OfficeServ uses two MGIs or 1 MPS call for connecting two external calls.	OfficeServ sends Re-Invite message to ITSP. OfficeServ uses one MGI for music on hold.
PBX Managed 1	OfficeServ handles connection on trunk-to-trunk call transfer. Special application for UK.	If user hangs up the call after transferring an inbound call to an outbound call, OfficeServ will maintain the connection on two external calls by using two SIP trunks resources.	OfficeServ sends Re-Invite message to ITSP. OfficeServ always uses two MGIs for connecting two external calls even MPS is enabled.	OfficeServ sends Re-Invite message to ITSP. OfficeServ uses one MGI for music on hold.

If SS type is set to "PBX Managed 1"/"PBX Managed 2" and MPS is enabled, ITSP's "Auto NAT detection" feature should be turned off. Otherwise, no audio can happen after external transfer.

When a user presses TRANSFER key, OfficeServ will first put call on hold. The media resource is switched from MPS to MGI. When two external calls are connected, OfficeServ will switch back to MPS connection. Some ITSPs with "Auto NAT" will not transit RTP stream to MPS even OfficeServ tells them to move to MPS. They want to receive RTP stream first before switching over. This issue can be prevented if "Auto NAT Detection" is turned off.

3.6.1. MOH Function

1. SIP Providers have the right to override the OfficeServ Media request when playing MOH. This causes confusion. Who is controlling MOH? A new setting since V4.53 has been added to give more options to deal with various SIP providers. OfficeServ has a new setting “**Hold Mode**” added to MMC 837 or IT/DM 5.2.13. It has the following characteristics:

    a. SENDONLY: Request to SIP Provider to play OfficeServ MOH (Provider can reply with inactive message and play their MOH to the distant party to save bandwidth).

    b. INACTIVE: Request to SIP Provider to play SIP Provider MOH.

    c. SENDRECV: Play MOH from OfficeServ. Provider sees this as two way conversation so they pass the MOH.

    d. This setting only applies when HOLD REINVITE is enabled in MMC 837 or IT/DM 5.2.13 when using MGI channels. When MPS is used HOLD REINVITE is always sent.

3.7. Call Forward

3.7.1. Supported by ITSP

If ITSP supports SIP call forward feature, the “**302 Response**” needs to be set to “**Enable**” in MMC 837. In this case, no resource from the OfficeServ 7000 system will be used.

3.7.2. Not Supported By ITSP

If ITSP doesn’t support SIP call forward feature, the OfficeServ 7000 system can be used to handle call forward features. Two OfficeServ 7000 SIP trunks will be used: one for inbound and one for outbound.
In this case, the external forward feature in MMC 400 must be enabled.

In MMC 840:

[8501] TRK ON/OFF

EFWD EXT CLI: ON

3.8. Privacy

These privacy settings are for system wide. There is no per call basic privacy setting.

3.8.1. Privacy Setting

The OfficeServ 7000 systems allow users to withhold the caller ID information when making a outbound call.

The "**PRIVACY**" setting in MMC 837 can be enabled to offer this feature. In this case, the OfficeServ 7000 system will use anonymous rather than the telephone number in the INVITE message.

3.8.2. P-Asserted-ID

Providing privacy in a SIP network is more complicated than in the PSTN. In SIP networks, the participants in a session are typically able to exchange IP traffic directly without involving any SIP service provider. P-Asserted-ID is used to address the problem of network asserted identification. Refer to RFC 3325 for more details.

If ITSP requires to use P-Asserted-ID, the "**ASSERTED ID**" field in MMC 837 must be selected.

ASSERTED ID:

- None: the field is not used
- PRIMARY: P-ASSERTED-ID header contains primary number and FROM header contains secondary number.
- ALTERNATE: P-ASSERTED-ID header contains secondary number and FROM header contains primary number.

3.9. Proxy Domain Name/Local Domain Name

A SIP message includes "from" and "to' headers. In these fields, there are usually telephone numbers with @domain-name. For example, 2145671234@samsung.com. Some carriers may want OS-system to send different domain names for "from" and "to" headers. For example, in the "to" header, there should be 2145679876@somecarrier.com which might be carrier's domain name, and in the "from" header, 2135671234@samsung.com which should be any domain name but carrier's domain name. For this example, you should enter somecarrier.com in proxy domain name and samsung.com in local domain name. If you want the same domain names for "from" and "to" headers, then you can leave blank in the local domain name.

3.10. FAX over IP

The OfficeServ 7000 systems support both T.38 and G.711 for FAX over IP network. Please check with the ITSP for the correct setting for FAX service.

To set T.38:

MMC 835

MGI16:T38 FAX USE

ENABLE

To set G.711:

MMC 835

MGI16:T38 FAX USE

DISABLE

3.11. Carrier EXCUSIVE Field

Enabling this option will allow OfficeServ to accept calls only from the IP addresses that were in the active SIP trunk provider (MMC 837) and SIP peers (MMC 833). It can prevent any unauthorized SIP peer access such as calls with “Asterisk” caller ID.

Sometimes, phones are ringing with “Asterisk” caller ID on the display. When one picks up the call, no one is on the other end. It is one of the scenarios of hackers trying to gain access to the phone systems through SIP peering connection.

Once enabling the SIP trunk/stack license, the SIP peering connection is automatically opened. If hackers obtains the OfficeServ static IP address, he/she can access to the OfficeServ stations without any configurations on the OfficeServ system. By default, OfficeServ system will accept any SIP messages.

In order to prevent unauthorized SIP peering access, one can enable the **Carrier Exclusive** field in IT5.2.12 or MMC 837. This option makes OfficeServ system to accept the SIP trunk calls from the registered ITSP and to accept the SIP peering calls only the the IP addresses are in the IT 5.2.17 or MMC 833.

Enable Carrier Exclusive option:

5.2.12.

Item	Item	Value
SIP Stack Configuration	Retrans T1 Time (100ms)	5
	Retrans T2 Time (100ms)	40
	Retrans T4 Time (100ms)	50
	General Ring Time (100ms)	50
	Invite Ring Time (100ms)	50
	Provisional Time (100ms)	1800
	Invite No Response Time (100ms)	50
	General No Response Time (100ms)	50
	Request Retry Time (100ms)	50
[837] SIP SIP Extension Configuration	Signal Port	5060
	IP-UMS/IVR Signal Port	5070
	SIP Expire Time (sec)	600
	NAT Reg Expire Time	60
	Carrier Exclusive	Enable
SIP Trunk Configuration	Default SIP Carrier	1
	iBG Expire Time (sec)	10
	Incoming Mode	Follow DID Trans
	Peer CLI Table	1
	Received CLI Forward On Alias	Disable

Filled the valid SIP peering IP address or leave them to 0.0.0.0. If all entries are 0.0.0.0, OfficeServ will not accept any SIP peering calls. In this example, OffcieServ will accept SIP peering call from IP address 65.65.65.64.

5.2.17.

Table Number	IP Address				Protocol
	Entry 1	Entry 2	Entry 3	Entry 4	
0	65.65.65.64	0.0.0.0	0.0.0.0	0.0.0.0	SIP
1	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
2	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
3	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
4	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
5	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
6	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
7	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
8	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
9	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
10	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
11	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
12	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
13	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
14	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
15	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
16	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
17	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
18	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
19	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
20	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
21	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP
22	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	SIP

4. Troubleshooting

4.1. No Dial Tone when Dial 8501 (Default SIP Trunk Number)

1. SIP stack license is invalid. SIP license is issued based on the MAC address of the MP. It is not transferable.
2. Make sure there are SIP trunks allocated in the MMC 841.
3. Make sure MGI card is there and in service.
    a. Make sure MGI and MP are in the same subnet range.
    b. Reboot MP or MGI card if needed.

4.2. "SVC AVAIL" in MMC 837 shows NO

1. "SVC AVAIL" field indicates the availability of the ITSP. If it shows NO, that means there is no SIP trunking services. Several things could go wrong. Check the following:
    a. MP and MGI card are physically connected to the Internet network.
    b. Internet is out of service. Use PC to check the Internet connection.
    c. OfficeServ system is not configured up properly.
2. Make sure "SIP SERVER" is enabled in MMC 837. It take several seconds to initialize changes.
3. Make sure the correct account information is in MMC 837.
4. Check if there is correct DNS IP address in MMC 837 if domain name is use in the proxy. Note, OfficeServ 7200 doesn't support DNS server; therefore, one must use the IP address for the proxy not domain name.
    a. Usually, the ITSP will provide the DNS address. If not, obtained from the ISP. Also check the Wireshark packets to make sure the DNS server returns the IP address for SIP server.
5. Use Wireshark to analyze the packets. Referred to the "SIP Service Technical Manual" for details.

4.3. Can't Receive Incoming Call

1. Make sure MMC 830 and 831 have the correct setting.
    a. Both MMC 830 and 831 have the same SYSTEM IP TYPE. Ex. "PRIVATE w PUBLIC".
    b. Public static IP address has registered to the ITSP.

4.4. One Way Audio or No Audio

1. Make sure the **MGI/OAS card has the latest software version** that match with MP software version.

2. Make sure MP, MGI/OAS card have the same SYSTEM IP TYPE setting.

    a. Use “PRIVATE w PUBLIC” if OfficeServ is behind a router or firewall.

    b. Use “PRIVATE ONLY” if only one IP address is used. This IP address can be a public or private IP address.

        i. The public IP address is used if there is no router/firewall between the OfficeServ and Internet.

        ii. The private address is used if the MPLS service is used.

3. Make sure there is no IP address and port conflict among MP and MGI/OAS cards.

    a. Check firewall setting.

    b. Make sure signal traffic (normally port 5060) is forwarded to MP.

    c. Make sure all voice traffic (RTP streams) are forwarded to the MGI or MPS channels.

        i. OAS card comes with the built-in MPS channels. The default MPS ports are 40000 which have the same default ports assignment as the embedded MPS channels. You will need to change the default MPS port number on the OAS card to different numbers.

        ii. Need to forward the port numbers on the router/firewall to the correct OfficeServ resources.

            1. For example: if OfficeServ system has the following assignment,

                a. MP

                    i. Public IP address is 65.65.65.64

                    ii. Private IP address is 10.10.10.2

                    iii. Embedded MPS port (16 channels) range is set 40000 to 40031

                b. OAS

                    i. Public IP address is 65.65.65.64

                    ii. Private IP address is 10.10.10.3

                    iii. MPS port (64 channels) range is set 40032 to 40159

                    iv. MGI port range is 30000 to 30031

                c. Router/Firewall port forwarding setting

                    i. Forward 65.65.65.64 port 5060 to 10.10.10.2 (MP)

ii. Forward 65.65.65.64 port 40000:40031 to 10.10.10.2 (MP)

iii. Forward 65.65.65.64 port 40100:40131 to 10.10.10.3 (OAS)

iv. Forward 65.65.65.64 port 30000:30031 to 10.10.10.3 (OAS)

4. If two public IP addresses are used for MP and MGI, the public IP address setting on the router must include the MGI public IP address as alias.

   a. For example,

	Router	**MP**	**MGI/OAS**
Local IP Address	10.10.10.1	10.10.10.2	10.10.10.3
Public IP Address	65.65.65.100 alias: 65.65.65.64 alias: 65.65.65.65	65.65.65.64	65.65.65.65
IP Type		Private with Public	Private with Public

5. Ask ITSP to turn "**Auto NAT Detection**" off.

Some ITSP "Auto NAT Detection" feature may cause no audio or one way audio issue in the following two scenarios:

a. ITSP doesn't use the negotiated audio ports for RTP, rather it monitors the originated audio ports.

   i. OfficeServ uses the MPS channels for the SIP trunk to IP phone connection. When call is on hold, OfficeServ uses MGI channel for music on hold (MOH). When use resumes the call, OfficeServ will send re-invite message to ITSP to change the audio from MGI port to MPS port. OfficeServ then switches the audio from MGI port to MPS port with some delay. Although ITSP agreed to use the new MPS port for audio, its "Auto NAT Detection" feature sees audio still coming from MGI port it will continue to communicate to the MGI port. Now OfficeServ has switched to MPS port but ITSP still communicates to the MGI port. When there is no auto NAT detection, ITSP will communicate with the negotiated port only.

b. ITSP doesn't handle external call transfer. Ie. ITSP doesn't support REFER message. OfficeServ needs to use two SIP trunks to connect two external call transfer.

   i. When user transfers an inbound call to an external number, OfficeServ will put the existing call on hold, which using MGI channel. When user completes the transfer and hangs up the call, OfficeServ will switch back to MPS channel. Some ITSP may not switch audio from MGI to MPS if "Auto NAT Detection" is enabled.

4.5. Incoming Calls Ring to the Operator Group Only

1. Make sure MMC 714 have correct DID directive.

4.6. No Caller ID or have Extra Digits on the Phone Display

1. Check the caller ID format from the ITSP.

   a. If E.164 format is used, the “+” sign must be removed from MMC 832.

   b. If account codes are prefixed to the incoming digits, the extra codes can be removed from MMC 832.

4.7. No Caller Name Only Caller Number

1. V4.22 software does not support DISPLAY NAME field in FROM header. The DISPLAY NAME field in the FROM header will be ignored.

2. V4.30 software support both call name and number.

3. Some ITSPs mat not deliver caller name as a standard feature.

4.8. Can’t Make Outgoing Call but Incoming Call is OK

1. Make sure “SVC AVAIL” is YES before making outgoing call. The SIP software will take about 30 seconds to initialize the task after saving the new SIP settings.

2. Make sure the “Outbound Proxy” field in MMC 837 is not empty.

3. Make sure there is CLI number in MMC 321. Many ITSPs require to have CLI before processing the call.

4. If there is outbound LCR digits in MMC 832, the digits must have the “SERVER USE” set YES.

5. If ITSP requires to use of the MP IP address in the FROM field rather than the OUT proxy IP address, then the SIP PEERING field in the MMC 837 must be enabled.

4.9. Receive “No User Response” while Making Outgoing Call

1. Make sure previous 3.8 is observed then check with the ITSP or network (router) settings. Make sure the account is in good standing.

    a. If the MP doesn’t receive progress message (Progress_in MP message) back from ITSP in 7 seconds, the OfficeServ will time out and shows “No User response”. This 7 seconds timer cannot be changed in MMC.

2. While making international cellular phone calls, sometimes the other service providers may take long time to respond. The OfficeServ will time out and shows “User no respond”.

    b. The timer for waiting Alert_in MP message is 15 seconds. It can be changed to a longer value in **MMC 501 VOIP_REROUTE_TM**. If the timer value is increased to 30 seconds and the ringing SIP message (180) comes in 28 seconds, then user will not hear ring back tone for 28 seconds.

4.10. DTMF is Not Working

1. DTMF type must match between OfficeServ and the ITSP. Check with your ITSP and adjust the OfficeServ MGI card setting in MMC 835 or 861.

4.11. Not Enough SIP Trunk Capacity

1. Make sure you have purchased enough SIP trunk license.
2. Check the maximum SIP trunk capacity for your OfficeServ system.
3. Make sure enough SIP trunks are allocated in MMC 857.

4.12. Poor Voice Quality

1. Most of poor voice quality issue is related to the network quality.
2. Make sure there is no software mismatched among OfficeServ equipments; especially, MP and MGI card..
3. Make sure there is no OfficeServ equipment hardware problems.
4. If internal calls between TDM phone and IP phone have good voice quality but SIP trunk calls have poor quality, then it is the outside Internet network issue. You will need to work with ISP/ITSP to solve the network issue.

4.13. Can’t Hear Announcement from Provider when Reach an Invalid Number

1. For outbound call, OfficeServ has an option to play the ringback tone from OfficeServ or play the real ring back tone from the far side.
2. VOIP REALRBT in MMC 210 should be set to ON to hear real ringback tone (or announcement) from the far side.

4.14. ITSP required "180" or "183" Ring Back Response

1. For inbound call when 183 is set, OfficeServ will provide the ringback tone to the caller.

2. For inbound call when 180 is set, ITSP will provide the ringback tone of the caller.

3. The option can be set from MMC 861 > VOIP RTP OPTION > SIP-T RBACK > 180

4. Some ITSPs requires OfficeServ to respond with "180" (not "183") ring back during the call process.

4.15. "302 Response" in MMC 837

1. This is ITSP call forward feature.

   a. If ITSPs support this feature, the external call forward will be handle by the ITSP. If call forward feature is set to an external number, any inbound call to this DID will not come to OfficeServ. ITSP will forward all calls from their end. No OfficeServ trunks will be used.
   b. If ITSPs don't support this feature, the call forward feature will be handled by the OfficeServ. That means the OfficeServ will use two trunks for the external call forwarding, one for inbound and other for outbound.
   c. The SVMi Find-Me feature has nothing to with "302 Response". Once the call reaches SVMi, all calls will use OfficeServ trunks.

4.16. Mystery Phone Call with "Asterisk" Caller ID on the Phone

1. Enable the **Carrier Exclusive** field in IT5.2.12 or MMC 837. "Exclusive".

4.17. Wireshark Trace

1. Wireshark traces from the OfficeServ system side is critical to analyze any issue. Wireshak traces from the ITSP side is also very helpful to troubleshooting the router/firewall issues. Please have the Wireshark traces from both sides and database file ready when seeking the technical support.