

T A B L E O F C O N T E N T S

SPECIAL APPLICATIONS SECTION

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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 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.

2.2 CALLER ID

The OfficeServ 7030 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

A 4TM trunk module

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](#)

The OfficeServ 7030 will display both Caller ID Name and Number simultaneously on the display the keysets. 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. This MMC is primarily used when displaying transferred calls.

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.

2.3 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.

2.4 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 digits stripped off using the modify 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.

2.5 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 506) 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 (MMC 607). 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 506 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 1005 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 1005 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.

2.6 OfficeServ 7030 Voice over IP (VoIP)

Introduction

This OfficeServ 7030 Technical Manual Special Applications section is intended to introduce qualified personnel to the following OfficeServ 7030 VoIP features:

- Supporting the VoIP trunking gateway.
- Supporting Samsung proprietary IP keyset. OfficeServ 7030 can support up to 16 IP stations, which can be combination of any of the following: ITP 5100 Series or Softphone.
- IP Networking: OfficeServ 7030 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, OfficeServ 7030, OfficeServ 7100, OfficeServ 7200 and OfficeServ 7400 systems. The OfficeServ 100, 500, and 7200 main software must be at version 2.69 or higher. The OfficeServ 7100 main software must be at version 3.34 or higher.

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

Overview

The MGI VoIP (Media Gateway Interface) 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 7030 systems via an IP network. The OfficeServ 7030 cannot support the MGI-16 card. However, the OfficeServ 7030 main unit comes with 4 MGI channels intergrated on board. These channels can be activated by the appropriate license key. The MGI also supports Samsung's proprietary ITP 5100 series IP keyphone, Wireless Handset, and Softphone. MGI 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. 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 7030 MGI channels 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 7030 MGI channels can utilize G.711, G.729A, G.729 or G.723 standards voice compression codec's. 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 MGI channels 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 data packets. Since voice conversation is usually considered "real time" these packets need to be delivered in a consistent manner with minimal delay.

In any IP network 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 7030 system in a proprietary IP message format to emulate full keyphone functionality. Working in conjunction with the OfficeServ 7030 main cabinet, the ITP keyphone can access all station features and trunk facilities on the OfficeServ 7030 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. All of the ITP models support PoE (Power Over Ethernet, IEEE 802.3af). This means that they can be powered by any PoE LAN switch instead of the AC adapter. Of course they can still be powered by the AC adapter connection to an AC outlet. There are three types of ITP 5100 series keyphones:

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

The **ITP-5121D** and **ITP-5107S** have 21 and 7 programmable buttons respectively and full traditional iDCS keyset functionality. The LCD display is a 2 line 16 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 through 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. Up to 99 buttons can be programmed on the ITP-5112L. In addition to the standard keyphone keys the navigation button also allows scrolling through various keyphone menu parameters. [See the ITP 5100 Series Keyphone User Guides for more details.](#)

Figure 2. ITP-5112L Keyphone



NOTE:

The OfficeServ 7030 supports one form of MGI channels:

1. On-board 4 MGI channels that are integrated on the 7030 main unit. This form of the MGI channels shares the same IP configuration and physical LAN connection as the 7030 main unit. The on-board MGI channels are activated by a license key.

MGI Operation Overview

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

The MGI channels can be used for several functions simultaneously or individually. Each MGI channel will auto negotiate the service of each channel for the type of usage required. This means that the same channel can use SIP trunking for one call and SPNet for the next call without any programming changes.

MGI Services:

- SIP Trunking Gateway
- ITP keyphone CODEC support
- VoIP Networking (SPNet)
- SIP Services/Softphones

OfficeServ 7030 Main Unit

The OfficeServ 7030 main unit supports the default LAN connection via the RJ 45 10/100 BaseT port on the face of the unit. This connects to the same IP network that the MGI channels are connected to. (ITP keyphones can be on the same or a different network). The OfficeServ 7030 system IP address is registered on the main unit. All IP connectivity is directed to the main unit's IP address. MMC 830: Ethernet Parameters, provides the selections for the system 7030 main unit IP address.

ITP Keyphone, MGI Channels and OfficeServ 7030 Main Unit Relationships

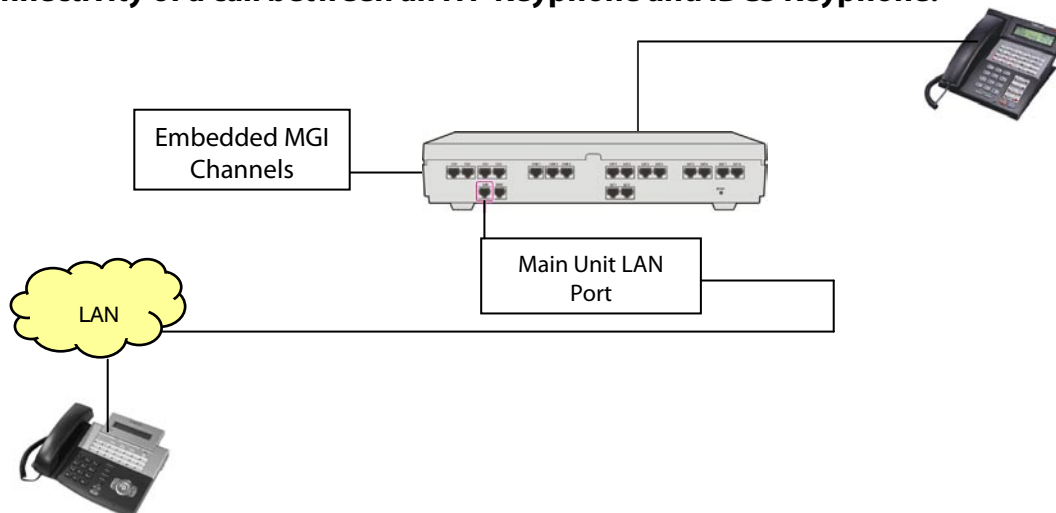
ITP Keyphone and iDCS Keyphone Operation

The ITP 5100 series keyphones use the MGI channels as the voice gateway when connected to the OfficeServ 7030 system stations or trunks. The OfficeServ 7030 main unit 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 MGI channel(s) used. There can be more ITP keyphones than available MGI 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 MGI channel that does not have a voice channel available, the ITP keyphone display will show “No MGI Channels Available”

The ITP 5100 series keyphones must be programmed to provide the keyphone IP address, subnet mask, router gateway (if used) and OfficeServ 7030 main unit (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 7030 system (MMC 840: IP SET INFO). The ID and Password are used to register the ITP keyphone to the OfficeServ 7030 and retain the number associated with that particular ITP keyphone. The ITP keyphones will register with the OfficeServ 7030 main unit and be provided a default extension number from the OfficeServ 7030 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 SET INFO, SYS IP OPTN. See Figure 4 for the relationship of ITP keyphone and iDCS keyphones.

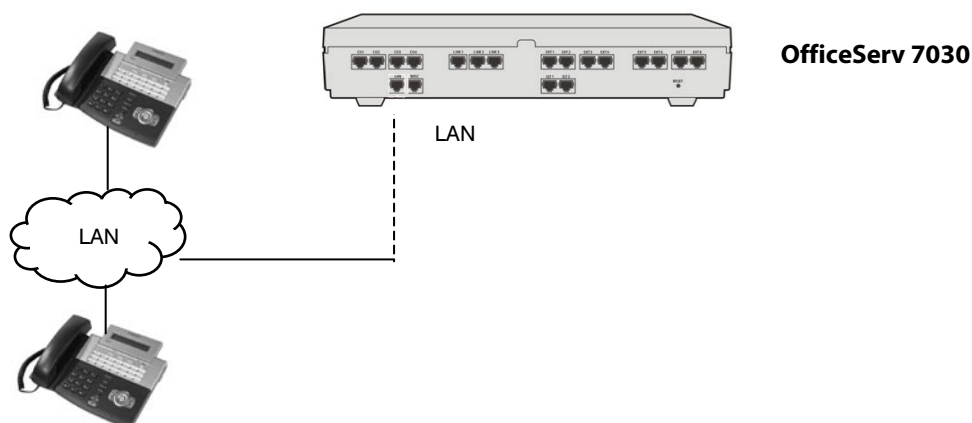
Figure 4. Connectivity of a call between an ITP Keyphone and iDCS Keyphone:



ITP Keyphone to ITP Keyphone Operation

When an ITP keyphone is connected and in conversation to another ITP keyphone the MGI audio codec is not involved when called on the private network side. The OfficeServ 7030 main unit is involved to provide call setup and tear down, programmable button functions, displays and system feature accesses. However, an MGI channel is used for system features such as conference and transfer or internal paging as needed. See Figure 5.

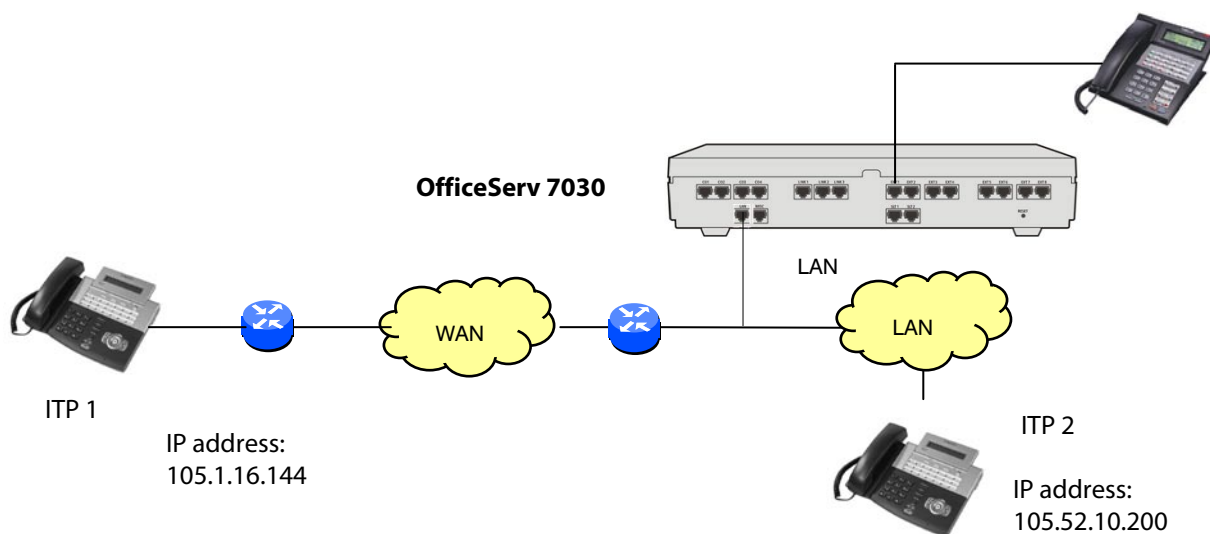
Figure 5. Connectivity between ITP Keyphones:



Remote ITP Keyphones

The ITP 5100 series key phones 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 OfficeServ 7030 main unit server IP address and have the matching ID and Password. The ITP Server is the address of the OfficeServ 7030 main unit supporting that ITP keyphone. The ID and Password are used to register the ITP keyphone to the OfficeServ 7030 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 OfficeServ 7030 main unit. 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 7030 MGI Functional Overview

MGI Channels Concepts

The MGI channels supports several VoIP services. In supporting multiple services the MGI channels are capable of using different protocols (IE Proprietary, SIP) automatically on a per port basis as needed. This allows the mixing of different standards on the same system.

MGI 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.](#)

SIP Trunking Gateway

The MGI channels 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 has an 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 MGI channels when calling a circuit switched digital keyphone or an outside trunk in the OfficeServ 7030 system. The 7030 main unit controls the setup messages and determines if the destination (IE iDCS keyphones or outside trunks) require the use of the MGI channels. The ITP keyphones only use the MGI channels when accessing the OfficeServ 7030 circuit switched system side. ITP to ITP calls do not use the MGI when called on a private network. This aspect of use can determine how many MGI channels need to be available based on ITP station traffic considerations. The number of ITP keyphones does not need to match the number of MGI channels unless a complete non-blocking application is required.

VoIP Networking Gateway

The MGI channels support the voice when connections between OfficeServ systems are set up to use the IP Networking Feature. Point to point calls are originated and terminated as VoIP trunks but the informational data is exchanged via IP packets between the processors.

Facsimile over IP

This option assigns the number of ports used to support IP facsimiles. The MGI channels support the T.38 facsimile over IP standard.

MGI Group Numbering Plan

When using MMC 615: MGI Group, an understanding of the multi-use aspect of the MGI 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 MGI channels Table 1 should be used as an example to understand default assignments. For clarity the first 4 port assignments are shown in the example.

MMC 615: MGI Group Assignments, selects which MGI channels will be accessed based on the call type. Default assignments allow all channels to connect to all services.

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

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

VoIP Network: This refers to channels used when using the OfficeServ Networking features on a private network. These ports will transport the voice side of an OfficeServ 7030 system networking call.

Public IP Network: This selects ports that will be used when using the OfficeServ Networking features via a public or exposed IP networking. These ports will transport the voice side of an OfficeServ 7030 system networking call.

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

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

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
381	381	381	3801	211	381	381	831	381	381	831	835
382	382	382	3802	212	382	382	832	382	382	832	836
383	383	383	3803	213	383	383	833	383	383	833	837
384	384	384	3804	214	384	384	834	384	384	834	838

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

For traffic considerations, (i.e. CODEC availability at any given time) specific channels may be selected to be accessed for defined uses. In the Table 2 example the four MGI channels will be segregated to allow the first four ports to be used for ITP keyphones and the last two ports as VoIP trunking gateway ports all on a private network. The other services are not used so they are removed for clarity.

Table 2: MGI 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 NETWORK 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	381		211							
3802	3802	382		212							
3803	3803						383			383	
3804	3804						384			384	

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

In the numbering plan example on table 2 there is a one to one relationship of available MGI channels 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.)

SIP Trunking Gateway

The OfficeServ 7030 MGI functions as a VoIP trunking gateway. The OfficeServ 7030 software considers the MGI channels as trunk ports. As a trunking gateway the MGI channels support the SIP standards on a per channel basis as needed. The OfficeServ 7030 MGI channels programming is similar to adding other cards in the OfficeServ 7030 system. Each MGI channel has ports assigned that are comparable to the system trunk ports. In a default system the OfficeServ 7030 will automatically identify that an MGI channel is present and assign trunk numbers to the available ports. The trunk numbers will be in the 38X range.

Programming MGI VoIP 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 7030 systems via VoIP can also be accomplished via MGI VoIP 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 MGI channels.

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 MGI channels used as trunking gateways. The outbound routing tables are referenced by MGI channels system wide.

A basic example of the direct station to station dialing string in an OfficeServ 7030 directing a call over an MGI channel to another OfficeServ 7030 MGI channel is as follows:

800 0 201

- 800 is the trunk group access code to the MGI VoIP trunks.
- 0 is the access code that references an IP address table.
- 201 is a station in the distant OfficeServ 7030 system.

In the above example the station caller will hear a second dial tone when 800 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 main unit of the

originating system and the far end system. 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 channels as a trunking gateway. IP Networking uses the communications between the main unit 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 7030 must have an IP address associated with the main unit. This is the same address using the on-board MGI channels.

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 junction or the meeting point of two networks. This meeting point can be a router or an MGI channel. When a request is made by a PC or other IP device 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 7030 and MGI, programming the gateway is equivalent to the router address to leave the local LAN and access another network.

ITP Keyphones and IP Network Connectivity

The OfficeServ 7030 MGI channels 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.

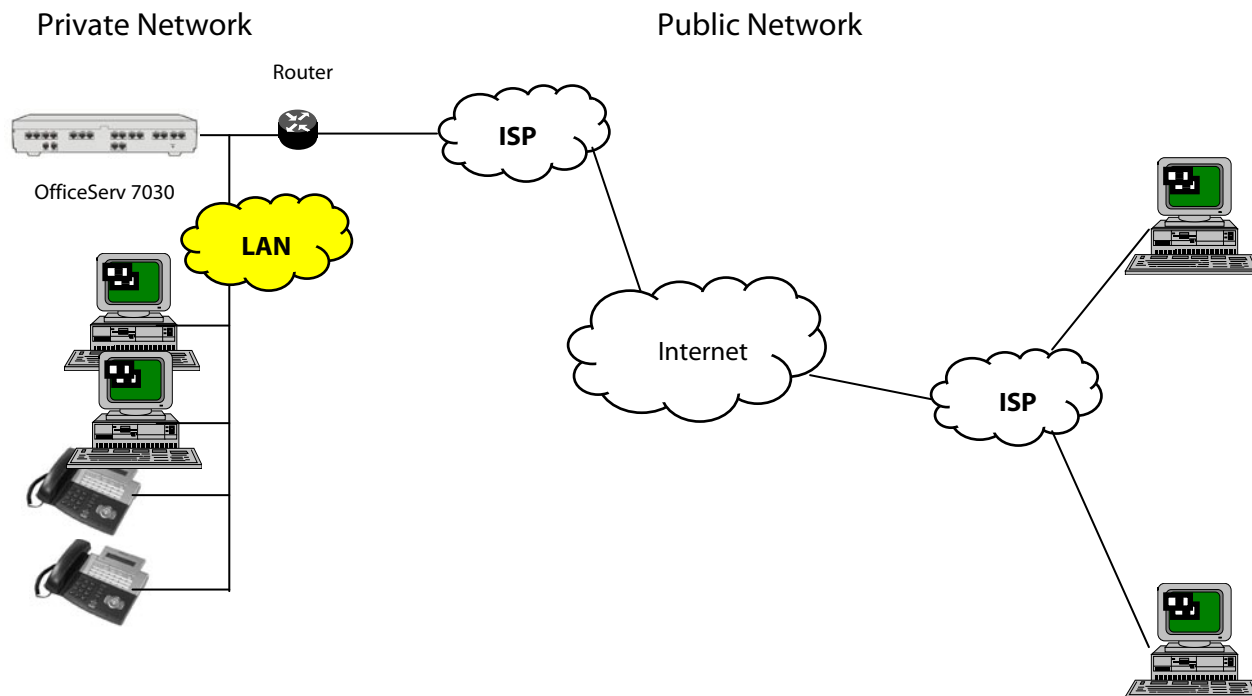
Public and Private IP Connectivity

The OfficeServ 7030 main unit, MGI 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 OfficeServ 7030 main unit. 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 OfficeServ 7030 main unit 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 7030 “public” and “private” do not necessarily actually mean public and private. These are relational terms to determine if the OfficeServ 7030 main unit 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 OfficeServ 7030 and ITP keyphones would be programmed as private. They are all on the same network.

If the OfficeServ 7030 main unit 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 OfficeServ 7030 and local ITP keyphones would be private. In this case the OfficeServ 7030 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 7030 main unit to convert the external IP address to the internal private addresses.

Customer Provided Routers

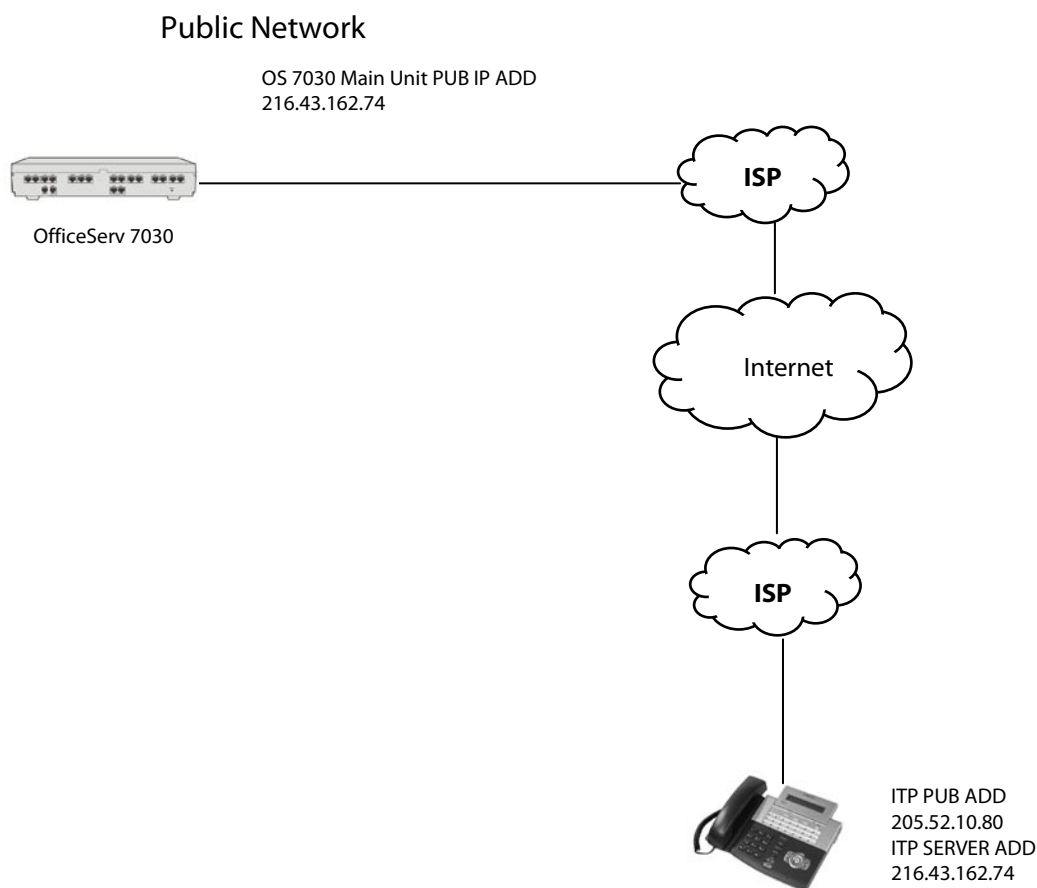
Exposure of the private network to the public can compromise the security of the private network. To use the Private with Public relationship the networks are usually separated by a router and/or firewall. The routers used **MUST** be able to support “reservations” or “port forwarding” feature. The Reservations 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 7030 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 OfficeServ 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 7030 system side. The remote location can have 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 OfficeServ 7030 main unit 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 7030 and ITP keyphone can be considered Private because they are all on the same network. When using the on-board MGI channels, then the MGI channels and main unit share the same IP configuration.

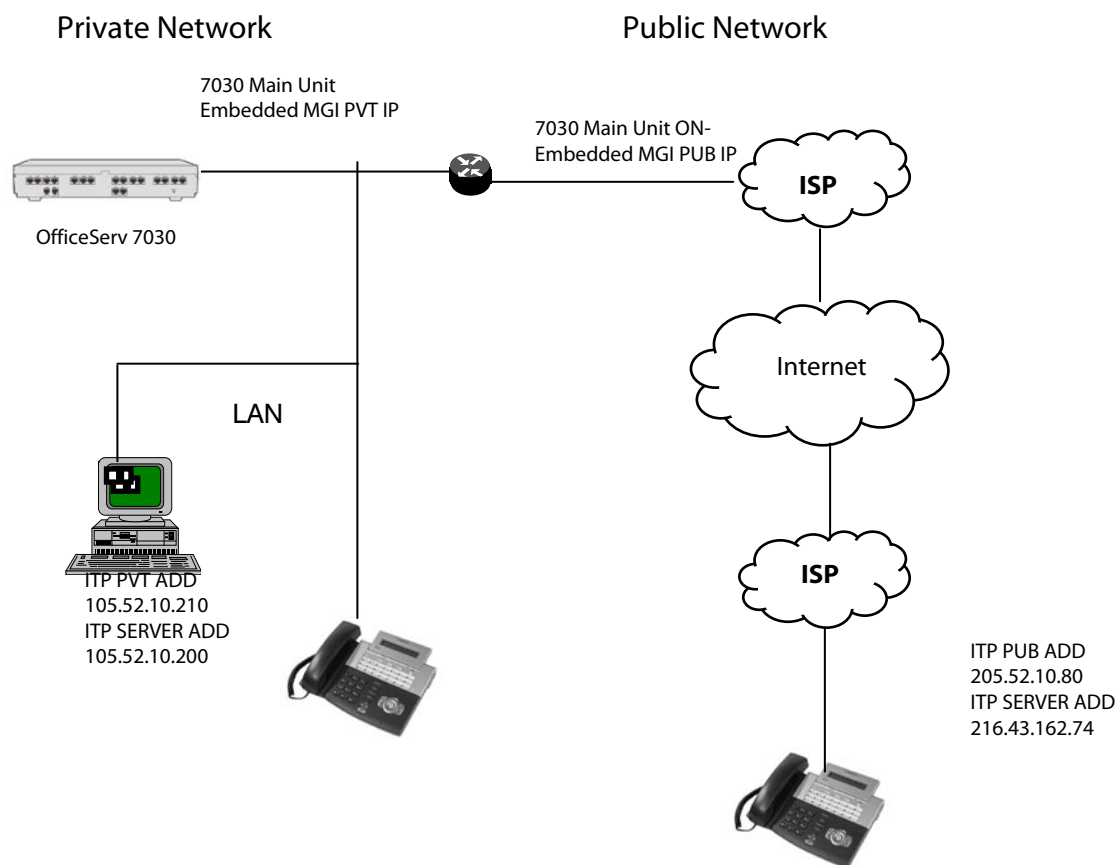
Figure 8. Remote ITP with Public Static IP Addresses



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

Figure 9 shows the relationship between the OfficeServ 7030 main unit and a remote ITP keyphone using a router to provide a separation between the Private and Public networks. With the OfficeServ 7030 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 OfficeServ 7030 Public IP address. The customer provided router in this case must “reserve” or forward the OfficeServ 7030 and MGI public IP addresses to the private OfficeServ 7030 addresses. Note that the remote ITP Server IP address is that of the public OfficeServ 7030.

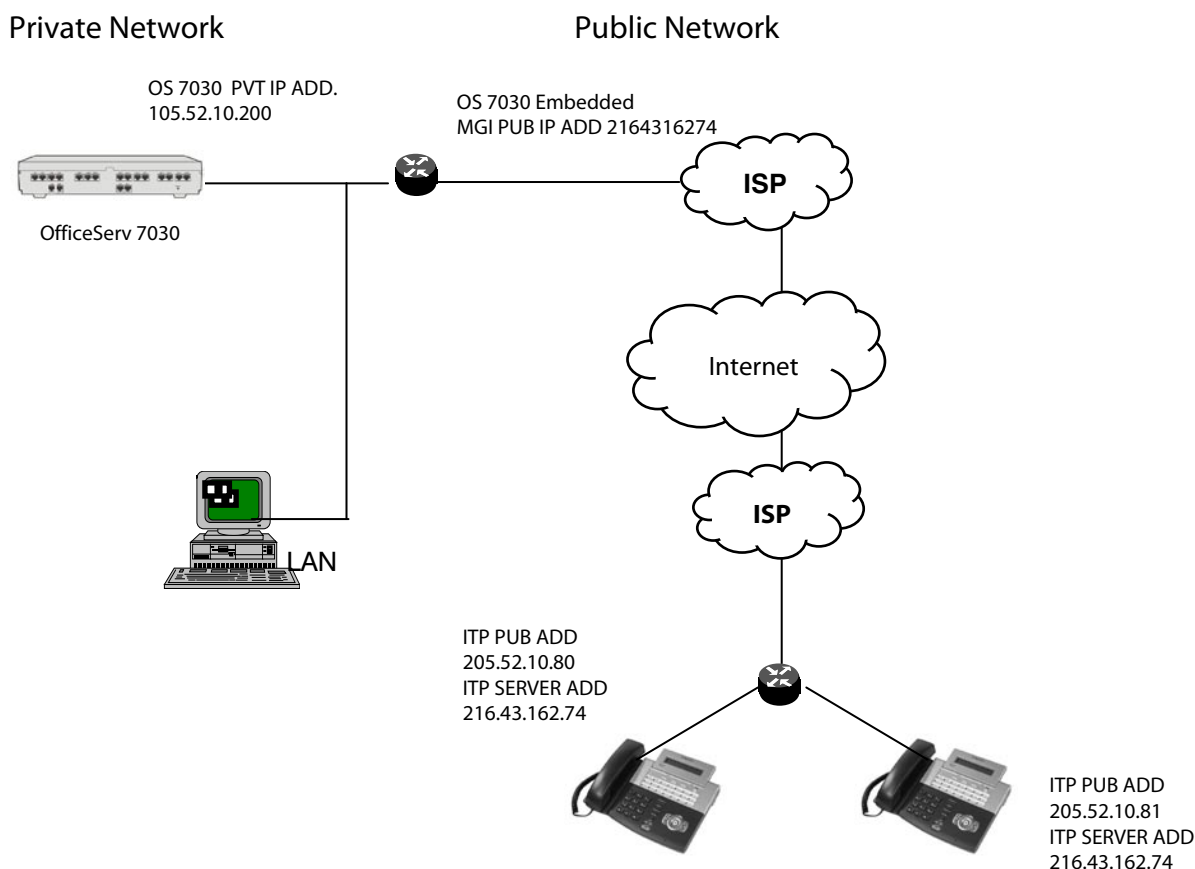
Figure 9. Private Network with Public Network ITP Keyphones



Remote ITP Keyphones with Reservations Router Addressing

Figure 10 shows Private with Public connectivity using router reservations or port forwarding at the OfficeServ 7030 end the remote ITP endpoint(s). In all cases, on a public application, a public IP addresses must be obtained for the OfficeServ 7030 main unit. 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 OfficeServ 7030 and ITPs) are using port forwarding or reservation to expose the OfficeServ 7030 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 7030 end and a NAT supporting router at the remote ITP endpoint. In public network cases a public IP addresses must be obtained for the OfficeServ 7030 main unit and the MGI channels. 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 OfficeServ 7030.

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 7030 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

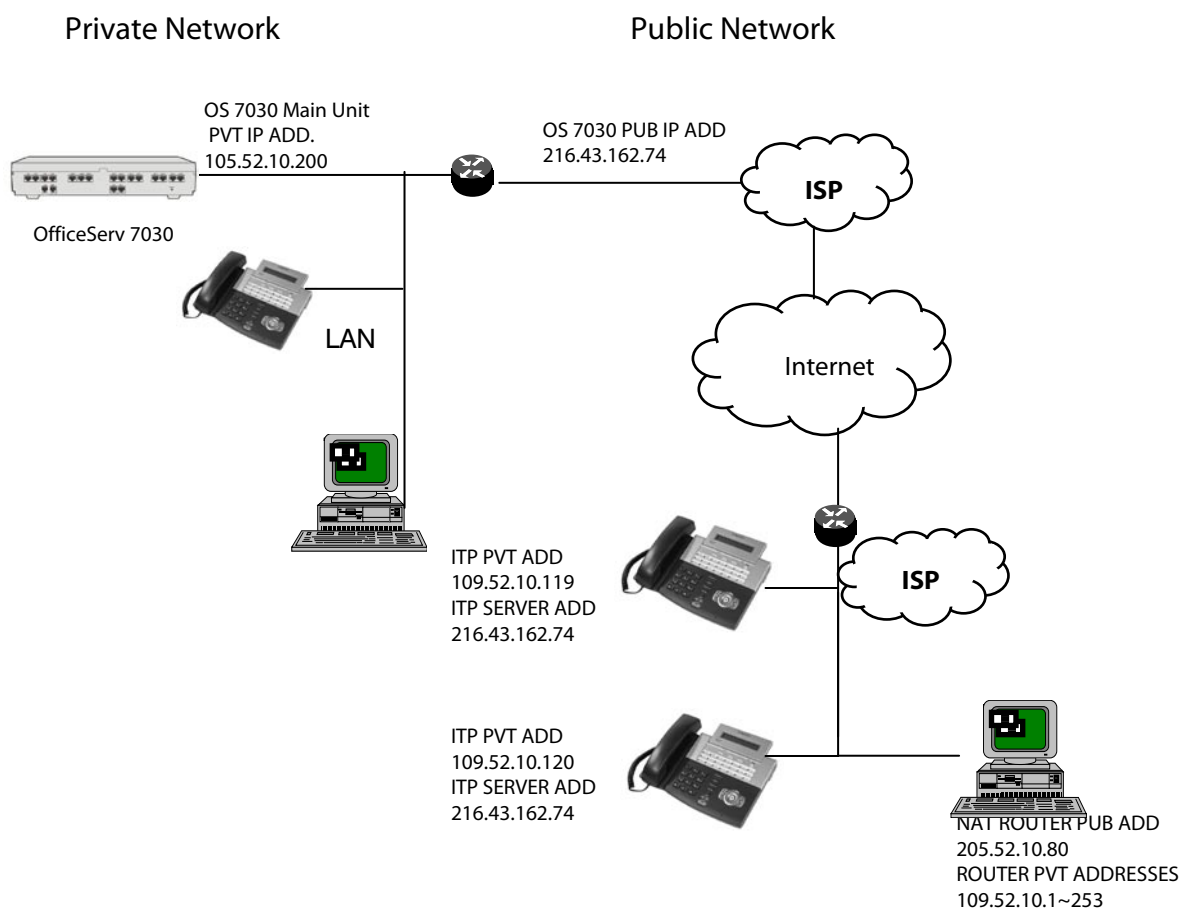
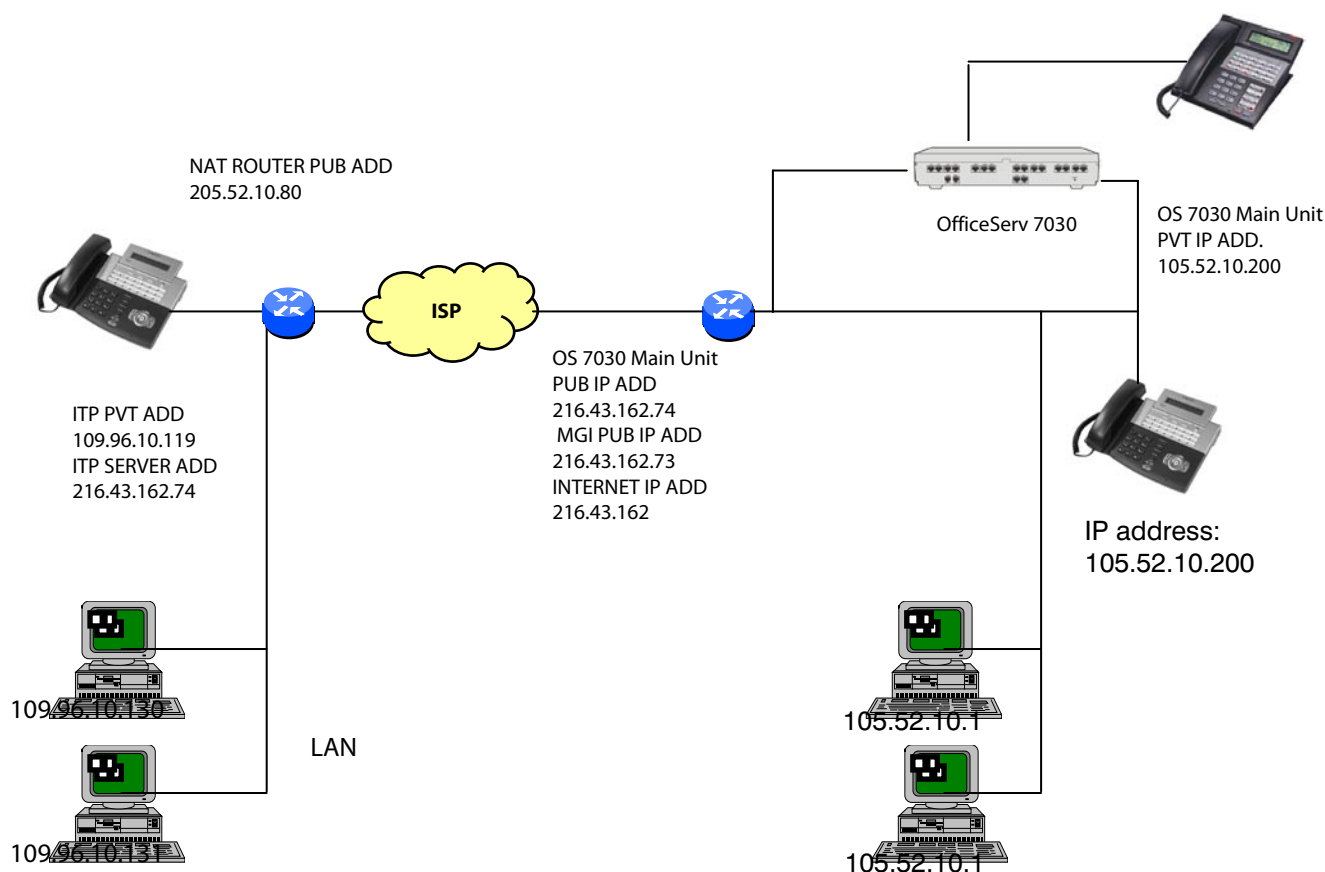


Figure 11b Private with Public Remote NAT Router Connections



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 7030 system side. One to expose the MGI channels to the public Internet and a second one to be used for OfficeServ 7030 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 OfficeServ 7030 main unit, then set:

- MMC 830 and MMC 831: IP type to "Private with Public" 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 7030 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 traverse **routers**, firewalls and to provide prioritization in certain environments. The IP ports and port types utilized by the OfficeServ 7030 are listed in Table 5.

System TCP/UDP Port numbers

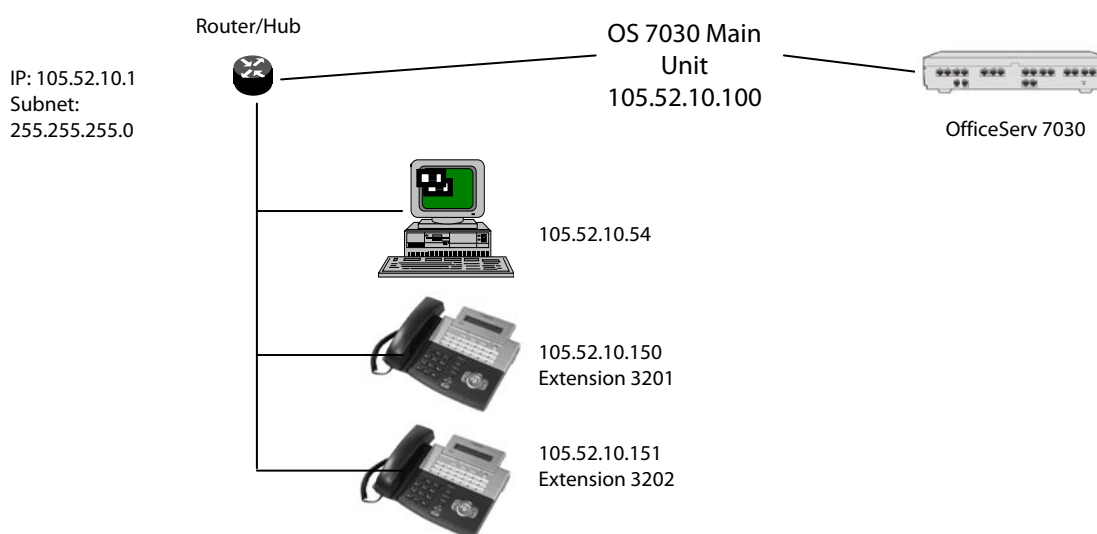
Table 5. OfficeServ 7030 TCP/RTP/UDP Port

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 [NOT USED in OS 7030]	1719	UDP	For connection with Gatekeeper
	1720	TCP	For connection setup
	1024 ~ 4999	TCP	For maintaining TCP Connection
SIP Gateway	5060	UDP	For connection setup
MGI (Embedded)	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	UDP	Signaling for Main Unit
	9000, 9001	RTP,RTCP	Voice data for MGI or ITP
CTI	5002	TCP	CTI Connection
Program Upload	5003	TCP	Program upload to NAND Flash
Installation Tool	5090	TCP	Connection port for the Installation Tool system programming application.
SMDR Print	5100	TCP	SMDR printout to IP connection
UCD Print	5101	TCP	UCD printout to IP connection

Service Type	Port Number	Protocol	Remarks
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

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 7030 system is as follows:

1. MMC 830 ETHERNET PARAMETERS

Note: When using the embedded MGI channels MMC 831 is “read only”. (The OfficeServ 7030 main unit’s IP will be used for the MGI function)

- SYSTEM IP ADDR: 105.52.10.100 (OS 7030 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 840: IP PHONE SETTINGS FOR EACH ITP ON THE PRIVATE NETWORK

- IP TYPE: Private

3. 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 12 MMCs directly related to the use and operation of the MGI channels.

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 OfficeServ 7030 system IP address, subnet, gateway and OfficeServ 7030 Public address (if required).

MMC 831 MGI PARAMETERS [NOT REQUIRED FOR OS 7030]

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 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 837 SIP OPTIONS

This MMC programs the SIP protocol parameters.

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 7030.

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.

2.7 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 7030 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

NOTE: PRI networking is not supported on the OfficeServ 7030.

Each node in the network must have at least one MGI channel activated. In each system, the OfficeServ 7030 main unit 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 OfficeServ 7030 main unit can communicate with the processor 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 processor as well as the IP addresses of the processor 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 the OfficeServ 7030 main unit.**

Configure the OS 7030 with the relevant IP information:

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

MMC 831. MGI PARAMETERS

Note: When using the MGI channels, this is a "Read-Only" MMC. The settings here will be the same as MMC 830.

Configure the MGI channels with the relevant IP information:

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

MMC 835. MGI DSP Options

Configure the MGI channels with the relevant IP information:

- AUDIO CODEC must be set for G.729A in each node for IP networking

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 381 and 382 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 (private/public).
- 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 node number of the system, 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 node number, the system regards it as the internal 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 MGI 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 MMM	❷	01:22 →09122 SZ:4 MAX:07 MB:N
---	----------------------------	---	----------------------------------

In display above, 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.
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. 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.

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.

10. 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	2

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. (Assigned at default)

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.

11. 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 408.
- 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 (repeat digits received).
- 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
50**	RP1~RP6 = B	0	Intercom to 001 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

NOTES

1. The OfficeServ 7030 does not support the SVMi-20E card. However, the OfficeServ 7030 can be a remote system that uses 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 7030 is a remote node using the Centralized Voicemail resources remotely via the SPNET.
2. It is recommended that the embedded OfficeServ 7030 Voicemail NOT be used as the Centralized Voicemail because of the 2 port voicemail limitation.

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 or auto setup 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. This is accomplished by changing the attendant destination in MMC 825.

Note: The OfficeServ 7030 should only be used as the remote node for Centralized Operator. The OfficeServ 7030 should not be used as the main node for centralized operator.

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 NTKW 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 NTKW 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 OfficeServ 7030 main unit can communicate with each other over TCP/IP. When using the on-board MGI channels, the MGI will share the same IP configuration as the OfficeServ 7030 main unit. This example assumes that nodes A & B are using the on-board MGI.*

NODE A:

Node id: 091
Station numbering plan: 2000~2099
VoIP Network trunk group: 803
OS 7030 Main Unit IP: 192.168.1.2

NODE B

Node id: 092
Station numbering plan: 2100~2110
VoIP Network trunk group: 803
OS 7030 Main Unit IP: 192.168.1.4

NODE A Programming:

- **MMC 830:** Program the IP information for OS 7030 main unit.
IP address: 192.168.1.2
Gateway: 192.168.1.1
Subnet mask: 255.255.255.0
*** Now select the reset option to reset the OS 7030
- **MMC 831:** "Read Only". Displays same IP information as programmed in MMC 830.
- **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 OfficeServ 7030 main unit 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.

STN NUM PLAN	Station numbers (2000 ~ 2099)		
TRKG NUMBER PLAN	803		
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	

- **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. **Be sure to set DISP=Y.**

Index	Station	Converted	SZ	Max	MB	DISP	
01	21	09221	4	7	N	Y	Stations in Node B

- **MMC 714:**

Entry	Digit	MOH	1~6	CW	Delete	Name
001	20**	-	B	-	0	-
002	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	Route	
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:** Set HHMM= 0000, LCRT=1 for each day of the week unless 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.

NODE B Programming:

- **MMC 830:** Program the IP information for OfficeServ 7030 main unit.
IP address: 192.168.1.4
Gateway: 192.168.1.1
Subnet mask: 255.255.255.0
*** Now select the reset option to reset the OS 7030
- **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 OS 7030 main unit 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.

STN NUM PLAN	Station numbers (2100 ~ 2399)
--------------	-------------------------------

TRKG NUMBER PLAN	803	
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. **Be sure to set DISP=Y.**

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	21**	-	B	-	0	-
002	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	Route	
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 unless 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).

2.8 NETWORKING

The networking feature package allows up to 99 OfficeServ 7030 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 connection. 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) 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 7030 does not support the SVMi-20E card. However, the OfficeServ 7030 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 7030 is a remote node using the Centralized Voicemail resources remotely via the SPNET.
2. It is recommended that the embedded OfficeServ 7030 Voicemail NOT be used as the Centralized Voicemail.
3. The OfficeServ 7030 does not support PRI or QSIG over PRI.

Enhanced Networking

OfficeServ 7030 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 is present, the CID information is passed across the network.
- This feature is supported over IP networking only on the OfficeServ 7030.
- MMC 406 is primarily used to terminate analog lines. Digital trunks will not follow the setting of MMC 406, but will follow 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.

<p>NOTE: Caution should be used when considering using the OfficeServ 7030 as Centralized Attendant due to the limitation in MGI resources.</p>
--

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 IP networking only on the OfficeServ 7030.

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 is present, the CID information is passed across the network.
- This feature is supported over IP networking only on the OfficeServ 7030.

MMC INFORMATION:

[MMC 607](#) (set UCD FINAL DEST)

[MMC 733](#) (set INVALID 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 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 IP networking only on the OfficeServ 7030.

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 and ITP phones.
- This feature is supported over IP networking only on the OfficeServ 7030.

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 of 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 centralized voicemail station group extender, for remote nodes).
- This feature is supported over IP networking only on the OfficeServ 7030.

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 IP networking only on the OfficeServ 7030.

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 on 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 on IP networking.
- When CID is 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 be 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.

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)

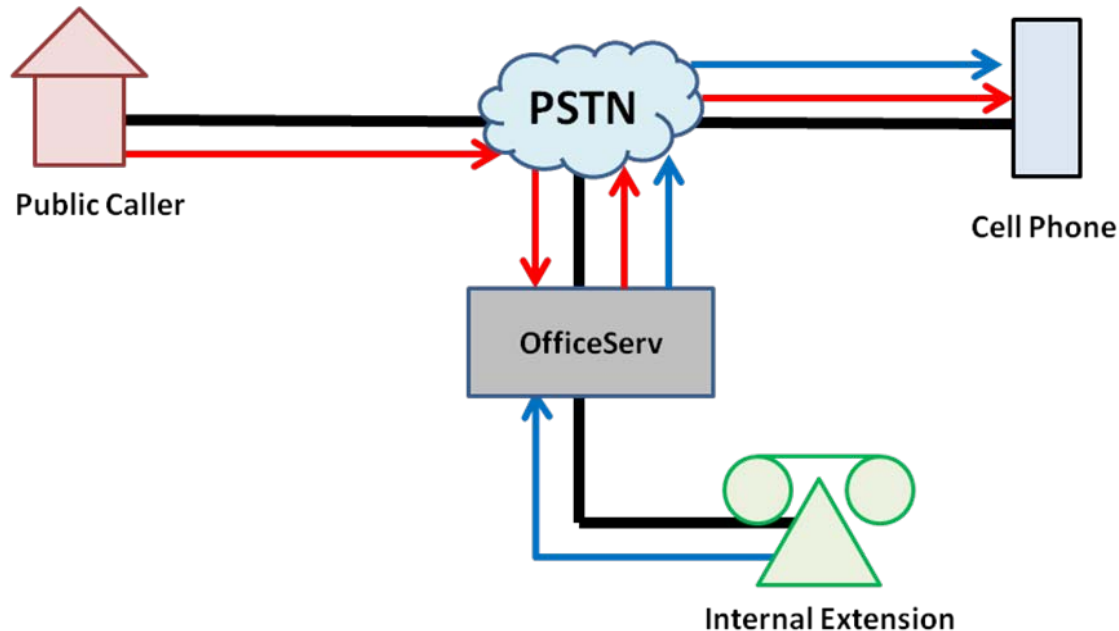
[MMC 419](#) (set TRUNK Priority)

2.9 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**

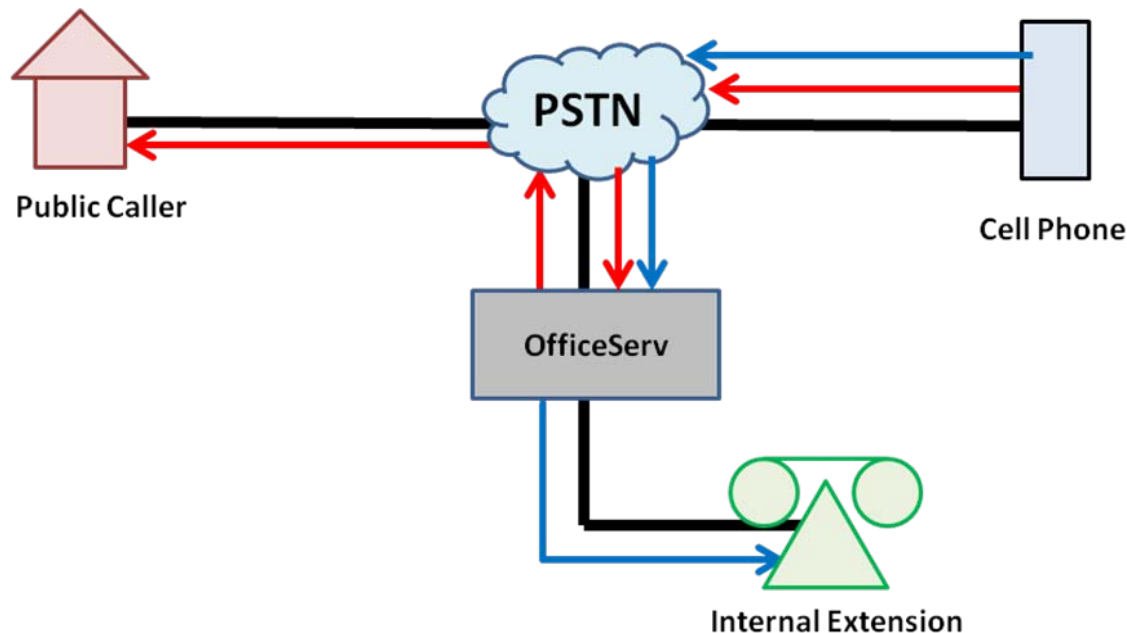
The red arrows show the call flow of a CO call. The public caller dials a DID number which has a destination of a MOBEX port (in this case a cell phone). When the call reaches the cell phone the call will display the caller ID of the public caller.**

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. 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

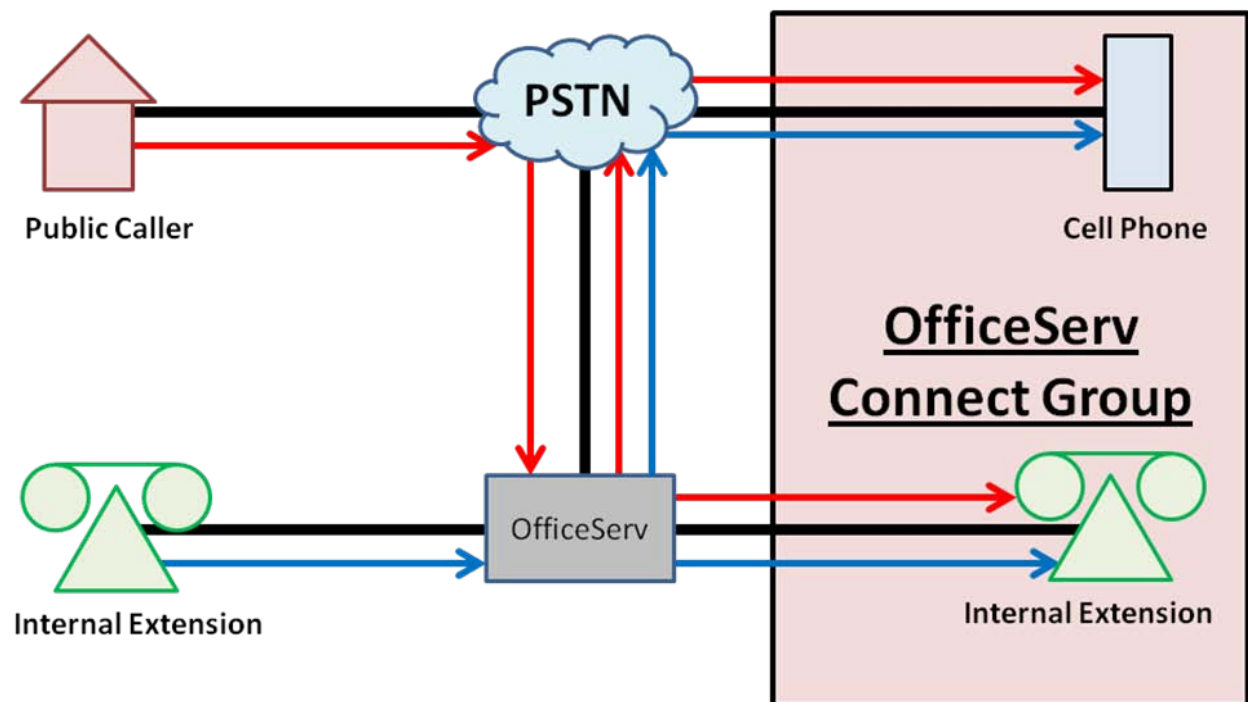
- 1. Call to CO will display CID of MOBEX port instead of Cell Phone**
- 2. Call to Internal Extension will show MOBEX Extension as CID**

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:



Simultaneously Ring Multiple Numbers (Max 5)

- 1. Incoming CO Call to Multiple Telephone Numbers**
- 2. Intercom call to Multiple Telephone Numbers**

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 trunking. Executive MOBEX requires a valid Executive MOBEX license.

****:** The public caller or internal extension caller ID can only be sent to the MOBEX port if your PRI service provider allows it. Some PRI 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.
- 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.

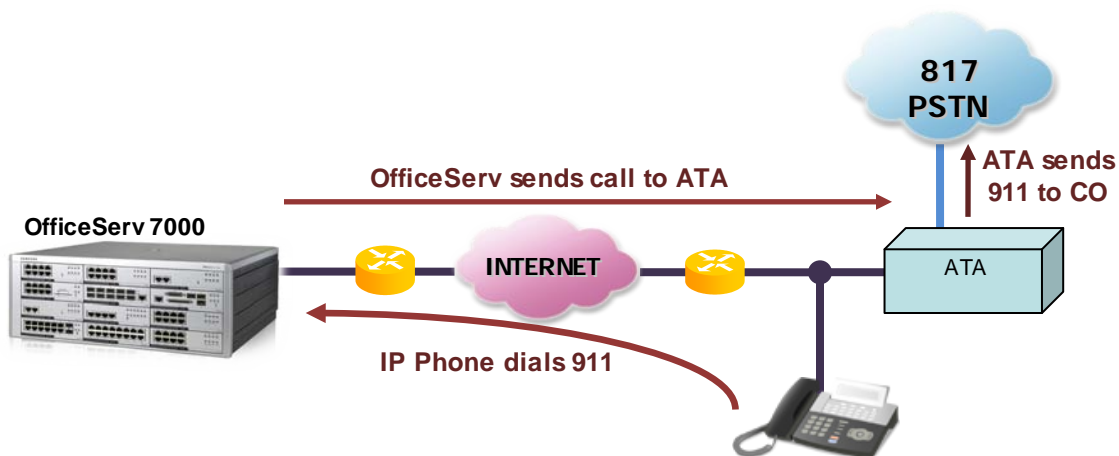
6. In MMC 860 enter the Executive MOBEX license
7. In MMC 328 set each of the 4 MOBEX ports to be Executive users. In the CLI field enter the 7 or 10 digit cell phone number. Set the Master station to the corresponding station from MMC 329 (2011 for 3011, etc). This will assign the MOBEX ports as Executive 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.

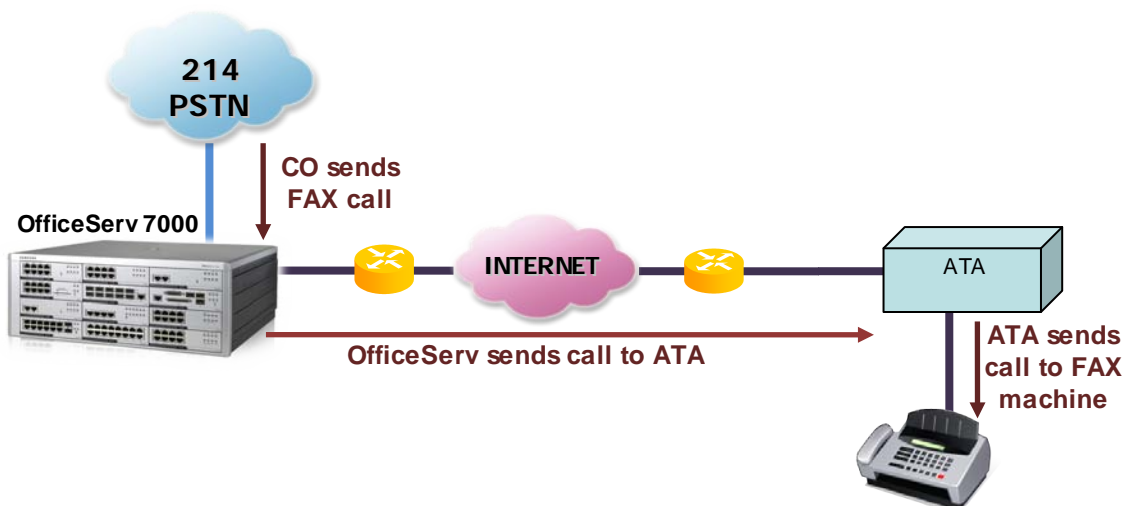
2.10 SIP CONNECTIVITY TO 3rd PARTY ATA DEVICES

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.



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 2.6**

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 <i>(the public IP of the OfficeServ MP)</i>
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:

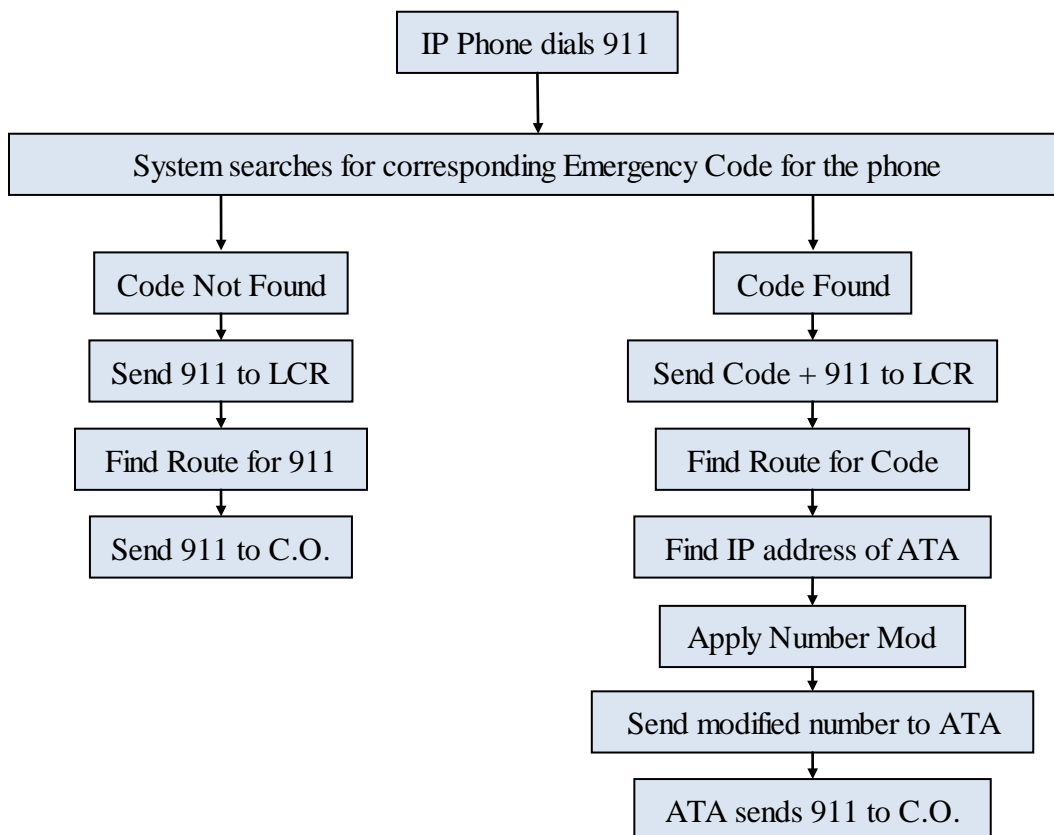
SIP Port:	5060 <i>(cannot use the same port from step 2)</i>
Proxy:	12.240.8.209
Display Name:	3303
User ID:	3303
Password:	0000
Use Auth ID:	YES
VoIP to PSTN Gateway Enable:	YES
One Stage Dialing:	YES

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 **LCRENABLE** 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.

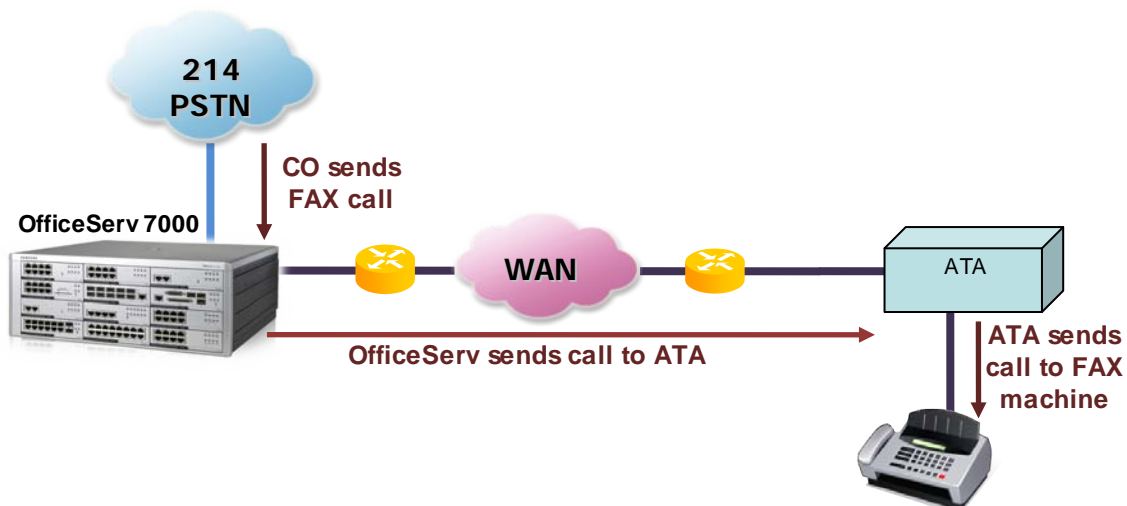
In order to properly route the call the OfficeServ will scan the Outbound Access Digits in **MMC 832** for a match. **4444** with a length of **7** is recognized, and the **4444** is stripped off. Digit string **411** is then passed on to **MMC 833** to find the IP address at table 000, entry 00 (**TB(000)IP 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.

OfficeServ™ 7000 Series SIP Trunking Applications & Best Practices



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 MG16/64 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

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 3 rd 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 3 rd 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:

- For SPNET to use MPS channels, one need to enable MPS on both MMC 861 (MPS SERVICE: ON) and MMC 820 (NO MGI:ON).
- Two MPS channels equivalent to one MPS call.** 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".

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 (MG1/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 NEGOT]	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.

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.	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.
	Preferred method if ITSP can support REFER message.			OfficeServ uses one MGI for music on hold.
PBX Managed 2	OfficeServ handles connection on trunk-to-trunk call transfer.	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 sends Re-Invite message to ITSP.
	Normal application.		OfficeServ uses two MGIs or 1 MPS call for connecting two external calls.	OfficeServ uses one MGI for music on hold.
PBX Managed 1	OfficeServ handles connection on trunk-to-trunk call transfer.	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 sends Re-Invite message to ITSP.
	Special application for UK.		OfficeServ always uses two MGIs for connecting two external calls even MPS is enabled.	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 EXCLUSIVE 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
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, OfficeServ 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.