

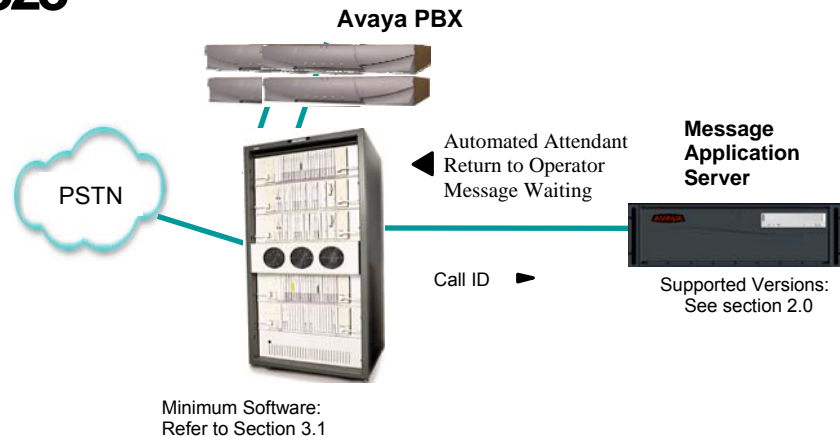


Avaya

Modular Messaging

Configuration Note 88014 – Version AR (1/10)

Avaya Definity G3, Prologix & S8300/S85x0/S84x0/S87x0 H.323



Overview

This Configuration Note is intended for Avaya certified Modular Messaging technicians/engineers who are familiar with Modular Messaging procedures and terminology. It also assumes that you are Avaya certified or very familiar with the features and functionality of the Avaya PBXs supported in this Configuration Note and the QSIG protocol.

Use this document in conjunction with *Modular Messaging Installation Guide* and the *Avaya PBX Administration Guide*.

Please read the entire document before attempting any configuration.

1.0 METHOD OF INTEGRATION

The H.323 integration provides connectivity with the Avaya PBX over an IP network. The connectivity between the Avaya Message Application Server (MAS) and the Avaya PBX is achieved over an IP-connected trunk defined as ISDN-PRI-equivalent tie lines. This integration passes call information and MWI using QSIG messages tunneled in the H.323 packet.

H.323 Trunks allows the Avaya PBX and the Avaya Message Application Server to communicate over a LAN.

Avaya MAS Requirements

Important when using Hyper-Threading capable systems

¹Release Note:

Should features of the integration not function optimally when integrated to a PBX or MM that may be operating on an unsupported software release as defined Section 2.0 and 3.1, customers will need to upgrade their PBX and/or MM to a supported software release.

PBX Hardware Requirements

PBX Software Requirements

2.0 AVAYA MESSAGE APPLICATION SERVER REQUIREMENTS

- Releases ¹: 1.1, 2.0, 3.x, 4.0, 5.x

Note: If using an S3500, or any hardware that is Hyper-Threading capable, Avaya strongly recommends Hyper-Threading be disabled. Please refer to the Installation Guide for detailed instructions.

3.0 PBX HARDWARE REQUIREMENTS

Before performing the installation ensure the customer site has passed the Avaya Network Evaluation.

Definity G3 and S8500/S8700:

- TN2302 IP Media Processor for voice processing
- TN799C C-LAN for signaling

S8300:

- PROCR

3.1 PBX SOFTWARE REQUIREMENTS

Minimum Software ¹:

Definity G3 & Prologix: G3V10.1 Load 48

S8xx0: MV1.1, CM2.x, CM3.x, CM 4.x, CM 5.x

PBX Software Packages:

- QSIG-Supplementary Services, QSIG Interworking with DCS, QSIG rerouting, QSIG transfer into Voice Mail, QSIG Value-Added comes with an optional "Advanced Networking Package."
- DCS+ for Multiple PBX Support (Refer to Section 9.0)
This option is **only** required for networking (multiple PBXs)

Note: Either DCS+ or QSIG Networks are necessary to support multiple PBXs (Centralized Voice Mail). No other networks are supported. Refer to Section 8.0 (Considerations) for additional info.

Important: Before ordering account teams should check with Avaya Services to determine if there are any applicable patches for customer specific configuration. Specifically if DCS+ or QSIG Networking is used.

3.2 LAN CONNECTIVITY

- Ethernet LAN connectivity - TCP/IP

3.3 CUSTOMER-PROVIDED EQUIPMENT

- Wiring necessary to support the physical LAN (CAT 3 minimum) Ethernet Hub (Location of 10baseT - Optional)

Supported integration features

4.0 SUPPORTED INTEGRATION FEATURES

[✓] Items are supported

System Forward to Personal Greeting

All Calls	[✓]
Ring/no answer	[✓]
Busy	[✓]
Busy/No Answer	[✓]

Station Forward to Personal Greeting

All Calls	[✓]
Ring/no answer	[✓]
Busy	[✓]

Auto Attendant	[✓]
Call Me	[✓]
Direct Call	[✓]
External Call ID (ANI)	[✓]
Fax	[]
Find Me	[✓]
Internal Call ID	[✓]
Message Waiting Indication (MWI)	[✓]
Multiple Call Forward	[✓]
Multiple Greetings	[✓]
N+1	[✓]
Outcalling	[✓]
Queuing	[]
Return to Operator	[✓]

IMPORTANT: PBX options or features not described in this Configuration Note are not supported with this integration. To implement options/features not described in this document, please contact the Avaya Switch Integration product manager.

PBX Configuration

5.0 SWITCH CONFIGURATION FOR IP INTEGRATION

The following tasks must be completed in the following order when programming the PBX to integrate. PBX programming is intended for certified PBX technicians/engineers.

- Verify customer option for H.323/QSIG trunking
- Administer QSIG TSC and Extension Length
- Assign Local Node Number
- Administer C-LAN and IP Media Processor circuit packs (G3 and S8500/S8700 only)
- Assign IP node names and IP addresses to C-LAN, IP Media Processor (G3 and S8500/S8700 only)
- Define IP interfaces (G3 and S8500/S8700 only)
- Administer IP Network Regions (Skip this step if default QOS is used)
- Administer Codec Type (Skip this step if default QOS is used)
- Add Message Server to the node names
- Create H.323 signaling group
- Create a new trunk group
- Modify the signaling group to add trunk group
- Create Route Pattern
- Modify AAR Analysis Table
- Modify AAR Digit Conversion Table
- Modify ARS Digit Conversion Table
- Define ISDN Numbering Format
- Create Coverage Path
- Create Hunt Group (Pilot Number)

Note: The screens shown in this section are taken from an Avaya Definity G3 administration terminal. Some parameters may not appear on all software releases.

NOTICE:

The screens in this Config Note are only for illustration purposes.

It is recommended that a qualified technician review the customer's configuration for accuracy.

***NOTE:**

DCS is only required for networking.

Stand-alone/single PBX do not require DCS, so those options may be set to "N"

****NOTE:**

Cvg of Calls Redirected Off-Net (CCRON) must be set to "y" if you are using *Find Me*.

5.1 VERIFY CUSTOMER OPTIONS FOR H.323/QSIG TRUNKING

Ensure all required software features are enabled on the PBX. Access the System Parameters Customer Options form. Below is an example of the forms required for QSIG integration, with the required features in **boldface**.

IMPORTANT: Only change the recommended fields.

display system-parameters customer-options Page 1 of 9
OPTIONAL FEATURES

G3 Version: V9 Maximum Ports: 3600
Location: 1 Maximum XMOBILE Stations: 100

IP PORT CAPACITIES

Maximum Administered IP Trunks: 150
Maximum Concurrently Registered IP Stations: 200
Maximum Administered Remote Office Trunks: 5
Maximum Concurrently Registered Remote Office Stations: 5

Maximum Number of DS1 Boards with Echo Cancellation: 332
Maximum VAL Boards: 10

display system-parameters customer-options Page 2 of 9
OPTIONAL FEATURES

Abbreviated Dialing Enhanced List? y	Attendant Vectoring? y
Access Security Gateway (ASG)? n	Audible Message Waiting? y
Analog Trunk Incoming Call ID? y	Authorization Codes? y
A/D Grp/Sys List Dialing Start at 01? n	CAS Branch? n
Answer Supervsn by Call Classifier? y	CAS Main? n
ARS? y	Change COR by FAC? n
ARS/AAR Partitioning? y	Cvg Of Calls Redirected Off-net? n**
ARS/AAR Dialing without FAC? n	DCS (Basic)? y*
ASAI Interface? y	DCS Call Coverage? y*
ASAI Proprietary Adjunct Links? y	DCS with Rerouting? y*
Async. Transfer Mode (ATM) PNC? n	DEFINITY Network Admin? n
Async. Transfer Mode (ATM) Trnkng? n	Digital Loss Plan Modification? n
ATM WAN Spare Processor? n	DS1 MSP? y
ATMS? y	DS1 Echo Cancellation? y

Avaya H.323 Integration

```
display system-parameters customer-options                               Page 3 of 9
                                OPTIONAL FEATURES

Emergency Access to Attendant? y                                ISDN-BRI Trunks? n
Enhanced EC500? y                                                ISDN-PRI? y
Extended Cvg/Fwd Admin? y                                       Malicious Call Trace? y
External Device Alarm Admin? n   Mode Code for Centralized Voice Mail? n
Flexible Billing? n
Forced Entry of Account Codes? n                               Multifrequency Signaling? y
Global Call Classification? n   Multimedia Appl. Server Interface (MASI)? n
Hospitality (Basic)? y         Multimedia Call Handling (Basic)? y
Hospitality (G3V3 Enhancements)? n   Multimedia Call Handling (Enhanced)? y
H.323 Trunks? y                                                Multiple Locations? n
                                                                    Personal Station Access (PSA)? n
IP Stations? y
ISDN Feature Plus? n
ISDN Network Call Redirection? n

(NOTE: You must logoff & login to effect the permission changes.)
```

```
display system-parameters customer-options                               Page 4 of 9
                                OPTIONAL FEATURES

PNC Duplication? n                                             Tenant Partitioning? y
                                                                    Terminal Trans. Init. (TTI)? y
Processor and System MSP? y                                     Time of Day Routing? y
Private Networking? y                                         Uniform Dialing Plan? y
                                                                    Usage Allocation Enhancements? y
R9.5 Capabilities? y                                         VAL Full 1-Hour Capacity? y
Remote Office? y
Restrict Call Forward Off Net? y                               Wideband Switching? y
Secondary Data Module? y                                       Wireless? y
Station and Trunk MSP? y
Station as Virtual Extension? y

(NOTE: You must logoff & login to effect the permission changes.)
```

```
display system-parameters customer-options                               Page 7 of 9
                                QSIG OPTIONAL FEATURES

Basic Call Setup? y
Basic Supplementary Services? y
Centralized Attendant? y
Interworking with DCS? y* (see note below)
Supplementary Services with Rerouting? y
Transfer into QSIG Voice Mail? y
Value-Added (VALU)? y
```

***Note:** This would be set to “y” only if you are using DCS to network Avaya PBXs.

❑ Change features and assign your private network access code, in this example we assigned **100**

```
change feature-access-codes                               Page 1 of 6
                                     FEATURE ACCESS CODE (FAC)
Abbreviated Dialing List1 Access Code: 101
Abbreviated Dialing List2 Access Code: 102
Abbreviated Dialing List3 Access Code: 103
Abbreviated Dial - Prgm Group List Access Code:
Announcement Access Code: 104
Answer Back Access Code:
Auto Alternate Routing (AAR) Access Code: 100
Auto Route Selection (ARS) - Access Code 1: 9           Access Code 2:
Automatic Callback Activation:                          Deactivation:
Call Forwarding Activation Busy/DA: 190    All: *9       Deactivation: #9
Call Park Access Code: *6
Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
CDR Account Code Access Code: 188
Change COR Access Code:
Change Coverage Access Code:

Data Origination Access Code:
Data Privacy Access Code:
Directed Call Pickup Access Code:
```

- Administer QSIG TSC and Extension Length. Assign a phantom endpoint extension for QSIG TSC(Temporary Signaling Connection). Enter any valid, unassigned extension.

```
display system-parameters features                       Page 6 of 11
                                     FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS

Send Non-ISDN Trunk Group Name as Connected Name? n
Display Connected Name/Number for ISDN DCS Calls? y
Send ISDN Trunk Group Name on Tandem Calls? n
CPN Replacement for Restricted Calls: #202
CPN Replacement for Unavailable Calls: #202
QSIG TSC Extension: 2799
MWI - Number of Digits Per Voice Mail Subscriber: 4 (see note below)

National CPN Prefix:
International CPN Prefix:
Pass Prefixed CPN to ASAI? n
Unknown Numbers Considered Internal for AUDIX? y           Max Length: 5
USNI Calling Name for Outgoing Calls? n
Path Replacement with Measurements? y
QSIG Path Replacement Extension: 2798 (vacant extension)
Path Replace While in Queue/Vectoring? n
```

NOTE: This parameter must match the number of digits used for mailbox/extension length. For **multiple length extensions** leave this field blank. **Important:** The option to leave this field blank requires Avaya CM 2.1 and later.

Access the System Parameter Coverage-Forwarding and change or ensure the following parameters are set appropriately:

Maintain SBA At Principal? **n*** (*see note on left*)

Coverage Of Calls Redirected Off-Net Enabled? **n**** (*see note on left*)

***NOTE:**

Maintain SBA at principal set to “n” ensures that when the call covers to voice messaging the appearance on the station is removed to ensure privacy. Doing this prevents someone from listening to the call as it is being recorded by the voice messaging system.

****NOTE:**

Coverage of Calls Redirected Off-Net (CCRON) must be set to “y” if you are using *Find Me*.

```
display system-parameters coverage-forwarding          Page 1 of 2
                SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS

    Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
    Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
    Coverage - Caller Response Interval (seconds): 2
    Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls: 3

COVERAGE
    Keep Held SBA at Coverage Point? y
    External Coverage Treatment for Transferred Incoming Trunk Calls? n
    Immediate Redirection on Receipt of PROGRESS Inband Information? n
    Maintain SBA At Principal? n*
    QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? n
    Station Hunt Before Coverage? n

FORWARDING
    Call Forward Override? n
    Coverage After Forwarding? n
```

```
display system-parameters coverage-forwarding          Page 2 of 2
                SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

    Coverage Of Calls Redirected Off-Net Enabled? n**
    Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? n
    Ignore Network Answer Supervision? n
    Disable call classifier for CCRON over ISDN trunks? n
```


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- Assign Local Node Number. Ensure the PBX has an assigned Local Node Number. If there is no assigned number, enter 1.

For G3 release 10 or earlier:

```
display dialplan

                                DIAL PLAN RECORD

                                Local Node Number: 1
                                ETA Node Number:
                                Uniform Dialing Plan: 4-digit
                                ETA Routing Pattern:
                                UDP Extension Search Order: local-extensions-first
FIRST DIGIT TABLE
First
Digit  - 1 -      - 2 -      - 3 -      - 4 -      - 5 -      - 6 -
1:
2:
3:
4:
5:
6:
7:
8:
9: dac
0: misc
*: misc
#: misc
```

FOR MV release 1.3 or later:

```
change dialplan parameters                                Page 1 of 1

                                DIAL PLAN PARAMETERS

                                Local Node Number: 1
                                ETA Node Number:
                                ETA Routing Pattern:
                                UDP Extension Search Order: local-extensions-first
6-Digit Extension Display Format: xx.xx.xx
7-Digit Extension Display Format: xxx-xxxx
```

- Administer C-LAN and IP Media Processor circuit packs (G3 and S8500/S8700 only)

```
display circuit-packs                                Page 2 of 5

                                CIRCUIT PACKS

                                Cabinet: 1
                                Carrier: B
Cabinet Layout: five-carrier                                Carrier Type: port

Slot Code  Sf Mode  Name                                Slot Code  Sf Mode  Name
00:  TN799  C      C-LAN                                11:
01:  TN2302      IP Media Processor  12:
02:
03:  TN793  B      ANALOG LINE                                13:
04:  TN464  G      DS1 INTERFACE                                14: TN464  G      DS1 INTERFACE
                                15: TN464  G      DS1 INTERFACE
```

- Assign IP Node names IP addresses to C-LAN, IP Media Processor (G3 and S8500/S8700 only). Enter the appropriate IP addresses for the installation.

```
display node-names ip
```

IP NODE NAMES	
Name	IP Address
clan1	148.147.6 .246
medpro1	148.147.6 .163

- Define IP interfaces (G3 and S8500/S8700 only). Enter the appropriate Gateway address for the installation.

```
display ip-interfaces
```

Page 1 of 15

IP INTERFACES

Enable	Eth	Pt	Type	Slot	Code	Sfx	Node Name	Subnet Mask	Gateway Address	Net Rgn
y	C-LAN	01B00	TN799	D	clan1			255.255.255.0	148.147.6 .1	1
y	MEDPRO	01B01	TN2302		medpro1			255.255.255.0	148.147.6 .1	1
n								255.255.255.0	.	.
n								255.255.255.0	.	.

- Skip this step if default QOS is used. Define IP Network Regions. In this example network region '6' is selected.

```
display ip-network-region 6
```

Page 1 of 2

IP Network Region

Region: 6
Name: msgserver

Audio Parameters
Codec Set: 6

UDP Port Range (Note: 5000-5999 is the only range allowed by MM)
Min: 5000
Max: 5999

DiffServ PHB Value: 46 Direct IP-IP Audio Connections? n
802.1p/Q Enabled? n IP Audio Hairpinning? n

NOTE1: If QOS is being administered, don't forget to include the network region in the CLAN and MedPro network region. In this example the CLAN and MedPro are using network region 6.

```
change ip-network-region 6
```

Page 2 of 2

Inter Network Region Connection Management

Region (Group Of 32)

Region	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1	2
001-032	6					6																
033-064																						
065-096																						
097-128																						
129-160																						
161-192																						
193-224																						
225-250																						

NOTE2: The IP Network Region screen is different with the S8300. In addition to the codec-set=6, direct-WAN=y, and WAN-BW-limits=NoLimit

- Skip this step if default QOS is used. Administer Codec type. In this example codec set '6' is used. The default for the ip-codec-set is G.711MU. This must match the settings on the MAS noted in Section 6.0

```
display ip-codec-set 6
```

IP Codec Set

Codec Set: 6

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
1:	G.711MU	n	3	30
2:				
3:				
4:				
5:				

- Add the MAS to the IP Nodes Names. Enter the IP address assigned to the Message Server.

```
change node-names ip
```

Page 1 of 1

IP NODE NAMES

NAME	IP Address
Default	0 .0 .0 .0
clan1	148.147.6 .246
medpro1	148.147.6 .163
msgserver	148.147.6 .123

- Create the signaling group for the H.323. The Near-end Node Name is the name assigned to the C-LAN above. The Far-end Node Name is the name assigned to the Message Server above. For the Far-end Network Region enter the number of the region created above. For this example signal group 299 was selected.

NOTE: If QOS was administered, make sure to enter the new network region in the Far-end Network Region field.

NCA-TSCs are used for QSIG.

CA-TSCs are used with DCS+

Trunk Group for NCA TSC
should have a value to indicate
trunk group only if using MWI
Interrogation.

```
display signaling-group 299

                                SIGNALING GROUP

Group Number: 299              Group Type: h.323
                                Remote Office? n          Max number of NCA TSC: 10
                                Max number of CA TSC: 10
                                Trunk Group for NCA TSC: 1
                                Trunk Group for Channel Selection: (configured on page 11)
                                Supplementary Service Protocol: b

Near-end Node Name: clan1 (procr w/S8300)  Far-end Node Name: msgserver
Near-end Listen Port: 1720                 Far-end Listen Port: 1720
                                           Far-end Network Region: 6
                                           (For the Far-end Network Region also SEE NOTE ABOVE)

LRQ Required? n                      Calls Share IP Signaling Connection? n
RRQ Required? n                      Bypass If IP Threshold Exceeded? n

DTMF over IP: out-of-band            Direct IP-IP Audio Connections? n
                                           IP Audio Hairpinning? n
                                           Interworking Message: PROGRESS
```

- Create a new trunk group. For this example trunk group 299 was selected.

```
add trunk-group 299                                     Page 1 of 22

                                TRUNK GROUP

Group Number: 299              Group Type: isdn          CDR Reports: n
Group Name: msgserver          COR: 1                    TN: 1          TAC: #296
Direction: two-way            Outgoing Display? n        Carrier Medium: IP
Dial Access? y                Busy Threshold: 255        Night Service:
Queue Length: 0
Service Type: tie              Auth Code? n              TestCall ITC: rest
                                Far End Test Line No:

TestCall BCC: 4
TRUNK PARAMETERS
Codeset to Send Display: 6      Codeset to Send National IEs: 6
Max Message Size to Send: 260
Supplementary Service Protocol: b  Digit Handling (in/out): overlap/enbloc

                                Trunk Hunt: cyclical      QSIG Value-Added? n
                                           Digital Loss Group: 13
Calling Number - Delete:      Insert:              Numbering Format: unk-unk
                                Bit Rate: 1200            Synchronization: async  Duplex: full
Disconnect Supervision - In? y Out? n
Answer Supervision Timeout: 0
```

Tip: If the numbering format is set to **unk-pvt** then the PBX looks to the Private-Numbering Table to build the number. The network Level must be **0** and the Level 2 and 1 code blank or NO number will be sent. This is where the PBX builds the called party number for the integration. If it is set to **unknown**, the PBX looks at the Public-Unknown table where an entry is required to build to the number. If there is no entry in this table to build the number, NO number is sent.

Note: If you are using CM 4.0 or higher see next page

```

add trunk-group 299                                     Page 2 of 22
TRUNK FEATURES
  ACA Assignment? n                                     Measured: none       Wideband Support? n
                                                    Internal Alert? n     Maintenance Tests? y
                                                    Data Restriction? n   NCA-TSC Trunk Member:
                                                    Send Name: n          Send Calling Number: y
  Used for DCS? n                                       Hop Dgt? n
Suppress # Outpulsing? n Numbering Format: unk-pvt (see Tip & NOTE below)
Outgoing Channel ID Encoding: exclusive                UII IE Treatment: service-provider

                                                    Replace Restricted Numbers? n
                                                    Replace Unavailable Numbers? n
                                                    Send Called/Busy/Connected Number: y
                                                    Hold/Unhold Notifications? y
                                                    Modify Tandem Calling Number? n
  Send UII IE? n
  Send UCID? n
Send Codeset 6/7 LAI IE? y

                                                    DS1 Echo Cancellation? n
Path Replacement with Retention? n
Path Replacement Method: always
                                                    Network (Japan) Needs Connect Before Disconnect? n
  
```

NOTE: The Numbering Format fields could be set according to the customer's environment. There are many variations plus other tables associated with these fields that must be considered also. Consult with a Software Specialists for proper programming.

```

add trunk-group 299                                     Page 6 of 22
                                                    TRUNK GROUP
                                                    Administered Members (min/max): 1/1
GROUP MEMBER ASSIGNMENTS                               Total Administered Members: 1

  Port      Code Sfx Name      Night      Sig Grp
1: ip       maschan1
2: ip       maschan2
3: ip       maschan3
...
19: ip      maschan19
20: ip      maschan20
  
```

NOTE: For multiple MAS, you must administer ports in a circular fashion (round robin) to distribute the load.

☐ Change the ISDN Numbering - Private Network form to configure and ensure the PBX for the proper Network Level to be used. Below is a copy of the ISDN Numbering - Private Network form with the required field in **boldface**.

```

display isdn private-numbering

ISDN NUMBERING - PRIVATE FORMAT

Network Level: 0
Level 2 Code:
Level 1 Code:

PBX Identifier:
Deleted Digits: 0
  
```

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NOTES for Avaya CM 4.0 and later

Network Levels and Level codes are now found in *system-parameters features* under *Parameters for Creating QSIG Selection Numbers*

Note: If the numbering format is set to **unk-pvt** then the PBX looks to the Private-Numbering Table to build the number. Network Level must not be left blank (in most cases this is set to **0**) or NO number will be sent. This is where the PBX builds the proper number (i.e., user's station number) for the integration to open the proper mailbox.

IMPORTANT

This screen supports Private Numbering Plans (PNP) allowing you to specify the digits to be put in the Calling Number information element (IE), the Connected Number IE, and the QSIG Party Number for extensions in the Private Numbering Plan.

Avaya CM supports private-network numbers up to 15 digits in length. If the total number — *including the level 1 and 2 prefixes, the Private Prefix (formerly known as PBX identifier), and the extension* — is more than 15 digits long, neither QSIG Party Numbers nor the information elements are created or sent.

display system-parameters features Page 8 of 17
FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS

Send Non-ISDN Trunk Group Name as Connected Name? n
Display Connected Name/Number for ISDN DCS Calls? y
Send ISDN Trunk Group Name on Tandem Calls? n
Send Custom Messages Through QSIG? y

PARAMETERS FOR CREATING QSIG SELECTION NUMBERS

Network Level: 0
Level 2 Code:
Level 1 Code:

QSIG/ETSI TSC Extension: 2998

MWI - Number of Digits Per Voice Mail Subscriber: 4 (see note below)

Feature Plus Ext:

National CPN Prefix:

International CPN Prefix:

Pass Prefixed CPN to ASAI? n

Unknown Numbers Considered Internal for AUDIX? n

USNI Calling Name for Outgoing Calls? y

Path Replacement with Measurements? y

QSIG Path Replacement Extension: 2798

Path Replace While in Queue/Vectoring? n

NOTE: This parameter must match the number of digits used for mailbox/extension length. For **multiple length extensions** leave this field blank (this requires Avaya CM 2.1 or later). However, please note **MM supports only one mailbox length**.

In **Avaya CM 4.0** the Private Numbering Form is now used to define the number format for **specific** trunk groups. In our example screen below, we have a 4-digit extension length that includes extensions from 2000 thru 5999. The 4-digit number will be part of the QSIG number information that will be passed to the MM for call integration.

If you set numbering format to "unk-pvt" on page 2 of the trunk group form (see earlier in this section), which is for the MM Trunk Group, this form must be completed so the CM knows how to build the *private format* number. For that reason, do not leave the form blank or the MM call integration will fail.

change private-numbering 3 Page 1 of 2
NUMBERING - PRIVATE FORMAT

Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len	
4	2	99		4	Total Administered: 4
4	3	99		4	Maximum Entries: 540
4	4	99		4	
4	5	99		4	

- Modify the signaling group to add the new trunk group.

```
display signaling-group 299

SIGNALING GROUP

Group Number: 299      Group Type: h.323
Remote Office? n      Max number of NCA TSC: 10
                      Max number of CA TSC: 10
                      Trunk Group for NCA TSC: 1
Trunk Group for Channel Selection: 299
Supplementary Service Protocol: b

Near-end Node Name: clan1      Far-end Node Name: msgserver
Near-end Listen Port: 1720      Far-end Listen Port: 1720
Far-end Network Region: 6
LRQ Required? n      Calls Share IP Signaling Connection? n
RRQ Required? n
Bypass If IP Threshold Exceeded? n

DTMF over IP: out-of-band      Direct IP-IP Audio Connections? n
                                IP Audio Hairpinning? n
                                Interworking Message: PROGRESS
```

- Create a Route Pattern for the trunk group created. For this example route pattern 299 is used, with trunk group 299 and the leading 3 digits are deleted from the aar number.

```
display route-pattern 299                                     Page 1 of 3
Pattern Number: 299

Grp. FRL NPA Pfx Hop Toll No.  Inserted      DCS/ IXC
No.   Mrk Lmt List Del  Digits  QSIG
1: 299 0      3      (this is an example,  y user
2: 300 0      3      see note below)      n user
3:                                     n user
4:                                     n user
5:                                     n user
6:                                     n user

BCC VALUE  TSC CA-TSC  ITC BCIE Service/Feature BAND  No. Numbering LAR
0 1 2 3 4 W Request Dgts Format Subaddress
1: y y y y y n n      rest      rehu
2: y y y y y n n      rest      none
3: y y y y y n n      rest      none
4: y y y y y n n      rest      none
5: y y y y y n n      rest      none
6: y y y y y n n      rest      none
```

Note: If adding additional MAS servers, ensure their Trunk Groups are added to this Route Pattern and follow the same programming as the 1st Trunk Group.

- Within the AAR Digit Analysis Table, create a dialed string that will map calls to the newly created Route Pattern. The dialed string created in the AAR Digit Analysis Table will be used later in the Hunt Group form that will define the MAS Hunt Group. Below is an example of an AAR dialed string in **boldface**.

add aar analysis 299							Page	1 of	2
AAR DIGIT ANALYSIS TABLE							Percent Full: 10		
	Dialed String	Total Min	Max	Route Pattern	Call Type	Node Num	ANI Reqd		
	5187200	7	7	299	aar		n		

- Modify the AAR Digit Conversion to allow MAS to dial and transfer to local PBX extensions. Ensure to administer a Matching Pattern for all extensions the voice messaging server (MAS) will be dialing.

display aar digit-conversion 0								Page	1 of	2
AAR DIGIT CONVERSION TABLE								Percent Full: 10		
Matching Pattern	Min	Max	Del	Replacement String	Net	Conv	ANI	Req		
2	4	4	0		ext	n		n		
3	4	4	0		ext	y		n		
5	4	4	0		ext	n		n		
7	4	4	0		ext	n		n		

- Modify the ARS Digit Conversion to allow MAS to dial and transfer to local PBX extensions. Ensure to administer a Matching Pattern for all extensions the voice messaging server (MAS) will be dialing.

display ars digit-conversion 1								Page	1 of	2
ARS DIGIT CONVERSION TABLE								Percent Full: 10		
Location: all										
Matching Pattern	Min	Max	Del	Replacement String	Net	Conv	ANI	Req		
2	4	4	0		ext	n		n		
3	4	4	0		ext	n		n		
5	4	4	0		ext	n		n		
7	4	4	0		ext	n		n		

- Add Hunt Group. Configure a Hunt Group to be used as the Call Coverage Point for the Call Coverage Path assigned to the MAS subscribers. This hunt group's extension number is going to be used as the MAS Access Number. For the Coverage Path enter the Coverage Path number to be created in the next step. Enter the dialed string created previously in the AAR Digit Analysis Table in the "Voice Mail Number" field on page 2 of the Hunt Group form. Also, in the "Routing Digit (e.g. AAR/ARS Access Code)" field of this form, enter your PBX's AAR Access Code as defined on page 1 of the Feature Access Codes form. This hunt group is configured with no members assigned to it, and should be configured as follows:

```
add hunt-group 5                                     Page 1 of 60
                                                    HUNT GROUP

Group Number: 5                                     ACD? n
Group Name: msgserver                               Queue? n (see note below)
Group Extension: 7200                               Vector? n
Group Type: ucd-mia                                Coverage Path:
TN: 1                                                Night Service Destination:
COR: 1                                              MM Early Answer? n
Security Code:
ISDN Caller Display: mbr-name
```

Note: Queue? should be set to "n" as recommended. Refer to Consideration 8.10 for further information.

NOTE: The "Voice Mail Number" entered here must be administered in the "Voice Mail System Configuration->PBX->OutgoingCall" tab of the MAS. The number that is entered in the "Outgoing Call" tab is the Voice Mail Number.

```
add hunt-group 5                                     Page 2 of 60
                                                    HUNT GROUP

Message Center: qsig-mwi
Voice Mail Number: 5187200
Routing Digits (e.g. AAR/ARS Access Code): 100

Send Reroute Request: y

LWC Reception: none
AUDIX Name:
Messaging Server Name:
```

- Setup a coverage path for the subscriber's extensions. Assign to it the hunt group created in the previous step.

```
display coverage path 2
```

COVERAGE PATH			
Coverage Path Number: 2			
Next Path Number:		Hunt after Coverage? n Linkage	
 COVERAGE CRITERIA			
Station/Group Status	Inside Call	Outside Call	
Active?	n	n	
Busy?	y	y	
Don't Answer?	y	y	Number of Rings: 4
All?	n	n	
DND/SAC/Goto Cover?	y	y	
 COVERAGE POINTS			
Terminate to Coverage Pts. with Bridged Appearances? n			
Point1: h5	Point2:	Point3:	
Point4:	Point5:	Point6:	

5.2 CONFIGURING ADDITIONAL MAS

The following tasks must be performed for each additional MAS.

- Add Message Server to the node names
 - Create H.323 signaling group
 - Change trunk group to include new signaling group ports.
-
- Add the new MAS to the IP Nodes Names

```
change node-names ip
```

Page 1 of 1			
IP NODE NAMES			
Name	IP Address	Name	IP Address
default	0 .0 .0 .0		. . .
clan1	148.147.6 .246		. . .
medpro1	148.147.6 .163		. . .
msgserver	148.147.6 .123		. . .
msgserverB	148.147.6 .125		. . .

- Create H.323 signaling group. Follow the same instructions as the first MAS.
- Add the new signaling group channels to the previously created trunk group. In order to distribute the load between MASs the ports must be administered in a circular fashion (round robin).

add trunk-group 299					Page 6 of 22	
					TRUNK GROUP	
					Administered Members (min/max): 1/3	
GROUP MEMBER ASSIGNMENTS					Total Administered Members: 3	
	Port	Code Sfx	Name	Night	Sig	Grp
1:	ip		ms1-1		299	
2:	ip		ms2-1		300	
3:	ip		ms3-1		301	
4:	ip		ms1-2		299	
5:	ip		ms2-2		300	
6:	ip		ms3-2		301	
..						
58:	ip		ms1-60		299	
59:	ip		ms2-60		300	
60:	ip		ms3-60		301	

5.3 SUBSCRIBER ADMINISTRATION

Subscriber administration has two parts: Administering the MWI, and assigning the call coverage path.

Follow these steps to program the subscribers stations assigned to the MAS:

change station 7175					Page 1 of 5	
					STATION	
Extension: 7175					Lock Messages? n	BCC: 0
Type: 6416D+					Security Code:	TN: 1
Port: 02B0209					Coverage Path 1: 2	COR: 1
Name: MM User					Coverage Path 2:	COS: 1
					Hunt-to Station:	
STATION OPTIONS						
Loss Group: 2					Personalized Ringing Pattern: 1	
Data Option: none					Message Lamp Ext: 7175	
Speakerphone: 2-way					Mute Button Enabled? y	
Display Language: english					Expansion Module? n	
					Media Complex Ext:	
					IP SoftPhone? n	
					Remote Office Phone? n	

change station 7175	Page 2 of 5
---------------------	-------------

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Per Station CPN – Send Calling Number must be set to “y” for the integration to work properly.

FEATURE OPTIONS		STATION
LWC Reception: none		Auto Select Any Idle Appearance? n
LWC Activation? y		Coverage Msg Retrieval? y
LWC Log External Calls? n		Auto Answer: none
CDR Privacy? n		Data Restriction? n
Redirect Notification? y		Idle Appearance Preference? n
Per Button Ring Control? n		
Bridged Call Alerting? y		Restrict Last Appearance? y
Active Station Ringing: single		
H.320 Conversion? n	►	Per Station CPN - Send Calling Number? y
Service Link Mode: as-needed		
Multimedia Mode: basic		Audible Message Waiting? n
MWI Served User Type: qsig-mwi		Display Client Redirection? n
		Select Last Used Appearance? n
		Coverage After Forwarding? s
Automatic Moves: no		Multimedia Early Answer? n
		Direct IP-IP Audio Connections? n
		IP Audio Hairpinning? n

Note: Per Station CPN should be set to Y.

Single Line sets should have field “**Message Waiting Indicator**” set to “**led**” or “**neon**,” depending on the type of telephone set used. Also, the “**Number of Rings**” field should be set to a minimum of 4 rings, to allow Personal Assistance to work properly.

Save these PBX changes.

Please refer to section 8.0 at the end of this document for special PBX programming considerations.

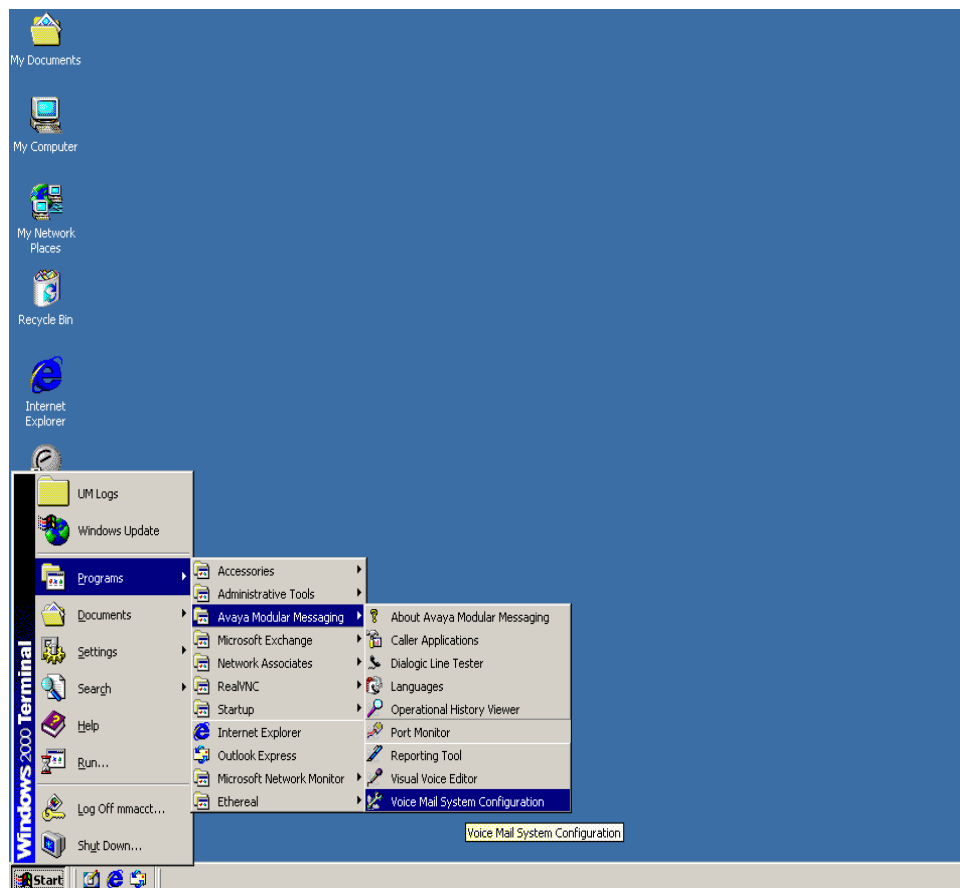
Configuring the Message Application Server

6.0 CONFIGURING THE MESSAGE APPLICATION SERVER

Configuring the MAS platform for proper PBX integration requires configuring several menus accessed within the **Voice Mail System Configuration** application, and a certified MM engineer. This must be performed for each MAS Voice Mail Domain (VMD).

PLEASE READ TEXT below the example illustrations/screens in this section. The entries needed for your system may be different than those shown in the example screens.

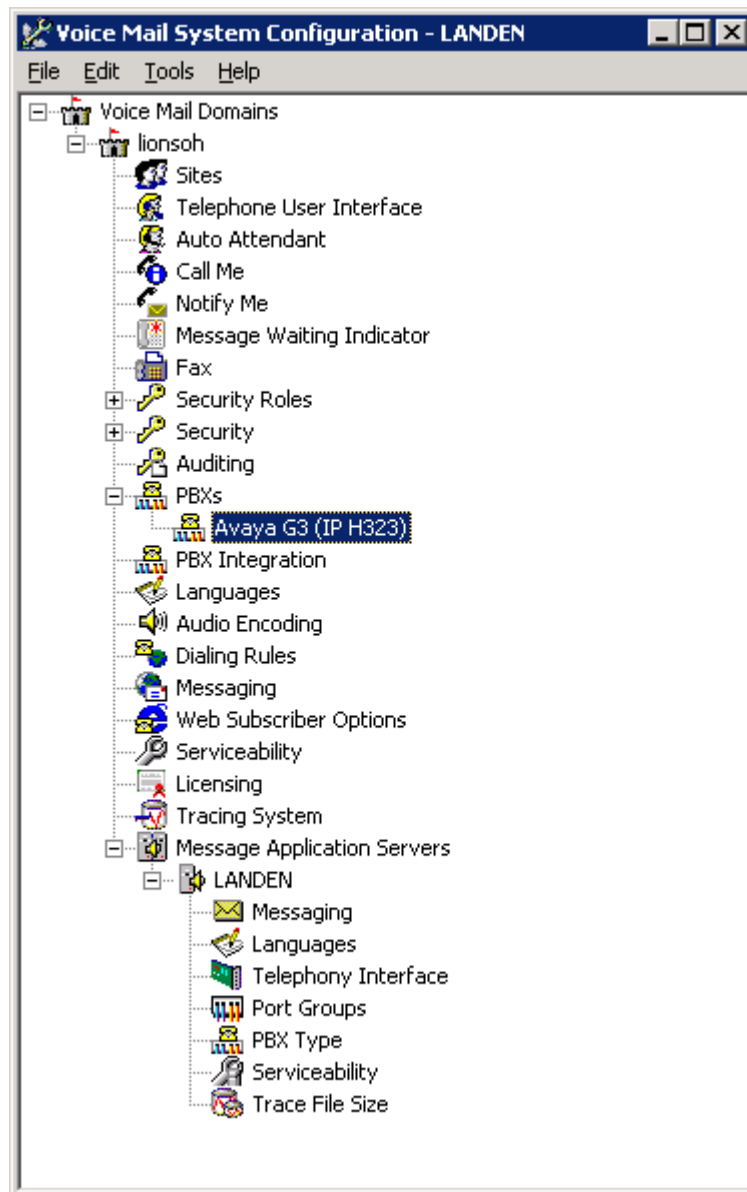
- Access the **Voice Mail System Configuration** application from the MAS program group:



Avaya H.323 Integration

Expand all fields so all-applicable options are visible.

Note: Starting with MM 5.0 additional Fields such as *Sites and PBX Integration* will appear on the VMSC screen.

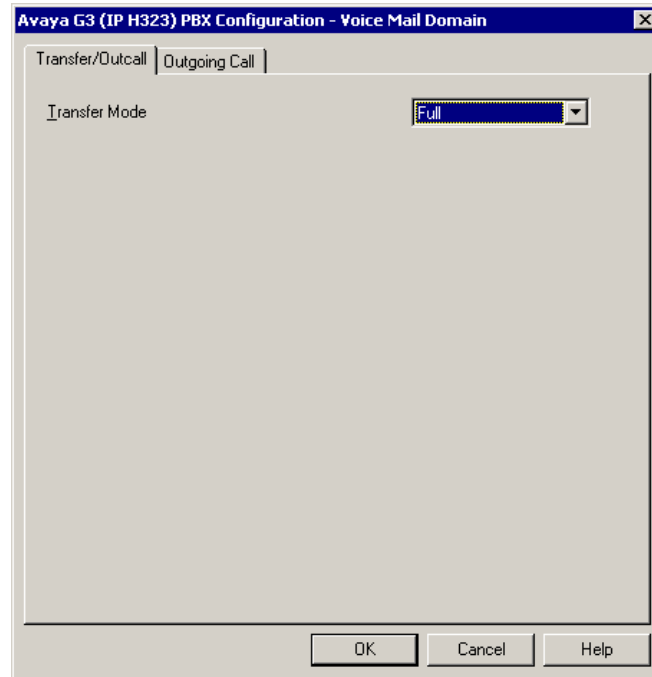


Ensure the new PBX is added as instructed by the Modular Messaging Installation guide. The new PBX should be:
Avaya G3 or MV (IP H323)

Select **Voice Mail Domains**

1. Expand **PBXs**
2. Select (double click) the **Avaya G3 or MV (IP H323) PBX**
3. Access the **Transfer/Outcall** tab

Transfer Mode = Full



NOTE: Administer transfers as FULL (Supervised transfer) to prevent callers from being disconnected when calls are re-routed back to the MAS. Transfers should only be administered as blind or partial when the transferred to numbers will not be re-routed to the MAS.

The following programming is a continuation from the Modular Messaging (MAS section) Installation Guide:

- Next access the **Outgoing Call** tab

The screenshot shows a dialog box titled "Avaya G3 (IP H323) PBX Configuration - Voice Mail Domain". It has three tabs: "Transfer/Outcall", "Tone Detection", and "Outgoing Call", with the "Outgoing Call" tab selected. The dialog contains the following fields:

- Layer1 Protocol**: A dropdown menu showing "G.711 u-law".
- BC Transfer Cap**: A dropdown menu showing "Speech".
- Number Type**: A dropdown menu showing "Local".
- Number Plan**: A dropdown menu showing "Private".
- Origin Number**: A text field containing "5187200".

At the bottom of the dialog are three buttons: "OK", "Cancel", and "Help".

1. **Layer 1 Protocol = G711u-law***
2. **Number Type = Local**
3. **Number Plan = Private**
4. **Origin Number = 5187200** (The number entered here should be the number entered in the "Voice Mail Number" field on page 2 of the Hunt Group form)
5. Select **OK** to save changes

* **u-law** is used in No. America & Japan; **a-law** primarily in Europe.

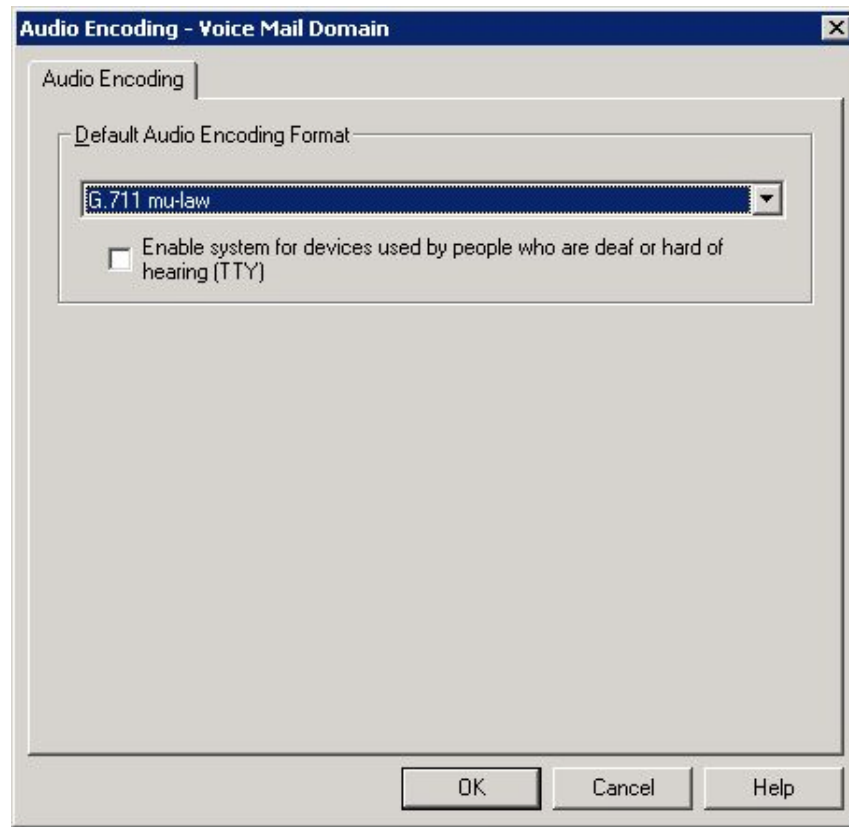
IMPORTANT: Confirm these values with PBX administrator/technician. All settings must match, or transfers and outcalling will not work.

Note: The MAS will prompt to restart the services. Wait until instructed below.

Avaya H.323 Integration

Next under Voice Mail Domain, select **Audio Encoding** and make sure the selection for Default Audio Encoding Format matches what you chose as the Layer 1 Protocol in the Outcalling Tab on the previous page.

Note: This format must be one of the codecs in the ip-codec-set in the PBX for this integration. Refer to *page 11 in this CN*.



- Next access the **Message Waiting Indicator (MWI)** tab

The screenshot shows a Windows-style dialog box titled "Message Waiting Indicator - Voice Mail Domain". It has two tabs: "General" and "Update Schedule", with "Update Schedule" currently selected. In the "Update Schedule" tab, there are several configuration options: a checked checkbox for "Enable Message Waiting Indicator (MWI)", a text field for "MAS MWI server" containing "LANDEN", a dropdown menu for "Scheduled MWI updates" set to "Active", an unchecked checkbox for "Limit requests", and a text field for "Maximum requests per minute" containing "60". Below these is a list box titled "Message Application Servers that support MWI" which contains the entry "LANDEN". At the bottom of the dialog are three buttons: "OK", "Cancel", and "Help".

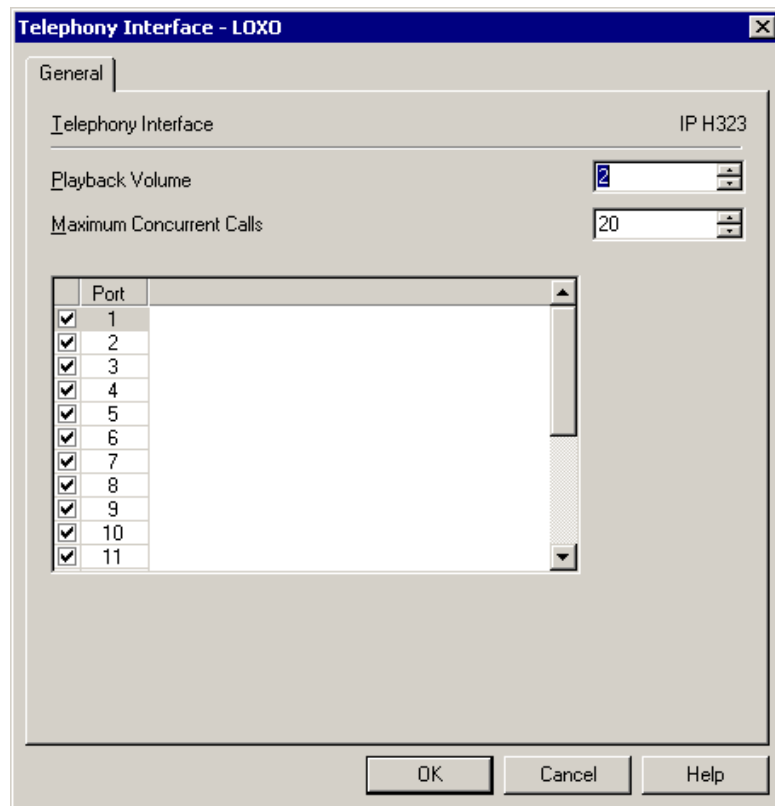
1. **Enable Message Waiting Indicator (MWI)** = Enable by checking the box
2. **MAS MWI Server** = Enter the name of the MWI server created during the installation procedure.
3. **Scheduled MWI updates: Active or Inactive** = Configure as per customer requirements.*
4. **Limit requests** = Leave Unchecked
5. **Maximum requests per Minute** = <grayed out>
6. **Message Application Servers that Support MWI** = This box should contain a list of MAS servers capable of placing MWI requests.
7. Select **OK** to save changes

*Note: The Scheduled MWI updates parameter is only available on MM 3.x

- Next access the **Telephony Interface (IP H323)**
- 1. **Playback Volume** = 2 (Default)
- 2. **Number of Ports** = **20** (maximum per MAS for MM2.0 or earlier)
-or- **30** (maximum per MAS for MM 3.0) **see note*

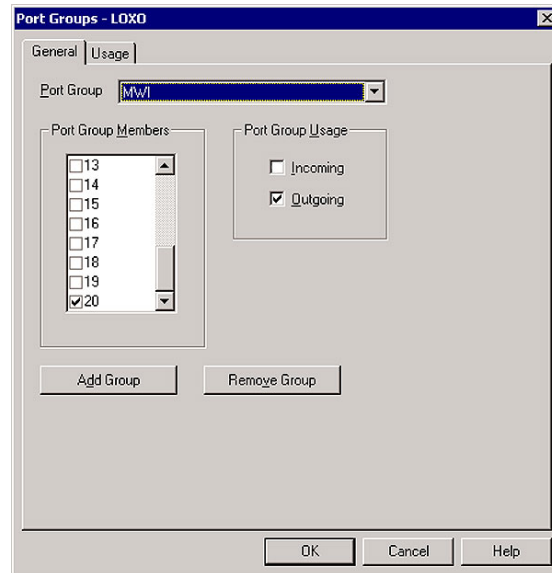
Important: The Ports are enabled by default. The MAS service must be restarted to allow port enabling/disabling.

***NOTE:** The max of 30 ports can only be attained when using an **MM 3.0 with an Avaya S3500 server**. (For additional details please refer to the *Avaya MM Concepts and Planning Guide*).

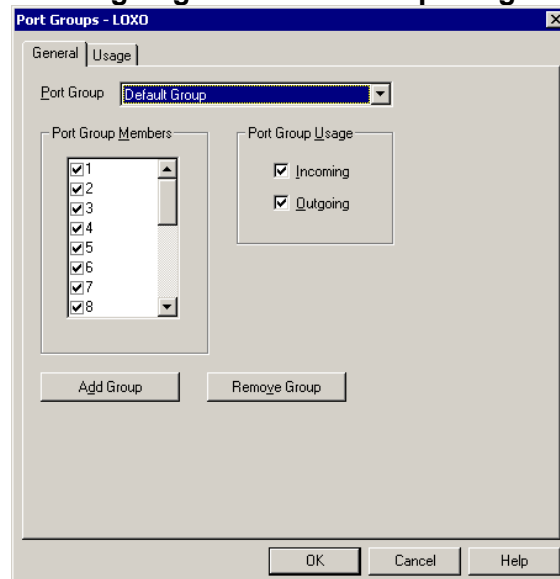


3. Select **OK** to save changes
4. Restart the MAS Service and then continue with the step below.

- Next access the **Port Groups** tab under the MAS name
 1. Click **Add Group** Button
 2. Name Group **MWI**
 3. Uncheck all of the ports, except the ports you will be using for MWI (please also refer to [Consideration 8.7](#))
Note: The MWI port Group Usage should be checked [✓] outgoing only.



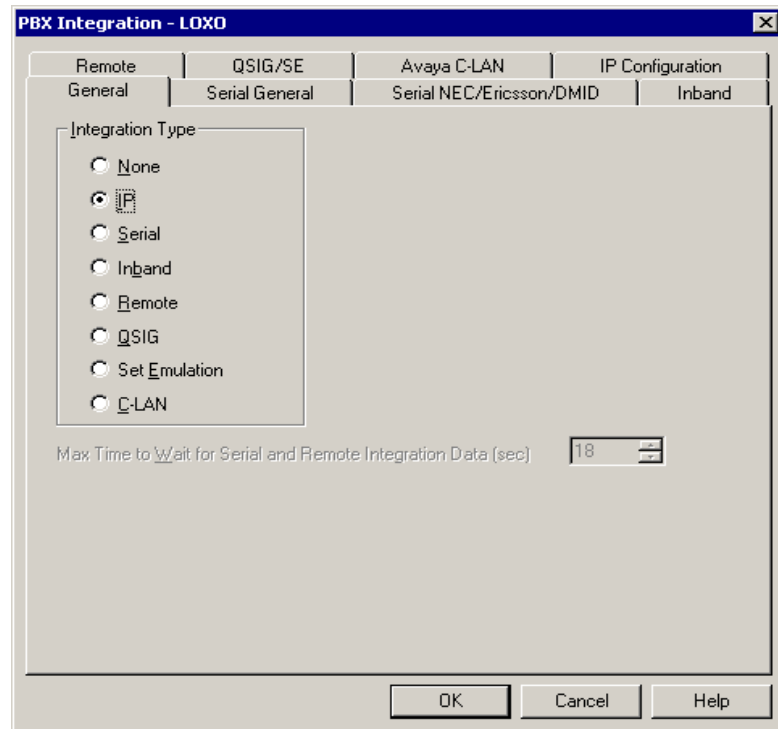
4. Then select the **Default Group** under **Port Groups** and ensure it is configured to meet the customer's need for **Incoming** and **Outgoing** under **Port Group Usage** and by checking all **Ports**



5. Select **OK** to save changes

- Next access **PBX Integration**

1. Within the **General** tab select **IP** for the **Integration Type**



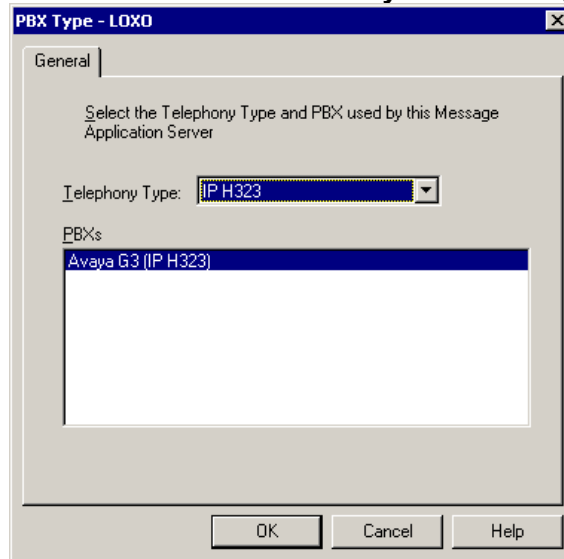
2. Next select the **IP Configuration** tab

The screenshot shows the 'PBX Integration - LOXO' dialog box with the 'IP Configuration' tab selected. The dialog has a tabbed interface with the following tabs: General, Serial General, Serial NEC/Ericsson/DMID, Inband, Remote, QSIG/SE, Avaya C-LAN, and IP Configuration. The IP Configuration tab is active, showing the following fields and controls:

- MAS Corporate IP Address: 148 . 147 . 22 . 21
- PBX IP Address: 148 . 147 . 22 . 54
- Port: 1720
- UDP Port Range: 5000 To: 5999
- Packet size (bytes): 30
- Enable Tunneling: ☒
- Enable Fast Start: ☒
- Silence Suppress: ☐
- Max MWI Sessions: 1
- Port Group Name: MWI (dropdown menu)
- IP Supported Codecs: G.711-uLaw-64k, G.711-ALaw-64k
- Buttons: Move Up, Move Down, Add..., Remove
- Bottom Buttons: OK, Cancel, Help

3. **Host IP Address** = IP address assigned to the **MAS CORP IP** (corporate network)
4. **PBX IP Address** = IP address assigned to the **PBX C-LAN** card or the **PROCR** (S8300)
5. **Port** = 1720
6. **UDP Port Range** = 5000-5999
7. **Packet size (bytes)** = 30
8. **Link test extension** = Leave blank
9. **Enable Tunneling** = Check box
10. **Enable Fast Start** = Check box
11. **Silence Suppress** = Uncheck box
12. **Max MWI Sessions** = 1
13. **Port Group Name** = MWI
14. **IP Supported Codec's** = Highlight G.711-uLaw-64k and Move Up
15. Select **OK** to save changes

- Next access the **General** tab within the **PBX Type** tab
 1. Scroll under the **Telephony Type** = IP H323
 2. Under **PBXs** ensure **Avaya G3 or MV (IP H323)** is selected



3. Select **OK** to save changes

After making these changes, return to “Configuring the voicemail system” within the S3400 Message Server Installation guide. Ensure you restart the Message Application Server services to apply these changes.

Important notes regarding this integration

8.0 CONSIDERATIONS

- 8.1 **Before performing the installation ensure the customer site has passed the Avaya Network Evaluation.**
- 8.2 **To minimize Quality Of Service (QoS) issues it is strongly recommended that the MAS and the IP PBX be connected on the same sub-net and to the same IP Network Switch.**
- 8.3 **To prevent callers from being disconnected or receiving wrong integration information when callers are re-routed back to VS, administer transfers as FULL.** Transfers should only be administered as blind or partial when the transferred to number is not re-routed to the VS.
- 8.4 **Currently Fax is not supported.**
- 8.5 **Support for Transfer to QSIG Voice Mail feature.** In order to provide support for the Transfer to QSIG Voice Mail feature on your QSIG integrated MAS solution, you need to ensure the following criteria are met:
 - Your switch will need to be running Definity software R10 or later.

- You will need to have the following features enabled on your switch:
 - o ISDN-PRI or ISDN-BRI
 - o QSIG Supplementary Services
 - o Transfer to QSIG Voice Mail
- A Feature Access code for Transfer to QSIG Voice Mail will need to be configured.
- Your MAS Voice Server will need to be using the QSIG-MWI hunt group integration as described in this document.

Example on how the feature works:

- Extension 7001 has voice mail but calls received are forwarded to extension 7002 rather than directly to voice mail.
- Callers to 7001 are answered at 7002 but the caller wishes to leave a voice mail for the person at 7001.
- The person at 7002 transfers the caller directly to the voice mail for 7001 by pressing the transfer button, then the code for transfer to voice mail (*8,*50, etc.), then presses transfer again.
- **The caller immediately hears the greeting of 7001 (no additional ringing) and can subsequently leave a message.**

For full details on how to configure and implement the Transfer to QSIG Voice Mail feature on your Avaya switch, please consult your Avaya Definity Sales or Support representative.

8.6 When multiple Avaya PBX's are arranged in a QSIG network, care must be taken to configure the QSIG tie trunks properly. In order to provide full feature functionality to all subscribers, the trunk group(s) assigned to the QSIG tie trunks connecting all Avaya PBX's in the network must match the configuration of the trunk group form (page 1 and 2 of the form) assigned to the MAS QSIG trunks. An example of the trunk group administration form is illustrated on the PBX programming section of this document.

8.7 When running MWI on your Voice Messaging Server (MAS) you should setup a separate Port Group dedicated for MWI. If you are using a single MAS, one port is adequate. For multiple MAS you may need to increase the number of MWI ports to 2 or 3; or if using MM 3.0 or higher, you can create an MWI port group on each MAS to handle MWI needs. Only outgoing needs to be checked [✓] in the Port Group Usage field. Additionally, **if you are experiencing glare** you should dedicate these ports for MWI by not including them in any other port group. Note: **Once the port group is created, this port group name must be entered in the PBX Integration -> IP Configuration tab -> Port Group Name field.**

8.8 Possible reasons for QSIG Path Replacement failure:

Note: Path Replacement is a function of the PBX. There are no Path Replacement settings on MM.

- When the QSIG Message Application Server Auto Attendant, Caller Application or Find Me features are transferring a call, you will see a second channel in Port Monitor appear busy until the transfer has been completed. After the transfer has been completed you will see that both channels are now idle in Port Monitor. This shows that the QSIG Path Replacement feature has completed successfully.
- However there are some switch configurations that will cause Path Replacement to fail, therefore your MM Server will stay bridge onto the transferred call keeping two channels busy in Port Monitor. The following list of features can cause Path Replacement to fail:
 - **Call Vector.** A call that has experienced SS-CT (Supplementary Service Call Transfer) that terminated to a vector and received answer treatment can have its path replaced. This is allowed only after a true user answers the call so that vector processing is completed. A vector step of recorded announcement or wait listening to music could cause a CONNECT message to be sent back and make the far end think ANF-PR (Path Replacement Additional Network Feature) could take place. If the PR Propose is received while still in vector processing, the vectoring PBX will deny the ANF-PR attempt but will initiate its own attempt at ANF-PR when a user does answer the call.
 - **Malicious Call Trace (MCT).** While Malicious Call Trace (MCT) is active and if there is a trunk involved in the call, MCT feature does not allow the trunk resources to be released from the switch-side to facilitate the tracing activity. Also, a MCT controller on switch A may request a controller on switch B to continue tracing a call that was tandem through switch B by providing the trunk member id they wish to have traced. So, ANF-PR shall not be done while MCT is active on the call.
 - **Restriction Features.** *Class of Restriction (COR).* The restrictions placed on routing calls are in affect for ANF-PR. Any call that cannot be originated or terminated because of COR on a regularly dialed basis will not be originated or terminated when that call is made ON BEHALF OF that terminal by ANF-PR.
 - *Voice Terminal Restrictions.* Voice Terminal Restrictions for the reroute of an ANF-PR call will be enforced.

- *Inward.* If a terminal is Inward restricted then that terminal would not be able to accept an incoming call. This would include a new path SETUP message. Therefore a Requesting PBX should not bother proposing ANFPR, since it is destined to fail.
- *Manual Terminating Line.* If a terminal is Manual Terminating restricted then that terminal would not be able to accept an incoming call except from an attendant. Thus a new path SETUP would be denied. Therefore a Requesting PBX should not bother proposing ANF-PR, since it is destined to fail.
- *Origination.* If a terminal is Origination restricted then that terminal would not be able to make an outgoing call. Thus originating a new path SETUP should be denied. Therefore a Cooperating PBX should reject the path replacement proposal/request.
- *Outward.* If a station is outward restricted at the cooperating end of an ANF-PR call then no ANF-PR SETUP should be attempted and ANF-PR should fail. Any call that cannot be originated outward (because of COR) on a regularly dialed basis will not be allowed to originate outward when that call is made ON BEHALF OF that terminal by ANF-PR.
- *Termination.* If a terminal is Termination restricted then that terminal would not be able to accept an incoming call. Thus an incoming new path SETUP would be denied. Therefore a Requesting PBX should not bother proposing ANF-PR, since it is destined to fail.
- *TAC (Trunk Access Code).* When an outgoing call is made using a TAC, or a call was extended by an attendant using DTGS (Direct Trunk Group Selection), the user has intentionally chosen a particular Trunk Group for the outgoing call. ANF-PR will not replace the path in this case.

8.9 H.323 integration (QSIG Protocol) does not support forwarding/transfer from a Vector. Currently, if calls are routed from a Vector to the QSIG link(s) connected to the MAS, the call will not pass the VDN as the called party ID. Applications requiring calls that are routed from Vectors to mailboxes on the MAS can be configured so as to route calls to phantom extensions (X-ports) configured to call-cover all-calls to the MAS hunt group.

Note: Patch 7960 corrects this. Avaya CM 2.0.1 and later releases include this fix/patch.

8.10 The Communication Manager does not support call queuing on QSIG trunks. Hence, calls cannot be queued to MAS ports. The user audible

behavior is that during peak traffic, when all MAS ports are busy, a caller will hear a fast busy. They should hang up and try at a later time.

- 8.11 H.323 integrations may not be reliable for TTY** if the IP network is unable to support uncompressed audio with no packet loss. For this reason we currently do not support TTY with this integration.
- 8.12 Call transfers may not display the Call ID to ringing phones.** The Call ID is not provided until the subscriber answers the phone.
- 8.13 Trunk-to-Trunk is not required to support Find Me if the minimum releases indicated below are met.** Previously, when a public network call arrived at an Avaya™ Communication Manager system and was routed via coverage to a QSIG trunk connected to a Modular Messaging system equipped with the Find Me feature, the Find Me feature would place the call to the user and connect the calling and called parties. When the Communication Manager received the Transfer Complete messages from the Messaging system, and Path Replacement was enabled on the Communication Manager, it would proceed with the Path Replacement. While performing this task, the Trunk-to-Trunk Transfer parameter would be checked, and if set to “none”, the call between the calling party and the found user would be torn down. A change was put into the following releases and load numbers to correct this:
- 1.3 Load 537.0
 - 2.0 Load 226.0
 - 2.1 Load 411.0
- When the Communication Manager System is running on one of these loads or a later one, the Path Replaced call will not be torn down.
- 8.14 The Communication Manager supports up to 28 lines of input within the DCS to QSIG TSC Gateway.** This limitation affects how many entries can be configured for remote locations in a centralized voice mail environment.
- 8.15 AUDIX TUI and CALL SENDER Feature** – When using an AUDIX TUI the Call Sender feature will not function. Instead the user will hear the message, “This call is experiencing difficulties. Please try again later. Please disconnect.” And the call will then be terminated. The cause of this is currently under investigation. Please contact an integration specialist for any updated information.
- 8.16 Re-Route Request set to “y”** (page 2 of the Hunt Group Form) **eliminates potential call tromboning** where 3-legs may remain active and Path Replacement in effect appears to have failed. Should remote sites calling the

MM experience call delay or calls not completing this may be due to a re-route request being sent back to the remote and that site not being administered with a dial plan to recognize the MM pilot number. Changing Send Re-Route Request to “n” is a common solution but recreates the tromboned call. To avoid issues it is best to have a proper dial plan so the remote PBX knows how to reroute the call when requested.

- 8.17 MM Caller Applications with Vectors** — when using MM Caller Applications with Vectors, we have found that in some cases Path Replacement may not occur. We have found that adding a “wait 10 seconds hearing music” step to the beginning of a vector” provides the time needed to hold the call in vector processing and allow path replacement to occur before hitting the messaging step. This answer supervision is to so we can path replace the call, before the vector sends the call anywhere else. No other option on the wait step will work.
- 8.18** In some instances when a call is transferred and the answering party places the call on hold too quickly, it may cause Path Replacement to fail. This will be seen as a QSIG Return Error generated by the Avaya CM. When the error reaches MM, the MM system does not tandem it back to the originating port on the CM thereby causing an issue that leads to the call being dropped. This issue was specific to the H.323 integration and was corrected in MM 4.0 SP 4 and MM 5.1.

9.0. ADDENDUM ON CONFIGURING MULTIPLE PBXS (DCS+ NETWORKING/INTERNETWORKING)

The following information is not intended for new installations. This Addendum assumes the customer has an existing Network already in place. Please refer to the appropriate PBX installation guide for brand new networking installations. Obtain a configuration printout of the existing network to use as a reference.

Ensure the integration is working properly within the PBX where the MAS will reside (Hub Node) before continuing with the networking configuration. Configure the remote switch (i.e. Node 2), Definity G3, Prologix, etc., by following the screens below.

☐ The Hub Node will not require changes to the Trunk Group; however configure the Signaling Group, which will be assigned to the DS1 channels. The following is an example of the changes highlighted in **boldface**:

NOTICE:

The screens in this Config Note are only for illustration purposes.

It is recommended that a qualified technician review the customer's CM QSig programming for accuracy.

change signaling-group 7				Page 1 of 5	
SIGNALING GROUP					
Group Number: 7		Group Type: isdn-pri			
Associated Signaling? y		Max number of NCA TSC: 16			
Primary D-Channel: 01A0824		Max number of CA TSC: 31			
				Trunk Group for NCA TSC: 4	
Trunk Group for Channel Selection: 4					
Supplementary Service Protocol: a					

change signaling-group 7				Page 2 of 5	
ADMINISTERED NCA TSC ASSIGNMENT					
Service/Feature:			As-needed Inactivity Time-out (min):		
TSC	Local				Mach.
Index	Ext.	Enabled	Established	Dest. Digits	Appl. ID
1:	2299	y	permanent	2100	dcs 1
2:	3081	y	permanent	3083	qsig-mwi 2
3:		n			
4:		n			
5:		n			
etc..					

Note: The **Group Type** will depend on the customer's environment. The trunks can either be T1, E1, IP, etc.. For example, they could be:

Group Type: **isdn-pri** or Group Type: **h.323**

Additionally, you should consult with a Software Specialist to ensure the Numbering Format of the trunk group is configured appropriately to route Call ID to the Modular Messaging from remote switches.

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- ☐ On the Hub Node program the ISDN QSIG TO DCS TSC GATEWAY. This defines the stations that are DCS to another Node (in our example DCS Node 2). This allows MWI to be directed via DCS to Node 2 from the MAS.

```
change isdn qsig-dcs-tsc-gateway
```

QSIG TO DCS TSC GATEWAY					
Subscriber Number	Sig Grp	TSC Index	Subscriber Number	Sig Grp	TSC Index
63xx	7	2			
6409	7	2			
6411	7	2			
6412	7	2			
6415	7	2			
6430	7	2			

Important: Use caution when completing this task. The data in this field will display exactly as it is entered. This is critical because when the switch makes a selection it will use the first match.

- ☐ On the Hub Node program the ISDN DCS TO DCS TSC GATEWAY. This defines the stations using DCS to another Node (in our example DCS Node 2). This allows incoming calls from DCS Node 2 to be directed to the MAS.

```
change isdn dcs-qsig-tsc-gateway
```

DCS TO QSIG TSC GATEWAY									
				AAR/ ARS					AAR/ ARS
Mach ID	Sig Grp	TSC Index	Voice Mail Number	Access Code	Mach ID	Sig Grp	TSC Index	Voice Mail Number	Access Code
2	7	2	4575678	107					

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- ☐ The Remote switch (DCS Node 2) does not require changes on the DCS Trunk Group incoming to the Hub Node. However the Signaling Group for this Trunk Group requires changes. The following is an example of the changes highlighted in **boldface**:

change signaling-group 7		Page 1 of 5
SIGNALING GROUP		
Group Number: 7	Group Type: isdn-pri	
Associated Signaling? y	Max number of NCA TSC: 16	
Primary D-Channel: 01A0824	Max number of CA TSC: 31	
Trunk Group for Channel Selection: 7	Trunk Group for NCA TSC: 7	
Supplementary Service Protocol: a		

display signaling-group 7		Page 2 of 5
ADMINISTERED NCA TSC ASSIGNMENT		
Service/Feature:		As-needed Inactivity Time-out (min):
TSC	Local	Mach.
Index	Ext.	Enabled
1:	2299	y
2:	6451	y
3:		n
4:		n
5:		n
etc..		

- ☐ Create a Hunt Group for messaging from the remote Node.

display hunt-group 2		Page 1 of 10
HUNT GROUP		
Group Number: 2	ACD? n	
Group Name: S3400 VOICEMAIL REMOTE	Queue? n	
Group Extension: 6300	Vector? n	
Group Type: ucd-mia	Coverage Path:	
TN: 1	Night Service Destination:	
COR: 8	MM Early Answer? n	
Security Code:		
ISDN Caller Display:		

display hunt-group 2		Page 2 of 10
HUNT GROUP		
Message Center: rem-vm		
Voice Mail Extension: 3000		
Send Reroute Request: y		
Calling Party Number to INTUITY AUDIX? y		
LWC Reception: none		

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Note: Ensure the voice mail extension is the pilot number of the voicemail system and not the lead number of the qsig-mwi hunt group at the host site.

```
display hunt-group 2                                     Page 3 of 10

                                HUNT GROUP
      Group Number: 2      Group Extension: 6300      Group Type: ucd-mia
Member Range Allowed: 1 - 200      Administered Members (min/max): 0 /0
                                Total Administered Members: 0

GROUP MEMBER ASSIGNMENTS
      Ext      Name (24 characters)      Ext      Name (24 characters)
1 :
2 :
3 :
4 :
5 :
6 :
7 :
8 :
9 :
10 :
11 :
12 :
13 :
14 :
15 :
16 :
17 :
18 :
19 :
20 :
21 :
22 :
23 :
24 :
25 :
26 :

At End of Member List
```

☐ The key in the remote Node is the method the remote Voice Mail Extension of the Hunt Group routing is configured. Change the Uniform Dial Plan and add RNX 457 and direct this to the AAR

```
display uniform-dialplan 0                               Page 1 of 2
                                UNIFORM DIAL PLAN TABLE
                                Percent Full: 0

Matching      Insert      Node      Matching      Insert      Node
Pattern  Len Del Digits Net Conv Num  Pattern  Len Del Digits Net Conv Num
2          4  0  221  aar  n          2          4  0  221  aar  n
30         4  0  221  aar  n          30         4  0  221  aar  n
3000      4  0  457  aar  n          3000      4  0  457  aar  n
31         4  0  221  aar  n          31         4  0  221  aar  n
5          4  0  221  aar  n          5          4  0  221  aar  n
63         4  0          ext  n          63         4  0          ext  n
64         4  0          ext  n          64         4  0          ext  n
          n          n          n          n          n          n
          n          n          n          n          n          n
          n          n          n          n          n          n
```


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- ☐ In the remote Node AAR Analysis table route 4573000 to route pattern 2.

display aar analysis 2						Page	1 of	2
AAR DIGIT ANALYSIS TABLE								
Percent Full:								9
Dialed	Total	Route	Call	Node	ANI			
String	Min Max	Pattern	Type	Num	Reqd			
4573000	7 7	2	aar	1	n			

- ☐ In the remote Node Route Pattern 2 insert the MAS AAR Access code of 107.

display route-pattern 2														Page	1 of	3
Pattern Number: 2																
Grp. No.	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/ IXC								
			Mrk	Lmt	List	Del	Digits	QSIG								
							Dgts	Intw								
1:	7	0					107							y		user
2:														n		user
3:														n		user
4:														n		user
5:														n		user
6:														n		user
	BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	BAND	No.	Numbering	LAR					
	0	1	2	3	4	W	Request		Dgts	Format						
									Subaddress							
1:	y	y	y	y	y	n	y	as-needed	rest			unk-unk				none
2:	y	y	y	y	y	n	n		rest							none
3:	y	y	y	y	y	n	n		rest							none
4:	y	y	y	y	y	n	n		rest							none
5:	y	y	y	y	y	n	n		rest							none
6:	y	y	y	y	y	n	n		rest							none

- ☐ Create a Call Coverage Path that will be assigned to the subscribers' stations. This Call Coverage Path will have the Remote Voice Mail Hunt Group as the Call Coverage Point. Below is an example of a Call Coverage Path.

```
display coverage path 3
```

COVERAGE PATH			
Coverage Path Number: 3		Hunt after Coverage? n	
Next Path Number:		Linkage	

COVERAGE CRITERIA			
Station/Group Status	Inside Call	Outside Call	
Active?	n	n	
Busy?	y	y	
Don't Answer?	y	y	Number of Rings: 4
All?	n	n	
DND/SAC/Goto Cover?	y	y	

COVERAGE POINTS		
Terminate to Coverage Pts. with Bridged Appearances? n		
Point1: h2	Point2:	Point3:
Point4:	Point5:	Point6:

- ☐ Configure the remote subscriber stations, assigning the newly created Call Coverage Path to them. They should also have LWC and the AUDIX Name set to AUDIX.

- continued on next page -

CHANGE HISTORY

Revision	Issue Date	Reason for Change
Version L	11/10/04	Added a new field on page 2 of Hunt group from: "Send Reroute Request". Also added an "Overview" section.
Version M	02/01/05	Removed support of the IP600 (S8100) PBX
Version N	03/15/05	Included MWI Limit Request programming, Included NOTE on page 3, updated 8.12
Version O	05/02/05	Updated page 2 of the Trunk Group.
Version P	09/02/05	Updated remote subscribers' configuration.
Version Q	09/23/05	Clarified UDP Port usage (5000-5999)
Version R	10/18/05	Updated to support ACM 3.0
Version S	2/17/06	Added Consideration 8.15 regarding Call Sender issue with AUDIX TUI
Version T	03/30/06	Updated Consideration 8.11 regarding TTY support
Version U	4/11/06	Added: <ul style="list-style-type: none"> MM 3.0 to support release section 2.0 New MWI screen shot with Scheduled MWI updates parameter noted for MM3.0
Version V	5/22/06	Changed Max port noted on page 24 from 20 to 30. Added Note to explain how to achieve 30 ports.
Version W	6/22/06	Changed IP Configuration tab screen shot (page 27) so packet size now shows 30 to match packet size as shown in item number 7 just below same screen shot.
Version X	6/29/06	Added note (in red) to Consideration 8.9
Version Y	8/11/06	Revised consideration 8.7 for clarity and added note to MWI port group usage settings (Section 6.0)
Version Z	9/01/06	Re-revised consideration 8.7, corrected MWI example screen Section
Version AA	1/24/07	Added note regarding Interworking in Section 5.0
Version AB	2/1/07	Added screens for <i>system-parameters coverage-forwarding</i> with explanation of <i>Maintain SBA at Principal</i> and <i>CCRON</i> enabled feature with explanation. Changed in Section 5.0: <ul style="list-style-type: none"> Added note for multiple length extensions on system-parameter features screen. Added sidebar adjacent signaling group form to explain NCA-TSC, CA-TSC, and Trunk Group for NCA-TSC Section 6.0 - Changed information clarifying what origin number entry should be on outgoing tab for PBX Type
Version AC	4/12/07	Added sidebar about CCRON needing to be turned on in system-parameters customer-options screen in addition to the one already placed adjacent the <i>display system-parameters coverage-forwarding</i> screen.
Version AD	6/1/07	Updated Consideration 8.12

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Version AE	6/20/07	Added note regarding Hyper-Threading systems in Section 2.0
Version AF	10/29/07	Highlighted CPN set to Y on station form in section 5.1 and added note for same.
Version AG	11/08/2007	Added new screens for Avaya CM 4.0 and related private-numbering format; updated sidebars; changed Dial Access parameter in trunk group screen to N.
Version AH	12/03/2007	Changed Consideration 8.12.
Version AI	05/05/2008	Updated to support MM 4.0
Version AJ	11/14/08	Added note about setting "Per Station CPN – Send Calling Number?" on Station Form to Yes in Section 5.0
Version AK	12/17/08	Updated screen shot for outgoing Tab in PBX Configuration in section 6.0 with updated Number Type and Number Plan as Local and Private respectively.
Version AL	2/02/09	Updated to support MM 5.0; Changed Limit Request for MWI in Section 6.0;
Version AM	02/12/09	Changed screen shot for VMSC and added sidebar in Section 6.0.
Version AN	03/18/09	Changed "Send Reroute Request" to y on the Hunt Group Form in Section 5.0; Added Consideration 8.16 regarding Re-Route Request
Version AO	04/08/09	Added Consideration 8.17 regarding potential issues that may arise when using MM Caller Applications with Vectors. Added Consideration 8.18 regarding potential Path Replacement Failure with transfer when the answering party places call on hold too quickly.
Version AP	07/09	Updated to support MM 5.1; changed wording in note on sidebar for VMSC in Section 6.0.
Version AR	01/19/10	Added note indicator to titles of Section 2.0 and 3.1; added corresponding note in sidebar; removed word "supported" in same sections for MAS releases and PBX Software releases. Updated section 3.1

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