**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**

With E1 QSIG integration, one digital pathway between the Avaya™ PBX and the Avaya Message Application Server (MAS) transmits both call information and voice communications. The pathway is provided by an ISDN digital link (QSIG), which provides channels that connect to the Dialogic E1 card. Within the D-Channel, routing information is sent to the MAS containing information regarding the source of the call with reason codes. The MAS processes call information from the supplementary code in the D-Channel, which routes call reasons directly to mailboxes. Message-Waiting indication is set and canceled using the supplementary code service. Voice is carried through the system in digital format.

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**Disclaimer:** Configuration Notes are designed to be a general guide reflecting AVAYA Inc. experience configuring its systems. These notes cannot anticipate every configuration possibility given the inherent variations in all hardware and software products. Please understand that you may experience a problem not detailed in a Configuration Note. If so, please notify the Technical Assistance Center at (800) 876-2835, and if appropriate we will include it in our next revision. AVAYA Inc. accepts no responsibility for errors or omissions contained herein.
2.0 AVAYA MESSAGE APPLICATION SERVER REQUIREMENTS

- Dialogic D/600JCT-1E1, or D/600JCT-2E1, or D/300JCT-E1*
- CT Bus cable (only required for multiple card installation)
- Supported Releases: 1.1, 2.0, 3.0, 3.1, 4.0, 5.x

* Not all cards support Fax. Please consult the MM Concepts and Planning Guide for details.

3.0 PBX HARDWARE REQUIREMENTS

- TN464 or MM710 circuits one per each Dialogic E1 Interface Card
- Cables:
  - RJ45 on the Dialogic (cable depends on PBX connection)

3.1 PBX SOFTWARE REQUIREMENTS

- Minimum Supported Software:
  - **Definity G3 & Prologix:** G3V10.1 Load 43
  - **S8x00/S87x0:** V1.1, V2.0, V3.x, V4.0

**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.

- ISDN PRI
- Private Networking
- Basic Call Setup
- Basic Supplementary Services
- Supplementary Services with Rerouting
- 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.
4.0 SUPPORTED INTEGRATION FEATURES

[✓] Items are supported

System Forward to Personal Greeting
- All Calls [✓]
- Ring/no answer [✓]
- Busy [✓]

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 [✓]
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.
5.0 CONFIGURING THE PBX TO INTEGRATE

The following programming is intended for certified PBX technicians/engineers.
The screens shown in this section are taken from an Avaya PBX administration
terminal. Some parameters may not appear on all software releases.

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. (see note on left)

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**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.

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**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.

---

The above information is provided by AVAYA Inc. as a guide. See disclaimer on page 1
### QSIG Optional Features

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Basic Call Setup?</td>
<td>y</td>
</tr>
<tr>
<td>Basic Supplementary Services?</td>
<td>y</td>
</tr>
<tr>
<td>Centralized Attendant?</td>
<td>y</td>
</tr>
<tr>
<td>Interworking with DCS?</td>
<td>n*</td>
</tr>
<tr>
<td>Supplementary Services with Rerouting?</td>
<td>y</td>
</tr>
<tr>
<td>Transfer into QSIG Voice Mail?</td>
<td>y</td>
</tr>
<tr>
<td>Value-Added (VALU)?</td>
<td>y</td>
</tr>
</tbody>
</table>

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

Define the QSIG TSC Extension and MWI Number of Digits per Voice Mail Subscriber.

The following is an example of the form with the required features in **boldface**.

### Change System-Parameters Features

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>QSIG TSC Extension:</td>
<td>2998</td>
</tr>
<tr>
<td>MWI - Number of Digits Per Voice Mail Subscriber:</td>
<td>4 (see note below)</td>
</tr>
</tbody>
</table>

**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.
Also access the System Parameter Coverage-Forwarding and change or ensure the following parameters are set appropriately:

**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.

☐ Change features and assign your private network access code, in this example we assigned 107
Install the E1 circuit pack, making sure they are “strapped” for 30-channel operation (E-1). Configure the DS1 circuits as follows:

```
display ds1 01b17
  DS1 CIRCUIT PACK

Location: 01B17                    Name: E1 QSIG
Bit Rate: 2.048                   Line Coding: hdb3
Signaling Mode: isdn-pri          Interface: peer-master
Connect: pbx                      Peer Protocol: Q-SIG
CentreVu Long Timers? n           Side: a
Interworking Message: PROGress    CRC? n
Interface Companding: alaw        Idle Code: 1111111
DCP/Analog Bearer Capability: 3.1kHz
Slip Detection? n                 Near-end CSU Type: other
```

Configure a Signaling Group that will be assigned to the DS1 channels. The Signaling Group should be configured as follows:

```
display signaling-group 1
  SIGNALING GROUP

Group Number: 9                  Group Type: isdn-pri
Associated Signaling? y          Max number of NCA TSC: 10
Primary D-Channel: 01b1716       Max number of CA TSC: 10
Trunk Group for NCA TSC          Trunk Group for NCA TSC: 1
Trunk Group for Channel Selection: 1  X-Mobility/Wireless Type: NONE
Supplementary Service Protocol: b
```

**Important:** The Max number of NCA TSC (Non Carrier Associated Temporary Signaling Channels) should be 1 or higher. This allows a path (1 or more) for MWI to operate.

---

**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.

The above information is provided by AVAYA Inc. as a guide. See disclaimer on page 1.
Create a trunk group, and assign the newly created DS1 channels to it. If the MAS will be configured to perform outcalls, ensure that the COR (Class of Restriction) assigned to this trunk group allows for outside trunk access. The trunk group must be configured as follows:

<table>
<thead>
<tr>
<th>Display Trunk Group 1</th>
<th>Page 1 of 22</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Trunk Group</strong></td>
<td></td>
</tr>
<tr>
<td>Group Number: 1</td>
<td></td>
</tr>
<tr>
<td>Group Name: <strong>QSIG - E1</strong></td>
<td></td>
</tr>
<tr>
<td>Direction: <strong>two-way</strong></td>
<td></td>
</tr>
<tr>
<td>Dial Access? n</td>
<td></td>
</tr>
<tr>
<td>Queue Length: 0</td>
<td></td>
</tr>
<tr>
<td>Service Type: <strong>tie</strong></td>
<td></td>
</tr>
<tr>
<td>Outgoing Display? y</td>
<td></td>
</tr>
<tr>
<td>Busy Threshold: 99</td>
<td></td>
</tr>
<tr>
<td>Carrier Medium: <strong>PRI/BRI</strong></td>
<td></td>
</tr>
<tr>
<td>TestCall BCC: 4</td>
<td></td>
</tr>
<tr>
<td>Codeset to Send Display: 6</td>
<td></td>
</tr>
<tr>
<td>Supplementary Service Protocol: b</td>
<td></td>
</tr>
<tr>
<td>Trunk Hunt: <strong>cyclical</strong></td>
<td></td>
</tr>
<tr>
<td>Digital Loss Group? n</td>
<td></td>
</tr>
<tr>
<td>Disconnect Supervision - In? y</td>
<td></td>
</tr>
<tr>
<td>Answer Supervision Timeout: 0</td>
<td></td>
</tr>
</tbody>
</table>

**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 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 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 the number. If there is no entry in this table to build the number, NO number is sent.

For Avaya CM 4.0 and later see related note below

**NOTE 1:** **NCA-TSC Trunk Member** should have the highest timeslot from the upper E1 span in the trunk group. For example, if there is one E1 span in the MAS then it should be 29, and if there are 2 spans then its should be 59.

**NOTE 2:** 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 Specialist for proper programming.
There should be 29 total ports for a single E1 span per MAS.

There should be 59 total ports if two E1 spans per MAS.

Do not administer (add) the highest order timeslot/port as a member in the trunk group.

- For an MAS with a single E1 card **Do Not** add port 30. Only 29 ports should be in the trunk group.
- For an MAS equipped with two E1 cards **Do Not** add Port 60. Only 59 ports should be in the trunk group.

The reason for this is due to a limitation with the Dialogic drivers it requires that we leave out the highest order timeslot to enable MWI, even though MWI is sent on the D-Channel.

Therefore, only 29 ports are available on an MAS with one E1 card, or 59 ports on an MAS with two E1 cards.

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**Note:** For full N+1 support in your MM environment you should set aside a number of ports from each MAS in your environment to be allocated for each of the different port types. When planning trunk group configuration for your multiple MAS environment care should be taken to ensure the same number of ports are allocated to the incoming trunk group for each MAS in your environment. For example, if MAS1 has 16-ports allocated to the incoming trunk group, MAS2 should also have 16-ports allocated to the incoming trunk group.
When running in a multiple MAS environment with N+1 support, you should setup the trunks groups as follows:

**Incoming Trunk group:** should be configured containing all of the required incoming ports from each MAS in your environment, the ports should be entered into the trunk group so Trunk group port-1 is the first port on MAS1, Trunk group port-2 is the first port on MAS2, Trunk group port-3 is the second port on MAS1, etc.

**Outgoing Trunk group:** should be configured with all of the required outgoing ports from each MAS in your environment, used for MM client outcalls and TUI transfers.

Refer to Consideration 8.10 for additional N+1 programming.

☐ 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**.

```plaintext
display isdn private-numbering

ISDN NUMBERING - PRIVATE FORMAT

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

PBX Identifier: 0
Deleted Digits: 0

The above information is provided by AVAYA Inc. as a guide. See disclaimer on page 1
```
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.

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.
Create a Route Pattern for the trunk group that was previously created for the DS1 channels. The Route Pattern must be configured as follows:

```
display route-pattern 1
Pattern Number: 12
Grp. FRL NPA Pfx Hop Toll No. Inserted
No. Mrk Lmt List Del Digits DCS/ IXC Intw
Dgts
1: 1 0 3 y user
2: 2 0 3 (this is an example,
y user
3: see note below)
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 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**.

```
display aar analysis 15
Percent Full: 3
Dialed Total Route Call Node ANI
String Min Max Pattern Type Num Req'd
4573000 7 7 1 aar n
```

**NOTE:** Avaya IP phones may experience DTMF failure when dialing the Hunt Group. They can dial the UDP number or the AAR # above to bypass the Hunt Group and use DTMF capability.
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. 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:

<table>
<thead>
<tr>
<th>Group Number: 1</th>
<th>ACD? n</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group Name: QSIG E1</td>
<td>Queue? n (see note below)</td>
</tr>
<tr>
<td>Group Extension: 3000</td>
<td>Vector? n</td>
</tr>
<tr>
<td>Group Type: ucd-mia</td>
<td>Coverage Path:</td>
</tr>
<tr>
<td>TN: 1</td>
<td>Night Service Destination:</td>
</tr>
<tr>
<td>COR: 1</td>
<td>MM Early Answer? n</td>
</tr>
<tr>
<td>Security Code: ISDN Caller Display: grp-name</td>
<td></td>
</tr>
</tbody>
</table>

**Note:** Queue? should be set to “n” as recommended. Refer to Consideration 8.9 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 minus the number of leading digits administered in the route-pattern to delete. In our example the Voice Mail Number is “4573000” and the number of digits to delete in the route-pattern is 3; therefore, the number entered in the “Outgoing Call” tab must be “3000.”

☐ Create a Call Coverage Path that will be assigned to the subscribers’ stations. This Call Coverage Path will have the MAS Hunt Group as the Call Coverage Point. Below is an example of a Call Coverage Path.

display coverage path 10

Coverage Path Number: 1
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
All?                   n              n
DND/SAC/Goto Cover?    y              y

Number of Rings: 4

COVERAGE POINTS

Terminate to Coverage Pts. with Bridged Appearances? n

Point1: h1        Point2:        Point3:        Point4:        Point5:        Point6:
Configure the subscriber stations, assigning the newly created Call Coverage Path (i.e. 10) to them. Also, ensure the following setting for subscribers:

<table>
<thead>
<tr>
<th>Extension: 3002</th>
<th>Lock Messages? n</th>
<th>BCC: 0</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type: 8410D</td>
<td>Security Code:</td>
<td>TN: 1</td>
</tr>
<tr>
<td>Port: 01A0102</td>
<td>Coverage Path 1: 10</td>
<td>COR: 1</td>
</tr>
<tr>
<td>Name: Smith John</td>
<td>Coverage Path 2:</td>
<td>COS: 1</td>
</tr>
</tbody>
</table>

**STATION OPTIONS**
- Loss Group: 2
- Data Module? n
- Display Module? y
- Speakerphone: 2-way
- Display Language: english

**STATION**

**FEATURE OPTIONS**
- LWC Reception: none
- Auto Select Any Idle Appearance? n
- LWC Activation? n
- Coverage Msg Retrieval? y
- CDR Privacy? n
- Auto Answer? none
- Redirect Notification? y
- Idle Appearance Preference? n
- Per Button Ring Control? n
- Bridged Call Alerting? n
- Restrict Last Appearance? y
- Active Station Ringing: single
- H.320 Conversion? n
- Per Station CPN – Send Calling Number? y

**MWI**
- Served User Type: qsig-mwi
- Display Client Redirection? n
- Select Last Used Appearance? n
- Coverage After Forwarding? s

**Note:** With QSIG integration the following fields are required for Message Waiting:

- **LWC Reception:** none
- **MWI Served User Type:** qsig-mwi

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 the Consideration section at the end of this document for special PBX programming considerations.
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.

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

**Special Note**: Verify the CRC value in the QTE.PRM is set correctly. If left uncorrected the Dialogic E1 boards will not initialize properly. The file is found in:

```
C:\Program Files\Dialogic\data\qte.prm
```

Open the qte.prm file. About 23 lines into the file text you will see a setting under the line as show in bold below.

```plaintext
;--- CRC ENABLE SWITCH (Parameter type 0FH - E1 only)
;--- Turns the transmission of the CRC4 pattern on or off and searches for such a pattern in the received signal.
;--- Possible values for the data are as follows:
;--- 00H = Turn CRC off.
;--- 01H = Turn CRC on.
;--- The default data value is 00H (Turn CRC off).
000F 01
```

The example number shown above in blue is incorrect. Change this from 000F 01 to 000F 00.

Save then file and then reboot the MAS.
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 (QSIG)**
1. Select the **Voice Mail Domain**
2. Expand **PBXs**
3. Select the newly created **Avaya G3 (Dialogic QSIG)**
4. Access the **General (QSIG) PBX Configuration** tab
5. **DTMF Inter-Digit Delay during Dialing (ms) = 80**
6. **DTMF Length during Dialing (ms) = 80**
7. **DTMF Length during Detection (ms) = 50**
Next access the **Transfer/Outcall** tab
**Transfer Mode** = Blind

Next access the **Tone Detection** tab
**Maximum Silence before Hanging Up (ms)** = 6000
Next access the **Outgoing Call** tab

1. **Layer Protocol** = G.711 A-Law
2. **BC Transfer Cap** = Speech
3. **Number Type** = Unknown
4. **Number Plan** = Unknown
5. **Origin Number** = 4573000 (The number entered here should be the number entered in the “Voice Mail Number” field on page 2 of the Hunt Group form)
6. Select **OK** to save changes

**Note:** The Layer 1 Protocol fields should match the Interface Companding setting selected on Page 1 of the DS1 Circuit Pack configuration screen on Page 7 of this configuration note.
Next access the **Message Waiting Indicator (MWI) tab**

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

**Support NOTE** for MM 1.1 & 2.0 only:
- Limit Requests should be checked.
- Maximum Requests per Minute should be set for 200.
Tip: To make the QSIG or set emulation telephony interface active, click the down arrow and click Make Active. If the QSIG or set emulation telephony interface is already active, this field does not appear.

- Next access the General tab within the Telephony Interface (Dialogic-QSIG) tab

1. **Playback Volume** = 2
2. **Maximum Concurrent Calls** = Enter the number of ports connected to the PBX (i.e. 29)
3. **Port** = Ports are enabled by default
   - **Note:** The MAS service must be restarted to allow port disabling
4. Select OK to save changes

**Note:** For a QSIG link, the total number of ports specified must occupy the first available set of contiguous channels. For example, if there are two boards on a system, one with 10 ports enabled and one with 29 ports enabled, then the first 10 ports will be enabled on one boards, and all 29 on the second board. Use the port checkboxes to enable or disable the correct ports.

The above information is provided by AVAYA Inc. as a guide. See disclaimer on page 1
Next access the Port Groups tab under the MAS name.

1. Click Add Group button
2. Name Group MWI
3. Within the new MWI Port Group uncheck all Ports except the MWI port. This will be the upper most port of the trunk group. Port 29 on a single board or Port 59 if 2 boards (see side NOTE). This port can receive calls, so ensure it is programmed for both Incoming and Outgoing under the Port Group Usage.
4. Return to the Port Groups General tab and ensure the Default Group under Port Groups is configured to meet the customer’s need for Incoming and Outgoing under Port Group Usage.

**NOTE:** The MWI port within the MWI Port Group is used by the MWI sub-system to control concurrent MWI requests. This does not affect incoming/outgoing traffic to the port in anyway.
5. Access the **Port Group** General tab and check all **Ports** (1-29 for a single span or 1-59 if 2 spans) including the MWI (29 or 59) port.

6. Select **OK** to save changes

**NOTE:** This Port Group configuration applies to each MAS

- continued on next page –
Next access the General tab within the PBX Integration

1. **QSIG** = Enable by checking the box

2. Access the QSIG/SE tab
3. Port Group Name = MWI
4. Max MWI Sessions = 1
5. Indicator On/Off signals must use same port = Leave Blank
6. MWI On Field = Leave as default (can’t be changed)
7. MWI Off Field = Leave as default (can’t be changed)
8. Select OK to save changes

- Next access the General tab within the PBX Type tab
  1. Telephony Type = Dialogic QSIG
  2. Under PBXs ensure Avaya G3 (Dialogic QSIG) is selected
  3. Select OK to save changes

After making these changes, return to “Configuring the voicemail system” within the Message Application Server Installation Guide. Ensure you are prompted to restart the Message Application Server services to apply these changes.
8.0 CONSIDERATIONS/ALTERNATIVES

**QSIG Integration 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.

**Outcalls to pagers placed over analog trunks may fail.** If the CO connecting the analog trunks to the PBX does not provide answer supervision, the MAS will not out-pulse DTMF digits to the pager terminal. This problem can be eliminated by installing a Call Classifier board in the PBX (if one is not already installed), enabling system parameter customer-option “Answer Supervision by Call Classifier”, and enabling “Answer Supervision” in the Trunk Group associated with the outgoing analog trunks accessed during the outcalls. Outcalls over digital trunks are not affected.

**Transfers to ringing use additional ports.** When performing unsupervised transfer, and the transferred-to extension forwards back to the MAS, additional ports are tied up on the MAS server, as “Path Replacement” does not occur. Two additional ports are used for each number dialed through the automated attendant, and all these ports are in use until the transferred call is answered or the caller disconnects from the message server. Customers should consider implementing supervised transfers, or installing additional ports. Note that with supervised transfers, callers are not provided with music on hold, but are instead prompted to wait during the silence. The called party will hear a “Connecting” prompt as he/she answers the call.

**When multiple 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 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.

8.5 **Message Waiting Interrogation** is a feature that allows the PBX to request message waiting indicator status of subscribers’ stations from the MAS. The following conditions will result in the PBX requesting message waiting indicators status:

- **Switch reload.** 15 minutes after a switch reload, a message waiting interrogation request will be sent to the MAS.

The above information is provided by AVAYA Inc. as a guide. See disclaimer on page 1.
- ** Interruption of service on a QSIG link.** 15 minutes after service is re-established on the QSIG link, a message waiting interrogation request is sent to the MAS.
- **Scheduled daily maintenance routine.** Once daily, during the scheduled maintenance period (usually around 1:00 A.M. on most switches), a message waiting interrogation request is sent to the MAS.

### 8.6 MAS can support multiple Avaya PBX’s in a “centralized” voice mail arrangement only if all PBX’s are connected using QSIG. If Avaya G3/MultiVantage PBX’s uses DCS for networking, QSIG integration cannot provide full functionality to remote PBX subscribers. Please refer to Section 9.0 to support a centralized environment using DCS+. When running MWI on your Voice Server you should setup a separate trunk group containing 2 channels from one of your QSIG trunks, these channels should not be apart of any other trunk group. You should then create a new MWI port group containing these channels and make the relevant changes in MAS VMCS.

### 8.7 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 R8 or later.

1. You will need to have the following features enabled on your switch:
   - ISDN-PRI or ISDN-BRI
   - QSIG Supplementary Services
   - Transfer to QSIG Voice Mail
2. A Feature Access code for Transfer to QSIG Voice Mail, will need to be configured.
3. 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.

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

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 initiated 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 ANF-PR, 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 allow 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 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.

8.10 When configuring an N+1 MM environment the following changes should be made to the MAS Service in the services applet, all steps should be completed:
1. Double click the Monitor icon on the desktop.
2. Select Services
3. Locate the Message Application Service
4. Right click the Message Application Service and select properties from the menu.
5. Select the Recovery tab.
6. For First Failure select Run File.
7. Select the Browse button and locate the QSIGRecover file:
   \Avaya_Support\Tools\QSIGRecover\QSIGRecover.exe
8. Enter the following in the Command Line Parameters box:
   If you have one E1 board in your MAS:
   /recover /boards 1
   If you have two E1 boards in your MAS:
   /recover /boards 2
   If you have three E1 boards in your MAS:
   /recover /boards 3
9. Check the Append fail count to end of command line checkbox.
10. Repeat steps 6 to 8 for Second Failure.
11. For Subsequent Failures select Reboot The Computer.
12. Select OK to save changes.
13. Select Run from the Start Menu
14. Enter cmd and select OK
15. At the command prompt navigate to the Avaya_Support\Registry_Keys folder.
16. Enter the following: stopdriversonshutdown.reg and press the return key.
   Note: This registry file will apply a change to the registry key to stop the Dialogic drivers when the MAS service is stopped.
17. Close the command prompt.

8.11 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.

8.12 Call transfers may not display the Call ID to ringing phones. The Call ID is not provided until the subscriber answers the phone.

- This issue was resolved in MM 3.0 where the Calling Party Number (number only, not the name) is displayed.
- Starting with MM 5.1, both name and calling party number will be displayed on ringing phones. Please note, for name to be displayed the PBX must send the MAS the Calling Name using the QSIG Name Identification Supplementary Services Protocol (ECMA-164).

8.13 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.

8.14 If the Avaya CM is networked to a non-Avaya PBX using H.323 trunking, please consult PSN001841u.

8.15 If the Avaya CM is networked to non-Avaya PBXs that use MM for voice messaging the Avaya CM integrated to MM should be at the following minimum releases:

- Avaya CM 3.1.4 SP 2 or later
- Avaya CM 3.1.5 SP0 or later
- Avaya CM4.0.3 base or later
- Avaya CM 4.0.4 base or later
- Avaya CM 5.0 SP3 or later
- Avaya CM 5.1.0 base or later such as 5.1.1 and 5.1.2

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.
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**:

```
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):
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..
```

**Notice:** 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**
- **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.
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

<table>
<thead>
<tr>
<th>Subscriber Number</th>
<th>Sig Grp</th>
<th>TSC Index</th>
</tr>
</thead>
<tbody>
<tr>
<td>63xx</td>
<td>7</td>
<td>2</td>
</tr>
<tr>
<td>6409</td>
<td>7</td>
<td>2</td>
</tr>
<tr>
<td>6411</td>
<td>7</td>
<td>2</td>
</tr>
<tr>
<td>6412</td>
<td>7</td>
<td>2</td>
</tr>
<tr>
<td>6415</td>
<td>7</td>
<td>2</td>
</tr>
<tr>
<td>6430</td>
<td>7</td>
<td>2</td>
</tr>
</tbody>
</table>
```

**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

<table>
<thead>
<tr>
<th>AAR/ARS</th>
<th>Mach Sig</th>
<th>TSC</th>
<th>Voice Mail Access</th>
<th>AAR/ARS</th>
<th>Mach Sig</th>
<th>TSC</th>
<th>Voice Mail Access</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>ID Grp Index</td>
<td>Number</td>
<td>Code</td>
<td></td>
<td>ID Grp Index</td>
<td>Number</td>
<td>Code</td>
</tr>
<tr>
<td>2</td>
<td>7</td>
<td>2</td>
<td>4575678</td>
<td>107</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```
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
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
Supplementary Service Protocol: a

display signaling-group 7
ADMINISTERED NCA TSC ASSIGNMENT
Service/Feature:        As-needed Inactivity Time-out (min):
TSC  Local  Ext.  Enabled Established Dest. Digits  Appl.       ID
Index  Index  Index  Index  Index  Enabled  Established  Dest. Digits appl.  ID
1:  2299  y  permanent  2100  dcs  1
2:  6451  y  permanent  3081  audix  1
3:  n  n
4:  n
5:  n
etc..
```

Create a Hunt Group for messaging from the remote Node.

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

<table>
<thead>
<tr>
<th>Ext</th>
<th>Name (24 characters)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td></td>
</tr>
<tr>
<td>7</td>
<td></td>
</tr>
<tr>
<td>8</td>
<td></td>
</tr>
<tr>
<td>9</td>
<td></td>
</tr>
<tr>
<td>10</td>
<td></td>
</tr>
<tr>
<td>11</td>
<td></td>
</tr>
<tr>
<td>12</td>
<td></td>
</tr>
<tr>
<td>13</td>
<td></td>
</tr>
<tr>
<td>14</td>
<td></td>
</tr>
<tr>
<td>15</td>
<td></td>
</tr>
<tr>
<td>16</td>
<td></td>
</tr>
<tr>
<td>17</td>
<td></td>
</tr>
<tr>
<td>18</td>
<td></td>
</tr>
<tr>
<td>19</td>
<td></td>
</tr>
<tr>
<td>20</td>
<td></td>
</tr>
<tr>
<td>21</td>
<td></td>
</tr>
<tr>
<td>22</td>
<td></td>
</tr>
<tr>
<td>23</td>
<td></td>
</tr>
<tr>
<td>24</td>
<td></td>
</tr>
<tr>
<td>25</td>
<td></td>
</tr>
<tr>
<td>26</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Matching Pattern Length Del Digits Net Conv Node</th>
<th>Insert Length Del Digits Net Conv Node</th>
<th>Num</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>4</td>
<td>0</td>
</tr>
<tr>
<td>30</td>
<td>4</td>
<td>0</td>
</tr>
<tr>
<td>3000</td>
<td>4</td>
<td>0</td>
</tr>
<tr>
<td>31</td>
<td>4</td>
<td>0</td>
</tr>
<tr>
<td>5</td>
<td>4</td>
<td>0</td>
</tr>
<tr>
<td>63</td>
<td>4</td>
<td>0</td>
</tr>
<tr>
<td>64</td>
<td>4</td>
<td>0</td>
</tr>
</tbody>
</table>
In the remote Node AAR Analysis table route 4573000 to route pattern 2.

<table>
<thead>
<tr>
<th>Dialed String</th>
<th>Total Min</th>
<th>Max</th>
<th>Route Pattern</th>
<th>Call Type</th>
<th>Node Num</th>
<th>ANI Reqd</th>
</tr>
</thead>
<tbody>
<tr>
<td>4573000</td>
<td>7</td>
<td>7</td>
<td>2</td>
<td>aar</td>
<td>1</td>
<td>n</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Pattern Number: 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Grp. FRL NPA Pfx Hop Toll No.</td>
</tr>
<tr>
<td>1: 7 0</td>
</tr>
<tr>
<td>2:</td>
</tr>
<tr>
<td>3:</td>
</tr>
<tr>
<td>4:</td>
</tr>
<tr>
<td>5:</td>
</tr>
<tr>
<td>6:</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>BCC VALUE</th>
<th>TSC</th>
<th>CA-TSC</th>
<th>ITC BCIE Service/Feature</th>
<th>BAND No.</th>
<th>Numbering</th>
<th>LAR Dgts</th>
<th>Format Subaddress</th>
</tr>
</thead>
<tbody>
<tr>
<td>1: y y y y y n</td>
<td>y</td>
<td>as-needed</td>
<td>rest</td>
<td>unk-unk</td>
<td>none</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2: y y y y y n</td>
<td>n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3: y y y y y n</td>
<td>n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>4: y y y y y n</td>
<td>n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5: y y y y y n</td>
<td>n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6: y y y y y n</td>
<td>n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
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.

<table>
<thead>
<tr>
<th>Coverage Path Number: 3</th>
<th>Hunt after Coverage? n</th>
</tr>
</thead>
<tbody>
<tr>
<td>Next Path Number:</td>
<td>Linkage</td>
</tr>
</tbody>
</table>

**Coverage Criteria**

<table>
<thead>
<tr>
<th>Station/Group Status</th>
<th>Inside Call</th>
<th>Outside Call</th>
</tr>
</thead>
<tbody>
<tr>
<td>Active?</td>
<td>n</td>
<td>n</td>
</tr>
<tr>
<td>Busy?</td>
<td>y</td>
<td>y</td>
</tr>
<tr>
<td>Don’t Answer?</td>
<td>y</td>
<td>y</td>
</tr>
<tr>
<td>All?</td>
<td>n</td>
<td>n</td>
</tr>
<tr>
<td>DND/SAC/Goto Cover?</td>
<td>y</td>
<td>y</td>
</tr>
</tbody>
</table>

**Coverage Points**

<table>
<thead>
<tr>
<th>Terminate to Coverage Pts. with Bridged Appearances? n</th>
</tr>
</thead>
<tbody>
<tr>
<td>Point1: h2</td>
</tr>
<tr>
<td>Point2:</td>
</tr>
<tr>
<td>Point3:</td>
</tr>
<tr>
<td>Point4:</td>
</tr>
<tr>
<td>Point5:</td>
</tr>
<tr>
<td>Point6:</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Revision</th>
<th>Issue Date</th>
<th>Reason for Change</th>
</tr>
</thead>
<tbody>
<tr>
<td>Version J</td>
<td>05/14/04</td>
<td>Grammar corrections.</td>
</tr>
<tr>
<td>Version K</td>
<td>06/25/04</td>
<td>Added ISDN formatting table.</td>
</tr>
<tr>
<td>Version L</td>
<td>09/16/04</td>
<td>Updated to meet MM R2.0</td>
</tr>
<tr>
<td>Version M</td>
<td>09/23/04</td>
<td>Added Consideration 8.11 &amp; 8.12</td>
</tr>
<tr>
<td>Version N</td>
<td>10/02/04</td>
<td>Added Consideration 8.13</td>
</tr>
<tr>
<td>Version O</td>
<td>10/20/04</td>
<td>Changed the Port Group Usage of the MAS. Also clarified the MWI port programming.</td>
</tr>
<tr>
<td>Version P</td>
<td>11/18/04</td>
<td>Added a new field on page 2 of Hunt group from: &quot;Send Reroute Request&quot;. Also added an &quot;Overview&quot; section.</td>
</tr>
<tr>
<td>Version Q</td>
<td>01/19/05</td>
<td>Changed the MWI port for bidirectional usage.</td>
</tr>
<tr>
<td>Version R</td>
<td>02/01/05</td>
<td>Changed the Port Group requirements and reversed dedicating Outgoing ports. Also, MWI ports are not required to be for Outgoing only. Removed support of the IP600 (S8100) PBX</td>
</tr>
<tr>
<td>Version S</td>
<td>04/22/05</td>
<td>Included NOTE on page 3, updated 8.12</td>
</tr>
<tr>
<td>Version T</td>
<td>05/02/05</td>
<td>Updated page 2 of the Trunk Group, and changed Subscriber programming fields.</td>
</tr>
<tr>
<td>Version U</td>
<td>07/01/05</td>
<td>Clarification of port programming</td>
</tr>
<tr>
<td>Version V</td>
<td>09/02/05</td>
<td>Updated remote subscribers' configuration.</td>
</tr>
<tr>
<td>Version W</td>
<td>10/18/05</td>
<td>Updated to support ACM 3.0</td>
</tr>
<tr>
<td>Version X</td>
<td>01/26/06</td>
<td>Corrected parameters on various screens in Section 5.0.</td>
</tr>
</tbody>
</table>
| Version Y | 04/11/06  | Added:  
  * MM 3.0 to support release section 2.0  
  * New MWI screen shot with Scheduled MWI updates parameter noted for MM3.0 |
<p>| Version Z | 06/29/06  | Added note (in red) to Consideration 8.1; also removed confusing portion of side note regarding Trunk Group members on page 9 |</p>
<table>
<thead>
<tr>
<th>Version</th>
<th>Date</th>
<th>Changes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Version AA</td>
<td>12/04/06</td>
<td>Added note for Interworking Section 5.0</td>
</tr>
<tr>
<td>Version AB</td>
<td>2/01/07</td>
<td>Changed in Section 5.0:</td>
</tr>
<tr>
<td></td>
<td></td>
<td>- Added note for multiple length extensions on system-parameter features screen.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>- Added sidebar adjacent signaling group form to explain NCA-TSC, CA-TSC, and Trunk Group for NCA-TSC</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Section 6.0 - Changed information clarifying what origin number entry should be on outgoing tab for PBX Type</td>
</tr>
<tr>
<td>Version AC</td>
<td>4/12/07</td>
<td>Added sidebar about CCRON needing to be turned on in system-parameters customer-options screen in addition to the one already placed adjacent the display system-parameters coverage-forwarding screen.</td>
</tr>
<tr>
<td>Version AD</td>
<td>5/21/07</td>
<td>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.</td>
</tr>
<tr>
<td>Version AE</td>
<td>6/1/07</td>
<td>Updated Consideration 8.12</td>
</tr>
<tr>
<td>Version AF</td>
<td>10/29/07</td>
<td>Updated information for station form</td>
</tr>
<tr>
<td>Version AG</td>
<td>12/03/07</td>
<td>Changed Consideration 8.12 to note fixed in MM 3.0</td>
</tr>
<tr>
<td>Version AH</td>
<td>1/04/08</td>
<td>Added note regarding Dialogic cards in Section 2.0</td>
</tr>
<tr>
<td>Version AI</td>
<td>03/03/08</td>
<td>Updated Title to S87x0 / 8x00</td>
</tr>
<tr>
<td>Version AJ</td>
<td>05/05/08</td>
<td>Updated to support MM 4.0</td>
</tr>
<tr>
<td>Version AK</td>
<td>05/16/08</td>
<td>Removed Dialogic Driver release from Section 2.0, updated PBX software requirements section 3.1</td>
</tr>
<tr>
<td>Version AL</td>
<td>06/11/08</td>
<td>Updated sidebar note in Section 5.0 detailing administration of trunk group and to leave out highest order timeslot/port.</td>
</tr>
<tr>
<td>Version AM</td>
<td>07/28/08</td>
<td>Added Consideration 8.14</td>
</tr>
<tr>
<td>Version AN</td>
<td>11/14/08</td>
<td>Added note about setting “Per Station CPN – Send Calling Number?” on Station Form to Yes in Section 5.0</td>
</tr>
<tr>
<td>Version AO</td>
<td>2/02/09</td>
<td>Updated to support MM 5.0; Changed Limit Request for MWI in Section 6.0; Added Consideration 8.15</td>
</tr>
<tr>
<td>Version AP</td>
<td>02/12/09</td>
<td>Changed screen shot for VMSC and added sidebar in Section 6.0.</td>
</tr>
<tr>
<td>Version AQ</td>
<td>03/18/09</td>
<td>Added Consideration 8.16 regarding Re-Route Request</td>
</tr>
<tr>
<td>Version AR</td>
<td>04/08/09</td>
<td>Added Consideration 8.17 regarding potential issues that may arise when using MM Caller Applications with Vectors</td>
</tr>
<tr>
<td>Version AS</td>
<td>06/02/09</td>
<td>Added note about reviewing/changing qte.prm file at beginning of Section 6.0</td>
</tr>
<tr>
<td>Version AT</td>
<td>07/09</td>
<td>Updated to support MM 5.1; changed wording in note on sidebar for VMSC in Section 6.0; Added note regarding collocated PBX/MM on page 1</td>
</tr>
<tr>
<td>Version AU</td>
<td>07/20/09</td>
<td>Updated Consideration 8.12 to reflect calling party number/name display on ringing phone and PBX requirements needed</td>
</tr>
</tbody>
</table>

The above information is provided by AVAYA Inc. as a guide. See disclaimer on page 1