Avaya™

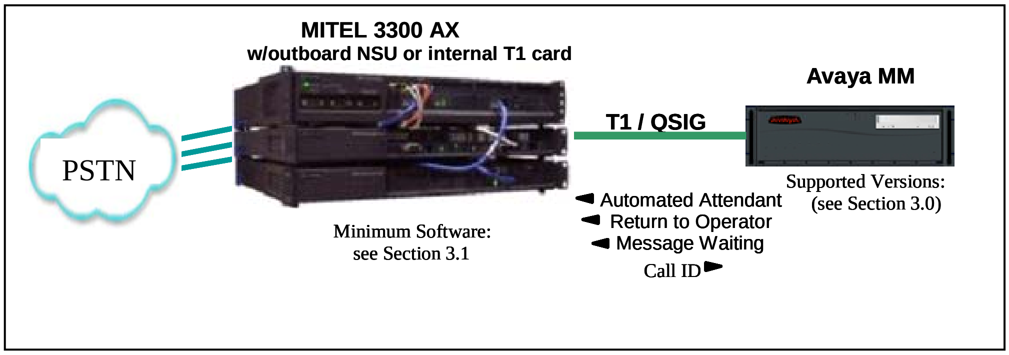
Modular Messaging

Configuration Note 88035 – Version D (1/10)

MITEL SX3300 AX Controller w/ NSU or Internal T1 Card

T1/QSIG

The PBX and MM are assumed to be collocated. For other configurations please consult with the Switch Integrations group.



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 MITEL PBX supported in this Configuration Note and the QSIG protocol.

Use this document in conjunction with *Modular Messaging Installation Guide* and the appropriate Mitel documentation.

Please read the entire document before attempting any configuration.

1.0 METHOD OF INTEGRATION

With T1 QSIG, one digital pathway between the PBX and Avaya Message Application Server transmits both call information and voice communications

With T1 QSIG integration, one digital pathway between the MITEL™ 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 T1 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.

MAS Requirements

[1] **Release 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

- Dialogic D/480JCT-1T1 or D/240JCT-T1
- CT Bus cable (only required for multiple card installation)
- Software Releases [1]: MM 3.x, 4.x, 5.x
- Dialogic Driver Version: 5.1.1 FP1 SU15

3.0 PBX HARDWARE REQUIREMENTS

- Mitel 3300 AX Controller
- Mitel outboard NSU* for T1 connectivity.

  *Note: Minimum firmware patch for NSU: 1.6.0.1x

  -or-
- Embedded Digital Trunk Module (EDT) for T1 connectivity.
  - The specific EDT part # depends 3300 model used. We have successfully used EDT P/N 50003560 w/Mitel 3300 Rel. 9.0.1.

Cables:

- RJ45 to RJ48C on the Dialogic (cable depends on PBX connections)

3.1 PBX SOFTWARE REQUIREMENTS

- Minimum Software [1]:
  - Mitel AX Controller 7.1 Version (w/NSU)
  - Mitel 3300 Release 9.0.1 with EDT *(see 3.0 PBX Hardware Reqs.)*

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Supported integration features

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 Indication (MWI)	[✓]
Multiple Call Forward	[✓]
Multiple Greetings	[✓]
N+1	[✓]
Outcalling	[✓]
Queuing	[ ]
Return to Operator	[✓]

* Untested

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.

NOTICE:

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

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

5.0 CONFIGURING THE PBX FOR INTEGRATION

The following programming is intended for certified PBX technicians/engineers. The information shown in this section is taken from MITEL PBX documentation and an implementation site. 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**.

The Mitel T1 QSIG integration with Modular Messaging required only minor changes to the Mitel 3300 AX default configuration. The following 5 items detail the changes required.

1) ARS Digit modification – Tone Plan Number – Included a 1 second pause before sending call data.
2) Enable Call Forward External in Station Class of Service (since we are forwarding to a trunk group)
3) Station Sets are forwarded to a Speed Dial Number which is pointed to the ARS number.
4) Straps on Comm Card were changed from "NT" to "LT"
5) Single line phones need to be enabled for MWI (display)

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- **Set up the T1 LINK**

-- Web Page Dialog

Link Descriptor Assignment

Number:	1
Address for Message Control:	A
BER - Maintenance Limit, 10**-n:	4
BER - Service Limit, 10**-n:	3
Data Call Alternate Digit Inversion:	⦿ No ○ Yes
Framing Losses in 24 hrs - Maintenance Limit:	255
Framing Losses in 24 hrs - Service Limit:	9000
Integrated Digital Access:	ISDN NODE
Vendor Inter-working Type:	
Satellite Link Delay:	⦿ No ○ Yes
Slip Rate - Maintenance Limit (slips/24hr.):	5000
Slip Rate - Service Limit (slips/24hr.):	7000
Alarm Debounce Timer - Service Limit (millisec.):	500
Voice Encoding:	ADI
Data Encoding:	Nil
QSIG Private Network Access:	○ No ⦿ Yes
Digital Link Fault Delay Timer (sec.):	240

Save Cancel

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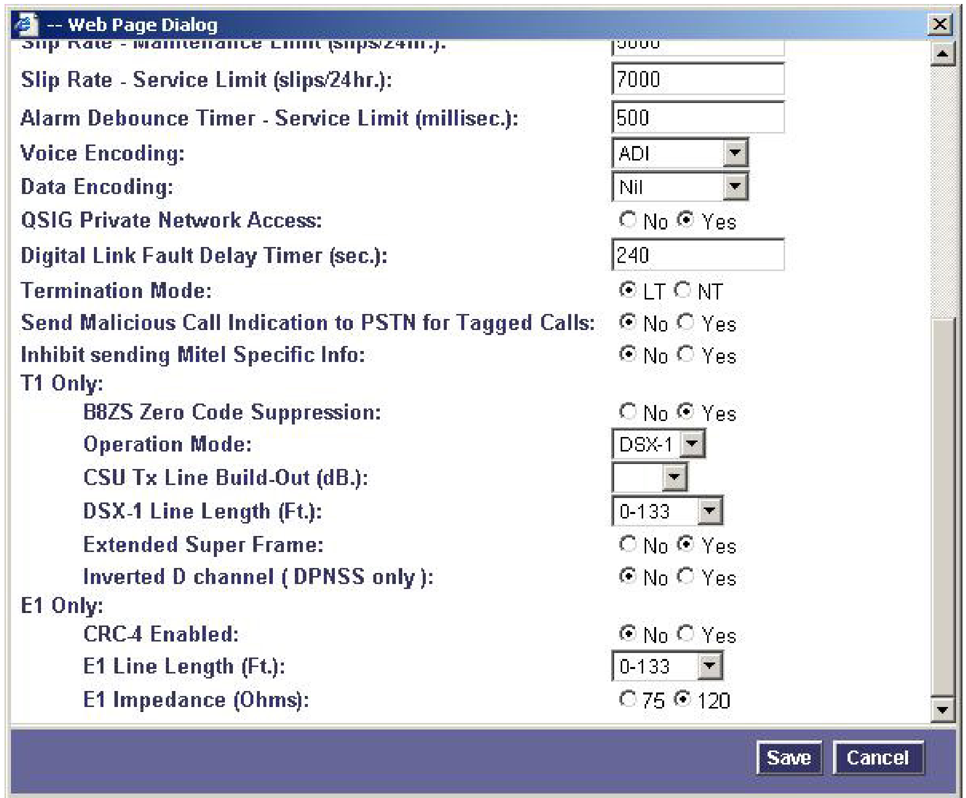

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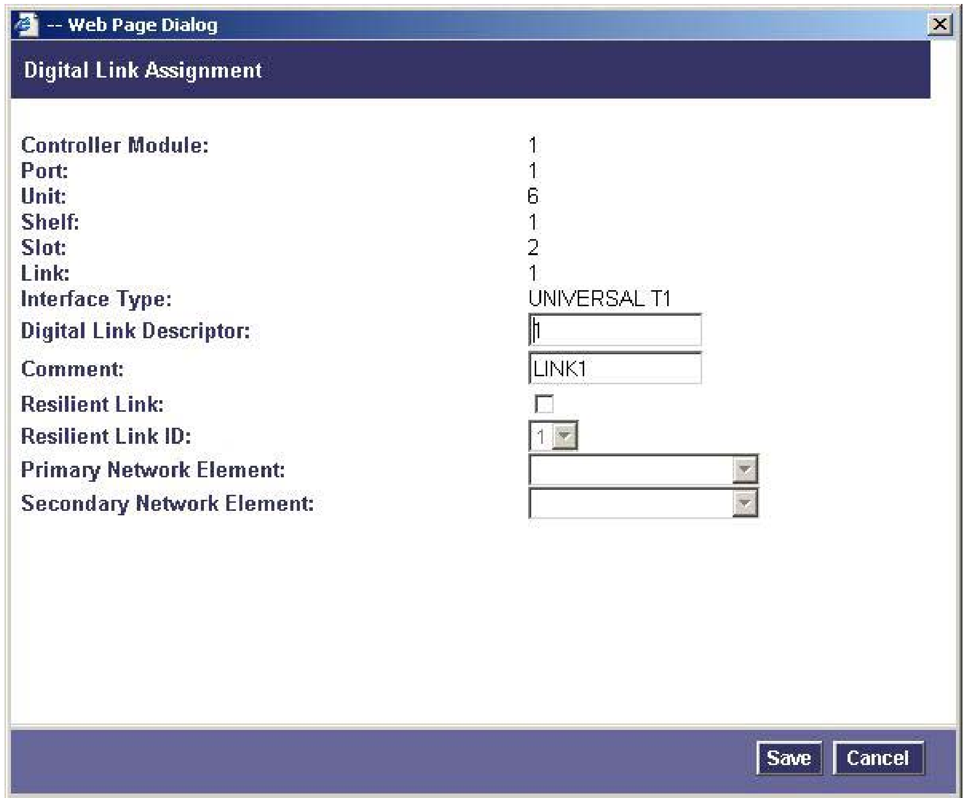
- **T1 Link assignment**



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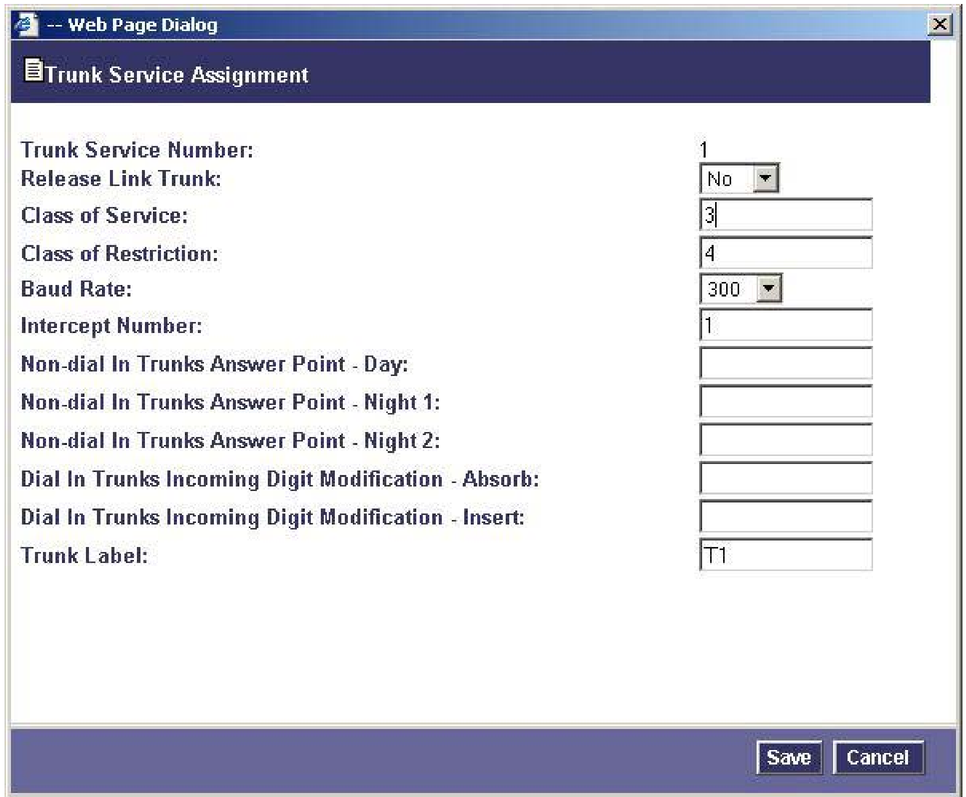
- **Trunk Services Assigment – This screen assigns the COS, COR and speed for service**



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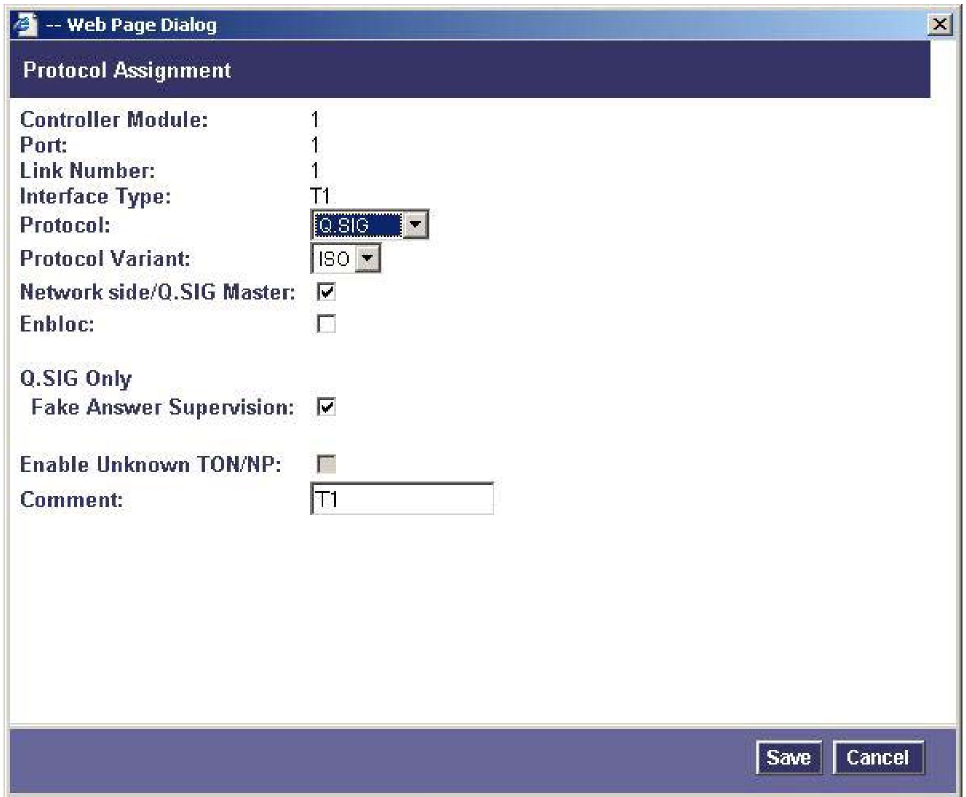
- **Protocol Assigment – Define the type of link (T1 or E1), QSIG protocol and ISO settings.**



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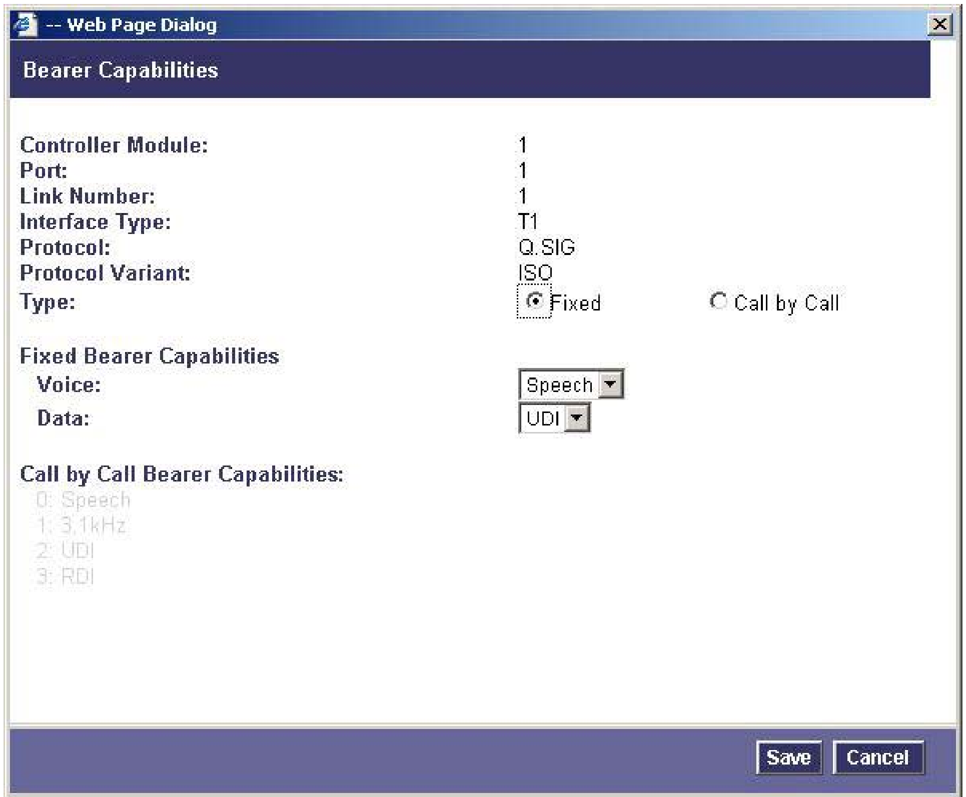
- **Bearer Capabilities – Service(s) the link will be used for.**



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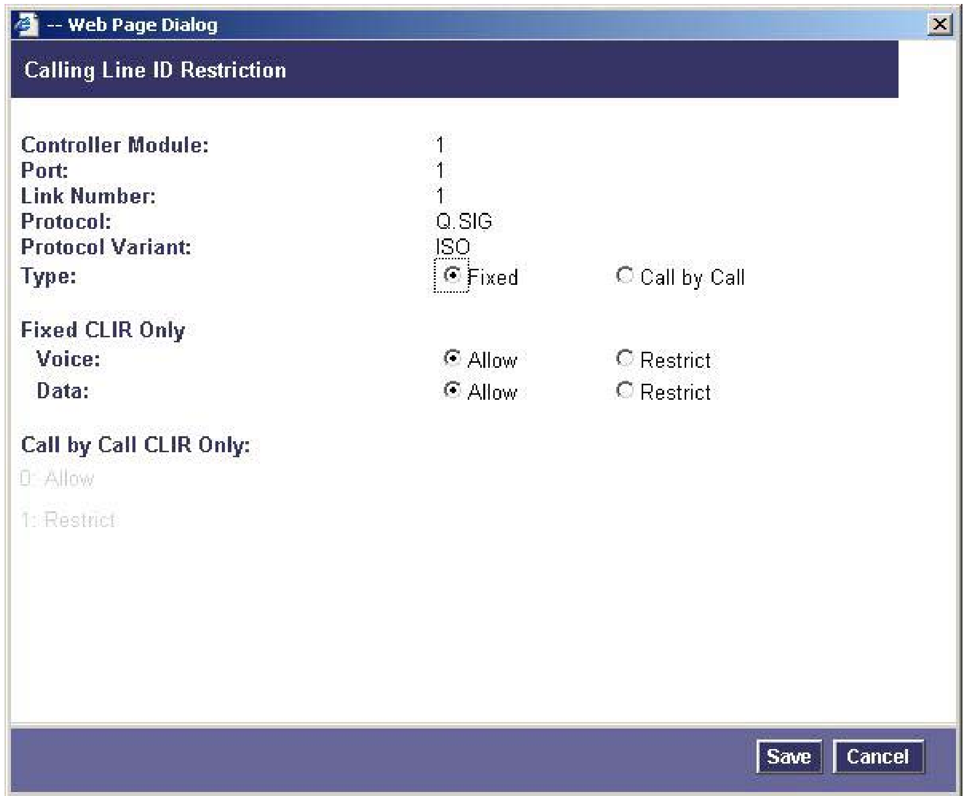
- **Calling Line ID Restriction -**



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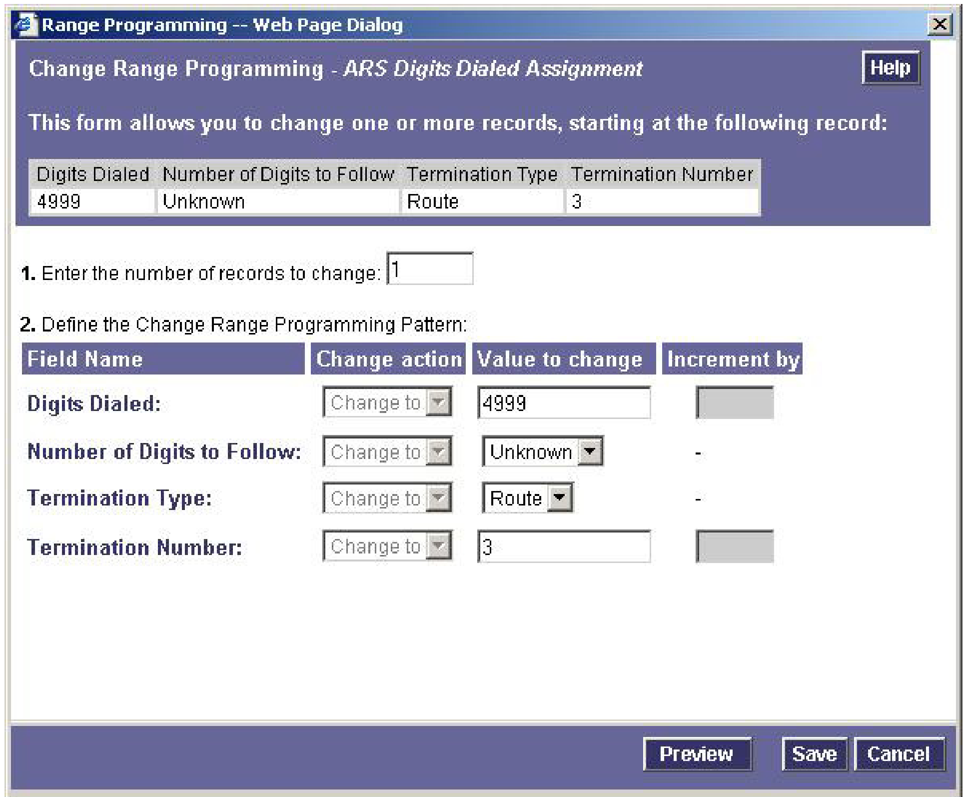
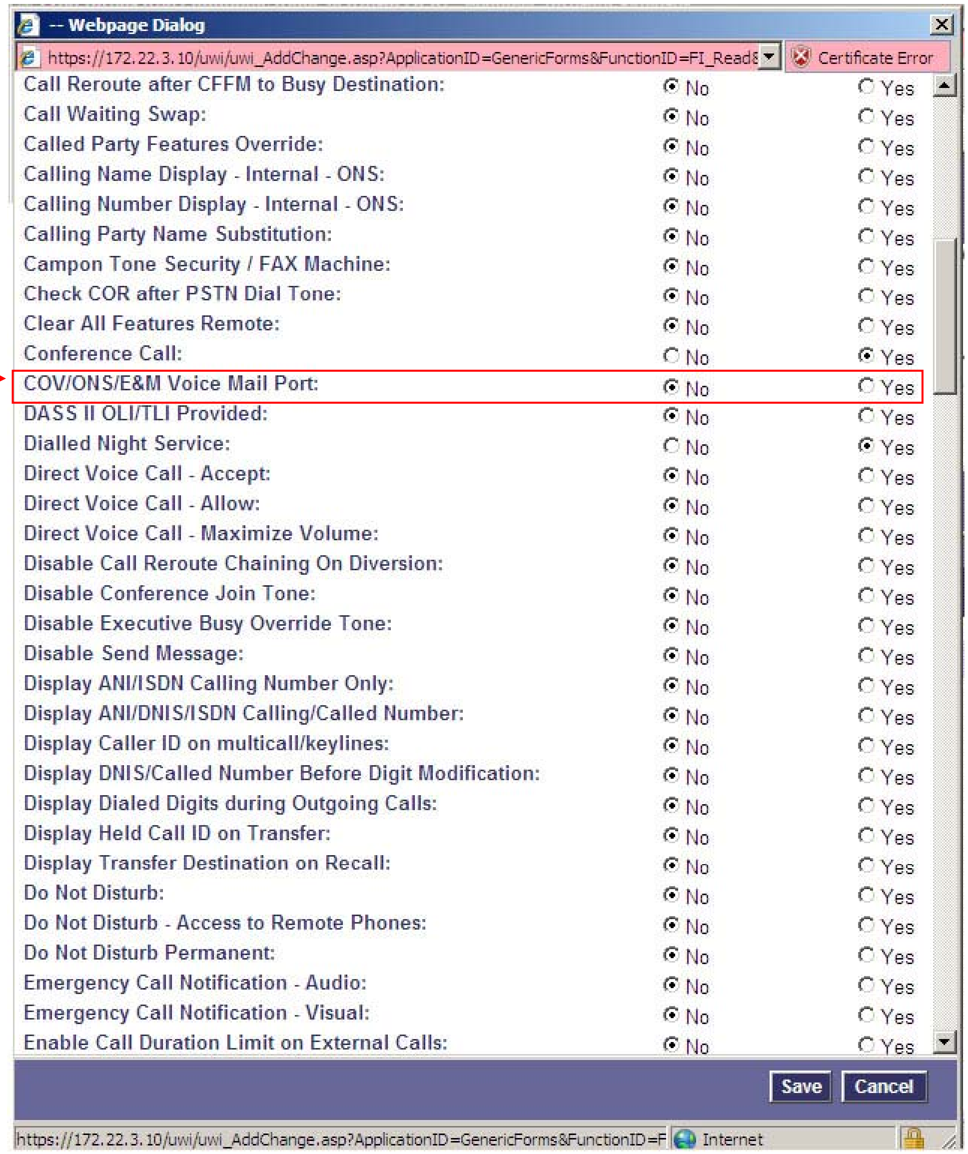
- **Range Programming – ARS programming**



NOTE:
The COS setting on the trunk for:
COV/ONS/E&M Voice Mail Port
should be set to "**No**."

We have found this resolved an issue at one site where the caller heard continuous ringing even though the call was answer and greeting was being played by MM.



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

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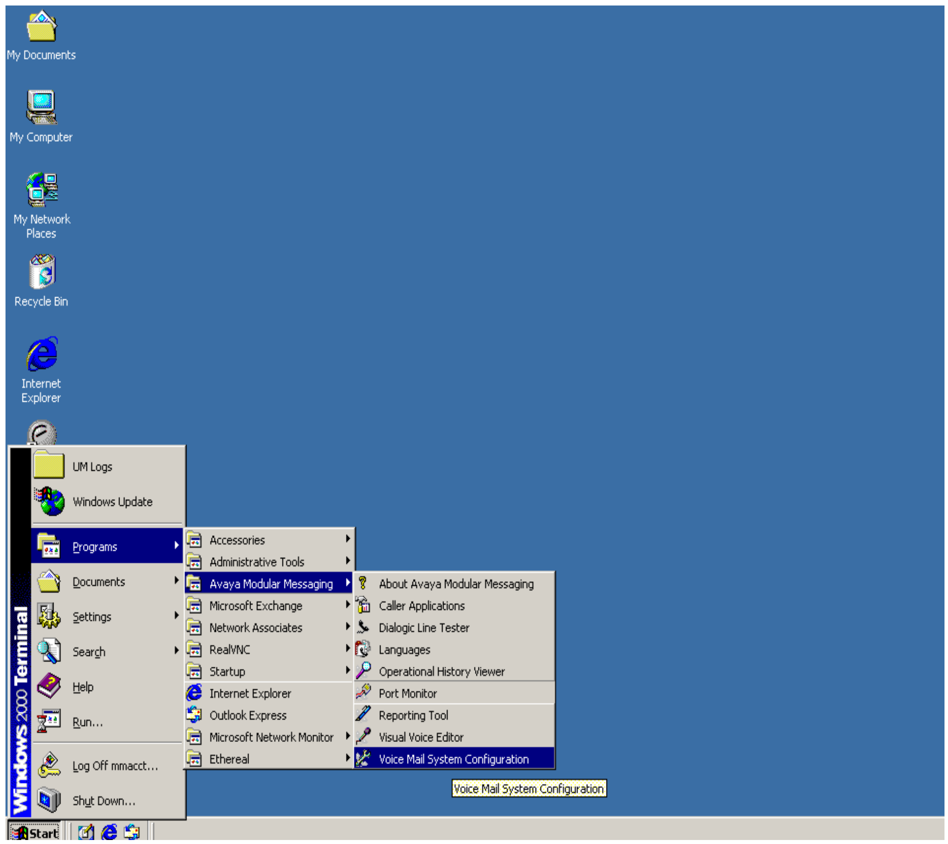
6.0 CONFIGURING THE MESSAGE APPLICATION SERVER

Configuring the MAS

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.



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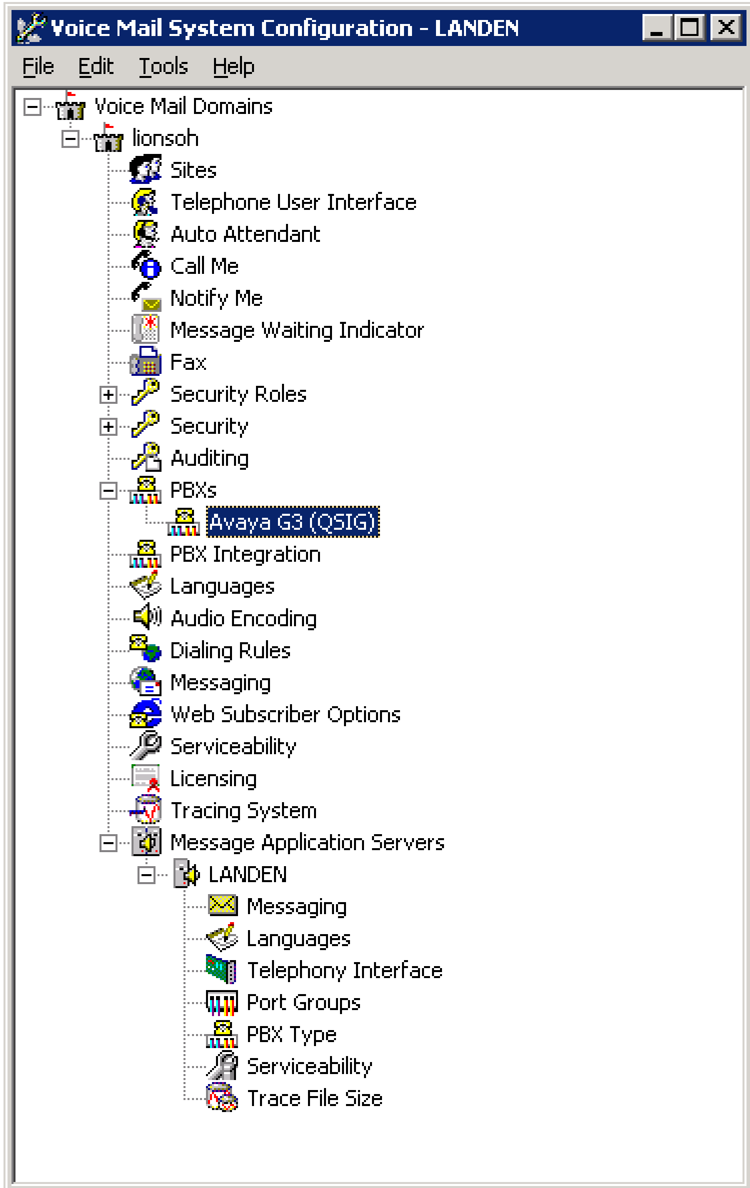
Expand all fields so all-applicable options are visible:



Note: Starting with MM 5.0 additional Fields such as *Sites*, *PBX Integration*, and *Dialing Rules* will appear on the VMSC screen.
If you do not see these you have an earlier MM release.

Ensure the new PBX is added as instructed by the Modular Messaging Installation guide. The new PBX should be:

Avaya G3 (QSIG)

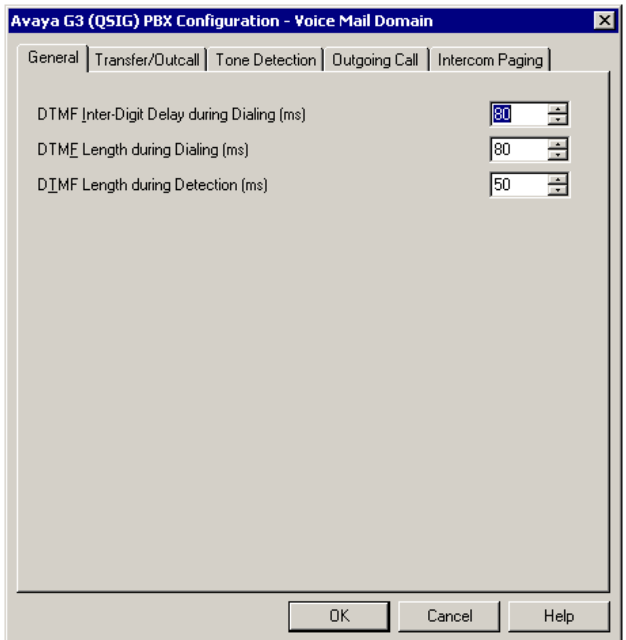
Note: This PBX type, **Avaya G3 (Dialogic QSIG)**, is the only valid choice for this integration at this time. A "**Mitel QSIG**" PBX type will be added at a later date.

Select the Voice Mail Domain

1. Expand **PBXs**
2. Select the newly created **Avaya G3 (Dialogic QSIG)**

Note: This PBX type, **Avaya G3 (Dialogic QSIG)**, is the only valid choice for this integration at this time. A "**Mitel QSIG**" PBX type will be added at a later date.

3. Access the **General (QSIG) PBX Configuration** tab
4. **DTMF Inter-Digit Delay during Dialing (ms)** = 80
5. **DTMF Length during Dialing (ms)** = 80
6. **DTMF Length during Detection (ms)** = 50



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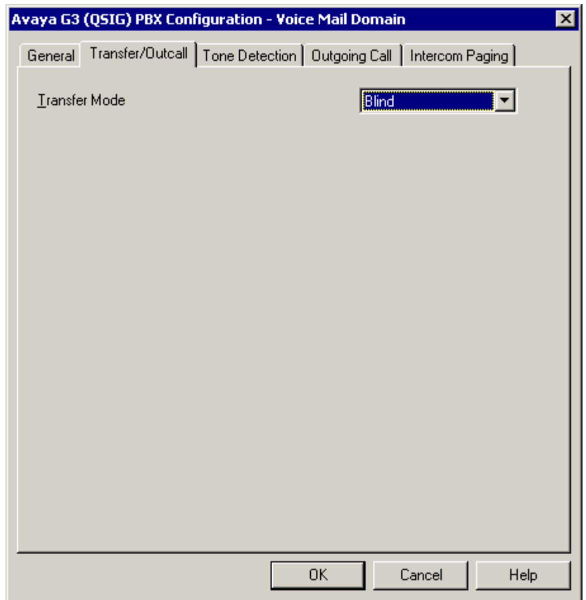
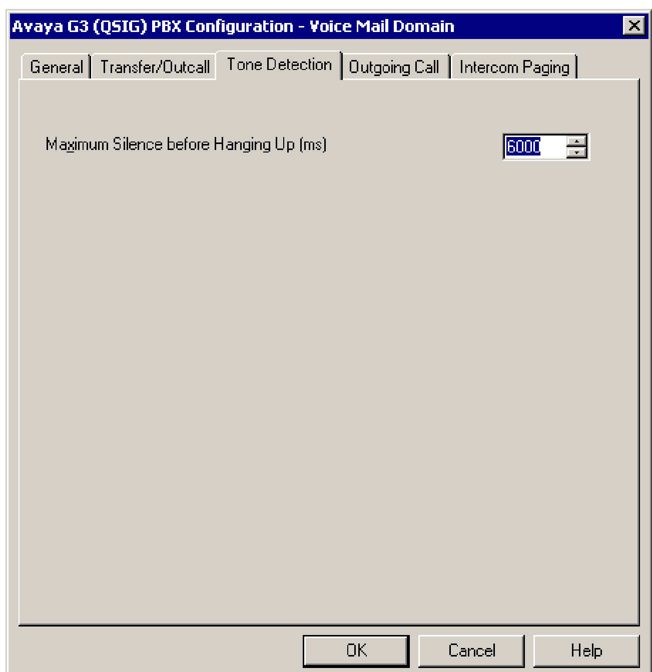
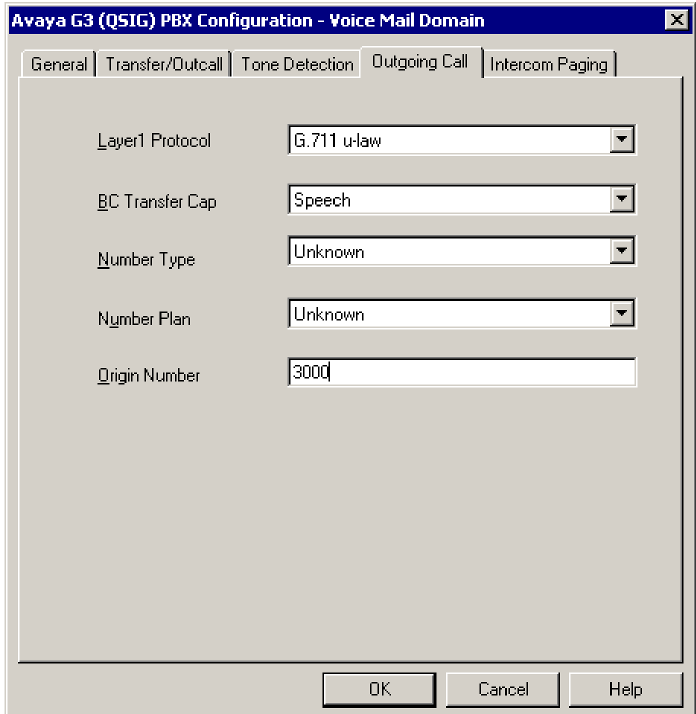
- Next access the **Transfer/Outcall** tab
  Transfer Mode = Blind



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



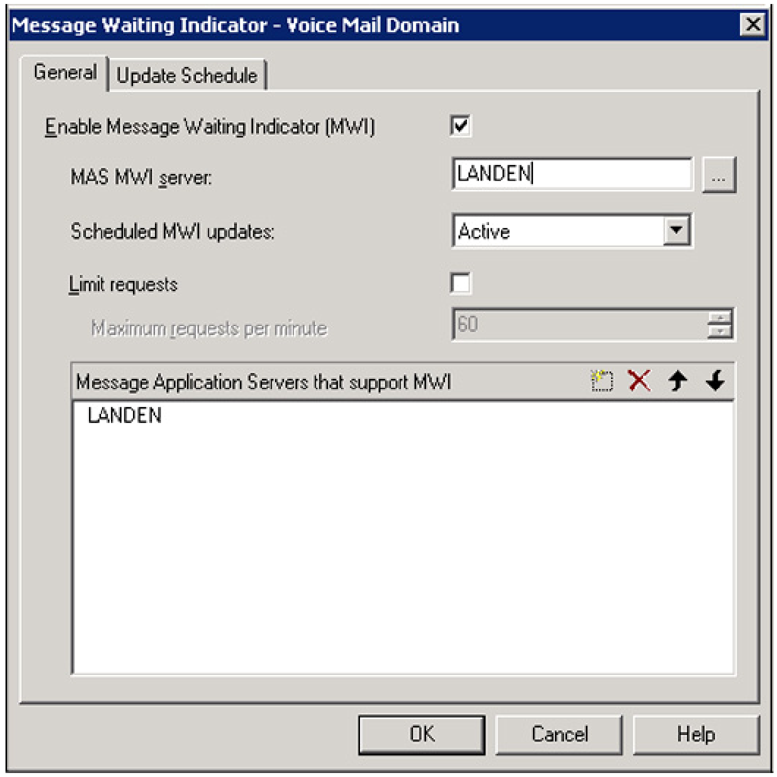
▪ Next access the **Outgoing Call** tab



1. **Layer1 Protocol** = G.711 u-Law
2. **BC Transfer Cap** = Speech
3. **Number Type** = Unknown
4. **Number Plan** = Unknown
5. **Origin Number** = 3000 (The number entered here must be the number entered in the "Voice Mail Number" field of the Hunt Group 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 here must be "3000"
6. Select **OK** to save changes

Note: The Layer1 Protocol field should match the Interface Companding setting selected on Page1 of the DS1 Circuit Pack configuration screen on Page 6 of this configuration note.

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

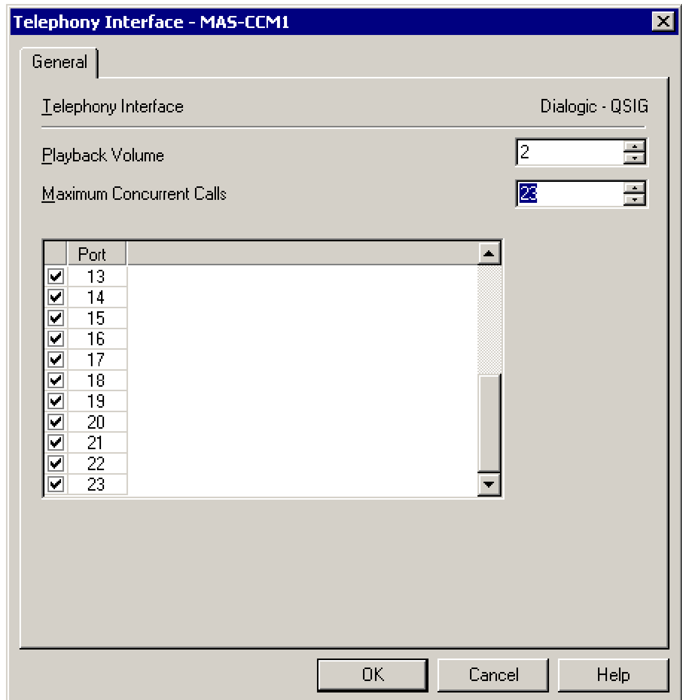


1. **Enable Message Waiting Indicator (MWI)** = Enable by checking the box
2. **MAS MWI Server** = Enter the name of the MWI server created during the installation procedure.
3. **Scheduled MWI updates: Active or Inactive** = Configure as per customer requirements.*
4. **Limit requests** = Leave Unchecked
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

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. 23)
3. **Port** = Ports are enabled by default

Note: The MAS service must be restarted to allow port disabling

4. Select **OK** to save changes

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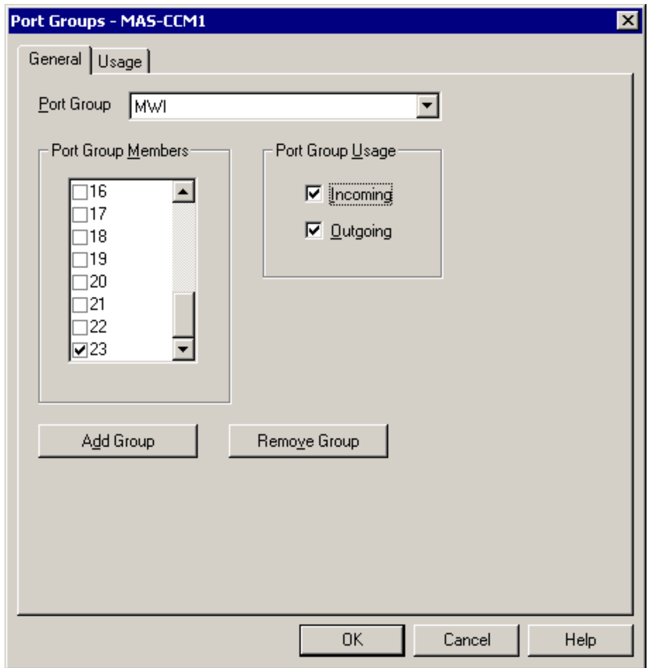
- Next access the **Port Groups** General tab



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. All MWI function is handled by the D-Channel.

1. Click **Add Group** button
2. Name Group **MWI**
3. Within the new **MWI** Port Group <u>uncheck</u> all **Ports** except the MWI port. This will be the upper most port of the Trunk Group. Port 23 on a single board, Port 46 if 2 boards, or Port 69 if 3 boards are used (see side NOTE).
4. 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**.

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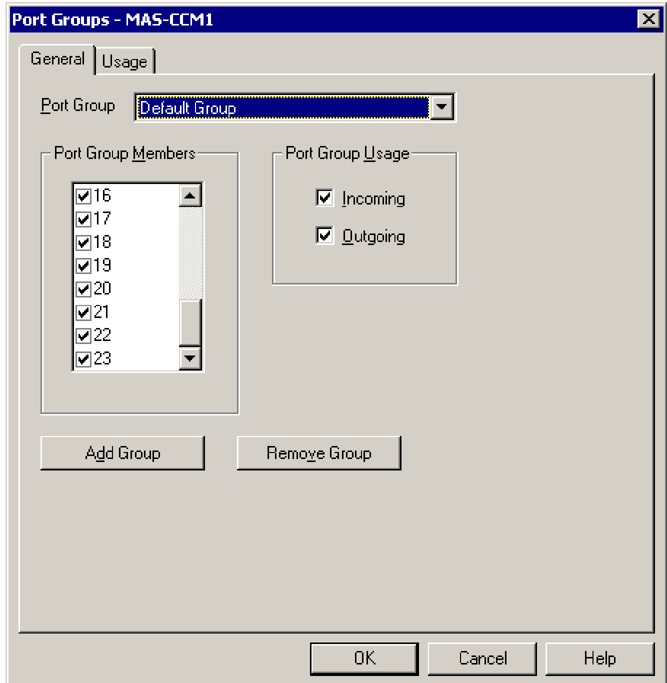

5. Next <u>check</u> all **Ports** (including the MWI port).

6. Select **OK** to save changes

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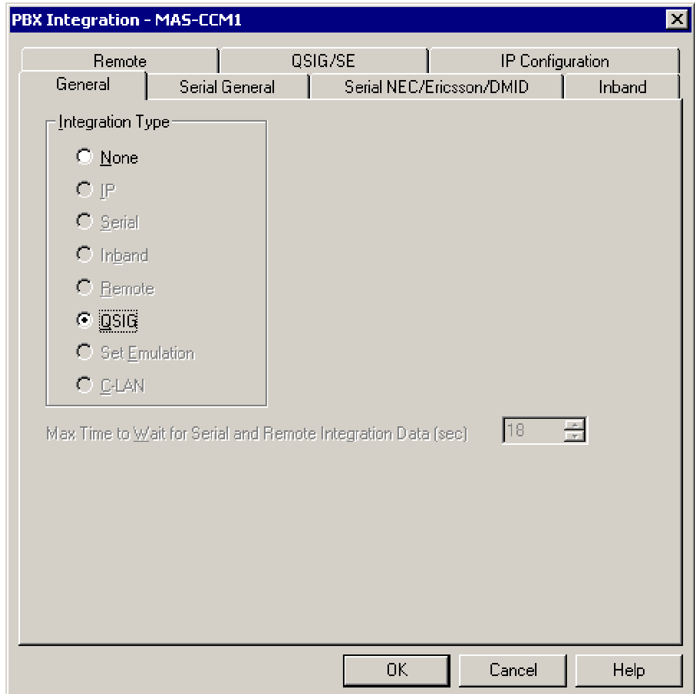
- Next access the **General** tab within the **PBX Integration**

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



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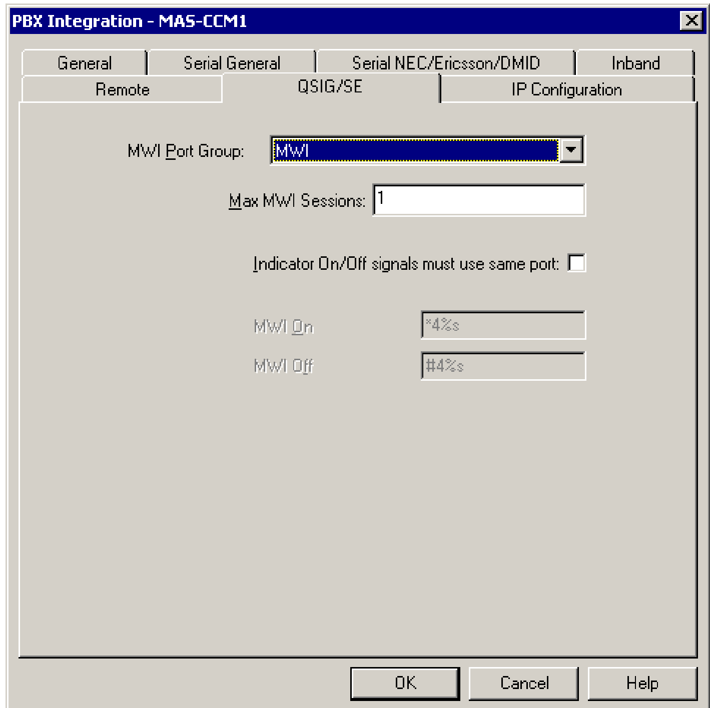
- Access the **QSIG/DSE** tab



1. **Port Group Name** = MWI
2. **Max MWI Sessions** = 1
3. **Indicator On/Off signals must use same port** = Leave Blank
4. **MWI On Field** = Leave as default (can't be changed)
5. **MWI Off Field** = Leave as default (can't be changed)
6. Select **OK** to save changes

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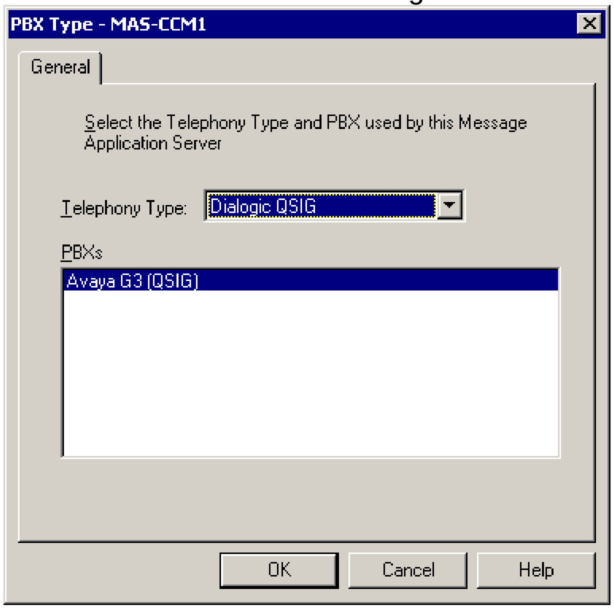
- Next access the **General** tab within the **PBX Type** tab

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.

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Important notes regarding this integration

8.0 CONSIDERATIONS

8.1 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.

8.2 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, as "Path Replacement" does not occur. Additional ports are used for each Find Me call process, and these ports are in use until the call is answered or the caller disconnects from the message server. Additional ports maybe required to support Find Me. 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.

8.3 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. The actual networking configuration is outside the scope of this Config Note.

8.4 Testing for QSIG Path Replacement operation. When your QSIG Modular Messaging systems 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 should see that both channels are now idle in Port Monitor, this shows that the QSIG Path Replacement feature has completed successfully.

Note: Path Replacement is a PBX function. There is no MM programming to support/control Path Replacement.

Certain PBX configurations may cause Path Replacement to fail. In these cases your MM Server may stay bridge onto the transferred call keeping two channels busy in Port Monitor.

8.5 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 T1 board in your MAS:
   /recover /boards 1
   If you have two T1 boards in your MAS:
   /recover /boards 2
   If you have three T1 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.

8.6 We have found that setting the COS on the trunk for the *COV/ONS/E&M Voice Mail Port* to "No" corrects an issue where the caller hears continuous ringing even though the call was answer and greeting was being played by MM. (*See screen shot and note at end of Section 5.0.)*

CHANGE HISTORY		
Revision	Issue Date	Reason for Change
Version A	**02/02/09**	Initial GA Release
Version B	**02/12/09**	Changed screen shot for VMSC and added sidebar in Section 6.0.
Version C	**07/09**	Updated for MM 5.1 and added Note and screen shot end of Section 5.0 about COS setting on trunk along with related Consideration 8.6; Added note regarding collocated PBX/MM on page 1; Updated PBX Hardware & Software sections 3.0 & 3.1
Version D	**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.

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