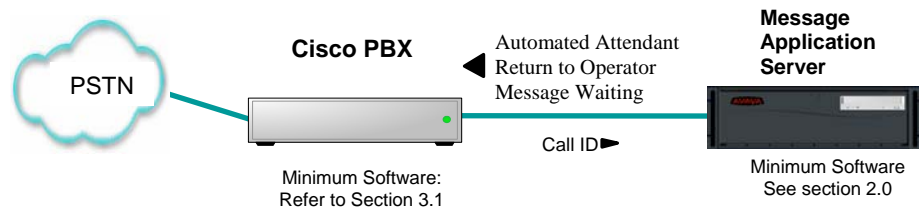


Configuration Note 88061 – Rev. K (07/08)

Cisco Call Manager

T-1 / QSIG



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 Cisco certified or very familiar with the features and functionality of the Cisco PBX supported in this Configuration Note and the QSIG protocol.

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

Please read the entire document before attempting any configuration.

1.0 METHOD OF INTEGRATION

With T1 QSIG integration, one digital pathway between the Cisco Call Manager 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.

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 2.0, 3.x, 4.0

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

MAS Requirements

PBX hardware requirements**3.0 PBX HARDWARE REQUIREMENTS**

- Cisco Catalyst 6500 switch with 6608 T1/E1 blades; or Cisco 2811 Integrated Services Router; or equivalent IOS-based MGCP gateways.
- **Cables:**
 - T1 (DSL) CABLE TYPE CMP (UL) 22AWG or equivalent

PBX software requirements**3.1 PBX SOFTWARE REQUIREMENTS**

- Supported Software:
Cisco CallManager 4.1(2)

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	[✓]

NOTICE:

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

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

**Gateway
Configuration
screen**

**IP & MAC Address
should be set
here. Leave
remaining items
as default.**

5.0 CONFIGURING THE CALL MANAGER FOR INTEGRATION

The following programming is intended for Cisco certified technicians/engineers. The screens shown in this section are taken from Cisco 6608 T1/E1 Gateway Configuration administration screens.

Parameters may vary with software releases.

Ensure all required software features are enabled on the Cisco Call Manager. 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: Change fields as shown.

Gateway Configuration

[Back to Find/List Gateway](#)
[Dependency Record](#)

Product : Cisco Catalyst 6000 T1 VoIP Gateway
Gateway : S0/DS1-0@SDA0001C9D93A9C
Device Protocol: Digital Access PRI
Registration: Registered with Cisco CallManager 172.20.236.2
IP Address: **172.20.236.15**

Status: Ready

Update

Delete

Reset Gateway

Device Information

MAC Address*	0001C9D93A9C
Description	Cat 6500 port 5/5
Device Pool*	Default
Call Classification*	Use System Default
Network Locale	United States
Media Resource Group List	< None >
Location	< None >
AAR Group	< None >
Load Information	

Multilevel Precedence and Preemption (MLPP) Information

The screen SHOT shown below was taken from a Cisco 2218 T1/E1 Gateway Configuration. It is provided purely as an example.

PARAMETERS MAY VARY WITH SOFTWARE RELEASES..

Gateway Configuration

[Back to MGCP Configuration](#)
[Back to Find/List Gateways](#)
[Dependency Records](#)

Assigned to Route Group:RG_USPL-Avaya_VoiceMail

Product : Cisco 2811
Gateway : S0/SU0/DS1-0@Rpl.com
Device Protocol: Digital Access PRI
Registration: Registered with Cisco CallManager 17.20.76.1
IP Address: 161.107.0.1

Status: Ready

UpdateDeleteReset Gateway

Device Information

End-Point Name*	S0/SU0/DS1-0@ Rpl.com
Description	S0/SU0/DS1-0@ Rpl.com
Device Pool*	DP_USPL-CCMS1
Call Classification*	Use System Default
Network Locale	< None >
Signal Packet Capture Mode	None
Packet Capture Duration	60
Media Resource Group List	< None >
Location	< None >
AAR Group	< None >
Load Information	
V150 (subset)	<input type="checkbox"/>

Set PRI Protocol
Type to: **PRI
QSIG T1**.

Set remaining
items as indicated.

Enter a valid Calling Search
Space that contains the
partition assigned to the voice
mail users' telephones.

Note: If this is left set to none then
MWI, transfers, MM Find
Me, etc. will not work. See
Consideration 8.3

Set items as
indicated.

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain (e.g., "0000FF")
MLPP Indication
MLPP Preemption

Interface Information

PRI Protocol Type*
Protocol Side*
Channel Selection Order*
Channel IE Type*
PCM Type*
Delay for first restart (1/8 sec ticks)
Delay between restarts (1/8 sec ticks)
☒ Inhibit restarts at PRI initialization
☐ Enable status poll

Call Routing Information

Inbound Calls

Significant Digits*
Calling Search Space
AAR Calling Search Space
Prefix DN

Outbound Calls

Calling Line ID Presentation*
Calling Party Selection*
Called party IE number type unknown*
Calling party IE number type unknown*
Called Numbering Plan*
Calling Numbering Plan*
Number of digits to strip*
Caller ID DN
SMDI Base Port*

PRI Protocol Type Specific Information

- ☐ Display IE Delivery
- ☐ Redirecting Number IE Delivery - Outbound
- ☐ Redirecting Number IE Delivery - Inbound
- ☒ Send Extra Leading Character In DisplayIE***
- ☐ Setup non-ISDN Progress Indicator IE Enable****
- ☐ MCDN Channel Number Extension Bit Set to Zero**
- ☐ Send Calling Name In Facility IE
- ☐ Interface Identifier Present**

Interface Identifier Value** Connected Line ID Presentation (QSIG Inbound Call)* **UUIE Configuration**

- ☐ Passing Precedence Level Through UUIE

Security Access Level

Ensure the Clock Reference is set to **Network**.

Set remaining items as shown.

Product Specific ConfigurationClock Reference* TX-Level CSU* FDL Channel* Framing* Audio Signal Adjustment into IP Network* Audio Signal Adjustment from IP Network* Yellow Alarm* Zero Suppression* Digit On Duration(50-500ms)* Interdigit Duration(50-500msec)* SNMP Community String Disable SNMP Set operations* ☐Debug Port Enable* ☒Hold Tone Silence Duration* Port Used for Voice Calls* ☒Port Used for Modem Calls* ☒Port Used for Fax Calls* ☒

Fax and Modem Parameters	
Fax Relay Enable*	<input checked="" type="checkbox"/>
Fax Error Correction Mode Override*	<input checked="" type="checkbox"/>
Maximum Fax Rate*	14400bps
Fax Payload Size*	20
Non Standard Facilities Country Code*	65535
Non Standard Facilities Vendor Code*	65535
Fax/Modem Packet Redundancy*	<input type="checkbox"/>
NSE Type*	Non-IOS Gateways

Playout Delay Parameters	
Initial Playout Delay*	40
Minimum Playout Delay*	20
Maximum Playout Delay*	150

Echo Cancellation Configuration	
Echo TailLength (ms)*	32 ms
Minimum ERL (db)*	6 db

* indicates required item
 ** applicable to DMS-100 protocol only
 *** applicable to DMS-100 protocol and DMS-250 protocol only

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration

For Cisco IP Telephony Solutions



Route Pattern Configuration

[Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern: 5566

Status: Insert completed

Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

[Copy](#) [Update](#) [Delete](#)

Pattern Definition

Route Pattern*	5566
Partition	< None >
Description	Avaya MM voicemail
Numbering Plan*	North American Numbering Plan
Route Filter	< None >
MLPP Precedence	Default
Gateway or Route List*	S0/DS1-0@SDA0001C9D93A9C (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern — Not Selected —
Call Classification*	OnNet
<input type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Device Override
<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code	

The Route Pattern Configuration is where you will enter the voice mail pilot number.

It is entered in the field adjacent the **Route Pattern***

<input type="checkbox"/>	Require Forced Authorization Code	
	Authorization Level	<input type="text" value="0"/>
<input type="checkbox"/>	Require Client Matter Code	
Calling Party Transformations		
<input type="checkbox"/>	Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
Calling Line ID Presentation	<input type="text" value="Allowed"/>	
Calling Name Presentation	<input type="text" value="Allowed"/>	
Connected Party Transformations		
Connected Line ID Presentation	<input type="text" value="Allowed"/>	
Connected Name Presentation	<input type="text" value="Allowed"/>	
Called Party Transformations		
Discard Digits	<input type="text" value=" < None >"/>	
Called Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
ISDN Network-Specific Facilities Information Element		
Carrier Identification Code	<input type="text"/>	
Network Service Protocol	<input type="text" value=" — Not Selected —"/>	
Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value=" — Not Selected —"/>	<input type="text" value=" < Not Exist >"/>	<input type="text"/>

* indicates required item.

Note: Make sure you configure a Route Pattern containing the newly configured gateway in the "Gateway or Route List" field. The digit string used in the "Route Pattern" field will be the voicemail pilot number.

**Cisco Call Manager
QSIG-related Service
parameters configuration**

Clusterwide Parameters (Feature - Forward)

Parameter Name	Parameter Value	Suggested Value
Forward Maximum Hop Count*	<input type="text" value="12"/>	12
Forward No Answer Timer (sec)*	<input type="text" value="12"/>	12
Max Forward Hops to DN*	<input type="text" value="12"/>	12
Retain Forward Information*	<input type="text" value="False"/>	False
Forward By Reroute Enabled*	<input type="text" value="True"/>	False
Forward By Reroute T1 Timer (sec)*	<input type="text" value="10"/>	10
Include Original Called Info for Q.SIG Call Diversions*	<input type="text" value="Always"/>	Only after the first diversion

**Cisco Call Manager Path
Replacement parameters
configuration.**

Note: Path Replacement requires Cisco Call Manager 4.1(2) software or newer.

Clusterwide Parameters (Feature - Path Replacement)

Parameter Name	Parameter Value	Suggested Value
Path Replacement Enabled*	<input type="text" value="True"/>	False
Path Replacement on Tromboned Calls*	<input type="text" value="True"/>	True
Start Path Replacement Minimum Delay Time (sec)*	<input type="text" value="0"/>	0
Start Path Replacement Maximum Delay Time (sec)*	<input type="text" value="0"/>	0
Path Replacement T1 Timer (sec)*	<input type="text" value="30"/>	30
Path Replacement T2 Timer (sec)*	<input type="text" value="15"/>	15
Path Replacement PINX ID	<input type="text" value="5555"/>	

Configuring the MAS

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. Expand all fields so all-applicable options are visible.

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

1. Select the **Voice Mail Domain**
 2. Expand **PBXs**
 3. Select the newly created **CISCO CM (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 mu-Law
 2. **BC Transfer Cap** = Speech
 3. **Number Type** = Unknown
 4. **Number Plan** = Unknown
 5. **Origin Number** = *
 6. Select **OK** to save changes
- Note:** Confirm these ISDN values with your PBX vendor. If these values do not match the ISDN protocol, then transfer and out calling will not work.
- * Enter the MAS server pilot number in the Origin Number field
- 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.

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 functionality is handled by the D-Channel.

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 will not appear.

3. **Scheduled MWI updates: Active or Inactive** = Configure as per customer requirements.*
 4. **Maximum Requests per Minute** = 200
 5. **Message Application Servers that Support MWI** = This box should contain a list of MAS servers capable of placing MWI requests.
 6. Select **OK** to save changes
- ***Note:** The Scheduled MWI updates parameter is only available on MM 3.x
- Next access the **Ports Group** tab
 1. Click **Add Group** button
 2. Name new 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 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**.
 5. Next check all **Ports**.
 6. Select **OK** to save changes
 - Next access the **QSIG General** tab within the **PBX Type** tab
 1. Telephony Type = **Dialogic QSIG**
 2. Under PBXs ensure **CISCO CM (QSIG)** is selected
 3. Select **OK** to save changes
 - 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
 - Next access the **General** tab within the **PBX Integration**
 1. **QSIG** = Enable by checking the box
 2. Access the **QSIG/DSE** tab
 3. Select **Port Group Name** = MWI (or name used in Port Groups)
 4. **Max MWI Sessions** = 1
 5. **Indicator On/Off signals must use same port** = Leave Blank
 6. **MWI On Field** = Leave as default
 7. **MWI Off Field** = Leave as default

8. 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

Important notes regarding
this integration

- 8.1 **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.](#)
- 8.2 **Transfers to ringing may use additional ports.** When Find Me is used additional ports are in use until the call is answered or the caller disconnects from the message server. Additional ports may be 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 **For MWI to properly function** all end user extensions (known as Directory Numbers [DNs] in CallManager) must be assigned to *partitions* within the Cisco CallManager that are accessible by the *Calling Search Space* assigned to the *Message Waiting On* and *Message Waiting Off numbers*.

Note: *Message Waiting On* and *Message Waiting Off numbers* are extensions configured on CallManager, under Feature>Voice Mail>Message Waiting, and are basically equivalent to MWI ON and OFF feature access codes.
- 8.4 **Cisco CallManager does not present iDivert-ed calls over QSIG** trunks as forwarded calls. These calls are presented as direct calls from the calling station instead of being presented as forwarded calls from the diverting station. Cisco is aware of this issue. Contact Cisco for further information.

- continued next page -

CHANGE HISTORY

Revision	Issue Date	Reason for Change
Version A	06/23/05	GA release
Version B	04/11/06	Added: <ul style="list-style-type: none"> • MM 3.0 to support release section 2.0 • New Scheduled MWI updates parameter updates noted for MM3.0
Version C	11/06/06	Changed PBX Type in Section 6.0 to Cisco CM (QSIG)
Version D	6/1/07	Updated Consideration 8.1
Version E	6/2/07	Updated diagram page 1
Version F	12/7/07	Added 2811 router to requirements, new note on Calling Search Space in Section 5, and example screen shot for 2811 router.
Version G	03/08	Updated Consideration 8.1
Version H	05/05/08	Updated to support MM 4.0
Version I	05/16/08	Removed Dialogic Driver release from Section 2.0
Version J	07/02/08	Removed check in section 4.0 for N+1 and Consideration 8.5 regarding N+1. Cisco CM is unable to create a trunk group so that it can distribute channels to multiple MMs.
Version K	07/03/08	Made screen shot change page 4

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