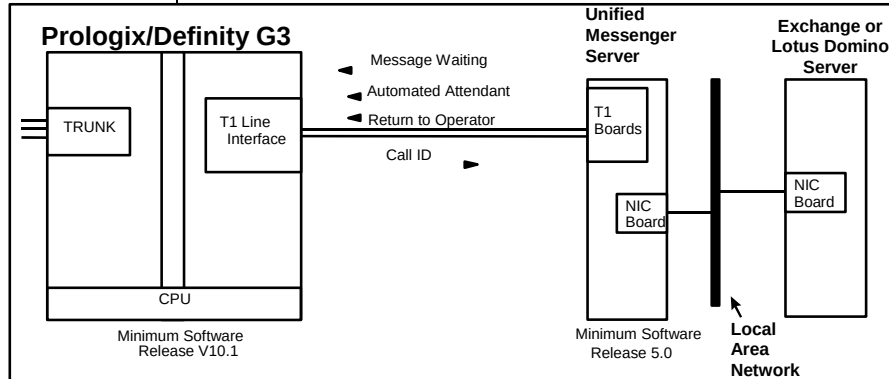


Configuration Note 8204 – Version H (10/03)

AVAYA Definity G3i/r - QSIG



With QSIG, one digital pathway between the PBX and the Unified Messenger server transmits both call information and voice communications

1.0 METHOD OF INTEGRATION

With QSIG integration, one digital pathway between the Definity PBX and Unified Messenger™ server transmits both call information and voice communications. The pathway is provided by an ISDN digital link, which provides 24 channels that connect to the Dialogic D/240JCT-T1 card. The D/240JCT-T1 card connects directly to the PBX using a QSIG link, which makes the Unified Messenger server appear as another PBX in the network. Within the D-Channel, routing information is sent so that the destination PBX has information regarding the source of the call and the reason for its arrival. The Unified Messenger server receives 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 UNIFIED MESSENGER SERVER REQUIREMENTS

- **Dialogic D/240JCT-T1**
- **CT Bus cable** (only required for multiple card installation)
- **Software Release 5.0** or higher

Unified Messenger Server requirements

PBX hardware requirements

3.0 PBX HARDWARE REQUIREMENTS

- TN464 Circuit Pack per each Dialogic T1 Interface Card
- **Cables:**
 - RJ45 to RJ48C on the Dialogic (cable depends on PBX connection)

PBX software requirements

3.1 PBX SOFTWARE REQUIREMENTS

- Minimum Supported Software: V10.1

Important: Before ordering account teams should check with Avaya Services to determine if there are any applicable patches for customer specific configuration, regardless of the PBX version. Especially if DCS+ or QSIG Networking will be 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)

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	[✓]
Busy/No Answer	[]

Auto Attendant	[✓]
Direct Call	[✓]
External Call ID (ANI)	[✓]
Internal Call ID	[✓]
Message Waiting	[✓]
Multiple Call Forward	[✓]
Multiple Greetings	[✓]
Outcalling	[✓]
Return to Operator	[✓]

5.0 CONFIGURING THE DEFINITY G3 TO INTEGRATE

The following PBX programming is intended for certified PBX technicians/engineers. The screens shown in this section are taken from an Avaya Definity G3 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 changed the recommended fields.

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.

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

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

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

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

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

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

```

display system-parameters customer-options
                                OPTIONAL FEATURES

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

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

```

```

display system-parameters customer-options
                                QSIG OPTIONAL FEATURES

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

```

☐ 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
                                FEATURE-RELATED SYSTEM PARAMETERS

                                ISDN PARAMETERS

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

                                QSIG TSC Extension: 2998
                                MWI - Number of Digits Per Voice Mail Subscriber: 4 (*see note below)
                                Feature Plus Ext:
                                National CPN Prefix:
                                International CPN Prefix:
                                Pass Prefixed CPN to ASAI? n
                                Unknown Numbers Considered Internal for AUDIX? n
                                USNI Calling Name for Outgoing Calls? y
                                Path Replacement with Measurements? y
                                QSIG Path Replacement Extension:
                                Path Replace While in Queue/Vectoring? n

```

Note: This parameter must match the number of digits used for mailbox/extension length.

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

Maintain SBA At Principal? **n**
Coverage Of Calls Redirected Off-Net Enabled? **n**

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

CALL COVERAGE/FORWARDING PARAMETERS

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

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

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

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

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

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

☐ Change features and assign your private network access code, in this example we assigned 107

```
change feature-access-codes                            Page 1 of 6
FEATURE ACCESS CODE (FAC)
    Abbreviated Dialing List1 Access Code: 101
    Abbreviated Dialing List2 Access Code: 102
    Abbreviated Dialing List3 Access Code: 103
    Abbreviated Dial - Prgm Group List Access Code:
    Announcement Access Code: 104
    Answer Back Access Code:
    Auto Alternate Routing (AAR) Access Code: 107
    Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
    Automatic Callback Activation:                      Deactivation:
    Call Forwarding Activation Busy/DA: 190 All: *9     Deactivation: #9
    Call Park Access Code: *6
    Call Pickup Access Code:
    CAS Remote Hold/Answer Hold-Unhold Access Code:
    CDR Account Code Access Code: 188
    Change COR Access Code:
    Change Coverage Access Code:

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

- ☐ Install the T1 circuit pack, making sure they are “strapped” for 24-channel operation (T-1). Configure the DS1 circuits as follows:

```

display ds1 03a11                               Page 1 of 2
                                DS1 CIRCUIT PACK

                                Location: 03A11           Name: QSIG
                                Bit Rate: 1.544           Line Coding: b8zs
                                Line Compensation: 1       Framing Mode: esf
                                Signaling Mode: isdn-pri
                                Connect: pbx              Interface: peer-master
                                CentreVu Long Timers? n    Peer Protocol: Q-SIG
                                Interworking Message: PROgress Side: a
                                Interface Companding: mulaw CRC? n
                                Idle Code: 11111111
                                DCP/Analog Bearer Capability: 3.1kHz

                                Slip Detection? n          Near-end CSU Type: other

```

```

display ds1 03a11                               Page 2 of 2
                                DS1 CIRCUIT PACK

                                ESF DATA LINK OPTIONS

                                Network Management Protocol: tabs
                                Send ANSI-T1.403 One-Second Performance Reports? n
                                Far-end CSU Address: b

                                INTEGRATED CSU OPTIONS

                                Transmit LBO: 0db
                                Receive ALBO: 26db
                                Upon DTE LOS: loopback

                                CPE LOOPBACK JACK OPTIONS
                                Supply CPE Loopback Jack Power? n

```

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

```

display signaling-group 1                       Page 1 of 5
                                SIGNALING GROUP

                                Group Number: 9           Group Type: isdn-pri
                                Associated Signaling? y    Max number of NCA TSC: 10
                                Primary D-Channel: 03a1124 Max number of CA TSC: 0
                                Trunk Group for Channel Selection: 1 Trunk Group for NCA TSC: 2
                                Supplementary Service Protocol: b X-Mobility/Wireless Type: NONE

```

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.

The MWI trunk group for NCA TSC must also be set.

☐ Create a trunk group, and assign the first 22 newly created DS1 channels to it. If the Unified Messenger 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:

```

display trunk-group 1                                     Page 1 of 10
                                     TRUNK GROUP

Group Number: 1                      Group Type: isdn      CDR Reports: y
  Group Name: UM QSIG                COR: 1    TN: 1      TAC: 15
  Direction: two-way                Outgoing Display? n  Carrier Medium: PRI/BRI
  Dial Access? y                    Busy Threshold: 99    Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n        TestCall ITC: rest
                                     Far End Test Line No:

TestCall BCC: 4
TRUNK PARAMETERS
  Codeset to Send Display: 6          Codeset to Send National IEs: 6
  Max Message Size to Send: 260      Charge Advice: none
Supplementary Service Protocol: b    Digit Handling (in/out): enbloc/enbloc

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

```

```

display trunk-group 1                                     Page 2 of 10
TRUNK FEATURES
  ACA Assignment? n                  Measured: none      Wideband Support? n
                                     Internal Alert? n    Maintenance Tests? y
                                     Data Restriction? n  NCA-TSC Trunk Member:
                                     Send Name: y          Send Calling Number: y
  Used for DCS? n                    Hop Dgt? n
Suppress # Outpulsing? n            Numbering Format: unk-pvt
Outgoing Channel ID Encoding: exclusive  UII IE Treatnt: service-provider

                                     Replace Restricted Numbers? n
                                     Replace Unavailable Numbers? n
                                     Send Connected Number: y

  Send UCID? n
Send Codeset 6/7 LAI IE? y          Ds1 Echo Cancellation? n

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

```

NOTE: There were no entries on Pages 3, 4, 5, for INCOMING CALL HANDLING TREATMENT

```

display trunk-group 1                                     Page 6 of 22
                                     TRUNK GROUP
                                     Administered Members (min/max): 1/23
GROUP MEMBER ASSIGNMENTS                                     Total Administered Members: 23

      Port      Code Sfx Name      Night      Sig Grp
1: 03A1101  TN464  F      Name      Night      Sig Grp
2: 03A1102  TN464  F
3: 03A1103  TN464  F
4: 03A1104  TN464  F
5: 03A1105  TN464  F
6: 03A1106  TN464  F
7: 03A1107  TN464  F
8: 03A1108  TN464  F
9: 03A1109  TN464  F
10: 03A1110 TN464  F
11: 03A1111 TN464  F
12: 03A1112 TN464  F
13: 03A1113 TN464  F
14: 03A1114 TN464  F
15: 03A1115 TN464  F

```

```

display trunk-group 1                                     Page 7 of 22
                                     TRUNK GROUP
                                     Administered Members (min/max): 1/22
GROUP MEMBER ASSIGNMENTS                                     Total Administered Members: 23

      Port      Code Sfx Name      Night      Sig Grp
16: 03A1116 TN464  F
17: 03A1117 TN464  F
18: 03A1118 TN464  F
19: 03A1119 TN464  F
20: 03A1120 TN464  F
21: 03A1121 TN464  F
22: 03A1122 TN464  F
23:
24:
25:
26:
27:
28:
29:
30:

```

Note: When running MWI on your Voice Server you should setup a separate trunk group containing 1 channel from one of your QSIG trunks; this channel should not be apart of any other trunk group and should be the highest port number on the board e.g. channel 23 for 1 board or channel 46 for two boards or channel 69 for 3 boards. You should then create a new MWI port group containing this channel and make the relevant changes in UM VMSC. Below is an example of the MWI trunk group:

```

display trunk-group 2                                     Page 1 of 22
                                     TRUNK GROUP

Group Number: 2                                           Group Type: isdn           CDR Reports: y
  Group Name: UM QSIG MWI                                COR: 1          TN: 1         TAC: 22
  Direction: two-way                                     Outgoing Display? y       Carrier Medium: PRI/BRI
  Dial Access? y                                         Busy Threshold: 99        Night Service:
Queue Length: 0
Service Type: tie                                         Auth Code? n             TestCall ITC: rest
                                     Far End Test Line No:
TestCall BCC: 4
TRUNK PARAMETERS
  Codeset to Send Display: 6                             Codeset to Send National IEs: 6
  Max Message Size to Send: 260                         Charge Advice: none
  Supplementary Service Protocol: b                     Digit Handling (in/out): enbloc/enbloc

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

```

```

display trunk-group 1                                     Page 2 of 22
TRUNK FEATURES
  ACA Assignment? n                                     Measured: none           Wideband Support? n
                                                         Internal Alert? n         Maintenance Tests? y
                                                         Data Restriction? n       NCA-TSC Trunk Member: 1
                                                         Send Name: y              Send Calling Number: y
  Used for DCS? n                                         Hop Dgt? n
  Suppress # Outpulsing? n                               Numbering Format: unk-pvt
Outgoing Channel ID Encoding: exclusive                 UII IE Treatment: service-provider

                                                         Replace Restricted Numbers? n
                                                         Replace Unavailable Numbers? n
                                                         Send Called/Busy/Connected Number: y

  Send UCID? n
Send Codeset 6/7 LAI IE? y                             Ds1 Echo Cancellation? n
                                                         Modify Reroute Number? y
Path Replacement with Retention? n
Path Replacement Method: always
                                                         Network (Japan) Needs Connect Before Disconnect? n

```

NOTE: There were no entries on Pages 3, 4, 5, for INCOMING CALL HANDLING TREATMENT.

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

```
display isdn private-numbering
```

ISDN NUMBERING - PRIVATE FORMAT

```

Network Level: 0          PBX Identifier:
Level 2 Code:           Deleted Digits: 0
Level 1 Code:

```

☐ 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
```

Page 1 of 3

Pattern Number: 1

Grp. No.	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Digits	DCS/ QSIG Intw	IXC
1:	1	0						y	user
2:								n	user
3:								n	user
4:								n	user
5:								n	user
6:								n	user

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

- ☐ 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 Unified Messenger Hunt Group. Below is an example of an AAR dialed string in **boldface**.

```
display aar analysis 15
```

Page 1 of 2

Percent Full: 3

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req
4573000	7	7	15	lev0		n

Note: The Call Type uses **lev0** when the trunk group is set for **unk-pvt** as it was programmed earlier.

- ☐ Within the AAR Digit Conversion Table, create entries that will allow the Unified Messenger server to dial internal extensions over the QSIG link.

```
list aar digit-conversion
```

AAR DIGIT CONVERSION REPORT							
Matching Pattern	Min	Max	Del	Replacement String	Net	Conv	ANI Req
0	1	28	0		ars	y	
1	4	28	0		ars	y	
2	4	4	0		ext	n	
3	4	4	0		ext	y	
4	4	4	0		ext	y	
5	4	4	0		ext	y	
6	4	4	0		ext	y	
7	4	4	0		ext	y	
x11	3	3	0		ars	y	

☐ Configure a Hunt Group to be used as the Call Coverage Point for the Call Coverage Path assigned to the Unified Messenger subscribers. This hunt group's extension number is going to be used as the Unified Messenger 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:

```
display hunt-group 1                                Page 1 of 60
                                                    HUNT GROUP

      Group Number: 1                                ACD? n
      Group Name:  QSIG T1                          Queue? n
      Group Extension: 3000                          Vector? n
      Group Type: ucd-mia                          Coverage Path:
      TN: 1                                Night Service Destination:
      COR: 1                                MM Early Answer? n
      Security Code:
      ISDN Caller Display: grp-name
```

```
display hunt-group 1                                Page 2 of 60
                                                    HUNT GROUP

      Message Center: qsig-mwi
      Voice Mail Number: 4573000
      Routing Digits (e.g. AAR /ARS Access Code): 107

      LWC Reception: none
```

```
display hunt-group 1                                Page 3 of 60
                                                    HUNT GROUP

      Group Number: 96      Group Extension: 3000      Group Type: ucd-mia
      Member Range Allowed: 1 - 1500      Administered Members (min/max): 0/0
                                                    Total Administered Members: 0
GROUP MEMBER ASSIGNMENTS
  Ext      Name (24 characters)              Ext      Name (24 characters)
  1:
  2:
  3:
  4:
  5:
  6:
  7:
  8:
  9:
  10:
  14:
  15:
  16:
  17:
  18:
  19:
  20:
  21:
  22:
  23:
At End of Member List
```

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

```
display coverage path 10
```

COVERAGE PATH			
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	Number of Rings: 4
All?	n	n	
DND/SAC/Goto Cover?	y	y	
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 to them. Also ensure the following setting for subscribers:

LWC Reception: spe (spe-msa for older PBX versions)

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.

Configuring the Unified Messenger

6.0 CONFIGURING THE UNIFIED MESSENGER SERVER

***Note:** The following screens reflect the latest version of the Unified Messenger. Please refer to the appropriate manual according to your system's release for older versions of Unified Messenger.*

- Configuring the Unified Messenger platform for proper PBX integration requires configuring several menus accessed within the **Voice Mail System Configuration** application.
- Access the **Voice Mail System Configuration** application from the Unified Messenger program group.

Note: Maximize the Voice Mail System Configuration application window. Also, expand all fields so that all applicable options are visible.

- Access the **Telephone User Interface**. Select **General**. Within this screen, set the number of digits in a mailbox. This number should match the number of digits of extension numbers on the customer's PBX. At this point, you can also choose to enable/disable the **Automated Attendant** and **Call Back Notification** features. All other fields and tabs are configurable according to your customer needs.
- Access the **Message Waiting Indicator** tab and set the following values:
 - **Enable Message Waiting Indicator (MWI)** = Enable by checking the box
 - MWI server** = Enter the name of the MWI server. Click  to locate the MWI server(s) installed. To add voice servers to the list of voice server that support MWI, click the Add icon  and type the name of the voice server.
 - Limit requests** = Leave Blank

Note: Refer to the Help screen under the Message Waiting Indicator section for Deleting voice servers and other MWI information.

- Return to the **Voice Mail Domain** and select **PBX's**. Access **Edit**, and select **Add New PBX**.
- Within the **Add New PBX** dialog box, select **Avaya G3 (QSIG)**
- From the **Voice Mail System Configuration** screen select the **Voice Mail Domain**, and then access **PBXs –Avaya G3 (QSIG)**
- Access the General **Avaya G3 (QSIG) PBX Configuration** tab and set the following values:
 - **DTMF Inter-Digit Delay during Dialing (ms)** = 80
 - DTMF Length during Dialing (ms)** = 80
 - DTMF Length during Detection (ms)** = 50

- Access the **Transfer/Outcall** tab and set the following values:
- **Transfer Mode = Blind**
- Access the **Tone Detection** tab and set the following values:
- **Maximum Silence before Hanging Up (ms) = 6000**
- Access the **Outgoing Call** tab and set the following values:

Layer Protocol = G.711 u-Law

BC Transfer Cap = Speech

Number Type = National

Number Plan = ISDN/Telephony (E.164/E.163)

Origin Number = 9999

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.

- Access the **General** tab within the **Telephony Interface** tab and set the following values:

Note: To activate the QSIG telephony interface, click on the down arrow and select **Make Active**. If the QSIG telephony interface is already active, this field does not appear.

Telephony Interface = Dialogic - QSIG

Playback Volume = 2

Maximum Concurrent Calls = Enter the number of ports connected to the PBX (i.e. 23)

Port = Enable ports by checking the checkbox next to the ports

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 23 ports enabled, then the first 10 ports will be enabled on one board, and all 23 on the second board. Use the port checkboxes to enable or disable the correct ports.

- Access the **General** tab within the **PBX Integration** property page and set the following values:
- **QSIG = Enable by checking the box**
- Select the **QSIG/SE** tab and set the following values:

Port Group Name = Enter the desire Port group Name

Max MWI Sessions = 1 (or the # of ports within the MWI port group)

Indicator On/Off signals must use same ports = Check if using MWI

MWI On = *4%s

MWI Off = #4%s

Note: The number of ports for MWI outgoing calls need to be reserved. Any port(s) can be used in the MWI port group, but they will need to be set to outgoing only and need to be in a separate trunk group on the switch.

- Access the **General** tab within the **PBX Type** tab and set the following values:
- Select: **Avaya G3 (QSIG)**

After making these changes, stop and restart the Unified Messenger server.

7.0 TESTING THE INTEGRATION

This test is simply to ensure the integration is working.

- ☐ Create two test mailboxes associated with two test extensions. Record a name and personal greeting for each mailbox.
- ☐ Ensure these test extensions have been forwarded under busy and ring-no-answer conditions to the pilot number on the UM system.
- ☐ Using one test extension, call the other test extension. Verify a personal greeting plays.
- ☐ Leave a message. Ensure MWI turns on.
- ☐ Verify the return-to-operator feature works properly.
- ☐ Call the voice-processing system from a test extension. Verify the recorded name is played immediately and the system prompts for a password.
- ☐ Review the message in the mailbox. Ensure MWI turns off.
- ☐ Delete the test mailbox and extensions.

8.0 CONSIDERATIONS/ALTERNATIVES

8.1 QSIG Integration does not support forwarding from a Vector.

Currently, if calls are routed from a vector to the QSIG link(s) connected to the Unified Messenger server, the call will not pass the VDN as the called party ID. Applications requiring calls that are routed from vectors to mailboxes on the Unified Messenger server can be configured so as to route calls to phantom extensions (X-ports) configured to call-cover all-calls to the Unified Messenger hunt group.

Testing the integration when
complete

Important notes regarding this
integration

8.2 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 Unified Messenger server 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.

8.3 Transfers to ringing use additional ports. When performing unsupervised transfer, and the transferred-to extension forwards back to the Unified Messenger server, additional ports are tied up on the UM 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.

8.4 When multiple Definity 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 Definity PBX's in the network must match the configuration of the trunk group form (page 1 and 2 of the form) assigned to the Unified Messenger QSIG trunks. An example of the trunk group administration form is illustrated on the PBX programming section of this document.

8.5 Support for Transfer to QSIG Voice Mail feature. In order to provide support for the Transfer to QSIG Voice Mail feature on your QSIG integrated Unified Messenger solution, you need to ensure the following criteria are met:

- Your switch will need to be running Definity software R8 or later.
- 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
- A Feature Access code for Transfer to QSIG Voice Mail, will need to be configured.
- Your Unified Messenger 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 transfer the caller directly to the voice mail for 7001 by pressing the transfer button, then the code for transfer to voice mail (*8,*50, etc.), then presses transfer again.

The caller immediately hears the greeting of 7001 (no additional ringing) and can subsequently leave a message.

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

8.6 Possible reasons for QSIG Path Replacement failure.

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 will see that both channels are now idle in Port Monitor, this shows that the QSIG Path Replacement feature has completed successfully.

However there are some switch configurations that will cause Path Replacement to fail, therefore your MM Server will stay bridge onto the transferred call, keeping two channels busy in Port Monitor. The following list of features can cause Path Replacement to fail:

Call Vector. A call that has experienced SS-CT (Supplementary Service Call Transfer) that terminated to a vector and received answer treatment can have its path replaced. This is allowed only after a true user answers the call so that vector processing is completed. A vector step of recorded announcement or wait listening to music could cause a CONNECT message to be sent back and make the far end think ANF-PR (Path Replacement Additional Network Feature) could take place. If the PR Propose is received while still in vector processing, the vectoring PBX will deny the ANF-PR attempt but will initiate its own attempt at ANF-PR when a user does answer the call.

Data Call Setup. ANF-PR platform will reject the ANF-PR invocation request if the old connection is a data call. ANF-PR may cause disruption in user information during switchover of the old connection to the new connection, so it will be undesirable to disrupt the data call. For a call where the Bearer Capability Class (BCC) or Information Transfer Capability, Octet 3 of Bearer Capability IE, indicates that the call is data call, ANF-PR shall not be allowed.

For a voice grade data and voice, DEFINITY uses the BCC value or Information Transfer Capability value of "3.1 KHZ audio", so using the bearer capability information would not be enough to distinguish between a voice call and voice grade data call. DEFINITY provides a feature "data pre-indication" which allows using the BCC of the data extension instead of the voice extension even though the call is originated using the voice extension. So, ANF-PR shall not be allowed on a call for which data pre-indication is activated.

Data Privacy. The "data privacy" feature protects analog data calls from being disrupted by the system's overriding or ringing features. For analog endpoints, which are used for data calls such as fax and data only, DEFINITY provides the feature "data privacy". So, ANF-PR shall not be allowed if "data privacy" is active on the old connection.

Data Restriction. The "data restriction" feature protects analog data calls from being disrupted by the system's overriding or ringing features. For analog endpoints, which are used for data calls such as fax and data only, DEFINITY provides the feature "data restriction". So, ANF-PR shall not be allowed if "data restriction" is active on the old connection.

Malicious Call Trace (MCT). While Malicious Call Trace (MCT) is active and if there is a trunk involved in the call, MCT feature does not allow the trunk resources to be released from the switch-side to facilitate the tracing activity. Also, a MCT controller on switch A may request a controller on switch B to continue tracing a call that was tandem through switch B by providing the trunk member id they wish to have traced. So, ANF-PR shall not be done while MCT is active on the call.

Restriction Features. *Class of Restriction (COR).* The restrictions placed on routing calls are in affect for ANF-PR. Any call that cannot be originated or terminated because of COR on a regularly dialed basis will not be originated or terminated when that call is made ON BEHALF OF that terminal by ANF-PR.

Voice Terminal Restrictions. Voice Terminal Restrictions for the reroute of an ANF-PR call will be enforced.

Inward. If a terminal is Inward restricted then that terminal would not be able to accept an incoming call. This would include a new path SETUP message. Therefore a Requesting PBX should not bother proposing ANFPR, since it is destined to fail.

Manual Terminating Line. If a terminal is Manual Terminating restricted then that terminal would not be able to accept an incoming call except from an attendant. Thus a new path SETUP would be denied. Therefore a Requesting PBX should not bother proposing ANF-PR, since it is destined to fail.

Origination. If a terminal is Origination restricted then that terminal would not be able to make an outgoing call. Thus originating a new path SETUP should be

denied. Therefore a Cooperating PBX should reject the path replacement proposal/request.

Outward. If a station is outward restricted at the cooperating end of an ANF-PR call then no ANF-PR SETUP should be attempted and ANF-PR should fail. Any call that cannot be originated outward (because of COR) on a regularly dialed basis will not be allowed to originate outward when that call is made ON BEHALF OF that terminal by ANF-PR.

Termination. If a terminal is Termination restricted then that terminal would not be able to accept an incoming call. Thus an incoming new path SETUP would be denied. Therefore a Requesting PBX should not bother proposing ANF-PR, since it is destined to fail.

TAC (Trunk Access Code). When an outgoing call is made using a TAC, or a call was extended by an attendant using DTGS (Direct Trunk Group Selection), the user has intentionally chosen a particular Trunk Group for the outgoing call. ANF-PR will not replace the path in this case.

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 UM will reside (Hub Node) before continuing with the networking configuration. Configure the remote switch (i.e. Node 2), Definity G3, Prologix, etc., by following the screens below.

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

NOTICE:

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

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

change signaling-group 7		Page 1 of 5	
SIGNALING GROUP			
Group Number: 7		Group Type: isdn-pri	
Associated Signaling? y		Max number of NCA TSC: 16	
Primary D-Channel: 01A0824		Max number of CA TSC: 31	
		Trunk Group for NCA TSC: 4	
Trunk Group for Channel Selection: 4			
Supplementary Service Protocol: a			
change signaling-group 7		Page 2 of 5	
ADMINISTERED NCA TSC ASSIGNMENT			
Service/Feature:		As-needed Inactivity Time-out (min):	
TSC	Local		Mach.
Index	Ext.	Enabled	Established
1:	2299	y	permanent
2:	3081	y	permanent
3:		n	
4:		n	
5:		n	
etc..			

Note: The **Group Type** will depend on the customer's environment. The trunks can either be T1, E1, IP, etc.. For example, they could be:
Group Type: **isdn-pri** or Group Type: **h.323**

☐ 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 UM.

change isdn qsig-dcs-tsc-gateway

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

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

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

change isdn dcs-qsig-tsc-gateway

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

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

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

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

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

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

display hunt-group 2		Page 2 of 10
HUNT GROUP		
Message Center: rem-vm		
Voice Mail Extension: 3000		
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                                     Page 3 of 10
                                     HUNT GROUP
      Group Number: 2      Group Extension: 6300      Group Type: ucd-mia
Member Range Allowed: 1 - 200      Administered Members (min/max): 0 /0
                                     Total Administered Members: 0
GROUP MEMBER ASSIGNMENTS
  Ext      Name (24 characters)      Ext      Name (24 characters)
  1 :                                           14 :
  2 :                                           15 :
  3 :                                           16 :
  4 :                                           17 :
  5 :                                           18 :
  6 :                                           19 :
  7 :                                           20 :
  8 :                                           21 :
  9 :                                           22 :
 10 :                                           23 :
 11 :                                           24 :
 12 :                                           25 :
 13 :                                           26 :

      At End of Member List
  
```

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

```

display uniform-dialplan 0                               Page 1 of 2
                                     UNIFORM DIAL PLAN TABLE
                                     Percent Full: 0
Matching      Insert      Node      Matching      Insert      Node
Pattern  Len Del Digits Net Conv Num Pattern  Len Del Digits Net Conv Num
  2         4  0  221  aar  n         2         4  0  221  aar  n
 30         4  0  221  aar  n        30         4  0  221  aar  n
3000       4  0  457  aar  n        3000       4  0  457  aar  n
 31         4  0  221  aar  n         31         4  0  221  aar  n
  5         4  0  221  aar  n         5         4  0  221  aar  n
 63         4  0      ext  n         63         4  0      ext  n
 64         4  0      ext  n         64         4  0      ext  n
                                     n
                                     n
                                     n
                                     n
  
```

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

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

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

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

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

```
display coverage path 3
```

COVERAGE PATH			
Coverage Path Number: 3		Hunt after Coverage? n	
Next Path Number:		Linkage	
COVERAGE CRITERIA			
Station/Group Status	Inside Call	Outside Call	
Active?	n	n	
Busy?	y	y	
Don't Answer?	y	y	Number of Rings: 4
All?	n	n	
DND/SAC/Goto Cover?	y	y	
COVERAGE POINTS			
Terminate to Coverage Pts. with Bridged Appearances? n			
Point1: h2	Point2:	Point3:	
Point4:	Point5:	Point6:	

- ☐ Configure the remote subscriber stations, assigning the newly created Call Coverage Path to them.

These screens should allow remote PBXs to integrate with the Unified Messenger. Follow section 7.0 to confirm the installation was done properly.

CHANGE HISTORY		
Revision	Issue Date	Reason for Change
DRAFT 0.1	02/10/02	Initial release for review/validation
DRAFT 0.2	03/24/02	Incorporate comments
DRAFT 0.3	05/17/02	Added PBX screen updates
DRAFT 0.4	08/19/02	Added Transfer to QSIG Consideration
DRAFT 0.5	09/20/02	Added TN464 as the supported PBX H/W
DRAFT 0.6	10/16/02	Updated Transfer to QSIG Consideration
DRAFT 0.7	11/21/02	Included DCS+ PBX programming info.
Version A	11/24/02	GA
Version B	01/03/03	Updated grammar errors.
Version C	02/17/03	Updated PBX supported version to 10.1
Version D	03/28/03	MWI Interrogation changes
Version E	05/21/03	MWI Port programming updated
Version F	05/27/03	Path Replacement consideration added
Version G	08/11/03	Cleaned up PBX Screens and Changed DCS+ Networking as per info provided by Tier 4 PBX engineering.
Version H	10/28/03	Updated Route Pattern for remote PBX (p25) to support DCS/QSIG Internetworking

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