Avaya Solution & Interoperability Test Lab

Configuring Avaya Communication Manager for Voice Mail Service and Dialing Transparency for Sites Using an Enterprise Survivable Server (ESS) - Issue 1.0

Abstract

These Application Notes complement previously published Application Notes by illustrating enhancements to Avaya Communication Manager that enable messaging service and dialing transparency for sites controlled by an Enterprise Survivable Server (ESS). The "Centralized Messaging Coverage for Survivable Servers" enhancement enables simplified provisioning of voice mail services for sites controlled by either an Avaya S8300 Media Server Local Survivable Processor (LSP) or an Enterprise Survivable Server (ESS). The same enhancement also enables dialing transparency for users served by a survivable server (LSP or ESS). Configuration steps are presented that allow a branch user to dial the extension of a user at the main site, whether service to the branch is currently being provided by the primary server at the main site or an ESS or LSP at a branch site. The sample configuration includes an Avaya S8700 Media Server pair and Avaya Modular Messaging at a main site, an Avaya Compact Modular Cabinet (CMC) Media Gateway and Avaya S8500 Media Server ESS at one branch site, and an Avaya G350 Media Gateway with S8300 Media Server LSP at another branch site.

1. Introduction and Scope

These Application Notes present a sample configuration for a network comprised of three sites. At Site A, an Avaya S8700 Media Server pair and Avaya Modular Messaging provide call processing and messaging services to all sites. Site B uses an Avaya G350 Media Gateway with an Avaya S8300 Media Server licensed as a Local Survivable Processor (LSP). Site C uses an Avaya CMC Media Gateway with an S8500 Media Server licensed as an Enterprise Survivable Server. Under normal operation, the Avaya S8700 Media Server controls the gateways and IP Telephones at Site B and Site C across a data WAN. The gateways at Site B and Site C are each equipped with a (simulated) Public Switched Telephone Network (PSTN) ISDN-PRI trunk interface for inbound and outbound calls. If a WAN failure occurs, isolating a branch site, the survivable processor at the site can provide call processing, with voice messaging services preserved, as described in these Application Notes. Configuration steps are also presented that allow users at Site B or Site C to dial a user at Site A in the same fashion (i.e., by extension) whether service is currently being provided by the primary servers at Site A or a survivable processor.

These Application Notes complement previously published Application Notes that all use a similar network configuration, and all relate to providing messaging service when a survivable processor becomes active. Refer to Section 9 for a list of references. These Application Notes use Avaya Communication Manager Release 3.1, which enables the “Centralized Messaging Coverage for Survivable Servers” enhancement to apply to Enterprise Survivable Servers. Reference [1] uses Avaya Communication Manager Release 3.0.1 to illustrate the use of the “Centralized Messaging Coverage for Survivable Servers” enhancement for branch office users served by an S8300 LSP. Reference [2] was written prior to the availability of the “Centralized Messaging Coverage for Survivable Servers” enhancement, and provides alternate solutions to similar issues.

The configuration steps that enable voice mail service and dialing transparency, even when the survivable processor is forced active, are presented in Section 5. Since a thorough understanding of the behavior while the ESS is active requires a foundational understanding of the entire configuration, Section 4 illustrates the relevant Avaya Communication Manager configuration supporting Modular Messaging voice mail service. Detailed status screens and call flow traces are presented in Section 7, to reinforce understanding of the configuration.

Section 6 contains screen captures illustrating aspects of the Modular Messaging configuration. The few Modular Messaging configuration screens illustrated in these Application Notes are intended to be descriptive of the configuration, but not prescriptive of a unique approach. Compared to reference [2] (i.e., before availability of “Centralized Messaging Coverage for Survivable Servers”), these Application Notes and reference [1] remove the prior requirement for additional DID numbers and multiple extensions associated with the same subscriber mailbox. Although Avaya Modular Messaging was used in this configuration, the enhancements to Avaya Communication Manager described in Section 5 are intended to work with a variety of messaging systems.

Figure 1 provides a high-level overview of the network used to verify these Application Notes. Although this network uses S8700 Media Servers with MCC1, CMC, and G350 Media Gateways, the same approach documented in these Application Notes could be used with other Avaya Media Servers and Media Gateways. For example, the same procedures could be followed if Site A used an S8500 Media Server and G650 Media Gateway, Site B used a G700 Media Gateway with an S8300 Media Server LSP, and Site C used a SCC or G650 Media Gateway with S87x0 Media Servers licensed as an ESS.

Except where noted, it is assumed that users at Site B and Site C receive inbound calls from the PSTN trunk interface in the local media gateway. That is, Site B users receive PSTN calls from the trunk in the G350 Media Gateway, and Site C users receive PSTN calls from the trunk in the CMC Media Gateway.

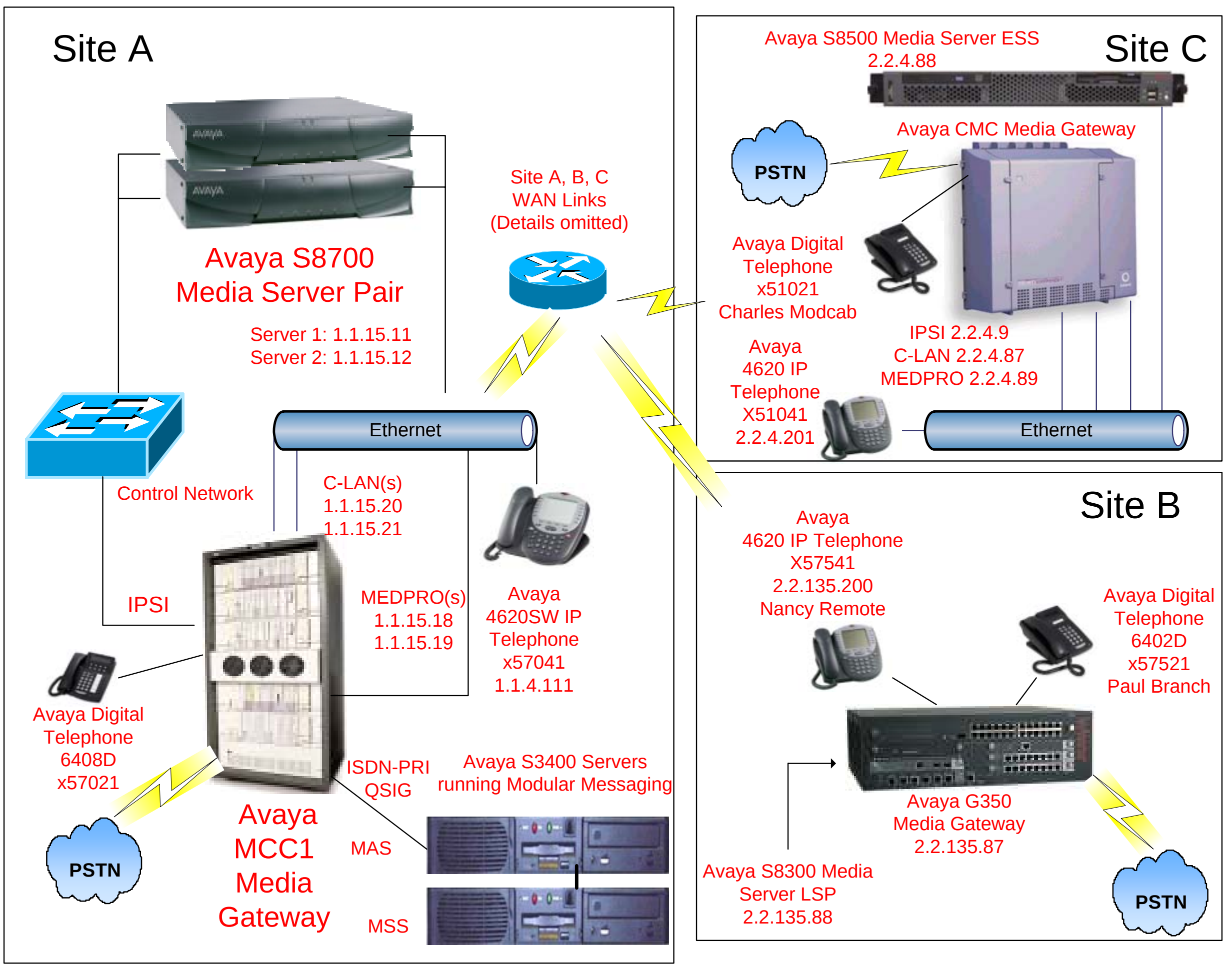


Figure 1: Network Overview

2. Equipment and Software Validated

Table 1 shows the relevant equipment and version information used in the sample configuration.

Network Component	Version Information
Avaya S8700 Media Server	Communication Manager 3.1 Load 628.3
Avaya S8500 Media Server ESS	Communication Manager 3.1 Load 628.3
Avaya S8300 Media Server LSP	Communication Manager 3.1 Load 628.3
Avaya TN2312BP IPSI in MCC1	FW 21
Avaya TN799DP C-LAN in MCC1	TN799DP HW01 FW015
Avaya TN2302AP MEDPRO in MCC1	TN2302AP HW03 FW093
Avaya TN464GP DS1 in MCC1 (to MAS)	HW2 FW17
Avaya TN2312BP IPSI in CMC	FW 21
Avaya TN799DP C-LAN in CMC	TN799DP HW01 FW012
Avaya TN2302AP MEDPRO in CMC	TN2302AP HW03 FW093
Avaya G350 Media Gateway	25.22.0
Avaya S3400 Servers running Modular Messaging	Modular Messaging 1.1
Avaya 4620SW IP Telephones	2.3
Avaya 6400 Series Digital Telephones	-

Table 1 – Equipment Version Information

3. Conventions and Assumptions

In these Application Notes, Avaya Communication Manager administration software screens are shown with a gray shaded background. These screens are also referred to as "SAT" (System Access Terminal) screens. In some instances, the information from the original screen has been edited for brevity and clarity in presentation. Unless otherwise noted, each screen capture is preceded by the text that references the screen capture.

Steps that aid in understanding the configuration, but are not central to the scope, are shown with "display" or "list" commands, to emphasize that this aspect should not be interpreted as a procedurally complete configuration flow. Configuration steps directly associated with enabling voice mail service and dialing transparency for sites served by an Enterprise Survivable Server are shown using the relevant "add" and "change" commands.

It is assumed that the appropriate license files and authentication files have been installed, and that login and password credentials are available to the reader. To verify that the installed license grants permission to use the features illustrated in these Application Notes, use the command "display system-parameters customer-options" as shown in the following series of screens. Ensure that the bolded fields show "y". If the bolded feature options are not enabled, a new license must be obtained to re-create the configuration described in these Application Notes.

On Page 3, ensure that ARS is enabled.

```
display system-parameters customer-options                     Page   3 of  11
                                OPTIONAL FEATURES
    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? n
        Analog Trunk Incoming Call ID? n Backup Cluster Automatic Takeover? n
 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  Computer Telephony Adjunct Links? n
          ARS/AAR Dialing without FAC? n      Co-Res DEFINITY LAN Gateway? n
          ASAI Link Core Capabilities? y   Cvg Of Calls Redirected Off-net? y
          ASAI Link Plus Capabilities? y                      DCS (Basic)? y
       Async. Transfer Mode (ATM) PNC? n                DCS Call Coverage? y
  Async. Transfer Mode (ATM) Trunking? y               DCS with Rerouting? y
              ATM WAN Spare Processor? n
                                ATMS? y    Digital Loss Plan Modification? y
                  Attendant Vectoring? y                          DS1 MSP? y
                                                      DS1 Echo Cancellation? Y
```

On Page 4, ensure that ISDN-PRI is enabled. The Modular Messaging integration in these Application Notes uses ISDN-PRI. ISDN-PRI trunks are also used in the Avaya CMC gateway for (simulated) PSTN access, but the enhancements described in Section 5 are not intended to be limited to ISDN trunks. In these Application Notes, IP Stations and Media Encryption over IP are used, but these are not required in general. To use the optional “origination mapping” illustrated in Section 7.2.10, the “Enhanced EC500” feature is required. Note that the “Enterprise Survivable Server” field should show “n” for the S8700 Media Servers, but show “y” for the S8500 Media Server licensed as an ESS.

```
display system-parameters customer-options                     Page   4 of  11
                                OPTIONAL FEATURES
   Emergency Access to Attendant? y                           IP Stations? y
           Enable 'dadmin' Login? y             Internet Protocol (IP) PNC? y
           Enhanced Conferencing? y                    ISDN Feature Plus? y
                  Enhanced EC500? y         ISDN Network Call Redirection? y
    Enterprise Survivable Server? n                      ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                             ISDN-PRI? y
              ESS Administration? y            Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                  Malicious Call Trace? y
     External Device Alarm Admin? n              Media Encryption Over IP? y
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y               Multifrequency Signaling? y
      Global Call Classification? n Multimedia Appl. Server Interface (MASI)? n
             Hospitality (Basic)? y        Multimedia Call Handling (Basic)? n
 Hospitality (G3V3 Enhancements)? y     Multimedia Call Handling (Enhanced)? n
                       IP Trunks? y
           IP Attendant Consoles? y
```

To use the routing configuration shown in Section 5, ensure that the bolded features are enabled on Page 5.

```
display system-parameters customer-options                      Page   5 of  11
                                OPTIONAL FEATURES

                Multinational Locations? n            Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n      Station as Virtual Extension? y
                     Multiple Locations? y
                                             System Management Data Transfer? y
          Personal Station Access (PSA)? y              Tenant Partitioning? y
                       Posted Messages? y       Terminal Trans. Init. (TTI)? y
                       PNC Duplication? n              Time of Day Routing? y
                   Port Network Support? y             Uniform Dialing Plan? y
                                               Usage Allocation Enhancements? y
               Processor and System MSP? y      TN2501 VAL Maximum Capacity? y
                     Private Networking? y
                     Processor Ethernet? y               Wideband Switching? n
                                                                   Wireless? y
                          Remote Office? n
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y
```

To implement the method of dialing transparency for non-DID users at Site A (as illustrated in Section 5), vectoring must be enabled as shown in the following screen.

```
display system-parameters customer-options                      Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES
                          Call Center Release: 3.0

                                ACD? y                          Reason Codes? n
                       BCMS (Basic)? y              Service Level Maximizer? n
         BCMS/VuStats Service Level? y            Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? n    Service Observing (Remote/By FAC)? n
                  Business Advocate? n            Service Observing (VDNs)? n
                    Call Work Codes? n                           Timed ACW? n
      DTMF Feedback Signals For VRU? n                   Vectoring (Basic)? y
                   Dynamic Advocate? n               Vectoring (Prompting)? y
       Expert Agent Selection (EAS)? y            Vectoring (G3V4 Enhanced)? n
                            EAS-PHD? n             Vectoring (3.0 Enhanced)? n
                   Forced ACD Calls? n    Vectoring (ANI/II-Digits Routing)? n
               Least Occupied Agent? n    Vectoring (G3V4 Advanced Routing)? n
          Lookahead Interflow (LAI)? n                    Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n     Vectoring (Best Service Routing)? n
    Multiple Call Handling (Forced)? n                 Vectoring (Holidays)? n
  PASTE (Display PBX Data on Phone)? n                Vectoring (Variables)? n
```

On Page 8, ensure that the desired QSIG features are enabled (i.e., for the Avaya Modular Messaging integration type used in this configuration).

```
display system-parameters customer-options                      Page   8 of  11
                             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
```

If calls arriving over the PSTN from Site C users to Site A trunks are to appear to have been originated from the extension of the Site C user, preserving "local station to station" characteristics for PSTN trunk calls from Site C to Site A when the S8500 ESS is active, then Extension to Cellular licenses must be available for Site C users. Section 7.2.10 presents the optional configuration.

```
display system-parameters customer-options                       Page   1 of  11
                                OPTIONAL FEATURES
     G3 Version: V13
       Location: 1                            RFA System ID (SID): 1
       Platform: 6                            RFA Module ID (MID): 1

                                                               USED
                                Platform Maximum Ports: 44000 171
                                      Maximum Stations: 36000 52
                              Maximum XMOBILE Stations: 20    3
                    Maximum Off-PBX Telephones - EC500: 10    3
```

4. Foundational Configuration of Avaya Communication Manager Supporting Figure 1

This section illustrates foundational aspects of the Avaya Communication Manager configuration for **Figure 1** that are not directly associated with the enhancements enabling survivable messaging and dialing transparency. Note that the integration with Modular Messaging and the Modular Messaging software are unchanged from the previously published Application Notes [1] and [2]. The Avaya Communication Manager enhancements for both LSP and ESS configurations can be utilized without posing new requirements on the messaging system. In **Figure 1**, the Avaya Messaging Application Server (MAS) is connected to the MCC1 at Site A using a T1/ISDN-PRI with QSIG enabled. Reference [4] can be consulted for a more procedural description of the QSIG method of integration, inclusive of Modular Messaging configuration.

4.1. DS1, Signaling Group, Trunk Group to Modular Messaging MAS

The following screen shows the DS1 board configuration for the physical interface to the Modular Messaging MAS. Note the use of QSIG protocol. The bit rate, line coding, and framing mode must match the provisioning of the MAS physical interface, with commonly used options shown.

```
display ds1 1a08                                                Page   1 of   2
                                DS1 CIRCUIT PACK
            Location: 01A08                            Name: T1-QSIG-MAS
            Bit Rate: 1.544                     Line Coding: b8zs
   Line Compensation: 1                        Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                         Interface: peer-master
   TN-C7 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
   Echo Cancellation? n
```

The following screen shows that signaling group 34 is an ISDN-PRI signaling group using the DS1 board located in slot 1A08 of the MCC1 Media Gateway. Supplementary Service Protocol B is selected for QSIG features.

```
display signaling-group 34
                                SIGNALING GROUP
 Group Number: 34               Group Type: isdn-pri
                    Associated Signaling? y         Max number of NCA TSC: 1
                       Primary D-Channel: 01A0824     Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC: 34
       Trunk Group for Channel Selection: 34      X-Mobility/Wireless Type: NONE
           Supplementary Service Protocol: b          Network Call Transfer? n
```

The following screens describe trunk group 34, the ISDN-PRI trunk group associated with signaling group 34. The reader may notice changes in the trunk group screens compared with reference [1], but these reflect changes in the layout of the screens in Avaya Communication Manager Release 3.1 compared with prior releases, not changes to the configuration.

```
display trunk-group 34                                          Page   1 of  21
                                TRUNK GROUP
Group Number: 34                   Group Type: isdn          CDR Reports: y
  Group Name: QSIG T1 TO S3400 MAS        COR: 1        TN: 1        TAC: 134
   Direction: two-way       Outgoing Display? n          Carrier Medium: PRI/BRI
 Dial Access? n               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
```

On Page 2, observe the use of Supplementary Service Protocol “b” associated with QSIG integration.

```
display trunk-group 34                                          Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 0     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
Incoming Calling Number - Delete:     Insert:                Format: lev0-pvt
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 0
```

From Page 3, observe that the "Calling Number" (ANI) is sent to Modular Messaging. This allows Modular Messaging to present the calling number information, where appropriate. For example, the calling number can be included in the "envelope information" for call answering messages.

```
display trunk-group 34                                          Page   3 of  21
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     Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: unk-pvt
 Outgoing Channel ID Encoding: exclusive     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                          Send Called/Busy/Connected Number: y
                                                  Hold/Unhold Notifications? y
             Send UUI IE? y                     Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                          Ds1 Echo Cancellation? n

                             Network (Japan) Needs Connect Before Disconnect? n
                                                       Apply Local Ringback? n
```

On Page 4, "Path Replacement" allows Modular Messaging and Avaya Communication Manager to release trunk members when no longer required to service the connection.

```
display trunk-group 34                                          Page   4 of  21
                       QSIG TRUNK GROUP OPTIONS
           Diversion by Reroute? y
               Path Replacement? y
Path Replacement with Retention? n
        Path Replacement Method: always
                            SBS? n
  Display Forwarding Party Name? y
    Character Set for QSIG Name: eurofont
```

The following screen shows that the members of trunk group 34 are associated with signaling group 34. Only the first several members of the trunk group are illustrated. As documented in reference [4], a separate trunk group (not illustrated here) can be created specifically for Message

Waiting Indication (MWI) purposes. The “port groups” on the Modular Messaging MAS are similarly configured, with some ports used for inbound / outbound calls, and others dedicated to the MWI function.

```
display trunk-group 34                                          Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/22
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  22

       Port    Code Sfx Name        Night           Sig Grp
  1: 01A0801  TN464  G                                34
  2: 01A0802  TN464  G                                34
  3: 01A0803  TN464  G                                34
  4: 01A0804  TN464  G                                34
  ………………………
```

4.2. Feature Access Codes

The following screen illustrates the feature access codes used as part of the configuration. The digit “8” is the AAR access code, and the digit “9” is the ARS access code.

```
display feature-access-codes                                    Page   1 of   7
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *5
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                  ………………………………………………………………………………………………………………
```

4.3. Hunt Group for Access to Modular Messaging MAS

The following screens illustrate the configuration for hunt group 34. As in reference [1], hunt group 34 will use extension 57555, as shown on Page 1. The configuration of hunt group 34 differs in these Application Notes compared to reference [1] in that a coverage path has been configured. Section 10 of reference [1] described an enhancement that would allow coverage to apply for a QSIG-integrated hunt group to Modular Messaging. This enhancement has become available, and is used in these Application Notes. The enhancement enables Site B or Site C users to dial the messaging system by extension (e.g., for message retrieval), even when the LSP or ESS is actively providing service.

```
change hunt-group 34                                            Page   1 of  60
                                  HUNT GROUP
            Group Number: 34                               ACD? n
              Group Name: QSIG T1 Hunt S3400 MAS         Queue? n
         Group Extension: 57555                         Vector? n
              Group Type: ucd-mia                 Coverage Path: 575
                      TN: 1         Night Service Destination:
                     COR: 1                   MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display: grp-name
```

On Page 2, note the bolded fields in the following screen. In this configuration, the "Routing Digits" field contains the AAR access code "8". The "Voice Mail Number" field contains the number "2347555", which will be used to route a call to this hunt group through AAR to a routing pattern containing the trunk group to Modular Messaging.

```
change hunt-group 34                                            Page   2 of  60
                                  HUNT GROUP
                            LWC Reception: none      AUDIX Name:

                           Message Center: qsig-mwi
                     Send Reroute Request: y
                        Voice Mail Number: 2347555
Routing Digits (e.g. AAR/ARS Access Code): 8           Provide Ringback? n
```

On Page 3 and beyond, note that it is not necessary to add "members" to this type of hunt group.

```
display hunt-group 34                                            Page   3 of  60
                                  HUNT GROUP
          Group Number: 34      Group Extension: 57555     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:                                     14:
   2:                                     15:
   3:                                     16:
......................................................................
  At End of Member List
```

4.4. Routing to Modular Messaging

The following screen illustrates the AAR analysis form, showing how the digits "2347555" (configured for hunt group 34) are routed. Calls to hunt group 34 will use route pattern 34.

```
display aar analysis 234                                         Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                           Percent Full:    2
          Dialed           Total     Route     Call   Node  ANI
          String         Min  Max  Pattern    Type   Num   Reqd
    234                  7    7      34       lev0         n
```

The following screen shows the configuration for route pattern 34. The digit manipulation causes hunt group extension 57555 to be sent over trunk group 34 to Modular Messaging. With respect to digit manipulation, customer circumstance may vary.

```
display route-pattern 34                                        Page   1 of   3
                    Pattern Number: 34  Pattern Name: QSIG S3400 MAS
                                          Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                                 DCS/ IXC
    No          Mrk Lmt List Del  Digits                                   QSIG
                             Dgts                                          Intw
 1: 34   0                    3   5                                         n   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  none      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
```

4.5. System Features Showing QSIG Parameters

The following screen illustrates the existence of a QSIG TSC extension, as well as the five-digit numbering for voice mail subscribers.

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

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

                              QSIG TSC Extension: 57600
 MWI - Number of Digits Per Voice Mail Subscriber: 5
                                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? n
                  QSIG Path Replacement Extension:
            Path Replace While in Queue/Vectoring? n
```

4.6. Node Names

The following screen illustrates a subset of the node names used in this configuration. These node names will appear in other screens depicting the configuration and verification.

```
list node-names                                                      Page   1
                        NODE NAMES
Type      Name             IP Address
IP        CLAN-EPN1        1  .1  .15 .20
IP        CLAN2-EPN1       1  .1  .15 .21
IP        Clan-CMC1        2  .2  .4  .87
IP        EPN1-PROWL1      1  .1  .15 .18
IP        EPN1-PROWL2      1  .1  .15 .19
IP        ESSCid002Sid003  2  .2  .4  .88
IP        Medpro-CMC1      2  .2  .4  .89
IP        G350-2-Right     2  .2  .135.87
IP        S8300-G350-LSP   2  .2  .135.88
```

4.7. Avaya Communication Manager Configuration for the G350 Media Gateway in Site B

The following screen illustrates the Avaya Communication Manager configuration of the G350 Media Gateway in Site B. The screen was captured after the G350 Media Gateway had registered with a C-LAN (1.1.15.20) in the MCC1 Media Gateway. Note the Network Region and Location number "3", also shown in bold.

```
display media-gateway 1
                              MEDIA GATEWAY
         Number: 1                                IP Address: 2  .2  .135.87
           Type: g350               FW Version/HW Vintage: 25 .22 .0  /1
           Name: G350-Right                      MAC Address: 00:04:0d:29:c9:91
      Serial No: 03IS07589448                   Encrypt Link? y
 Network Region: 3                                  Location: 3
     Registered? y              Controller IP Address: 1  .1  .15 .20
  Recovery Rule: 2                                 Site Data:
      Slot   Module Type                Name
       V1:   S8300                      ICC MM
       V2:
       V3:
       V4:   MM710                      DS1 MM
       V5:
       V6:   MM312                      DCP MM
       V7:   1T+2L-Integ-Analog         ANA IMM
       V8:
       V9:   gateway-announcements      ANN VMM
```

4.8. IP Network Map

Site B uses the IP Address range 2.2.135.100 – 2.2.135.254 for Avaya IP Telephones. Site C uses the IP Address range 2.2.4.201 – 2.2.4.254. The bolded rows of the “ip-network-map” shown below associate these ranges with network region 3 (Site B) and network region 8 (Site C). The Site B IP Telephones are mapped to the same network region assigned to the Site B G350 Media Gateway. The Site C IP Telephones are mapped to the same network region as the TN2302AP MEDPRO in the Avaya CMC Media Gateway in Site C.

```
change ip-network-map                                           Page   1 of  32
                         IP ADDRESS MAPPING
                                                              Emergency
                                     Subnet                   Location
 From IP Address  (To IP Address   or Mask)  Region    VLAN   Extension

 2  .2  .4  .201  2  .2  .4  .254             8        n
 2  .2  .135.100  2  .2  .135.254             3        n
    .   .   .         .   .   .                        n
```

4.9. Network Region Configuration

In the network of **Figure 1**, Site B is configured in network region 3, and Site C is configured in network region 8. The configuration of network region 3 is illustrated in the following screen. Observe from the bold Location field that network region 3 is configured for location 3. This allows IP Telephones in network region 3 to be associated with location 3. IP Telephones at Site B are considered to be in network region 3 as a result of the IP network map configured in the previous section. Codec set 1 will be used for connections within region 3 (i.e., IP media connections local to Site B).

```
change ip-network-region 3                                       Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location: 3       Authoritative Domain:
    Name: G350-2-Right
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 34      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
             Keep-Alive Count: 5
```

On Page 2, the node name of the S8300 Media Server LSP is configured. This enables Avaya Communication Manager to provide the IP address of the S8300 Media Server LSP to the IP Telephones in network region 3. When the data WAN is down, the IP Telephones can automatically re-register to the S8300 LSP.

```
change ip-network-region 3                                          Page   2 of  19
                               IP NETWORK REGION
INTER-GATEWAY ALTERNATE ROUTING
 Incoming LDN Extension:
 Conversion To Full Public Number - Delete:    Insert:
 Maximum Number of Trunks to Use:

 BACKUP SERVERS IN PRIORITY ORDER    SECURITY PROCEDURES
 1   S8300-G350-LSP                  1   pin-eke
 2                                   2   any-auth
 3                                   3
 4                                   4
 5
 6
```

The following screen illustrates the inter-region configuration for network region 3. Although not the focus of these Application Notes, inter-region connection management allows for configuration of Call Admission Control (i.e., limits based on bandwidth or number of calls), Inter-Gateway Alternate Routing (i.e., alternate TDM paths to be used when Call Admission Control thresholds are reached) and codec set selection (i.e., determining the G.7xx codec to use, encryption parameters, etc.). Observe that calls between region 3 (Site B) and region 1 (Site A) and region 3 and region 8 (Site C) will use codec-set 3 (shown in Section 4.10).

```
change ip-network-region 3                                          Page   3 of  19
                   Inter Network Region Connection Management

 src dst  codec  direct                                       Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions     Gateway    IGAR
 3   1     3       y           :NoLimit                                     n
 3   2     1       y           :NoLimit                                     n
 3   3     1
 3   4     4       y           :NoLimit                                     n
 3   5
 3   6
 3   7
 3   8     3       y           :NoLimit                                     n
………………………………………………………………………………………………………………………
```

The configuration of network region 8 is illustrated in the following screen. Observe from the bold Location field that network region 8 is configured in location 4. This allows IP Telephones in network region 8 to be associated with location 4. Fundamentals of location based routing are summarized in Section 5.7. IP Telephones at Site C are considered to be in network region 8 as a result of the IP network map. Codec set 1 will be used for connections within region 8 (i.e., IP media connections local to Site C).

```
change ip-network-region 8                                          Page   1 of  19
                               IP NETWORK REGION
  Region: 8
Location: 4       Authoritative Domain:
    Name: CMC-and-S8500-ESS
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3028
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 34      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

The following screen illustrates the inter-region configuration for network region 8. Observe that calls between region 8 (Site C) and region 1 (Site A) and region 8 and region 3 (Site B) will use codec-set 3.

```
change ip-network-region 8                                          Page   3 of  19
                   Inter Network Region Connection Management

 src dst  codec  direct                                      Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions    Gateway    IGAR
 8   1     3       y           :NoLimit                                    n
 8   2
 8   3     3       y           :NoLimit                                    n
 8   4
 8   5
 8   6
 8   7
 8   8     1
 8   9
```

4.10. Codec Set Configuration

Codec set 1 is used for intra-region connectivity for all three sites. Codec set 1 has been configured for G.711MU.

```
change ip-codec-set 1                                           Page   1 of   2
                          IP Codec Set
    Codec Set: 1
    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
 3:
 4:
 5:
 6:
 7:
     Media Encryption
 1: none
 2:
 3:
```

Codec set 3 is used for inter-region connectivity. Codec set 3 has been configured for G.729 and media encryption (see reference [3] for Application Notes focusing on media encryption using a similar network configuration). Under normal conditions, when a user at Site B or Site C is interacting with the Modular Messaging MAS at Site A, the VoIP connection will follow the codec set 3 configuration shown below.

```
change ip-codec-set 3                                           Page   1 of   1
                          IP Codec Set
    Codec Set: 3
    Audio      Silence      Frames   Packet
    Codec      Suppression  Per Pkt  Size(ms)
 1: G.729           n         2        20
 2:
 3:
 4:
 5:
 6:
 7:
     Media Encryption
 1: aes
 2: aea
 3:
```

4.11. ISDN-PRI Trunk Group in the G350 Media Gateway at Site B

The following screens show that signaling group 8 and trunk group 8 define an ISDN-PRI link to the (simulated) PSTN, using the MM710 Media Module in the G350 Media Gateway in Site B.

```
display signaling-group 8                                      Page   1 of   5
                                SIGNALING GROUP
 Group Number: 8              Group Type: isdn-pri
                    Associated Signaling? y          Max number of NCA TSC: 0
                       Primary D-Channel: 001V424     Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 8       X-Mobility/Wireless Type: NONE
          Supplementary Service Protocol: a          Network Call Transfer? n
```

Trunk group 8 can be used for inbound trunk calls to Site B users, and outbound PSTN calls from Site B users, per the configured routing. For example, this is the trunk group that will be used for Site B connections to Site A, when the data WAN is down and the S8300 Media Server LSP is active. The following screen shows the configuration of trunk group 8.

```
display trunk-group 8                                          Page   1 of  21
                                TRUNK GROUP
Group Number: 8                    Group Type: isdn          CDR Reports: y
  Group Name: Simulated-PSTN              COR: 1        TN: 1        TAC: 108
   Direction: two-way       Outgoing Display? n         Carrier Medium: PRI/BRI
 Dial Access? n               Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n           TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

The following screen shows that the calling number will be sent to the PSTN over this trunk group. This is bolded to highlight the fact that Modular Messaging can still provide the calling number as part of the "envelope information" for a message delivered over the PSTN while the S8300 LSP is active, provided the PSTN delivers the calling number (ANI) for PSTN calls from Site B to Site A. This will be necessary if calls arriving over the PSTN from Site B users to Site A are to appear as "station to station" calls using the optional configuration shown in Section 7.2.10.

```
display trunk-group 8                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                                          Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: n      Send Calling Number: y
            Used for DCS? n                            Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider
       Charge Conversion: 1
           Decimal Point: none                     Replace Restricted Numbers? n
         Currency Symbol: $                       Replace Unavailable Numbers? n
             Charge Type: Dollars                       Send Connected Number: y
Network Call Redirection: none                      Hold/Unhold Notifications? y
             Send UUI IE? y                       Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                              Ds1 Echo Cancellation? n
                                          US NI Delayed Calling Name Update? n

                             Network (Japan) Needs Connect Before Disconnect? n
                                                        Apply Local Ringback? n
```

The following screen shows the first several members of trunk group 8, using the channels on the MM710 in slot 1V4 of the G350 Media Gateway in Site B.

```
display trunk-group 8                                           Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/8
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   8

       Port    Code Sfx Name        Night           Sig Grp
  1: 001V401  MM710                                   8
  2: 001V402  MM710                                   8
  3: 001V403  MM710                                   8
  4: 001V404  MM710                                   8
  ..............................................................................
```

4.12. ISDN-PRI Trunk Group in the CMC at Site C

The CMC in Site C also maintains a trunk to the (simulated) PSTN. The following screens show that signaling group 48 and trunk group 48 define an ISDN-PRI link to the (simulated) PSTN at Site C.

```
display signaling-group 48                                      Page   1 of   5
                                SIGNALING GROUP
 Group Number: 48               Group Type: isdn-pri
                     Associated Signaling? y          Max number of NCA TSC: 0
                        Primary D-Channel: 03A0624     Max number of CA TSC: 0
                                                     Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 48      X-Mobility/Wireless Type: NONE
          Supplementary Service Protocol: a          Network Call Transfer? n
```

Trunk group 48 can be used for inbound trunk calls to Site C users, and outbound PSTN calls from Site C users, per the configured routing. For example, this is the trunk group that will be used for Site C connections to Site A, when the data WAN is down and the S8500 ESS is active. The following screens show the configuration of trunk group 48.

```
display trunk-group 48                                          Page   1 of  21
                                TRUNK GROUP
Group Number: 48                   Group Type: isdn          CDR Reports: y
  Group Name: SIM-PSTN-FOR-CMC            COR: 1        TN: 1        TAC: 148
   Direction: two-way        Outgoing Display? y         Carrier Medium: PRI/BRI
 Dial Access? n                Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

The following screen shows that the calling number will be sent to the PSTN over this trunk group. Modular Messaging can provide the calling number as part of the “envelope information” for a message delivered over the PSTN while the S8500 ESS is actively providing service to Site C, assuming the PSTN delivers the calling number (ANI) from Site C to Site A. This will also be necessary if calls arriving from Site C users to Site A are to appear as “station to station” calls using the optional advanced configuration shown in Section 7.2.10.

```
display trunk-group 48                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                                         Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: n      Send Calling Number: y
            Used for DCS? n                            Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: n
Network Call Redirection: none                    Hold/Unhold Notifications? n
             Send UUI IE? y                     Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                           Ds1 Echo Cancellation? n

                                          US NI Delayed Calling Name Update? n
                             Network (Japan) Needs Connect Before Disconnect? n
                                                        Apply Local Ringback? n
```

The following screen shows the first several members of trunk group 48, using the channels of the DS1 circuit pack installed in slot 3A06 of the CMC Media Gateway in Site C.

```
display trunk-group 48                                          Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/8
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   8

       Port    Code Sfx Name        Night           Sig Grp
  1: 03A0601  TN464  F                                48
  2: 03A0602  TN464  F                                48
  3: 03A0603  TN464  F                                48
  4: 03A0604  TN464  F                                48
  ..............................................................................
```

4.13. Cabinet Configuration for CMC in Site C

The following screen illustrates the cabinet configuration for the CMC in Site C. Devices such as digital telephones and trunks that connect directly to the CMC gateway have location 4 and network region 8 assigned. Although a CMC gateway is used in these Application Notes, the same approach may be used for other gateways such as MCC, SCC, or G650 Media Gateways (i.e., any other IPSI connected port networks with local ESS and local trunks).

```
display cabinet 3
                                    CABINET
 CABINET DESCRIPTION
                  Cabinet: 3
           Cabinet Layout: cmc-carrier-stack
             Cabinet Type: cmc-portnetwork

    Survivable Remote EPN? n
                 Location: 4         IP Network Region: 8

                     Room: by-cmc       Floor:             Building:
 CARRIER DESCRIPTION
   Carrier       Carrier Type        Number

      C        not-used              PN  03
      B        not-used              PN  03
      A        cmc-control           PN  03
      D        not-used              PN  03
      E        not-used              PN  03
```

4.14. IPSI Configuration for CMC in Site C

The following screen illustrates the IPSI configuration for the CMC in Site C.

```
display ipserver-interface 3
          IP SERVER INTERFACE (IPSI) ADMINISTRATION - PORT NETWORK 3

                                IP Control? y            Socket Encryption? y
 Ignore Connectivity in Server Arbitration? n                   Enable QoS? n
 Primary IPSI
 ------------
  Location:  3A02
      Host: 2.2.4.9
   DHCP ID: ipsi-A03a
```

4.15. C-LAN Configuration for CMC in Site C

The following screen illustrates attributes of the C-LAN configuration for the CMC in Site C.

```
display ip-interface 3a10
                                  IP INTERFACES
                  Type: C-LAN
                  Slot: 03A10
           Code/Suffix: TN799  D
             Node Name: Clan-CMC1
            IP Address: 2  .2  .4  .87
           Subnet Mask: 255.255.255.0                           Link: 4
       Gateway Address: 2  .2  .4  .2
  Enable Ethernet Port? y                        Allow H.323 Endpoints? y
        Network Region: 8                         Allow H.248 Gateways? y
                  VLAN: n                          Gatekeeper Priority: 5
```

4.16. MEDPRO Configuration for CMC in Site C

The following screen illustrates attributes of the MEDPRO configuration for the CMC in Site C.

```
display ip-interface 3A05
                                  IP INTERFACES
                  Type: MEDPRO
                  Slot: 03A05
           Code/Suffix: TN2302
             Node Name: Medpro-CMC1
            IP Address: 2  .2  .4  .89
           Subnet Mask: 255.255.255.0
       Gateway Address: 2  .2  .4  .2
  Enable Ethernet Port? y
        Network Region: 8
```

4.17. Communication Manager Configuration for ESS for Site C

The following screens illustrate attributes of the ESS configuration. The configuration is performed using Avaya Communication Manager on the S8700 Media Server in Site A. In this configuration, the S8500 Media Server at Site C is licensed as an ESS, and configured to provide survivable service to Site C users only, should the IPSI in the CMC at Site C lose connectivity to Site A. Customer circumstances may vary with respect to the scope of survivability provided by a particular ESS. In the context of illustrating the survivable messaging feature, it suffices that Site C is isolated from other sites by a WAN failure, causing the ESS in Site C to provide active service to all Site C users and trunks.

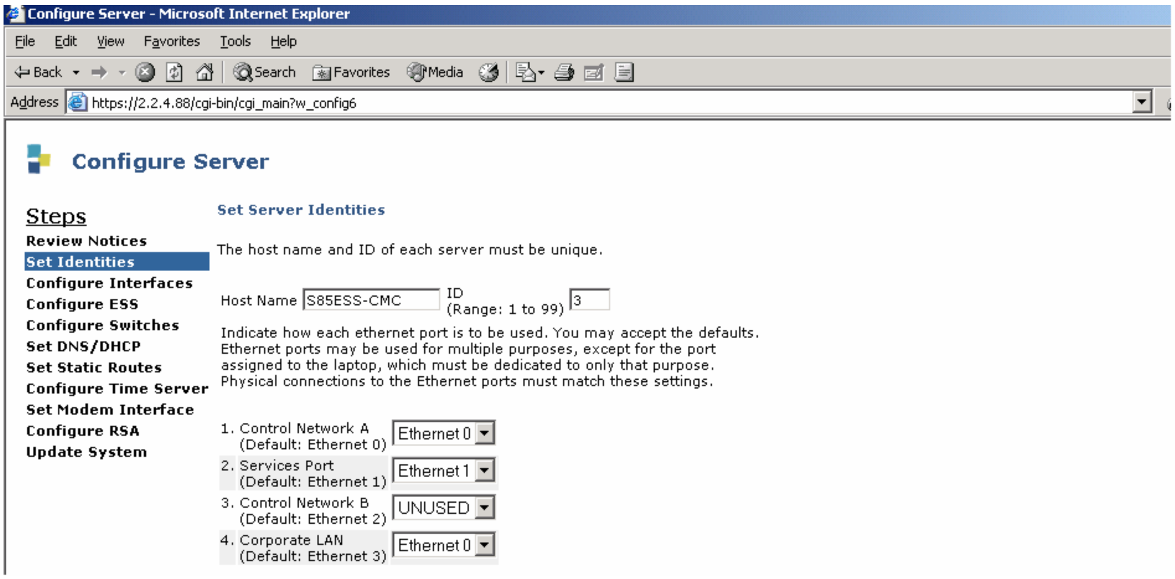
```
display system-parameters ess                                   Page   1 of   7
 ENTERPRISE SURVIVABLE SERVER INFORMATION

 Cl  Plat    Server   A             Server   B             Pri  Com Sys Loc Loc
 ID  Type    ID    Node Name        ID    Node Name        Scr      Prf Prf Only
 -------------------------------------------------------------------------------
                                 MAIN SERVERS
 1   Duplex  1     1  .1  .15 .11   2     1  .1  .15 .12
                        ENTERPRISE SURVIVABLE SERVERS

 2   Simplex 3     ESSCid002Sid003                        1    3   n   n   y
```

The following screen illustrates the ability to assign port networks to a community, providing a means to control the scope of an ESS. In this configuration, the CMC in Site C is port network 3, and the S8500 ESS in Site C has been configured for local control of community 3.

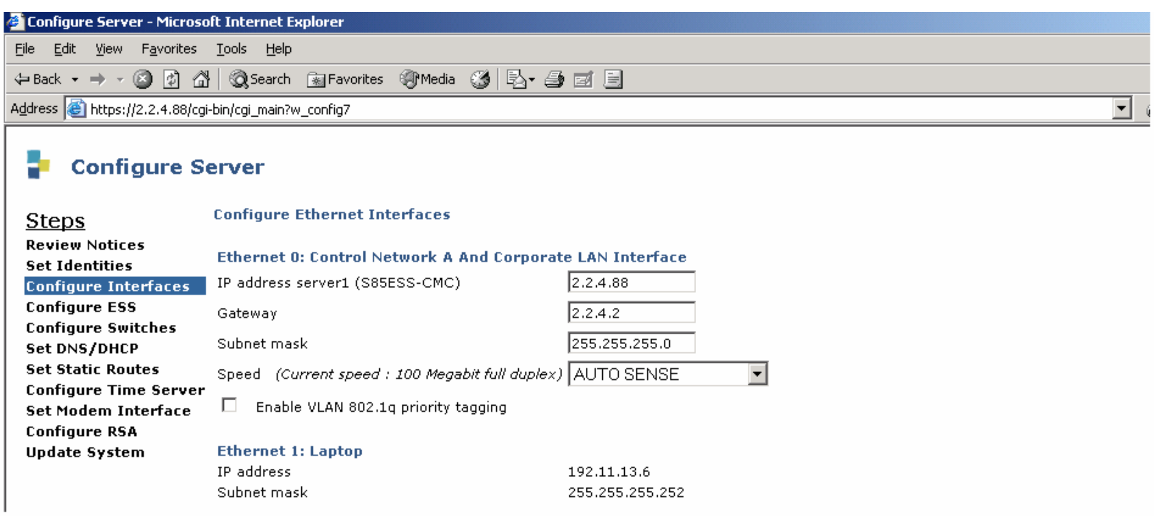
```
display system-parameters ess                                   Page   6 of   7
              COMMUNITY ASSIGNMENTS FOR PORT NETWORKS
 PN Community      PN Community      PN Community      PN Community     PN Community
 ------------      ------------      ------------      ------------     ------------
  1: 1             14: 1             27: 1             40: 1            53: 1
  2: 1             15: 1             28: 1             41: 1            54: 1
  3: 3             16: 1             29: 1             42: 1            55: 1
  4: 1             17: 1             30: 1             43: 1            56: 1
  5: 1             18: 1             31: 1             44: 1            57: 1
  6: 1             19: 1             32: 1             45: 1            58: 1
  7: 1             20: 1             33: 1             46: 1            59: 1
  8: 1             21: 1             34: 1             47: 1            60: 1
  9: 1             22: 1             35: 1             48: 1            61: 1
 10: 1             23: 1             36: 1             49: 1            62: 1
 11: 1             24: 1             37: 1             50: 1            63: 1
 12: 1             25: 1             38: 1             51: 1            64: 1
 13: 1             26: 1             39: 1             52: 1
```

4.18. S8500 Media Server Web Configuration

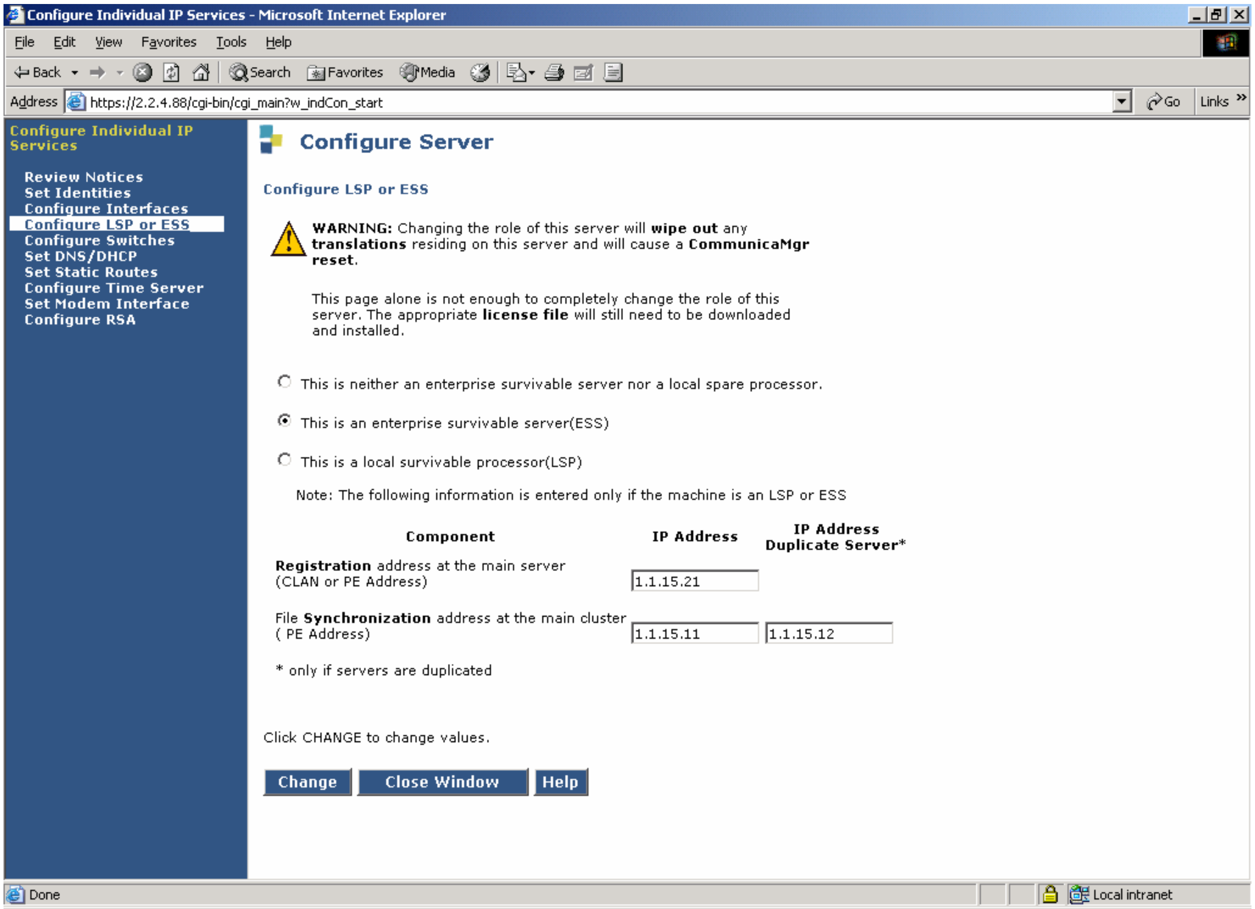
The following screens illustrate attributes of the web configuration for the Site C S8500 Media Server. The server "ID" 3 is unique (i.e., different from server IDs 1 and 2 assigned to the S8700 Media Servers at Site A). The server ID shown below also matches the "Server ID" field for cluster 2 in the "system-parameters ess" form shown in the previous section.



The following screen illustrates the IP address configuration, which again matches the IP address assigned to the node name for cluster 2 in the "system-parameters ess" screen shown in the previous section.



In the following screen, note that the S8500 is configured as an ESS, with the address of a C-LAN at Site A, and the customer network interface addresses of the S8700 Media Servers at Site A.



5. Avaya Communication Manager Configuration For Survivable Messaging and Dialing Transparency

This section and the associated verifications in Section 7 are the principal areas of divergence from References [1] and [2]. In Reference [1], the corresponding Section 5 illustrates the configuration for Site B, enabling survivable messaging when the Local Survivable Processor is active. While very similar, the focus here is on the configuration enabling survivable messaging and dialing transparency for Site C when the Enterprise Survivable Server is active. The following paragraphs summarize the expected behavior resulting from the configuration.

Under normal conditions, the IPSI in the CMC Media Gateway in Site C is controlled over the data WAN by Avaya Communication Manager at Site A. The S8700 Media Server processes calls on behalf of users at all sites. If a call is made from a user at Site A to a user at Site C, and the call is redirected to voice mail, an intra-site connection is made between the Site A originator and the Modular Messaging MAS in Site A. If a call is made from a user at Site C to another user at Site C, and the call is redirected to voice mail, an inter-site VoIP connection is made over the data WAN to Site A, enabling the call originator connected at Site C to interact with Modular Messaging at Site A. The S8500 ESS maintains an automatically synchronized copy of the Avaya Communication Manager translations centrally managed by the Site A primary cluster.

If a persistent data WAN outage occurs at Site C, the IPSI in the CMC Media Gateway at Site C will be unable to communicate with Site A. The S8500 Media Server ESS in Site C is capable of providing service under such conditions. The IPSI in the CMC Media Gateway automatically "reaches up" for service from the S8500 ESS.

When the data WAN to Site C is down, and the S8500 ESS is actively providing service, Site C access to other sites including Site A must be via PSTN trunks in the Site C CMC Media Gateway. The CMC Media Gateway has an ISDN-PRI to the (simulated) PSTN. If a call is made from a user at Site C to another user at Site C, and this call is redirected to voice mail, an inter-site PSTN connection is made using the ISDN-PRI trunks in the CMC Media Gateway, enabling the call originator to interact with Modular Messaging at Site A. With the data WAN down, the supplementary information about a call that enables Modular Messaging to identify the correct subscriber mailbox under normal operation is not available. Therefore, to retain the normal caller experience, reaching the proper subscriber mailbox for a Site C user, even when the WAN is down and the ESS is processing calls, an alternate means must be established. This alternate means is the subject of the following sub-sections.

If a Site C user dials the extension of a Site A user when the S8500 ESS is active, the Site A user must be reached over the PSTN rather than via an inter-site VoIP connection. The configuration enabling voice mail service and dialing transparency uses enhancements to remote call coverage.

5.1. Coverage Path For Site A Subscriber Extensions with DID

The following screens illustrate the configuration for coverage path 34, which will be assigned to Site A users who can be reached via Direct Inward Dialing (DID). That is, these users can be reached directly via a PSTN number. Coverage path 34 includes hunt group 34 as the first coverage point, and a remote coverage entry "r34" as the second coverage point. The remote coverage entry will be configured such that a call from a Site C user can automatically route to the PSTN number of the called user assigned this coverage path, when the S8500 ESS at Site C is active. That is, when the S8500 ESS is active, remote coverage entry "r34" enables "dialing transparency" for Site C users that dial the extension of Site A users that have DID numbers. In a configuration with additional branches whose DID numbers also arrive at Site A trunks, this same approach can be used to allow dialing transparency from Site C users to other branch sites under control of the primary server at Site A.

```
change coverage path 34
                                 COVERAGE PATH
                   Coverage Path Number: 34
                                                  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: 3
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n

  Point1: h34       Rng:    Point2: r34       Rng:    Point3:
  Point4:                   Point5:                   Point6:
```

Many Site A users with DID can be assigned this same coverage path to simplify administration. In **Figure 1**, the Avaya IP Telephone user with extension 57041 and fictitious PSTN DID number 732-555-7041 will be among users with coverage path 34 assigned, as illustrated in the following screen.

```
display station 57041                                          Page   1 of   4
                                     STATION
Extension: 57041                           Lock Messages? n         BCC: 0
     Type: 4620                            Security Code: 1234       TN: 1
     Port: S00008                        Coverage Path 1: 34        COR: 1
     Name: Mr. Midspan                   Coverage Path 2:           COS: 1
                                         Hunt-to Station:
STATION OPTIONS
              Loss Group: 19             Personalized Ringing Pattern: 1
                                                     Message Lamp Ext: 57041
            Speakerphone: 2-way                   Mute Button Enabled? y
        Display Language: english                    Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal                  Media Complex Ext:
   Survivable Trunk Dest? y                              IP SoftPhone? y
                                                   IP Video Softphone? n
```

The following screen illustrates Page 2 for station 57041, highlighting the use of “qsig-mwi” for QSIG-based interrogation and auditing of the user’s message waiting indicator status.

```
change station 57041                                            Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                    Idle Appearance Preference? n
 Per Button Ring Control? y                  Bridged Idle Line Preference? n
   Bridged Call Alerting? n                      Restrict Last Appearance? n
  Active Station Ringing: single         Conf/Trans on Primary Appearance? n
                                                       EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced                 Audible Message Waiting? n
    MWI Served User Type: qsig-mwi              Display Client Redirection? n
                                               Select Last Used Appearance? n
             IP Hoteling? n                      Coverage After Forwarding? y

 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
 Emergency Location Ext: 57041     Always Use? n      IP Audio Hairpinning? y
```

5.2. Coverage Path For Site A Subscriber Extensions without DID

The following screens illustrate the configuration for coverage path 35, which will be assigned to Site A users who cannot be reached via Direct Inward Dialing (DID). That is, these users *cannot* be reached directly via a PSTN number. Like coverage path 34, coverage path 35 includes hunt group 34 as the first coverage point, but a different remote coverage entry "r35" is used as the second coverage point. The remote coverage entry "r35" will be configured such that a call from a Site C user can automatically route to the PSTN number of an Avaya Communication Manager Vector Directory Number (VDN). When the S8500 ESS at Site C is active, the call can automatically be routed to the Site A PSTN number of the VDN. The Site C system will wait for the Site A vector to answer, pause, and then send the extension of the called user via end-to-end signaling. At the main site, the receiving VDN is associated with a call vector that collects the digits and routes the call to the called user. In sum, when the S8500 ESS at Site C is active, remote coverage entry "r35" will enable "dialing transparency" for calls dialed by extension from Site C users to Site A users who do not have DID numbers.

```
change coverage path 35                                         Page   1 of   1
                                 COVERAGE PATH
                   Coverage Path Number: 35
                                                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
   Holiday Coverage?             n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n

  Point1: h34       Rng:    Point2: r35       Rng:    Point3:
  Point4:                   Point5:                   Point6:
```

Many Site A users without DID can be assigned this same coverage path to simplify administration. In **Figure 1**, the user with extension 57021 will be among the users with coverage path 35 assigned, as illustrated in the following screen.

```
change station 57021                                          Page   1 of   4
                                     STATION
Extension: 57021                           Lock Messages? n          BCC: 0
     Type: 6408D+                          Security Code: 1234        TN: 1
     Port: 01B1701                       Coverage Path 1: 35         COR: 1
     Name: Don Peterson                  Coverage Path 2:            COS: 1
                                         Hunt-to Station:
STATION OPTIONS
              Loss Group: 2              Personalized Ringing Pattern: 1
             Data Module? n                          Message Lamp Ext: 57021
            Speakerphone: 2-way                   Mute Button Enabled? y
        Display Language: english
                                                    Media Complex Ext:
                                                         IP SoftPhone? y
                                                   IP Video Softphone? n
```

Page 2 for station 57021 is similar to Page 2 for station 57041, shown in Section 5.1.

5.3. Coverage Path For Site C Users

The following screens illustrate the configuration for coverage path 36, which will be assigned to Site C users. Coverage path 36 also includes hunt group 34 as the first coverage point, and a remote coverage entry “r36” as the second coverage point. The remote coverage entry will be configured such that a call to a Site C user can automatically route to the PSTN number of the Modular Messaging system at the main site when the S8500 ESS at Site C is active. The S8500 ESS will wait for answer, and automatically send the end-to-end digits representing the called user’s mailbox number (i.e., extension) to the Modular Messaging system. In sum, remote coverage entry “r36” will enable calls to Site C users to reach the proper user’s mailbox, without requiring the calling user to re-enter information, and without requiring an additional DID number for each such user, as was the case in reference [2].

```
change coverage path 36                                       Page   1 of   1
                                 COVERAGE PATH
                   Coverage Path Number: 36
                                                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: 2
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n

  Point1: h34       Rng:    Point2: r36       Rng:    Point3:
  Point4:                   Point5:                   Point6:
```

Many Site C users can be assigned this same coverage path to simplify administration. Indeed, depending on the remote coverage configuration, it may be possible to assign this same coverage path to users at other sites that require messaging service when a survivable processor is active. In these Application Notes, and the companion reference [1], users at both Site B (LSP) and Site C (ESS) may share this same coverage path. In **Figure 1**, the telephone user with extension 51021 is among the users with coverage path 36 assigned, as illustrated in the following screen.

```
change station 51021                                          Page   1 of   4
                                     STATION
Extension: 51021                           Lock Messages? n         BCC: 0
     Type: 6408D+                          Security Code: 1234       TN: 1
     Port: 03A0801                       Coverage Path 1: 36        COR: 1
     Name: Charles Modcab                Coverage Path 2:           COS: 1
                                         Hunt-to Station:
STATION OPTIONS
              Loss Group: 2              Personalized Ringing Pattern: 1
             Data Module? n                          Message Lamp Ext: 51021
            Speakerphone: 2-way                   Mute Button Enabled? y
        Display Language: english
                                                    Media Complex Ext:
                                                         IP SoftPhone? y
                                                   IP Video Softphone? n
```

Page 2 for station 51021 is similar to Page 2 for station 57041, shown in Section 5.1.

5.4. Remote Coverage Entries

The “Centralized Messaging Coverage for Survivable Servers” feature enhances the coverage remote form by allowing special characters such as “L”, “%”, “,” and “D” to be programmed. The letter “L” at the beginning of a coverage remote entry can be used to ensure that only an S8300 LSP (Release 3.0.1) or either an S8300 LSP or S8500 ESS (Release 3.1) will use the entry. A primary server (e.g., Site A) will consider an entry beginning with “L” as unavailable. The letter “%” signifies “wait for answer”. The character “,” instructs Avaya Communication Manager to pause, which can be useful after a “wait for answer” to help ensure the far-end is prepared to receive subsequent digits. The letter “D” denotes the called party’s extension. The use of the letter “D” allows the same coverage path and coverage remote entry to be used by many users sharing common characteristics. The following screen illustrates the use of these characters.

```
change coverage remote 1                                          Page   1 of  23
                       REMOTE CALL COVERAGE TABLE
                       ENTRIES FROM 1    TO 1000
01: 50069                 16:                       31:
02:                       17:                       32:
03:                       18:                       33: 97325557555
04:                       19:                       34: L825D
05: 55020                 20:                       35: L77556%,D
06:                       21:                       36: L77555%D
07:                       22:                       37:
08:                       23:                       38:
09:                       24:                       39:
10:                       25:                       40: 97325557551
11:                       26:                       41: L91234567890%,D
12:                       27:                       42: L97325551234%D#
13:                       28:                       43:
14:                       29:                       44:
15:                       30:                       45:
```

The entry for "r34" is used as the second point in coverage path 34, which is assigned to users at Site A that have DID numbers. The entry for "r34" is "L825D", interpreted as follows. The letter "L" ensures that the main server will not use this entry (i.e., in this configuration, the entries "r34", "r35", and "r36" all begin with "L" and will only be used when the S8300 LSP or S8500 ESS is actively processing calls). The number "8" is the AAR access code. The digits 25 together with the letter "D" result in a number beginning with 25 that varies based on the number dialed. In this configuration, extensions of the form 5xxxx may use this coverage remote entry. If a call is made from a Site C user currently served by the S8500 ESS to a Site A user assigned coverage path 34, the S8500 ESS will consider the Site A user unavailable, and the call will enter coverage processing. The first point in coverage is the hunt group to Modular Messaging, which is also unavailable from the perspective of the S8500 ESS (i.e., always the case in this example configuration). The second point in coverage is coverage remote entry "r34" which the ESS will route out trunk group 48 to the PSTN DID number of the user at Site A, based on the routing of the number 255XXXX through AAR. This gives the Site C caller in ESS mode "dialing transparency" for calls to Site A DID users. That is, the Site C user can call the Site A user in the same fashion whether the primary server or ESS is active. If the Site A user does not answer, the call will follow coverage at the main site as well. In general, the Modular Messaging system will be available to the main site, and therefore the call can reach the Site A user's mailbox. If the Modular Messaging system was unavailable, note that the "L" prevents the main site from considering the "r34" entry, when the main site is processing an unanswered incoming call to a Site A user assigned coverage path 34.

The entry for "r36" is used as the second point in coverage for coverage path 36, which is assigned to users at Site C (and Site B as shown in reference [1]). The entry "r36" contains "L77555%D". As before, the letter "L" is used to ensure that the entry is only used when a survivable processor is active. The number "77555" is an arbitrarily chosen unused extension valid within the dial plan. This number will be programmed in the Uniform Dial Plan table so that routing directs the call to the Site A PSTN number of the Modular Messaging hunt group. The "%" character means "Wait for Answer". The "D" after the "%" character causes the S8500 ESS (or S8300 LSP as shown in [1]) to "send the called party number" as end-to-end digits (e.g., DTMF). The net result is that calls that use the "r36" coverage remote entry will be directed to

Modular Messaging, and Modular Messaging will automatically receive the digits associated with the called user's mailbox. Therefore, Modular Messaging provides the appropriate greeting to leave a message in the mailbox of the called user, without requiring additional input from the caller.

The entry for "r35" is used as the second point in coverage for coverage path 35, which is assigned to users at Site A that do not have DID numbers. The entry "r35" contains "L77556%,D". As before, the letter "L" is used to ensure that the entry is only used when a survivable processor is active. The number "77556" is an arbitrarily chosen unused extension valid within the dial plan. This number will be programmed in the Uniform Dial Plan table so that routing directs the call to the Site A PSTN number of a Vector Directory Number (VDN) with extension 57556. The "%" character will cause the S8500 ESS (or S8300 LSP) to "wait for the vector to answer". The "," character is a "pause". In this configuration, the pause is inserted to ensure that the vector at the main site is ready to receive the digits representing the called party's extension. As before, the "D" after the "%" character causes the S8500 ESS (or S8300 LSP) to "send the called party number". The net result is that calls that use the "r36" coverage remote entry will be redirected to the main site VDN. The vector associated with the VDN will collect the called party's extension and route the call to the called party automatically. The called party will have the opportunity to answer the call. If the called party does not answer, the call is redirected to the called user's mailbox on Modular Messaging.

Although unused by call processing in these Application Notes, the entries for "r41" and "r42" in the preceding screen have been shown in bold to illustrate additional enhancements to Avaya Communication Manager Release 3.1 as compared with Release 3.0.1 (used in Reference [1]). In Release 3.0.1, long entries with special characters such as those shown in "r41" and "r42" would have been prohibited. In Release 3.1, the effective length allowed for entries that use special characters has been increased, thus allowing longer strings such as those shown. The entry "r42" illustrates another enhancement. In Release 3.0.1, there were restrictions imposed upon the use of the "#" special character that have been relaxed in Release 3.1. For example, "r42" shows the "#" character being sent as an end-to-end digit, which may be a valuable enhancement depending on the needs of the receiving system (e.g., an alternate messaging telephony user interface, or a system requiring an end-of-dialing indicator to expedite routing).

5.5. Routing the UDP and AAR Numbers in Coverage Remote Form

The prior section used arbitrary UDP and AAR numbers in the coverage remote form to facilitate routing. Although ARS numbers can be used directly in the coverage remote form, UDP and AAR were used to illustrate alternative approaches. This section illustrates the routing associated with the UDP and AAR numbers. For the UDP numbers in the following screen, the bold rows match the patterns 77555 and 77556, delete the first digit (7), and insert the digits 555, passing the resultant number to ARS for processing.

```
change uniform-dialplan 7                                       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
  77555      5   1   555   ars  n                                        n
  77556      5   1   555   ars  n                                        n
```

The following screen shows an ARS entry for handling calls to the fictitious number 555-7555 resulting from the digit manipulation from the previous screen. In the example configuration, the route pattern chosen will depend on location attributes. For example, route pattern 555 will be chosen in location 3 (Site B) based on the configuration of the screen below.

```
change ars analysis 5557555 location 3                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  3           Percent Full:    2
          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    5557555                7    7     555      natl         n
```

The following screen shows that route pattern 555 includes trunk group 8, the ISDN-PRI trunk group in the G350 Media Gateway in Site B. In this case, the digits 732 are inserted so that the number 732-555-7555 is sent to the (simulated) PSTN.

```
change route-pattern 555                                        Page   1 of   3
                    Pattern Number: 555 Pattern Name: To-Main-LSP-Loc3
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 8    0                        732                                n   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 PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                       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
```

The following screen shows another location-specific ARS entry for handling calls to the same fictitious number 555-7555. Route pattern 575 will be chosen in location 4 (Site C) based on the following configuration.

```
change ars analysis 5557555 location 4                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  4             Percent Full:    2
          Dialed            Total      Route     Call   Node  ANI
          String           Min  Max  Pattern    Type   Num   Reqd
    5557555                7    7     575       natl         n
```

An alternative approach that does not use location based routing is illustrated below for routing calls to the fictitious number 555-7556. In this case, the location parameter is omitted, and the global ARS table directs the call to route pattern 575.

```
change ars analysis 5557556                                     Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all           Percent Full:    2
          Dialed            Total      Route     Call   Node  ANI
          String           Min  Max  Pattern    Type   Num   Reqd
    5557556                7    7     575       natl         n
```

The following screen shows that route pattern 575 includes trunk group 48, the ISDN-PRI trunk group in the CMC Media Gateway in Site C, and trunk group 8, the ISDN-PRI in the G350 Media Gateway in Site B. The digits 732 are inserted so that an appropriate number is sent to the PSTN. While customer circumstances may vary, the available methods of routing and digit manipulation allow flexibility to accommodate the enterprise dial plan and PSTN requirements. At the end of this section, additional considerations are presented in the context of route pattern 255, which shares similar characteristics.

```
change route-pattern 575                                        Page   1 of   3
                    Pattern Number: 575 Pattern Name: Cover-to-PSTN
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 48   0                        732                                    n   user
 2: 8    0                        732                                    n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                           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
```

The following screen shows the routing associated with coverage remote entry “r34” to the AAR number “L825D” where the dialed number represented by “D” is of the form 5XXXX.

```
change aar analysis 255                                         Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    2
          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    255                   7    7     255      aar          n
```

The following screen illustrates route pattern 255. The digit manipulation deletes the arbitrary AAR number 255, and inserts 732-555, the fictitious PSTN digit pattern used in the sample configuration. For example, if extension 57041 is dialed, the number 732-555-7041 will be sent to the PSTN. Note that this configuration is similar to the configuration used in reference [1] where Site B was the only survivable site, except that trunk group 48 (in the CMC in Site C) has been inserted as the first routing pattern preference. With this configuration, if the S8500 ESS is active due to a WAN failure to Site C, then trunk group 48 will be in-service and trunk group 8 (in the G350 in Site B) will be out of service (i.e., since Site C is isolated by the WAN failure). Therefore, trunk group 48, local to Site C, will be chosen for Site C callers using this route pattern. If the S8300 LSP is active due to a WAN failure to Site B, then trunk group 48 will be out of service, but trunk group 8 will be in-service. In this case, trunk group 8, local to Site B, will be chosen for Site B callers using this route pattern. Depending on the configuration of the media gateway controller list of the G350 Media Gateway in Site B and the type of failure, it may be possible for the G350 Media Gateway to be registered to the S8500 ESS rather than the S8300 LSP in other failure scenarios. For example, if the media gateway controller list for the G350 in Site B included C-LANs in the CMC at Site C, and there was a significant failure at Site A rendering the S8700 Media Servers unusable (i.e., rather than a WAN outage isolating Site C), the G350 could register with the S8500 ESS through a C-LAN at Site C. In this case, absent other configuration, a call from a user at Site B that uses route-pattern 255 (e.g., a call to a user at Site A) could use the trunk in the CMC media gateway in Site C. Customer circumstances and objectives will vary, and this document is intended to be illustrative rather than prescriptive. The foregoing discussion has been included to illustrate some of the considerations that warrant planning.

```
change route-pattern 255                                        Page   1 of   3
                    Pattern Number: 255 Pattern Name: To-Main-DID
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                               DCS/ IXC
    No          Mrk Lmt List Del  Digits                                 QSIG
                             Dgts                                        Intw
 1: 48   0                    3   732555                                  n   user
 2: 8    0                    3   732555                                  n   user
 3:                                                                       n   user
 4:                                                                       n   user
 5:                                                                       n   user
 6:                                                                       n   user

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

5.6. Vector Directory Number and Vector for Site A Non-DID Users

The following screen illustrates a sample Vector Directory Number 57556 reachable via fictitious PSTN number 732-555-7556. An active S8500 ESS (or S8300 LSP) can automatically call this VDN when a Site C (or Site B) user dials the extension of a Site A user that has been assigned coverage path 35 (i.e., the coverage path for non-DID users at Site A). This VDN is associated with vector 56.

```
add vdn 57556                                                 Page   1 of   2
                            VECTOR DIRECTORY NUMBER
                             Extension: 57556
                                  Name: For non-DID xparency
                         Vector Number: 56
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                    TN: 1
                              Measured: none
```

The following screen illustrates vector 56. Step 1 collects the 5-digit extension expected from the active survivable processor. Step 2 routes the call to the extension represented by the collected digits. If there is no answer, coverage is allowed so that Modular Messaging can answer with the called user's greeting. Step 3 is included so that the call can still be routed to Modular Messaging, even if the collected digits do not represent a routable number. Under normal conditions, step 3 will not be executed.

```
change vector 56                                                 Page   1 of   3
                                  CALL VECTOR

    Number: 56               Name: Non-DIDxparency
                   Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? n   G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n
 Variables? n   3.0 Enhanced? n
01 collect      5    digits after announcement none
02 route-to     digits with coverage y
03 route-to     number 57555            with cov n if unconditionally
04 stop
.................................................................
```

5.7. Location-Based Routing for Direct-dialed ARS Calls

In general, location-based routing can be used to cause Avaya Communication Manager to consider the location of the call originator in routing decisions. Some sites, with limited data WAN bandwidth, may prefer that all calls to a public network telephone number use a PSTN trunk in the gateway serving the call originator. Other sites linked by a robust data WAN may prefer that calls proceed as VoIP calls over the data WAN to "hop-off" at the site where the PSTN leg of the call is least expensive. Location-based routing allows routing decisions to consider the location as well as the dialed number. This allows the same telephone number dialed directly by users at different locations to be routed differently.

In the context of these Application Notes, routing is configured such that direct dialed ARS calls originating from Site B (location 3) users or trunks to the DID numbers at Site A will route out the ISDN-PRI trunk (trunk group 8) in the G350 Media Gateway at Site B. Direct dialed ARS calls originating from Site C (location 4) users or trunks to the DID numbers at Site A will route out the ISDN-PRI trunk (trunk group 48) in the CMC Media Gateway at Site C. The two entries in the following screen show that calls from Site B (location 3) users or trunks to 732-555-75XX or 732-555-70XX will route to route pattern 36. In this case, the configured ranges include the PSTN DID number 732-555-7555 for the main Modular Messaging hunt group, as well as DID numbers such as 732-555-7041 for telephone users at Site A.

```
change ars analysis 7 location 3                                Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  3            Percent Full:    2
          Dialed            Total     Route    Call   Node  ANI
          String           Min  Max  Pattern   Type   Num   Reqd
    73255570               10   10    36       natl         n
    73255575               10   10    36       natl         n
```

The two entries in the following screen show that calls from Site C (location 4) users or trunks to the same 732-555-75XX or 732-555-70XX ranges will route to route pattern 46.

```
change ars analysis 7 location 4                                Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  4            Percent Full:    2
          Dialed            Total     Route    Call   Node  ANI
          String           Min  Max  Pattern   Type   Num   Reqd
    73255570               10   10    46       natl         n
    73255575               10   10    46       natl         n
```

The following screen shows that route pattern 36 includes trunk group 8, the ISDN-PRI trunk group in the G350 Media Gateway in Site B. In this example, no digit deletion and insertion was necessary. Customer circumstances may vary.

```
change route-pattern 36                                         Page   1 of   3
                    Pattern Number: 36  Pattern Name: G350-SIM-PSTN
                                          Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 8    0                                                             n   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  n            rest                                        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
```

The following screen shows that route pattern 46 includes trunk group 48, the ISDN-PRI trunk group in the CMC Media Gateway in Site C.

```
change route-pattern 46                                         Page   1 of   3
                    Pattern Number: 46  Pattern Name: To-PSTN-out-CMC
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 48   0                                                           n   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 PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                                       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
```

5.8. Dialing Transparency for Calls to Modular Messaging (extension)

This section includes the changes associated with dialing transparency for calls to the Modular Messaging hunt group extension, x57555. In Section 4.3, hunt group 34 has been assigned coverage path 575. The following screen illustrates coverage path 575, which directs calls to remove coverage entry “r75”.

```
change coverage path 575                                        Page   1 of   1
                                COVERAGE PATH
                   Coverage Path Number: 575
                                                 Hunt after Coverage? y
                       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: 2
                All?             n             n
 DND/SAC/Goto Cover?             y             y
   Holiday Coverage?             n             n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? y

  Point1: r75       Rng:    Point2:                   Point3:
  Point4:                   Point5:                   Point6:
```

Remote coverage entry “r75” is configured with the fictitious PSTN number of the Modular Messaging hunt group, preceded by the ARS access code “9”, as shown in the screen below.

```
change coverage remote 1                                        Page   2 of  23
                       REMOTE CALL COVERAGE TABLE
                       ENTRIES FROM 1    TO 1000
46:                       61:                       76:
47:                       62:                       77:
48:                       63:                       78:
49:                       64:                       79:
50:                       65:                       80:
51:                       66:                       81:
52:                       67:                       82:
53:                       68:                       83:
54:                       69:                       84:
55:                       70:                       85:
56:                       71:                       86:
57:                       72:                       87:
58:                       73:                       88:
59:                       74:                       89:
60:                       75: L97325557555          90:
```

The ARS analysis of the number directs the call to Route Pattern 33, as shown below.

```
change ars analysis 7325557                                     Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all           Percent Full:    2
          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    7325557               10   10    33       natl         n
```

Route-pattern 33 is configured with trunk group 48 in the CMC at Site C as the first choice, and trunk group 8 in the G350 at Site B as the second choice, for the same reasons described for route-pattern 255 in Section 5.5.

```
change route-pattern 33                                         Page   1 of   3
                    Pattern Number: 33  Pattern Name: VM-To-Sim-PSTN
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                                DCS/ IXC
    No          Mrk Lmt List Del  Digits                                  QSIG
                             Dgts                                         Intw
 1: 48   0                                                                 n   user
 2: 8    0                                                                 n   user
 3:                                                                        n   user
 4:                                                                        n   user
 5:                                                                        n   user
 6:                                                                        n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                            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
```

5.9. Saving Configuration Changes

The command “save translation all” can be used to save configuration changes to all Avaya Communication Manager servers in the configuration, including the S8300 Local Survivable Processor and S8500 Enterprise Survivable Server. To have translations automatically sent to the S8300 LSP and S8500 ESS as part of a regularly scheduled translation save, ensure the bolded field is set to “y” in the following screen.

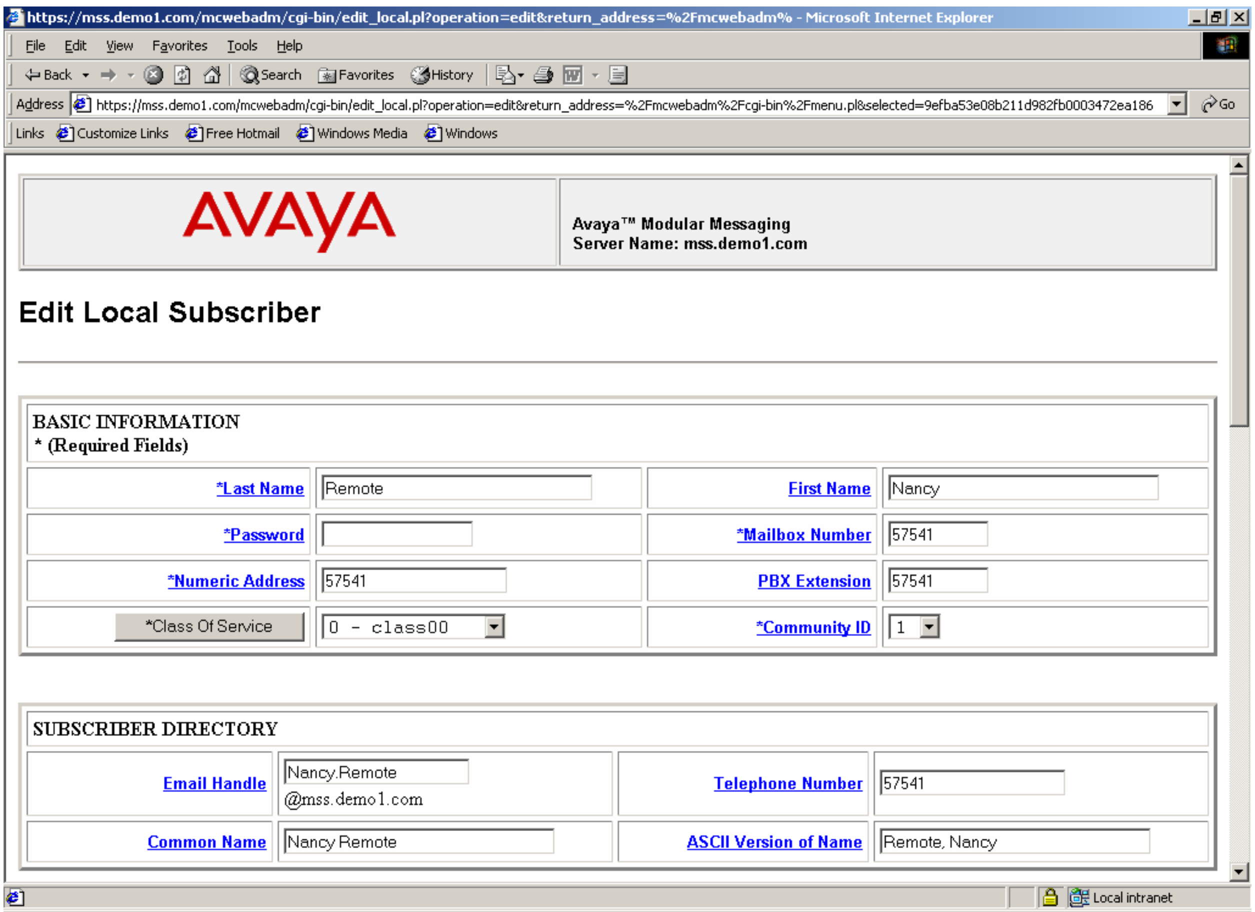
```
display system-parameters maintenance                          Page   1 of   3
                   MAINTENANCE-RELATED SYSTEM PARAMETERS
OPERATIONS SUPPORT PARAMETERS
     CPE Alarm Activation Level: none
SCHEDULED MAINTENANCE
                                          Start Time: 22 : 00
                                           Stop Time: 00 : 00
                                    Save Translation: daily
 Update LSP and ESS Servers When Saving Translations: y
                          Command Time-out (minutes): 120
                         Control Channel Interchange: no
                      System Clocks/IPSI Interchange: no
```

6. Modular Messaging Configuration

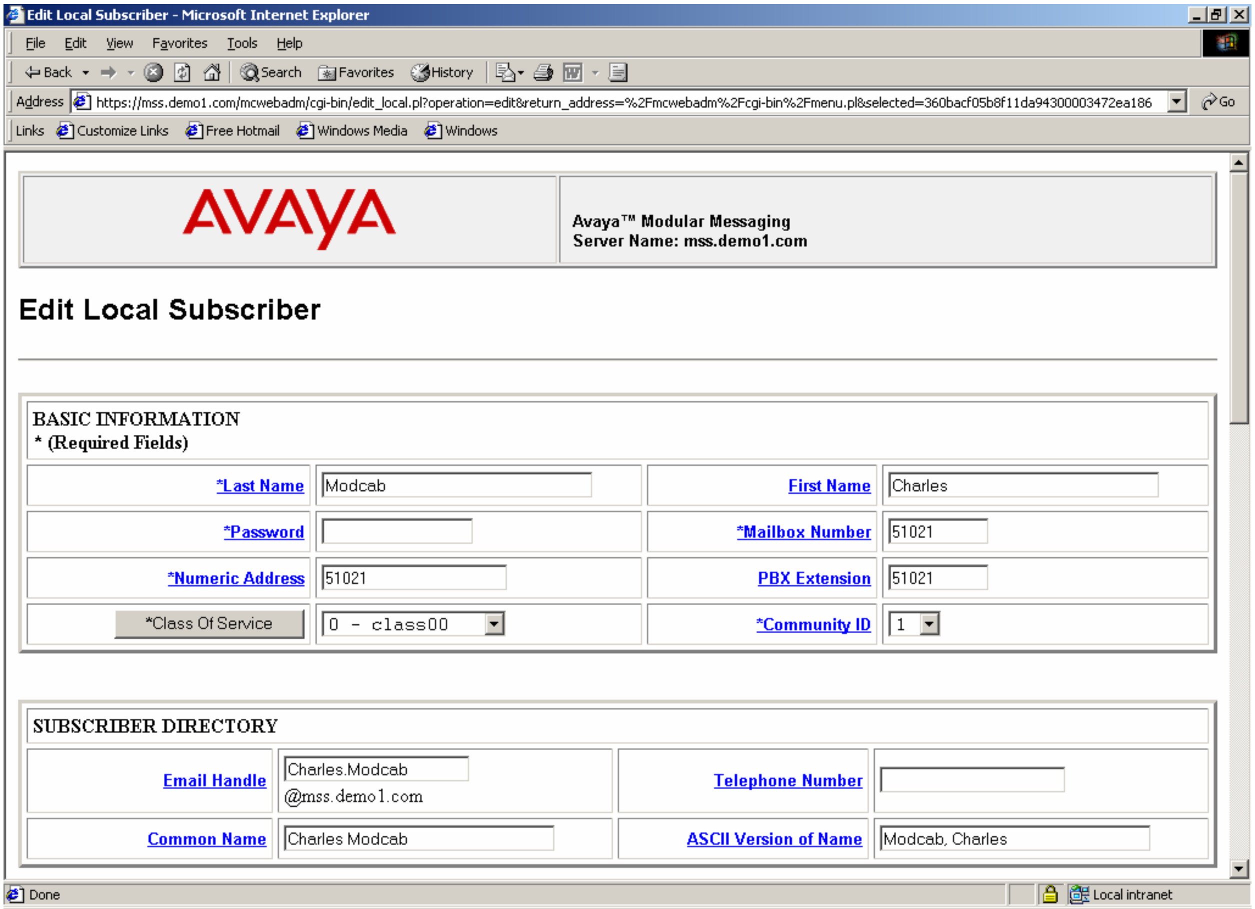
As previously noted, the basic Modular Messaging configuration for ISDN/QSIG integration is covered in detail in other configuration notes. Please consult reference [4]. Although not the focus of these Application Notes, Section 6.2 includes a few pertinent screens illustrating the basic configuration used in verifying these Application Notes.

6.1. Example Modular Messaging Configuration for Site B and Site C Users

The following screen illustrates the configuration for the Site B user with extension 57541 and name “Nancy Remote”. The configuration is performed on the Avaya Message Storage Server (MSS) via a web browser. Log in to the MSS. Click the link “Global Administration”. On the screen presented, click the link “Subscriber Management”. Enter the subscriber mailbox “57541” and click “Edit”. The following screen illustrates basic information for a previously configured subscriber mailbox. Throughout this configuration, it is assumed that the Modular Messaging mailbox is the same as the extension number within Avaya Communication Manager.



Similarly, the following screen illustrates the configuration for the Site C user with extension 51021 and name "Charles Modcab".



6.2. General Modular Messaging Configuration

This section includes basic Modular Messaging configuration screens for reference. From the Modular Messaging MAS, select **Start → Programs → Avaya Modular Messaging → Voice Mail System Configuration**. **Figure 2** illustrates the configurable entities.

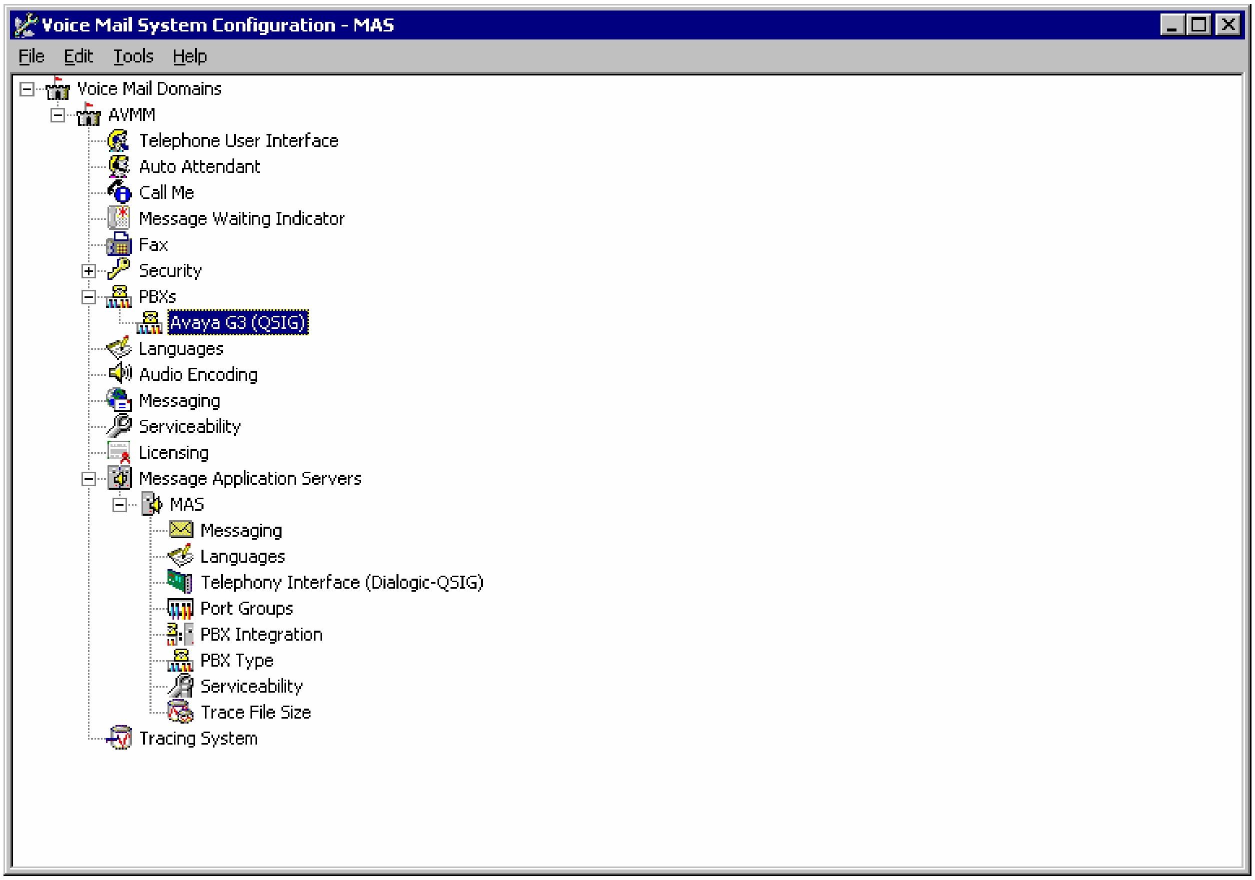
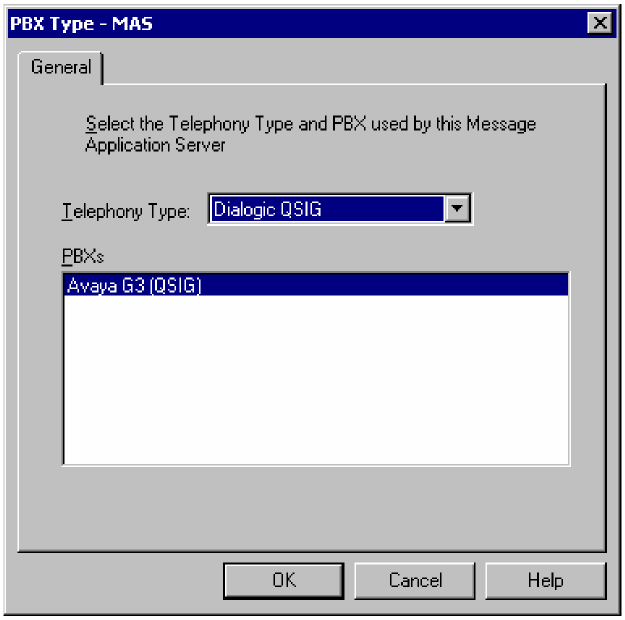
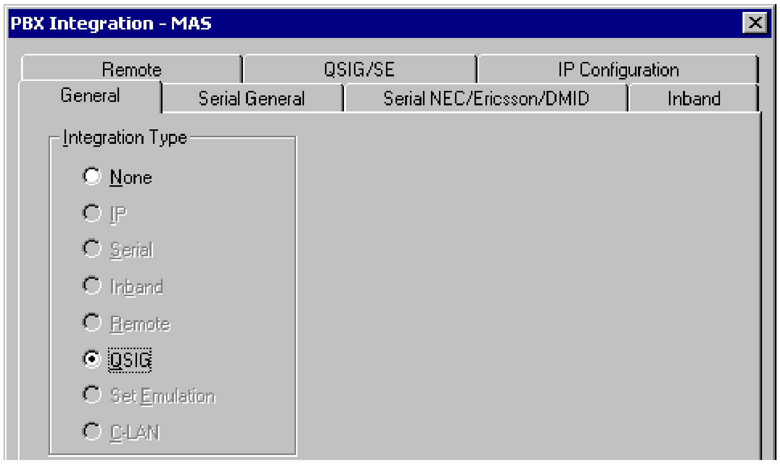


Figure 2: MAS System Configuration

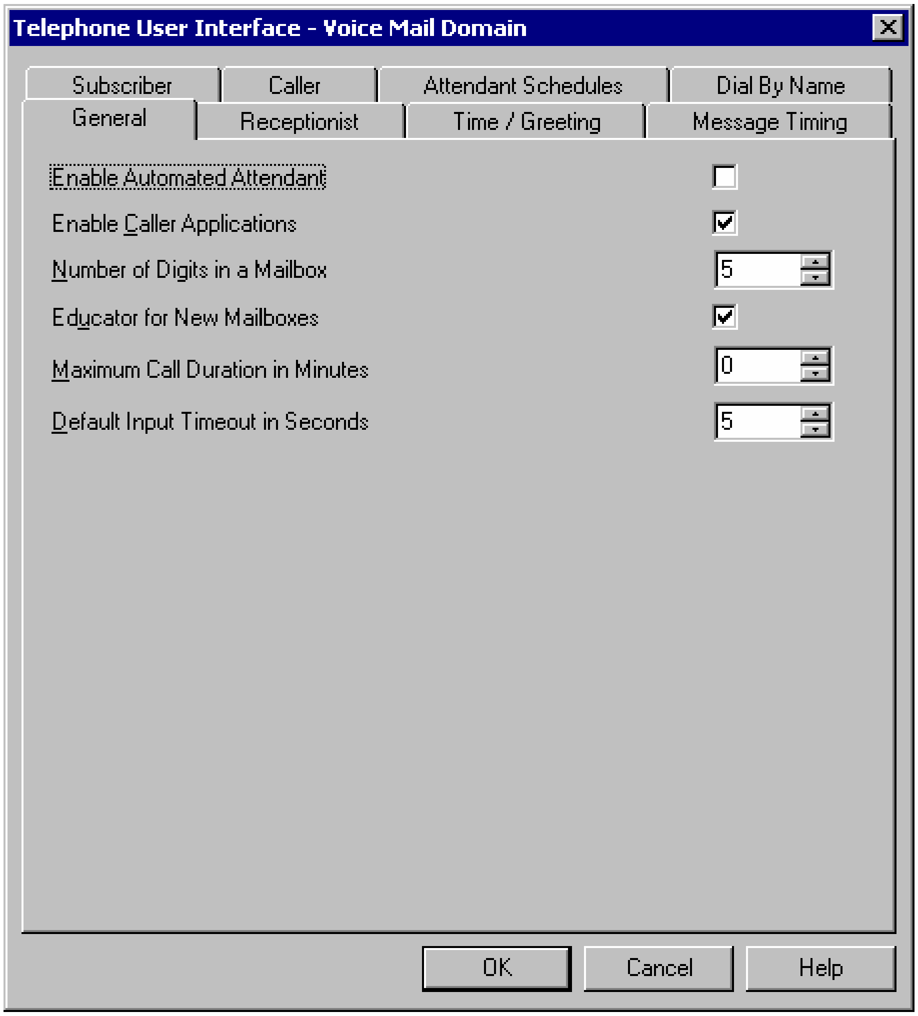
The following two screens illustrate aspects of the QSIG PBX integration used in these Application Notes. Double-click "PBX Type" from the screen shown in **Figure 2** to verify the following QSIG-related configuration.



Double-click "PBX Integration" from the screen shown in **Figure 2** to verify the following QSIG-related configuration.



The following screen illustrates the general Telephony User Interface parameters used during the verification of these Application Notes. Double-click “Telephone User Interface” from the screen shown in **Figure 2** to display the configuration.



7. Verifications and Advanced Topics

The illustrated configuration has been verified. The following subsections illustrate the expected behavior using the configuration presented in these Application Notes. Two main sub-sections follow. The first illustrates aspects of normal operation, when the data WAN is up, and Sites A, B, and C are controlled by the S8700 Media Server at Site A. The second main sub-section illustrates the behavior when the data WAN to Site C is down, and Site C is receiving service from the S8500 ESS. Whereas reference [1] focused on illustrating the behavior for LSP configurations (Site B), these Application Notes focus on the behavior for ESS configurations (Site C). The behavior when the data WAN to Site B is down has been omitted, since it has already been presented in Reference [1].

7.1. Normal Operation, Data WAN Up at Sites A, B, C

The following subsections show detailed screens captured from the S8700 Media Server in Site A, when the S8700 Media Server is processing calls for all sites. These screens illustrate the expected behavior when the WAN is up.

7.1.1. G350 Media Gateway at Site B Registered to C-LAN in Site A

The following screen shows that the G350 Media Gateway at Site B has registered to the C-LAN with IP-Address 1.1.15.20 in Site A.

```
display media-gateway 1
                              MEDIA GATEWAY
         Number: 1                          IP Address: 2  .2  .135.87
           Type: g350             FW Version/HW Vintage: 25 .22 .0  /1
           Name: G350-Right                MAC Address: 00:04:0d:29:c9:91
      Serial No: 03IS07589448             Encrypt Link? y
 Network Region: 3                             Location: 3
    Registered? y             Controller IP Address: 1  .1  .15 .20
  Recovery Rule: 2                            Site Data:
      Slot   Module Type               Name
       V1:   S8300                     ICC MM
       V2:
       V3:
       V4:   MM710                     DS1 MM
       V5:
       V6:   MM312                     DCP MM
       V7:   1T+2L-Integ-Analog        ANA IMM
       V8:
       V9:   gateway-announcements     ANN VMM
```

7.1.2. S8300 LSP and S8500 ESS Registered with Up-to-Date Translations

The following screen shows that the survivable processors are registered, along with the last translation updates. If re-creating and testing the configuration in these Application Notes, ensure that the survivable processors have the current copy of any configuration changes that have been entered into Avaya Communication Manager on the S8700 Media Server.

```
list survivable-processor
                             SURVIVABLE PROCESSORS
 Name           Type       IP Address     Reg LSP         Translations     Net
                                              Act         Updated          Rgn
 S8300-G350-LSP  LSP       2  .2  .135.88  y   n          22:00 2/2/2006    3
 ESSCid002Sid003 ESS       2  .2  .4  .88  y              22:00 2/2/2006    8
```

7.1.3. CMC Media Gateway (Site C) Controlled by S8700 (Site A)

The following screens show that the IPSI in the CMC (port network 3) is receiving service from the S8700 Media Server.

```
list ipserver-interface
                       IP SERVER INTERFACE INFORMATION

Port Pri/   Primary/         Primary/         Primary/                State Of
Ntwk Sec    Secondary        Secondary        Secondary Serv  Control Health
Num  Bd Loc IP Address       Host Name        DHCP ID   State State   C P E G
---- ------ ---------------  ---------------- --------- ----- ------- -------
  1   1AXX  198.151.254.101  198.151.254.101  ipsi-A01a  IN   actv-aa 0.0.0.0

  2   1E02  n/a

  3   3A02  2.2.4.9          2.2.4.9          ipsi-A03a  IN   actv-aa 0.0.0.0
```

In the following status screen, observe that the IPSI in port network 3 is being controlled by Cluster 1 (Site A) but is also communicating with Cluster 2 (Site C). Should the IPSI in Site C lose connectivity to Cluster 1, the IPSI will “reach up” to Cluster 2 (the S8500 ESS in Site C).

```
status ess port-networks

Cluster ID 1             ESS PORT NETWORK INFORMATION

                  Port IPSI   Pri/ Pri/     Cntl Connected
   Com  Intf Intf Ntwk Gtway  Sec  Sec      Clus Clus(ter)
PN Num  Loc  Type Ste  Loc    Loc  State      ID IDs

 1   1  1AXX IPSI up    1AXX  1AXX actv-aa  1   1
 2   1  1E03 EI   up    1AXX
 3   3  3A02 IPSI up    3A02  3A02 actv-aa  1   1   2
```

7.1.4. S8500 ESS In-Service with Up-to-Date Translations

The following screen shows that the S8500 ESS is registered, along with the last translation update to the translation file of the ESS.

```
status ess clusters
Cluster ID   1             ESS CLUSTER INFORMATION
                     Active
  Cluster            Server               Translations        Software
    ID     Enabled?    ID   Registered?     Updated           Version

     1        y        2        y       22:00 2/2/2006    R013x.01.0.628.3
     2        y        3        y       22:00 2/2/2006    R013x.01.0.628.3
```

7.1.5. Registered IP Stations Illustrating Network Region Assignment

The following screen illustrates the registration status of various IP Telephones, along with the network region assignments. Observe that the bold Site C user with extension 51041 is registered to the C-LAN with IP address 2.2.4.87 in the CMC in Site C. This user is in region 8. The bold Site B user with extension 57541 is registered to the C-LAN with IP address 1.1.15.21 in the MCC in Site A, and is in network region 3 (as configured in the ip-network-map).

```
list registered-ip-stations
                            REGISTERED IP STATIONS

Station  Set     Product    Prod  Station          Net Orig    Gatekeeper     TCP
Ext      Type    ID         Rel   IP Address       Rgn Port    IP Address     Skt
51041    4620    IP_Phone   2.300 2.2.4.201        8           2.2.4.87        y
57041    4620    IP_Phone   2.300 1.1.4.111        1           1.1.15.20       y
57042    4624    IP_Phone   1.830 1.1.1.33         1           1.1.15.20       y
57541    4620    IP_Phone   2.300 2.2.135.200      3           1.1.15.21       y
57542    4620    IP_Phone   2.300 2.2.135.201      3           1.1.15.20       y
57543    4610    IP_Phone   2.300 2.2.135.202      3           1.1.15.21       y
57544    4610    IP_Phone   2.300 2.2.135.203      3           1.1.15.21       y
57545    4602+   IP_Phone   1.800 2.2.135.204      3           1.1.15.20       y
```

7.1.6. Site A User 57041 Calls a Site C User, Cover to Modular Messaging

When the WAN is up to Site C and the S8700 Media Server is processing calls for both Site A and Site C, a call from a user at Site A to a user at Site C that covers to Modular Messaging at Site A will not require inter-site VoIP resources once the coverage completes. That is, the final connection of the Site A originator to Modular Messaging will be local to Site A, as shown with the "list trace" and "status station" screens in this section.

The trace shows a call from IP Station 57041 at Site A to Digital station 51021 at Site C. Initially, the call proceeds as an inter-region G.729 call between the Site A IP station and the Site C MEDPRO (2.2.4.89). When the call is not answered, the call covers to coverage path 36, whose first point in coverage is the Modular Messaging hunt group. The call routes to trunk group 34, the ISDN-PRI at Site A with QSIG integration to Modular Messaging. The final IP leg of the connection is an intra-region G.711MU connection from the IP Telephone at Site A to a MEDPRO resource in the MCC1 at Site A (1.1.15.19).

The caller leaves a message, and the Message Waiting Lamp on station 51021 is illuminated.

```
list trace station 57041                                           Page   1
                                LIST TRACE
time            data
15:45:44     active station    57041 cid 0x74
15:45:44     G711MU ss:off ps:20 rn:1/1 1.1.4.111:2242 1.1.15.18:2708
15:45:46     dial 51021
15:45:46     ring station    51021 cid 0x74
15:45:46     G729A ss:off ps:20 rn:1/8 1.1.4.111:2242 2.2.4.89:2424
             VOIP data from: 2.2.4.89:2424
15:45:55     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:30 WC:0 Avg:0
15:45:55     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 WC:0 Avg:0
15:45:57     no answer station    51021 cid 0x74
15:45:57     coverage-path  36 point 1  cid 0x74
15:45:57     call-forwarding 8234
15:45:57     term trunk-group 34    cid 0x74
15:45:57     xoip: fax:Relay modem:off tty:US 2.2.4.89:2460 (igc)
15:45:57     xoip: fax:Relay modem:off tty:US 1.1.15.18:2712 (igc)
15:45:57     G729 ss:off ps:20 rn:8/1 2.2.4.89:2460 1.1.15.18:2712
15:45:57     call-forwarding 82347555
15:45:57     route-pattern  34 preference 1  cid 0x74
15:45:57     seize trunk-group 34 member 12  cid 0x74
15:45:57     Calling Number & Name 57041 Mr. Midspan
15:45:57     G711MU ss:off ps:20 rn:1/1 1.1.4.111:2242 1.1.15.19:2132
15:45:57     Proceed trunk-group 34 member 12  cid 0x74
15:45:57     active trunk-group 34 member 12  cid 0x74
             VOIP data from: 1.1.15.19:2132
15:46:06     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 WC:1 Avg:0
15:46:06     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 WC:0 Avg:0
             VOIP data from: 1.1.15.19:2132
15:46:16     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 WC:1 Avg:0
15:46:16     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 WC:0 Avg:0
15:46:19     idle station    57041 cid 0x74
```

The “status station 57041” output for a similar (but separate) call is illustrated in the following screens. From Page 1, observe that the connected port is one of the members of trunk group 34.

```
station 57041                                               Page   1 of   7
                              GENERAL STATUS
     Administered Type: 4620               Service State: in-service/off-hook
        Connected Type: 4620           TCP Signal Status: connected
             Extension: 57041
                  Port: S00008        Parameter Download: complete
           Call Parked? no                 SAC Activated? no
      Ring Cut Off Act? no            CF Destination Ext:
Active Coverage Option: 1
          EC500 Status: N/A        Off-PBX Service State: N/A
       Message Waiting:
   Connected Ports: 01A0813
```

The remaining pages of “status station 57041” show more detailed information. The following screen is included to reinforce the observation that calls from Site A users to Site C users that cover to voice mail release the initial connection legs over the inter-region data WAN, once the call covers to Modular Messaging at Site A.

```
status station 57041                                            Page   6 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: S00008
S00008:TX:1.1.4.111:2242/g711u/20ms
01A0701:RX:1.1.15.19:2136/g711u/20ms   TX:tdm:a63
01A0813:RX:tdm:a63

dst port: 01A0813
```

7.1.7. Site C User Calls Another Site C User, Cover to Modular Messaging

When the WAN to Site C is up and the S8700 Media Server is processing calls for Site A and Site C, a call from a user (or trunk) at Site C to a user at Site C that covers to Modular Messaging at Site A will result in an inter-site, inter-region VoIP connection. That is, the final connection of the Site C originator to Modular Messaging will use the data WAN, as shown with the "list trace" and "status station" screens in this section.

The following trace shows a call from Site C IP Telephone 51041 to Site C digital telephone 51021. Initially, the call proceeds as an intra-site, intra-region G.711 call between the Site C IP Telephone (2.2.4.201) and the Site C MEDPRO (2.2.4.89). When the call is not answered, the call covers to coverage path 36, whose first point in coverage is the Modular Messaging hunt group. The call routes to trunk group 34, the ISDN-PRI at Site A with QSIG integration to Modular Messaging. The final IP leg of the connection is an inter-region Advanced Encryption Standard (AES) encrypted G.729 connection from the Site C IP Telephone (2.2.4.201) to a MEDPRO in the MCC1 at Site A (1.1.15.19). The caller leaves a message, and the Message Waiting Lamp on station 51021 is illuminated.

```
list trace station 51041                                               Page   1
                                LIST TRACE
time            data
15:52:40     active station    51041 cid 0x78
15:52:40     G711MU ss:off ps:20 rn:8/8 2.2.4.201:2334 2.2.4.89:2504
15:52:43     dial 51021
15:52:43     ring station    51021 cid 0x78
             VOIP data from: 2.2.4.89:2504
15:52:51     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 WC:0 Avg:0
15:52:51     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 WC:0 Avg:0
15:52:54     no answer station    51021 cid 0x78
15:52:54     coverage-path  36 point 1  cid 0x78
15:52:54     call-forwarding 8234
15:52:54     term trunk-group 34    cid 0x78
15:52:54     xoip: fax:Relay modem:off tty:US 2.2.4.89:2508 (igc)
15:52:54     xoip: fax:Relay modem:off tty:US 1.1.15.18:2724 (igc)
15:52:54     G729 ss:off ps:20 rn:8/1 2.2.4.89:2508 1.1.15.18:2724
15:52:54     call-forwarding 82347555
15:52:54     route-pattern  34 preference 1  cid 0x78
15:52:54     seize trunk-group 34 member 14  cid 0x78
15:52:54     Calling Number & Name 51041 John Survivor
15:52:54     G729A ss:off ps:20 rn:8/1 2.2.4.201:2334 1.1.15.19:2140
15:52:54     Proceed trunk-group 34 member 14  cid 0x78
15:52:54     active trunk-group 34 member 14  cid 0x78
.......................................................................
             VOIP data from: 1.1.15.19:2140
             15:53:12     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:30 WC:0 Avg:0
15:53:12     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 WC:0 Avg:0
15:53:18     idle station    51041 cid 0x78
```

The “status station 51041” output for a similar (but separate) call is illustrated in the following screens. From Page 1, it can be observed that the connected port is a member of the trunk group to Modular Messaging.

```
status station 51041                                            Page   1 of   7
                              GENERAL STATUS
     Administered Type: 4620                 Service State: in-service/off-hook
        Connected Type: 4620             TCP Signal Status: connected
             Extension: 51041
                  Port: S00006        Parameter Download: complete
           Call Parked? no                   SAC Activated? no
      Ring Cut Off Act? no            CF Destination Ext:
Active Coverage Option: 1

          EC500 Status: N/A        Off-PBX Service State: N/A
       Message Waiting:
   Connected Ports: 01A0815
```

The remaining pages show more detailed information. This is presented to reinforce the observation that calls from Site C users or trunks to other Site C users that cover to voice mail result in an inter-site, inter-region VoIP connection using the data WAN. In this case, the connection is between the IP Telephone in Site C and a MEDPRO in the MCC1 in Site A.

```
status station 51041                                            Page   6 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: S00006
S00006:TX:2.2.4.201:2334/g729a/20ms/aes
01A0701:RX:1.1.15.19:2144/g729a/20ms/aes   TX:tdm:a53
01A0815:RX:tdm:a53

dst port: 01A0815
```

7.1.8. Calls from Site C Users Directly to Modular Messaging (by extension)

When the WAN is up to Site C and the S8700 Media Server is processing calls for Site A and Site C, a call from a user at Site C to the Modular Messaging hunt group extension 57555 will result in an inter-site, inter-region VoIP connection. That is, the final connection of the Site C originator to Modular Messaging will use the data WAN, as shown with the “status station” screens in this section.

The following status screens show the result of a call from Site C digital telephone 51021 to the Modular Messaging hunt group extension 57555. The call routes to trunk group 34, the ISDN-PRI at Site A with QSIG integration to Modular Messaging. The final connection is an inter-region AES-encrypted G.729 connection from the Site C MEDPRO (2.2.4.89) to a MEDPRO in the MCC1 at Site A (1.1.15.18). The caller is known to Modular Messaging, and need only enter the password to log in. The caller listens to new messages, and the Message Waiting Lamp on the station is extinguished.

```
status station 51021                                          Page   1 of   4
                              GENERAL STATUS
     Administered Type: 6408D+            Service State: in-service/off-hook
        Connected Type: 6408 S/N
             Extension: 51021
                  Port: 03A0801       Parameter Download: complete
           Call Parked? no                 SAC Activated? no
      Ring Cut Off Act? no            CF Destination Ext:
Active Coverage Option: 1

          EC500 Status: disabled   Off-PBX Service State: in-service/idle
       Message Waiting: AUDIX
   Connected Ports: 01A0816
```

The following screen shows additional connection details.

```
status station 51021                                          Page   3 of   4
                      SRC PORT TO DEST PORT TALKPATH
src port: 03A0801
03A0801:TX:tdm:a93
03A0501:RX:tdm:a93   TX:2.2.4.89:2520/g729/20ms/aes
01A0601:RX:1.1.15.18:2732/g729/20ms/aes   TX:tdm:a253
01A0816:RX:tdm:a253
dst port: 01A0816
```

7.2. Data WAN Down to Site C, S8500 ESS Processing Site C Calls

The following subsections illustrate behavior when a failure is introduced, making it impossible for the IPSI in Site C to reach Avaya Communication Manager in Site A. For example, this would be the case if the data WAN to Site C is out-of-service. Unless stated otherwise, screens in this section were captured from the active S8500 Media Server ESS. Upon log in, a screen similar to the following will be displayed.

```

           Copyright (c) 1992 - 2006 Avaya Inc. All Rights Reserved.
Except where expressly stated otherwise, this Product is protected by copyright
and other laws respecting proprietary rights. Unauthorized reproduction,
transfer, and or use can be a criminal, as well as a civil, offense under the
applicable law. Certain software programs or portions thereof included in this
Product may contain software distributed under third party agreements ("Third
Party Components"), which may contain terms that expand or limit rights to use
certain portions of the Product ("Third Party Terms"). Information identifying
Third Party Components and the Third Party Terms that apply to them are
available on Avaya's web site at: http://support.avaya.com/ThirdPartyLicense/.


        * Enterprise Survivable Server (ESS) - Translations cannot be saved *

License-Error: Enterprise Survivable Server Controlling Port Networks
System Administration Will Be Blocked in Approximately 30 days
Contact Your Service Representative Immediately.
```

7.2.1. Site C IPSI Under Control of S8500 ESS

The following screen captures were taken while the WAN to Site C was still out of service. The next screen shows that the Site C IPSI is controlled by cluster 2, the S8500 ESS in Site C. Also observe that Cluster ID 1 is not listed as a connected cluster (i.e., since the WAN is still down). The Compact Modular Cabinet, port network 3, is controlled by Cluster 2, the S8500 ESS.

```
status ess port-networks

Cluster ID 2              ESS PORT NETWORK INFORMATION

                  Port IPSI   Pri/ Pri/     Cntl Connected
   Com  Intf Intf Ntwk Gtway  Sec  Sec      Clus Clus(ter)
PN Num  Loc  Type Ste  Loc    Loc  State      ID IDs

 1   1  1A01 EI   down        1AXX active
 2   1  1E03 EI   down  1AXX
 3   3  3A02 IPSI up    3A02  3A02 actv-aa  2   2
```

Observe that Cluster ID 2 is not registered with the primary server cluster at Site A.

```
status ess clusters
Cluster ID   2              ESS CLUSTER INFORMATION
                     Active
  Cluster            Server               Translations        Software
    ID     Enabled?    ID   Registered?     Updated           Version
     2        y         3        n       16:03 2/3/2006    R013x.01.0.628.3
```

Avaya Communication Manager on the S8500 ESS is controlling only the IPSI in the CMC Media Gateway. Since only the WAN to Site C is down, the MCC Media Gateways at Site A continues to be controlled by the S8700 Media Servers in Site A (not shown).

```
list ipserver-interface
                       IP SERVER INTERFACE INFORMATION

Port Pri/   Primary/         Primary/         Primary/                State Of
Ntwk Sec    Secondary        Secondary        Secondary Serv  Control Health
Num  Bd Loc IP Address       Host Name        DHCP ID   State State   C P E G
---- ------ ---------------  ---------------- --------- ----- ------- -------
  1   1AXX  198.151.254.101  198.151.254.101  ipsi-A01a  OUT  active  0.1.1.0

  2   1E02  n/a

  3   3A02  2.2.4.9          2.2.4.9          ipsi-A03a  IN   actv-aa 0.0.0.0
```

7.2.2. Site C IP Telephone Registration to S8500 ESS (via C-LAN)

The following screen shows the IP Telephone in Site C registered to the S8500 ESS. Although the "Gatekeeper IP Address" is the same as when the WAN was up, remember that this screen is captured from the active S8500 ESS.   The IP Telephone registers with a C-LAN in the CMC, due to the prior establishment of connectivity between the IPSI in the CMC with the S8500 ESS as illustrated in the previous section.

```
list registered-ip-stations
                            REGISTERED IP STATIONS
Station  Set     Product    Prod  Station          Net Orig    Gatekeeper     TCP
Ext      Type    ID         Rel   IP Address       Rgn Port    IP Address     Skt
51041    4620    IP_Phone   2.300 2.2.4.201        8           2.2.4.87        y
```

7.2.3. Status of Site A Trunk Group 34 to Modular Messaging As Seen By the Active S8500 ESS in Site C

The following screen shows that all members of trunk group 34 to Modular Messaging at Site A are considered out-of-service (OOS), from the perspective of the S8500 ESS, when the ESS is active. As a result, when hunt group 34 pointing to ISDN-PRI trunk group 34 is the first point in coverage, Avaya Communication Manager will proceed to the next point in coverage.

```
status trunk 34                                                        Page   1
                             TRUNK GROUP STATUS
Member   Port     Service State      Mtce Connected Ports
                                     Busy
0034/001 01A0801  OOS/FE-idle        no
0034/002 01A0802  OOS/FE-idle        no
..........................................................................
```

7.2.4. Status of Site C Trunk Group 48 to the PSTN As Seen By the Active S8500 ESS in Site C

The following screen shows that members of trunk group 48, the ISDN-PRI providing connectivity to the PSTN, are considered in-service, from the perspective of the S8500 ESS, when the ESS is active. As a result, the configuration described in this document will allow Avaya Communication Manager to route calls to other sites using the PSTN.

```
status trunk 48
                             TRUNK GROUP STATUS
Member   Port     Service State      Mtce Connected Ports
                                     Busy
0048/001 03A0601  in-service/idle    no
0048/002 03A0602  in-service/idle    no
0048/003 03A0603  in-service/idle    no
..........................................................................
```

7.2.5. Site C User Dials Extension of a Site A User That Has a DID Number

This section illustrates the behavior when a Site C user dials the extension of a Site A user that is configured with coverage path 34, the coverage path assigned to users at Site A that have a DID number. The following screen shows a trace taken from the S8500 ESS for a call from Site C x51041 to Site A x57041. For this call, the calling user sees a display showing the name translated for the dialed extension and the word "cover". For example, the display on the calling user is <Mr. Midspan   cover>. Absent further configuration, the called user 57041 sees a display showing the calling party information consistent with an ISDN-PRI delivered PSTN trunk call from Site C to Site A. For example, the display on the called user can be <CALL FROM 7325551041> assuming the calling party number (but not the name) was delivered over the PSTN (see Section 7.2.10 for a means to preserve station-to-station display characteristics).

The called user can answer the call. If the called user does not answer, the call covers to Modular Messaging, which directs the call to the mailbox for 57041, consistent with an incoming trunk call following coverage (see Section 7.2.10 for a means to preserve internal coverage behavior for such a call). If a message is left in the mailbox, the Modular Messaging system lights the message-waiting lamp of extension 57041. The envelope information can contain the calling party number, if delivered by the PSTN from Site C to Site A.

```
list trace station 51041                                                   Page   1
                                LIST TRACE
time            data
17:06:53     active station    51041 cid 0x5
17:06:53     G711MU ss:off ps:20 rn:8/8 2.2.4.201:2366 2.2.4.89:2180
17:06:56     call-forwarding
17:06:56     term station    57041 cid 0x5
17:06:56     call-forwarding 8234
17:06:56     term trunk-group 34    cid 0x5
17:06:56     call-forwarding
17:06:56     term station    57041 cid 0x5
17:06:56     call-forwarding 8255
17:06:56     term trunk-group 48    cid 0x5
17:06:56     dial 82557041 route:AAR
17:06:56     route-pattern  255 preference 1  cid 0x5
17:06:56     seize trunk-group 48 member 2  cid 0x5
17:06:56     Setup digits 7325557041
17:06:56     Calling Number & Name 7328551041 NO-CPName
17:06:56     Proceed trunk-group 48 member 2  cid 0x5
17:06:56     Alert trunk-group 48 member 2  cid 0x5
             VOIP data from: 2.2.4.89:2180
.................................................................................
17:07:12     active trunk-group 48 member 2  cid 0x5
             VOIP data from: 2.2.4.89:2180
17:07:14     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 WC:1 Avg:0
17:07:14     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 WC:0 Avg:0
             VOIP data from: 2.2.4.89:2180
17:07:37     idle station    51041 cid 0x5
```

7.2.6. Site C User Dials Extension of a Site A User That Has No DID Number

This section illustrates the behavior when a Site C user dials the extension of a Site A user that is configured with coverage path 35, the coverage path assigned to users at Site A that do not have a DID number. The following screen shows a trace taken from the S8500 ESS for a call from Site C x51021 to Site A x57021.

The calling user sees a display showing the called party's name as translated for the dialed extension and the word "cover". Transparent to the caller, the call is delivered to the vector at Site A and the S8500 ESS sends the called user's extension. The vector routes the call to the called user. As described in the previous section, the called user 57021 sees a display consistent with an incoming trunk call from Site C to Site A (see also Section 7.2.10). The called user can answer the call. If the called user does not answer, the call covers to Modular Messaging, which directs the call to the mailbox for 57021. If a message is left in the mailbox, Modular Messaging lights the message-waiting lamp of extension 57021.

In the trace, the bolded row shows the result of the coverage remote entry "L77556%,D" for a call to 57021. The "%" character is represented in the trace by "~m", the pause character by "~p", and the "D" character represents the dialed number 57021.

```
list trace station 51021
                                LIST TRACE
time            data
17:12:20     tone-receiver     03A0202 cid 0x8
17:12:20     active station    51021 cid 0x8
17:12:23     call-forwarding
17:12:23     term station    57021 cid 0x8
17:12:23     call-forwarding 8234
17:12:23     term trunk-group 34    cid 0x8
17:12:23     dial
17:12:23     term station    57021 cid 0x8
17:12:23     dial 77556 route:ARS
17:12:23     term trunk-group 48    cid 0x8
17:12:23     dial 77556~m~p57021 route:ARS
17:12:23     route-pattern  575 preference 1  cid 0x8
17:12:23     seize trunk-group 48 member 3  cid 0x8
17:12:23     Calling Number & Name 7328551021 NO-CPName
17:12:23     Proceed trunk-group 48 member 3  cid 0x8
17:12:24     active trunk-group 48 member 3  cid 0x8
17:13:08     idle station    51021 cid 0x8
```

The following screen shows a VDN trace of a similar call taken from the active S8700 Media Server at the main site.

```
list trace vdn 57556
                                LIST TRACE
time      vec st data
14:59:39    0  0 ENTERING TRACE cid 402
14:59:39   56  1 vdn e57556 bsr appl   0 strategy 1st-found override n
14:59:39   56  1 collect
14:59:41   56  2 route-to
14:59:41   56  3 LEAVING VECTOR PROCESSING cid 402
14:59:41   56  3 TRACE COMPLETE cid 402
```

The following screen shows the incoming trunk trace for a similar call taken from the active S8700 Media Server at Site A. From this trace, the routing of the call first to the VDN (57556) and then to the "D" dialed user (57021) is evident. In this trace, the call was not answered, and the trace shows the call going to Modular Messaging (via trunk group 34) at Site A.

```
list trace tac 187                                                    Page   1
                                      LIST TRACE
time            data
17:05:24     Calling party trunk-group 87 member 1  cid 0x99
17:05:24     Calling Number & Name 7328551021 NO-CPName
17:05:24     active trunk-group 87 member 1  cid 0x99
17:05:24     dial 57556
17:05:24     term vector 56    cid 0x99
17:05:24     G711MU ss:off ps:20 rn:1/1 2.2.85.3:2244 1.1.15.18:2964
17:05:26     dial 57021
17:05:26     ring station    57021 cid 0x99
17:05:26     G711MU ss:off ps:20 rn:1/1 1.1.4.111:2242 1.1.15.18:2968
17:05:47     no answer station    57021 cid 0x99
17:05:47     coverage-path  35 point 1  cid 0x99
17:05:47     call-forwarding 8234
17:05:47     term trunk-group 34    cid 0x99
17:05:47     dial 82347555 route:AAR
17:05:47     route-pattern  34 preference 1  cid 0x99
17:05:47     seize trunk-group 34 member 22  cid 0x99
17:05:47     Setup digits 57555
17:05:47     Calling Number & Name 7328551021 NO-CPName
17:05:48     Proceed trunk-group 34 member 22  cid 0x99
17:05:48     active trunk-group 34 member 22  cid 0x99
17:06:05     idle trunk-group 87 member 1  cid 0x99
```

7.2.7. Call to Site C User That Covers to Modular Messaging over PSTN

This section illustrates the behavior when a user at site C calls another user (x51021) at Site C. The called user does not answer, and the call proceeds to coverage. The Modular Messaging hunt group is not available to the S8500 ESS, so the call proceeds to the second point in coverage, which directs the call out the PSTN to Modular Messaging. Modular Messaging delivers the call to the mailbox for user x51021, as a result of the "D" digits being sent after Modular Messaging answers. The caller leaves a message. **Note that Modular Messaging cannot update the message-waiting lamp for the Site C user when the S8500 ESS is active**. The proper message waiting lamp status is updated when Site C returns to normal service. Once the data WAN to Site C is restored, the return of the Site C IPSI to the control of the primary servers at Site A can either be manual or automatic (via scheduled return).

The following trace illustrates the behavior when Site C x51041 dials Site C x51021. Note the use of coverage path 36, and the arbitrary UDP number 77555 routed through ARS to routing pattern 575, containing trunk group 48.

```
list trace station 51041                                              Page   1
                                LIST TRACE
time            data
17:25:39     active station    51041 cid 0x14
17:25:39     G711MU ss:off ps:20 rn:8/8 2.2.4.201:2366 2.2.4.89:2248
17:25:40     dial 51021
17:25:40     ring station    51021 cid 0x14
             VOIP data from: 2.2.4.89:2248
17:25:49     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 WC:0 Avg:0
17:25:49     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 WC:0 Avg:0
17:25:51     no answer station    51021 cid 0x14
17:25:51     coverage-path  36 point 1  cid 0x14
17:25:51     call-forwarding 8234
17:25:51     term trunk-group 34    cid 0x14
17:25:51     dial
17:25:51     term station    51021 cid 0x14
17:25:51     dial 77555 route:ARS
17:25:51     term trunk-group 48    cid 0x14
17:25:51     dial 77555~m51021 route:ARS
17:25:51     route-pattern  575 preference 1  cid 0x14
17:25:51     seize trunk-group 48 member 3  cid 0x14
17:25:51     Calling Number & Name 7328551041 NO-CPName
17:25:51     Proceed trunk-group 48 member 3  cid 0x14
17:25:51     active trunk-group 48 member 3  cid 0x14
             VOIP data from: 2.2.4.89:2248
.....................................................................................
17:26:19     Jitter:0 0 0 0 0 0 0 0 0 0 0: Buff:8 WC:1 Avg:0
17:26:19     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 WC:0 Avg:0
17:26:20     idle station    51041 cid 0x14
```

7.2.8. Site C User Dialing Modular Messaging Voice Mail Access Number

In Reference [1] using Avaya Communication Manager 3.0.1 (load 346), different approaches to preserving dialing transparency to the voice mail hunt group were presented. However, those alternate approaches did not use the set of “coverage remote” enhancements. This section illustrates an enhancement available in Avaya Communication Manager 3.1 that enables the “coverage remote” approach to apply for calls to the extension of the voice mail hunt group, when the voice mail hunt group is inaccessible to a survivable processor. For example, in the sample configuration, users can continue to dial extension 57555 to reach Modular Messaging, whether service is currently being provided by Site A or a survivable processor. Recall that when Site C is receiving survivable service from the S8500 ESS (e.g., due to WAN outage), the message waiting lamp status of a telephone at Site C will not necessarily be an accurate reflection of whether messages are waiting.

The following trace shows Site C user 51021 calling 57555. The call is automatically routed out ISDN-PRI trunk group 48 in the CMC Media Gateway, and the full PSTN routable number (732-555-7555) to reach Modular Messaging at Site A is sent out.

```
list trace station 51021                                          Page   1
                                LIST TRACE
time            data
15:48:56     tone-receiver     03A0201 cid 0x22
15:48:56     active station    51021 cid 0x22
15:48:58     call-forwarding
15:48:58     term hunt-group 34    cid 0x22
15:48:58     call-forwarding 8234
15:48:58     term trunk-group 34    cid 0x22
15:48:58     call-forwarding
15:48:58     term hunt-group 34    cid 0x22
15:48:58     call-forwarding 973255575
15:48:58     term trunk-group 48    cid 0x22
15:48:58     dial 97325557555 route:ARS
15:48:58     route-pattern  33 preference 1  cid 0x22
15:48:58     seize trunk-group 48 member 3  cid 0x22
15:48:58     Setup digits 7325557555
15:48:58     Calling Number & Name 7328551021 NO-CPName
15:48:58     Proceed trunk-group 48 member 3  cid 0x22
15:48:59     active trunk-group 48 member 3  cid 0x22
15:49:54     idle station    51021 cid 0x22
```

Without further configuration, when the Site C user dials Modular Messaging and the call arrives to Site A from the PSTN, Modular Messaging will greet the user similar to when the user dialed from home. That is, the mailbox of the calling user will be unknown to Modular Messaging, so the user will be required to enter the mailbox (extension) and password, and not simply the password. Once logged in, the user can listen to messages, but Modular Messaging will not be able to change the message waiting status of a Site C telephone, until Site C returns to normal operation (i.e., data WAN is restored, and the IPSI in Site C returns the CMC gateway to the control of the primary cluster at Site A). However, with optional configuration of an “off-pbx station-mapping” for the Site C calling user, as shown in Section 7.2.10, it is possible for Modular Messaging to know the mailbox of the calling user. This would allow Modular Messaging to greet the caller in the same fashion as when Site C is under control of the primary server at Site A.

7.2.9. Timing Considerations When Using the % and D Characters

In the illustrated configuration, a pause character (“,”) was inserted after the wait for answer (“%”) character for calls that are directed to the Site A VDN serving non-DID users. This pause was added to mitigate the possibility of the S8500 ESS system sending the DTMF digits before the receiving system is fully prepared to collect the digits. In the case of the VDN, there is no “penalty” for the insertion of the pause, save for the small additional delay before ringing the called party.

In some configurations, it may be necessary to insert the pause character for calls directed to a messaging system as well. However, in the case of a messaging system, there may be a “penalty” to inserting the pause. If the pause is used, the caller will still be directed automatically to the called user’s mailbox when the ESS is active, but due to the pause, the caller may hear the beginning of the messaging system's default greeting. This default greeting would be interrupted automatically and followed by the expected subscriber mailbox answer. If the pause is not needed, the caller will be directed to the expected subscriber mailbox answer, and will not hear the beginning of the messaging system's default greeting. In practice, the pause should only be inserted if empirical experience suggests that the pause is required to ensure accuracy of the automated digit generation and collection process.

If it is determined that a pause is necessary for messaging system collection accuracy, and the possibility of hearing the beginning of the default messaging system greeting is unacceptable, calls can be directed to an application rather than the messaging system's main number. For example, the S8500 ESS could direct calls to a Modular Messaging caller application that delays audible feedback for several seconds before collecting the digits sent from the S8500 ESS that represent the mailbox number.

7.2.10. Preserving "Station-to-Station" Behavior Using Off-PBX Telephone Station Mapping

Optionally, the concept of "off-pbx station mapping" can be used to allow calls from Site C (or Site B) to Site A to behave similarly with respect to displays, ringing patterns, coverage criteria, and any other feature that treats "internal" and "external" calls differently. As noted in Section 3, the optional configuration in this section is subject to the availability of an EC500 license for each user that will have the public network formatted calling number mapped to the corresponding local extension. Under normal conditions, when Site A and Site C are under the control of the primary cluster at Site A, a call from Site C to Site A is an "internal" call. Displays, ringing patterns, call coverage, and other features would reflect a local "extension to extension" call. However, absent the use of "off-pbx station mapping", when an extension-dialed call arrives from Site C to Site A over the PSTN (e.g., when the ESS is active), Site A would treat the call as an incoming trunk call. Ringing, display, coverage and other features that distinguish internal vs. external calls would have the "external call" characteristic. If it is desirable to retain the internal "extension to extension" properties of a Site C to Site A call when the S8500 ESS is active, the "off-pbx station mapping" can be used to map the Site C user's PSTN-formatted calling party number to the Site C user's local extension. This same mechanism is used by the Extension to Cellular feature to allow a call from a cellular telephone to appear to have been originated from the "desk telephone" of the cellular telephone user, when the cellular telephone dials an enterprise user.

As an example, if the objective is for any call from Site C station 51021 to users at Site A to appear identical when the S8500 ESS is actively processing calls, the following entry could be added to the "off-pbx-telephone station-mapping" form. Assume the PSTN-formatted identification associated with extension 51021 is 7328551021. When a call arrives with a calling party identification of "7328551021", Avaya Communication Manager will behave as if the call had been originated by local station extension 51021. Note that this would apply both for direct dialed calls from Site C user 51021 to a full Site A PSTN number (e.g., 9-732-555-7041) as well as calls routed through the "coverage remote" form as detailed in Section 5.

```
change off-pbx-telephone station-mapping 51021                    Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 51021         EC500          -  7328551021         ars         1
```

The following screen shows Page 2, highlighting the use of "origination" mapping mode.

```
change off-pbx-telephone station-mapping 51021                    Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 51021         2         origination    all         both
```

Other implications with this mapping in place for a call from Site C user 51021 in ESS mode to Site A extensions are as follows:

- If Site C user 51021 were to call the Site A Modular Messaging number (x57555), as described in Section 7.2.8, the mailbox of the originator (i.e., 51021) would be conveyed to Modular Messaging automatically. The user need only enter the password.
- The name and extension translated for station user 51021 would appear in displays, the same as if both Site C and Site A were under the control of Site A.
- Call coverage would follow internal rather than external coverage criteria.
- Other Avaya Communication Manager features that distinguish internal vs. external call behavior would behave like an internal call.
- Modular Messaging will have the usual calling party information, potentially enabling other features to operate normally. For example, when the Site A user retrieves the message, the originator can be identified by spoken name and Modular Messaging will present the "reply options". If "Notify Me" or "Call Me" rules apply for the calling party, the notifications can still occur when the calling party is served by an S8500 ESS, provided the calling party's PSTN-formatted ANI is mapped to the extension via the "off-pbx station-mapping".

8. Conclusion

As illustrated in these Application Notes, Avaya Communication Manager and Avaya Modular Messaging enable customers with geographically distributed sites to achieve centralized management of their call processing and messaging applications. The configuration described in these Application Notes allows a voice mail subscriber located in a branch office site to receive Modular Messaging voice mail services, even when the data WAN linking the branch site to the Modular Messaging at the central site is down. Calls originated by branch users can be automatically routed out PSTN trunks in the branch Media Gateway when the data WAN is down and either an Enterprise Survivable Server or a Local Survivable Processor is active. As described in these Application Notes, callers can be automatically directed to the proper mailbox and personal greeting of a branch user, even when a survivable processor is actively processing calls. In addition, branch users can have "dialing transparency" allowing calls to be dialed by extension, even when a survivable processor is actively providing service.

9. References

The following references are among the Application Notes available at http://www.avaya.com.

Reference [1] was written based on Avaya Communication Manager 3.0.1 load 346, which allowed the "coverage remote" enhancements to apply to users server by S8300 Local Survivable Processors (but not Enterprise Survivable Servers).

[1] "Configuring Avaya Communication Manager for Voice Mail Service and Dialing Transparency for Sites Using Avaya Media Gateways with Local Survivable Processors, Issue 1.0"

Reference [2] was written prior to the enhancements in Avaya Communication Manager 3.0.1, and provides alternate solutions to similar issues.

[2] "Configuring Avaya Communication Manager and Avaya Modular Messaging for Voice Mail Service to Sites Using Avaya Media Gateways with Local Survivable Processors, Issue 1.0"

Reference [3] focuses on media encryption using a network configuration that is very similar to the one depicted in **Figure 1** of these Application Notes.

[3] "Configuring Avaya Communication Manager for Media Encryption – Issue 1.0"

The following references can be found at the Avaya support web site, http://support.avaya.com

[4] "Avaya Definity G3, IP600, S8300/S8700, & Prologix – T1/QSIG", Configuration Note 88003, Version G, 12/03

[5] "Avaya Enterprise Survivable Servers (ESS) User Guide", Document ID 03-300428, Issue 1.1, June 2005.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com