Avaya Solution & Interoperability Test Lab

Application Notes for Visionutveckling VIP 2000 with Avaya Communication Manager – Issue 1.0

Abstract

These Application Notes describe the compliance testing of the Visionutveckling VIP 2000 with Avaya Communication Manager. These Application Notes contain a description of the configurations for both VIP 2000 and Avaya Communication Manager which were used for testing. The testing which was performed tested the major functions of the VIP 2000 product.

Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the configuration used to enable the Visionutveckling VIP 2000 server to interoperate with Avaya Communication Manager. VIP 2000 is a voicemail system which also provides additional features, such as:

- The ability to queue a call to wait until a busy user is free
- Breakthrough to diverted user
- Conferencing

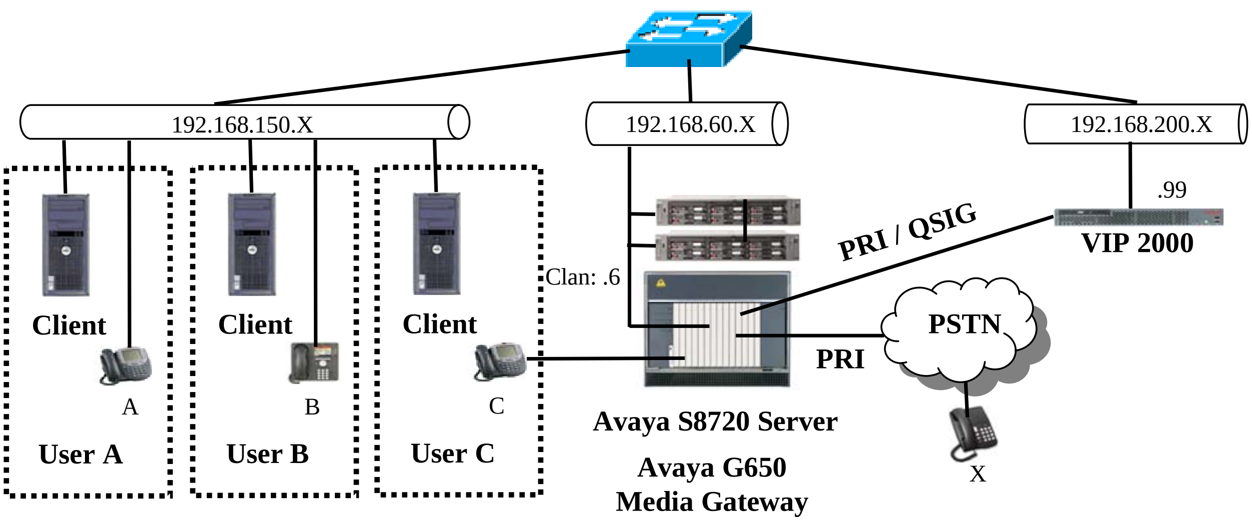

Figure 1: **VIP 2000 Test Configuration**

In the above diagram, VIP 2000 is connected to the Avaya S8720 server via a PRI interface on the Avaya G650 Gateway, from which it receives voicemail calls.

The following table contains details about the telephones which were used for testing.

Phone	Model	Extension	PSTN
A	4620SW IP	60121	069 907 39887 60121
B	9640	60093	069 907 39887 60093
C	2410	60007	069 907 39887 60007
X			069 7505 6174

Table 1: Extensions Used for Testing

The following table contains a list of extensions which were used to activate VIP 2000 functions.

HG	Extn	DDI	Usage
55	65555	5555	A-Num route
77	60077	0077	Voicemail Coverage
80	60080	5678	Call flow: Conference

Table 2: Functional Extensions

2. Equipment and Software Validated

Software Component	Version
Avaya Communication Manager	R015x.01.1.415.1
Avaya TSAPI Client	4.2.1-338
Avaya TN2312BP IP Server Interface	HW15/FW042
Avaya TN799DP Control LAN	HW01/FW026
Avaya TN2302AP Media Processor	HW20/FW033
Avaya TN2464CP DS1 Interface	HW01/FW19
Avaya 46xxSW IP Telephone	2.887
Avaya 96xx IP Telephone	2.0.3.0
Asterisk	1.4.19
Sangoma E1 Interface	2.3.4-16
Visionutveckling VIP 2000SW	8.2
Visionutveckling VIP 2000 platform OS	Linux Cent OS 4.5 Kernel 2.6.9

Table 3: Hardware/Software Component Versions

3. Configuration

3.1. Configure Avaya Communication Manager

The configuration and verification operations illustrated in this section were all performed using the Avaya Communication Manager System Administration Terminal (SAT) via SSH port 5022.

The information provided in this section describes the configuration of Avaya Communication Manager for this solution. For all other provisioning information such as installation and configuration, please refer to the product documentation in references **[1]** and **[2]**.

3.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer options** command to verify that Avaya Communication Manager is configured to meet the minimum requirements to run VIP 2000. Those items shown in **bold** indicate required values or minimum capacity requirements. If these are not met in the configuration, please contact an Avaya representative for further assistance.

Parameter	Usage
Maximum Concurrently Registered IP Stations (p.2)	This must be sufficient to support the total number of IP stations.
Computer Telephony Adjunct Links? (p.3)	This parameter must be set to “y”.
IP Stations? (p.4)	This parameter must be set to “y”.
ISDN-PRI? (p.4)	This is required to allow the allocation of the PRI trunk to be attached to VIP 2000.
IP_Phone (p.10)	This parameter must be set the number of IP stations.

Table 4: System-Parameters Customer-Options Parameters

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   60
           Maximum Concurrently Registered IP Stations: 12000 4
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 10    0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 1000  255
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 10    0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 2: System-Parameters Customer-Options Screen, Page 2

```
display system-parameters customer-options                      Page   3 of  11
                                OPTIONAL FEATURES

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

Figure 3: System-Parameters Customer-Options Screen, Page 3

```
display system-parameters customer-options                      Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                                IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                          ISDN Feature Plus? n
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                            ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                   ISDN-PRI? y
              ESS Administration? n              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? n
     External Device Alarm Admin? n                 Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n                  Multifrequency Signaling? y
      Global Call Classification? n        Multimedia Call Handling (Basic)? n
              Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? n
 Hospitality (G3V3 Enhancements)? n              Multimedia IP SIP Trunking? n
                      IP Trunks? y

           IP Attendant Consoles? y
```

Figure 4: System-Parameters Customer-Options Screen, Page 4

```
display system-parameters customer-options                      Page  10 of  11
                     MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID  Rel. Limit          Used
IP_API_A      : 1000            0
IP_API_B      : 1000            0
IP_API_C      : 1000            0
IP_Agent      : 1000            0
IP_IR_A       : 1000            0
IP_Phone      : 12000           4
IP_ROMax      : 12000           0
IP_Soft       : 1000            0
IP_eCons      : 128             0
oneX_Comm     : 12000           0
```

Figure 5: System-Parameters Customer-Options Screen Page 10

3.1.2. Configure Dial Plan and Call Routing

Use the **change dialplan analysis** command to specify that dialed strings for the number plan, as shown in the following table.

Dialed String	Usage
0	The prefix for PSTN numbers.
6	The leading digit of local extensions, as listed in **Table 1**.
*18	The Trunk Access Code which is used to access the VIP 2000 trunk.
*9	The Trunk Access Code which is used to access the PSTN trunk.

Table 5: Dial Plan Parameters

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     0            1    fac
     6            5    ext
     *18          3    dac
     *9           2    dac
```

Figure 6: Dialplan Analysis Screen

Use the **change aar analysis** command to select routing pattern "18" for numbers which have the leading dialed string "78".

```
change aar analysis 0                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                           Percent Full:    3

          Dialed           Total     Route    Call   Node   ANI
          String         Min  Max  Pattern    Type   Num    Reqd
    78                   6    6     18        aar           n
```

Figure 7: AAR Analysis Form

Use the **change route-pattern** command to route numbers using Routing Pattern "18" via Trunk Group "18", as shown in the following table.

Parameter	Usage
Pattern Name	Specify an appropriate name to identify the routing pattern.
Grp No	Specify "18".
No. Del Dgts	Specify "2" to delete the leading digits which were used to select the trunk.
TSC	Specify "y".
CA-TSC Request	Specify "as-needed".
Format	Specify "lev0-pvt".

Table 6: Route-Pattern 18 Parameters

```
change route-pattern 18                                         Page   1 of   3
                    Pattern Number: 18  Pattern Name: LINUX
                             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: 18   0                    2                                         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 M 4 W     Request                                Dgts Format
                                                       Subaddress
 1: y y y y y n  y  as-needed rest                              lev0-pvt  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
```

Figure 8: Route-Pattern 18 Form

3.1.3. Configure IP Network Interface

Use the **change node-names ip** command to configure IP address, as shown in the following table.

Parameter	Usage
clan	Enter the IP address of the CLAN interface of the S8720.

Table 7: Node-Names IP Parameters

```
change node-names ip                                        Page   1 of   2
                                 IP NODE NAMES
    Name              IP Address
clan                192.168.60.6
default             0.0.0.0
```

Figure 9: Node-Names IP Screen

3.1.4. Configure PRI Interface to the VIP 2000 Server

Use the **add ds1 <media module hardware address>** command to configure the DS1 interface card to serve as a Primary Rate ISDN interface. Assign those values for this command as shown in the following table.

Parameter	Usage
Bit Rate	Assign the bit rate to "2.048", as required to connect to the VIP 2000 E1 interface card.
Line Coding	Assign the Line Coding to "hdb3", as required to connect to the VIP 2000 E1 interface card. This must match the value used for the "FE_LCODE" parameter shown in **Figure 27.**
Name	Assign a name to be used to identify the card.
Signaling Mode	Assign the signaling mode to "isdn-pri". This must match the value used for the "CommPort" parameter shown in **Figure 27.**
Connect	Specify the connection is to a "pbx"
Interface	Specify the G350 is to serve as the "peer-master".
Peer Protocol	Specify the "Q-SIG" protocol is to be used.
Interface Companding	Specify "alaw" speech encoding is to be used.
CRC?	Specify a cyclic-redundancy-check sequence is not to be sent.
Idle Code	Specify that an idle sequence of "01010100" is to be sent on the interface when no data is being transmitted.
Channel Numbering	Specify that "timeslot" channel numbering is to be used.
Slip Detection?	Specify that slip detection is to be used.

Table 8: DS1 Parameters for PRI Interface

```
add ds1 01a08                                                  Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 01A08                              Name: LINUX
            Bit Rate: 2.048                       Line Coding: hdb3

      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: alaw                               CRC? n
           Idle Code: 01010100              Channel Numbering: timeslot
                              DCP/Analog Bearer Capability: 3.1kHz

                                              T303 Timer(sec): 4
                                             Disable Restarts? n

      Slip Detection? y                 Near-end CSU Type: other

   Echo Cancellation? n
```

Figure 10: Ds1 Form for PRI Interface

Use the **add trunk-group** command to configure the Trunk Group to the VIP 2000 Server. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type (p.1)	Specify the Group Type as “isdn”
Group Name (p.1)	Select an appropriate name to identify the device.
TAC (p.1)	Specify a trunk access code that can be used to provide dial access to the trunk. This value is included in the dial plan configured in **Figure 6**.
Carrier Medium (p.1)	Specify a Carrier Medium of “PRI/BRI”, as PRI will be used for this trunk.
Dial Access (p.1)	Allow dial access to the trunk by dialing the trunk access code.
Service Type (p.1)	Designate the trunk as a “tie” line to a peer system.
Supplementary Service Protocol (p.2)	Specify a Supplementary Service Protocol of “b” for QSIG.
Digit Handling (p.2)	Specify “enbloc/enbloc” to use block sending of dialed digits.
Send Name (p.3)	Specify “y” so that the name of the caller is sent for outgoing calls.
Send Calling Number (p.3)	Specify “y” so that the number of the caller is sent for outgoing calls.
Format (p.3)	Specify “unknown” to use unknown dialing plan for calls in both directions.
Supress # Outpulsing? (p.3)	Specify “y”.
Send Connected Number (p.3)	Specify “y” so that the number of the connected party is sent to the caller.
QSIG Value-Added? (p.4)	Specify “y”.
Group Member Assignments (p. 5, 6)	Assign the interface ports on the E1 interface to the trunk group members. Note that port 16 is used for the D channel, which must be assigned to the signaling group associated with this trunk.

Table 9: Trunk-Group Parameters for PRI Interface

```
add trunk-group 18                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 18                   Group Type: isdn          CDR Reports: y
  Group Name: LINUX                       COR: 1        TN: 1        TAC: *18
   Direction: two-way        Outgoing Display? n         Carrier Medium: PRI/BRI
 Dial Access? y              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n           TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

Figure 11: Trunk-Group Form for PRI Interface, Page 1

```
add trunk-group 18                                              Page   2 of  21
      Group Type: isdn

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

            Trunk Hunt: cyclical
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
          Administer Timers? n        CONNECT Reliable When Call Leaves ISDN? n
```

Figure 12: Trunk-Group Form for PRI Interface, Page 2

```
add trunk-group 18                                              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: 10
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n             Hop Dgt? n     Send EMU Visitor CPN? n
   Suppress # Outpulsing? y    Format: unknown
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send 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

    Apply Local Ringback? n
 Show ANSWERED BY on Display? y
                             Network (Japan) Needs Connect Before Disconnect? n
```

Figure 13: Trunk-Group Form for PRI Interface, Page 3

```
add trunk-group 18                                           Page   4 of  21
                       QSIG TRUNK GROUP OPTIONS




   TSC Method for Auto Callback: drop-if-possible
           Diversion by Reroute? y
               Path Replacement? y
Path Replacement with Retention? n
        Path Replacement Method: better-route
                           SBS? n
  Display Forwarding Party Name? y
    Character Set for QSIG Name: eurofont
               QSIG Value-Added? y
   QSIG-Value Coverage Encoding: proprietary
```

Figure 14: Trunk-Group Form for PRI Interface, Page 4

```
add trunk-group 18                                              Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/30
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  30

       Port    Code Sfx Name        Night           Sig Grp
  1: 01A0801  TN2464 C                                18
  2: 01A0802  TN2464 C                                18
  3: 01A0803  TN2464 C                                18
  4: 01A0804  TN2464 C                                18
  5: 01A0805  TN2464 C                                18
  6: 01A0806  TN2464 C                                18
  7: 01A0807  TN2464 C                                18
  8: 01A0808  TN2464 C                                18
  9: 01A0809  TN2464 C                                18
 10: 01A0810  TN2464 C                                18
 11: 01A0811  TN2464 C                                18
 12: 01A0812  TN2464 C                                18
 13: 01A0813  TN2464 C                                18
 14: 01A0814  TN2464 C                                18
 15: 01A0815  TN2464 C                                18
```

Figure 15: Trunk-Group Form for PRI Interface, Page 5

```
add trunk-group 18                                              Page   6 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/30
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  30

       Port    Code Sfx Name        Night           Sig Grp
 16: 01A0817  TN2464 C                                18
 17: 01A0818  TN2464 C                                18
 18: 01A0819  TN2464 C                                18
 19: 01A0820  TN2464 C                                18
 20: 01A0821  TN2464 C                                18
 21: 01A0822  TN2464 C                                18
 22: 01A0823  TN2464 C                                18
 23: 01A0824  TN2464 C                                18
 24: 01A0825  TN2464 C                                18
 25: 01A0826  TN2464 C                                18
 26: 01A0827  TN2464 C                                18
 27: 01A0828  TN2464 C                                18
 28: 01A0829  TN2464 C                                18
 29: 01A0830  TN2464 C                                18
 30: 01A0831  TN2464 C                                18
```

Figure 16: Trunk-Troup Form for PRI Interface, Page 6

Use the **add signaling-group** command to allocate a signaling group to this trunk.

Parameter	Usage
Group Type	Specify "isdn-pri" for ISDN primary rate.
Primary D-Channel	Assign port 16 of the DS1 interface as the D channel. This must match the "TDMV_DCHAN" parameter shown in **Figure 27**.
Trunk Group for Channel Selection	Specify "18" as the Trunk Group to be used for channel selection.
TSC Supplementary Service Protocol	Specify "b" to designate use of the QSIG protocol.

Table 10: Signaling-Group Parameters for PRI Interface

```
add signaling-group 18                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 18             Group Type: isdn-pri
                    Associated Signaling? y          Max number of NCA TSC: 10
                       Primary D-Channel: 01A0816     Max number of CA TSC: 10
                                                   Trunk Group for NCA TSC: 18
       Trunk Group for Channel Selection: 18
      TSC Supplementary Service Protocol: b          Network Call Transfer? n
```

Figure 17: Signaling-Group Form for PRI Interface

Use the **change private-numbering** command to specify that the calling party number is to be sent for calls which are made via the E1 trunk to VIP 2000 trunk.

Parameter	Usage
Ext Len	Specify “5” as the length of local extensions.
Ext Code	Specify “6” as the leading digit of local extensions.
Trk Grp	Specify “18” as the trunk which connects to the LINUX server.
Total Len	Specify “5” as the length of local extensions.

Table 11: Private-Numbering Parameters for PRI Interface

```
change private-numbering 0                                          Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 7  1                                            7     Total Administered: 10
 5  2                                            5        Maximum Entries: 540
 5  3                                            5
 7  5                                            7
 5  6                18                          5
 6  6                                            6
 7  6                                            7
 6  81                                           6
 7  82                                           7
 7  83                                           7
```

Figure 18: Private-Numbering Form for PRI Interface

3.1.5. Configure Stations

3.1.5.1 Configure H.323 IP Stations

Use the **add station** command to create IP stations for extensions A and B, as shown in **Table 1**.

Parameter	Usage
Extension	Use an unused extension which is compatible with the dial plan.
Type	Use a type value which corresponds to the physical station to be used.
Name	Any alphanumeric string can be assigned as an extension name, which is used for identification purposes.
Security Code	Enter an appropriate numeric string to be used as a security code.
Coverage Path 1	Enter the number of the coverage path which is defined in **Figure 23**

Table 12: Configuration IP Stations

```
add station 60121                                               Page   1 of   5
                                     STATION

Extension: 60121                         Lock Messages? n               BCC: 0
     Type: 4620                          Security Code: 12106            TN: 1
     Port: S00101                      Coverage Path 1: 999             COR: 1
     Name: extn 60121                  Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 60121
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n




                                              Customizable Labels? y
```

Figure 19: IP Station Screen

3.1.5.2 Configure Digital Stations

Use the **add station** command to create an IP station for extension C, as shown in **Table 1**

Parameter	Usage
Extension	Use an unused extension which is compatible with the dial plan.
Type	Use a type value which corresponds to the physical station to be used.
Name	Any alphanumeric string can be assigned as an extension name, which is used for identification purposes.
Coverage Path 1	Enter the number of the coverage path which is defined in **Figure 23**

Table 13: Configuration IP Stations

```
add station 60007                                               Page   1 of   5
                                     STATION

Extension: 60007                         Lock Messages? n               BCC: 0
     Type: 2410                          Security Code:                  TN: 1
     Port: 01A0507                     Coverage Path 1: 999             COR: 1
     Name: extn 60007                  Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 2          Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 60007
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english

          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n

                                                          IP Video? n


                                               Customizable Labels? y
```

Figure 20: Digital Station Screen

3.1.6. Configure Functional DDI Hunt Groups

Avaya Communication Manager communicates with VIP 2000 using the ISDN call facility offered by the PRI interface interconnecting these components. VIP 2000 interprets the diverting-party extensions (hereafter referred to as "DDI numbers") for calls that it receives from its E1 trunk connected to Avaya Communication Manager as function codes. VIP 2000 can be configured to perform specific functions based on these DDI numbers. The "switchtable" configuration file shown in **Figure 26** can be configured to activate services based on the DDI number. When a call is received by VIP 2000, the "switchtable" is examined "from the bottom up", looking for an entry with a "C-num" value which matches the DDI number for the call. If no matching entry is found, the top entry ("vmail", the voicemail service) is activated to process the call. The voicemail service also processes the VIP 2000 "call tree" which is managed by the VIP 2000 Call Director, as described in **Section 3.3.2**.

Thus, VIP 2000 voicemail and Call Director operations can be initiated by Avaya Communication Manager by calling to VIP 2000 via the E1 trunk with the DDI number which corresponds to the operation to be performed. This can be configured in Avaya Communication Manager by allocating a hunt group for each VIP 2000 operation which is to be performed and giving it an extension contained within the local dial plan so that it can be called by local users. The hunt group "Message Center" parameter is set to "qsig-mwi" and the "Voice Mail Number" parameter to the DDI of the service to be called, prepended by the AAR prefix used to select the VIP 2000 routing pattern. This routing pattern removes the prefix prior to routing the call to VIP 2000.

Use the **add hunt-group** command to create hunt a group for each VIP 2000 DDI facility shown in **Table 2**, using the values also contained in that table. **Figure 21** illustrates the settings for the first entry in **Table 2**. Repeat this operation for the remainder of the entries in the **Table 2**.

Parameter	Usage
Group Name (p.1)	Any alphanumeric string can be used as a Group Name.
Group Extension (p.1)	Use an unused extension which is compatible with the dial plan. This number can be dialed by users to activate the VIP 2000 functions.
Message Center (p.2)	Specify "qsig-mwi".
Voice Mail Number (p.2)	Compose a number consisting of "78" which is used to route calls to the VIP 2000 trunk via aar (see **Figure 7**) followed by the DDI code which is used by VIP 2000 to match "C-num" entries in the "switchtable", as shown in **Figure 26**.

Table 14: DDI Hunt Groups

```
change hunt-group 55                                            Page   1 of  60
                                  HUNT GROUP

            Group Number: 55                                ACD? n
              Group Name: A-Num Route                     Queue? n
         Group Extension: 65555                          Vector? n
              Group Type: ucd-mia                  Coverage Path:
                      TN: 1         Night Service Destination:
                     COR: 1                  MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display:
```

Figure 21: DDI Hunt Group Screen, Page 1

```
change hunt-group 55                                            Page   2 of  60
                                  HUNT GROUP

                       LWC Reception: none           AUDIX Name:



                      Message Center: qsig-mwi
                     Send Reroute Request: y
                   Voice Mail Number: 785555
Routing Digits (e.g. AAR/ARS Access Code):              Provide Ringback? n
                TSC per MWI Interrogation? n
```

Figure 22: DDI Hunt Group Screen, Page 2

3.1.7. Configure Voicemail Coverage

Use the **add coverage-path** command to create a coverage path to route calls to voicemail.

Parameter	Usage
COVERAGE CRITERIA	Assign the coverage criteria as required for users.
COVERAGE POINTS	Enter "h55" to assign a coverage point to the hunt group (see **Section 3.1.6**) to serve as the voicemail interface to VIP 2000.

Table 15: Voicemail Coverage Path Parameters

```
change coverage path 999                                        Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 999
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?             y              y
               Busy?             y              y
        Don't Answer?            y              y        Number of Rings: 2
                All?             n              n
 DND/SAC/Goto Cover?             n              n
   Holiday Coverage?             n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h55            Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

Figure 23: Voicemail Coverage Path

3.2. Configure VIP 2000 Server

Insert the VIP 2000 software installation CD. When the installation wizard starts, click "Install the VIP 2000 Database" and accept all the default settings.

3.2.1. Asterisk directory

The files shown below are installed in the directory /etc/asterisk as part of the installation process.

Those parameters which must be manually edited are show in the following table.

Parameter	Usage
LP_FORMATS	Enter “pcm64a,alaw” to specify the G.711A codec.
LP_NUMBERPLAN	(1=ISDN)
LP_NUMBERTYPE	(0=Unknown)
LP_PABX_TYPE	QSIG: decode both EMCA/ISO, send ISO

Table 16: wtlp_config.avaya_qsig Configuration Parameters

```
[general]
LP_ERROR_DESTS=taf
LP_TRACE_DESTS=taf
LP_TRACE=0xffff
LP_SERVICE=switch
LP_SERVICE_CALLEND=tv_eos
LP_FORMATS=pcm64a,alaw
LP_NATIONALITY=se
LP_NUMBERPLAN=![0-9]{6,40}!1!,1
LP_NUMBERTYPE=![0-9]{6,40}!2!,0
LP_BTRANSFORMS=!^(9999999999.*)$!@\1@local_start!,!^([0-9]{6,40})!00\1!,!^([^@].
*)!g1/\1!,!^@(.*)$!\1!
LP_BTRANSFORMS_TECH=!^(.*)/!\1!,!^9999999999(.*)$!Local!,Zap
LP_BTRANSFORMS_BYPASS=!^([0-9]{1,5})$!*60*\1#!,!(.*)!g1/\1!
LP_BTRANSFORMS_BYPASS_TECH=!^(.*)/!\1!,Zap
LP_PABX_TYPE=QSIG/ISO
LP_LICENSE_MGR=lm
LP_DDI_TOTAL_MAXTIME=10000
LP_DDI_VALID=[0-9]{4}
LP_NETWORK_FLAGS=1
LP_ANUMBER=2000
LP_CALLERNAME=VIP2000
LP_LOCAL_NUMBERPLAN=1
LP_LOCAL_NUMBERTYPE=2
LP_SOCK_CHUNKSIZE=160
LP_SOCK_PREFETCH_CHUNKS=50
LP_LOGFILE=/u/vip2000/logs/lp_logfile%m
LP_ACCEPT_POSTDELAY=500
LP_ACCEPT_PREDELAY=1000
LP_DUMMY_CHANNEL_TECH=Zap
LP_DUMMY_CHANNEL_PHONENR=g1/1234
LP_ZAP_CHANNEL_EXCLUSIVE=NO
LP_ALCATEL_QSIG_MEW_TYPE_OF_SERVER=0
LP_ALCATEL_QSIG_MEW_MESSAGE_CAT=0
LP_ALCATEL_QSIG_MEW_CALLING_NR=3999
LP_ALCATEL_QSIG_MEW_CALLING_PLAN=0x0
LP_ALCATEL_QSIG_MEW_CALLING_PRES=0x0
LP_ALCATEL_QSIG_MEW_CALLED_PLAN=0x0
LP_ALCATEL_QSIG_MEW_CALLED_PRES=0x0
LP_ROUTEOPT_ECHO_DELAY_MS=0
LP_USE_FALLBACK=YES
LP_INTRUSION_PROTECTION_LEVEL_UUI=1
LP_ANALOG_HOLD=YES
LP_DIALSTRING_HOLD=&ww
LP_DIALSTRING_RETRIEVE=ww&ww
LP_DIALSTRING_TRANSFER=H
LP_CA_LVL3_CONFIRM_PHRASE=/u/vip2000/phrases/se/alaw/voicemail/press_hash_pause.pcm64a
LP_CA_LVL1_MAXTIME=2500
```

Figure 24: wtlp_config.avaya_qsig Configuration File

The values for the Zapata.config_avaya_qsig configuration file, shown in the following table must be manually edited.

Parameter	Usage
group	This specifies the first trunk group.
switchtype	Set this parameter to “qsig”
signalling	Set this parameter to “pri_cpe” to specify “pri” for primary rate, and “cpe” to specify “peer-slave”.
context	Specify “zapata_avaya_qsig”.
channel	Specify “1-15, 17-31” to indicate which B channels are to be used by the trunk.

Table 17: Configuration Zapata.conf_avaya_qsig

```
group = 1
switchtype=qsig
signalling=pri_cpe
context=zapata_avaya_qsig
channel => 1-15,17-31
channel => 32-46,48-62
channel => 63-77,79-93
channel => 94-108,110-124
```

Figure 25: Zapata.conf_avaya_qsig Configuration File

3.2.2. Config directory

The “config” directory is contained in the directory into which VIP 2000 was installed.
The “switchtable” file is used to specify what VIP 2000 is to do with calls which it receives. The diverting party number for incoming calls is matched against the “C-num” entries of the “switchtable”. Upon receiving a call, VIP 2000 examines the “C-num” entries in the table, from bottom to top. When a matching entry is found, the program indicated by the “Service to call” entry is activated to process the call. If no match is found, the “top” entry in the table is used. The following table describes the function of the “Service to call” application for each of the entries.

Service to call	Usage
Vmail	This is the voicemail service, which also processes user-defined Call Flow trees.
Reject	The “Reject” service is used to terminate a call without alerting. This can be done by transferring the call to “2000”.
Mobile_int	Call deflection to DIVNUM parameter. This parameter is used to deflect calls via QSIG path replacement.

Table 18: Configuration for switchable File

```
#------------------------------------------------------------------------------
# time | In  | A-  | B-  | C-   | Typ | Org | Service  | Args |
#      | Out | num | num | num  |     |     | to call  |      |
#------------------------------------------------------------------------------
 *     *     [.]*  [.]*  [.]*   *     *     vmail      "-c I <ANR> <BNR> <CNR> <TYPE> <ORIGIN>"
 *     *     [.]*  [.]*  ^2000$ *     *     reject     "-a 2 busytone.pcm64a"
 *     *     [.]*  [.]*  ^4321$ *     *     mobile_int "-c I <ANR> <BNR> <CNR> <TYPE>  external
-a ASSOC=assoc_mobile:ACTION=imm_divert:ERR=2:ANSW=NO:BTYPE=MOBEXT:DIVNUM=60093"
```

Figure 26: switchtable Configuration File

3.2.3. wanpipe2.conf Configuration File

The /etc/wanpipe/wanpipe2.conf is an editable text file which contains parameters for the QSIG link to Avaya Communication Manager, as described in the table below.

Section	Parameter	Value	Usage
wanpipe2	CommPort	PRI	Primary rate interface.
	FE_MEDIA	E1	E1 interface.
	FE_LCODE	HDB3	Specify the protocol to be the same as the “Line Coding” parameter in **Figure 10**.
	FE_FRAME	NCRC4	This value indicates “No CRC” which must be the same as the “CRC?” parameter in **Figure 10**.
	LBO	120OH	Impedance.
	TE_SIG_MODE	CCS	“Common Channel Signaling”, required for QSIG
	TDMV_DCHAN	16	“D” channel port. This must match the value configured for the “Primary D-Channel” parameter in **Figure 17**.
w2g1	ACTIVE_CH	ALL	Turn on all B channels
	TDMV_ECHO_OFF	NO	Turn off SW echo cancellation.
	TDMV_HWEC	YES	Turn on HW echo cancellation.

Table 19: wanpipe2.conf Parameters

```
 [interfaces]
w2g1 = wanpipe2, , TDM_VOICE, Comment
[wanpipe2]
CARD_TYPE       = AFT
S514CPU         = A
CommPort        = PRI
AUTO_PCISLOT    = NO
PCISLOT         = 9
PCIBUS          = 5
FE_MEDIA        = E1
FE_LCODE        = HDB3
FE_FRAME        = NCRC4
FE_LINE         = 2
TE_CLOCK        = NORMAL
TE_REF_CLOCK    = 0
TE_HIGHIMPEDANCE        = NO
LBO             = 120OH
TE_SIG_MODE     = CCS
FE_TXTRISTATE   = NO
MTU             = 1500
UDPPORT         = 9000
TTL             = 255
IGNORE_FRONT_END = NO
TDMV_SPAN       = 2
TDMV_DCHAN      = 16
[w2g1]
ACTIVE_CH       = ALL
TDMV_ECHO_OFF   = NO
TDMV_HWEC       = YES
```

Figure 27: wanpipe2.conf Configuration File

3.3. VIP 2000 Administration

To start the VIP 2000 administration facility, browse to the URL "http://<VIP2000 IP Address>/cgi-bin/WRAP/r8-sysadmin.html". When the login dialog appears, enter an appropriate username and password, and click "Log in".

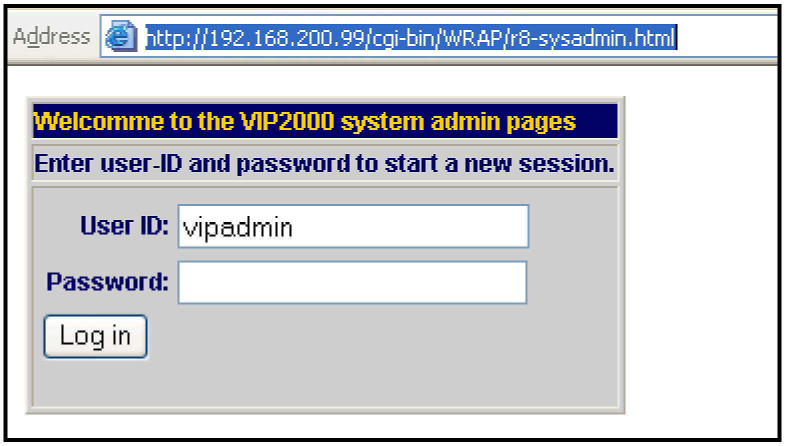

Figure 28: VIP Admin Login Dialog

3.3.1. Add Extensions

From the top-level menu, click “Extensions” from the left frame.

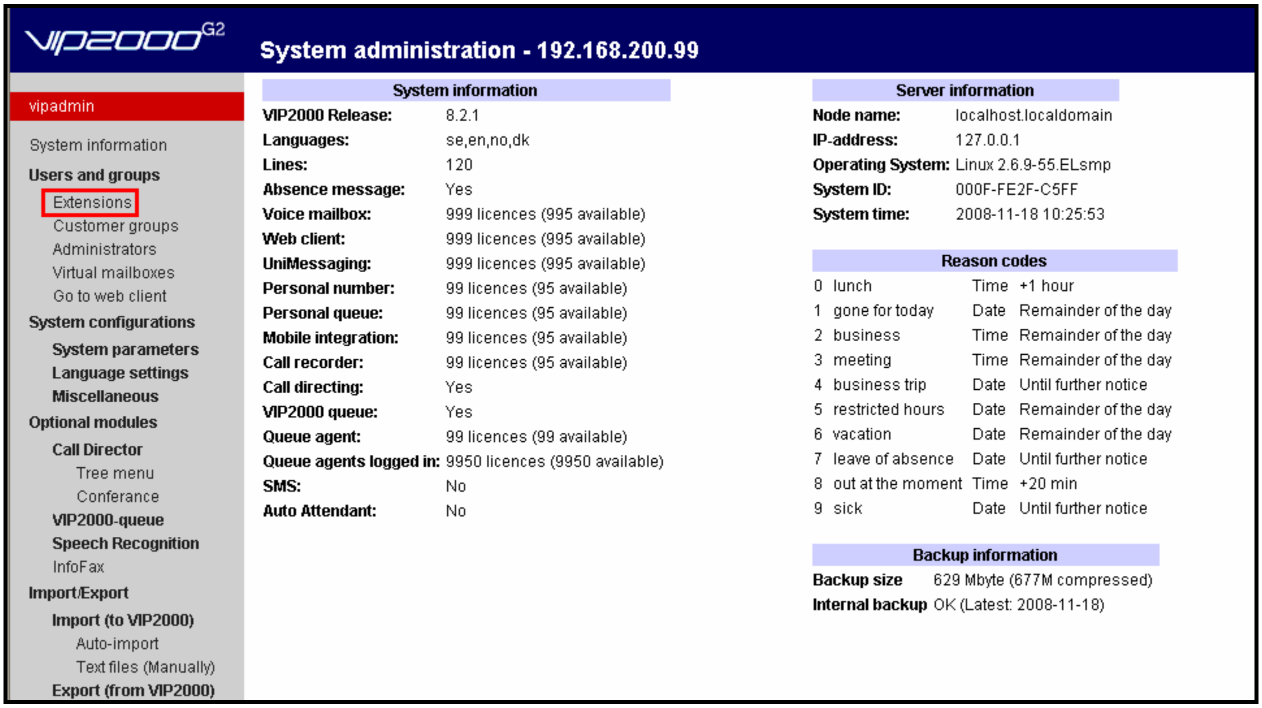

Figure 29: Administer Extensions

Click “Add”.

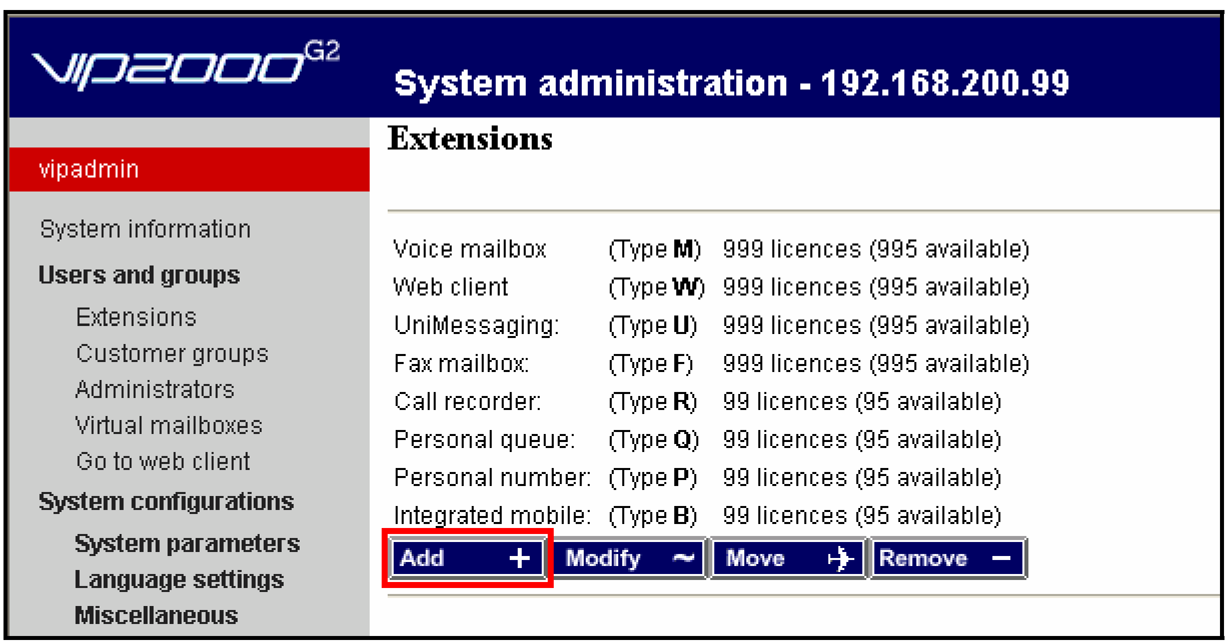

Figure 30: Add Extensions

Add an extension for each of the functions shown in **Table 2**, using the settings shown below.

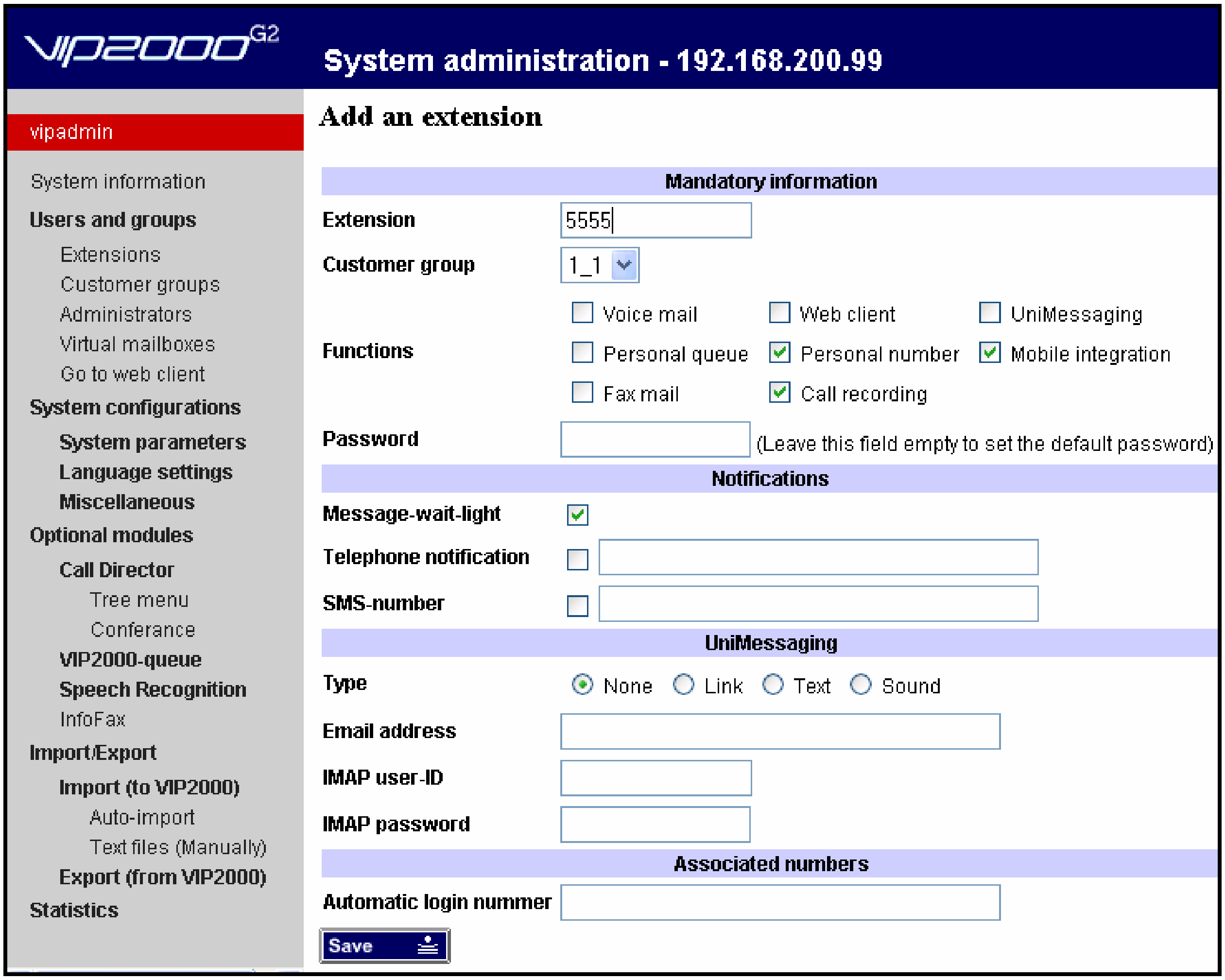

Figure 31: Add Function Extension

Add a user extension for each of the users shown in **Table 1**, using the settings shown below.

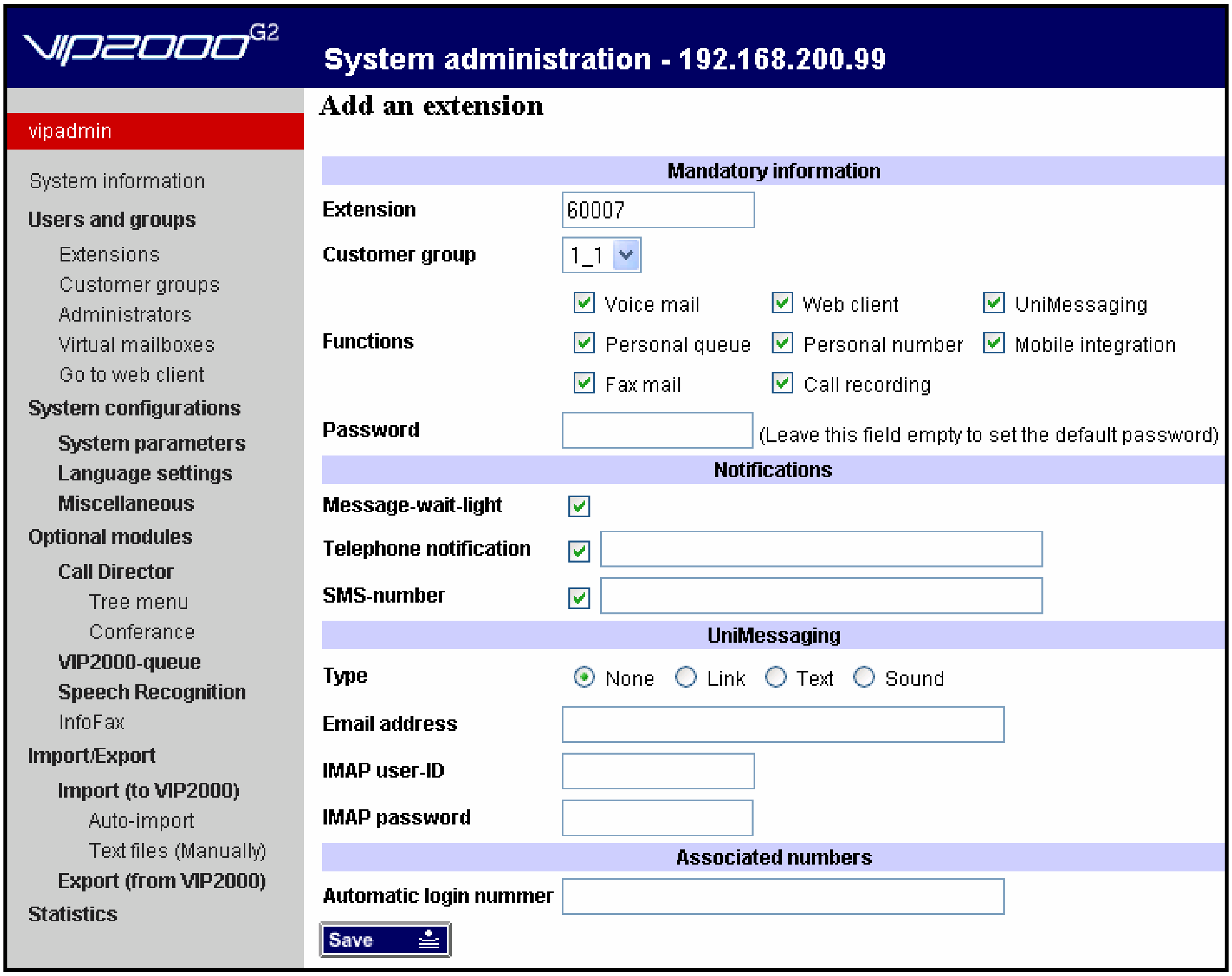

Figure 32: Add User Extension

3.3.2. Configure Call Director

As described in **Section 3.1.6,** the DDI number associated with calls received by VIP 2000 is used as a function code. This code is used to select a program from the “switchtable”, described in **Figure 26,** to process the call. If no entry for the DDI number is contained in the table, the program for the first entry, vmail (the voicemail application), is selected to process the call. Embedded within the VIP 2000 voicemail application is the Call Director.

The Call Director is a component of the VIP 2000 voicemail facility “vmail”. The “vmail” facility uses the DDI associated with redirecting number of calls that it receives via its E1 trunk from Avaya Communication Manager to identify the purpose of the calls that it receives. Calls for DDI “0077” have been sent to “coverage” due to absence or another reason which prevented the called party from answering the call. Such calls will be provided voicemail service by “vmail”.

Other DDI values are examined by the VIP 2000 Call Director to determine if they correspond to user-defined function codes. Such DDI codes can be associated with corresponding extensions by Avaya Communication Manager, as described in **Section 3.1.6**. These DDI codes must be configured by the Call Director administration facility.

Click “Call Director” -> “Tree menu” from the left frame.

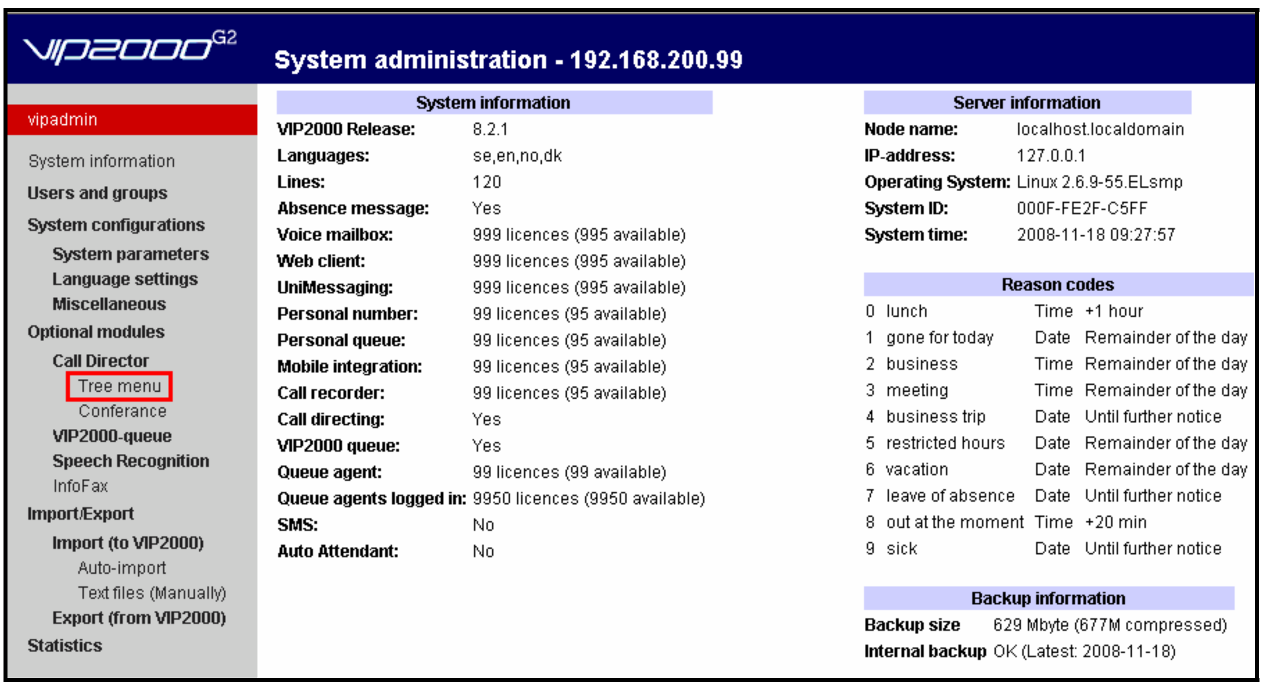

Figure 33: VIP Admin Introductory Screen

The remainder of this section of the document describes how to configure Dialog Director to associate DDI codes with specific functions that it is to perform.

3.3.2.1 Add Conference Function

Enter the "Number" of the DDI assigned to the "Conference" function in **Section 3.1.6** and click "Create Node".

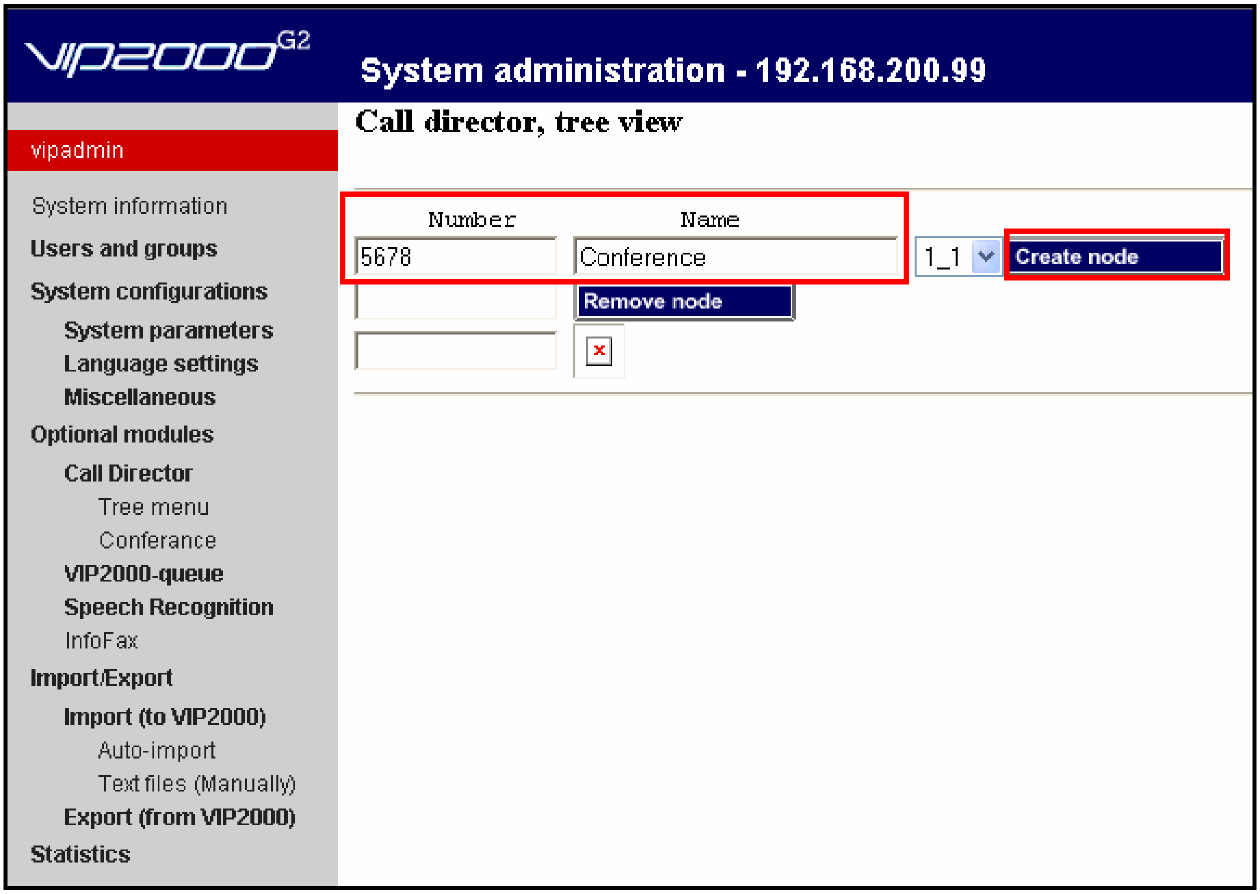

Figure 34: Call Director Screen

Select “Conference” from the drop-down menu corresponding to “button 0” and click “Update node”. On completion, click “Tree menu”.

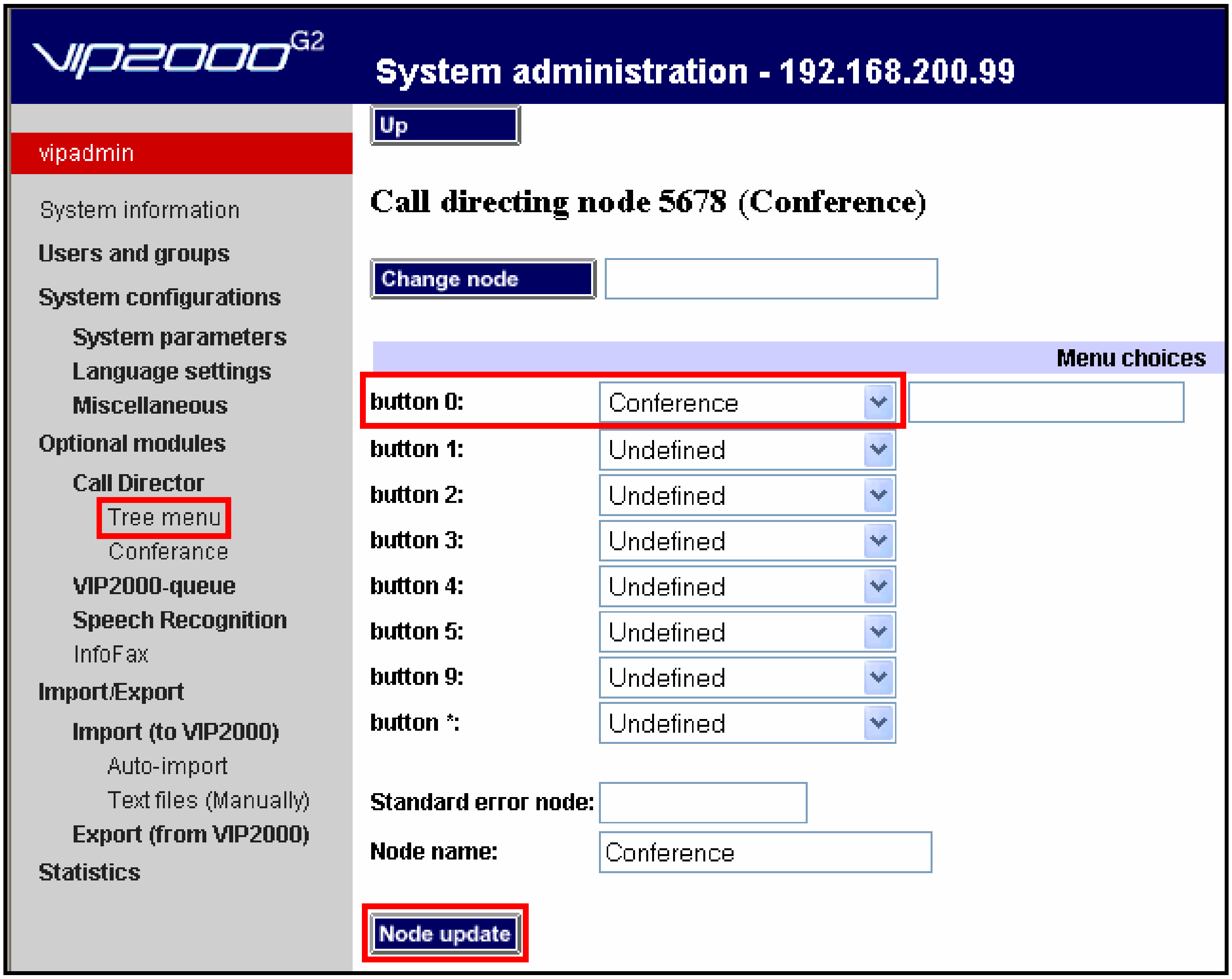

Figure 35: Create Conference

3.3.2.2 Add “A-Num” Routing Function

This routing function allows calls to be directed to specific destinations dependent on the number of the calling party. In order to use this function, a text file must be imported into VIP 2000 which indicates which Call Director node is to be activated when a call from a specific number is received. The file contains a text line for each possible destination to which calls can be directed. The first entry on each line is the Call Director node to which the call is to be directed, and the second entry the calling party number. The last entry contains a single number which is the Call Director node to which the call is to be directed if none of the other entries match the calling party number. Lines beginning with “#” are ignored. The Call Director node number is identical to the DDI assigned to the function.

A VIP 2000 Import operation can be performed from the top level VIP 2000 menu item “Import/Export” -> “Import (to VIP 2000)” -> “Text files (manually)”.

The following file causes calls from Frankfurt to be directed to node “1234” and calls from the UK to be directed to node “5678”.

```
# Calls from Frankfurt
1234   069

# Calls from England
5678   0044

#Otherwise
9999
```

Figure 36: A-Num Route Configuration File

After the A-Num configuration file has been imported, specify a Call Director node which is to be used for this function. Enter the “Number” of the DDI assigned to the “A-Num Route” function in **Table 2** and click “Create Node”.

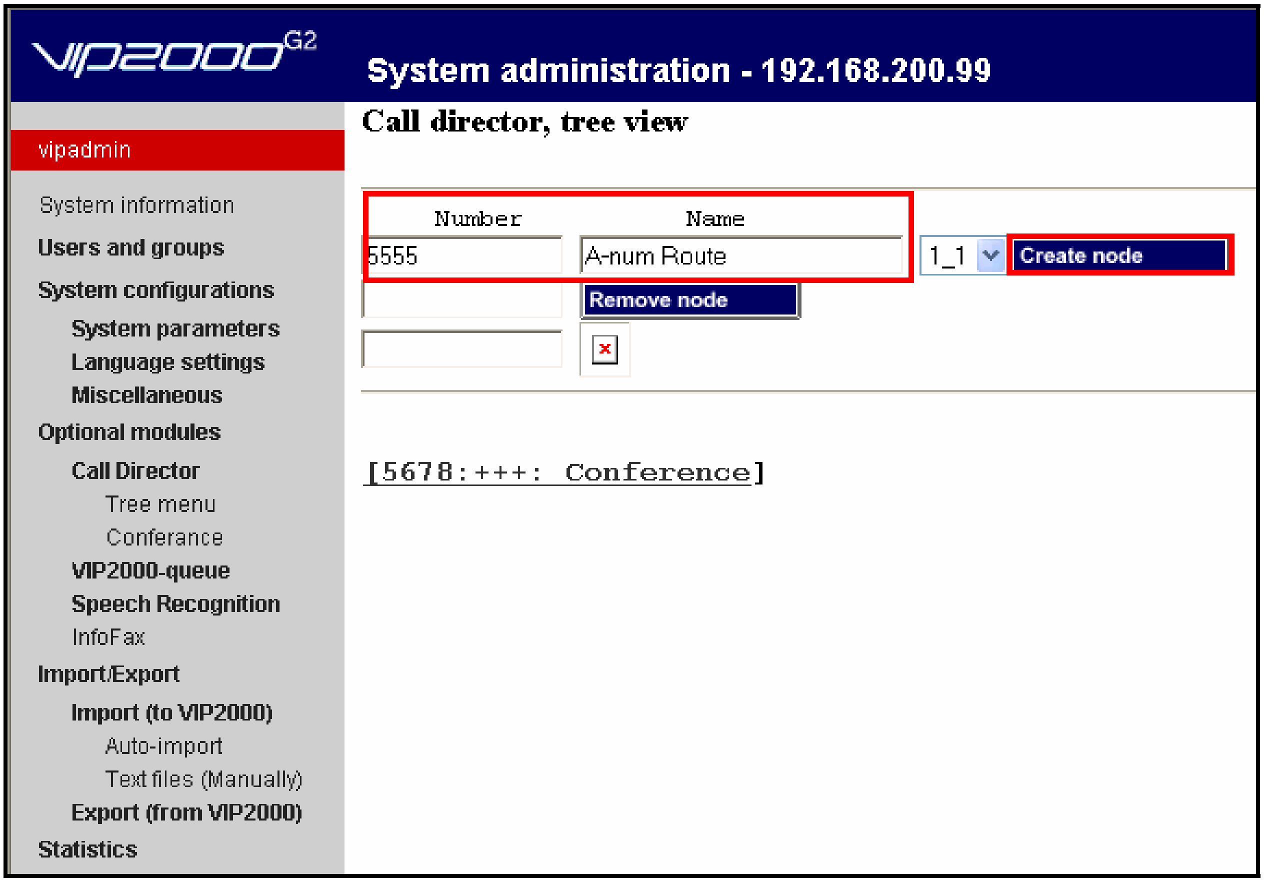

Figure 37: Create A-Num Routing Function

After the Call Director nodes have been defined, the Tree Menu should appear as shown below.

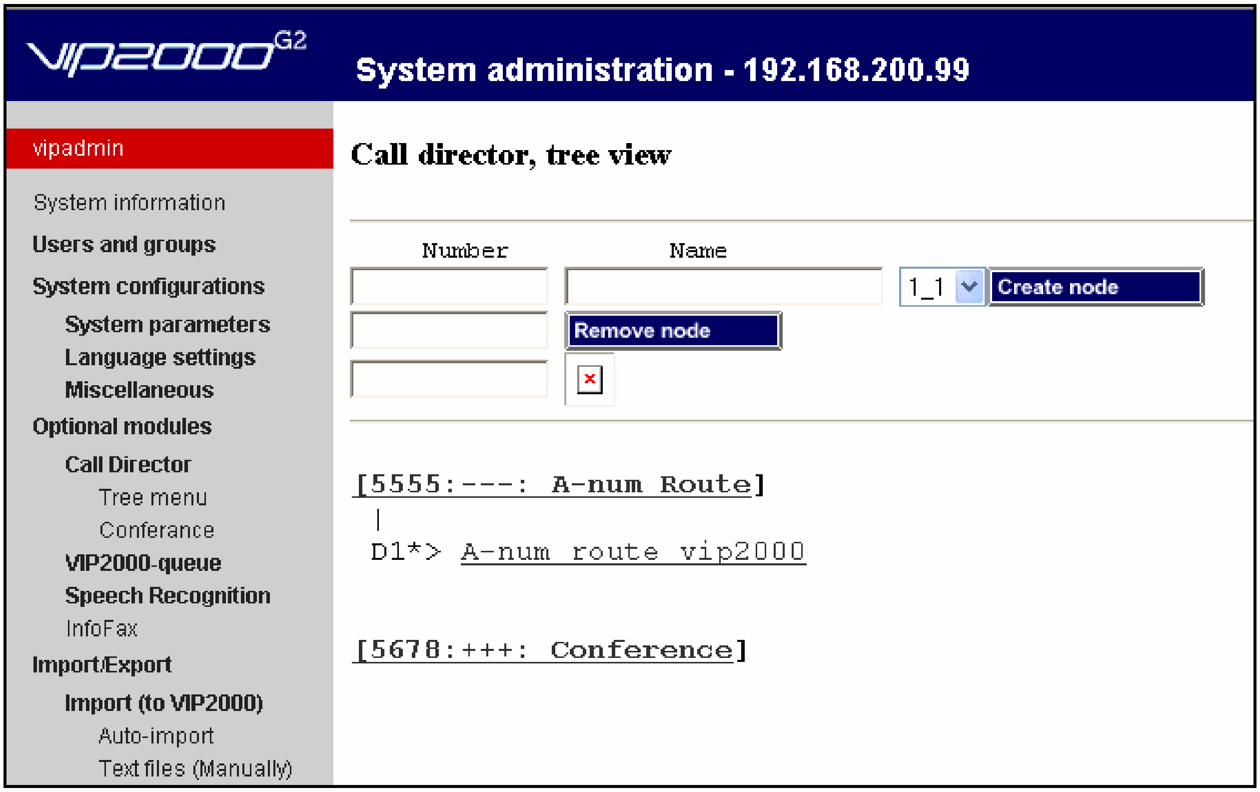

Figure 38: Create A-Num Routing Function

4. Interoperability Compliance Testing

The objective of the compliance testing done on the Visionutveckling VIP 2000 product was to verify that it is compatible with Avaya Communication Manager. This includes verifying that the essential VIP 2000 features function properly when used with Avaya Communication Manager, and that Avaya Communication Manager features are not hindered by the interaction with VIP 2000. Furthermore, VIP 2000's ability to recover from failures was verified.

4.1. General Test Approach

The test method employed can be described as follows:

- Avaya Communication Manager was configured to support various local IP telephones, as well as a networked PBX connection and a PSTN connection.
- A PSTN interface was attached to Avaya Communication Manager, which was used to communicate with external telephones.
- The following test scenarios were used to test the various VIP 2000 features:
  - Correct voicemail response for coverage from busy user
  - Correct voicemail response for coverage no answer
  - Correct voicemail response for unconditional forward

    - Queue call waiting for busy user
    - Conferencing via Call Tree
    - A-Number routing via Call Tree
    - Immediate diversion via QSIG
    - Call reject
    - Break-through to diverted party
    - Call route optimization via QSIG path replacement
- VIP 2000's robustness was tested by verifying its ability to recover from interruptions to its LAN connection between the VIP 2000 and the network as well as its E1 connection to Avaya Communication Manager
- VIP 2000's robustness was further tested by verifying its ability to start automatically.

All testing was performed manually. The tests were all functional in nature, and no performance testing was done.

4.2. Test Results

All tests which were performed produced the expected result.

5. Verification Steps

The following steps can be performed to verify the correct installation and configuration of VIP 2000:

- Use the Avaya Communication Manager SAT function "status trunk" command to verify that all of the members of the E1 trunk to VIP 2000 are in "in-service/idle" condition.
- Use the Avaya Communication Manager SAT function "status signaling-group" command to verify that all VIP 2000 signaling group is "in-service".
- Verify that it is possible to call the VIP 2000 voicemail number and receive the voicemail greeting.

6. Support

Support for VIP 2000 is available at:

Visionutveckling
support@visionutveckling.se
Phone: +46 303 389 000
Fax: +46 303 72 92 60

7. References

[1] *Administrator Guide for Avaya Communication Manager*, January 2008, Issue 4.0, Document Number 03-300509.

[2] *Feature Description and Implementation for Avaya Communication Manager*, January 2008, Issue 6, Document Number 555-245-205.

[3] *4600 Series IP Telephone LAN Administrator Guide*, October 2007, Issue 7, Document Number 555-233-507.

[4] VIP 2000 Brochure, December 2008. Available at http://www.vision8020.se/misc/Vision%208020_eng_2008-11.pdf

8. Conclusion

These Application Notes describe the compliance testing of the Visionutveckling VIP 2000 with Avaya Communication Manager. A detailed description of the configuration required for both the Avaya and the Visionutveckling equipment is documented within these Application Notes. The VIP 2000 passed all of the tests performed, which included both functional tests and tests to verify the ability of VIP 2000 to recover from failures.

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 DevConnect Program at devconnect@avaya.com.