Avaya Multi-Vantage / Communication Manager H.323 Configuration Guide For Use with AT&T IP Flexible Reach and IP Toll Free

Issue 2.14
3/15/2007

TABLE OF CONTENTS

1 Introduction

This document provides a configuration guide to assist Avaya Multi-Vantage administrators in connecting to AT&T IP Flexible Reach and the AT&T IP Toll Free Services.

2 Special Notes

Emergency 911/E911 Services Limitations

While AT&T IP Flexible Reach services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide in detail to understand these limitations and restrictions.

Fax Not Supported on G.729 Calls

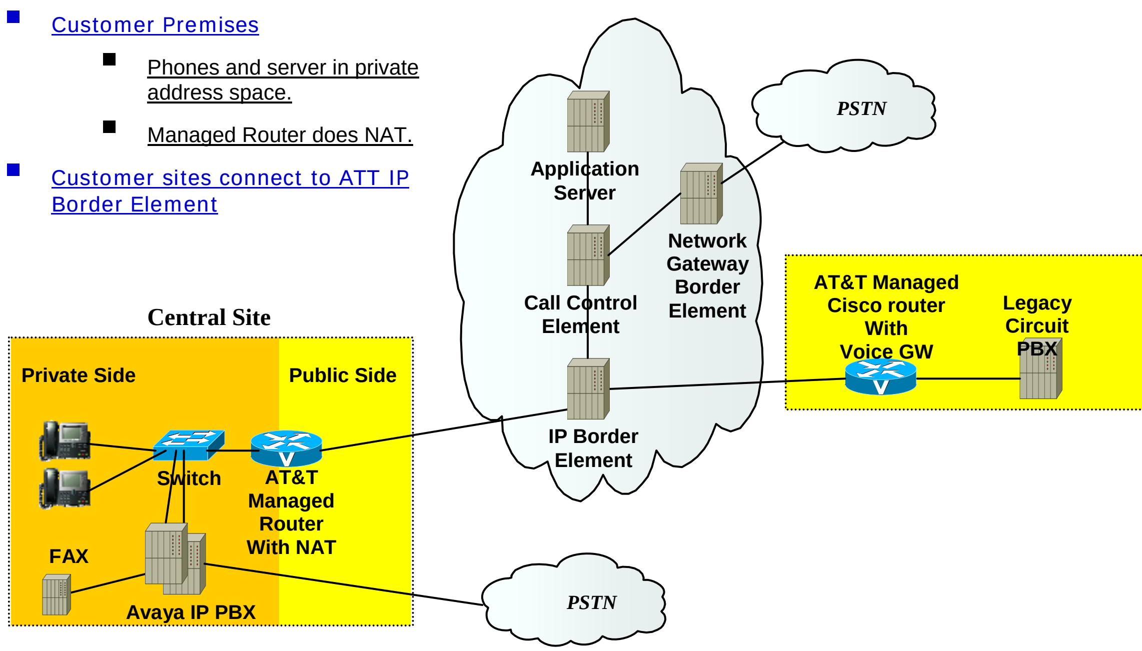
On G.729 voice calls, the AT&T IP Flexible Reach Service supports a change to the G.711 ulaw codec when fax is detected. This change is not currently supported by Avaya on H.323 trunks.

Customer IP Subnets Must Be IP Routable to the AT&T Network

Avaya does support the ability to connect non IP routable subnets. In theory, the Avaya MV/CM can act like a session border controller and connect IP subnets that otherwise cannot route to each other. However, this feature is not supported on incoming calls over an H.323 trunk configured with LRQ. Thus this is not supported with the AT&T network. **All customer subnets must be IP routable to the AT&T network.** In addition, it is recommended that the customer put all boards used in the AT&T call flow in the same “port network”. This eliminates any issues going between port networks.

3 Overview

This section provides a service overview of the Avaya Multi-Vantage integration with AT&T IP Flexible Reach and IP Toll Free Services.



The Avaya customer premises shall consist of the following components.

- Avaya IP phones – These phones use the Avaya proprietary H.323 protocol to communicate to the Avaya IP PBX for call feature and routing support. These phones can be connected to an Avaya switch that supplies in-line power to the phones.
- Avaya digital phones – These phones use the Avaya TDM protocol.
- Analog phones – These phones are standard analog phones.
- Avaya IP PBX – This is the central site unit which consists of the following.
    - Processor –
        - For IP Flexible Reach, the processor may be a Definity processor card (which does not support survivable remote sites) or an 8500/8700 Linux processor (which does support survivable remote sites). A standalone 8300 may also be used as a central site.

        - For IP Toll Free, the processor may be an 8500/8700 Linux processor (which does support survivable remote sites). The 8500/8700 supports integrated ACD/IVR functionality.
    - IP cards (CLAN for VOIP signaling and Medpro for VOIP media). These are used in the Definity and 8500/8700 environments. The 8300 has built in the VOIP signaling and media processing.
    - IPSI card – Provides communication between Definity shelf with IP and TDM/analog cards and 8500/8700 processor.
    - T1 voice card for connection to the local PSTN. Typically a TN464 series DS1 card is used in the 8500/8700/Definity carrier for the PSTN connection. The local PSTN connection shall be used for the following call scenarios.
        - Inbound PSTN calls destined to the Avaya phones.
        - Outbound local PSTN calls from the Avaya phones if desired.
        - Outbound local N11 (i.e. 411, 911) calls from Avaya phones phones.
- AT&T Managed Router (AT&T managed) – This is the router managed by AT&T. The router shall perform packet marking and QOS for voice. This router will support NAT (static NAT for Avaya Medpro card (media) and static NAT for the Avaya CLAN (signaling) in the VoPNT VPN configuration). IOS 12.4.6T6 was used in testing

The following routing scenarios are supported by the Avaya MV IP PBX and **DO NOT** use the AT&T Call Control. This scenario shall use the G.711 codec.

- Local AVAYA MV IP PBX phone to AVAYA MV IP PBX phone

The following routing scenarios are supported by the AVAYA MV IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729B, G729 or G.711 ulaw codec shall be used.

- AVAYA MV IP PBX phones to PSTN (domestic US and international).
- AVAYA MV IP PBX phones to legacy PBX site with Cisco gateway.
- Legacy PBX site with Cisco gateway to AVAYA MV IP PBX phones.
- AVAYA MV IP PBX phones at one AVAYA MV IP PBX IP PBX site to AVAYA MV IP PBX phones at another IP PBX site.

If the customer has subscribed to the AT&T IP Local feature of the IP Flexible Reach service, then the following routing scenarios are supported by the AVAYA MV IP PBX IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729B or G.711 ulaw codec may be used.

- Inbound PSTN to AVAYA MV IP PBX phone
- Outbound local PSTN calls from the AVAYA MV IP PBX phones.
- Outbound local N11 (i.e. 411, 911) calls from the AVAYA MV IP PBX phones.

4 Basic Configuration Guide

This specifies the Avaya Multi-Vantage screens that must be configured and updated to support the AT&T IP Flexible Reach and the IP Toll Free Services.

4.1 Avaya Multi-Vantage Version and Feature Requirements

The Avaya Multi-Vantage (MV) must be running:

- version 2.1 (build 414.1) (8300/8500/8700)
- version 3.0 (build 340.3) (8500/8700)
- version 3.1.2 (build 632.1) (8500/8700/8300)

The IP media processor card should be running vintage HW20 FW107 or later.

You can check the version of MV by running the “list configuration software-versions” command from the site administration interface. The results are shown next.

Version 2.1 example

```
SOFTWARE VERSION
   Memory Resident: R012x.01.1.414.1
     Disk Resident: R012x.01.1.414.1
```

Version 3.0 example

```
SOFTWARE VERSION
   Memory Resident: R013x.00.0.340.3
     Disk Resident: R013x.00.0.340.3
```

Version 3.1.2 example

```
SOFTWARE VERSION
   Memory Resident: R013x.01.2.632.1
     Disk Resident: R013x.01.2.632.1
```

You can check the vintage of the IP Media Processor card by running the “list config all” command from the site administration interface. You will need to page to the screen that shows the IP Media Processor card. The results are shown next.

```
                    SYSTEM CONFIGURATION

Board                                                     Assigned Ports
Number   Board Type               Code     Vintage    u=unassigned t=tti p=psa

01B11    CONTROL-LAN               TN799DP HW01 FW015 u  u  u  u  u  u  u  u
                                                      u  u  u  u  u  u  u  u
                                                      17
01B12    IP MEDIA PROCESSOR       TN2302AP HW20 FW107 01 02 03 04 05 06 07 08
```

At minimum, the following features and hardware will need to be present to support IP Trunking. Note that on Avaya G700 and G350 Gateways, CLAN and Media Processor resources are embedded as part of the standard product offer. Contact your Avaya sales team for traffic engineering and upgrade support in the event that you encounter problems with the following features and hardware.

- 178786 -- SA8507 - Support H245 DTMF Tones - This feature provides the capability to use the H.245 link to pass DTMF tones. In order to make sure that this feature has been installed, run the “display system-parameters special-applications” command and make sure that the output contains the following entry.

  (SA8507) - H245 Support With Other Vendors? y

- 151423 -- Control LAN (C-LAN) Circuit Pack (TN799DP or later) The Control-LAN (C-LAN) circuit pack, TN799DP, provides call control for all H.323-based IP endpoints connected to the S8500. The TN799DP contains programmable firmware and connects to the LAN at 10/100 Mbps. The number of C-LANs required depends on the number of devices connected, which consume "sockets," and whether other IP-based devices are employed. In the latter case, it may be advantageous to segregate IP voice control traffic from device control traffic as a safety measure. The C-LAN differs from an IP Media Processor in that the former controls the call and the latter provides the codecs used for the call's audio. To take advantage of downloadable firmware capability, customers must have at least one TN799DP or later C-LAN and access to the Internet.

- 150940 -- IP Media Processor (TN2302AP) Media Servers using IP-port network connectivity require resources on an IP Media Processor, TN2302AP circuit pack for inter-port network bearer communications. (This is not the case for direct connect configurations.) The TN2302AP Media Processor includes a 10/100 BaseT Ethernet interface to support H.323 endpoints for IP (Internet Protocol) trunks and H.323 end-points. At least one TN2302AP is required per IP connected port network, but the quantity may be higher, depending on the number of H.323 endpoints. The TN2302AP can perform echo cancellation, silence suppression, DTMF detection, and conferencing. V45 firmware and later also supports RTCP protocol, which is required for Avaya Integrated Management suite's VoIP Monitoring Manager, an IP traffic monitoring tool. The TN2302AP supports firmware download.

4.2 IP Nodes Names

A series of IP nodes names and their corresponding IP address must be configured for the following.

- Processor (8300)
- CLAN card (8500/8700/Definity)
- Prowler/Medpro card (8500/8700/Definity)
- AT&T IP Border Element

The command “change node-names ip” is used to configure node names for IP.

Sample nodes names and IP addresses are shown next.

PLEASE CONTACT YOUR CUSTOMER CARE REPRESENTATIVE FOR THE IPBE IP ADDRESSES FOR YOUR SPECIFIC PBX.

```
8300 configuration

    Name                 IP Address
procr                  172.16 .5  .12
attbe1                 12.194.180.4
attbe2                 12.194.177.4
```

```
8500/8700/Definity configuration

    Name                 IP Address
C-LAN                  172.16 .5  .10
PROWLER                172.16 .5  .11
attbe1                 12.194.180.4
attbe2                 12.194.177.4
```

4.3 IP Codec Set

An IP codec set must be configured. The following codecs (g.729b, g729 and g.711 ulaw) should be configured for use with the AT&T Border Element.

Note that for inbound calls to Avaya, Avaya picks the chosen codec based on the priority order configured in the Avaya codec set and not based on the order sent by the caller (the AT&T Network in this case). Thus if G.711 ulaw is preferred for inbound calls, then that codec should be specified as the highest priority in the codec set. Also note AT&T will always offer G.729 (without annex b) for calls from the AT&T Network to the PBX.

The command “change ip-codec-set <set #>” is used to configure the codec set. The parameters for the ip codec set are shown next.

```
    Codec Set: 1

    Audio      Silence      Frames   Packet
    Codec      Suppression  Per Pkt  Size(ms)
1: G.729B            n          2        20
2: G.729             n          2        20
3: G.711MU           n          2        20


                Mode                Redundancy
    FAX         off                  1
    Modem       off                  0
    TDD/TTY     US                   3
    Clear-channel   n                0
```

4.4 IP Network Region

An IP network region must be configured. The command "change ip-network-region <set #>" is used to configure the IP network region. The parameters for the IP network region are shown next. Key parameters are:

- Codec set – Set to the value of the codec to be used to connect to the AT&T BE.
- UDP port min – Set to 16384
- UDP port max – Set to 32767
- IP audio hairpining – Set to n since NAT is being used.
- Inter-region IP-IP Direct Audio – Set to no since shuffling (aka direct audio) is not supported.

With direct audio set to no, all audio packets are passed through the medpro card. **In this mode, there is a limitation on the number of simultaneous calls per medpro card. The following table provides the call limits by medpro card type.**

Medpro card type	G.711 call limit	G.729 call limit
TN2302AP	32	16
TN2602AP - 80 channel	40	40
TN2602AP - 320 channel	160	140

```
     Name: NET REGION 1
                                Intra-region IP-IP Direct Audio: no
AUDIO PARAMETERS                Inter-region IP-IP Direct Audio: no
   Codec Set: 1                            IP Audio Hairpinning? n
UDP Port Min: 16384
UDP Port Max: 32767                      RTCP Reporting Enabled? y
                                 RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS           Use Default Server Parameters? y
 Call Control PHB Value: 34
        Audio PHB Value: 46
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6      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
```

4.5 IP Interfaces

The IP interfaces can be viewed with the "list ip-interface" command. The IP interface can be changed using the "change ip-interface" command with the type field. The following IP interfaces are required.

- 8300 – The IP interface of the processor is required (see below). The network region must be one of the network regions previously defined.

```
ON Type  Slot  Code Sfx Node Name       Subnet Mask     Gateway Address Rgn
VLAN
-- ----  ----  ---- --- --------------- --------------- --------------- --- ---
-
y PROCR  01A00 ?      ? 172.16.5.12     255.255.255.0   172.16.5.1      1   0
```

- 8500/8700/Definity – The IP interface of the C-LAN and Medpro/Prowler cards are required (see below). The network region must be one of the network regions previously defined. The network regions for the 2 cards must be the same.

```
ON Type  Slot  Code Sfx Node Name       Subnet Mask     Gateway Address Rgn
VLAN
-- ----  ----  ---- --- --------------- --------------- --------------- --- ---
-
y C-LAN  01C08 TN799  D C-LAN           255.255.255.0   172.16.6.1      1   n
                        172.16.6.112
y MEDPRO 01C07 TN2302   PROWLER         255.255.255.0   172.16.6.1      1   n
                        172.16.6.113
```

4.6 Signaling Group

One signaling group must be configured for each AT&T Border Element. The command "change signaling-group <signaling-group #>" is used to configure the signaling group. The parameters for the signaling group are shown next.

- Trunk group for channel selection – Set to trunk group used for incoming calls.
- Near-end node name – In the case of the 8500/8700/Definity, this is the node name of C-LAN card. In the case of the S8300, this is the node name of the S8300 processor.
- Near end list port – Set to 1719.
- Far end node name – Node name of AT&T BE.
- Far end listen port – Set to 1719.
- Far-end Network Region – Set to the appropriate network region.
- LRQ required – Set to y.
- H245 control addr on facility – Set to y. This is required to use H245 for DTMF.
- Bypass if IP threshold exceeded – Set to n. This prevents pings to the AT&T BE, These pings are not supported by the AT&T BE.
- DTMF over IP – Set to out of band.
- Direct IP-IP audio connection – Set to n.
- IP audio hairpinning – Set to n. If desired, this parameter may also be set to y. It should not affect connectivity to the AT&T network.

```
change signaling-group 1                                           Page   1 of   5
                                SIGNALING GROUP

 Group Number: 1             Group Type: h.323
                          Remote Office? n          Max number of NCA TSC: 0
                                    SBS? n           Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 1
          Supplementary Service Protocol: a


        Near-end Node Name: procr              Far-end Node Name: attbe1
      Near-end Listen Port: 1719             Far-end Listen Port: 1719
                                          Far-end Network Region: 1
              LRQ Required? y             Calls Share IP Signaling Connection? n
              RRQ Required? n                   H245 Control Addr On FACility? y
                                               Bypass If IP Threshold Exceeded? n

              DTMF over IP: out-of-band         Direct IP-IP Audio Connections? n
                                                         IP Audio Hairpinning? n
                                                Interworking Message: PROGress
```

```
change signaling-group 2                                       Page   1 of   5
                               SIGNALING GROUP

 Group Number: 2             Group Type: h.323
                          Remote Office? n         Max number of NCA TSC: 0
                                    SBS? n          Max number of CA TSC: 0
                                                  Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 2
          Supplementary Service Protocol: a


        Near-end Node Name: procr             Far-end Node Name: attbe2
      Near-end Listen Port: 1719            Far-end Listen Port: 1719
                                         Far-end Network Region: 1
              LRQ Required? y             Calls Share IP Signaling Connection? n
              RRQ Required? n                  H245 Control Addr On FACility? y
                                             Bypass If IP Threshold Exceeded? n

              DTMF over IP: out-of-band        Direct IP-IP Audio Connections? n
                                                         IP Audio Hairpinning? n
                                              Interworking Message: PROGress
```

4.7 Trunk Group

A trunk group must be configured. The command “change trunk <trunk #>” is used to configure the trunk group. The parameters for the trunk group are shown next.

- Group Type – Set this to isdn.
- Direction – Set this to two-way.
- Carrier Medium – Set this to IP.
- Dial access – Set this to y.
- Service Type – Set this to tie.
- Codeset to send display – If “send name” is required, set codeset to 0.
- Send name – For the 8300, this can be set to y if required. However for the 8500/8700/Definity being used for the Cisco PIX firewall for NAT, this should be set to n. **These MV platforms (8500/8700/Definity) cannot send more than 260 bytes in a packet. If the send name is set to y, the call setup packet may be broken into 2 packets. Multi-packet messages cannot be NATed when using the Cisco PIX firewall resulting in call failures.**
- Send calling number – Can be set to y for all platforms if required.
- Group member assignments – The group member assignments must point to the appropriate signaling group (e.g. signaling group 1 for attbe1 and signaling group 2 for attbe2).

```
change trunk-group 1                                            Page   1 of  22
                      TRUNK GROUP

Group Number: 1                   Group Type: isdn         CDR Reports: y
  Group Name: OUTSIDE CALL               COR: 1        TN: 1        TAC: 201
   Direction: two-way        Outgoing Display? n        Carrier Medium: IP
 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


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: a     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical                     QSIG Value-Added? n
                                                   Digital Loss Group: 18
Calling Number - Delete:     Insert:                 Numbering Format:
              Bit Rate: 1200         Synchronization: async     Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

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

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

             Send UUI IE? y
               Send UCID? n
 Send Codeset 6/7 LAI IE? y



                     SBS? n  Network (Japan) Needs Connect Before Disconnect? n
```

```
change trunk-group 1                                            Page   6 of  22
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/5
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   5

       Port     Code Sfx Name        Night            Sig Grp
  1: T00001                                              1
  2: T00002                                              1
  3: T00003                                              1
  4: T00004                                              1
```

```
              1                                               Page    1 of   22
                         TRUNK GROUP

Group Number: 2                    Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1        TN: 1          TAC: 201
   Direction: two-way        Outgoing Display? n         Carrier Medium: IP
 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


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: a     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical                   QSIG Value-Added? n
                                                 Digital Loss Group: 18
Calling Number - Delete:     Insert:               Numbering Format:
              Bit Rate: 1200        Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

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

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

             Send UUI IE? y
               Send UCID? n
 Send Codeset 6/7 LAI IE? y



                     SBS? n  Network (Japan) Needs Connect Before Disconnect? n
```

```
change trunk-group 2                                                Page    3 of   22
                      INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called        Del  Insert           Per Call  Night
 Feature        Len      Number                              CPN/BN    Serv
```

```
change trunk-group 2                                            Page   6 of  22
                                TRUNK GROUP
                                     Administered Members (min/max):   1/5
GROUP MEMBER ASSIGNMENTS                 Total Administered Members:   5

       Port    Code Sfx Name         Night            Sig Grp
  1: T00001                                             2
  2: T00002                                             2
  3: T00003                                             2
  4: T00004                                             2
```

4.8 Route Pattern

Depending on the dial plan, a route pattern may need to be configured. Calls are mapped to a route pattern based on the outbound digit analysis of the switch (using ARS, AAR, etc). A sample AAR digit analysis entry is shown next. This table is accessed using the “list aar analysis” command.

```
                          AAR DIGIT ANALYSIS REPORT

              Dialed             Total         Route     Call      Node
              String           Min     Max    Pattern    Type     Number

        825                     7       19     57         aar
```

The parameters for the route pattern are shown next.

- Group number – Trunk group numbers listed in order of priority. This configuration specifies the alternate routing to the dual AT&T Border Elements.
- LAR – In the LAR column, “next” is specified for the first trunk specified (trunk 1). This enables “fail over” from trunk 1 to trunk 2.

```
             Pattern Number: 57   Pattern Name:
                                        Secure SIP? n
   Grp FRL NPA Pfx Hop Toll No.  Inserted                        DCS/ IXC
   No          Mrk Lmt List Del  Digits                          QSIG
                            Dgts                                 Intw
1: 1    0                    3                                   n    user
2: 2    0                    3                                   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                                          next
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
```

4.9 Calling Number Configuration

In order to send a calling number, the public-unknown-number screen must be used as shown next. This configuration may be required to support the Virtual Telephone Number (VTN) feature.

```
change public-unknown-numbering                                 Page   1 of   8
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                   Total
Total
Ext Ext    Trk       CPN            CPN Ext Ext    Trk       CPN            CPN
Len Code   Grp(s)    Prefix         Len Len Code   Grp(s)    Prefix         Len

 4 3                 732368          10
 4 3050              7323682059      10
```

As shown above, the calling number can be configured a number of different ways. In all cases, the rule applies to 4 digit extensions that start with “3”.

- The first example shows the use of a CN prefix which is pre-pended to the actual extension.
- The second example shows the mapping from an internal extension to a completely different public calling party number.

4.10 Incoming Call Routing on Virtual Telephone Number

When using the AT&T Virtual Telephone Number feature (VTN), the AT&T network will send the call to the PBX using a full E.164 public number. This number can be mapped to an internal extension on the following screens.

Version 2.1
In version 2.1, use the incoming call handling treatment on the trunk group screen as shown next. This example maps the public called number "7323682059" to the internal extension "3050" for internal routing.

```
change trunk-group 1                                          Page    3 of   22
                       INCOMING CALL HANDLING TREATMENT
 Service/          Called     Called         Del  Insert            Per Call   Night
 Feature            Len       Number                                CPN/BN     Serv
tie                 10        7323682059      10  3050
```

Version 3.0/3.1.2
In version 3.0/3.1.2, use the change inc-call-handling-trmt command screen as shown next. This example maps the public called number "7323682059" to the internal extension "3050" for internal routing.

```
change inc-call-handling-trmt trunk-group 1                  Page    1 of   30
                       INCOMING CALL HANDLING TREATMENT
 Service/          Called     Called         Del  Insert            Per Call   Night
 Feature            Len       Number                                CPN/BN     Serv
tie               10          7323682059     10   3050
```

5 ACD/IVR Configuration

This section provides software requirements and important configuration screens used to connect the AT&T IP Toll Free network to an Avaya ACD/IVR application. This section also provides an example of a simple ACD/IVR application. It does not provide all full documentation for the creation of an ACD/IVR application. Consult the appropriate Avaya documentation for a full description of ACD/IVR applications.

5.1 Avaya Multi-Vantage Feature Requirements

In order to the support the ACD/IVR functionality, the Avaya Call Center feature set must be supported. The following screens show the features available on the Avaya Call Center system used during lab testing.

```
display system-parameters customer-options                      Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 3.0

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


display system-parameters customer-options                      Page   7 of  11
                         CALL CENTER OPTIONAL FEATURES

           VDN of Origin Announcement? y                           VuStats? y
               VDN Return Destination? y         VuStats (G3V4 Enhanced)? y
```

In order to play announcements as part of the IVR, an announcement board is required as shown next.

```
list configuration all                                                  Page   4

                              SYSTEM CONFIGURATION

Board                                                           Assigned Ports
Number   Board Type                Code      Vintage     u=unassigned t=tti p=psa

01B05    VAL-ANNOUNCEMENT          TN2501AP HW03 FW007 01 02 03 04 05 06 07 08
                                                       09 10 11 12 13 14 15 16
                                                       17 18 19 20 21 22 23 24
                                                       25 26 27 28 29 30 31 32
                                                       33
```

5.2 Incoming Call Routing on ACD/IVR Calls

For incoming ACD/IVR calls, use the change inc-call-handling-trmt command screen as shown next. This example maps the called number string "000003010014" to the internal extension "3701" for internal routing. The "3701" number is a vector directory number (VDN; see next section) which then initiates the ACD/IVR application.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert            Per Call  Night
 Feature         Len       Number                              CPN/BN    Serv
tie               12    000003010014     12     3701
```

5.3 Vector Directory Number

The vector directory number is a called number which maps to a vector ("61" in this example). The vector implements the ACD/IVR application.

```
display vdn 3701                                                Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 3701
                                  Name: CAllWorkerDemo
                         Vector Number: 61
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                    TN: 1
                              Measured: none


         VDN of Origin Annc. Extension:
                             1st Skill:
                             2nd Skill:
                             3rd Skill:
```

5.4 Vector

This section shows an example of a vector that implements an ACD/IVR application. In this example, the first vector (i.e. "61") plays an announcement and collects a digit to determine the next vector in the application.

```
display vector 61                                               Page   1 of   3
                                  CALL VECTOR

    Number: 61              Name: AvayaCD2
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 collect      1    digits after announcement 5060     for none
03 goto vector 71 @step 1  if digits           =      1
04 goto vector 72 @step 1  if digits           =      2
05 goto vector 73 @step 1  if digits           =      3
06 busy
```

Continuing with the example, vector 71 (below) queues a call to a specific skill. A skill is assigned to an agent (see next section).

```
display vector 71                                               Page   1 of   3
                                  CALL VECTOR

    Number: 71              Name: Skill 1 Queue
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01
02
03 queue-to     skill 1    pri l
04 wait-time    10  secs hearing ringback
05 announcement 5017
06 wait-time    20  secs hearing music
07 goto step   5           if unconditionally
```

5.5 Hunt Group (aka Skill Set)

A hunt group (also referred to as a skill set) is configured as shown next. The screen shown next is a sample configuration for the skill 1 that is referenced in the vector.

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

            Group Number: 1                              ACD? y
              Group Name: Test Skill 1                 Queue? y
         Group Extension: 6001                        Vector? y
              Group Type: ead-mia
                      TN: 1
                     COR: 1                   MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: 1
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

5.6 Announcements

The screen shown next is a sample configuration for the announcements that are referenced in the vector.

```
display announcements                                         Page   1 of  16
                          ANNOUNCEMENTS/AUDIO SOURCES

Ann.                                                                 Group/
No.  Ext.     Type       COR TN   Name                        Q QLen Pr Rt Port


8    5017     integrated 1   1    ACDwaiting                  n NA    n  64 01B05
15   5060     integrated 1   1    CallWorkerDemo              n NA    n  64 01B05
```

5.7 Agents LoginID

This section is the continuation of the ACD/IVR example. The agent loginID configuration is used to define an agent login id and assign it to one or more skills. In this example, agent loginID 5001 is assigned a single skill (i.e. skill 1).

```
display agent-loginID 5001                                      Page   1 of   2
                                 AGENT LOGINID

                Login ID: 5001                                        AAS? n
                    Name: Test Agent 1                              AUDIX? n
                      TN: 1                                 LWC Reception: spe
                     COR: 1                        LWC Log External Calls? n
           Coverage Path:                      AUDIX Name for Messaging:
           Security Code:
                                               LoginID for ISDN Display? n
                                                                Password:
                                                 Password (enter again):
                                                             Auto Answer: station
                                                       MIA Across Skills: system
                                              ACW Agent Considered Idle: system
                                              Aux Work Reason Code Type: system
                                                Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system



display agent-loginID 5001                                      Page   2 of   2
                                 AGENT LOGINID
      Direct Agent Skill:
Call Handling Preference: skill-level              Local Call Preference? n

    SN      SL          SN      SL          SN      SL          SN      SL
 1: 1       1       16:                 31:                 46:
```

6 Troubleshooting

This section provides some tips about troubleshooting problems

6.1 List Trace

The “list trace” command is used to make sure an outgoing or incoming call is used the correct trunk group. The format of the command is:

list trace tac <tac # from the trunk group profile>