AT&T VOIP
Avaya Multi-Vantage 4.0
Configuration Guide
For Use with AT&T
IP Toll Free with IP Transfer Connect
(SIP Version)

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TABLE OF CONTENTS

1 Introduction

This document provides a configuration guide to assist Avaya Multi-Vantage (MV) / Communication Manager (CM) administrators in connecting to the AT&T IP Toll Free Service using SIP.

2 Special Notes

Fax Not Supported
Fax is not supported since fax testing was not done.

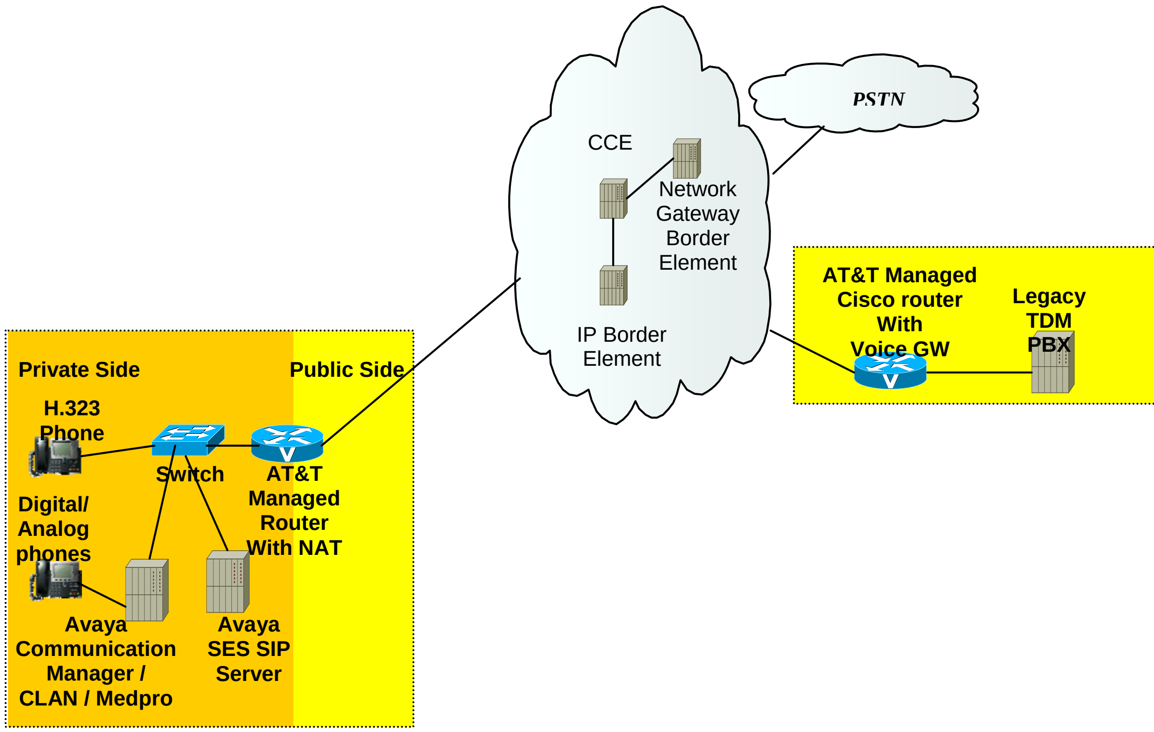
Avaya SIP IP Phones are Not Supported
Avaya SIP IP phones were not tested and are not supported. Avaya H.323 IP phones, digital phones and analog phones are supported. Note that most Avaya installations use H.323 IP phones.

Direct media (Shuffling) is Not Supported
Direct media (i.e. shuffling) is not supported.

G.726 not supported
G.726 has not yet been tested with Avaya.

3 Overview

This section provides a service overview of the Avaya Multi-Vantage integration with AT&T IP Toll Free using SIP. The components are shown next.



The Avaya customer premises shall consist of the following components.

- Avaya H.323 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. **Note that SIP phones are not supported.**
- Avaya digital phones – These phones use the Avaya TDM protocol.
- Analog phones – These phones are standard analog phones. Fax machines connect as an analog phone.
- Avaya IP PBX – This consists of the following.
    - Processor – This is an 8500/8700 Linux processor or 8300/8400 processor with GXXX gateway.
    - SIP Server – This server is known as the SES server. The server provides the SIP signaling interface to the AT&T network.

  - IP cards for 8500/8700 (CLAN for VOIP signaling and Medpro for VOIP media).
  - IPSI card – Provides communication between the card shelf and the 8500/8700 processor.
  - Analog ports for connection to fax machines.
- 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 static NAT for the Medpro card and static NAT for the Avaya SES (SIP signaling).

4 Configuration Guide

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

4.1 Generic Communication Manager Configuration

4.1.1 Version and Feature Requirements

The Avaya Communication Manager (CM) must be running version 4.0.

You can check the version of CM by running the "list configuration software-versions" command from the site administration interface.

```
    SOFTWARE VERSION
       Memory Resident: R014x.00.0.730.5
         Disk Resident: R014x.00.0.730.5
```

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

01C07    IP MEDIA PROCESSOR        TN2302AP HW20 FW107 01 02 03 04 05 06 07 08
01C08    CONTROL-LAN                TN799DP HW01 FW015 u  u  u  u  u  u  u  u
```

The recommended media card firmware and hardware vintage are available from Avaya at the following link.

http://support.avaya.com/elmodocs2/comm_mgr/CM_SW_FW_Compatibility_Matrix.doc#_Toc166034753

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."

- 151423 -- Control LAN (C-LAN) Circuit Pack (TN799DP or later) The Control-LAN (C-LAN) circuit pack, 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-- Media Servers using IP-port network connectivity require resources of an IP Media Processor; either the TN2302AP or TN2602AP circuit pack for inter-port network bearer communications. (This is not the case for direct connect configurations.) At least one TN2302AP (or TN2602) is required per IP connected port network, but the quantity may be higher, depending on the number of H.323 endpoints. These IP Media Processor cards perform echo cancellation, silence suppression, DTMF detection, conferencing. and supports RTCP protocol, which is required for Avaya Integrated Management suite's VoIP Monitoring Manager, an IP traffic monitoring tool. They support firmware download.

4.1.2 IP Nodes Names

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

- CLAN card (8500/8700)
- Prowler/Medpro card (8500/8700)
- SES server

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

Sample nodes names are shown next.

```
                        NODE NAMES

Type     Name                 IP Address
IP       C-LAN                172.16 .6  .112
IP       PROWLER              172.16 .6  .113
IP       ses                  172.16 .6  .118
```

4.1.3 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	t.38-standard	1
Modem	off	0
TDD/TTY	off	0
Clear-channel	n	0

4.1.4 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:

- Authoritative Domain: Set to”attccs.com”. Must match the domain in the SES.
- 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 hairpinning – Set to y or n.
- Intra-region IP-IP Direct Audio: no
- Inter-region IP-IP Direct Audio: no

When direct audio (i.e. shuffling) is disabled, all audio packets are passed through the 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	160

```
                                IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: attccs.com
    Name: Avaya Trial
                                Intra-region IP-IP Direct Audio: no
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: no
      Codec Set: 1                         IP Audio Hairpinning? y
   UDP Port Min: 16384
   UDP Port Max: 32767                   RTCP Reporting Enabled? y
DIFFSERV/TOS PARAMETERS          RTCP MONITOR SERVER PARAMETERS
 Call Control PHB Value: 34       Use Default Server Parameters? y
        Audio PHB Value: 1
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 0      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
```

4.1.5 IP Interfaces

The IP interfaces can be viewed with the “list ip-interface all” command.. The following IP interfaces are required.

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
                         IP-Address
-- ----   ----  ---- --- --------------- --------------- --------------- --- ---
-
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
```

4.1.6 Signaling Group

One signaling group must be configured for each SES. 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 – This is the node name of C-LAN card.
- Near end list port – Set to 5061.
- Far end node name – Leave this blank.
- Bypass if IP threshold exceeded – Set to n.
- DTMF over IP: Set to rtp-payload.
- Direct IP-IP audio connection – Set to n.
- IP audio hairpinning –This can be y or n.

```
                              SIGNALING GROUP

 Group Number: 1             Group Type: sip
                        Transport Method: tls






   Near-end Node Name: C-LAN                Far-end Node Name: ses
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
       Far-end Domain:

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? n
                                                       IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

4.1.7 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 SIP.
- Direction – Set this to two-way.
- Carrier Medium – Set this to IP.
- Dial access – Set this to n.
- Service Type – Set this to tie.

- Codeset to send display – If “send name” is required, set codeset to 0.
- Send name – This can be set to y if required.
- 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.

```
                                TRUNK GROUP

Group Number: 1                  Group Type: sip          CDR Reports: y
  Group Name: S8500 to SIP SES            COR: 1        TN: 1        TAC: 268
   Direction: two-way       Outgoing Display? n
 Dial Access? n               Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                                        Signaling Group: 1
                                                      Number of Members: 16
TRUNK PARAMETERS

     Unicode Name? y

                                             Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
```

```
TRUNK FEATURES
          ACA Assignment? n            Measured: internal
                                                          Maintenance Tests? y



                     Numbering Format: public



                                                 Replace Unavailable Numbers? n
```

```
                                TRUNK GROUP
                                      Administered Members (min/max):   1/16
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  16

       Port             Name
  1: T00420             S8500 to S
  2: T00421             S8500 to S
  3: T00422             S8500 to S
  4: T00423             S8500 to S
  5: T00424             S8500 to S
```

4.1.8 Incoming Call Routing

The AT&T network will send the call to the PBX using the IPTF DNIS (also known as signaled digits outpulsed or SDOP) as provided by AT&T Customer Care. Note that the DNIS will always be pre-pended with 5 leading zeroes (i.e. “00000”). This number can be mapped to an internal extension on the following screens.

Use the change inc-call-handling-trmt command screen as shown next. This example maps the DNIS “0017” (i.e. 000000017 after the zeroes are added) to the internal extension “3400” 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             9  000000017        9   3400
```

4.2 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.

4.2.1 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
```

4.2.2 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 DNIS plus 5 zeroes “000000027” 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                9  000000027        9   3701
```

4.2.3 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:
```

4.2.4 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
```

4.2.5 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:
```

4.2.6 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
```

4.2.7 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:
```

4.3 IP Transfer Connect Requirements

This section provides software requirements and important configuration screens used to enable the IP Transfer Connect (IPC) feature of the AT&T IP Toll Free network. This section also provides sample vectors for invoking IPC features. The IPC feature consists of the following basic capabilities.

- When a call is received from the IP Toll Free service at the customer premises equipment (CPE), the Avaya CPE may initiate a pre-answer transfer (also known as a redirect). This redirect contains the following key information.
    - Speed dial code (SDC) – The SDC is an abbreviation that specifies the destination to which the call should be transferred. The SDC is assigned by the customer and given to AT&T during IPC provisioning. Once the redirect is received, the AT&T network transfers the call to the destination specified by the SDC.
    - User to user information (UUI) – The CPE may optionally return UUI in the redirect. The AT&T network forwards the UUI when it transfers the call to the destination specified in the SDC.
- When a call is received from the IP Toll Free service at the customer premises equipment (CPE), the Avaya CPE may initiate a post-answer transfer (also known as a courtesy transfer). This courtesy transfer contains the following key information.
    - Speed dial code (SDC) – The SDC is an abbreviation that specifies the destination to which the call should be transferred. The SDC is assigned by the customer and given to AT&T during IPC provisioning. Once the courtesy transfer is received, the AT&T network transfers the call to the destination specified by the SDC.
    - User to user information (UUI) – The CPE may optionally return UUI in the courtesy transfer. The AT&T network forwards the UUI when it transfers the call to the destination specified in the SDC.

Note: All of the examples in the following subsections show a SDC of "91919191919". **Vectors with this SDC must be configured prior to test and turn up**. **Customer provided SDCs can be 1 to 10 digits. The SDC for test and turn up (i.e. 91919191919) is an exception to this rule. Furthermore, there is a limit of three courtesy transfer attempts in a vector.**

4.3.1 Avaya Multi-Vantage Feature Requirements

In order to the support the IPC, the Avaya network call redirection (NCR) feature set must be supported. A license for this feature (SA8898) must be obtained from Avaya (highlighted in red below). The following screen shows how to check for the presence of this feature.

```
display system-parameters special-applications                  Page
7 of   7
                             SPECIAL APPLICATIONS

                      (SA8896) - IP Softphone Lamp Control? n
  (SA8898) - SIP Service Provider Network Call Redirection? y
      (SA8917) - LSP Redirect using special coverage point? n
     (SA8928) - Display Names on Bridged Appearance Labels? n
            (SA8931) - Send IE with EC500 Extension Number? N
```

In addition to this license, the NCR feature must be turned on. This is done on the trunk form (highlighted in red below) associated with the SIP trunk that points to the SES server. The required field is on page 4 of the trunk form.

```
change trunk-group 69                                           Page
4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
Service Provider Network Call Redirection? y
             Telephone Event Payload Type: 127
```

Finally, shuffling (aka direct IP-IP audio connections, highlighted in red below) must be turned “off” when using the NCR feature. Shuffling does not work properly when the NCR feature is turned on. Shuffling is best turned off on the signaling group screen (as shown in the following example).

```
change signaling-group 69                                   Page   1 of   1
                               SIGNALING GROUP

 Group Number: 69             Group Type: sip
                        Transport Method: tls




   Near-end Node Name: C-LAN                    Far-end Node Name: ccs
 Near-end Listen Port: 5061                   Far-end Listen Port: 5061
                                           Far-end Network Region: 1
       Far-end Domain:

                                           Bypass If IP Threshold Exceeded? y

         DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? n
                                                      IP Audio Hairpinning? y
 Enable Layer 3 Test? n
 Session Establishment Timer(min): 120
```

4.3.2 Incoming Call Routing on IPC Calls

For incoming IPC calls, use the change inc-call-handling-trmt command screen as shown next. This example maps the IPTF DNIS (also known as signaled digits outpulsed or SDOP) plus 5 zeroes “000000027” to the internal extension “5123” for internal routing. The “5123” number is a vector directory number (VDN; see next section) which then initiates the vector.

```
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                9   000000027        9    5123
```

4.3.3 Vector Directory Number

The vector directory number is a called number which maps to a vector. This sample VDN points to vector 123 which is the IPC example shown in the next section. A unique SDOP, VDN and vector is required for each example.

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

                             Extension: 5123
                                  Name: NCR redirect no UUI
                         Vector Number: 123
                   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:
```

4.3.4 IP Redirect without UUI

The following vector performs a redirect without UUI.

```
change vector 123                                               Page   1 of   6
                                  CALL VECTOR

    Number: 123              Name: NCR redirect no UUI
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 #    redirect -no answer -   this is a pre-answer
02 route-to     number ~r91919191919          with cov n if unconditionally
03 #    Play error only if routing fails
04 announcement 6198
```

The steps of this vector do the following.

- Step 01 – This step is a comment.
- Step 02 – This step returns a redirect back to the AT&T network with SDC "91919191919". Once test and turn up is completed, you can then put in the desired SDC.
- Step 03 – This step is a comment.
- Step 04 – This step points to a pre-configured announcement that plays if the redirect step fails.

4.3.5 IP Redirect with UUI

In order to send UUI, a variable must be defined in the "change variables" screen. The following example shows the creation of a variable named "A" which is 5 characters long.

```
change variables                                              Page   1 of  39
                              VARIABLES FOR VECTORS

Var Description                 Type    Scope Length Start Assignment       VAC
A   UsertoUserTest              asaiuui L     5      1
```

The following vector performs a redirect with UUI using the previously defined variable A.

```
change vector 124                                             Page   1 of   6
                                  CALL VECTOR

    Number: 124             Name: NCR redirect no UUI
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 #    redirect -no answer -   this is a pre-answer
02 set          A      = digits CATR  67890
03 route-to     number ~r91919191919          with cov n if unconditionally
04 #    Play error only if routing fails
05 announcement 6198
```

The steps of this vector do the following.

- Step 01 – This step is a comment.
- Step 02 – This step sets the UUI variable "A" to a value of "67890".
- Step 03 – This step returns a redirect back to the AT&T network with SDC "91919191919" with the UUI.
- Step 04 – This step is a comment.
- Step 05 – This step points to a pre-configured announcement that plays if the redirect step fails.

4.3.6 IP Courtesy Transfer without UUI

The following vector performs a courtesy transfer without UUI.

```
change vector 125                                                Page   1 of   6
                                  CALL VECTOR

    Number: 125              Name: NCR-Refer-UUI
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 #    Refer with ringback, announcement, and Refer with UUI
02 wait-time    3   secs hearing ringback
03 announcement 6122
04 #    Refer occurs since this is post answer
05 route-to     number ~r91919191919          with cov n if unconditionally
06 #    Play error only if routing fails
07 announcement 6198
```

The steps of this vector do the following.

- Step 01 – This step is a comment.
- Step 02 – This step specifies that the caller will hear ringback for 3 seconds.
- Step 03 – This step points to a pre-configured announcement that answers the call.
- Step 04 – This step is a comment.
- Step 05 – This step returns refer message (post answer courtesy transfer) back to the AT&T network with SDC “91919191919”.
- Step 06 – This step is a comment.
- Step 07 – This step points to a pre-configured announcement that plays if the transfer step fails.

4.3.7 IP Courtesy Transfer with UUI

In order to send UUI, a variable must be defined in the “change variables” screen. The following example shows the creation of a variable named “A” which is 5 characters long.

```
change variables                                                 Page   1 of  39
                              VARIABLES FOR VECTORS

Var Description                 Type    Scope Length Start Assignment       VAC
A   UsertoUserTest              asaiuui L     5      1
```

The following vector performs a redirect with UUI using the previously defined variable A.

```
change vector 126                                              Page   1 of   6
                                  CALL VECTOR

    Number: 126              Name: NCR-Refer-UUI
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 #    Refer with ringback, announcement, and Refer with UUI
02 wait-time    3   secs hearing ringback
03 announcement 6122
04 #    Refer occurs since this is post answer -- uui sent with var A contents
05 set          A      = digits CATR  12345
06 route-to     number ~r91919191919           with cov n if unconditionally
07 #    Play error only if routing fails
08 announcement 6198
```

The steps of this vector do the following.

- Step 01 – This step is a comment.
- Step 02 – This step specifies that the caller will hear ringback for 3 seconds.
- Step 03 – This step points to a pre-configured announcement that answers the call.
- Step 04 – This step is a comment.
- Step 05 – This step sets the UUI variable “A” to a value of “12345”.
- Step 06 – This step returns refer message (post answer courtesy transfer) back to the AT&T network with SDC “91919191919” with the UUI.
- Step 07 – This step is a comment.
- Step 08 – This step points to a pre-configured announcement that plays if the transfer step fails.

4.3.8 IP Courtesy Transfer with Multiple Attempts

The following vector performs multiple courtesy transfer attempts without UUI. Only the last attempt is successful.

```
change vector 122                                                Page   1 of   6
                                  CALL VECTOR

    Number: 122              Name: NCR 3 REFERS
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 #    shows multiple refers
02 wait-time    3   secs hearing ringback
03 announcement 6122
04 route-to     number ~r11111          with cov n if unconditionally
05 route-to     number ~r22222          with cov n if unconditionally
06 route-to     number ~r33333          with cov n if unconditionally
07 route-to     number ~r91919191919           with cov n if unconditionallly
08 announcement 6198
```

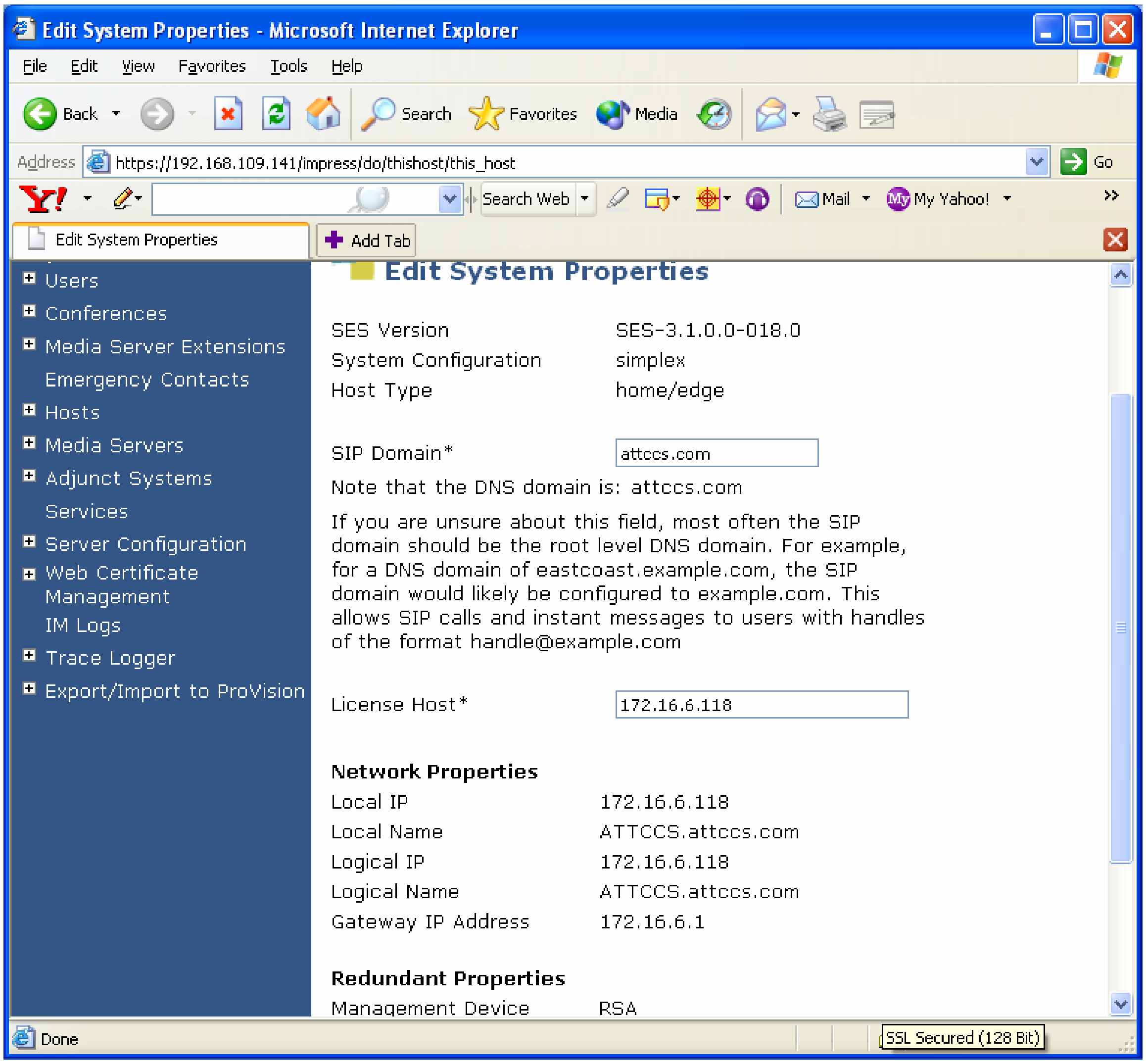
The steps of this vector do the following.

- Step 01 – This step is a comment.
- Step 02 – This step specifies that the caller will hear ringback for 3 seconds.
- Step 03 – This step points to a pre-configured announcement that answers the call.
- Step 04 – This step returns refer message (post answer courtesy transfer) back to the AT&T network with SDC “11111” without the UUI.
- Step 05 – If the previous step fails, this step returns refer message (post answer courtesy transfer) back to the AT&T network with SDC “22222” without the UUI.
- Step 06 – If the previous step fails, this step returns refer message (post answer courtesy transfer) back to the AT&T network with SDC “33333” without the UUI.
- Step 07 – If the previous step fails, this step returns refer message (post answer courtesy transfer) back to the AT&T network with SDC “91919191919” without the UUI.
- Step 08 - This step points to a pre-configured announcement that plays if the transfer step fails.

4.4 SES (SIP Enablement Server) Configuration

4.4.1 Version and Feature Requirements

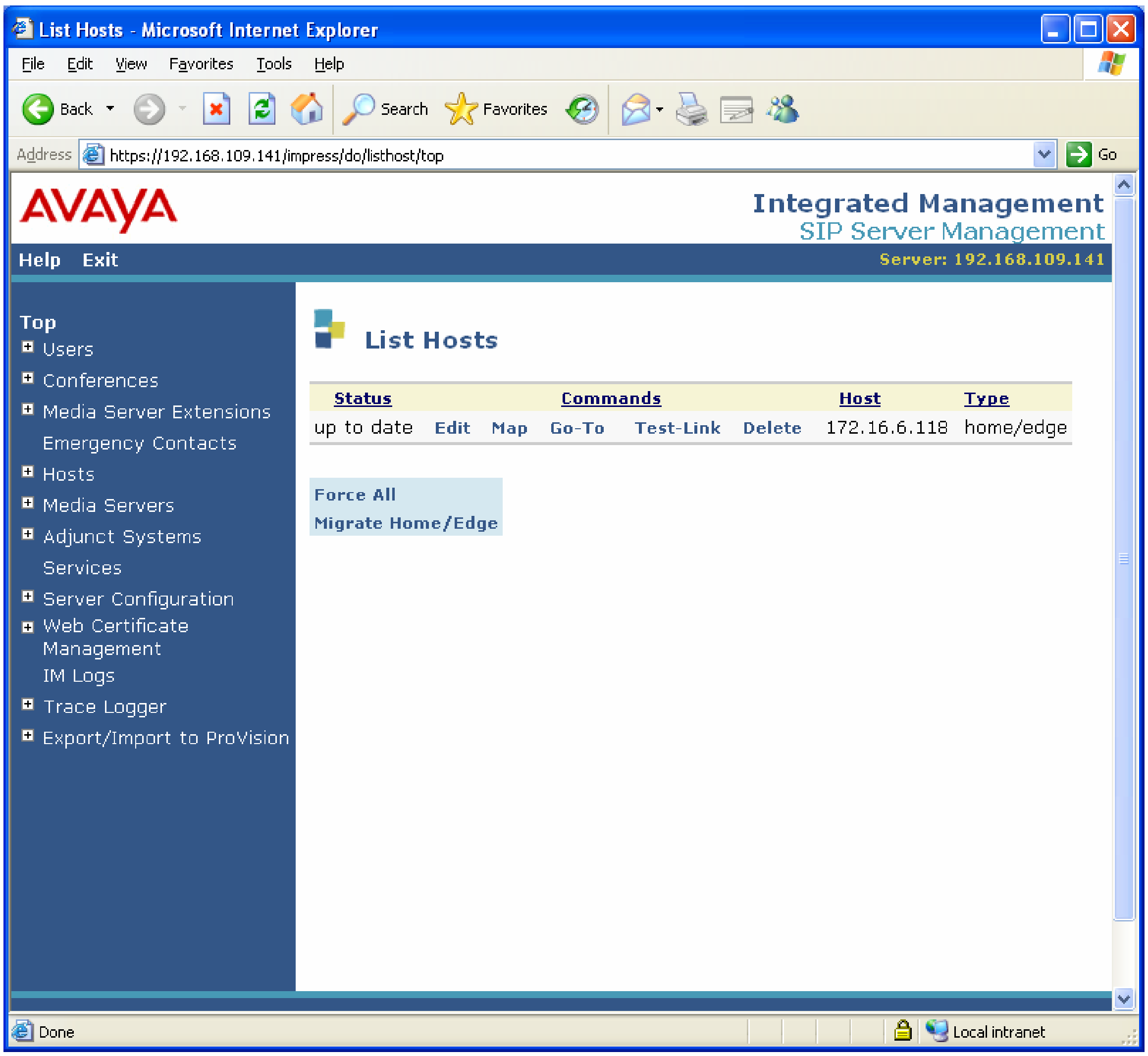
The SES should be running version SES-3.1.0.0-018.0 as shown below on the "server configuration / system properties" screen.



4.4.2 Host Screens

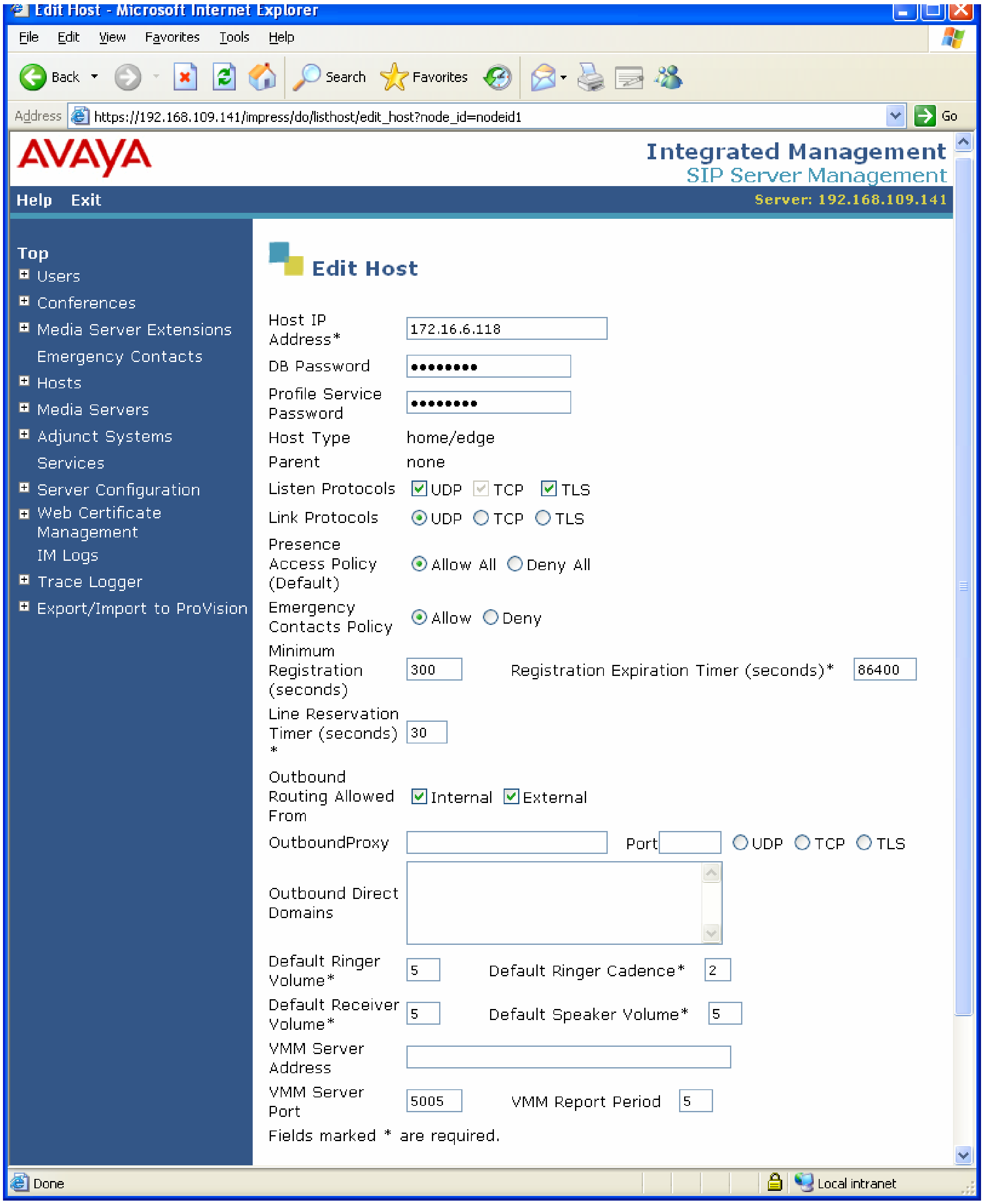
4.4.2.1 Accessing Host Screens

The host parameters are accessed by selecting “Host”. You will receive the “List Hosts” screen.



4.4.2.2 Host Parameters

Next you can select the "edit" option to get the following screen.

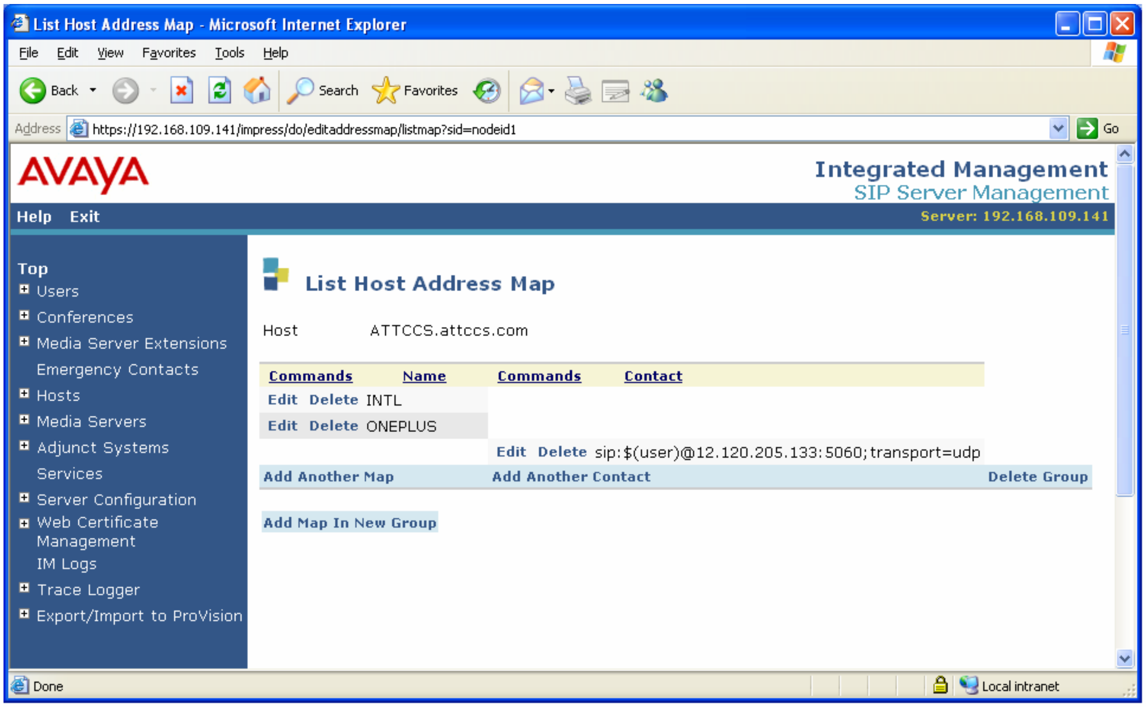


Set parameters are shown above. Key parameters are:

Host name – IP address of SES.
Listen protocols – Check all 3 boxes.
Link protocol is udp.
OutboundProxy – Not required.

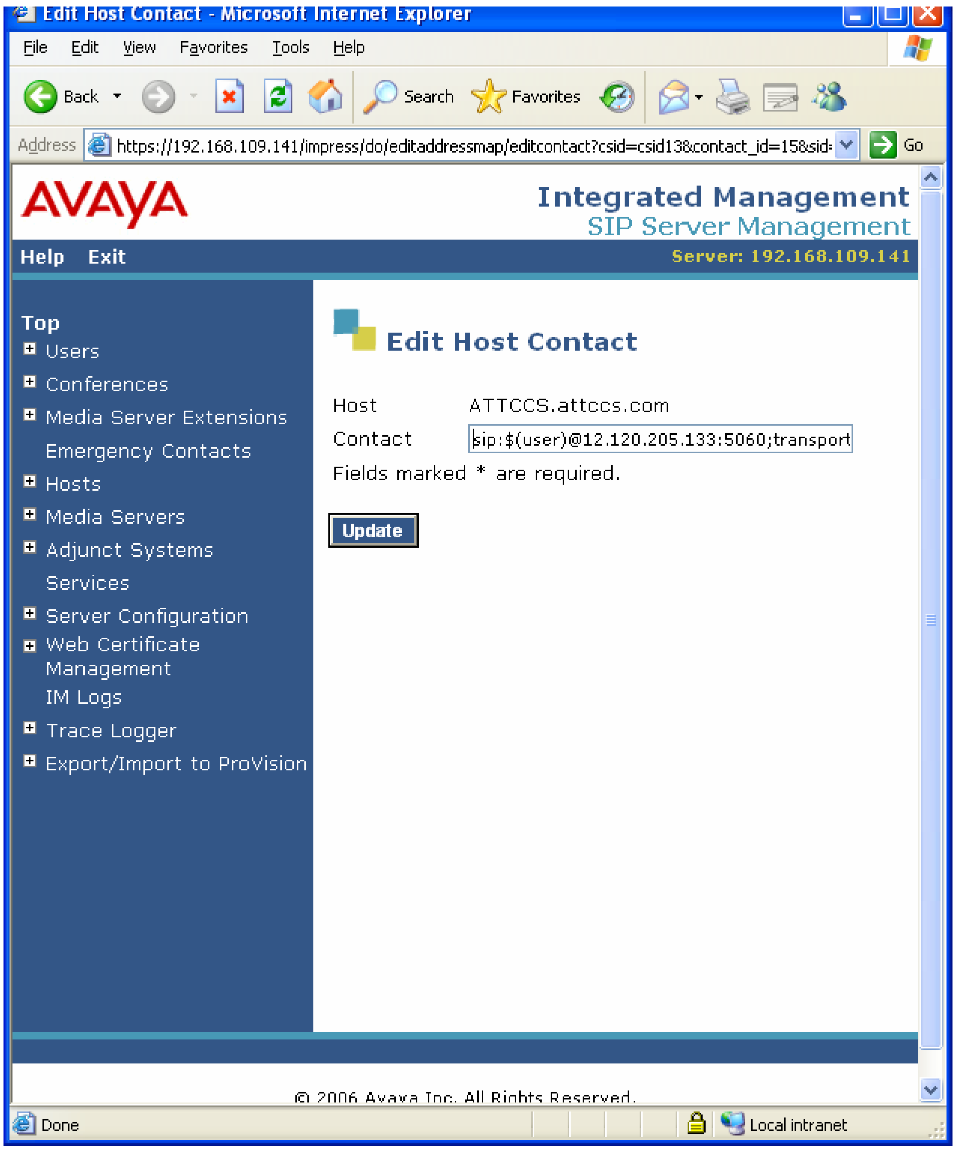
4.4.2.3 Host Address Map Parameters

From the “List Hosts” screen, you select “map” option to receive the following screen.



4.4.2.4 Host Contact Parameters

From the list screen, you should then select “edit” (next to contact) to retrieve the host contact entry or “add another contact” to create a new one. The contact entry provides the location of this host (i.e. IP address). This contact entry points to the AT&T border element. **There must be a contact entry for each of the 2 AT&T Border Elements provided by AT&T Customer Care.**



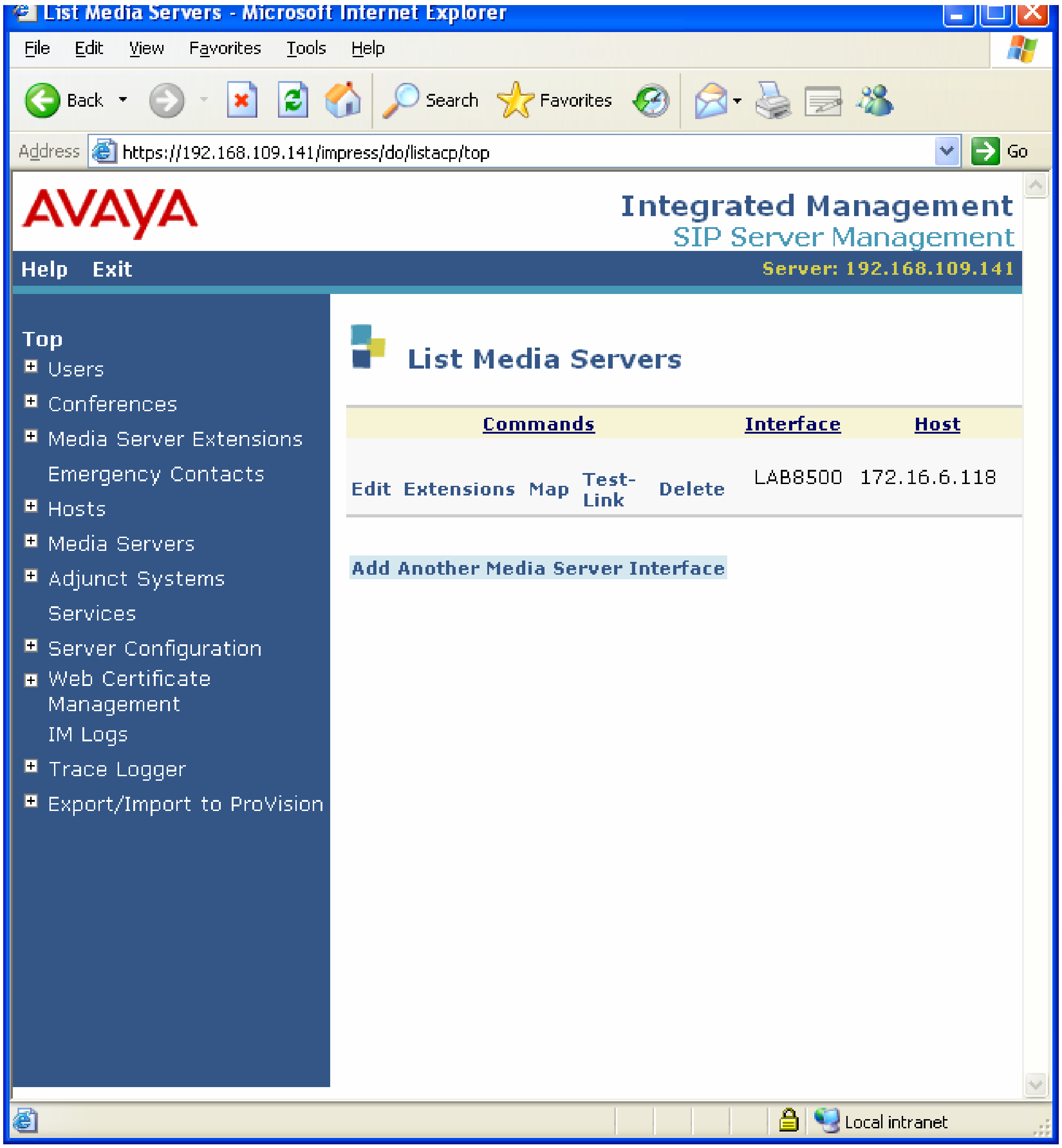
Key parameters are:

Contact – This parameter sets the Contact header. The format is shown above. **Please contact AT&T Customer Care to obtain the IP Border Element Address**.

4.4.3 Media Server Screens

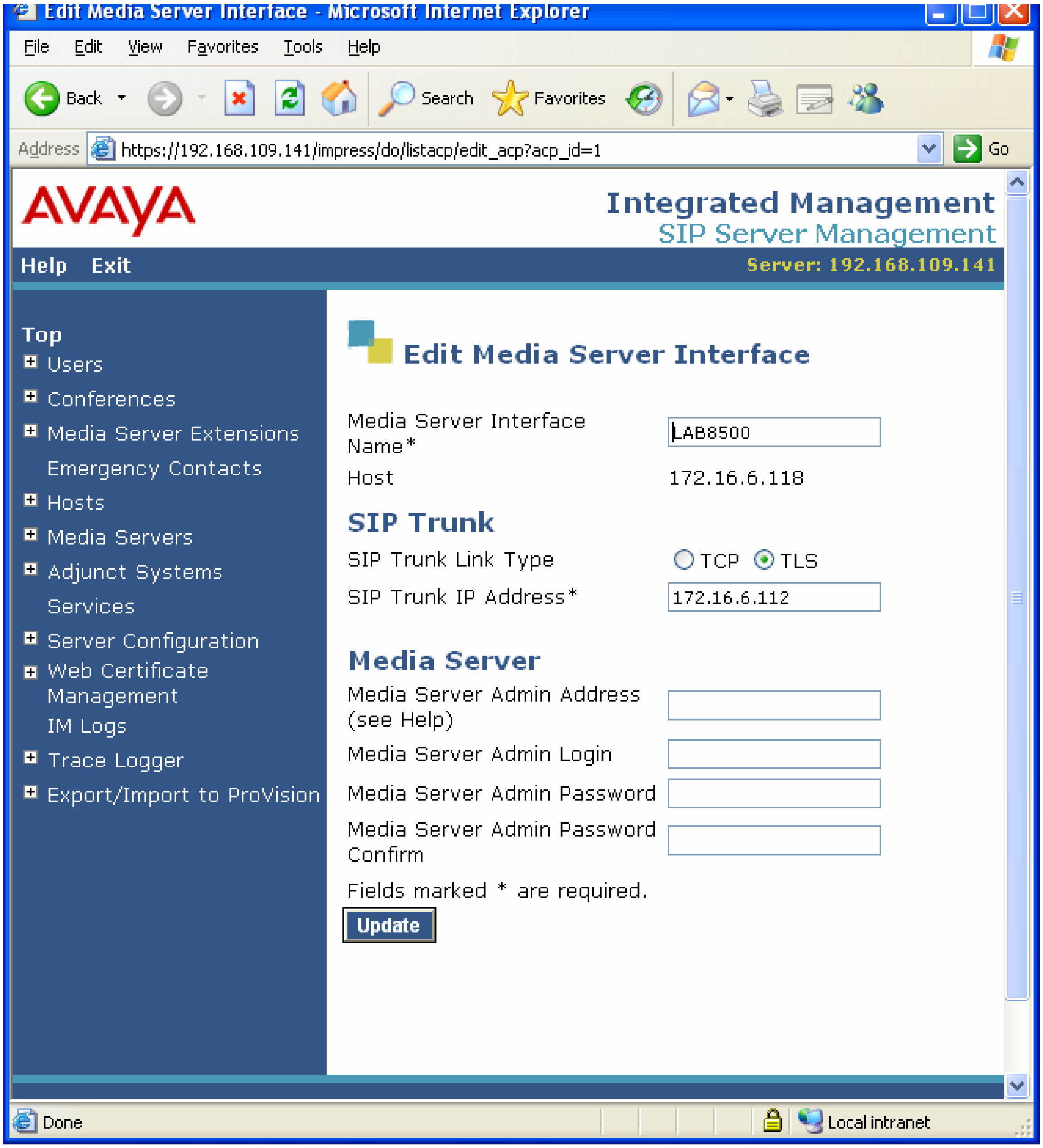
4.4.3.1 Accessing Media Server Screens

The media parameters are accessed by selecting “Media Server”. You will receive the “List Media Servers” screen.



4.4.3.2 Media Server Parameters

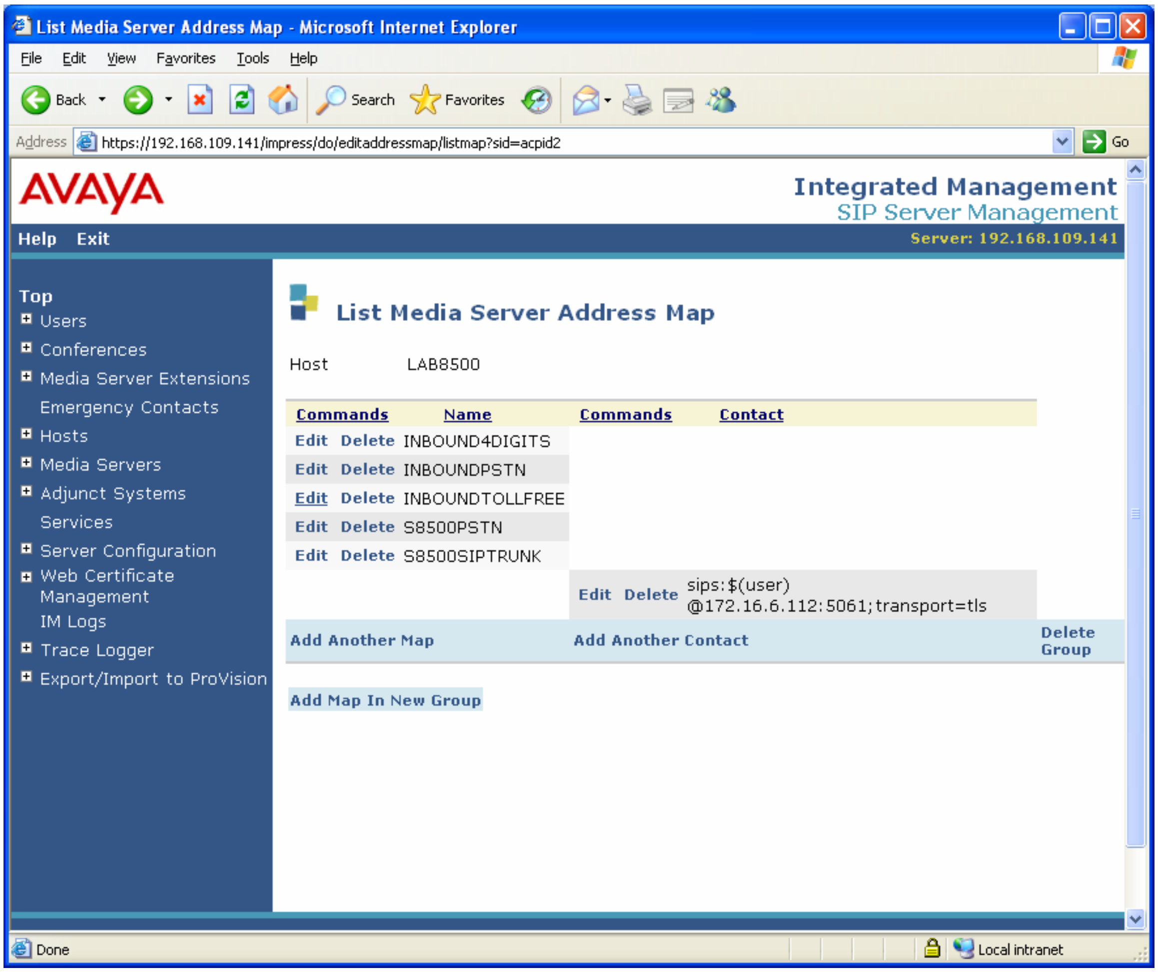
Next you can select the “edit” option to get the following screen.



Key parameters are:
Link Type – Set this to “TLS”.
Name of IP address – This is the IP address of the CLAN board on the Communication Manager.

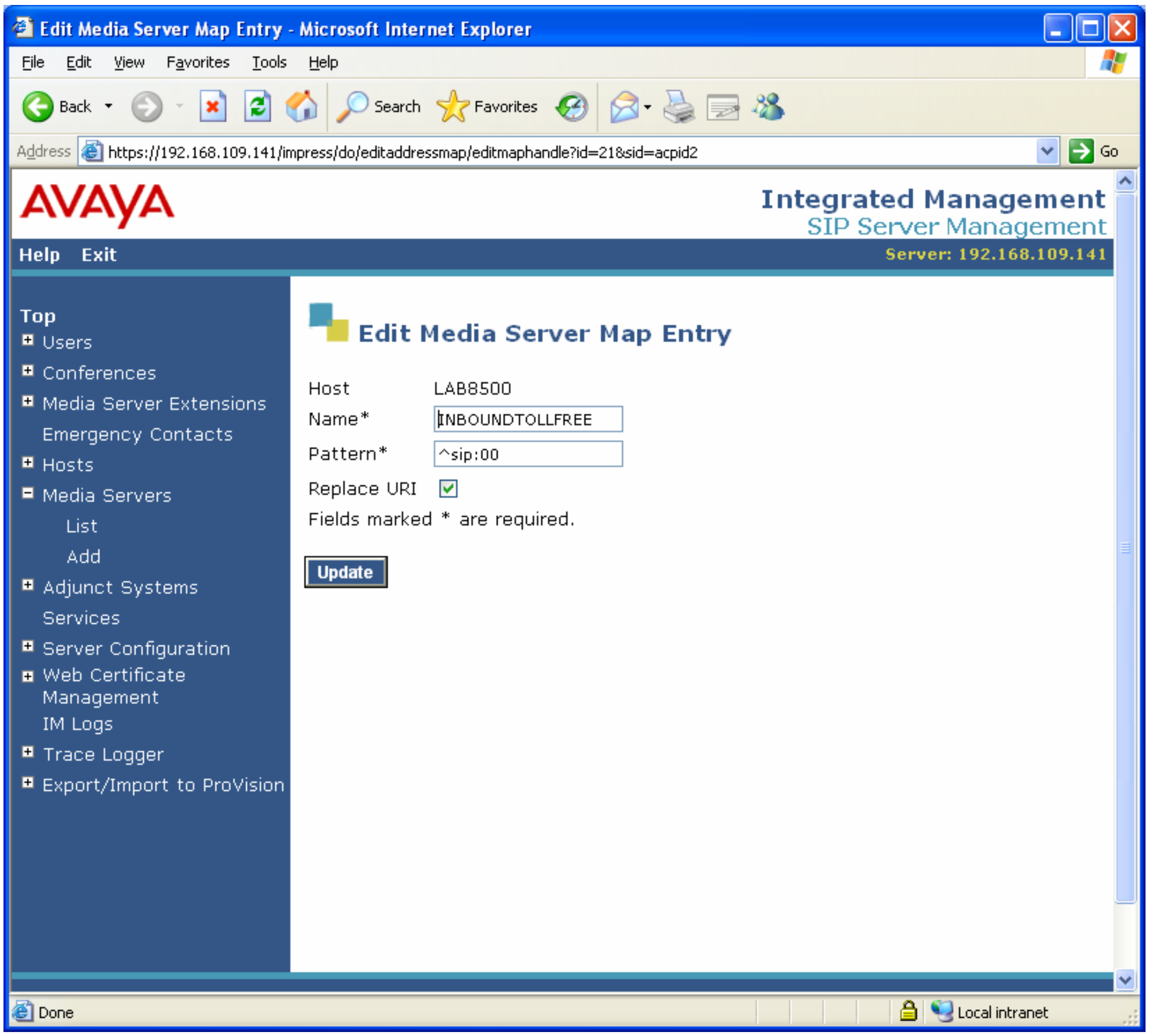
4.4.3.3 Media Server Address Map Parameters

From the “List Media Servers” screen, you select “map” option to receive the following screen. These entries provide the called number patterns that should be sent to the CLAN in the CM (Communication Manager).



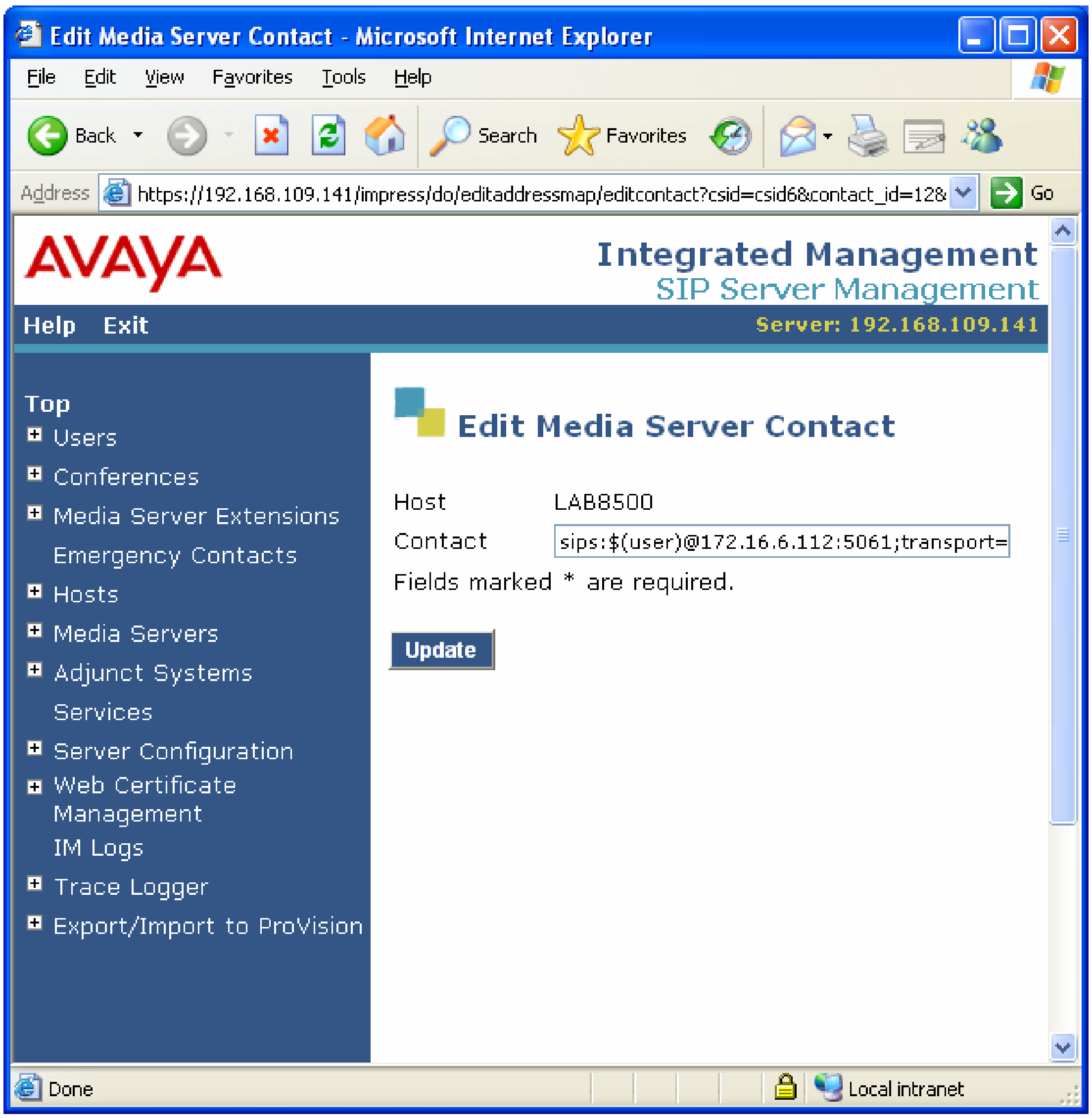
4.4.3.4 Sending Inbound Calls to the CLAN

Inbound calls received from the AT&T Network by the SES must be sent to the CLAN on the CM. The following media server map example directs all IP Toll Free calls dialed with the "00" prefix to the CLAN in the CM.



4.4.3.5 Media Server Contact

From the list screen, you should then select “edit” (next to contact) to retrieve the media server contact entry or “add another contact” to create a new one. The contact entry provides the location of this media (i.e. IP address). This contact entry points to the CLAN in the CM.



Key parameters are:
Contact – This parameter sets the Contact header. The format is shown above. The IP address is the address of the CLAN in the CM.

4.4.4 Trusted Host Configuration

The AT&T network must be configured as a trusted host. Each of the 2 border elements provided by AT&T must be configured as trusted hosts.

To configure a trusted host, use the trustedhost command in the Linux shell of Avaya SIP Enablement Services. The command:

trustedhost –a 12.120.205.133 –n 172.16.6.118 –c Your_Proxy

is used to add the trust relationship. The –a argument specifies the address to be trusted; –n specifies the SES IP address; –c adds a comment.

The trustedhost –L command allows the list of trusted hosts to be displayed.

admin@k2> trustedhost -a 12.120.205.133 -n 172.16.6.118 –c Your_Proxy

12.120.243.114 is added to trusted host list.

admin@k2> trustedhost -L

Third party trusted hosts.

Trusted Host	SES Host Name	Comment
12.120.205.133	172.16.6.118	Your_Proxy

For completeness, the –d argument allows the trust relationship to be deleted. For, example,

trustedhost –d 12.120.205.133 –n 172.16.6.118

removes the trust relationship added above.

After configuring a trusted host, the user must go to the Administration web interface and click on the Update link for the changes to take effect. This is required even though the trusted host was configured via the Linux shell.

5 Troubleshooting

This section provides some tips about troubleshooting problems

5.1 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>