AT&T VOIP
Avaya Multi-Vantage 3.1.2
Configuration Guide
For Use with AT&T
IP Flexible Reach Service
(SIP Version)

Issue 1.5
1/30/2008

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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 Flexible Reach Service using SIP.

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.

Calling Number Restricted / Private not supported

For calls from Avaya to AT&T, the calling party number cannot be marked as restricted. This function is not supported.

Fax Status

For calls setup as G.729, switchover to G.711 (via re-invite) is supported. G3 Fax is supported except on calls between 2 Avaya PBXs. SG3 fax is supported except for calls received by the Avaya PBX. SG3 fax is supported for calls originated by the Avaya PBX.

Due to inhomogeneous hardware standards, AT&T cannot give any guarantees for FAX transport reliability.

Mid Call DTMF Issue

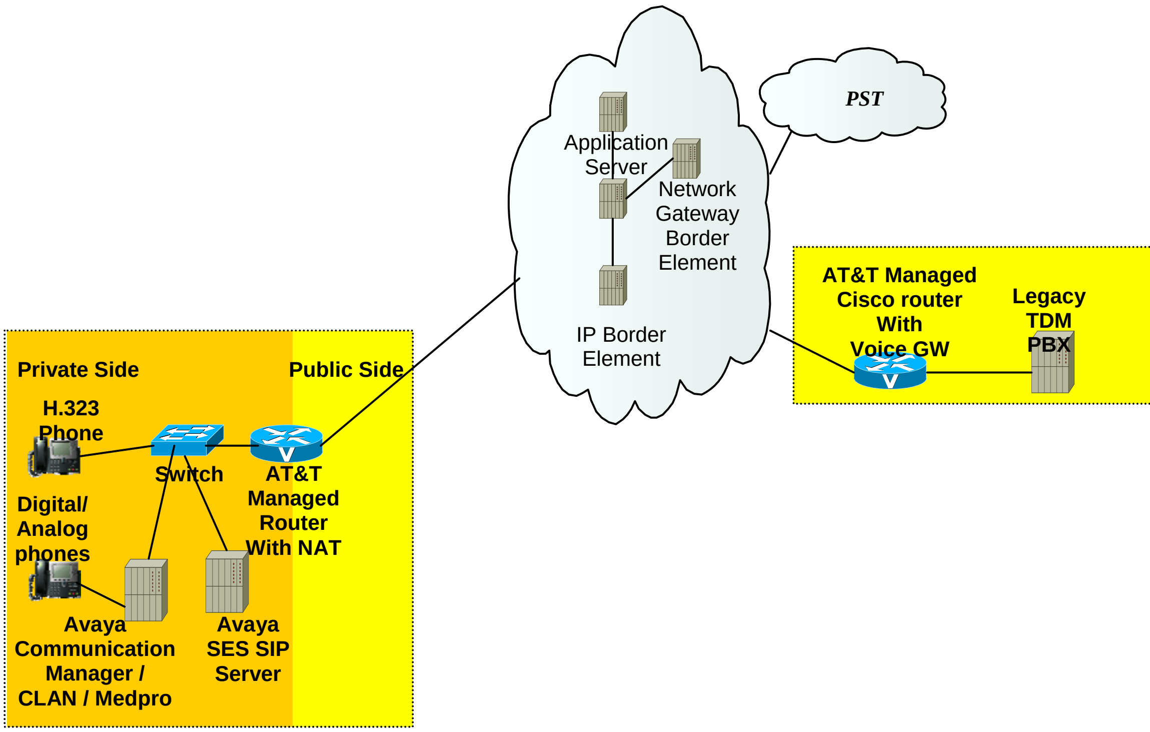
When direct media (i.e. shuffling turned on) is configured with an H.323 IP phone, the first digit from the H.323 IP phone is not passed by the AT&T network. Subsequent digits are passed successfully. The work around for this is to turn off shuffling or retry the digits. A fix is planned for this issue.

Failover to an Alternate AT&T Border Element

Currently, there is no tested configuration with the Avaya MV/CM SIP trunk for connecting to an alternate AT&T Border Element if the primary border element is not available.

3 Overview

This section provides a service overview of the Avaya Multi-Vantage integration with AT&T IP Flexible Reach 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.
    - o Processor – This is an 8500/8700 Linux processor or 8300/8400 processor with GXXX gateway.
    - o SIP Server – This server is known as the SES server. The server provides the SIP 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 dynamic (many to one) NAT for the Medpro cards and H.323 phones 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 Flexible Reach Services using SIP.

4.1 Communication Manager Configuration

4.1.1 Version and Feature Requirements

The Avaya Communication Manager (CM) must be running version 3.1.2.

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

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

```
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

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
```

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 an 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. The second page of the configuration shows 2 different configuration. The first instance shows the configuration for G.711 fax and the second configuration shows the configuration for T.38 fax.

```
    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                     0
    Modem           off                     0
    TDD/TTY         US                      3
    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 the IP address of the SES server. This setting is required for compatibility with the IP FlexReach Service.
- 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.
- Intra-region IP-IP Direct Audio: yes
- Inter-region IP-IP Direct Audio: yes

With H.323 IP phones, direct audio (i.e. shuffling) from the IP phones is supported subject to the limitations described in the Special Notes section.

With digital or analog phones or if 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: 172.16.6.118
    Name: Avaya Trial
                                Intra-region IP-IP Direct Audio: yes
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: yes
      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 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
                         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 – Node name of SES Server.
- Bypass if IP threshold exceeded – Set to n.
- DTMF over IP: Set to rtp-payload.
- Direct IP-IP audio connection – Set to y.
- IP audio hairpinning – Set to y.

```
                                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: attccs.com

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? y
 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 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

        869                     14     21      1       aar
```

The parameters for the route pattern are shown next.

- Group number – Trunk group number.

```
                                  Pattern Number: 1   Pattern Name: S8500 TO
SIPSES
                             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: 1    0                    3                                   n   user
 2:                                                               n   user
 3:                                                               n   user
 4:                                                               n   user
 5:                                                               n   user
 6:                                                               n   user

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

4.1.9 Calling Number Configuration

In order to send a calling number, the public-unknown-number screen must be used as shown next.

```
                     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                 7323683601      10
 4 3                 732368          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 single CN for all calls.
- The second example shows the use of a CN prefix which is pre-pended to the actual extension.

4.2 Incoming Call Routing

When using the AT&T Virtual Telephone Number feature (VTN), the AT&T network will send the call to the PBX using a 10 digit PSTN number. This number can be mapped to an internal extension on the following screens.

When using non virtual TNs, AT&T can send 7 digits or less. The PBX administrator must tell AT&T Customer Care the number of digits required.

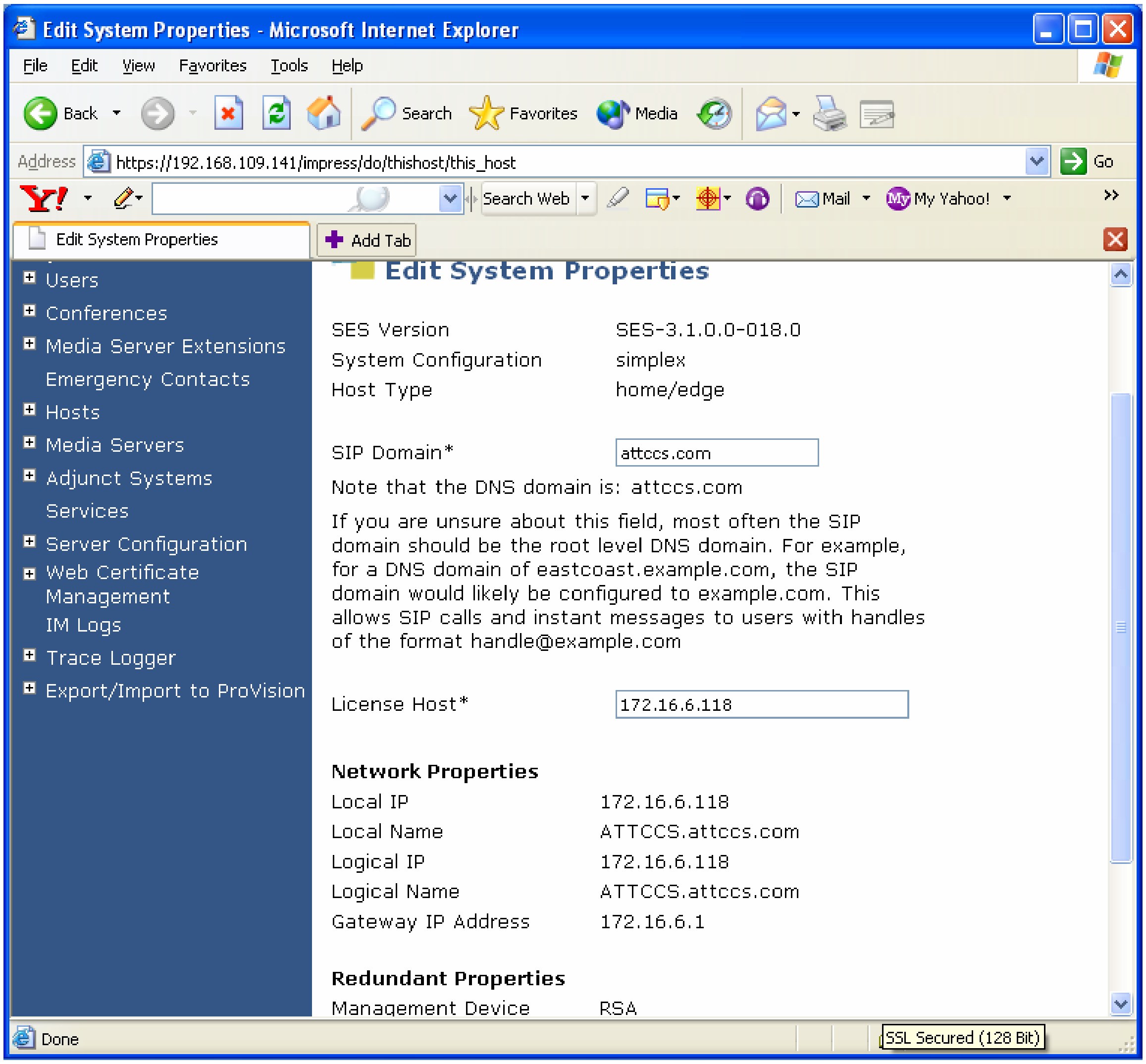
Use the change inc-call-handling-trmt command screen as shown next. This example maps the public called number “7323680470” to the internal extension “3650” and the called number “0471” to “3651” for internal routing.

```
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         7323680470    10   3650
tie             4          0471         4    3651
```

4.3 SES (SIP Enablement Server) Configuration

4.3.1 Version and Feature Requirements

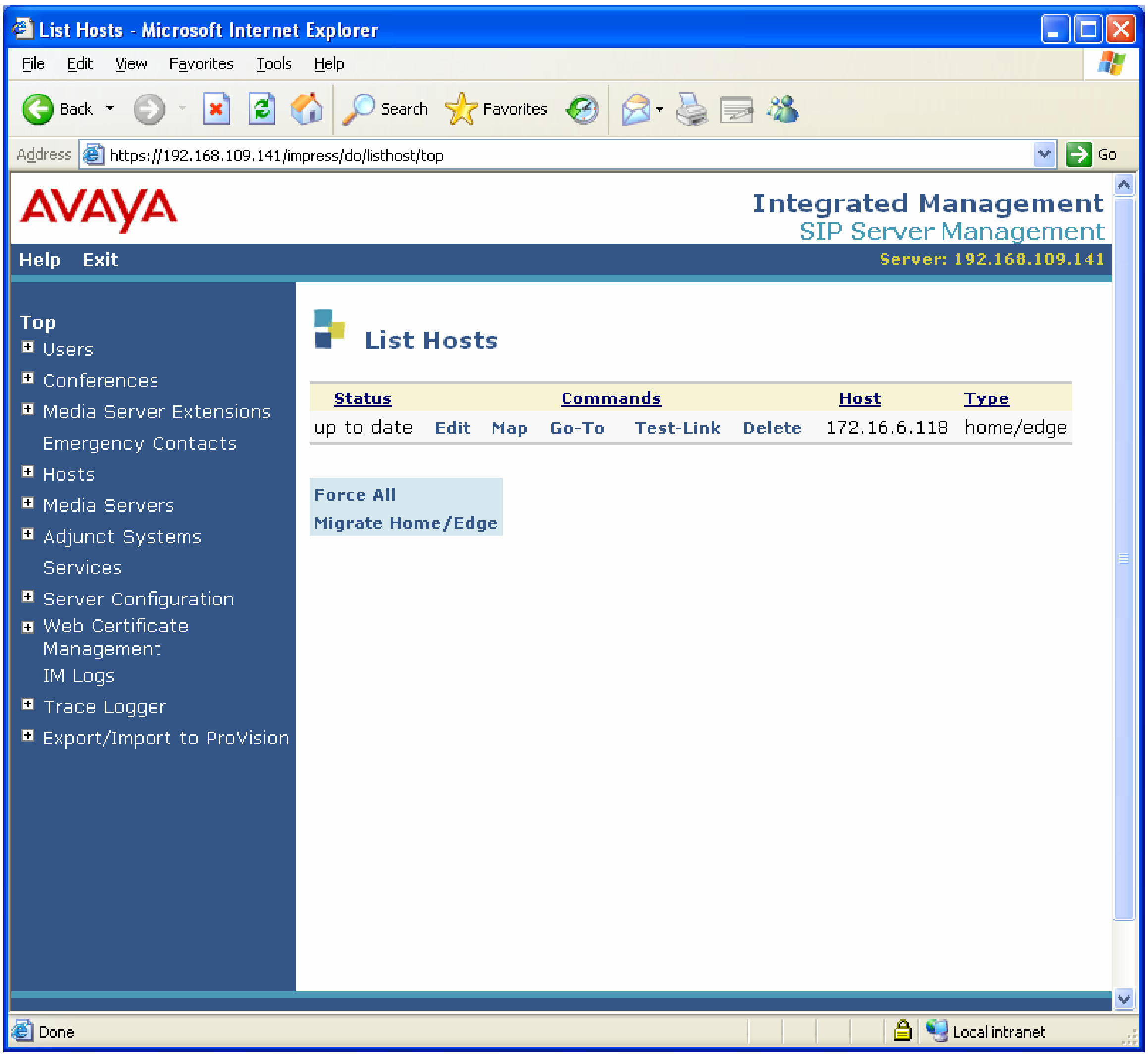
The SES should be running version SES-3.1.0.0-018.0 as shown below on the "server configuration / system properties" screen.



4.3.2 Host Screens

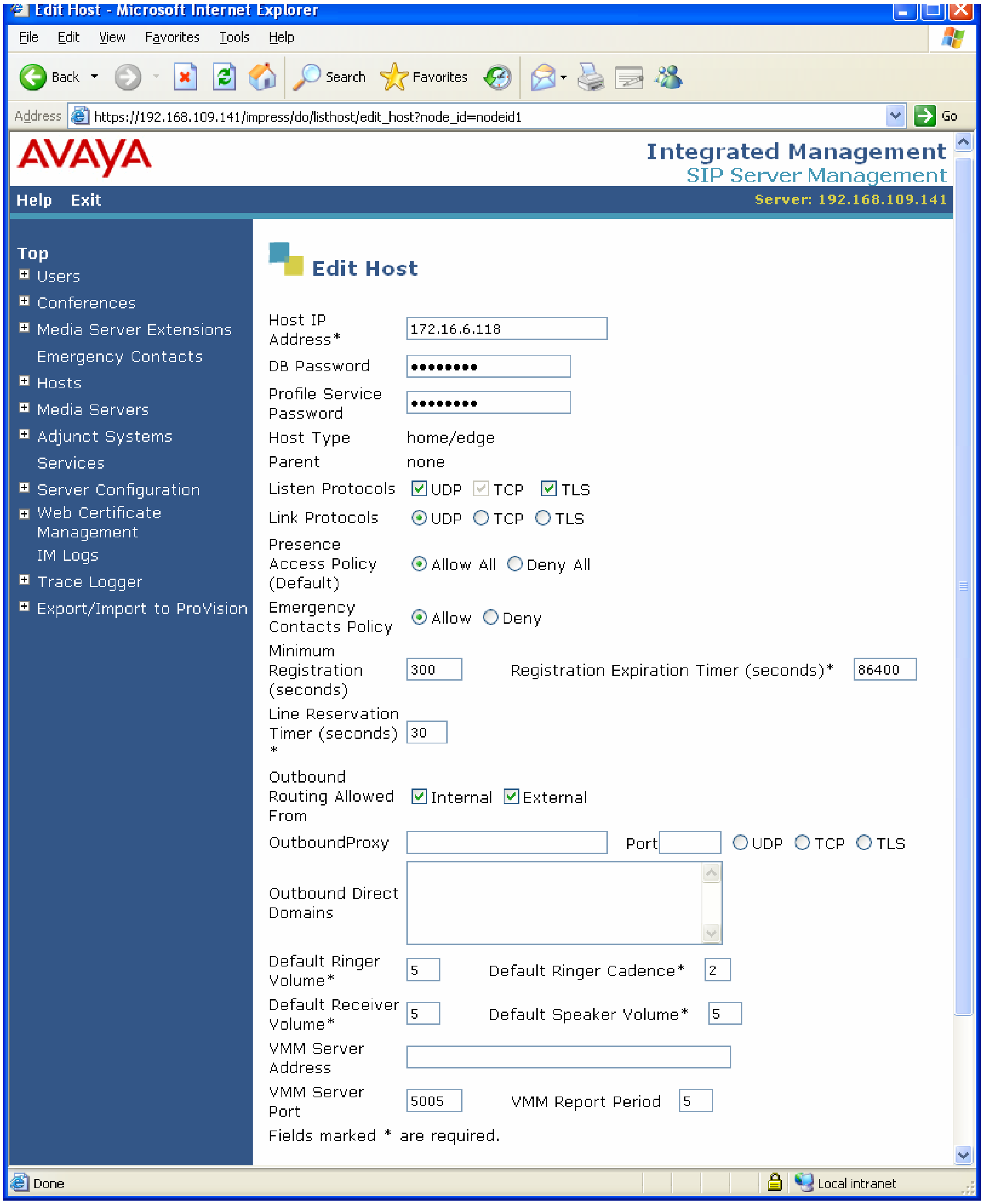
4.3.2.1 Accessing Host Screens

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



4.3.2.2 Host Parameters

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

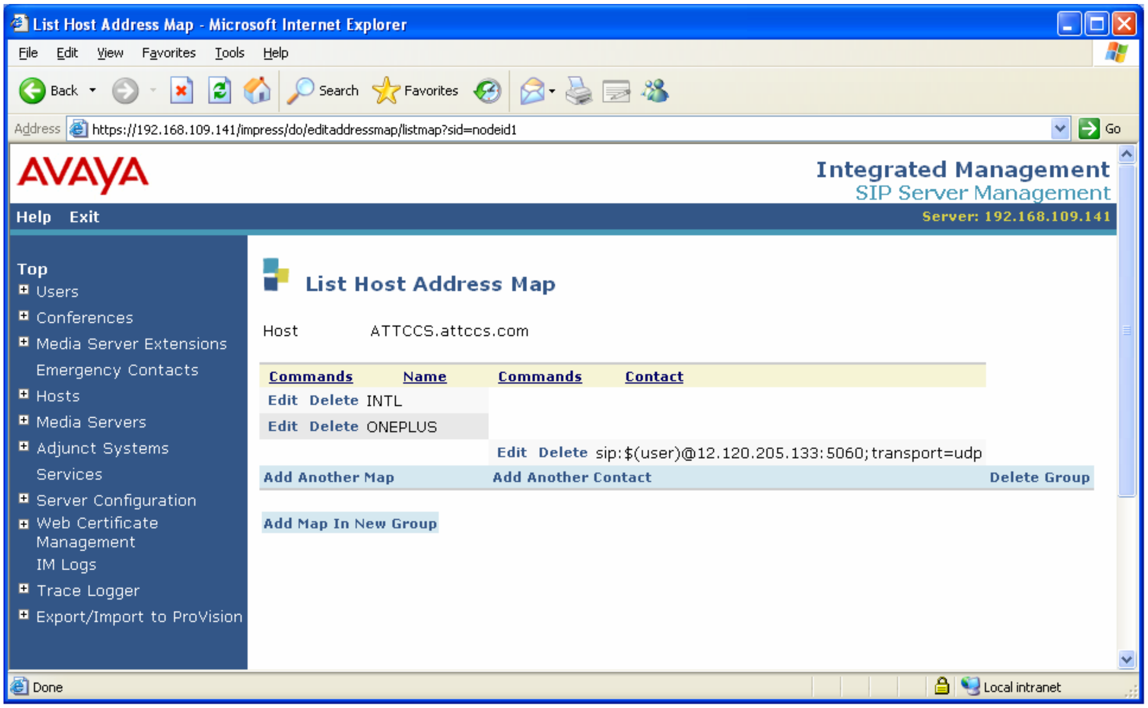


Set parameters are shown above. Key parameters are:

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

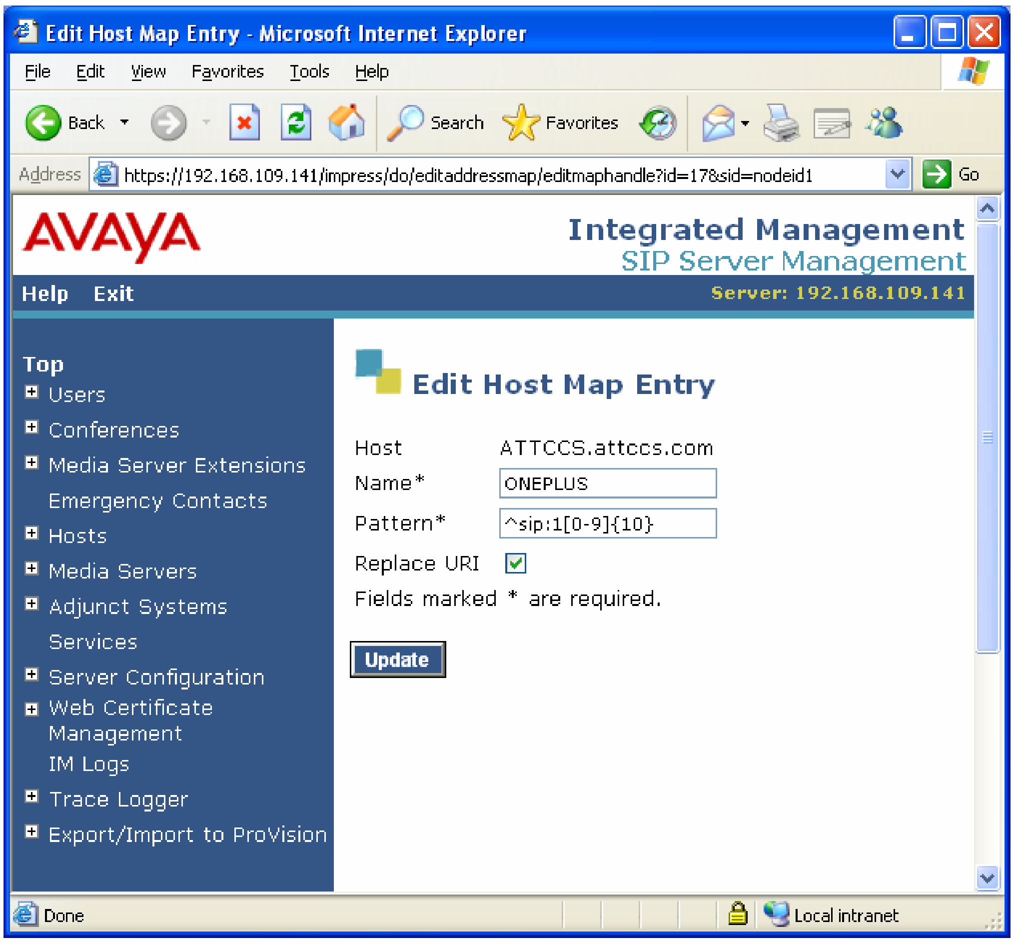
4.3.2.3 Host Address Map Parameters

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



4.3.2.4 Directing Calls to the AT&T Network

From this screen, you should then select “edit” (to the left of the Name heading) to retrieve the host map entry or “add another map” to create a new one. The host map entry contains the SIP URI patterns (i.e. called number) that should be matched to send calls to the AT&T Network.

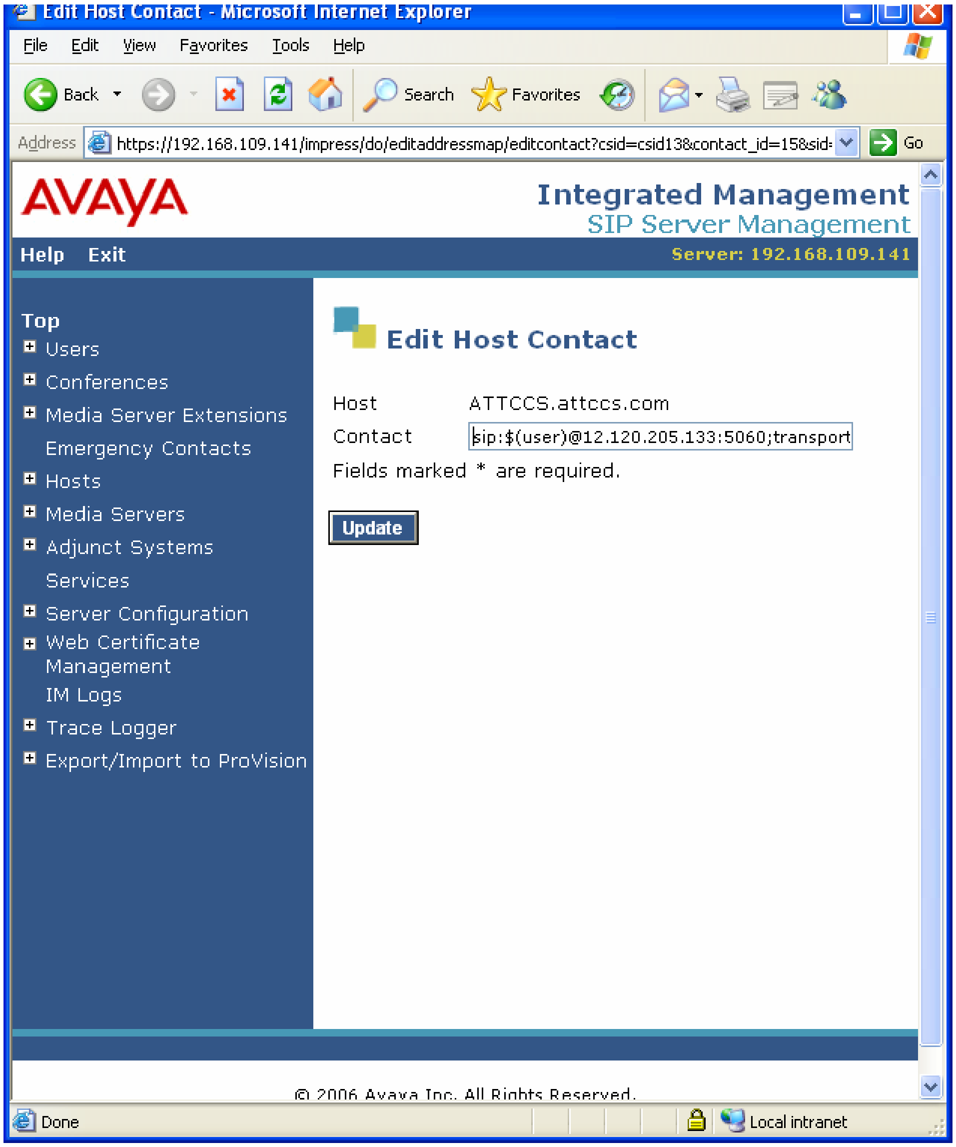


Key parameters on the host map screen are:

Host map name – Name of the host map.
Pattern – This parameter displays the format of the SIP URI. In the example, the format is 1 followed by 0 to 9 repeated 10 times.

4.3.2.5 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.



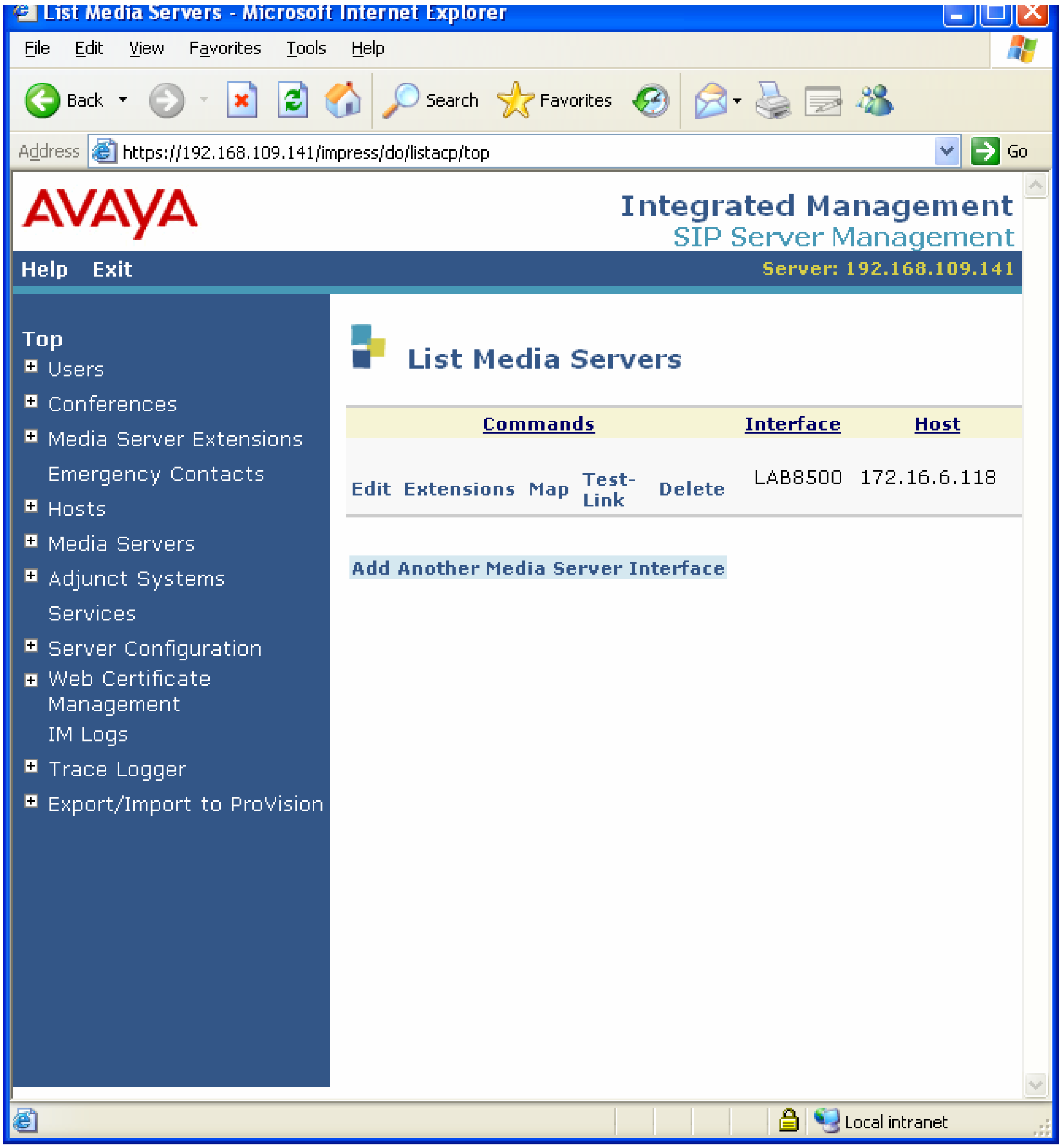
Key parameters are:

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**.

4.3.3 Media Server Screens

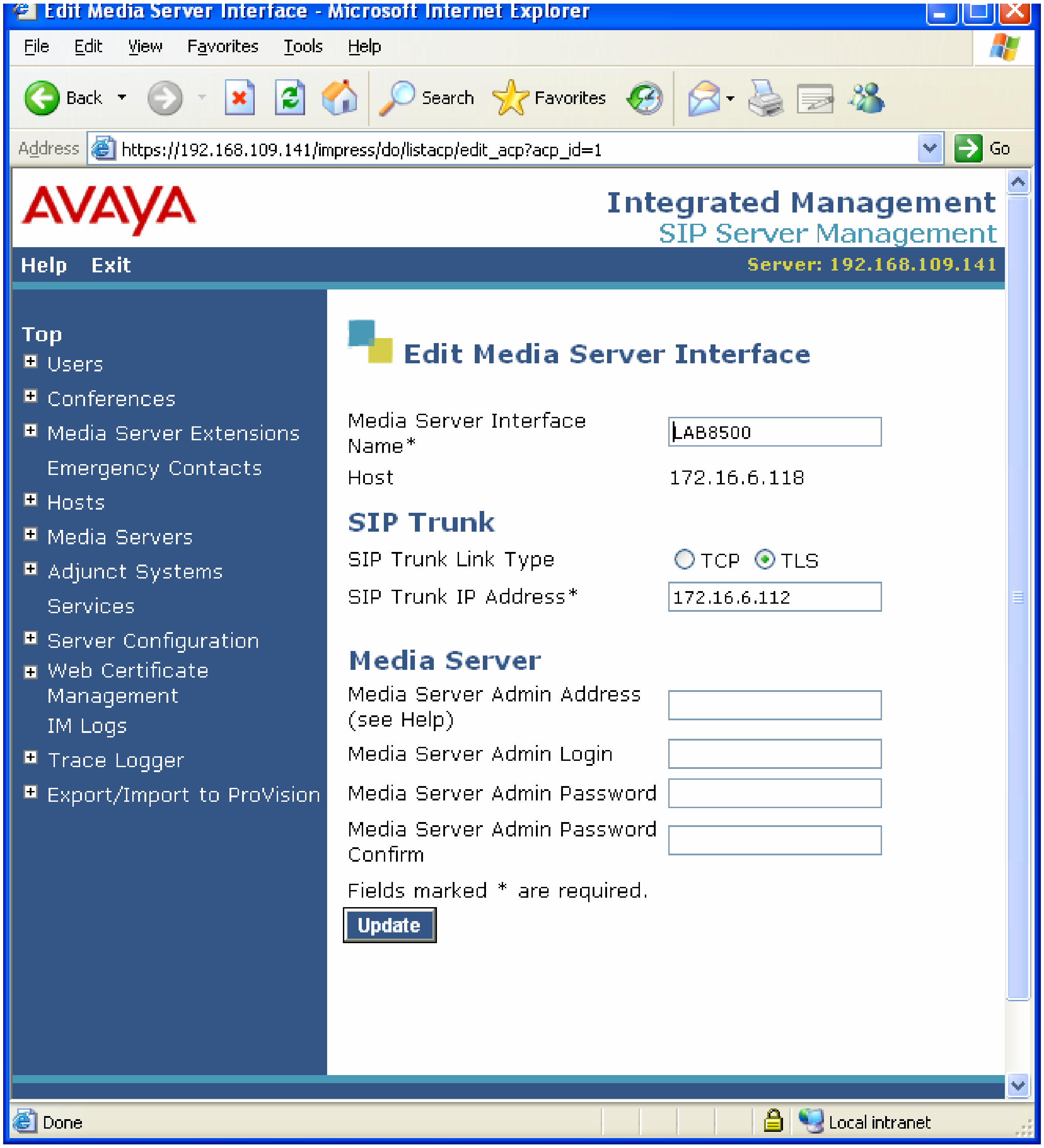
4.3.3.1 Accessing Media Server Screens

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



4.3.3.2 Media Server Parameters

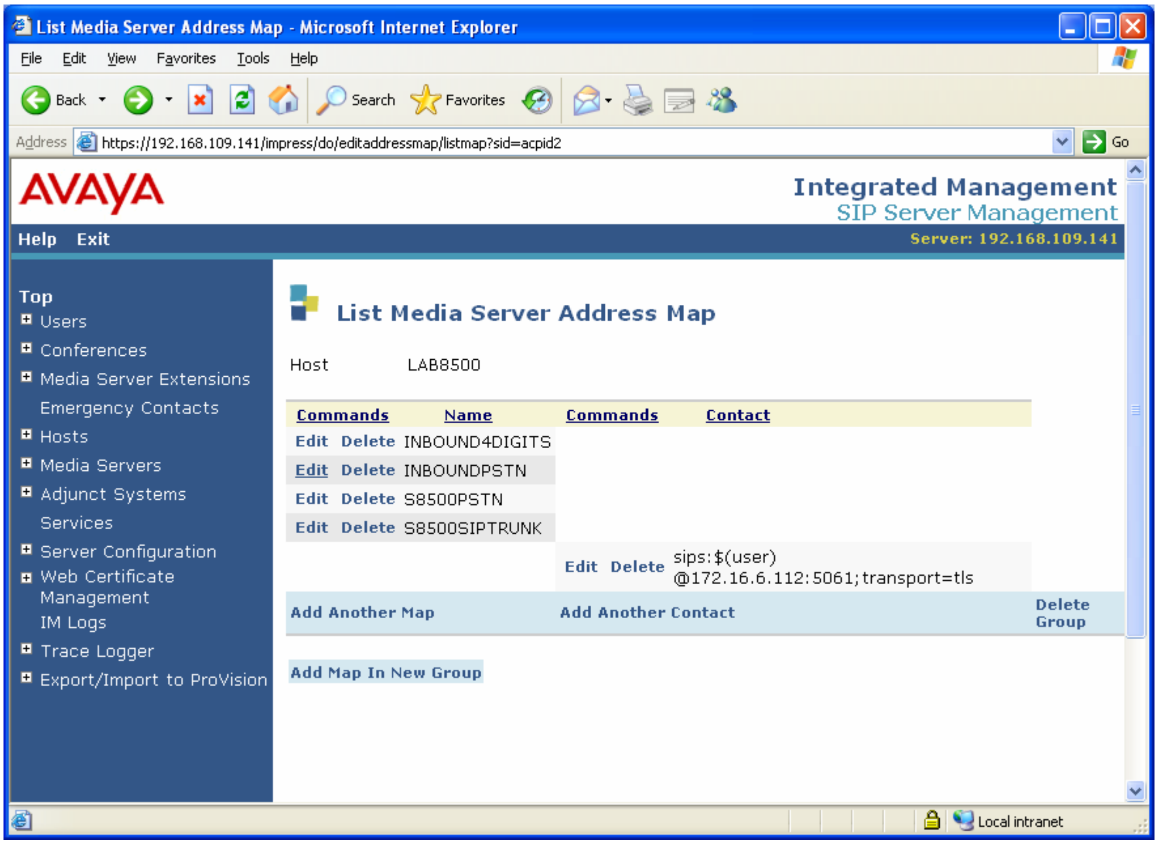
Next you can select the “edit” option to get the following screen.



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.

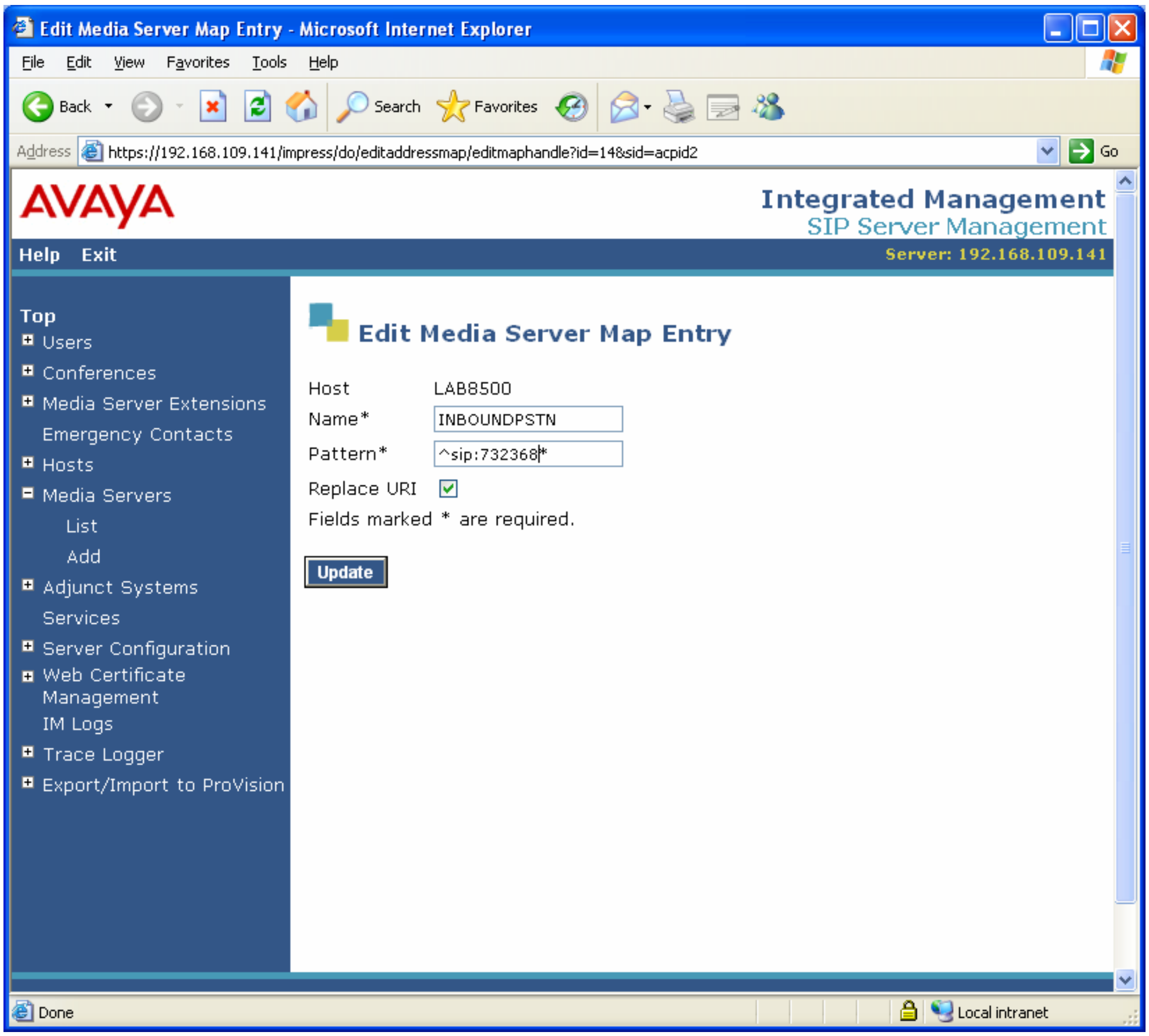
4.3.3.3 Media Server Address Map Parameters

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



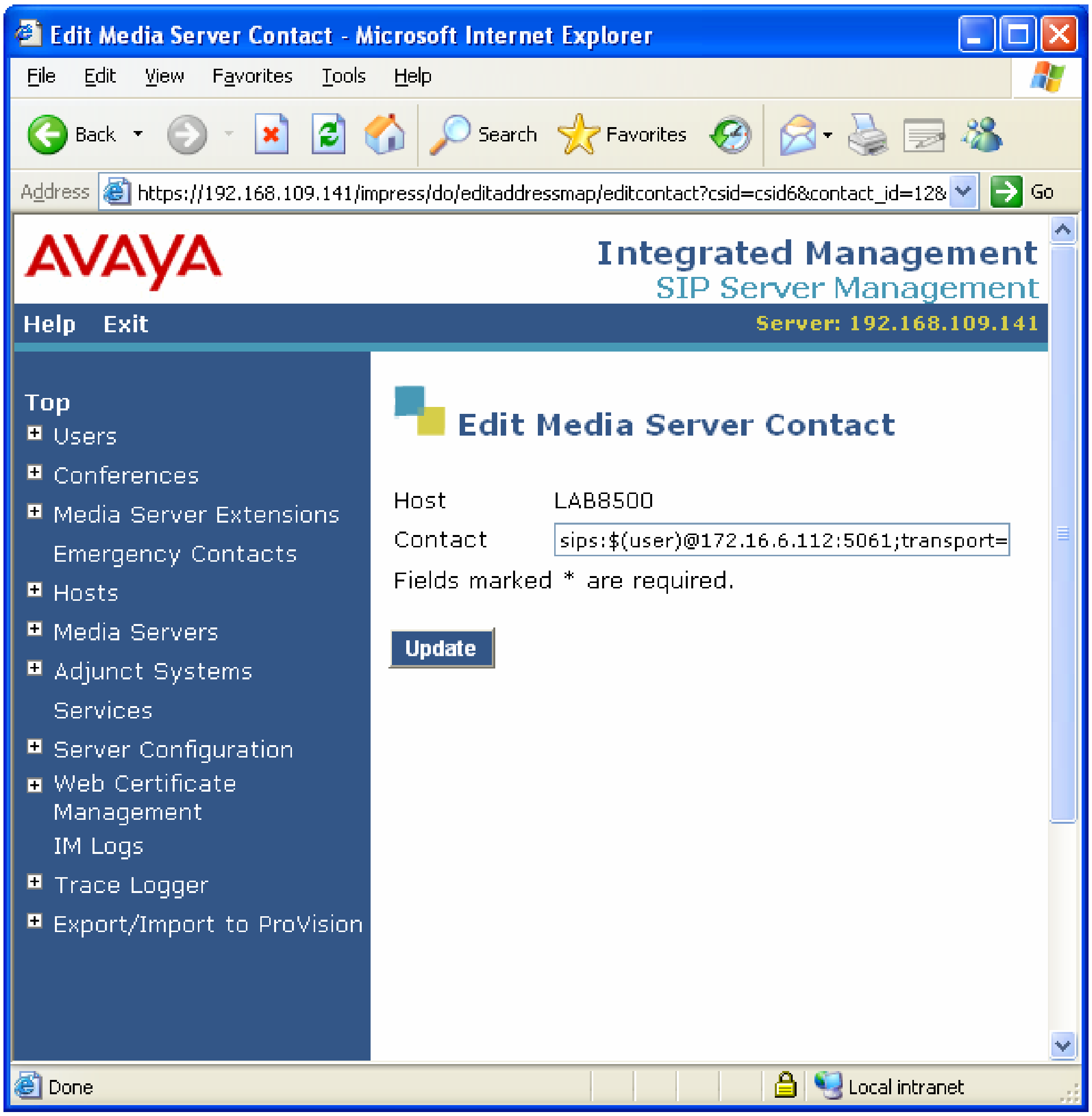
4.3.3.4 Sending Inbound Calls to the CLAN

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 calls dialed with the 732368 prefix to the CLAN in the CM.



4.3.3.5 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.

4.3.4 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>

This Customer Configuration Guide ("CCG") is offered as a convenience to AT&T's customers. The specifications and information regarding the product in this CCG are subject to change without notice. All statements, information, and recommendations in this CCG are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided “AS IS”. Users must take full responsibility for the application of the specifications and information in this CCG.

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