Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the AudioCodes MP-118 Analog VoIP Gateway with Avaya SIP Enablement Services and Avaya Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring the AudioCodes MP-118 Analog VoIP Gateway with Avaya SIP Enablement Services and Avaya Communication Manager.

The AudioCodes MP-118 Analog VoIP Gateway serves as a gateway between legacy analog endpoints/trunks at a branch location and a VoIP infrastructure at a main location using the Session Initiation Protocol (SIP). The MP-118 has 4 FXS (analog endpoint) ports and 4 FXO (POTS trunk) ports. The compliance test focused on the interoperability of the FXS ports. The FXO ports were configured only as a failover path to the PSTN if the data WAN is unavailable and SIP calls can not be made. The ability of these FXO ports to provide local PSTN access for the branch as part of normal operation is not covered by these Application Notes.

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

1. Introduction

These Application Notes describe the procedure for configuring the AudioCodes MP-118 Analog VoIP Gateway with Avaya SIP Enablement Services (SES) and Avaya Communication Manager.

The AudioCodes MP-118 Analog VoIP Gateway serves as a gateway between legacy analog endpoints/trunks at a branch location and a VoIP infrastructure at a main location using the Session Initiation Protocol (SIP). The MP-118 has 4 FXS (analog endpoint) ports and 4 FXO (POTS trunk) ports. The compliance test focused on the interoperability of the FXS ports. The FXO ports were configured only as a failover path to the PSTN if the data WAN is unavailable and SIP calls can not be made. The ability of these FXO ports to provide local PSTN access for the branch as part of normal operation is not covered by these Application Notes. The data WAN being unavailable and causing a failover can be due to three types of failures at the branch: a failure of the WAN link, a failure of the LAN providing access to the WAN or power loss of the MP-118. In the case of power loss, the FXS ports are connected directly to the corresponding FXO ports (i.e., FXS port 1 is connected to FXO port 1, FXS port 2 is connected to FXO port 2, etc.).

The MP-118 registers with the Avaya SES as a SIP endpoint for each analog endpoint connected to it. When a call is placed from an analog telephone, the MP-118 will send SIP signaling messages to the Avaya SES to setup the call. Once the call has been setup, the MP-118 converts the analog signal from the analog telephone to a series of voice samples sent in data packets over the data network using the Real Time Protocol (RTP).

1.1. Configuration

Figure 1 illustrates the configuration used in these Application Notes. In the sample configuration, two sites are connected via an IP network. The main site has an Avaya SES and an Avaya S8300 Media Server running Avaya Communication Manager in an Avaya G350 Media Gateway. Endpoints include an Avaya 4600 Series IP Telephone (with SIP firmware), an Avaya 4600 Series IP Telephone (with H.323 firmware), an Avaya 6408D Digital Telephone, and a fax machine. An ISDN-PRI trunk connects the media gateway to the PSTN.

The branch site has an AudioCodes MP-118 Analog VoIP Gateway with two analog telephones, and a fax machine. The branch site also has two Avaya 4600 Series IP Telephones (with SIP firmware). The MP-118 connects the branch site to the PSTN via an FXO (POTS) trunk which is only used if the data WAN is unavailable. The other three FXO ports were not connected. Normally, all outbound calls from the branch location pass across the data WAN, through the Main site and reach the PSTN via an ISDN-PRI trunk at the Main site. Conversely, all inbound calls for the branch site come from the PSTN across a trunk connected to the Main site then passed over the data WAN to the branch.

All SIP telephones and analog telephones at both sites are registered directly to Avaya SES and are administered as Outboard Proxy SIP (OPS) stations in Avaya Communication Manager. The SIP telephones at the branch site use the IP network gateway as the default gateway. The MP-118 only supports the operation of the analog endpoints and is not required for the SIP telephones. As a result, if the data WAN is unavailable, the MP-118 allows the analog telephones to keep functioning but the SIP telephones will not.

One of the DID numbers of the ISDN-PRI trunk to the Main Site is mapped to a telephone extension at the Main Site. The other is mapped to a telephone extension at the Branch Site. The DID number of the POTS line is mapped to an extension at the Branch Site which could be used if the data WAN is unavailable.

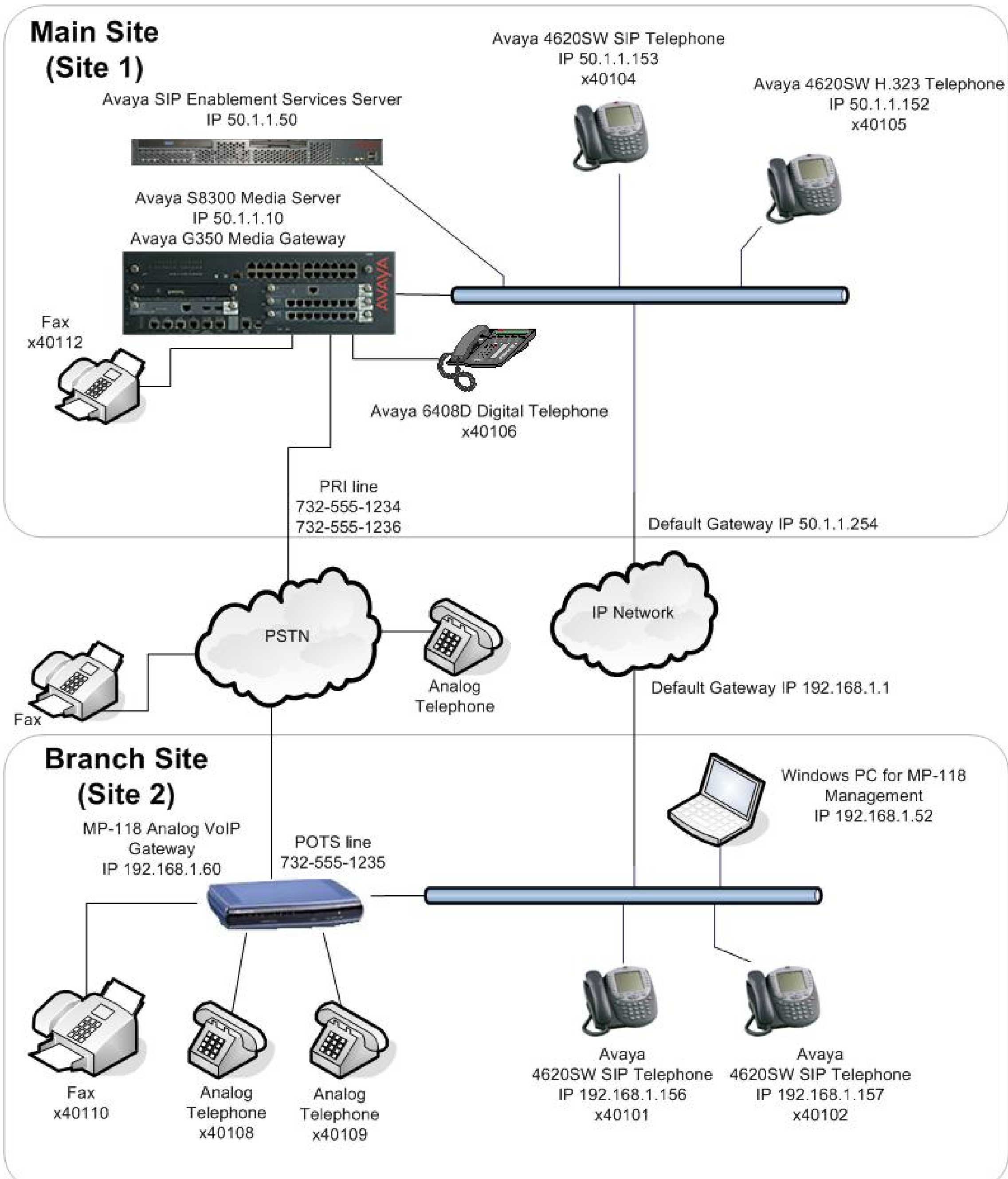


Figure 1: MP-118 Test Configuration

2. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration provided:

Equipment	Software/Firmware
Avaya S8300 Media Server with Avaya G350 Media Gateway	Avaya Communication Manager 3.1.2 (R013x.01.2.635.0) This is a post-GA build needed to support T.38 fax with the MP-118.
Avaya SIP Enablement Services (SES)	3.1 (build 18)
Avaya 4620SW IP Telephones	SIP version 2.2.2 H.323 version 2.3
Avaya 6408D Digital Telephone	-
Analog Telephones	-
Analog Fax Machines	-
Windows PCs	Windows XP Professional
AudioCodes MP-118 Analog VoIP Gateway	5.00A.011.008

3. Configure Avaya Communication Manager

The communication between Avaya Communication Manager and Avaya SES is via a SIP trunk group. All SIP signaling for calls between Avaya Communication Manager and the MP-118 passes through Avaya SES via this trunk group. This section describes the steps for configuring this trunk group and associated signaling group. In addition, this section describes the configuration of stations as OPS stations, which is required for each analog telephone, and fax machine connected to the MP-118.

The following configuration of Avaya Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration in this section, perform a **save translation** command to make the changes permanent.

Step	Description

1.	Use the **display system-parameters customer-options** command to verify that sufficient SIP trunk capacity exists. On Page 2, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to another SIP telephone will use two SIP trunks. A call between a non-SIP telephone and a SIP telephone will only use one trunk. In this solution, each analog endpoint at the branch counts as a SIP telephone.
	
	The license file installed on the system controls the maximum permitted. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.
	
	`display system-parameters customer-options                      Page   2 of  10`
	`                                OPTIONAL FEATURES`
	
	`IP PORT CAPACITIES                                            USED`
	`                     Maximum Administered H.323 Trunks: 100   10`
	`           Maximum Concurrently Registered IP Stations: 20    0`
	`             Maximum Administered Remote Office Trunks: 0     0`
	`Maximum Concurrently Registered Remote Office Stations: 0     0`
	`              Maximum Concurrently Registered IP eCons: 0     0`
	`  Max Concur Registered Unauthenticated H.323 Stations: 0     0`
	`                  Maximum Video Capable H.323 Stations: 0     0`
	`                   Maximum Video Capable IP Softphones: 0     0`
	`                       Maximum Administered SIP Trunks: 100   24`
	
	`   Maximum Number of DS1 Boards with Echo Cancellation: 0     0`
	`                             Maximum TN2501 VAL Boards: 0     0`
	`                    Maximum G250/G350/G700 VAL Sources: 5     1`
	`           Maximum TN2602 Boards with 80 VoIP Channels: 0     0`
	`          Maximum TN2602 Boards with 320 VoIP Channels: 0     0`
	`   Maximum Number of Expanded Meet-me Conference Ports: 10    0`
	
	`        (NOTE: You must logoff & login to effect the permission changes.)`

2.	Use the **change node-name ip** command to assign the node name and IP address for Avaya SES at the enterprise site. In this case, ***SES*** and ***50.1.1.50*** are being used, respectively. The node name ***SES*** will be used throughout the other configuration forms of Avaya Communication Manager. In this example, ***procr*** and ***50.1.1.10*** are the name and IP address assigned to the Avaya S8300 Media Server.
	
	`change node-names ip                                             Page   1 of   1`
	`                                  IP NODE NAMES`
	`    Name``              `**`IP Address`**`         Name              IP Address`
	`SES``                 `**`50 .1  .1  .50`**`                              .   .   .`
	`default             0  .0  .0  .0                               .   .   .`
	`procr``               `**`50 .1  .1  .10`**`                              .   .   .`

Step	Description
3.	Use the **change ip-network-region *n*** command, where ***n*** is the number of the region to be changed, to define the connectivity settings for all VoIP resources and IP endpoints within the region. Select an IP network region that will contain the Avaya SES server. The association between this IP network region and the Avaya SES server will be done on the **Signaling Group** form as shown in Step 6. In the case of the compliance test, the same IP network region that contains the Avaya S8300 Media Server and Avaya IP Telephones was selected to contain the Avaya SES server. By default, the Media Server and IP telephones are in IP Network Region 1. On the **IP Network Region** form: ▪ The **Authoritative Domain** field is configured to match the domain name configured on Avaya SES. In this configuration, the domain name is ***devcon.com***. This name will appear in the "From" header of SIP messages originating from this IP region. ▪ By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G350 Media Gateway. This is true for both intra-region and inter-region IP-IP Direct Audio. Shuffling can be further restricted at the trunk level on the **Signaling Group** form. ▪ The **Codec Set** is set to the number of the IP codec set to be used for calls within this IP network region. If different IP network regions are used for the Avaya S8300 Media Server and the Avaya SES server, then Page 3 of each **IP Network Region** form must be used to specify the codec set for inter-region communications. ▪ The **Audio PHB Value** is ***46,*** which translates to a DiffServ header value of 0xb8. ▪ The default values can be used for all other fields.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
Region: 1
Location: 1       Authoritative Domain: devcon.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3027
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 34      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                         RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Step	Description

4.	Use the **change ip-codec-set *n*** command, where ***n*** is the codec set value specified in Step 3, to enter the supported audio codecs for calls routed to Avaya SES. Multiple codecs can be listed in priority order to allow the codec to be negotiated during call establishment. The list should include the codecs the enterprise wishes to support within the normal trade-off of bandwidth versus voice quality. The example below shows the values used in the compliance test.
	`change ip-codec-set 1                                          Page   1 of   2`
	`                          IP Codec Set`
	`    Codec Set: 1`
	`    Audio        Silence      Frames   Packet`
	`    Codec        Suppression  Per Pkt  Size(ms)`
	` 1: G.711MU          n         2        20`
	` 2: G.729AB          n         2        20`
	` 3:`

5.	On Page 2, the **FAX Mode** field must be set to ***t.38-standard*** to support the fax machines. The **Modem** field should be set to ***off***. The screen below shows the setting used for the fax testing.
	`change ip-codec-set 1                                          Page   2 of   2`
	`                          IP Codec Set`
	`                              Allow Direct-IP Multimedia? n`
	`                    Mode                Redundancy`
	`    FAX             t.38-standard        0`
	`    Modem           off                  0`
	`    TDD/TTY         US                   3`
	`    Clear-channel   n                    0`

Step	Description
6.	Use the **add signaling group *n*** command, where ***n*** is the number of an unused signaling group, to create the SIP signaling group as follows: ▪ Set the **Group Type** field to ***sip***. ▪ The **Transport Method** field will default to ***tls*** (Transport Layer Security). TLS is the only link protocol that is supported for communication between Avaya SES and Avaya Communication Manager. ▪ Specify the Avaya S8300 Media Server (node name ***procr***) and the Avaya SES Server (node name ***SES***) as the two ends of the signaling group in the **Near-end Node Name** and the **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in Step 2. For alternative configurations that use a C-LAN board, the near (local) end of the SIP signaling group will be the C-LAN board instead of the Media Server. ▪ Ensure that the recommended TLS port value of ***5061*** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields. ▪ In the **Far-end Network Region** field, enter the IP network region value assigned in the **IP Network Region** form in Step 3. This defines which IP network region contains the Avaya SES server. If the **Far-end Network Region** field is different from the near-end network region, the preferred codec will be selected from the IP codec set assigned for the inter-region connectivity for the pair of network regions. ▪ Enter the domain name of Avaya SES in the **Far-end Domain** field. In this configuration, the domain name is ***devcon.com***. This domain is specified in the Uniform Resource Identifier (URI) of the SIP "To" header in the INVITE message. ▪ The **Direct IP-IP Audio Connections** field is set to ***y***. ▪ The **DTMF over IP** field must be set to the default value of ***rtp-payload*** for a SIP trunk. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833. ▪ The default values for the other fields may be used.

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

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




   Near-end Node Name: procr                 Far-end Node Name: SES
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
       Far-end Domain: devcon.com

                                            Bypass If IP Threshold Exceeded? n

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

Step	Description
7.	Add a SIP trunk group by using the **add trunk-group *n*** command, where ***n*** is the number of an unused trunk group. For the compliance test, trunk group number 1 was chosen. On Page 1, set the fields to the following values: ▪ Set the **Group Type** field to ***sip***. ▪ Choose a descriptive **Group Name**. ▪ Specify an available trunk access code (**TAC**) that is consistent with the existing dial plan. ▪ Set the **Service Type** field to ***tie***. ▪ Specify the signaling group associated with this trunk group in the **Signaling Group** field as previously specified in Step 6. ▪ Specify the **Number of Members** supported by this SIP trunk group. As mentioned earlier, each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to another SIP telephone will use two SIP trunks. A call between a non-SIP telephone and a SIP telephone will only use one trunk. In this solution, each analog endpoint at the branch counts as a SIP telephone. ▪ The default values may be retained for the other fields.

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

Group Number: 1                  Group Type: sip          CDR Reports: y
  Group Name: To SES 50.1.1.50            COR: 1       TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                        Night Service:
Queue Length: 0
Service Type: tie                 Auth Code? n

                                                     Signaling Group: 1
                                                   Number of Members: 24
```

Step	Description
8.	On Page 3: ▪ Verify the **Numbering Format** field is set to ***public***. This field specifies the format of the calling party number sent to the far-end. ▪ The default values may be retained for the other fields.

```
add trunk-group 1                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y



                     Numbering Format: public
                                                 Prepend '+' to Calling Number? n


                                                   Replace Unavailable Numbers? n
```

Step	Description
9.	Use the **change public-unknown-numbering 0** command to define the full calling party number to be sent to the far-end. Add an entry for the trunk group defined in Step 7. In the example shown below, all calls originating from a 5-digit extension beginning with 4 and routed across trunk group 1 will be sent as a 5 digit calling number. This calling party number will be sent to the far-end in the SIP "From" header.

```
change public-unknown-numbering 0                              Page   1 of   2
                      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

 5 4       1                         5
```

Step	Description
10.	Create a route pattern that will use the SIP trunk that connects to Avaya SES. In general, a route pattern is not required for calling between SIP endpoints registered to the Avaya SES. This includes the dialing scenarios performed in the compliance test. However, some transfer scenarios using alpha-numeric handles (i.e., user names) instead of extensions require a default route pattern. The creation of this default route pattern is included here for completeness. To create a route pattern, use the **change route-pattern *n*** command, where ***n*** is the number of an unused route pattern. Enter a descriptive name for the **Pattern Name** field. Set the **Grp No** field to the trunk group number created for the SIP trunk. Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of ***0*** is the least restrictive level. The default values may be retained for all other fields.

```
change route-pattern 1                                         Page   1 of   3
                    Pattern Number: 3   Pattern Name: SIP
                             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                                                           n   user
 2:                                                                  n   user
 3:                                                                  n   user
 4:                                                                  n   user
 5:                                                                  n   user
 6:                                                                  n   user

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

Step	Description

11.	Use the **change locations** command to assign the default SIP route pattern to the location. In the compliance test, all SIP endpoints whether at the main or branch site are part of a single location defined in Avaya Communication Manager. This location uses the default name of ***Main*** and is shown in the example below. Enter the route pattern number from the previous step in the **Proxy Sel. Rte. Pat.** field. The default values may be retained for all other fields.
	
	`change locations                                          Page   1 of   4`
	`LOCATIONS`
	`ARS Prefix 1 Required For 10-Digit NANP Calls? y`
	`Loc.  Name           Timezone Rule  NPA  ARS   Attd         Pre-    Proxy Sel.`
	`No.                   Offset             FAC   FAC          fix     Rte. Pat.`
	`1:   Main            + 00:00   0                                   1`
	`2:`
	`3:`

12.	All SIP stations are configured as OPS stations on Avaya Communication Manager. This includes the analog telephones, and fax machine connected to the MP-118, which appear as SIP stations to Avaya Communication Manager.
	
	Use the **display system-parameters customer-options** command to verify Avaya Communication Manager has sufficient OPS capacity available to add the OPS stations needed for the SIP and analog endpoints at the branch office in **Figure 1**. If there is insufficient capacity, contact an authorized Avaya sales representative or business partner to make the appropriate changes.
	
	`display system-parameters customer-options                 Page   1 of  10`
	`OPTIONAL FEATURES`
	`G3 Version: V13`
	`Location: 1                          RFA System ID (SID): 1`
	`Platform: 13                         RFA Module ID (MID): 1`
	`USED`
	`Platform Maximum Ports: 900   121`
	`Maximum Stations: 450   41`
	`Maximum XMOBILE Stations: 0     0`
	`Maximum Off-PBX Telephones - EC500: 50    0`
	`Maximum Off-PBX Telephones -   OPS: 50    23`
	`Maximum Off-PBX Telephones - SCCAN: 0     0`

Step	Description
13.	To add a station, use the **add station *n*** command where ***n*** is an unused extension number. Use the default value of ***6408D+*** for the **Type** field. Enter an ***X*** in the **Port** field. This indicates a station is being added without identifying a physical port for the station to use. Enter a descriptive name in the **Name** field. The default values may be retained for all other fields.

```
add station 40108                                               Page   1 of   4
                                     STATION

Extension: 40108                          Lock Messages? n          BCC: 0
     Type: 6408D+                         Security Code:             TN: 1
     Port: X                            Coverage Path 1: 1          COR: 1
     Name: Branch 1                     Coverage Path 2:            COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 2             Personalized Ringing Pattern: 1
             Data Module? n                         Message Lamp Ext: 40108
            Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: english

                                                   Media Complex Ext:
                                                        IP SoftPhone? n
```

Step	Description
14.	On Page 2, set **Restrict Last Appearance** to ***n***. This will allow the last call appearance to be used for either an incoming or outgoing call.

```
add station 40108                                               Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: audix           Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                    Idle Appearance Preference? n
 Per Button Ring Control? n                  Bridged Idle Line Preference? n
   Bridged Call Alerting? y                      Restrict Last Appearance? n
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic                    Audible Message Waiting? n
    MWI Served User Type:                      Display Client Redirection? n
             AUDIX Name: IA770                Select Last Used Appearance? n
                                                 Coverage After Forwarding? s

                                              Direct IP-IP Audio Connections? y
 Emergency Location Ext: 40108                         IP Audio Hairpinning? n
```

Step	Description

15.	On Page 3, under **BUTTON ASSIGNMENTS**, create the appropriate number of call appearances for the SIP endpoint being configured. In general, the appropriate number of call appearances on Avaya Communication Manager is the same as the number of call appearances supported by the endpoint. To create a call appearance, enter ***call-appr*** as the button assignment. The example below shows the configuration of one of the analog endpoints connected to the MP-118. The analog endpoints used in the compliance test were all configured with two call appearances.
	
	There are two Feature Name Extensions (FNEs) that require the assignment of feature buttons in order to operate. The Automatic Callback FNE requires the assignment of an auto-cback button. The Conference On Answer FNE requires the assignment of a no-hld-cnf button. Both of these button assignments are shown in the example below.
	<pre>add station 40108 Page 3 of 4
	                                     STATION
	 SITE DATA
	       Room: Headset? n
	       Jack: Speaker? n
	      Cable: Mounting: d
	      Floor: Cord Length: 0
	   Building: Set Color:
	
	ABBREVIATED DIALING
	     List1: List2: List3:
	
	BUTTON ASSIGNMENTS
	 1: call-appr 5: auto-cback
	 2: call-appr 6: no-hld-cnf
	 3: 7:
	 4: 8:</pre>

16.	Map the Avaya Communication Manager extension to the Avaya SES media server extension defined in Section 4, Step 8 with the **add off-pbx-telephone station-mapping** command. Enter the values as shown below:
	
	▪ **Station Extension**: Avaya Communication Manager extension
	▪ **Application**: ***OPS***
	▪ **Phone Number**: Avaya SES media server extension
	▪ **Trunk Selection**: The SIP trunk group number
	▪ **Configuration Set**: Enter a valid configuration set. The compliance test used configuration set 1 which contained the default values.
	<pre>add off-pbx-telephone station-mapping Page 1 of 2
	                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
	
	 Station Application Dial Phone Number Trunk Configuration
	 Extension Prefix Selection Set
	 40108 OPS - 40108 1 1
	                               -</pre>

Step	Description

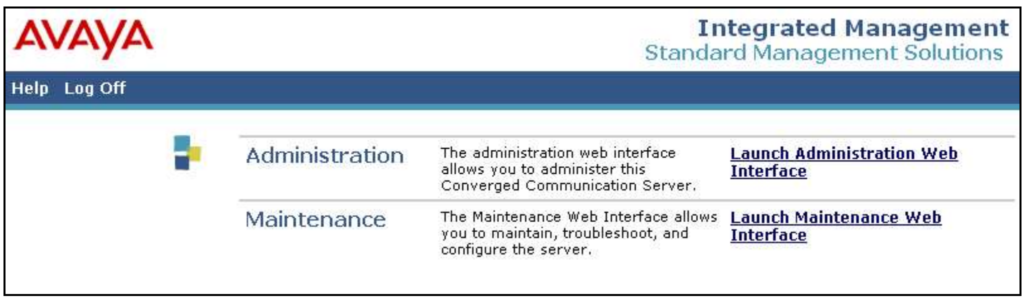
17.	On Page 2, set the **Call Limit** to the number of call appearances set on the station form in Step 15. Verify that the **Mapping Mode** is set to ***both***. This setting allows the OPS station to both originate and terminate calls.
	
	`add off-pbx-telephone station-mapping                          Page   2 of   2`
	`                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION`
	
	` Station       Call       Mapping      Calls       Bridged`
	` Extension     Limit      Mode         Allowed     Calls`
	` 40108        2         both          all         both`

18.	Repeat Steps 13 -17 for each remaining endpoint located at the branch office. The branch office has five user endpoints: two analog telephones connected to the MP-118 (x40108 and x40109), two Avaya 4600 Series SIP Telephones (x40101 and x40102), and a fax machine (x40110). In this configuration, the MP-118 gateway itself must also be registered as a separate endpoint. For the compliance test, extension 44444 was created for this purpose with the same characteristics as the Avaya SIP telephones at this site.

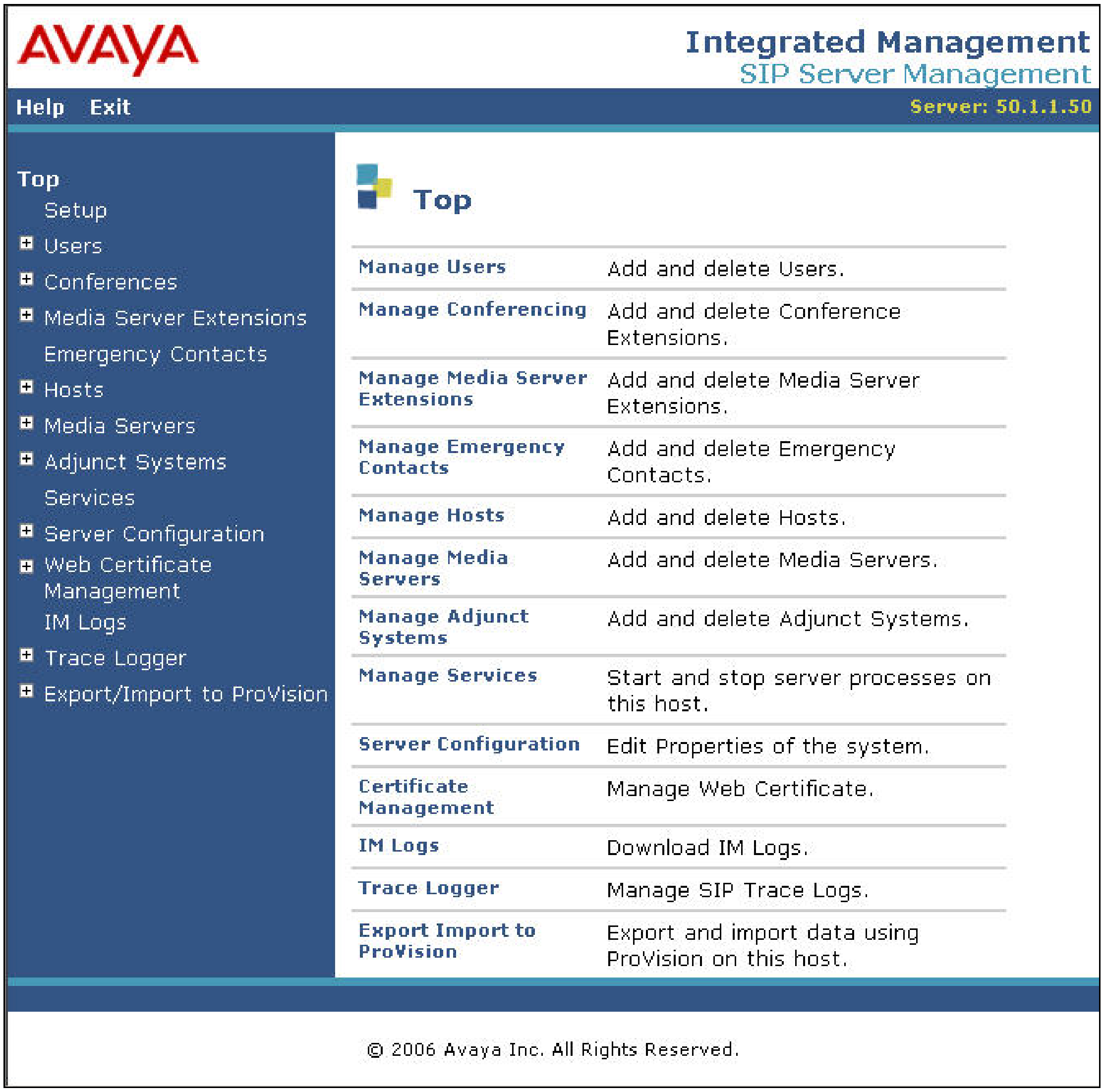
19.	To map a DID number to a station at the main or branch office, use the **change inc-call-handling-trmt trunk-group *n*** command, where ***n*** is the trunk group number connected to the PSTN from the Avaya G350 Media Gateway. The compliance test used trunk group 2 to connect to the PSTN. This trunk group configuration is not shown in these Application Notes. The example below shows two incoming 11-digit numbers being deleted and replaced with the extension number of the desired station.
	
	`change inc-call-handling-trmt trunk-group 2                    Page   1 of   3`
	`                       INCOMING CALL HANDLING TREATMENT`
	` Service/      Called    Called       Del  Insert`
	` Feature        Len      Number`
	` tie             11  17325551234      11   40104`
	` tie             11  17325551236      11   40108`

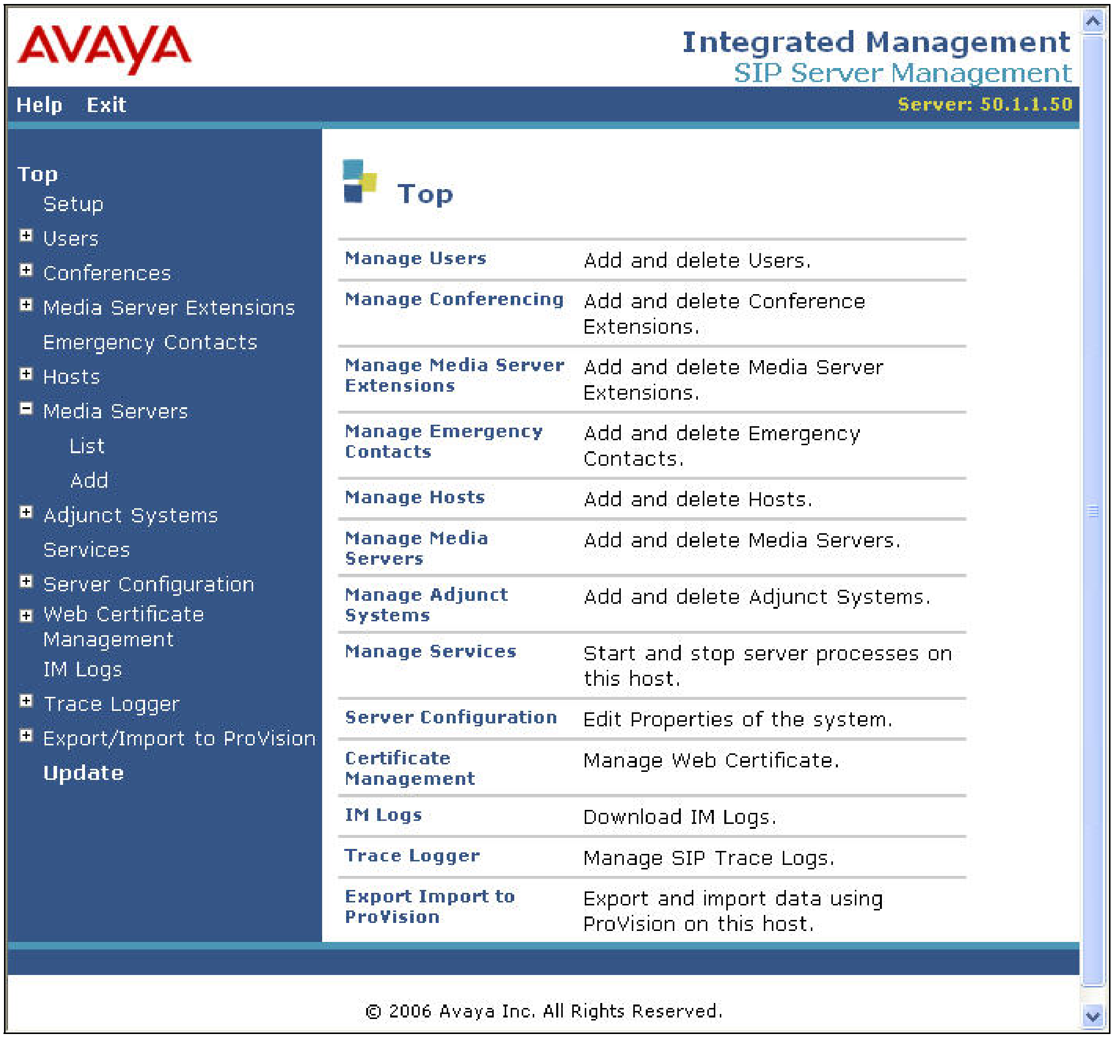
4. Configure Avaya SES

This section covers the configuration of Avaya SES. Avaya SES is configured via an Internet browser using the administration web interface. It is assumed that Avaya SES software and the license file have already been installed on the server. During the software installation, the installation script is run from the Linux shell of the server to specify the IP network properties of the server along with other parameters. For additional information on these installation tasks, refer to [5].

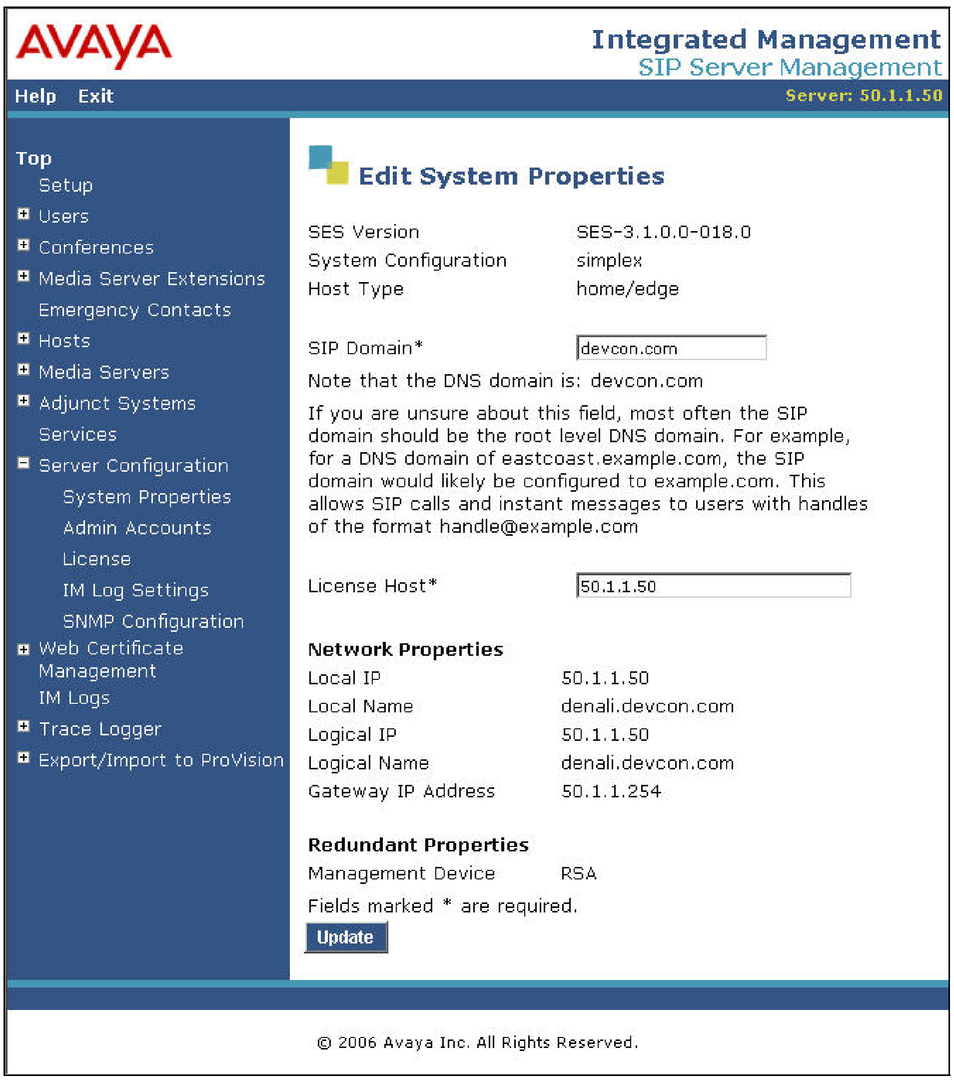
Step	Description
1.	Access the Avaya SES administration web interface by entering http://<*ip-addr*>/admin as the URL in an Internet browser, where <*ip-addr*> is the IP address of the Avaya SES server. Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main page as shown below. 

Step	Description
2.	The Avaya SES Administration Home Page will be displayed as shown below.



Step	Description
3.	After making changes within Avaya SES, it is necessary to commit the database changes using the **Update** link that appears when changes are pending. Perform this step by clicking on the **Update** link found in the bottom of the blue navigation bar on the left side of any of the Avaya SES administration pages as shown below. It is recommended that this be done after making each set of changes described in the following steps. 

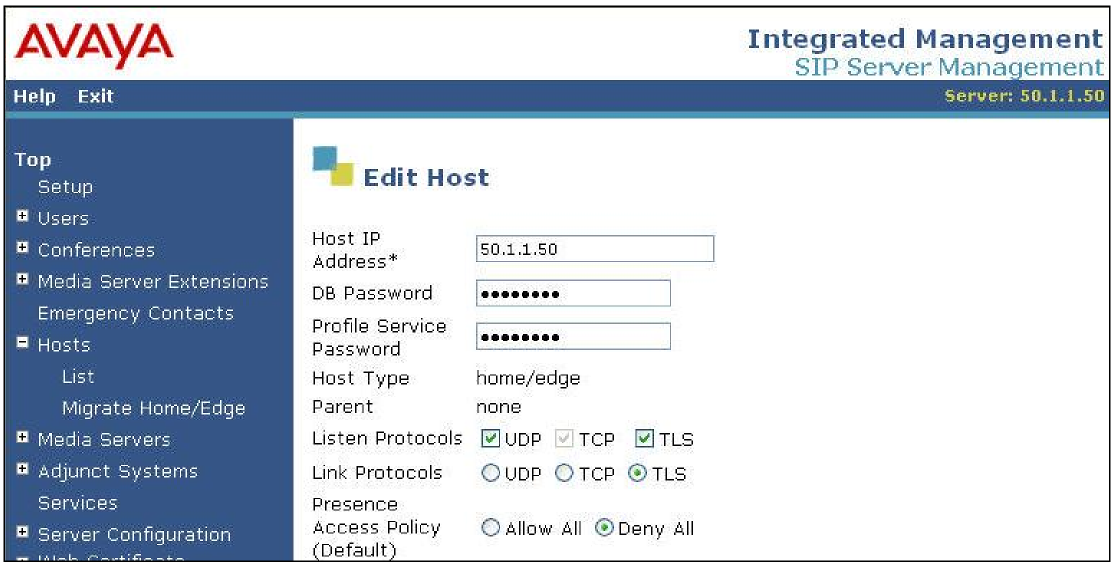
Step	Description
4.	From the left pane of the administration web interface, expand the **Server Configuration** option and select **System Properties**. The **Edit System Properties** page displays the software version in the **SES Version** field and the network properties entered during the installation process. On the **Edit System Properties** page: ▪ Enter the **SIP Domain** name assigned to Avaya SES. This must match the **Authoritative Domain** field configured on Avaya Communication Manager shown in Section 3, Step 3. ▪ Enter the **License Host** field. This is the host name, the fully qualified domain name, or the IP address of the SIP proxy server that is running the WebLM application and has the associated license file installed. ▪ After configuring the **Edit System Properties** page, click the **Update** button.



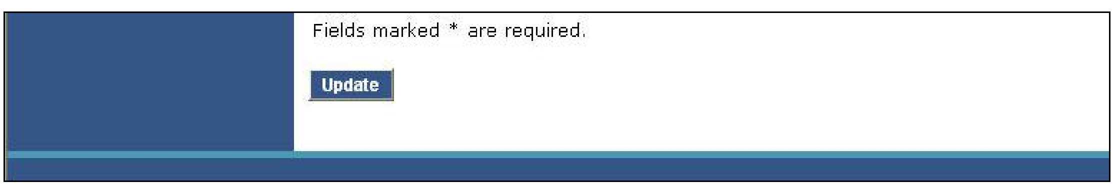
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>
After setting up the domain on the Edit System Properties page, create a host computer entry for Avaya SES. The following example shows the Edit Host page since the host had already been added to the system.

The Edit Host page shown below is accessible by clicking on the Hosts → List link in the left pane and then clicking on the Edit link under the Commands section of the subsequent page that is displayed.

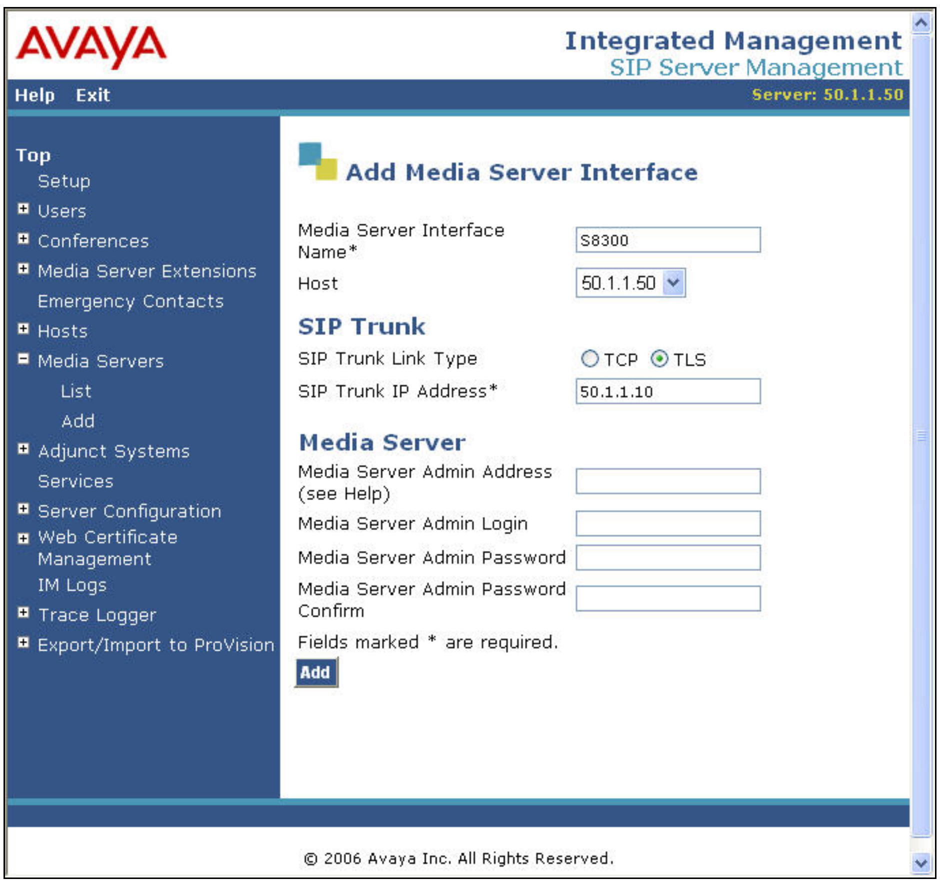
In the Host IP Address field, enter the IP address of the Avaya SES.
Enter the DB Password that was specified during the system installation.
The default values for the other fields may be used.

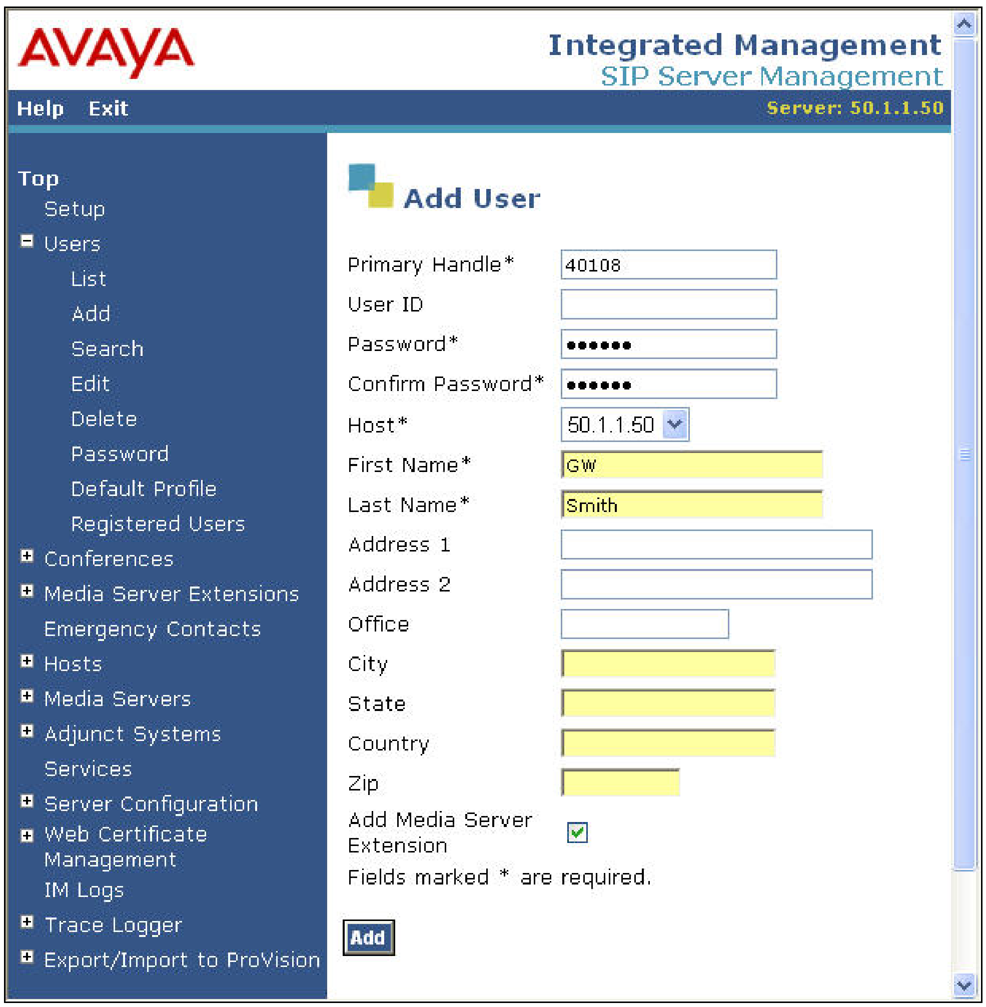


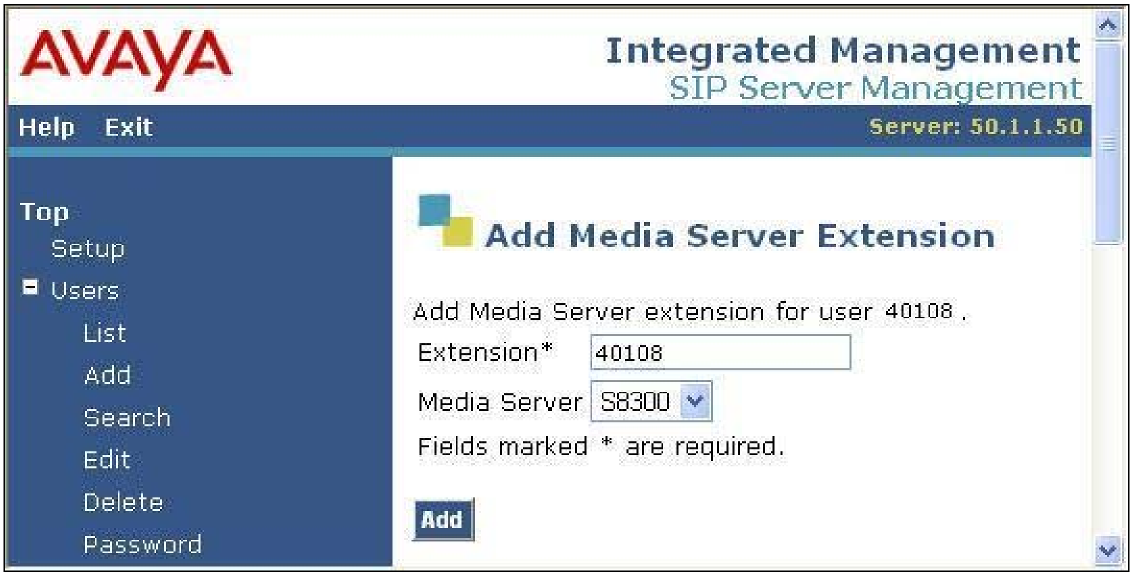
Scroll down to the bottom of the page and click the Update button.



</td></tr>
</table>

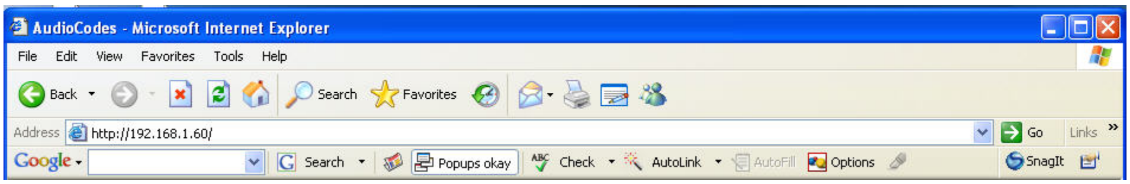
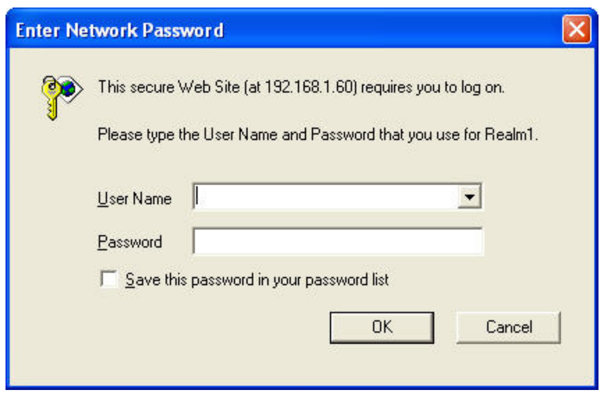
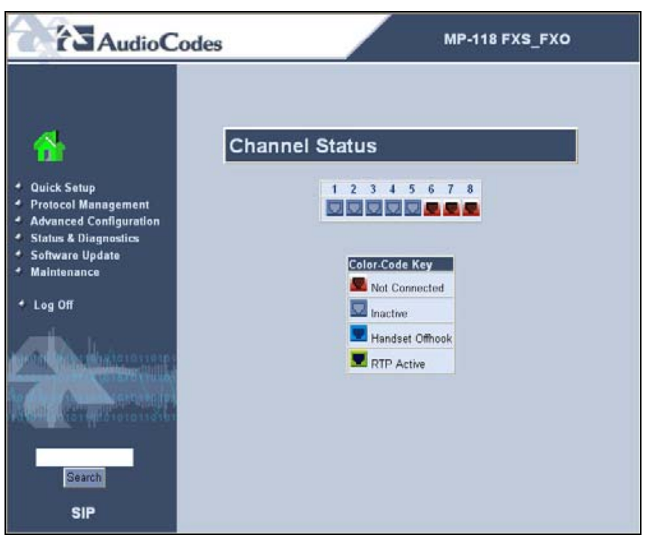
Step	Description
6.	From the left pane of the administration web interface, expand the **Media Servers** option and select **Add** to add the Avaya Media Server to the list of media servers known to Avaya SES. Adding the media server will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager. On the **Add Media Server Interface** page, enter the following information: ▪ A descriptive name in the **Media Server Interface Name** field (e.g. S8300). ▪ In the **Host** field, select the Avaya SES server from the pull-down menu that will serve as the SIP proxy for this media server. Since there is only one Avaya SES server in this configuration, the **Host** field is set to the host shown in Step 5. ▪ Select ***TLS*** (Transport Link Security) for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer. TLS is the only link protocol that is supported for communication between Avaya SES and Avaya Communication Manager. ▪ Enter the IP address of the Avaya S8300 Media Server in the **SIP Trunk IP Address** field. In alternative configurations that use a C-LAN board, the **SIP Trunk IP Address** would be the IP address of the C-LAN board. ▪ The default values may be retained for all other fields. ▪ After completing the **Add Media Server Interface** page, click the **Add** button. 

Step	Description
7.	A user must be added on Avaya SES for each of the extensions at the branch office created on Avaya Communication Manager in Section 3, Steps 13 – 17 including the extension created for the MP-118 gateway itself. From the left pane, navigate to **Users** → **Add**. Enter the values as shown below. ▪ **Primary Handle**: Enter the extension for this user. ▪ **Password**: Enter a valid password for logging into the SIP endpoint. ▪ **Confirm Password**: Re-enter the password. ▪ **Host**: Select the Avaya SES server from the pull-down menu. ▪ **First Name**: Any descriptive name. ▪ **Last Name**: Any descriptive name. Check the **Add Media Server Extension** checkbox. Click the **Add** button to proceed. A confirmation window will appear. Click **Continue** on this new page to proceed. 

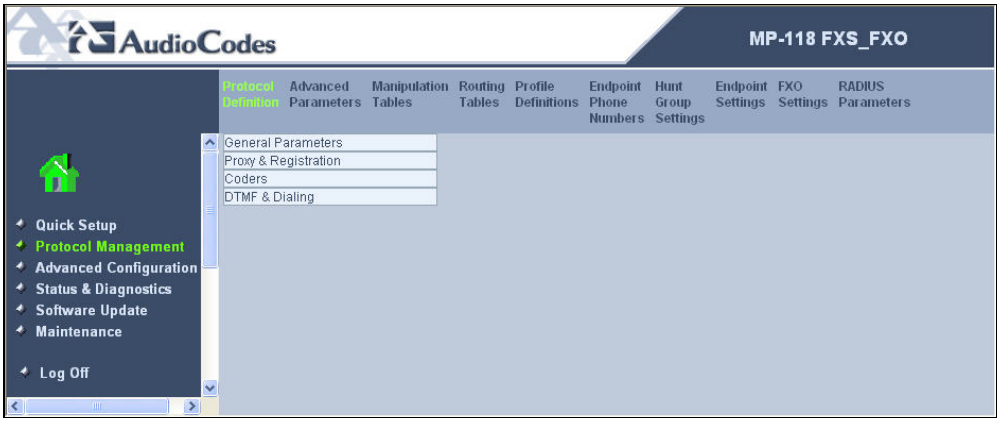
Step	Description
8.	The **Add Media Server Extension** page will appear. In the **Extension** field, enter the same extension used in the previous step. In the **Media Server** field, select from the pull-down menu the name of the media server added in Step 6. Click the **Add** button to complete the operation. 
9.	Repeat Steps 7 - 8 for each of the remaining stations at the branch office.

5. Configure the MP-118

This section describes the procedures for configuring the MP-118. These procedures assume the MP-118 has been installed using the procedures documented in [7] and has been assigned an IP address.

Step	Description
1.	The configuration of the MP-118 is done via a Web browser. To access the device, enter the IP address of the MP-118 in the **Address** field of the browser. 
2.	The following pop-up window will appear. Login with the proper credentials. 
3.	The MP-118 main page will appear as shown below. 

Step	Description
4.	The network settings that were configured during installation can be viewed by selecting **Advanced Configuration** in the left pane then navigating to **Network Settings → IP Settings** in the right pane. If necessary, changes can be made to the settings on this page followed by clicking **Submit**. For the compliance test, the **IP Address**, **Subnet Mask** and **Default Gateway Address** were set to values consistent with the test configuration shown in **Figure 1**. The **Media Premium QoS** must match the **Audio PHB Value** set on Avaya Communication Manager in Section 3, Step 3. Default values may be retained for all other values. 

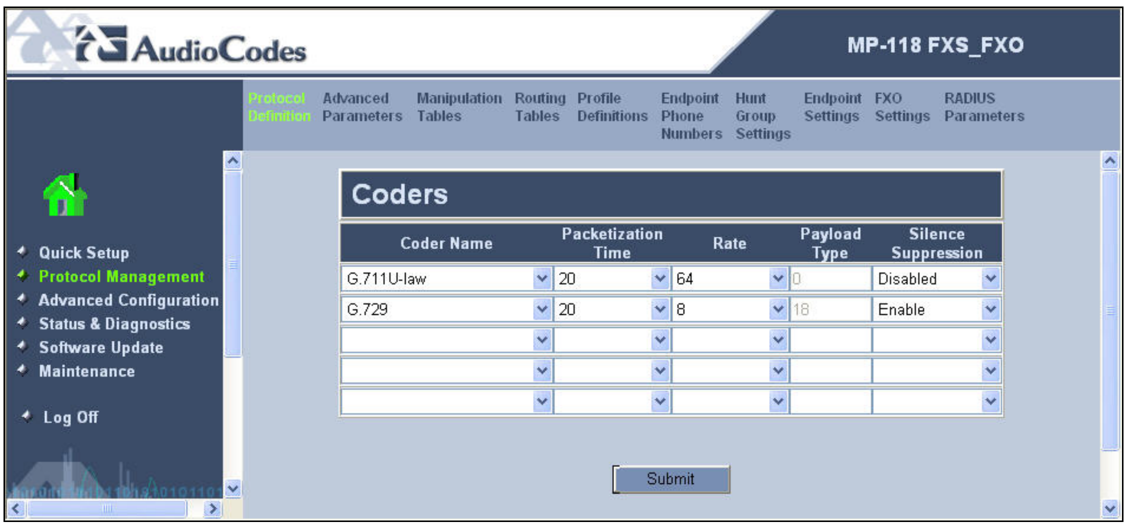
Step	Description
5.	**Protocol Definition** To access these parameters, select **Protocol Management** in the left pane and navigate to **Protocol Definition** in the right pane. The pull-down choices for **Protocol Definition** are shown below. 

Step	Description
6.	**Proxy and Registration** From the menu shown in Step 5, navigate to **Protocol Definition → Proxy & Registration**. Configure the parameters as described below. ▪ For the **Enable Proxy** field, select ***Use Proxy*** from the pull-down menu. ▪ In the **Proxy IP Address** field, enter the IP address of the Avaya SES. ▪ For the **Enable Proxy Keep Alive** field, select ***Using Options*** from the pull-down menu. The MP-118 will use the SIP OPTIONS message as a handshake mechanism with the Avaya SES to determine if the SIP connection is up. If the connection is down, the MP-118 will failover to the FXO ports. ▪ For the **Always Use Proxy** field, select ***Enabled***. ▪ For the **Send All Invite to Proxy** field, select ***Yes***. This directs the MP-118 to send all INVITE requests to the Avaya SES, including those generated as the result of a transfer or redirect. ▪ For the **Enable Registration** field, select ***Yes***. This will allow the MP-118 to register the FXS endpoints with the Avaya SES. ▪ In the **Registrar IP Address** field, enter the IP address of the Avaya SES. Default values may be retained for all other fields. Scroll down to continue configuring parameters on the lower half of the screen. 

Step	Description
7.	Configure the parameters as described below. ▪ In the **Gateway Name** field, enter the IP address of the Avaya SES. ▪ In this configuration, the MP-118 requires that a separate extension be assigned to the gateway itself. In the **User Name** field, enter the extension created for this purpose in Section 3, Step 18. ▪ In the **Password** field, enter the password for the above user. ▪ For the **Authentication Mode,** select ***Per Endpoint.*** The MP-118 will authenticate each endpoint separately. Click **Submit**. 

Step	Description
8.	**General Protocol Parameters** From the menu shown in Step 5, navigate to **Protocol Definition → General Parameters**. Configure the parameters as described below. ▪ For the **Enable Early Media** field, select ***Enabled***. If enabled, the MP-118 sends Session Description Protocol (SDP) information in the 18x responses allowing the media stream to be set-up prior to answering the call. ▪ If fax support is needed, select ***T.38 Relay*** as the **Fax Signaling Method**. ▪ Select ***No*** for the **Use "user=phone" in SIP URL** field. Default values may be retained for all other fields. Scroll down to the bottom of the page and click **Submit** (not shown). 

Step	Description
9.	**Codecs** From the menu shown in Step 5, navigate to **Protocol Definition → Coders**. In the screen below, select the list of preferred codecs to be used by the MP-118 with the most preferred codec at the top and working downward to the least preferred. This list must have an overlap with the list provided on Avaya Communication Manager in Section 3, Step 4. The codec is selected from the pull-down menu under the **Coder Name** field. The codec list used for the compliance test is shown in the example below. ***G.711U-law*** was selected as the most preferred followed by ***G.729***. For the G.729 codec, the **Silence Suppression** field was set to ***Enable***. With silence suppression enabled, the MP-118 uses the equivalent of a G.729AB codec. Default values were retained for all other fields. Click **Submit**.



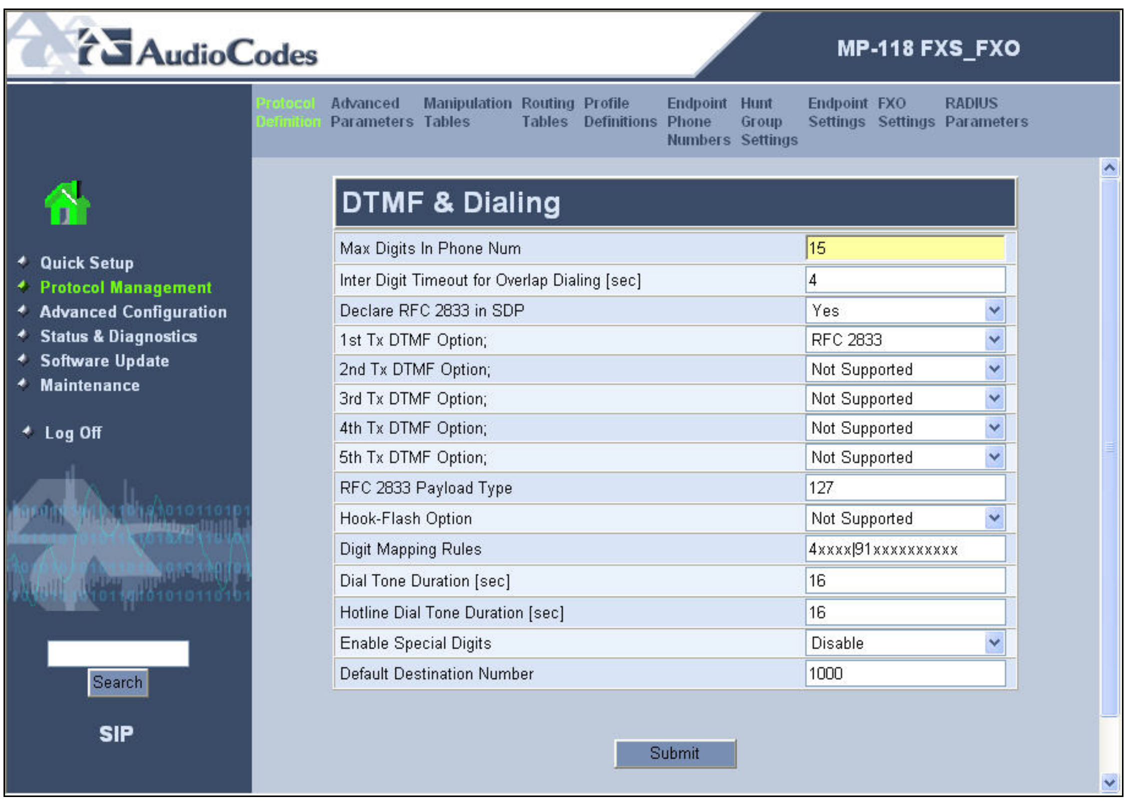
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>10.</td><td>

DTMF and Dialing

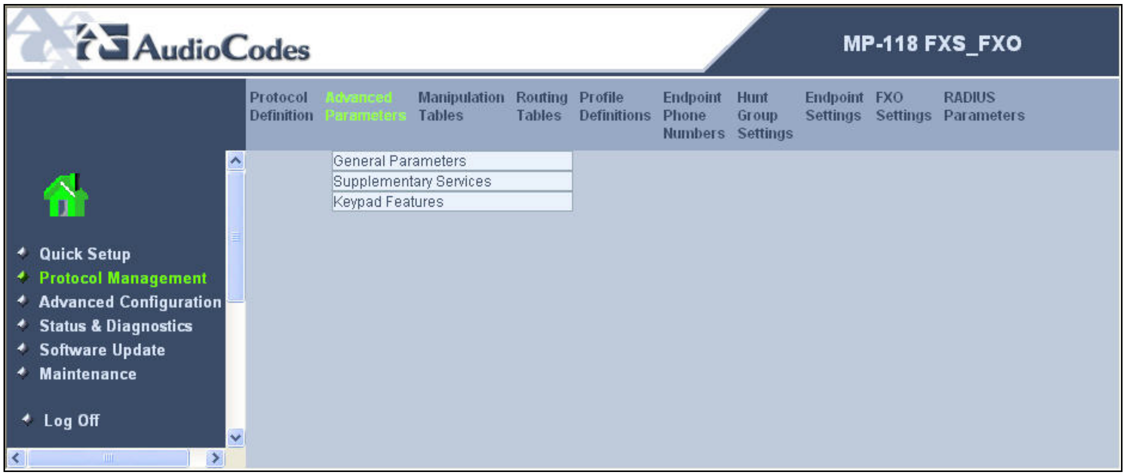
From the menu shown in Step 5, navigate to **Protocol Definition → DTMF & Dialing**. Configure the parameters as described below.

- In the **Max Digits in Phone Num** field, enter the maximum number of digits that can be dialed.
- For the **Declare RFC 2833 in SDP** field, select ***Yes***.
- For the **1st Tx DTMF Option** field, select ***RFC 2833***. This selects RFC 2833 as the preferred DTMF transmission method.
- Select ***127*** as the **RFC 2833 Payload Type** to match the value used by the Avaya SIP Telephones. Media may not be redirected (shuffled) in all scenarios from Avaya Communication Manager to the endpoints if this value is not the same as the SIP Telephones.
- Assign the digit map pattern to the **Digit Mapping Rules** field. If the dialed digits matches any pattern in the digit map, the MP-118 stops collecting digits and places the call. The digit map may contain up to 52 patterns each separated by a vertical bar (|). The maximum length of the entire digit map is limited to 152 characters. For the compliance test, two patterns were defined. One pattern matches any 5 digit number beginning with 4 (4xxxx). The other pattern matches any 12 digit number starting with 91 (91xxxxxxxxxx)

Default values may be retained for all other fields. Click **Submit**.



</td></tr>
</table>

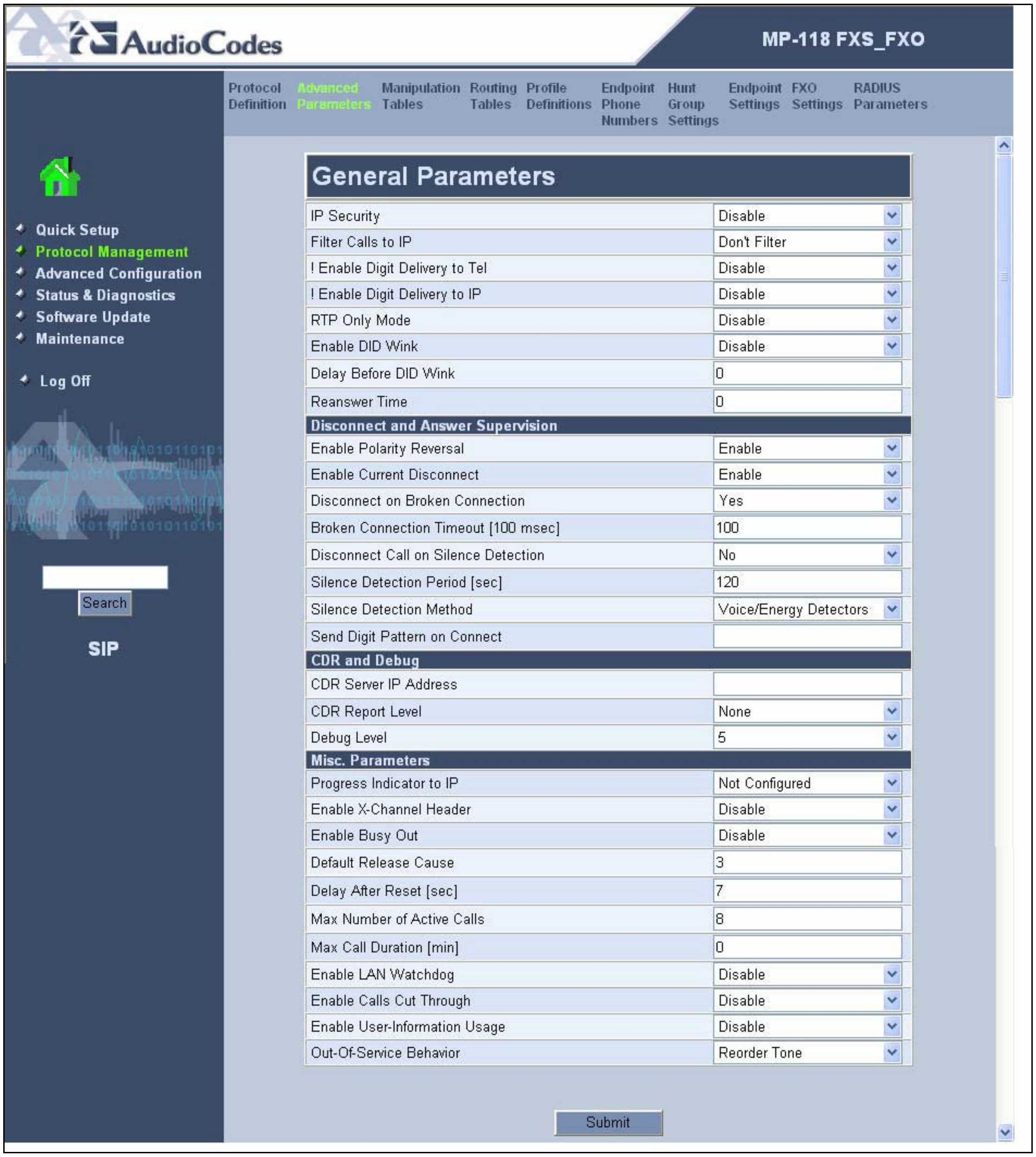
Step	Description
11.	**Advanced Parameters** To access these parameters, select **Protocol Management** in the left pane and navigate to **Advanced Parameters** in the right pane. The pull-down choices for **Advanced Parameters** are shown below. 

Step	Description
12.	**General Advanced Parameters**

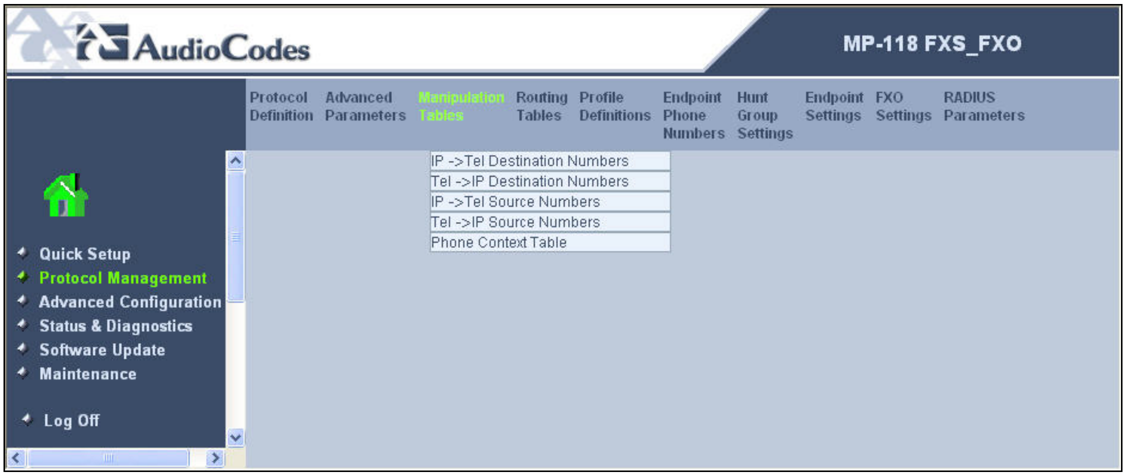
From the menu shown in Step 11, navigate to **Advanced Parameters → General Parameters**. Configure the parameters as described below.

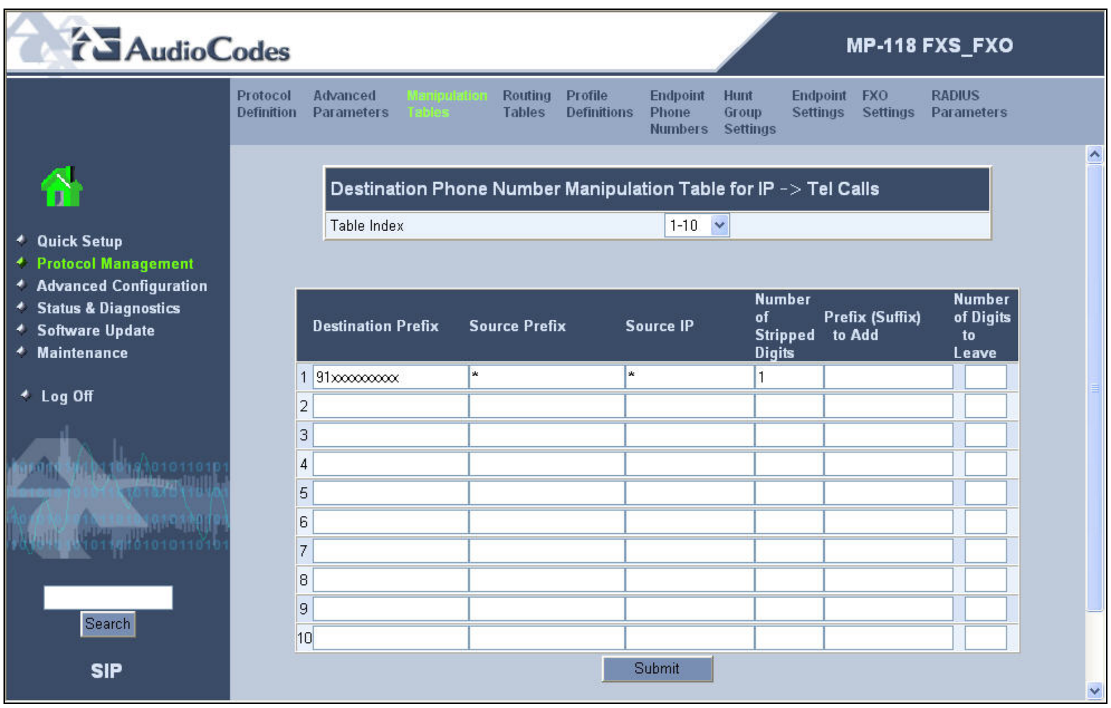
- Select ***Enable*** for the **Enable Polarity Reversal** and **Enable Current Disconnect** fields. This will allow the MP-118 to provide the proper disconnect indication to various line types.
- In the **Max Number of Active Calls** field, enter a value that is equal to or greater than the maximum number of ports (FXS + FXO) available on the gateway. For the compliance test, there were 4 FXS ports and 4 FXO ports.

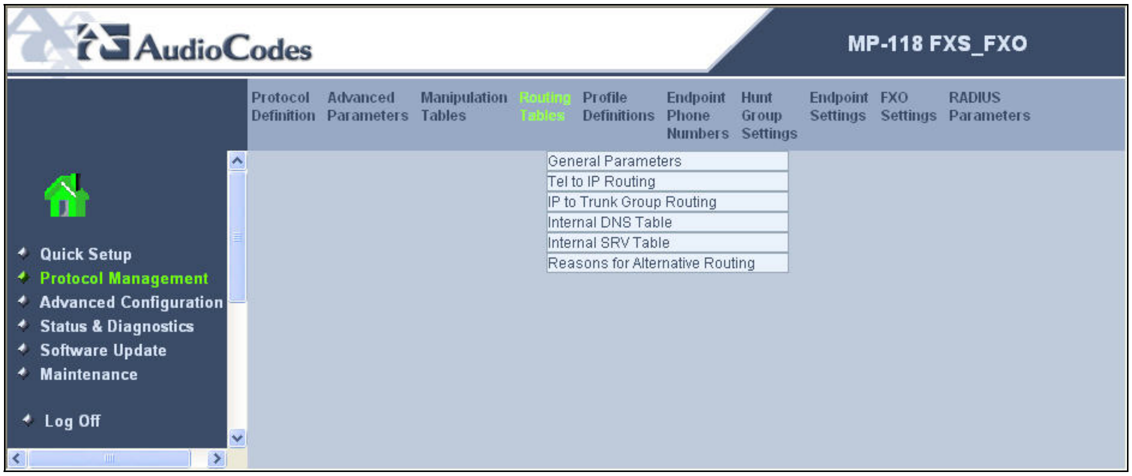
Default values may be retained for all other fields. Click **Submit**.

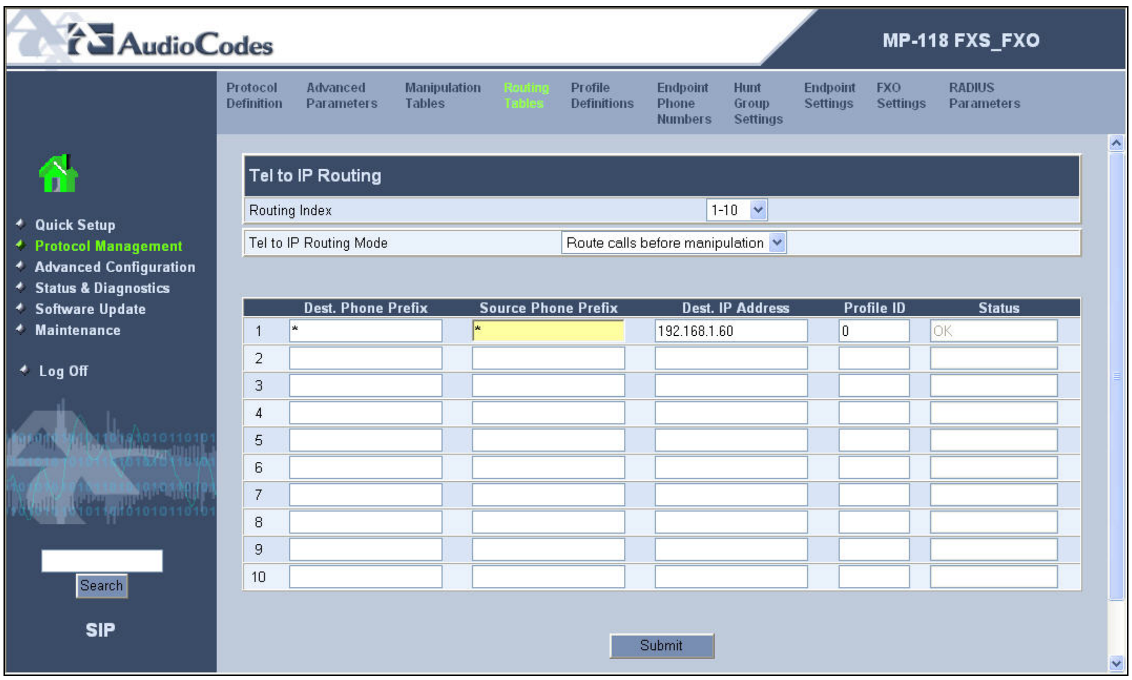


Step	Description
13.	**Supplementary Services** From the menu shown in Step 11, navigate to **Advanced Parameters → Supplementary Services**. Configure the parameters as described below. ▪ If the analog phones connected to the MP-118 support Caller ID then set the **Enable Caller ID** field to ***Enabled***. For the compliance test, this field was set to ***Disabled*** since none of the analog phones used had a Caller ID display. Caller ID can also be controlled on a per port basis by navigating to **Protocol Management → Endpoint Settings → Generate Caller ID to Tel Table** (not shown here). ▪ Select ***Enabled*** for the **Enable MWI** and **MWI Display** fields if the analog phones support a visual MWI indicator. For the compliance test, even though these fields were enabled, MWI was only tested for stutter dial tone. ▪ Hold, Transfer, Call Forwarding and Call Waiting are enabled by default. Default values may be retained for all other fields. Scroll down to the bottom of the page and click **Submit** (not shown).

Step	Description
14.	**Manipulation Tables** To access these parameters, select **Protocol Management** in the left pane and navigate to **Manipulation Tables** in the right pane. The pull-down choices for the **Manipulation Tables** are shown below. 

Step	Description
15.	**IP to Telephone Destination Numbers** From the menu shown in Step 14, navigate to **Manipulation Tables → IP to Tel Destination Numbers**. This table defines the digit manipulation that will be performed on IP to telephony calls. The **Destination Prefix**, **Source Prefix** and **Source IP** columns define which calls to manipulate. The 3 remaining columns define the digit manipulation to apply to these calls. In the example below, all calls dialed with twelve digits beginning with 91 coming from any source extension and source IP will have the first digit stripped off. This manipulation is required to allow the branch user to dial the same outbound number when the data WAN is available or not. Under normal conditions, a user dials 9 + the 11-digit PSTN number for an outbound call. The call is forwarded to the Avaya Communication Manager where the 9 is stripped off before being routed to the PSTN. If the data WAN is unavailable, the call is still originated as an IP call but can not be sent to the Main site. The MP-118 will use the table shown below to strip the preceding 9 off of this IP call before routing it to the FXO port for the PSTN. Click **Submit**. 

Step	Description
16.	**Routing Tables** To access these parameters, select **Protocol Management** in the left pane and navigate to **Routing Tables** in the right pane. The pull-down choices for the **Routing Tables** are shown below. 

Step	Description
17.	**Telephony to IP Routing** From the menu shown in Step 16, navigate to **Routing Tables → Tel to IP Routing**. This table defines the mapping of analog calls to an IP address that will process the IP leg of the call if the proxy defined in Step 6 (Avaya SES) is not available. The **Dest. Phone Prefix** and **Source Phone Prefix** columns define which calls are mapped to the IP address in the **Dest. IP Address** column. In the example below, the table entry maps calls from any destination prefix, or any source prefix to IP address 192.168.1.60 which is the address of the MP-118. When the MP-118 processes the call, the MP-118 will use the table in Step 18 to route the call to a FXO port. Thus, Steps 17 – 18 provide the routing necessary to support the failover functionality to the PSTN. The default values may be used for the **Profile ID** column. Click **Submit**. 

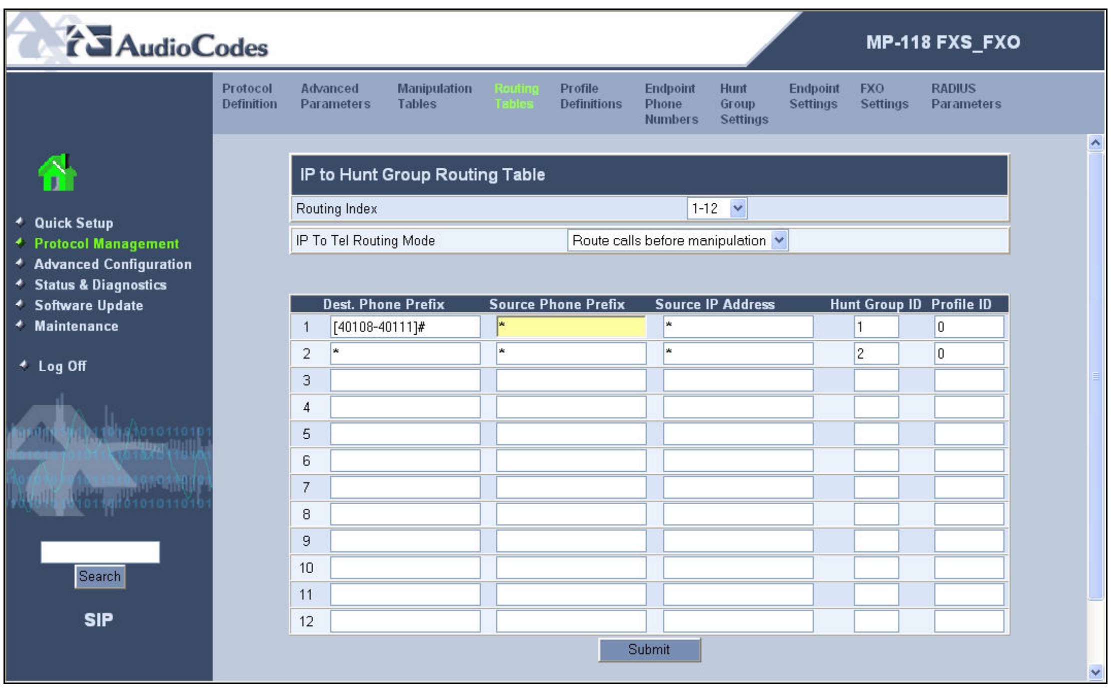
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>18.</td><td>

IP to Hunt Group Routing Table
From the menu shown in Step 16, navigate to **Routing Tables → IP to Trunk Group Routing**.

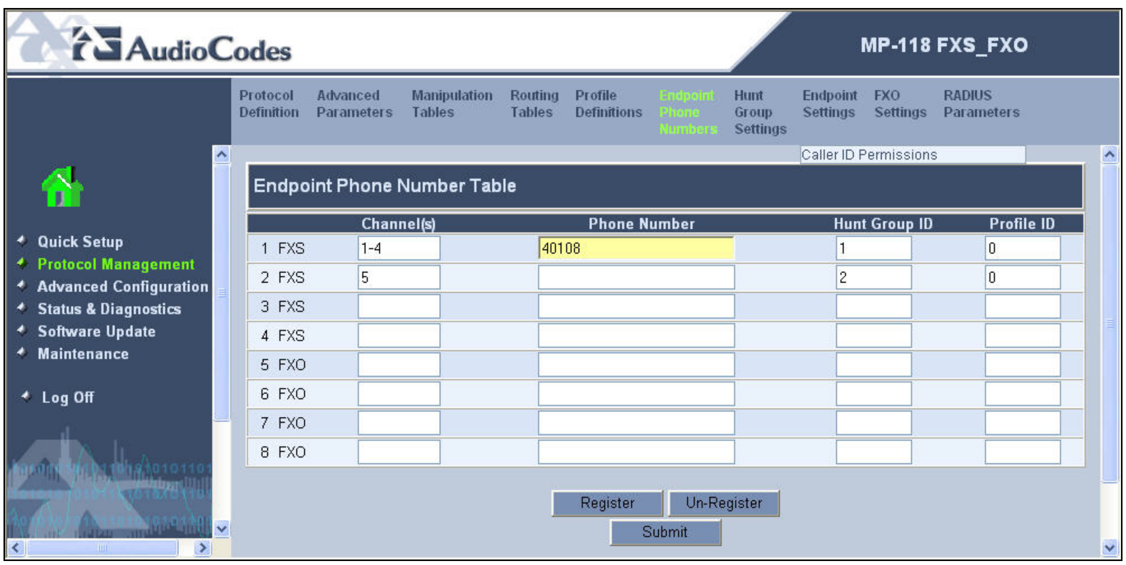
This table defines the mapping of IP calls to a group of channels or “hunt group”. In Step 19, the FXS ports are assigned to hunt group 1 and the FXO ports are assigned to hunt group 2. The **Dest. Phone Prefix**, **Source Phone Prefix** and **Source IP Address** columns define which calls are mapped to the hunt group in the **Hunt Group ID** column. In the first entry in the example below, all calls to any extension 40108 - 40111 from any source extension and source IP will be routed to hunt group 1. The # at the end of the **Dest. Phone Prefix** field indicates “end of number”. Thus, the dialed number must be an exact match with an extension and not simply a match on the prefix of 40108 - 40111.

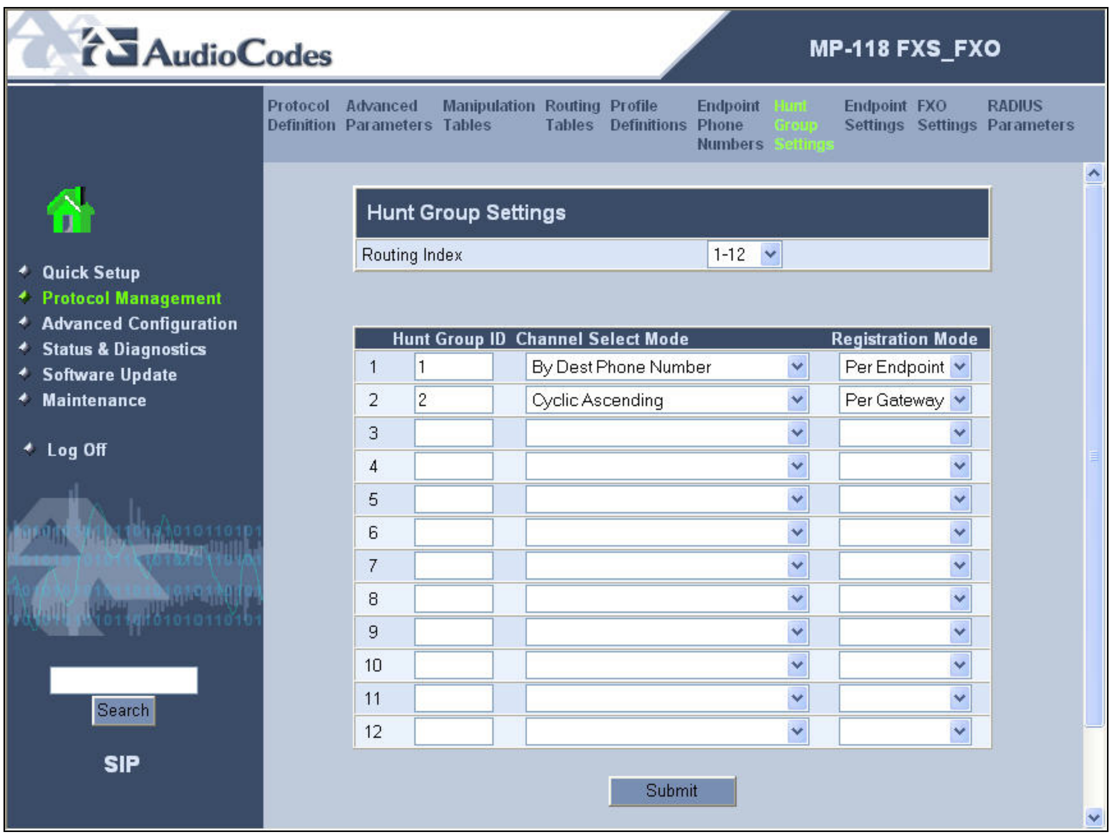
The second entry routes any other call to hunt group 2. Even though Avaya Communication Manager will not route any other calls to the MP-118 except calls to the extensions 40108 – 40111, this entry is needed for the failover case. If a user dials an outbound number and it can not reach the main site, this table will direct the call to hunt group 2 which contains the FXO ports.

Click **Submit**.

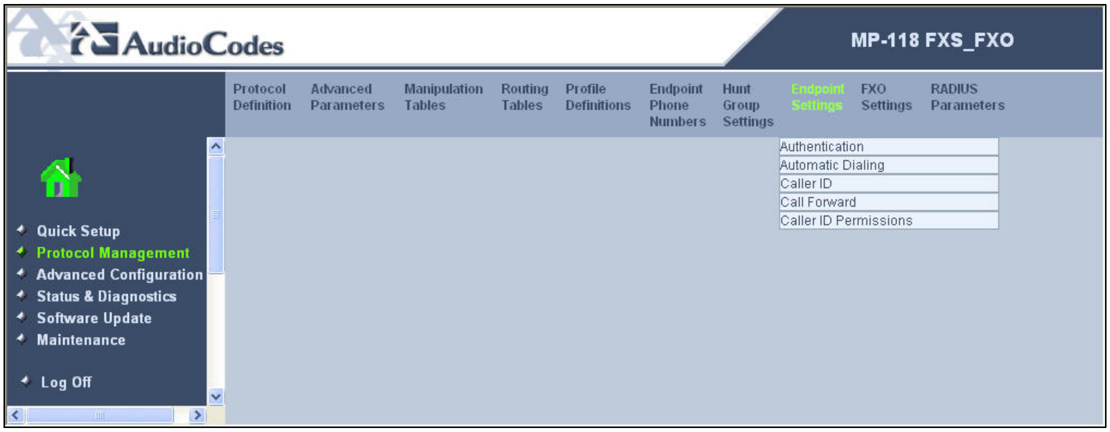


</td></tr>
</table>

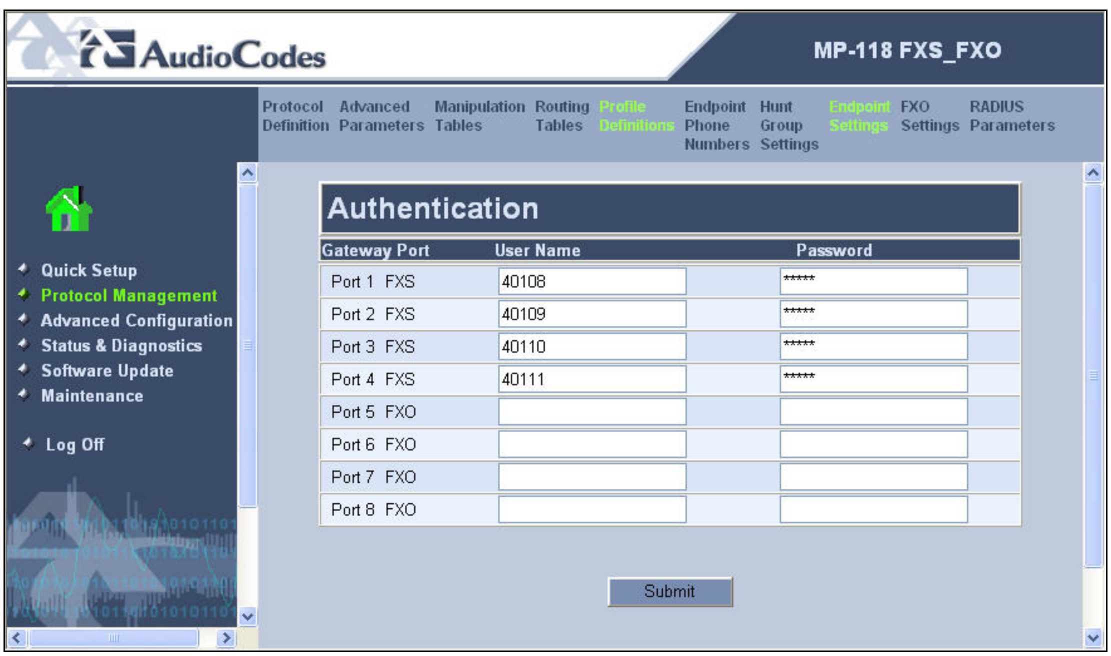
Step	Description
19.	**Endpoint Phone Numbers** Select **Protocol Management** in the left pane and navigate to **Endpoint Phone Numbers** in the right pane. The Endpoint Phone Number Table maps a particular channel/port to a phone number and hunt group. In the **Channel(s)** column, enter a range of channels to be assigned. In the **Phone Number** column, enter the starting extension for the range of extensions. In the **Hunt Group ID** column, enter the hunt group that contains these extensions. In the example below, the first entry assigns channels ***1 – 4*** with extensions ***40108 – 40111*** to hunt group ***1***. The second entry assigns channel ***5*** (the first FXO port) to hunt group ***2***. Since only the first FXO port was connected to the PSTN, only channel 5 was enter in this table. The **Phone Number** field is blank for this entry. The originating number for calls originating from channel 5 is set to the Caller ID information received from the PSTN trunk. Click **Submit**. 

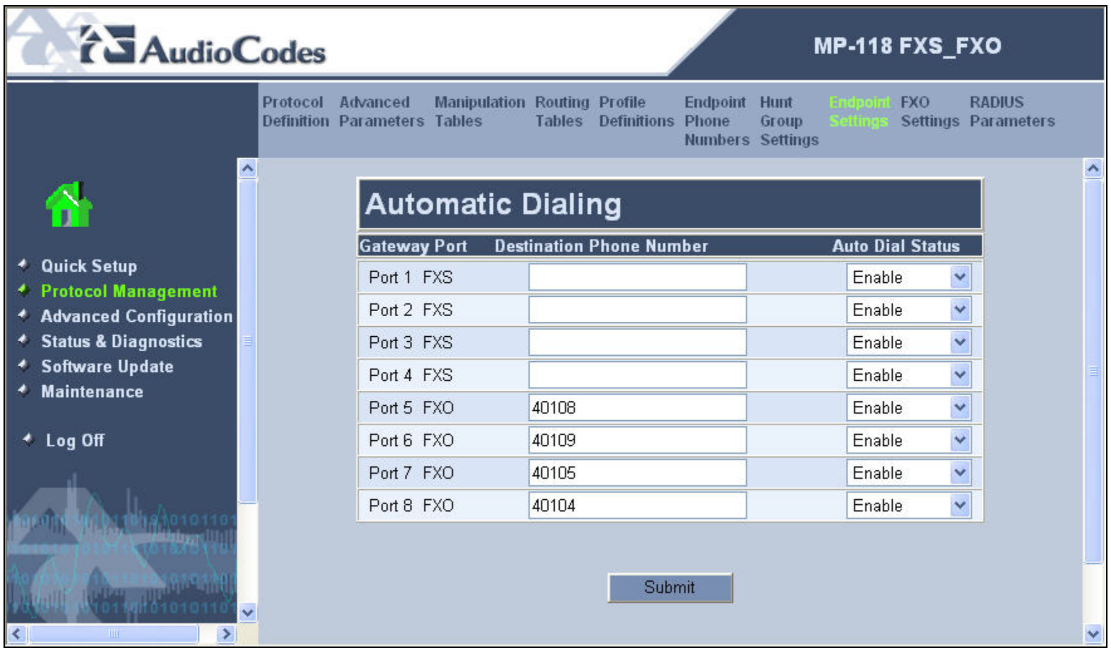
Step	Description
20.	**Hunt Group Settings** Select **Protocol Management** in the left pane and navigate to **Hunt Group Settings** in the right pane. Configure the parameters are described below. ▪ For **Hunt Group ID 1** which contain the FXS (endpoint) ports, select the **Channel Select Mode** as ***By Dest Phone Number***. Thus, each port in this hunt group will only be selected if its destination phone number is dialed. Select the **Registration Mode** to be ***Per Endpoint***. ▪ For **Hunt Group ID 2** which contain the FXO (POTS) ports, select the **Channel Select Mode** as ***Cyclic Ascending***. The ports in this hunt group are treated as a pool, and each will be selected in cyclic ascending order. Select the **Registration Mode** to be ***Per Gateway***. This allows the MP-118 to register once for all the FXO ports using the gateway extension entered in Step 7. The MP-118 requires that both the FXS and FXO ports be registered since registration was enabled in Step 6. For the compliance test, it was chosen to register once for all the FXO ports since it is not intended that the FXO ports be accessed using the gateway extension. Click **Submit**. 

Step	Description
21.	**Endpoint Settings** To access these parameters, select **Protocol Management** in the left pane and navigate to **Endpoint Settings** in the right pane. The pull-down choices for **Endpoint Settings** are shown below.



Step	Description
22.	**Authentication** From the menu shown in Step 21, navigate to **Endpoint Settings → Authentication**. The Authentication page defines a username and password combination for authentication of each MP-118 port with the Avaya SES. Enter a **User Name** and **Password** that matches the values configured on Avaya SES in Section 4, Step 7. Click **Submit**.

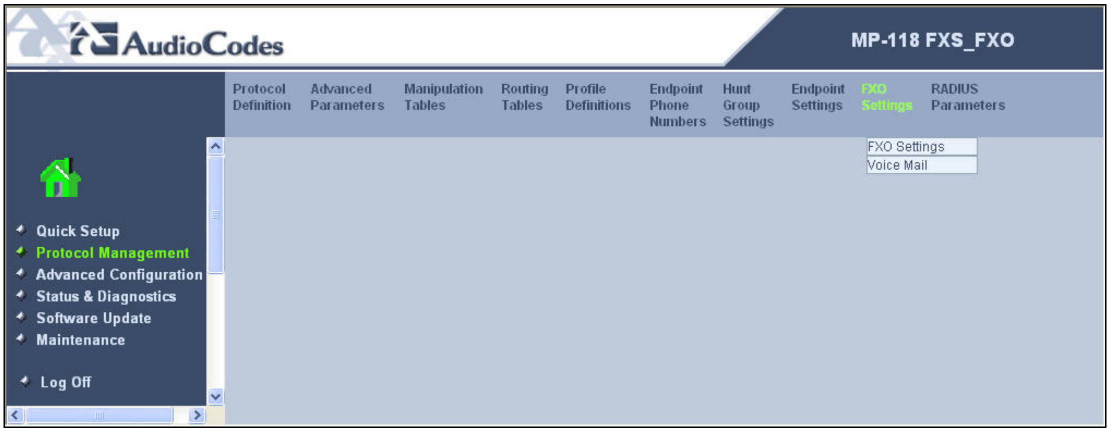


Step	Description
23.	**Automatic Dialing** From the menu shown in Step 21, navigate to **Endpoint Settings → Automatic Dialing**. The Automatic Dialing page provides the mapping of incoming calls on the FXO ports to a branch extension when the data WAN is unavailable. In the example below, each FXO port is mapped to a different extension at the branch location. All ports were mapped even though only one FXO port was connected. The destination extension is placed in the **Destination Phone Number** column. Click **Submit**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>24.</td><td>

FXO Settings

To access these parameters, select **Protocol Management** in the left pane and navigate to **FXO Settings** in the right pane. The pull-down choices for **FXO Settings** are shown below.



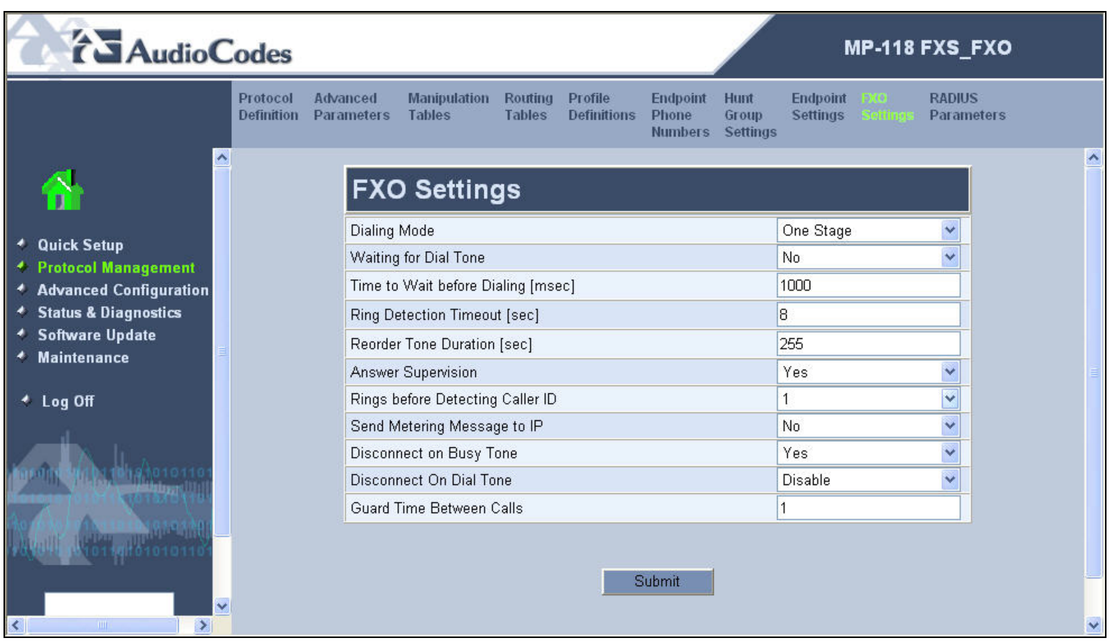
</td></tr>
<tr><td>25.</td><td>

FXO Settings

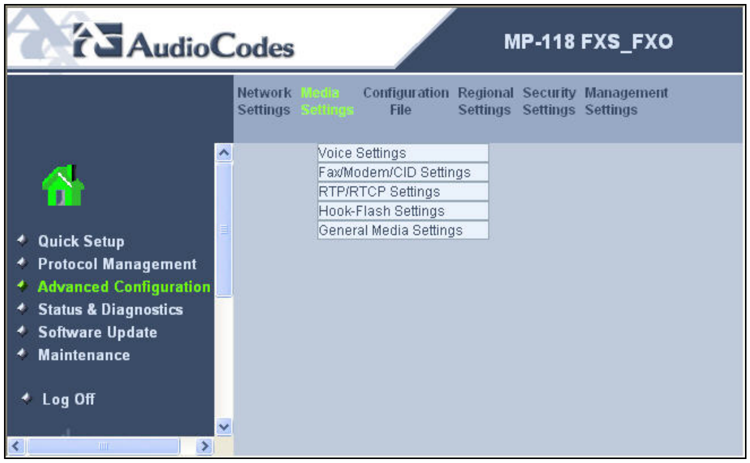
From the menu shown in Step 24, navigate to **FXO Settings → FXO Settings**. Configure the parameters as described below.

- Select ***One Stage*** for the **Dialing Mode**. The MP-118 will not prompt the user with a second dial tone and will send all digits in a single request including any access code prefix.
- Select ***Yes*** for **Answer Supervision**.

Default values may be retained for all other fields. Click **Submit**.



</td></tr>
</table>

Step	Description
26.	**Media Settings** To access these parameters, select **Advanced Configuration** in the left pane and navigate to **Media Settings** in the right pane. The pull-down choices for **Media Settings** are shown below. 

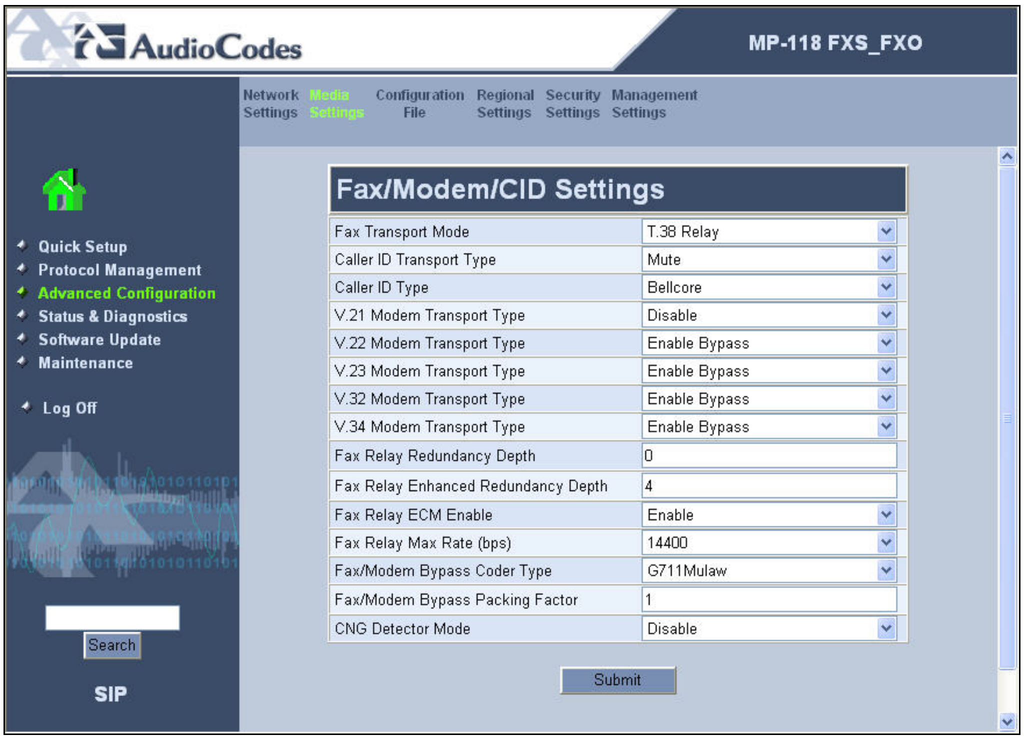
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>27.</td><td>

Fax/Modem/CID Settings

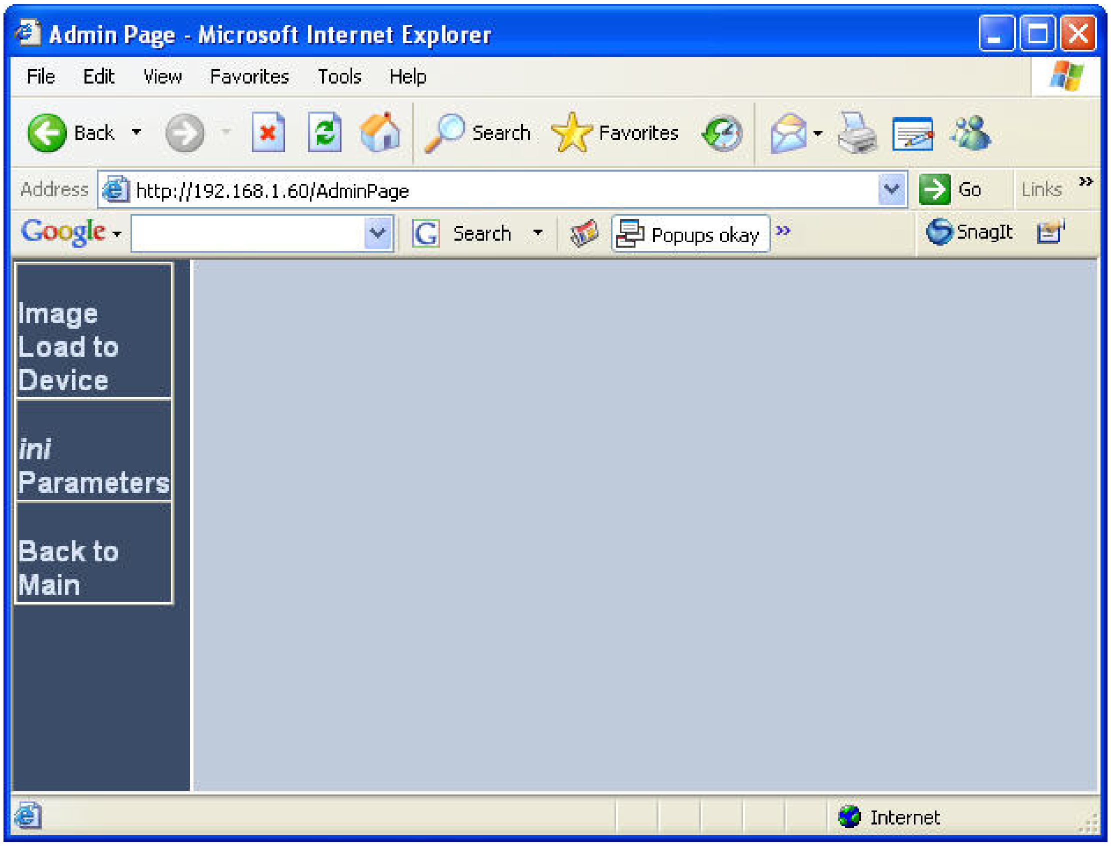
From the menu shown in Step 26, navigate to **Media Settings → Fax/Modem/CID Settings**. Configure the parameters as described below.

- Select ***T.38 Relay*** for the **Fax Transport Mode** to support faxing across the data WAN.
- Select ***Enable Bypass*** for the **V.22 – V.34 Modem Transport Types** to support modem calls over SIP planned in future Avaya Communication Manager releases.
- Select ***G711Mulaw*** for the **Fax/Modem Bypass Coder Type**.

Default values may be retained for all other fields. Click **Submit**.

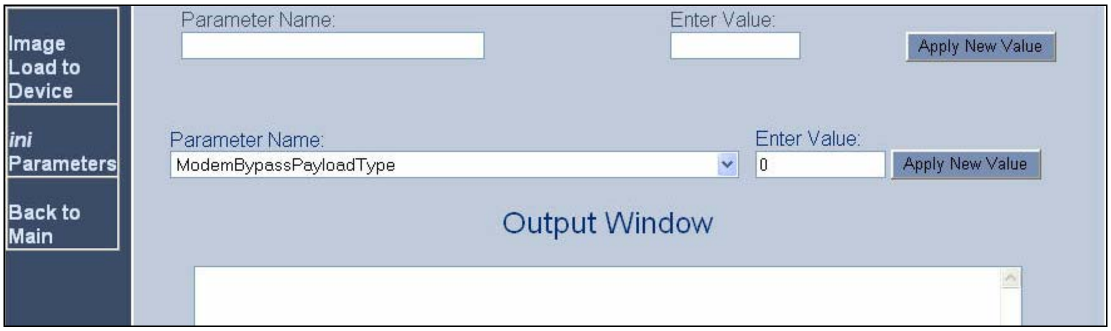


</td></tr>
</table>

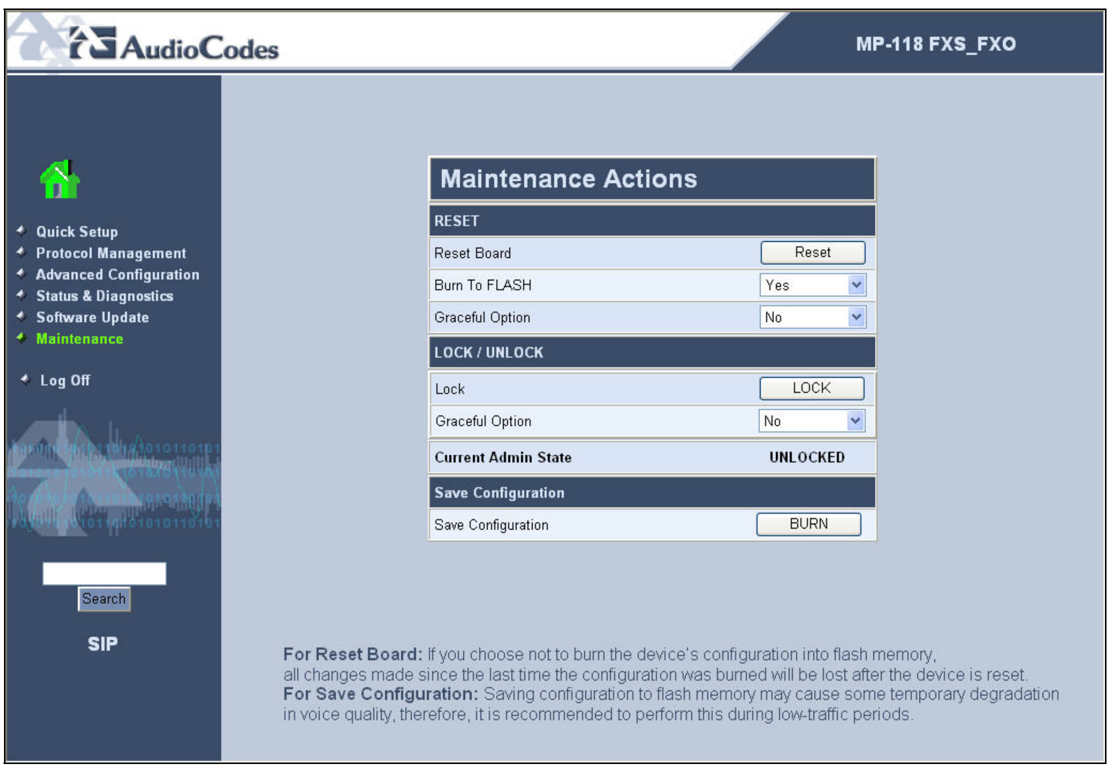
Step	Description
28.	For proper modem interoperability over SIP planned for future Avaya Communication Manager releases, an additional parameter must be changed in the ini file. From a web browser, enter <*ip_address*>/*AdminPage* in the **Address** field where <*ip_address*> is the IP address of the MP-118. The main page will appear as shown below. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>29.</td><td>
Navigate to ini Parameters in the left pane. In the Parameter Name field, select ModemBypassPayloadType from the pull-down menu. Enter 0 in the Enter Value field. Click on Apply New Value.

A value of 0 in the payload type indicates a G.711mu call. By default, the MP-118 uses a proprietary value for the payload type of modem calls so they can be distinguished from other G.711mu calls.



</td></tr>
<tr><td>30.</td><td>
To save the configuration, return to the administrative web page and select Maintenance in the left pane. Click the BURN button.



</td></tr>
</table>

6. Interoperability Compliance Testing

This section describes the compliance testing used to verify the interoperability between the AudioCodes MP-118, Avaya SIP Enablement Services (SES) and Avaya Communication Manager. This section covers the general test approach and the test results.

6.1. General Test Approach

The general test approach was to make calls to/from the telephones connected through the MP-118 at the branch site using various codec settings and exercising common PBX features. This testing included the analog telephones, and Avaya SIP telephones. The calls were made to/from the main site, the PSTN and within the branch site. The same test cases, where applicable, were repeated with a simulated data WAN outage.

6.2. Test Results

The AudioCodes MP-118 successfully passed compliance testing. The following features and functionality were verified using an MP-118 analog endpoint when the data WAN was available.

- Calls to/from the main site
- Calls to/from the PSTN
- Intra-branch calls
- G.711mu and G.729AB codec support
- Proper recognition of DTMF transmissions
- Local device support for Hold, Transfer, and Call Waiting
- Proper operation of voicemail with message waiting indicators (MWI). For the analog phones, MWI was provided via stutter dial tone.
- Call Forwarding provided by Avaya Communication Manager.
- Conferencing (Avaya SIP telephone initiates a conference that includes an MP-118 analog endpoint)
- Extended telephony features using Avaya Communication Manager Feature Name Extensions such as Conference On Answer, Call Park, Call Pickup, Automatic Redial and Send All Calls. For more information on FNEs, please refer to [6].
- T.38 fax support
- Proper system recovery after a MP-118 restart

The following features and functionality were verified using an MP-118 analog endpoint when a simulated data WAN failure was introduced.

- Automatic failover to the POTS line to complete calls to the main site and PSTN using full 11 digit dialing. Incoming calls to the branch are limited to the single POTS number assigned to the branch.
- Intra-branch calls
- Local device support for Hold, Transfer, and Call Waiting

The following observations were made during the compliance test:

- The analog endpoints connected to the MP-118 do not support initiating a conference call.
- When the data WAN is unavailable and an MP-118 analog endpoint calls the PSTN and does an unattended transfer to another MP-118 analog endpoint, the originator of the transfer does not hear ringback.

- When using the Call Park feature from an MP-118 analog endpoint via the Avaya Communication Manager FNE, the user is required to wait a couple of seconds before hanging up after hearing the feature confirmation tones. Otherwise, the MP-118 does not disconnect the call as expected but instead attempts a transfer.
- To support T.38 faxing with the MP-118, a non-GA release of Avaya Communication Manager was required.

7. Verification Steps

The following steps may be used to verify the configuration:

- From the Avaya Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling group is in-service.
- From the Avaya Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group is in-service.
- From the Avaya SES web administration interface, verify that all endpoints behind the MP-118 are registered with the Avaya SES.
- Verify that calls can be placed to/from the analog endpoints behind the MP-118 and the main branch.
- Verify that calls can be placed to/from the analog endpoints behind the MP-118 and the PSTN.
- Verify that calls can be placed from the analog endpoints behind the MP-118 when a simulated data WAN failure is introduced.

8. Support

For technical support on the MP-118, contact AudioCodes via the support link at www.audiocodes.com.

9. Conclusion

These Application Notes describe the procedures required to configure the AudioCodes MP-118 Analog VoIP Gateway to interoperate with Avaya SIP Enablement Services and Avaya Communication Manager. The AudioCodes MP-118 successfully passed compliance testing with the observations documented in Section 6.2.

10. Additional References

[1] *Feature Description and Implementation For Avaya Communication Manager*, Doc # 555-245-205, Issue 4.0, February 2006.

[2] *Administrator Guide for Avaya Communication Manager*, Doc # 03-300509, Issue 2.1, May 2006

[3] *Avaya Communication Manager Advanced Administration Quick Reference*, Doc # 03-300364, Issue 2, June 2005 Release 3.0

[4] *Avaya IA 770 INTUITY AUDIX Messaging Application*, Doc # 11-300532, May 2005

[5] *Installing and Administering SIP Enablement Services R3.1*, Doc# 03-600768, Issue 1.5, February 2006

[6] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, version 6.0, Doc # 210-100-500, Issue 9, June 2005

[7] *LTRT-59802 MediaPack Fast Track Installation Guide*, Version 5.0.

[8] *LTRT-65406 MediaPack SIP User's Manual*, Version 5.0.
[9] *LTRT-65607 MediaPack & Mediant 1000 SIP Analog Gateways Release Notes,* Version 5.0.

Product documentation for Avaya products may be found at http://support.avaya.com.

Product documentation for AudioCodes MP-118 Analog VoIP Gateway products may be found at http://www.audiocodes.com.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Developer*Connection* Program at devconnect@avaya.com.