Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Multi-Tech MultiVOIP SS with Avaya SIP Enablement Services and Avaya Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring the Multi-Tech MultiVOIP SS SIP Gateway and Server with Avaya SIP Enablement Services and Avaya Communication Manager.

The Multi-Tech MultiVOIP SS SIP Gateway and Server 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). It also supports SIP endpoints at the branch by serving as a local outbound proxy and provides local survivability for these SIP endpoints should the data WAN connection to the main site fail. Thus, allowing the SIP endpoints at the branch to continue to operate and make calls. The MultiVOIP SS represents a family of gateways that support 2, 4 or 8 analog ports with each port configurable as a FXS or FXO port.

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

1. Introduction

These Application Notes describe the procedures for configuring the Multi-Tech MultiVOIP SS SIP Gateway and Server with Avaya SIP Enablement Services (SES) and Avaya Communication Manager.

The Multi-Tech MultiVOIP SS SIP Gateway and Server 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). It also supports SIP endpoints at the branch by serving as a local outbound proxy and provides local survivability for these SIP endpoints should the data WAN connection to the main site fail. This allows the SIP endpoints at the branch to continue to operate and make calls. The MultiVOIP SS represents a family of gateways that support 2, 4 or 8 analog ports with each port configurable as a FXS or FXO port.

The FXO ports are intended to be used for local inbound and outbound PSTN access for the users located at the branch. Users at the main location will obtain PSTN access from the main location. Thus, a call placed from each location dialed with the same destination digit string will be routed differently based on where the call originates. This requires the use of the Multiple Locations feature on Avaya Communication Manager. The FXO ports are also used as a failover path if the data WAN is unavailable and SIP calls cannot be made.

The MultiVOIP SS registers with the Avaya SES as a SIP endpoint for each FXS or FXO port which is enabled. On the Avaya SES, the FXS ports are configured as users with media server extensions and the FXO ports are configured as users without media server extensions. Thus, when calls are routed to Avaya Communication Manager, calls from the FXS ports appear as calls from extensions and calls from the FXO ports appear as trunk calls.

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 S8500 Server running Avaya Communication Manager with an Avaya G650 Media Gateway. Endpoints include Avaya 4600 Series IP Telephones (with H.323 and SIP firmware), Avaya 9600 Series IP Telephones (with H.323 and SIP firmware), an Avaya one-X Desktop Edition SIP, and a fax machine. An ISDN-PRI trunk connects the media gateway to the PSTN.

The branch site has a MultiVOIP SS with two analog telephones, and/or a fax machine. The branch site also has two Avaya 4600 Series IP Telephones (with SIP firmware), an Avaya 9600 Series IP Telephone (with SIP firmware) and an Avaya one-X Desktop Edition SIP. The MultiVOIP SS connects the branch site to the PSTN via an FXO (POTS) trunk.

As mentioned previously, the MultiVOIP SS registers with the Avaya SES as a SIP endpoint for each FXS or FXO port which is enabled. In addition, SIP endpoints at the branch site are configured with the MultiVOIP SS as their call server and the MultiVOIP SS will register with the Avaya SES on behalf of these endpoints as well. Thus, all SIP traffic between the branch site and the Avaya SES will pass through the MultiVOIP SS.

The PSTN numbers assigned to the ISDN-PRI trunk at the main site are mapped to telephone extensions at the main site. The PSTN number assigned to the POTS line at the branch site is mapped to an extension at the branch site.

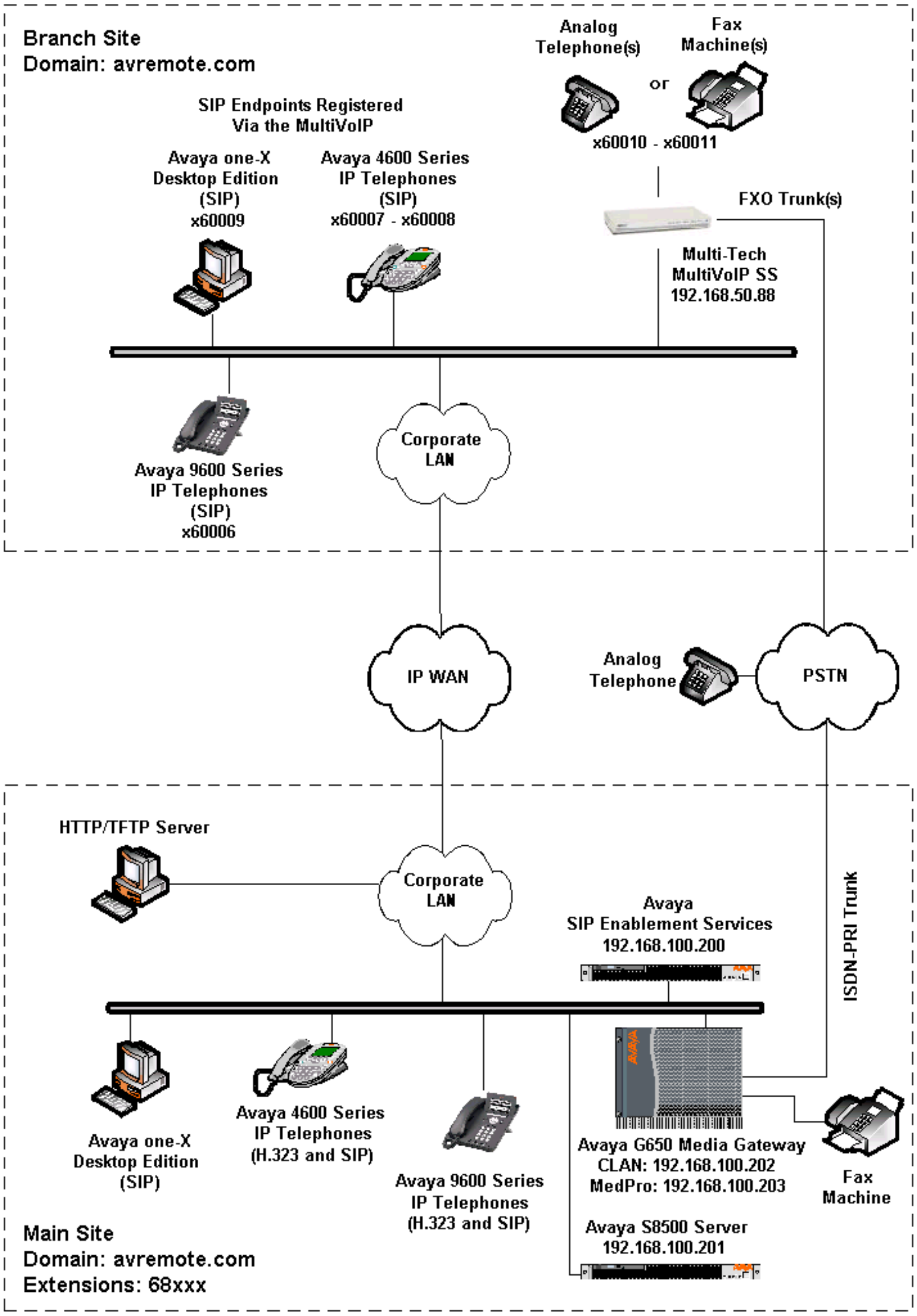


Figure 1: MultiVOIP SS Test Configuration

1.2. Operation

As mentioned previously, PSTN calls are routed based on the dialed digits as well as where the call originates. The Multiple Location feature and Automatic Route Selection (ARS) are used to provide this functionality. In addition, two SIP trunk groups are created between Avaya Communication Manager and Avaya SES. The first trunk group is the initial SIP trunk group required of any SIP installation. It is associated with a SIP signaling group with the far-end domain set to the domain of the Avaya SES. The second trunk group is associated with a signaling group that uses the IP address of the MultiVOIP SS as the far-end domain. This second domain information is passed to Avaya SES in the SIP signaling messages for calls using this second trunk group. This information allows the Avaya SES to route these calls to the MultiVOIP SS.

The call flow for an inbound and outbound call from the MultiVOIP SS is described below using this configuration. It should be noted that even though the call originates and terminates on the MultiVOIP SS, the call is sent to Avaya Communication Manager for processing. This allows Avaya Communication Manager to apply access and routing criteria, as well as make available a wide range of PBX features to the call.

Outbound call to the PSTN using the MultiVOIP SS FXO ports:

1. An analog telephone connected to the MultiVOIP SS at the branch dials an ARS access code + a PSTN number.
2. The MultiVOIP SS initiates a SIP call to the Avaya SES with the SIP domain specified in the Avaya SES. This is the same SIP domain with which the endpoint registered (e.g., avremote.com).
3. The Avaya SES recognizes the call is from a registered user in its own domain which has a media server extension assigned to it. Thus, Avaya SES routes the call automatically to Avaya Communication Manager.
4. Based on the access code, Avaya Communication Manager uses ARS to route the call. Based on the location where the call originates, the Avaya Communication Manager will select the proper ARS table unique to that location. This ARS table will point to a route pattern specific for that location.
5. The route pattern will point to the SIP trunk group that is unique for that location which has the far-end domain set to the IP address of the MultiVOIP SS.
6. Based on this route pattern, Avaya Communication Manager routes the call back to the Avaya SES using this new trunk group. This has the effect of changing the SIP domain in the SIP signaling messages to the IP address of the MultiVOIP SS instead of avremote.com.
7. The Avaya SES receives the call and determines it is not for its own domain (avremote.com) so forwards it to the new domain (the MultiVOIP SS IP address).
8. The MultiVOIP SS receives the call with the PSTN number and routes it to an available trunk.

Inbound call from the PSTN using the MultiVOIP SS FXO ports:

1. A PSTN caller dials a PSTN number that is associated with one of the POTS trunks connected to the MultiVOIP SS. The MultiVOIP SS maps this incoming call to an extension which can be located at either site. This example assumes the mapped extension is one of the

analog telephones connected to the MultiVOIP SS. The MultiVOIP SS will initiate a SIP call to the Avaya SES for this extension.

2. The Avaya SES recognizes the call is for a registered user in its own domain which has a media server extension assigned to it. Thus, Avaya SES routes the call automatically to Avaya Communication Manager.
3. The Avaya Communication Manager receives the call and determines that the destination extension is an OPS station associated with the SIP trunk connected to the Avaya SES. The Avaya Communication Manager then routes the call back to the Avaya SES.
4. The Avaya SES then routes the call to the registered IP address for that extension (the MultiVOIP SS).

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8500 Server with Avaya G650 Media Gateway	Avaya Communication Manager 5.0 with Service Pack (00.0.825.4-15467)
Avaya SIP Enablement Services (SES)	5.0 with Service Pack (SES-00.0.825.4-SP2d)
Avaya 4610SW IP Telephone (SIP) Avaya 4620SW IP Telephones (SIP)	2.2.2
Avaya 4625SW IP Telephone (H.323)	2.8.3
Avaya 9620 IP Telephone (H.323) - Avaya one-X Deskphone Edition	1.5
Avaya 9620 IP Telephones (SIP) - Avaya one-X Deskphone Edition (SIP)	2.0.3
Avaya one-X Desktop Edition (SIP soft phone)	2.1 SP2
Analog Telephones	-
Analog Fax Machines	-
Windows PCs	Windows XP Professional
Multi-Tech MultiVOIP SS (MVP 410-SS)	3.11.1t

3. Configure Avaya Communication Manager

This section describes the Avaya Communication Manager configuration at the main site to support the network shown in **Figure 1**. It assumes the procedures necessary to support SIP and connectivity to Avaya SES have been performed as described in [3]. It also assumes that an off-PBX station (OPS) has been configured on Avaya Communication Manager for each internal SIP endpoint in the configuration as described in [3] and [5].

This section is divided into three parts. **Section 3.1** will summarize the user-defined parameters used in the installation procedures that are important to understanding the solution as a whole. It will not attempt to show the installation procedures in their entirety. It will also describe any deviations from the standard procedures, if any. **Section 3.2** and **3.3** are specific to the MultiVOIP SS configuration. **Section 3.2** will describe the configuration needed for multiple location support and call routing based on origination location. **Section 3.3** will describe the configuration of the remote SIP endpoints.

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

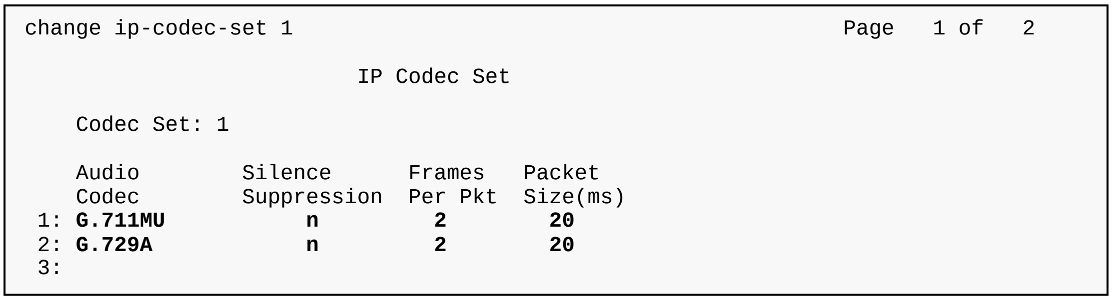
3.1. Summary of Initial SIP Installation

This section summarizes the applicable user-defined parameters used during the SIP installation procedures.

Step	Description
1.	**IP network region** The Avaya S8300 Server, Avaya SES and IP (H.323/SIP) endpoints were located in a single IP network region (IP network region 1) using the parameters described below. Use the **display ip-network-region** command to view these settings. The example below shows the values used for the compliance test. ▪ The **Location** field was set to 1. This associates this IP network region with location 1. ▪ The **Authoritative Domain** field was configured to match the domain name configured on Avaya SES. In this configuration, the domain name is ***avremote.com***. This name appears in the “From” header of SIP messages originating from this IP region. ▪ A descriptive name was entered for the **Name** field. ▪ **IP-IP Direct Audio** (shuffling) was enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. This was done for both intra-region and inter-region IP-IP Direct Audio. This is the default setting. Shuffling can be further restricted at the trunk level on the **Signaling Group** form. ▪ The **Codec Set** field was set to the IP codec set to be used for calls within this IP network region. In this case, IP codec set 1 was selected. If different IP network regions are used for the Avaya S8300 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 default values were used for all other fields.

```
display ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avremote.com
    Name: Main
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? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      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
2.	**Codecs** IP codec set 1 was used for the compliance test. Multiple codecs were listed in priority order to allow the codec used by a specific call to be negotiated during call establishment. The list includes 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. It should be noted that when testing the use of each individual codec, only the codec under test was included in the list.



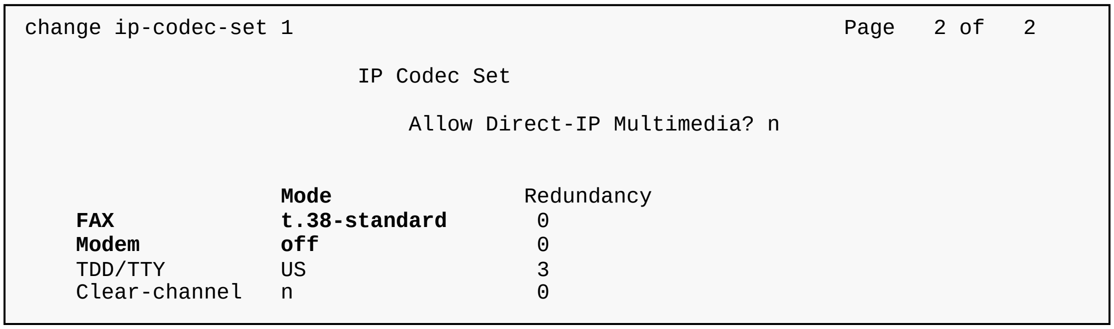
```
change ip-codec-set 1                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 1

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

3.	**Fax** On **Page 2**, the **FAX Mode** field was set to ***t.38-standard*** to support the fax machines. The **Modem Mode** field should be set to ***off***. The screen below shows the settings used for the compliance test.



```
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
4.	**Signaling Group** For the compliance test, signaling group 1 was used for the signaling group associated with the SIP trunk group between Avaya Communication Manager and Avaya SES. Signaling group 1 was configured using the parameters highlighted below. All other fields were set as described in [3]. ▪ The **Group Type** was set to ***sip***. ▪ The **Transport Method** was set to the recommended default value of ***tls*** (Transport Layer Security). As a result, the **Near-end Listen Port** and **Far-end Listen Port** are automatically set to ***5061.*** ▪ The **Near-end Node Name** was set to ***clan1***. This node name maps to the IP address of the CLAN circuit pack used to connect to Avaya SES. Node names are defined using the **change node-names ip** command. ▪ The **Far-end Node Name** was set to ***ses***. This node name maps to the IP address of Avaya SES as defined using the **change node-names ip** command. ▪ The **Far-end Network Region** was set to ***1***. This is the IP network region which contains Avaya SES. ▪ The **Far-end Domain** was set to ***avremote.com***. This is the domain configured on Avaya SES. This domain is sent in the "To" header of SIP INVITE messages for calls using this signaling group. ▪ **Direct IP-IP Audio Connections** was set to ***y***. This field must be set to ***y*** to enable media shuffling on the SIP trunk. ▪ The **DTMF over IP** field was set to the default value of ***rtp-payload***. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833. ▪ The default values were used for all other fields.

```
display signaling-group 1
                                SIGNALING GROUP

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



   Near-end Node Name: clan1                Far-end Node Name: ses
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
      Far-end Domain: avremote.com

                                             Bypass If IP Threshold Exceeded? n

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

Step	Description
5.	**Trunk Group** For the compliance test, trunk group 1 was used for the SIP trunk group between Avaya Communication Manager and Avaya SES. Trunk group 1 was configured using the parameters highlighted below. All other fields were set as described in [3]. On **Page 1**: ▪ The **Group Type** field was set to ***sip***. ▪ A descriptive name was entered for the **Group Name**. ▪ An available trunk access code (TAC) that was consistent with the existing dial plan was entered in the **TAC** field. ▪ The **Service Type** field was set to ***tie***. ▪ The **Signaling Group** was set to the signaling group shown in the previous step. ▪ The **Number of Members** field contained the number of trunks in the SIP trunk group. It determines how many simultaneous SIP calls can be supported by the configuration. 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. ▪ The default values were used for all other fields.

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

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: SES                         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: 10
```

Step	Description
6.	**Trunk Group – continued** On **Page 3**: ▪ The **Numbering Format** field was set to ***public***. This field specifies the format of the calling party number sent to the far-end. ▪ The default values were used for all other fields.

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



                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
```

Step	Description
7.	**Public Unknown Numbering** Public unknown numbering defines the calling party number to be sent to the far-end. An entry was created that will be used by the trunk group defined in **Step 5**. In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed across any trunk group (**Trk Grp** column is blank) will be sent as a 5-digit calling number. This calling party number is sent to the far-end in the SIP “From” header.

```
display public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                     Total Administered: 1
 5  6                                       5           Maximum Entries: 9999
```

Step	Description
8.	**Route Pattern** A route pattern was configured that will route calls to the SIP trunk that connects to Avaya SES. This route pattern is used as a default route for SIP calls in **Step 9**. The example below shows the default SIP route pattern used for the compliance test. A descriptive name was entered for the **Pattern Name** field. The **Grp No** field was set to the trunk group created for the SIP trunk. The Facility Restriction Level (**FRL**) field was set 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 were used for all other fields.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: SES
                             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 M 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
```

9.	**Locations** By default, Avaya Communication Manager creates a location 1 with the name ***Main***. As part of the SIP installation procedures, the **change locations** command is used to assign a default SIP route pattern for this location. The default route pattern is the route pattern configured in **Step 8** and is entered in the **Proxy Sel Rte Pat** field. The default values were used for all other fields.

```
display locations                                               Page   1 of  16
                                  LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc  Name           Timezone Rule  NPA  ARS  Atd      Disp   Prefix   Proxy Sel
 No                   Offset             FAC  FAC      Parm              Rte Pat
 1:  Main           + 00:00   0                         1                 1
 2:                     :
 3:                     :
```

Step	Description
10.	**Routing Outgoing Calls to the PSTN** ARS was used to route calls to the PSTN. If the Multiple Location feature is enabled (see **Section 3.2**, **Step 1**), ARS supports multiple ARS Digit Analysis Tables. A separate table is supported for each location as well as a general table which is used if a matching entry cannot be found in the location specific table. In the compliance test, the location 1 table was blank so that location 1 (the main site) would use the general table for ARS routing. PSTN numbers that begin with 1732852 and 1763717 were used for testing. In the example below the general table is shown, as indicated by the **Location** field being set to ***all***. In this example, the first entry indicates that PSTN numbers that begin with 1732852 and 11 digits long use route pattern 4. The second entry indicates that PSTN numbers that begin with 1763717 and 11 digits long also use route pattern 4. Route pattern 4 routes calls to the ISDN-PRI trunk between the main site and the PSTN shown in **Figure 1**. The configuration of the PRI trunk is beyond the scope of these Application Notes.

```
display ars analysis 1732                                     Page   1 of   2
                              ARS DIGIT ANALYSIS TABLE
                                    Location:  all          Percent Full:    1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    1732852               11   11    4        fnpa         n
    1763717               11   11    4        fnpa         n
```

Step	Description
11.	**Mapping Incoming PSTN Calls to Local Extensions** PSTN numbers were mapped to extensions at the main or branch office using the **change inc-call-handling-trmt trunk-group *n*** command, where ***n*** is the trunk group connected to the PSTN from the Avaya 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   68001
 tie              11  17325551235       11   68003
```

3.2. Multiple Location Support

This section describes the configuration necessary to define the physical branch location as a second logical location within Avaya Communication Manager with separate routing tables from the main site. These steps are not necessary if routing based on the call origination location is not required. In the case where it is not necessary to route calls based on the call origination location, the two physical locations could still be represented by a single logical location within Avaya Communication Manager.

Step	Description
1.	**Enable Multiple Locations** Use the **display system-parameters customer-options** command to verify that the **Multiple Locations** field on **Page 5** has been set to ***y***. If it is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.
2.	**Create Second Location** Use the **change locations** command to define a second location and assign the default SIP route pattern to the location. In the compliance test, the second location is named ***Multitech***. Enter the same default route pattern that is used for location ***Main*** in the **Proxy Sel Rte Pat** field. The default values may be retained for all other fields.

```
change locations                                                Page   1 of  16
                                   LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc  Name           Timezone Rule  NPA  ARS  Atd      Disp   Prefix   Proxy Sel
 No                   Offset             FAC  FAC      Parm              Rte Pat
 1:  Main            + 00:00   0                        1                 1
 2:  Multitech       + 00:00   0                        1                 1
 3:                      :
```

Step	Description
3.	**IP Network Region** Use the **change ip-network-region** command to define an IP network region for location 2 which represents the branch site. Use the same values shown in **Section 3.1**, **Step 1** for IP network region 1 with the following exceptions. Set the **Location** field to ***2*** and the **Name** field to a descriptive name for the branch site. For the compliance test, IP network region 3 was configured for location 2.

```
change ip-network-region 3                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location: 2       Authoritative Domain: avremote.com
    Name: Multitech
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      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.	**IP Network Region – Continued** On **Page 3**, define the codec set to be used when placing calls between IP network region 3 and IP network region 1 (shown in **Section 3.1**, **Step 1**). Locate the entry in the table for **src rgn 3** and **dst rgn 1** and enter the desired codec set. Default values may be used for all other fields. A corresponding entry will be automatically created on **Page 3** of the IP Network Region 1 form for calls using source region 1 (**src rgn 1**) and destination region 3 (**dst rgn 3**).

```
change ip-network-region 3                                     Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video                           Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Intervening-regions  CAC IGAR
 3   1    3     y    NoLimit                                                  n
 3   2
 3   3    3
```

Step	Description
5.	**IP Network Map** Use the **change ip-network-map** command to define what IP address range will be assigned to IP network region 3. Any IP address not defined in the table defaults to IP network region 1. For the compliance test, all IP addresses on the 192.168.50.0 network are located at the branch site. Enter ***192.168.50.0*** in the **From IP Address** field. Enter ***192.168.50.255*** in the **To IP Address** field. Set the **Region** field to ***3***.

```
change ip-network-map                                          Page   1 of  32
                         IP ADDRESS MAPPING

                                                             Emergency
                                    Subnet                   Location
 From IP Address  (To IP Address   or Mask)  Region    VLAN   Extension
 192.168.50 .0    192.168.50 .255             3        n
```

6.	**Codecs** Use the **change ip-codec-set** command to set the values for ip-codec-set 3 to be the same as ip-codec-set 1 as shown in **Section 3.1**, **Steps 2 – 3**.
7.	**Signaling Group** Create a new SIP signaling group using the same procedure as shown in **Section 3.1**, **Step 4**. Use the same parameters with the following exception. Set the **Far-end Domain** field to the IP address of the MultiVOIP SS. This signaling group is used by PSTN outbound calls that are routed back to the MultiVOIP SS for accessing the PSTN via the MultiVOIP SS FXO port. The compliance test used signaling group 50 as shown below.

```
add signaling-group 50                                         Page   1 of   1
                                SIGNALING GROUP

 Group Number: 50             Group Type: sip
                        Transport Method: tls



   Near-end Node Name: clan1                Far-end Node Name: ses
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 3
       Far-end Domain: 192.168.50.88

                                           Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                      IP Audio Hairpinning? n
        Enable Layer 3 Test? n
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>8.</td><td>
Trunk Group
Create a new trunk group using the same procedure as shown in Section 3.1, Step 5 - 6. Use the same parameters with the following exceptions. Use unique values for the Group Name and TAC fields. Set the Signaling Group field to the signaling group number created in the previous step. The compliance test used trunk group 50 with the following values.

Group Name: <i>Multitech</i>
TAC: <i>150</i>
Signaling Group: <i>50</i>

<pre>
add trunk-group 50   Page   1 of 21
                                TRUNK GROUP

Group Number: 50   Group Type: sip   CDR Reports: y
  Group Name: Multitech   COR: 1   TN: 1   TAC: 150
   Direction: two-way   Outgoing Display? n
 Dial Access? n   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n

                                                       Signaling Group: 50
                                                     Number of Members: 10
</pre>
</td></tr>
<tr><td>9.</td><td>
Public Unknown Numbering
This second trunk will use the public unknown numbering entry created in Section 3.1, Step 7 since this entry applies to all trunks.
</td></tr>
</table>

Step | **Description**

10. **Route Pattern**

Create a route pattern for use by ARS when routing PSTN calls to the branch site. To do this, 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 created in **Step 8**. 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 Prefix Mark (**Pfx Mrk**) field is set to ***1***. The Prefix Mark defines how to process the 1 on user dialed 1 + 10 digit calls. If the Prefix Mark is set to 1, then the user dialed 1 is not suppressed and all 11 digits are sent to the far end. In the **Inserted Digits** field, enter a **9**. This is required by the MultiVOIP SS for proper routing. The default values may be retained for all other fields.

```
change route-pattern 50                                         Page   1 of   3
                    Pattern Number: 50  Pattern Name: Multitech
                             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: 50   0       1                9                                     n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

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

11. **ARS Routing**

Use the **change ars analysis *n* location 2** command to add an entry in the location 2 specific ARS Digit Analysis Table for the dialed string beginning with ***n***. In the example shown, the first entry defines PSTN numbers that begin with 1732852 and 11 digits long will use route pattern 50. The second entry defines that PSTN numbers that begin with 1763717 and 11 digits long will also use route pattern 50. Set the **Call Type** field to ***fnpa*** for both. Route pattern 50 routes calls to the second SIP trunk between the main site and the branch with the far-end domain set to the IP address of the MultiVOIP SS. The default values may be retained for all other fields.

```
change ars analysis 17 location 2                               Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  2             Percent Full:    1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    1732852               11   11    50       fnpa         n
    1763717               11   11    50       fnpa         n
```

3.3. OPS Configuration

This section describes the configuration of OPS stations, which is required for each analog endpoint connected to the MultiVOIP SS and each SIP endpoint. These Application Notes assume that all necessary configuration has been performed for the SIP endpoints at the main location including the creation of OPS stations. This section will only focus on the endpoints at the branch.

Step	Description
1.	**System Parameters** All SIP stations are configured as OPS stations on Avaya Communication Manager. This includes the analog telephones, and fax machine connected to the FXS ports of the MultiVOIP SS, 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  11
                                OPTIONAL FEATURES

     G3 Version: V15                              Software Package: Standard
       Location: 1                             RFA System ID (SID): 1
       Platform: 12                            RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 3200  120
                                      Maximum Stations: 2400  50
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 0     0
                    Maximum Off-PBX Telephones -   OPS: 300   34
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

Step	Description
2.	**Stations** To add a station, use the **add station *n*** command where ***n*** is an unused extension number. In the **Type** field, enter the phone type. For the analog endpoints connected to the MultiVOIP SS, a phone type of ***4620*** was used. Enter ***IP*** in the **Port** field. Enter a descriptive name in the **Name** field. In the case of the Avaya one-X Desktop Edition, the **IP SoftPhone** field must be set to ***y***. The default values may be retained for all other fields. The example below shows the configuration of one of the analog endpoints.

```
add station 60010                                               Page   1 of   5
                                     STATION

Extension: 60010                         Lock Messages? n               BCC: 0
     Type: 4620                          Security Code:                  TN: 1
     Port: IP                          Coverage Path 1:                 COR: 1
     Name: SIP60010                    Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 60010
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n


                                              Customizable Labels? y
```

Step	Description
3.	**Stations – Continued** 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. Set the **Bridged Call Alerting** field to ***y***. This will allow this station to ring on a bridged call.

```
add station 60010                                               Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe              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
                                                         EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type:                       Display Client Redirection? n
              AUDIX Name:                      Select Last Used Appearance? n
                                                 Coverage After Forwarding? s

                                               Direct IP-IP Audio Connections? y
  Emergency Location Ext: 60010         Always Use? n IP Audio Hairpinning? n
```

Step	Description
4.	**Stations – Continued** On **Page 3**, under BUTTON ASSIGNMENTS, create 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 MultiVOIP SS. The analog endpoints used in the compliance test were all configured with two call appearances. Some Feature Name Extensions (FNEs) require the assignment of feature buttons in order to operate. The Automatic Callback FNE requires the assignment of an ***auto-cback*** button. This button assignment is shown in the example below.

```
add station 60010                                              Page   4 of   5
                                     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:
 3:                                     7:
 4:                                     8:
```

Step	Description
5.	**Off-pbx Station Mapping** Map the Avaya Communication Manager extension to the Avaya SES media server extension defined in **Section 4.2**, **Step 2** 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 defined in **Section 3.1**, **Step 5**. ▪ **Configuration Set**: Enter a valid configuration set which contain the default values.

```
add off-pbx-telephone station-mapping 60010                   Page   1 of   2
                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station        Application Dial   CC  Phone Number      Trunk       Config
Extension                  Prefix                       Selection   Set
60010             OPS           -     60010              1          1
                               -
```

Step	Description
6.	**Off-pbx Station Mapping - Continued** On **Page 2**, set the **Call Limit** to the number of call appearances set on the station form in **Step 4**. Verify that the **Mapping Mode** is set to ***both***. This setting allows the OPS station to both originate and terminate calls. Set the **Bridged Calls** field to ***both*** to allow bridging on this extension. The default values may be retained for all other fields.

```
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
 30108        2         both           all         both
```

Step	Description
7.	Repeat **Steps 2 - 6** for each remaining endpoint located at the branch office. The branch office has four user endpoints: two analog endpoints connected to the MultiVOIP SS used for telephones or a fax machine (x60010 and x60011), two Avaya 4600 Series SIP Telephones (x60007 and 60008), one Avaya 9600 Series SIP Telephone (x60006) and a Avaya one-X Desktop Edition SIP (x60009).

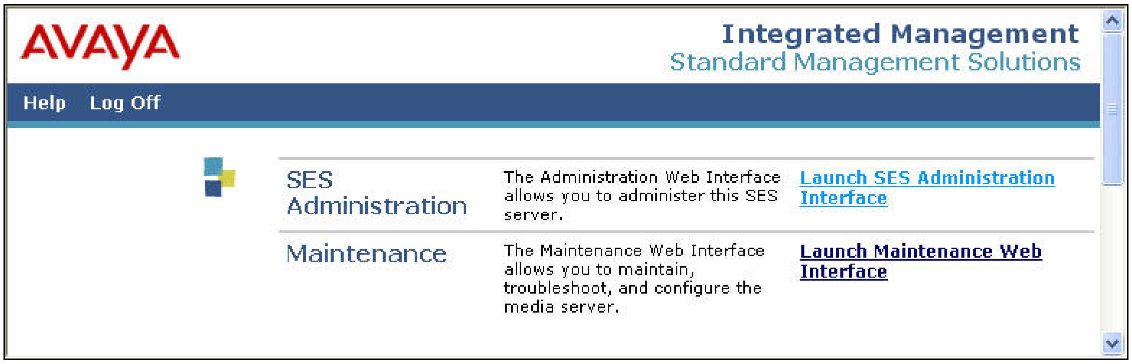
4. Configure Avaya SIP Enablement Services

This section covers the configuration of Avaya SES at the main site. Avaya SES is configured via an Internet browser using the administration web interface. It is assumed that the Avaya SES software and the license file have already been installed on the server. During the software installation, an installation script is run from the Linux shell of the server to specify the IP network properties of the server along with other parameters. In addition, it is assumed that the setup screens of the administration web interface have been used to initially configure Avaya SES. For additional information on these installation tasks, refer to [4].

This section is divided into two parts. **Section 4.1** will summarize the user-defined parameters used in the installation procedures that are important to understanding the solution as a whole. It will not attempt to show the installation procedures in their entirety. It will also describe any deviations from the standard procedures, if any. **Section 4.2** will describe procedures beyond the initial SIP installation procedures that are necessary for interoperating with the MultiVOIP SS. This includes configuration of the SIP endpoints at the branch. The creation of users and media server extensions for the SIP endpoints at the main site are not covered here. These procedures are covered in [4].

4.1. Summary of Initial Configuration Parameters

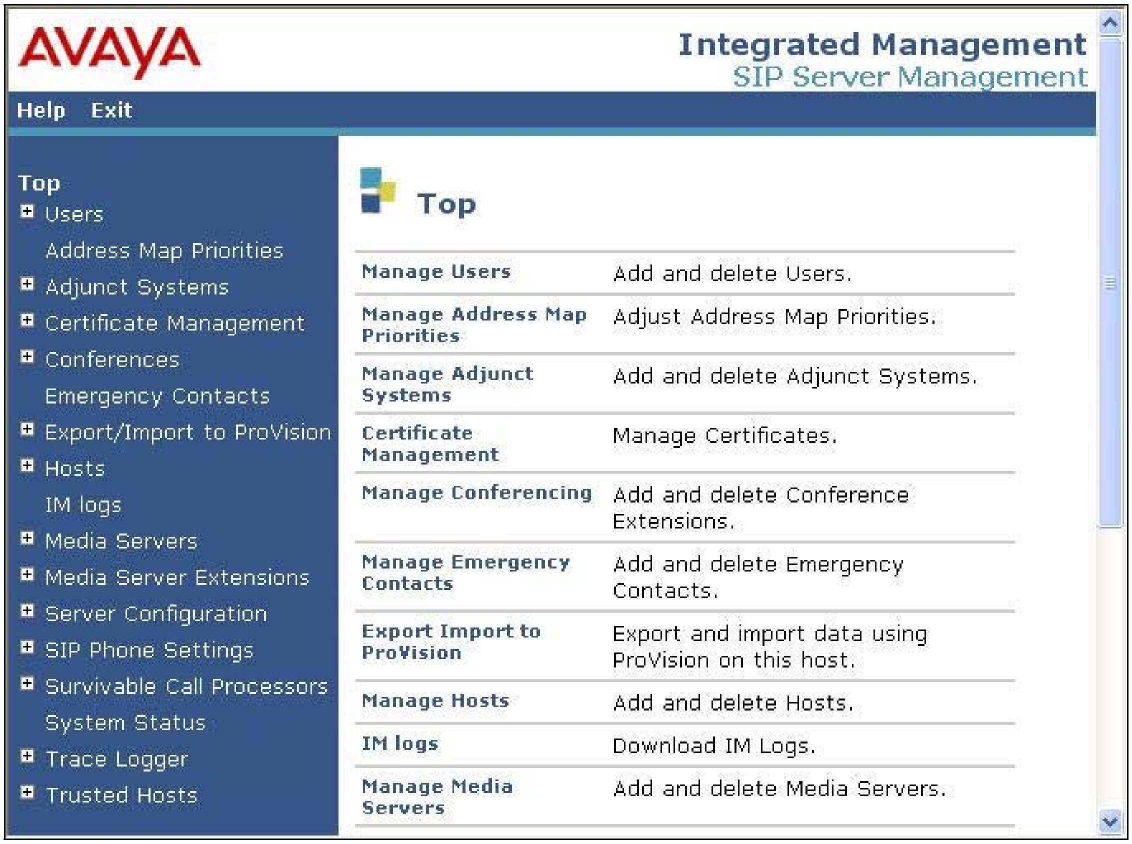
This section summarizes the applicable user-defined parameters used during the SIP installation procedures.

Step	Description
1.	**Login** 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 SES Administration Interface** link from the main page as shown below.  SES Administration — The Administration Web Interface allows you to administer this SES server. — Launch SES Administration Interface Maintenance — The Maintenance Web Interface allows you to maintain, troubleshoot, and configure the media server. — Launch Maintenance Web Interface

Step **Description**

2. **Top Page**

The Avaya SES **Top** Page will be displayed as shown below.



Manage Users	Add and delete Users.
Manage Address Map Priorities	Adjust Address Map Priorities.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Certificate Management	Manage Certificates.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Export Import to ProVision	Export and import data using ProVision on this host.
Manage Hosts	Add and delete Hosts.
IM logs	Download IM Logs.
Manage Media Servers	Add and delete Media Servers.

3. **Initial Configuration Parameters**

As part of the Avaya SES installation and initial configuration procedures, the following parameters were defined. Although these procedures are out of the scope of these Application Notes, the values used in the compliance test are shown below for reference. After each group of parameters is a brief description of how to view the values for that group from the Avaya SES **Top** page shown in the previous step.

- SIP Domain: ***avremote.com***
  (To view, navigate to **Server Configuration→System Properties**)

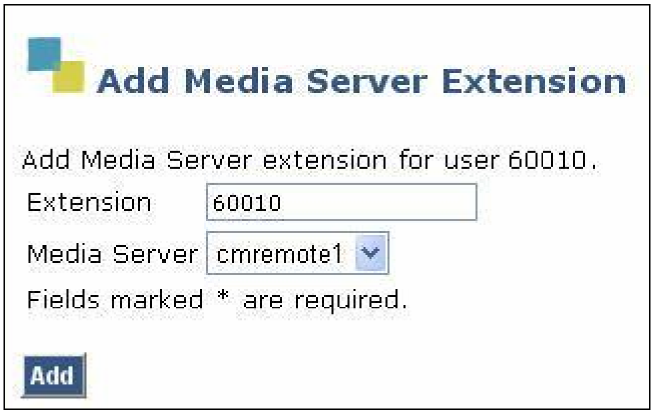
- Host IP Address (SES IP address): ***192.168.100.200***
- Host Type: ***SES combined home-edge***
  (To view, navigate to **Hosts→List**; click **Edit**)

- Media Server (Avaya Communication Manager) Interface Name: ***cmremote1***
- SIP Trunk Link Type: ***TLS***
- SIP Trunk IP Address (CLAN IP address): ***192.168.100.202***
  (To view, navigate to **Media Servers→List**; click **Edit**)

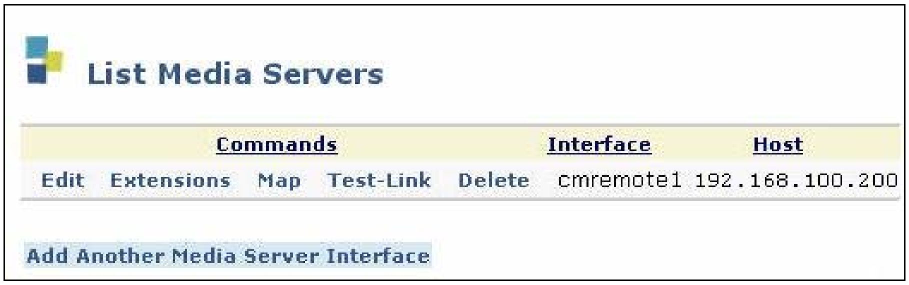
4.2. MultiVOIP SS Specific Configuration

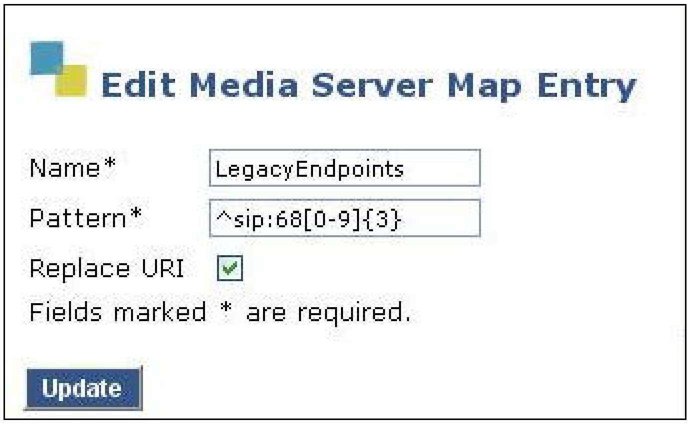
This section describes additional configuration necessary for interoperating with the MultiVOIP SS. In particular, this section describes the configuration of user and media server extensions for the FXS/FXO ports of the MultiVOIP SS and the SIP endpoints at the branch.

Step	Description
1.	**SIP Users For FXS (station) Ports and SIP Endpoints** A user must be added on Avaya SES for each of the extensions at the branch office created on Avaya Communication Manager in **Section 3.3**, **Steps 2 – 7**. 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.  Primary Handle* 60010 User ID Password* •••••• Confirm Password* •••••• Host* 192.168.100.200 First Name* ext60010 Last Name* SIP Address 1 Address 2 Office City State Country Zip Survivable Call Processor none Add Media Server Extension ☑ Fields marked * are required.

Step	Description
2.	**Media Server Extension** The **Add Media Server Extension** page will appear. In the **Extension** field, enter the Avaya Communication Manager extension associated with this user created in **Section 3.3**, **Step 2**. In the **Media Server** field, select from the pull-down menu the name of the media server shown in **Section 4.1**, **Step 3**. Click the **Add** button to complete the operation.  Add Media Server Extension Add Media Server extension for user 60010. Extension 60010 Media Server cmremote1 Fields marked * are required.
3.	Repeat **Steps 1 - 2** for each of the remaining stations at the branch office.

Step	Description
4.	**SIP User for FXO (Trunk) Port** In addition, a user must be added on Avaya SES for each of the FXO (trunk) ports on the MultiVOIP SS. These users will not have media server extensions assigned to them. From the left pane, navigate to **Users → Add**. Enter the values as shown below. ▪ **Primary Handle**: Enter all or part of the PSTN number assigned to the FXO port. This digit string will be the SIP user name that this port uses to register with the Avaya SES. ▪ **Password**: Enter a valid password for this user. ▪ **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. Do not 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.  Add User Primary Handle* 91763 User ID Password* •••••• Confirm Password* •••••• Host* 192.168.100.200 First Name* Multitech Last Name* Trunk Address 1 Address 2 Office City State Country Zip Survivable Call Processor none Add Media Server Extension ☐ Fields marked * are required. Add
5.	Repeat **Step 4** for any remaining FXO ports on the MultiVOIP SS. The compliance test used only one of the available FXO ports.

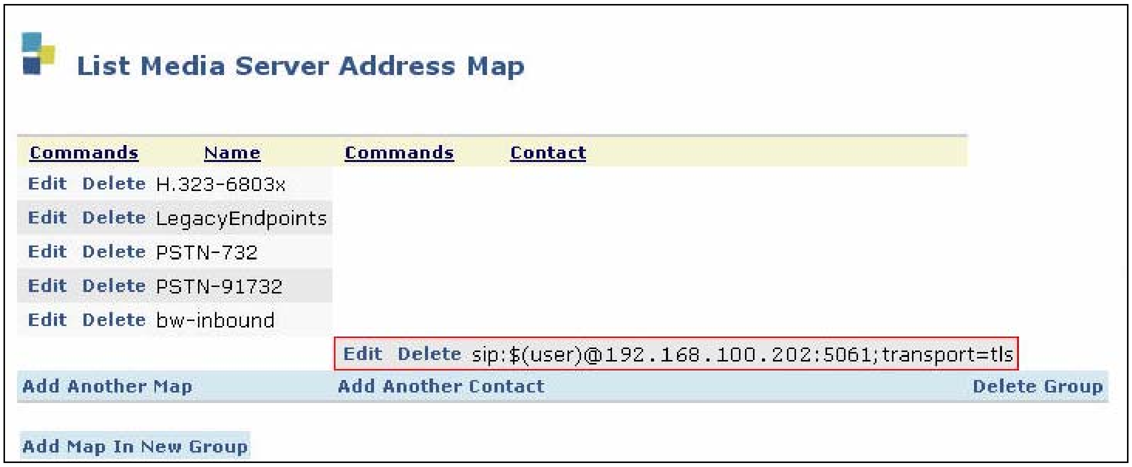
Step	Description
6.	**Media Server Address Map - Overview** A media server address map is needed to route calls from the FXO ports on the MultiVOIP SS to a non-SIP phone at the main site. This is because neither the caller nor the called party is a registered user on Avaya SES with a media server extension assigned to it. Thus, Avaya SES does not know to route this call to Avaya Communication Manager. Thus to accomplish this task, a media server address map is needed. To view the configured media server address maps, navigate to **Media Server→List** in the left pane. Click the **Map** link next to the media server name described in **Section 4.1**, **Step 3**.  List Media Servers Commands Interface Host Edit Extensions Map Test-Link Delete cmremote1 192.168.100.200 Add Another Media Server Interface
7.	**Media Server Address Map – List** The **List Media Server Address Map** page appears showing the list of configured media server address maps. Each map defines criteria for matching calls based on the contents of the SIP Request-URI received by Avaya SES. If a call matches the map, then the call is directed to the **Contact**. In the example below, five maps are shown. Only the maps named ***LegacyEndpoints*** and ***PSTN-91732*** were used for the compliance test. To view or edit the call matching criteria of the map, click the **Edit** link next to the map name.  List Media Server Address Map Commands Name Commands Contact Edit Delete H.323-6803x Edit Delete LegacyEndpoints Edit Delete PSTN-732 Edit Delete PSTN-91732 Edit Delete bw-inbound Edit Delete sip:$(user)@192.168.100.202:5061;transport=tls Add Another Map Add Another Contact Delete Group Add Map In New Group

Step	Description
8.	**Media Server Address Map – Criteria** The content of the media server address map is described below. • **Name**: Contains any descriptive name • **Pattern**: Contains an expression to define the matching criteria for calls to be routed from the MultiVOIP SS FXO port to Avaya Communication Manager at the main site. Information on the syntax used for address map patterns can be found in [4]. • **Replace URI**: Check the box. If any changes are made, click **Update**. For the address map named ***LegacyEndpoints***, the pattern will match any URI that begins with ***sip:68*** followed by any digit between ***0-9*** for the next ***3*** digits.  Edit Media Server Map Entry Name* LegacyEndpoints Pattern* ^sip:68[0-9]{3} Replace URI ☑ Fields marked * are required. Update For the address map named ***PSTN-91732***, the pattern will match any URI that begins with ***sip:91732***.  Edit Media Server Map Entry Name* PSTN-91732 Pattern* ^sip:91732 Replace URI ☑ Fields marked * are required. Update

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>9.</td><td>

Media Server Address Map – Contact

For the first Media Server Address Map created, the **Contact** information is populated automatically and its syntax is shown in the example below. The user portion of the original request URI is substituted for ***$(user)*** in the expression. The contact specifies the media server IP address (***192.168.100.202***), port number (***5061***) and transport protocol to use to reach the media server. The media server IP address corresponds to the **SIP Trunk IP Address** shown in **Section 4.1**, **Step 3**.



List Media Server Address Map

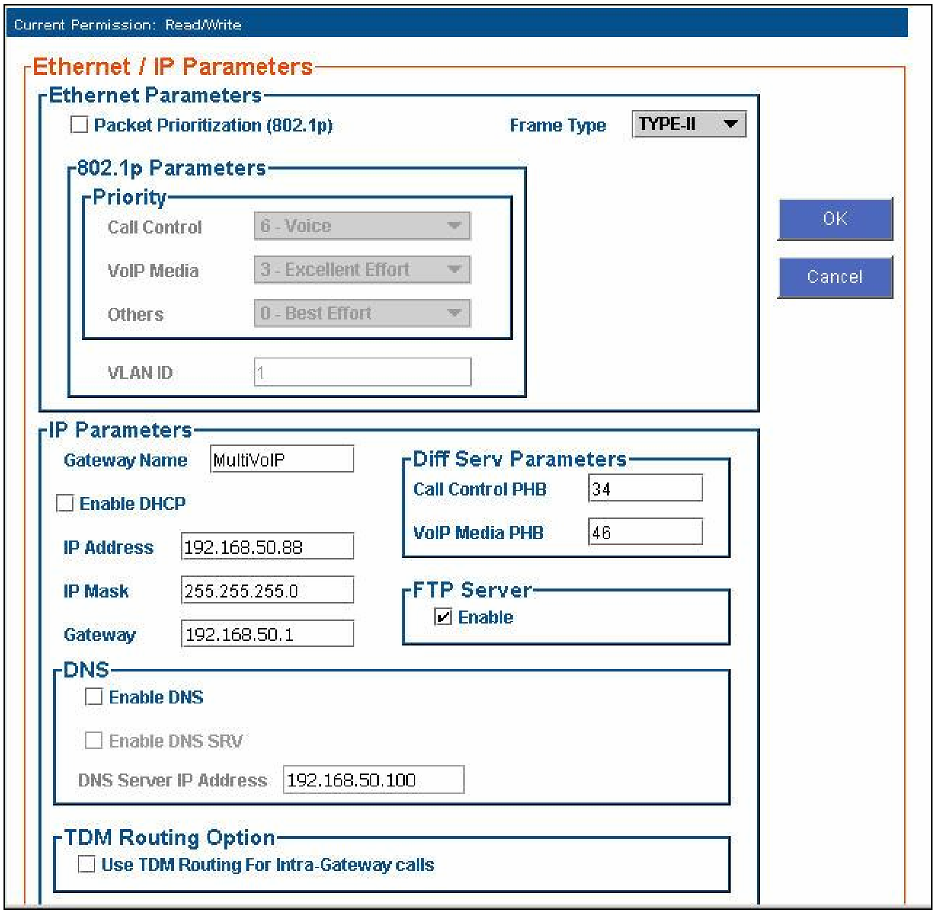
<table>
<tr><th>Commands</th><th>Name</th><th>Commands</th><th>Contact</th></tr>
<tr><td>Edit Delete</td><td>H.323-6803x</td><td></td><td></td></tr>
<tr><td>Edit Delete</td><td>LegacyEndpoints</td><td></td><td></td></tr>
<tr><td>Edit Delete</td><td>PSTN-732</td><td></td><td></td></tr>
<tr><td>Edit Delete</td><td>PSTN-91732</td><td></td><td></td></tr>
<tr><td>Edit Delete</td><td>bw-inbound</td><td></td><td></td></tr>
<tr><td></td><td></td><td>Edit Delete</td><td>sip:$(user)@192.168.100.202:5061;transport=tls</td></tr>
<tr><td>Add Another Map</td><td></td><td>Add Another Contact</td><td>Delete Group</td></tr>
</table>

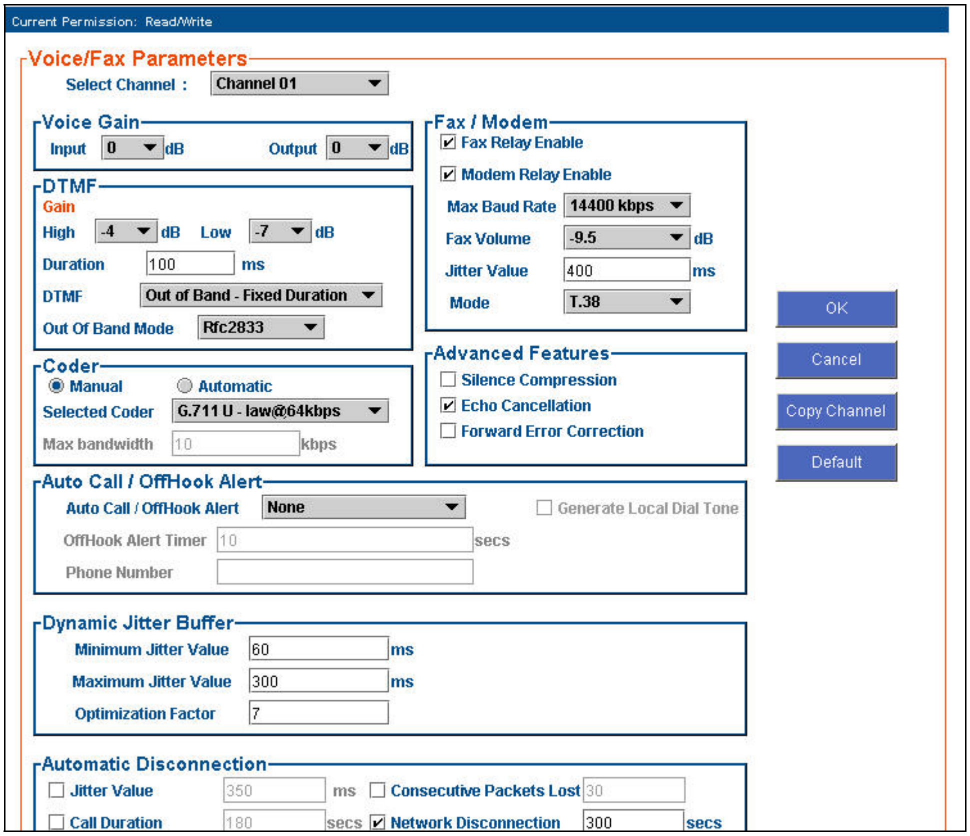
</td></tr>
</table>

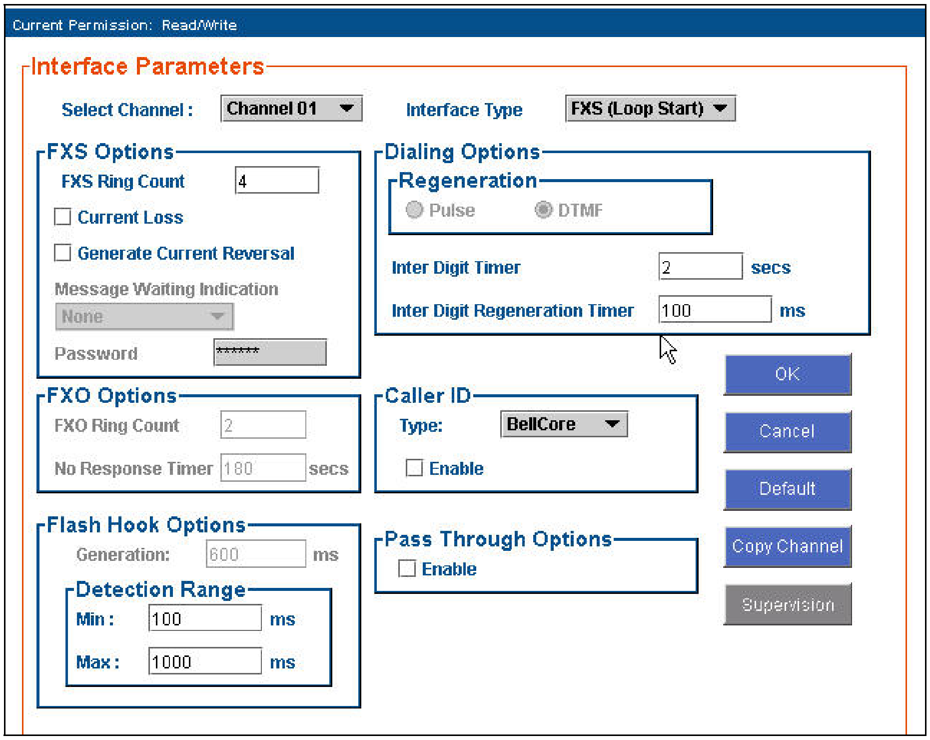
5. Configure the Multi-Tech MultiVOIP SS

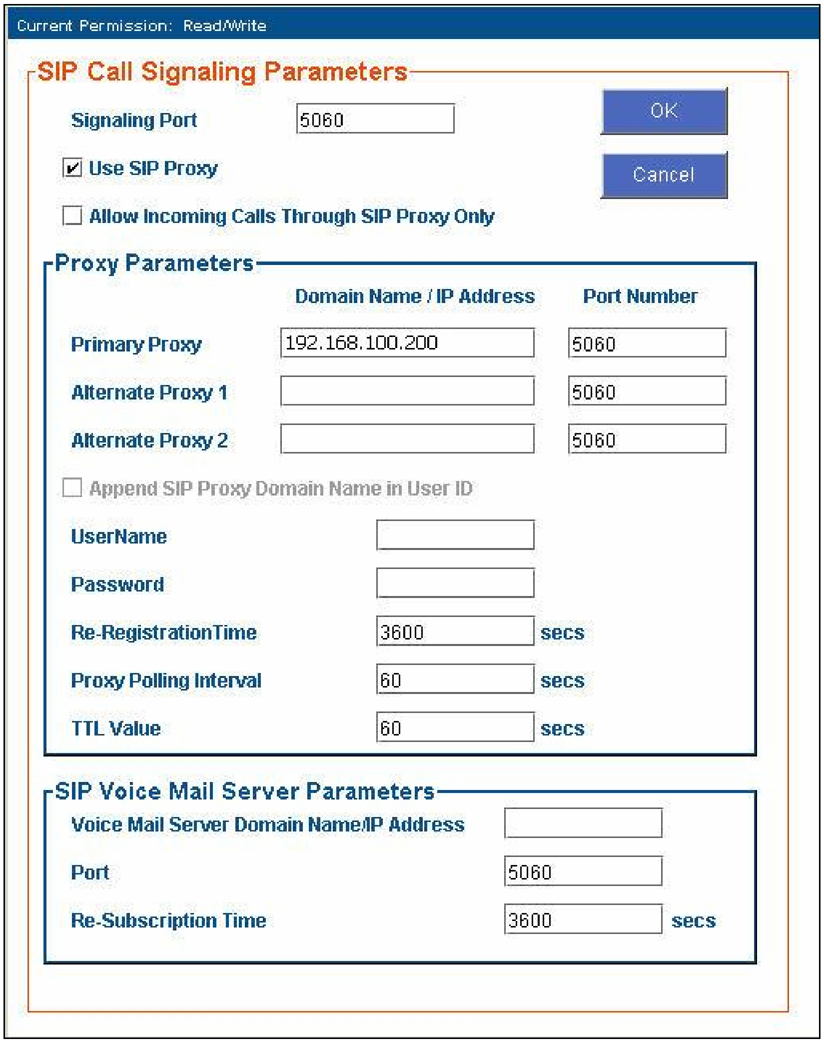
This section describes the configuration of the MultiVOIP SS.

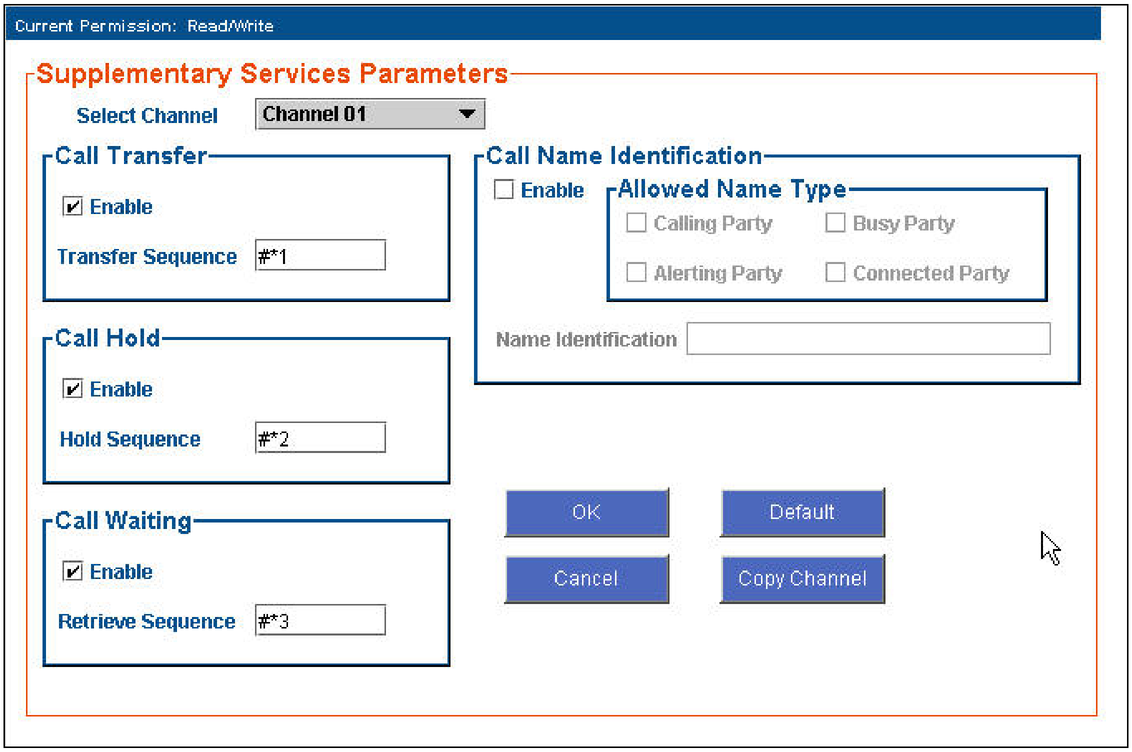
Step	Description
1.	**Login** The MultiVOIP SS is configured via a web interface accessed by entering the IP address of the MultiVOIP SS into an Internet browser. Log in with the appropriate credentials. The main page will appear as shown below. Subsequent screens will be accessed by navigating the menu in the left pane.  MultiVOIP-SS Web Based Configuration Multi-Tech Systems Inc., 2205 Woodale Drive, Mounds View, MN 55112 USA TEL : {763} 785-3500 FAX : {763} 785-3702 Tech Support : 800 972-2439 Web site : http://www.multitech.com FTP site : ftp://ftp.multitech.com Tech Support : tsupport@multitech.com

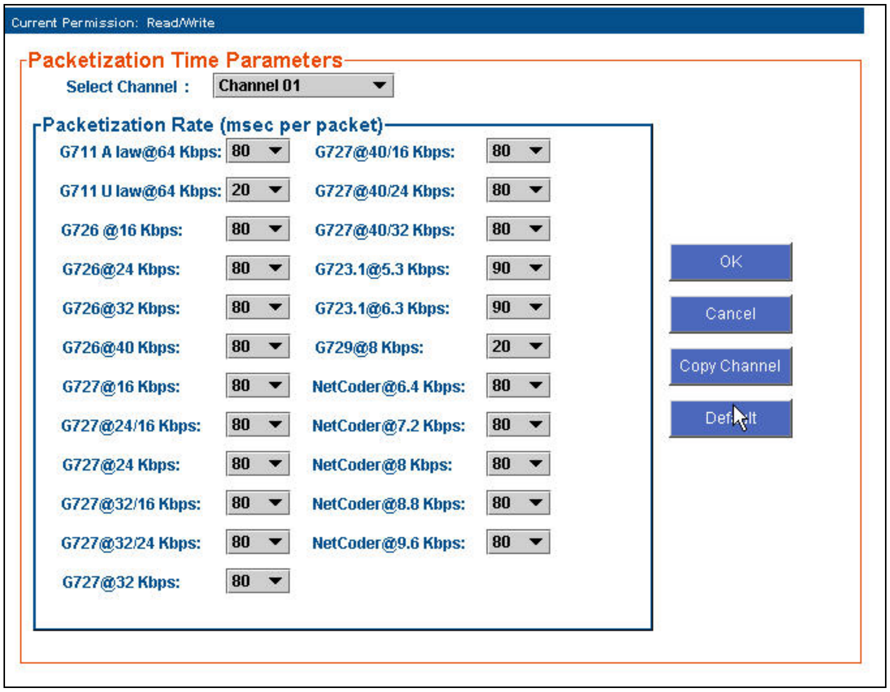
Step	Description
2.	**Network Parameters** Navigate to **Configuration→Ethernet/IP** in the left pane. The **IP Address**, **IP Mask** and **Gateway** fields are set to values consistent with **Figure 1**.  Current Permission: Read/Write Ethernet / IP Parameters Ethernet Parameters ☐ Packet Prioritization (802.1p) Frame Type TYPE-II 802.1p Parameters Priority Call Control 6 - Voice VoIP Media 3 - Excellent Effort Others 0 - Best Effort VLAN ID 1 IP Parameters Gateway Name MultiVoIP ☐ Enable DHCP IP Address 192.168.50.88 IP Mask 255.255.255.0 Gateway 192.168.50.1 Diff Serv Parameters Call Control PHB 34 VoIP Media PHB 46 FTP Server ☑ Enable DNS ☐ Enable DNS ☐ Enable DNS SRV DNS Server IP Address 192.168.50.100 TDM Routing Option ☐ Use TDM Routing For Intra-Gateway calls

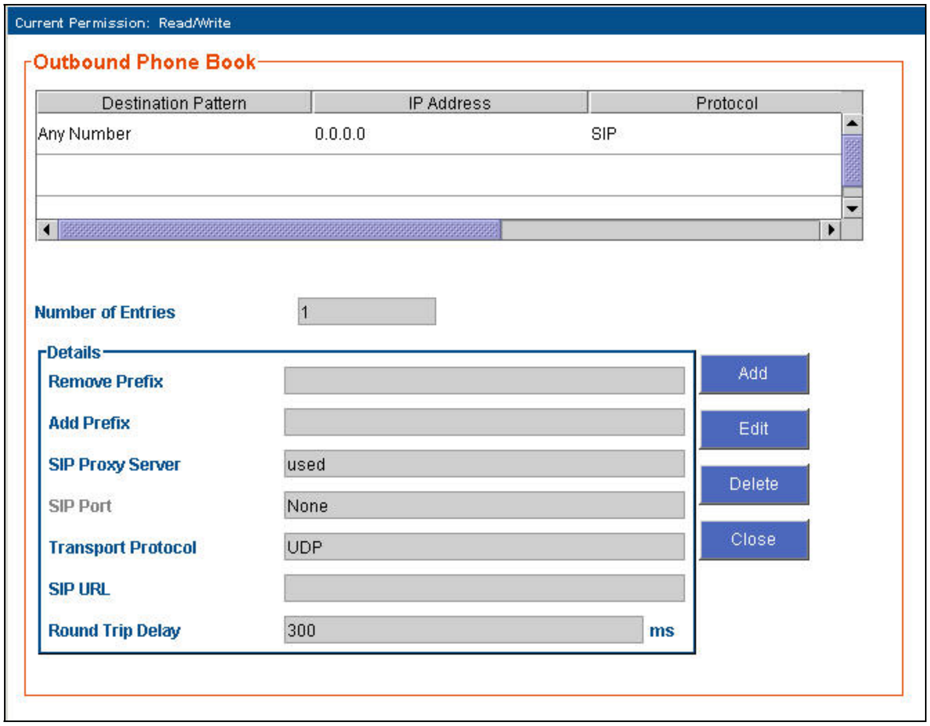
Step	Description
3.	**Voice/Fax** Navigate to **Configuration → Voice/Fax** in the left pane. For the compliance test, these parameters were set the same for each channel used. Select the channel to be configured from the **Select Channel** pull-down menu. Select the desired codec from the **Selected Coder** pull-down menu. Only one codec can be selected at a time. To support fax, locate the **Fax/Modem** section and select **T.38** from the **Mode** pull-down menu. Default values may be used for all other fields. Click the **OK** button.  Current Permission: Read/Write Voice/Fax Parameters Select Channel : Channel 01 Voice Gain Input 0 dB Output 0 dB DTMF Gain High -4 dB Low -7 dB Duration 100 ms DTMF Out of Band - Fixed Duration Out Of Band Mode Rfc2833 Coder ● Manual ○ Automatic Selected Coder G.711 U - law@64kbps Max bandwidth 10 kbps Auto Call / OffHook Alert Auto Call / OffHook Alert None ☐ Generate Local Dial Tone OffHook Alert Timer 10 secs Phone Number Dynamic Jitter Buffer Minimum Jitter Value 60 ms Maximum Jitter Value 300 ms Optimization Factor 7 Automatic Disconnection ☐ Jitter Value 350 ms ☐ Consecutive Packets Lost 30 ☐ Call Duration 180 secs ☑ Network Disconnection 300 secs Fax / Modem ☑ Fax Relay Enable ☑ Modem Relay Enable Max Baud Rate 14400 kbps Fax Volume -9.5 dB Jitter Value 400 ms Mode T.38 Advanced Features ☐ Silence Compression ☑ Echo Cancellation ☐ Forward Error Correction

Step	Description
4.	**Interface Parameters** Navigate to **Configuration → Interface** in the left pane. Select the channel to be configured from the **Select Channel** pull-down menu. Select the interface type for this channel from the **Interface Type** pull-down menu. For the compliance test, channels 1 and 2 were set to **FXS (Loop Start)** and channel 4 was set to **FXO**. The example below shows the settings for channel 1. Default values may be used for all other fields. Click the **OK** button.  Current Permission: Read/Write Interface Parameters Select Channel : Channel 01 Interface Type FXS (Loop Start) FXS Options FXS Ring Count 4 ☐ Current Loss ☐ Generate Current Reversal Message Waiting Indication None Password ****** FXO Options FXO Ring Count 2 No Response Timer 180 secs Flash Hook Options Generation: 600 ms Detection Range Min : 100 ms Max : 1000 ms Dialing Options Regeneration ○ Pulse ● DTMF Inter Digit Timer 2 secs Inter Digit Regeneration Timer 100 ms Caller ID Type: BellCore ☐ Enable Pass Through Options ☐ Enable

Step	Description
5.	**SIP Signaling** Navigate to **Configuration→SIP Call Signaling**. Check the box for **Use SIP Proxy**. Enter the IP address of the Avaya SES in the **Primary Proxy** field. Default values may be used for all other fields. Click the **OK** button.  Current Permission: Read/Write SIP Call Signaling Parameters Signaling Port: 5060 ☑ Use SIP Proxy ☐ Allow Incoming Calls Through SIP Proxy Only Proxy Parameters Domain Name / IP Address; Port Number Primary Proxy: 192.168.100.200; 5060 Alternate Proxy 1: ; 5060 Alternate Proxy 2: ; 5060 ☐ Append SIP Proxy Domain Name in User ID UserName: Password: Re-RegistrationTime: 3600 secs Proxy Polling Interval: 60 secs TTL Value: 60 secs SIP Voice Mail Server Parameters Voice Mail Server Domain Name/IP Address: Port: 5060 Re-Subscription Time: 3600 secs

Step	Description
6.	**Supplementary Services** Navigate to **Configuration→Supplementary Services**. For the compliance test, these parameters were set the same for each channel used. Select the channel to be configured from the **Select Channel** pull-down menu. Check the **Enable** box for **Call Transfer**, **Call Hold** and **Call Waiting**. Default values may be used for all other fields. Click the **OK** button.  Current Permission: Read/Write Supplementary Services Parameters Select Channel: Channel 01 Call Transfer ☑ Enable Transfer Sequence: #*1 Call Hold ☑ Enable Hold Sequence: #*2 Call Waiting ☑ Enable Retrieve Sequence: #*3 Call Name Identification ☐ Enable Allowed Name Type ☐ Calling Party ☐ Busy Party ☐ Alerting Party ☐ Connected Party Name Identification:

Step	Description
7.	**Packetization** Navigate to **Advanced→Packetization Time**. For the compliance test, these parameters were set the same for each channel used. Select the channel to be configured from the **Select Channel** pull-down menu. Set the **Packetization Rate** for **G711Ulaw@64 Kbps** and **G729@8 Kbps** to ***20*** msec/packet. Default values may be used for all other fields. Click the **OK** button.  Select Channel : Channel 01 G711 A law@64 Kbps: 80 G711 U law@64 Kbps: 20 G726 @16 Kbps: 80 G726@24 Kbps: 80 G726@32 Kbps: 80 G726@40 Kbps: 80 G727@16 Kbps: 80 G727@24/16 Kbps: 80 G727@24 Kbps: 80 G727@32/16 Kbps: 80 G727@32/24 Kbps: 80 G727@32 Kbps: 80 G727@40/16 Kbps: 80 G727@40/24 Kbps: 80 G727@40/32 Kbps: 80 G723.1@5.3 Kbps: 90 G723.1@6.3 Kbps: 90 G729@8 Kbps: 20 NetCoder@6.4 Kbps: 80 NetCoder@7.2 Kbps: 80 NetCoder@8 Kbps: 80 NetCoder@8.8 Kbps: 80 NetCoder@9.6 Kbps: 80

Step	Description
8.	**Outbound Phone Book** Navigate to **Phone Book→Outbound Phone Book→List Entries**. The outbound phone book defines how calls are routed from the analog ports to the IP interface. Entries can be added or viewed by clicking the **Add** or **Edit** button respectively. The example below shows that for the compliance test, the **Destination Pattern** was set to ***Any Number***, **IP Address** was set to ***0.0.0.0*** and the **SIP Proxy Server** field was set to ***used***. This indicates that any dialed number from the analog ports is not routed to a specific IP address but instead is forwarded to the defined SIP proxy. Default values were used for all other fields. 

Current Permission: Read/Write

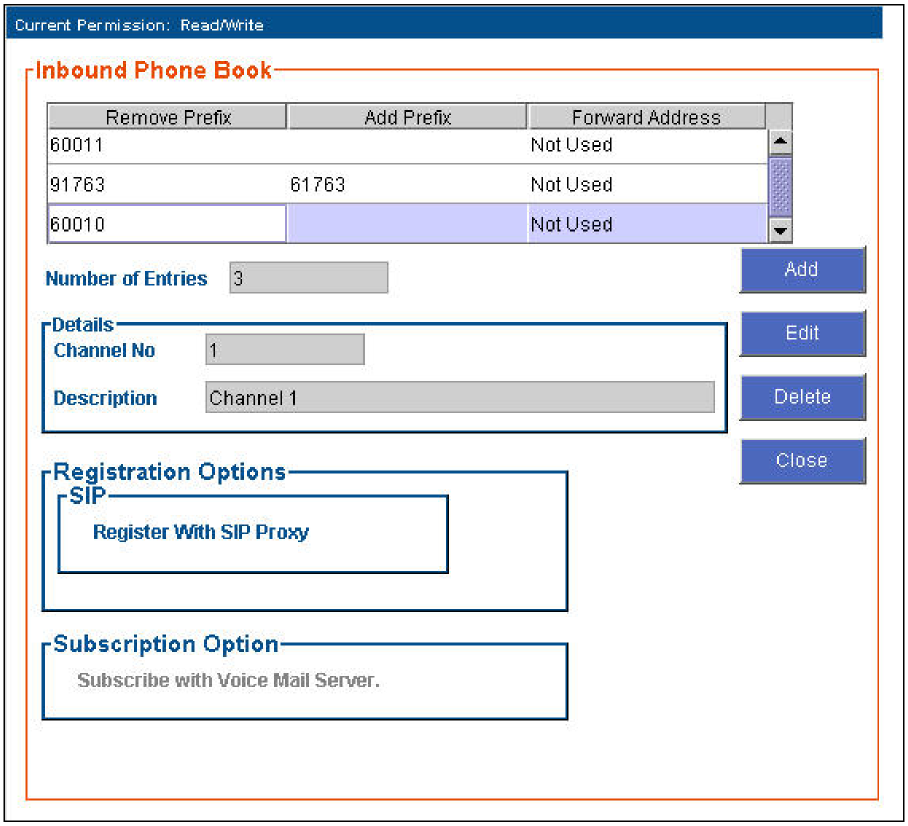
Outbound Phone Book

Destination Pattern	IP Address	Protocol
Any Number	0.0.0.0	SIP

Number of Entries 1

Details

Remove Prefix	
Add Prefix	
SIP Proxy Server	used
SIP Port	None
Transport Protocol	UDP
SIP URL	
Round Trip Delay	300 ms

Step	Description
9.	**Inbound Phone Book** Navigate to **Phone Book→Inbound Phone Book→List Entries**. Entries can be added or viewed by clicking the **Add** or **Edit** button respectively. The example below shows that for the compliance test, three entries were created; one for each channel used. Highlighting an entry displays its details in the lower half of the screen. The inbound phone book defines how calls are routed (and digits are manipulated) from the IP interface to the analog ports. The example below shows the details for the entry of ***60010***. The **Remove Prefix** field shows the digit string that will be matched and removed on incoming calls. The **Channel No** field in the **Details** section shows the destination channel. Thus, the highlighted entry indicates that incoming SIP calls that begin with ***60010*** will be routed to channel ***1***. The prefix ***60010*** will also be removed but it is no longer needed since the call is directed to a FXS station port. In addition, **Register With SIP Proxy** is shown under **Registration Options**. Thus, the **Remove Prefix** value will be used as the user name for registering this channel to Avaya SES. The password is not shown on this summary screen but can be entered when the entry is created. 

Current Permission: Read/Write

Inbound Phone Book

Remove Prefix	Add Prefix	Forward Address
60011		Not Used
91763	61763	Not Used
60010		Not Used

Number of Entries 3

Details

Channel No 1

Description Channel 1

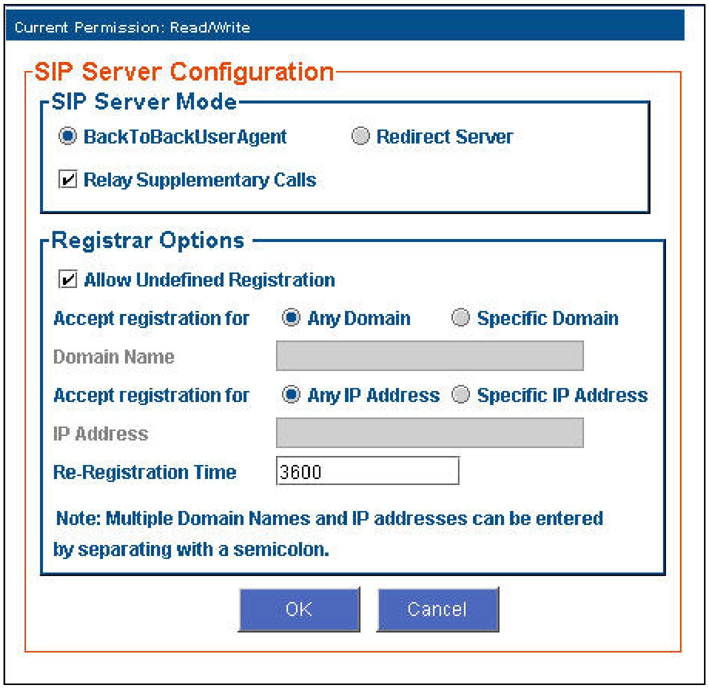
Registration Options

SIP

Register With SIP Proxy

Subscription Option

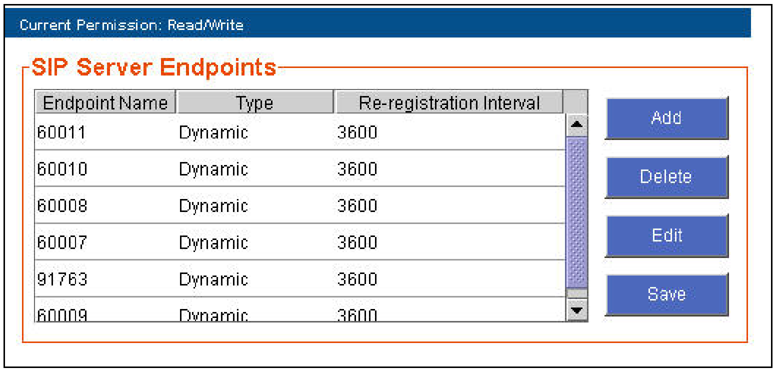
Subscribe with Voice Mail Server.

Step	Description
10.	**Inbound Phone Book - Continued** The Inbound Phone Book entry for prefix 60011 is created with the same settings as 60010 with the exception that the calls are directed to channel 2. The Inbound Phone Book entry for prefix 91763 is directed to channe1 4 (the FXO trunk port). The removed prefix is added back in the **Add Prefix** field with the ***9*** being replaced by a ***6*** (See **Step 9**). The prefix of 6 was the access code used by the equipment connected at the other end of the MultiVOIP SS FXO trunk to route the call to the PSTN. This channel is also configured to register with Avaya SES with the user name of 91763. It should be noted that inbound SIP calls destined for the FXO port (PSTN) must match an entry in the Inbound Phone Book in order to be allowed. Thus, the example used in the compliance test only allows PSTN calls to 1763xxxxxxx numbers (after the access code of 9 or 6 is removed). It does not allow calls to any arbitrary PSTN number. To allow dialing to any PSTN number via the FXO port, additional entries or a more broadly defined prefix would need to be used in the Inbound Phone Book.
11.	**SIP Server Configuration** Navigate to **SIP Server→Configuration**. In the **SIP Server Mode** section, select the **BackToBackUserAgent** and **Relay Supplementary Calls** options. Under **Registrar Options**, select **Allow Undefined Registration**. This will allow SIP endpoints to register with the MultiVOIP SS even if an endpoint entry has not been created on the MultiVOIP SS for that endpoint. Default values may be used for all other fields. Click the **OK** button.  Current Permission: Read/Write SIP Server Configuration SIP Server Mode ● BackToBackUserAgent ○ Redirect Server ☑ Relay Supplementary Calls Registrar Options ☑ Allow Undefined Registration Accept registration for ● Any Domain ○ Specific Domain Domain Name Accept registration for ● Any IP Address ○ Specific IP Address IP Address Re-Registration Time 3600 Note: Multiple Domain Names and IP addresses can be entered by separating with a semicolon.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>12.</td><td>

SIP Endpoints

Navigate to **SIP Server→Predefined Endpoints**. The SIP Server Endpoints table includes an entry for each endpoint known to the MultiVOIP SS whether it is a SIP telephone or one of the MultiVOIP SS analog ports. Entries can be added or viewed by clicking the **Add** or **Edit** button respectively. The example below shows the endpoints used for the compliance test. The password for each entry is not shown on this summary screen but can be entered when the entry is created. Default values were used for all other fields.



<table>
<tr><th>Endpoint Name</th><th>Type</th><th>Re-registration Interval</th></tr>
<tr><td>60011</td><td>Dynamic</td><td>3600</td></tr>
<tr><td>60010</td><td>Dynamic</td><td>3600</td></tr>
<tr><td>60008</td><td>Dynamic</td><td>3600</td></tr>
<tr><td>60007</td><td>Dynamic</td><td>3600</td></tr>
<tr><td>91763</td><td>Dynamic</td><td>3600</td></tr>
<tr><td>60009</td><td>Dynamic</td><td>3600</td></tr>
</table>

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

6. Interoperability Compliance Testing

This section describes the compliance testing used to verify the interoperability of the Multi-Tech MultiVOIP SS with 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 through the MultiVoIP at the branch site using various codec settings and exercising common PBX features. 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 MultiVOIP SS successfully passed compliance testing. The following features and functionality were verified when the data WAN was available.

- Calls to/from branch endpoints (analog and SIP) and the main site.
- Calls to/from analog endpoints at the branch (MultiVOIP SS FXS ports) and the PSTN (MultiVOIP SS FXO ports).
- Calls to/from SIP endpoints at the branch and the PSTN (MultiVOIP SS FXO ports).
- Intra-branch calls with both analog and SIP endpoints
- G.711mu and G.729AB codec support
- Proper recognition of DTMF transmissions
- 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.
- Conferencing (Avaya SIP telephone initiates a conference that includes an MultiVOIP SS analog endpoint or a FXO port)
- Extended telephony features using Avaya Communication Manager Feature Name Extensions such as Call Forwarding, Call Park, Call Pickup, Automatic Redial and Send All Calls. For more information on FNEs, please refer to [5].
- T.38 fax support
- Proper system recovery after a MultiVOIP SS restart

The following features and functionality were verified 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.
- Calls to/from analog endpoints at the branch (MultiVOIP SS FXS ports) and the PSTN (MultiVOIP SS FXO ports).
- Calls to/from SIP endpoints at the branch and the PSTN (MultiVOIP SS FXO ports).
- Intra-branch calls with both analog and SIP endpoints
- Local MultiVOIP SS support for Hold, Transfer, and Call Waiting
- Conferencing (Avaya SIP telephone initiates a conference that includes an MultiVOIP SS analog endpoint or a FXO port)

The following observations were made during the compliance test:

- The MultiVOIP SS supports only one codec at a time. It can not be configured with a list of codecs for negotiation.
- The MultiVOIP SS does not support initiating a conference using flash hook. In addition, the Avaya Communication Manager Conference On Answer feature is not supported.

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 MultiVOIP SS are registered with the Avaya SES.
- Verify that calls can be placed to/from the analog and SIP endpoints at the branch and the main site.
- Verify that calls can be placed to/from the analog and SIP endpoints at the branch and the PSTN.
- Verify that calls can be placed from the analog and SIP endpoints at the branch when a simulated data WAN failure is introduced.

8. Support

For technical support on the MultiVOIP SS, contact Multi-Tech via the support link at www.multitech.com.

9. Conclusion

These Application Notes describe the procedures required to configure the Multi-Tech MultiVOIP SS SIP Gateway and Server to interoperate with Avaya SIP Enablement Services and Avaya Communication Manager. The MultiVOIP SS 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 6.0, January 2008.

[2] *Administrator Guide for Avaya Communication Manager*, Doc # 03-300509, Issue 4, January 2008.

[3] *SIP Support in Avaya Communication Manager Running on Avaya S8xxx Servers*, Doc # 555-245-206, Issue 8, January 2008.

[4] *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services*, Doc # 03-600768, Issue 5, January 2008.

[5] *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

[6] *MultiVOIP User Guide for MVP210/410/810 Family*, PN: S0000383 Rev.C, February 18, 2007.

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

Product documentation for the MultiVOIP SS may be found at http://www.multitech.com.

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