Avaya Solution & Interoperability Test Lab

Configuring SIP IP Telephony Using Avaya Aura™ SIP Enablement Services, Avaya Aura™ Communication Manager, and Cisco ATA 186 Analog Telephone Adapters - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to connect Cisco ATA 186 Analog Telephone Adapters to a SIP infrastructure consisting of an Avaya Aura ™ SIP Enablement Services (SES) and Avaya Aura ™ Communication Manager. Cisco ATA 186 is directly registered to Avaya SIP Enablement Services (SES) as SIP endpoint with Avaya Analog phones connected to Cisco ATA 186. For registration Cisco ATA 186 uses UDP.

1. Introduction

With the introduction of the SIP protocol standard that supports telephony as well as a wide range of other communication modes, there is a much broader range of SIP telephones and gateways available to customers. There will be sales opportunities involving customers who wish to purchase the Avaya SIP offer, but already own analog telephones and analog gateways other than those offered by Avaya. Customers may be interested in replacing their existing telephony infrastructure (e.g., Cisco Unified Communications Manager) with Avaya servers, but wish to re-use the existing non-Avaya gateways. In addition, the Off-PBX Station (OPTIM) feature set can be extended from Avaya Aura™ Communication Manager to analog telephones connected to these gateways, providing enhanced calling features in advance of SIP protocol definitions and implementation by gateway manufacturers. These Application Notes describe the configuration steps for using Cisco ATA 186 Analog Telephone Adapters with Avaya Aura™ SIP Enablement Services and Avaya Aura™ Communication Manager. Only those configuration steps pertinent to interoperability of Cisco and Avaya equipment are covered. General administration information can be found in the product documentation as well as the specific references listed in References section.

1.1. Configuration

The configuration used as an example in these Application Notes is shown in **Figure 1**. The diagram indicates logical signaling connections. With the exception of the Avaya analog telephones and fax machines, all components are physically connected to an Avaya G650 Media gateway and are administered as a single subnet. Each Cisco ATA is configured to register to SIP Enablement Services. The ATA supports two analog ports, representing each port as a separate SIP telephone. An analog telephone and a fax machine are connected to the ports of each ATA. Each ATA port is administered as an OPTIM station on Avaya Aura™ for Midsize Enterprises running on an Avaya S8800 Server. The Avaya Modular Messaging application resides on the S8730 Media Server and is used to support voice messaging. The PC supports a TFTP server as well as a web browser for administration of the network components.

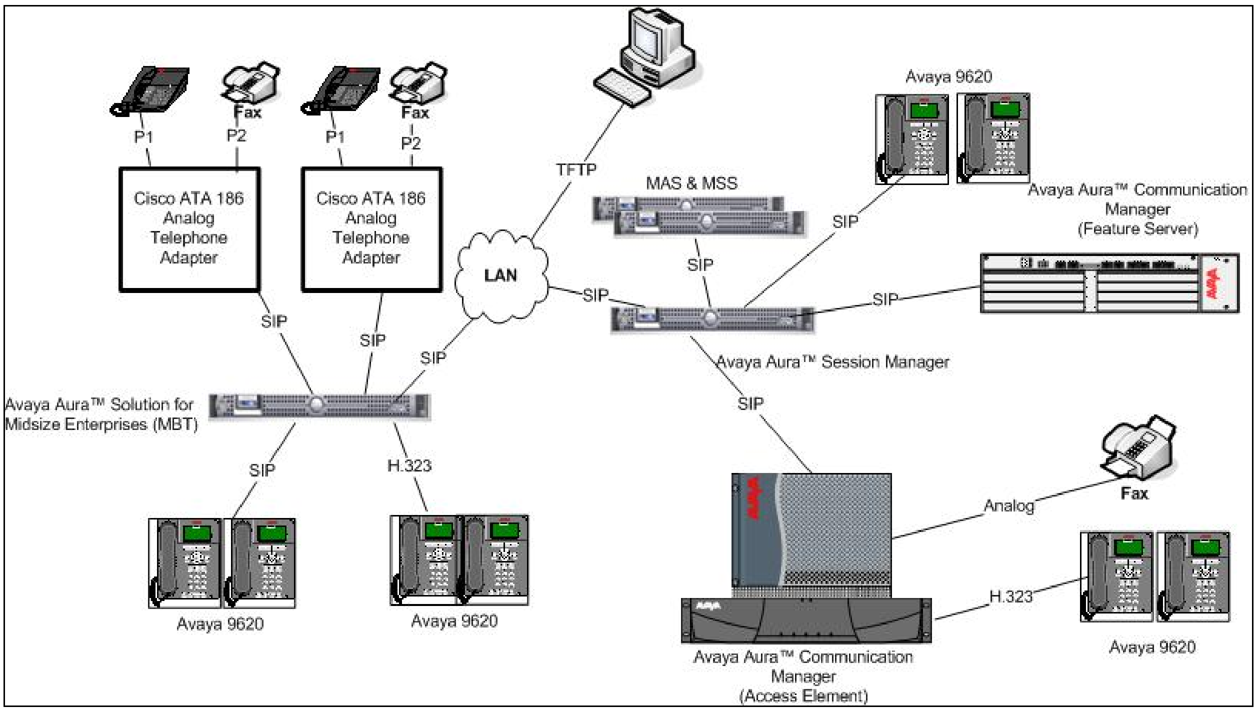


Figure 1: Avaya SIP Test Configuration with Cisco ATA 186 Analog Telephone Adapters

2. Equipment and Software Validated

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

Network Component	Software Version
Avaya 9600-Series Telephones	3.002 (H.323)
Avaya 9600-Series Telephones	2.5.0 (SIP)
Avaya Aura™ Solution for Midsize Enterprises S8800 server	5.2.1.2.5
Avaya S8720 Server	Avaya Aura™ Communication Manager 5.2.1 (S8720-015-02.1.016.4 with update 17774)
Avaya G650 Media Gateway IPSI TN2312BP MedPro TN2602AP CLAN TN799DP ANALOG TN793CP	 HW15 FW049 HW08 FW049 HW01 FW034 HW09 FW008
Avaya Modular Messaging on Avaya S8730 Messaging Servers (MAS and MSS)	Avaya Modular Messaging 5.2 (9.2.150)
Avaya Aura™ Communication Manager Feature Server	Avaya Aura™ Communication Manager 5.2.1 (S8720-015-02.1.016.4 with update 17774)
Avaya Aura™ System Manager Server S8510	Avaya Aura™ System Manager 5.2.0.1- SP0
Avaya Aura™ Session Manager Server S8510	Avaya Aura™ Session Manager 5.2.0.1- SP0
Cisco ATA 186 Analog Telephone Adapter	3.2.1 (SIP)

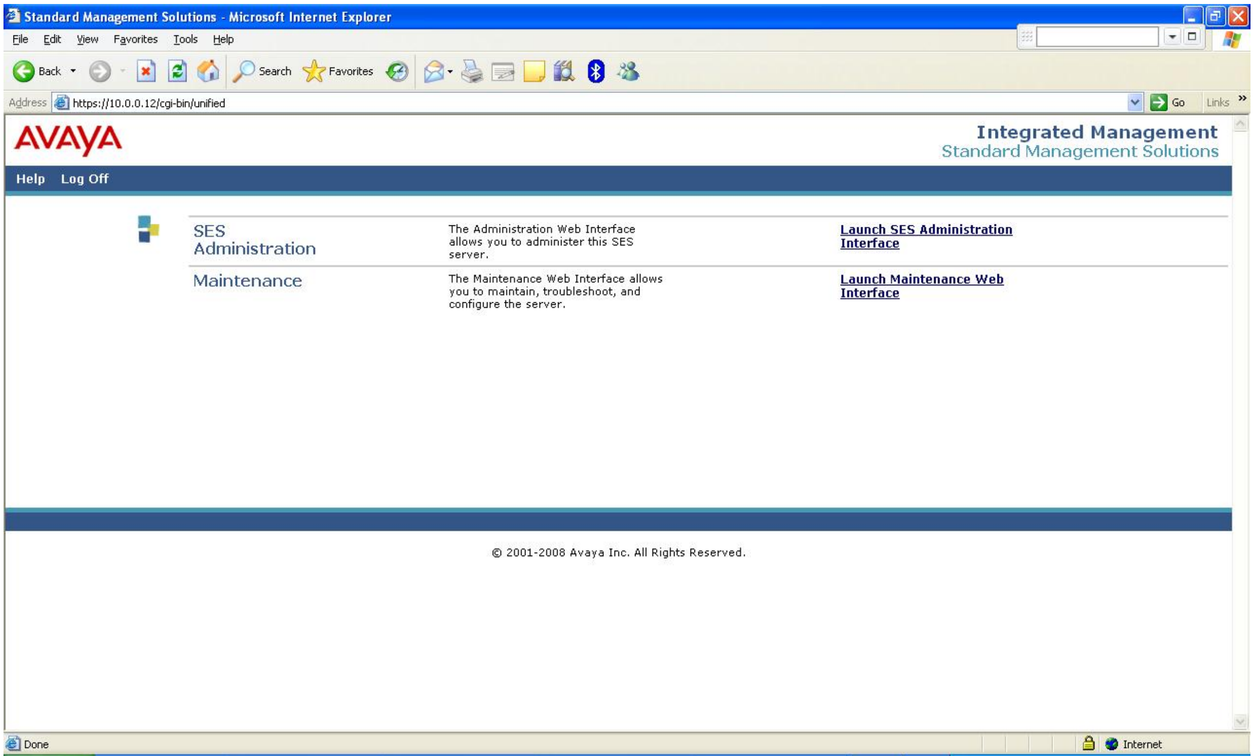
3. Administer Avaya Aura™ SIP Enablement Services

The following steps describe configuration of SIP Enablement Services for use with the Cisco 186 ATA phone and fax connected to ATA. SES is configured via an Internet browser using the administrator web interface. It is assumed that SES software and the license file have already been installed on the server. Access the SES administration web interface by entering **http://<SES-ip-addr>/admin** as the URL in an Internet browser.

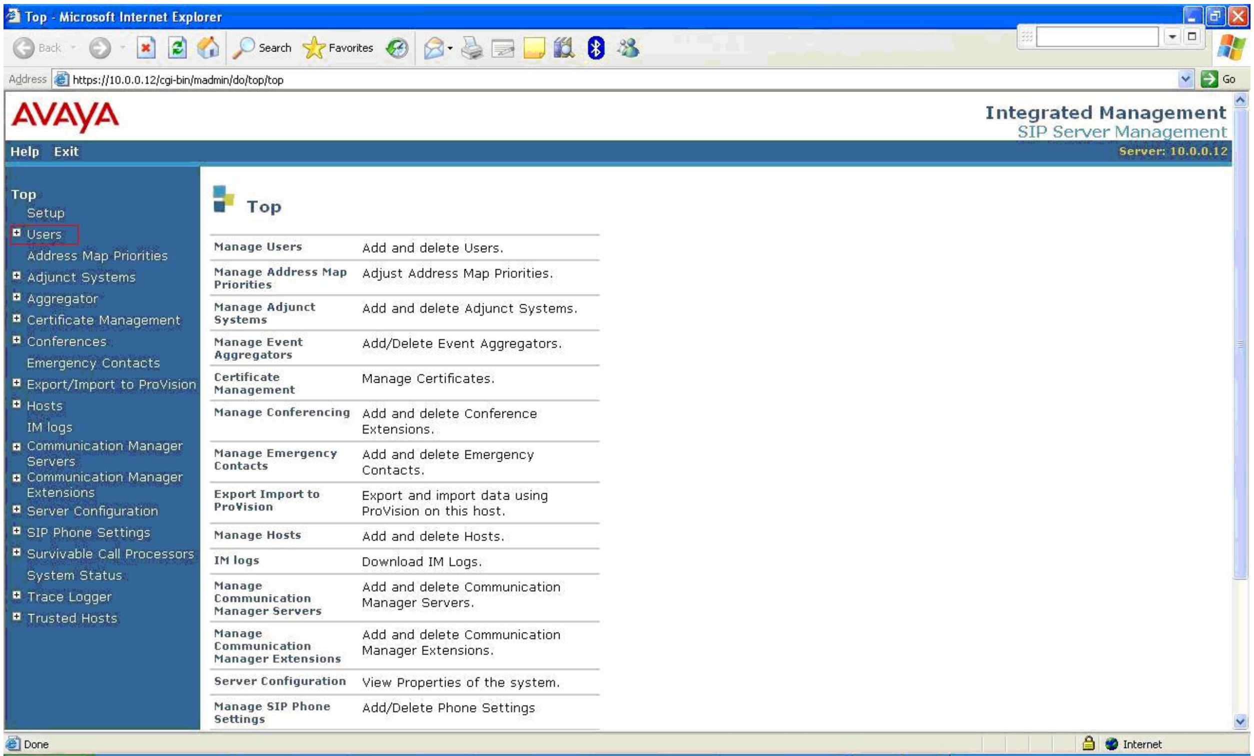
- Administer SIP OPTIM Users

3.1. Administer SIP OPTIM Users

Log in with appropriate credentials and then select the **Launch SES Administration Interface** link from the main page.

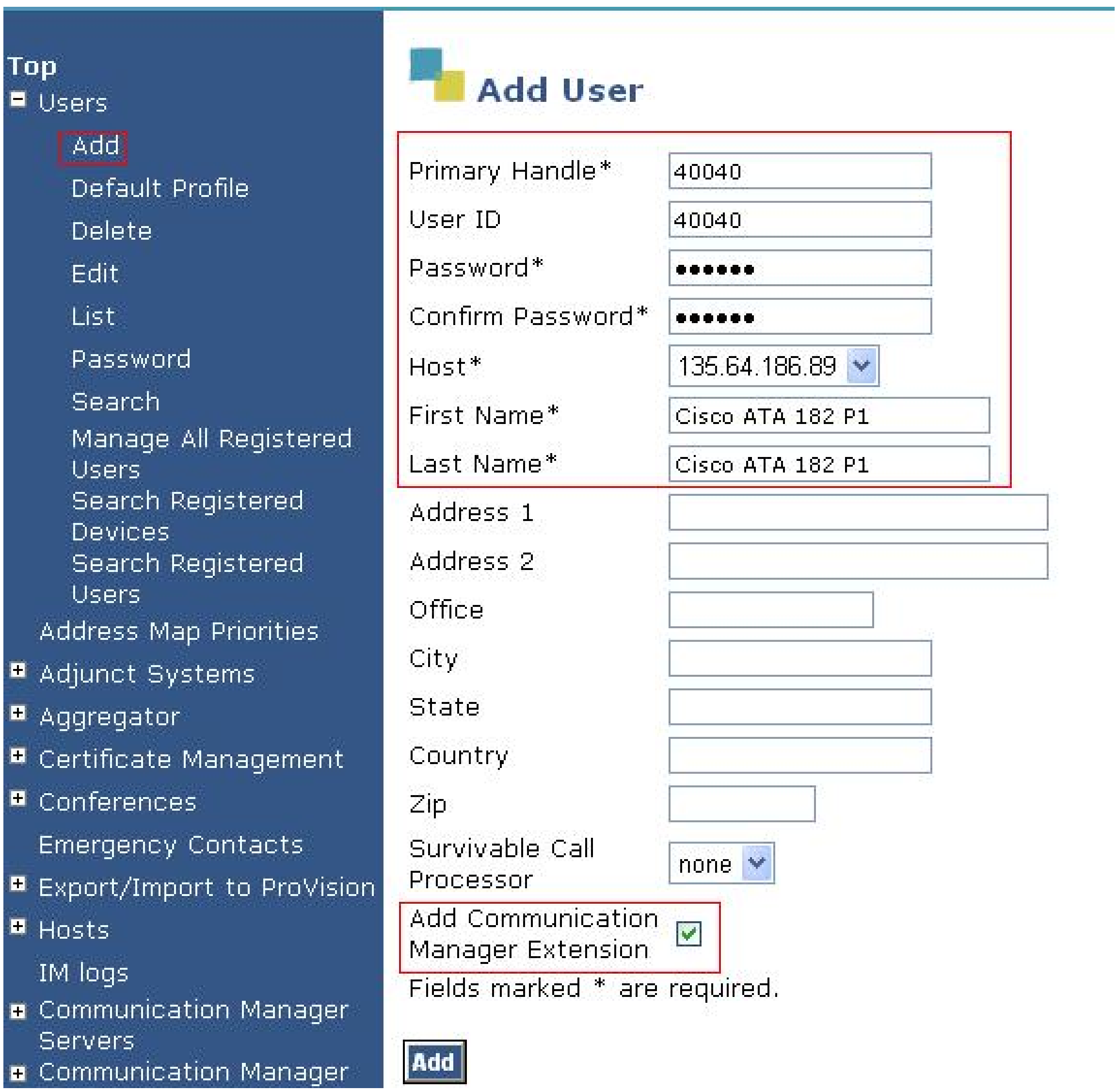


From the main page, click on **Users** as shown in the sample configuration below.



On the left panel expand **Users.** From **Users** click **Add**. Enter the required details as shown in the sample configuration below and click **Add**. Repeat the same steps for other SIP OPTIM users.

- **Primary Handle:** **40040** User Extension
- **User ID:** **40040** User Extension
- **Password:** Extension Password
- **Host:** **135.64.186.89**, i.e. SES IP Address
- **Add Communication Manager Extension:**
  Tick the check box. This will prompt administrator to create extension as shown in the next page.



Click **Continue** (not shown). The **Add Communication Manager Extension** screen will be displayed. Enter the required details as shown in the sample configuration. Click **Add** and **Continue.**

4. Configure Avaya Aura™ for Midsize Enterprises Communication Manager

This section highlights administering the OPTIM features on Communication Manager for Cisco ATA 186.

4.1. Verify OPTIM Capacity

Use the **display system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones – OPTIM** has been set to the value that has been licensed, and that this value will accommodate addition of the ATA-supported analog ports. Note that there are two ports on each Cisco ATA 186.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V15                                Software Package: Standard
       Location: 2                               RFA System ID (SID): 1
       Platform: 25                              RFA Module ID (MID): 1

                                                                  USED
                                Platform Maximum Ports: 44000 107
                                      Maximum Stations: 2400  15
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 2400  2
                    Maximum Off-PBX Telephones - OPTIM: 2400  10
                    Maximum Off-PBX Telephones - PBFMC: 2400  2
```

4.2. Administer the Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions.

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
       0          1    fac
       1          5    ext
       2          5    aar
       209        5    ext
       3          5    aar
       4          5    ext
       5          5    ext
       6          5    ext
       7          5    aar
       8          5    aar
       9          1    fac
       *          3    dac
       #          3    dac
```

4.3. Administer Coverage Path

Configure the coverage path to be used for the voice messaging hunt group, which is group **h1** in the sample configuration. The default values shown for **Busy?**, **Don't Answer?**, and **DND/SAC/Goto Cover?** can be used for the **Coverage Criteria**. In this case, the **Number of Rings** before the call goes to voice messaging has been extended from the default of 2 to 4 rings.

```
change coverage path 1                                          Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 1
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?             n              n
               Busy?             y              y
       Don't Answer?             y              y         Number of Rings: 4
                All?             n              n
 DND/SAC/Goto Cover?             y              y
   Holiday Coverage?             n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h1             Rng: 2  Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

4.4. Administer Stations

Use the **add station** command to add a station for each ATA port to be supported. Assign the same extension as the Communication Manager extension administered in SIP Enablement Services. Use 9620 for the Station, and be sure to include the **Coverage Path** for voice messaging if applicable. The **Name** field is optional and is shown on the display of Avaya non-SIP telephones when receiving calls from this station. Use defaults for the other fields on **Page 1.**

```
add station 40040                                          Page   1 of   5
                                     STATION

Extension: 40040                         Lock Messages? n               BCC: 0
     Type: 9620                          Security Code:                  TN: 1
     Port: S00017                      Coverage Path 1: 1               COR: 1
     Name: Cisco ATA1                  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: 40040
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                         IP SoftPhone? n

                                                        IP Video? n


                                             Customizable Labels? y
```

On **Page 2**, if this telephone will have a bridged appearance for another telephone (see **Page 3** for this station), then **Bridged Call Alerting** should be set to **y**, so that this telephone will ring when the other telephone is called. Note that no other operational behaviors of the bridged appearance feature apply to SIP telephones (e.g., off-hook indication, bridge-on, etc.). By default, the last call appearance is reserved for outgoing calls from the telephone. If it is desirable to allow an incoming call to use the last available call appearance when all others are occupied, set the **Restrict Last Appearance** field to **n**. In this mode, all call appearances are available for making or receiving calls. Enter the **"sip-adjunct"** administered for this system in **MWI Served User Type**.

```
change station 40040                                           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? y
       Service Link Mode: as-needed                        EC500 State: enabled
         Multimedia Mode: enhanced                 Audible Message Waiting? n
    MWI Served User Type: sip-adjunct           Display Client Redirection? n
                                               Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
                                                   Multimedia Early Answer? n
                                               Direct IP-IP Audio Connections? y
  Emergency Location Ext: 40040         Always Use? n IP Audio Hairpinning? n
```

On **Page 4,** under the heading **BUTTON ASSIGNMENTS**, fill in the number of call appearances (**call-appr** buttons) that are to be supported for the telephone. For the ATA 186 the recommended number is 2. Configure brdg-appr on this extension for 40001. In the sample configuration, 40001 is another SIP endpoint.

```
change station 40040                                            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                          4:
 2: call-appr                          5:
 3: brdg-appr  B:1  E:40001            6:


    voice-mail Number:
```

4.5. Administer the OPTIM

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extension (**40040**) to the same SIP Enablement Services extension. Enter the field values shown. For the sample configuration, the **Trunk Selection** value indicates the SIP trunk group. The **Configuration Set** value can reference a set that has the default settings in Communication Manager.

```
change off-pbx-telephone station-mapping 40040                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 40040            OPTIM            -    40040            3           1
```

On **Page 2**, change the **Call Limit** to match the number of **call-appr** entries in the **add station** form. Also make sure that **Mapping Mode** is set to **both** (the default value for a newly added station).

```
change off-pbx-telephone station-mapping 40040             Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 40040         OPTIM      2        both         all         both
```

5. Configure the Cisco ATA 186 Analog Telephone Adapter

5.1. Registration and Basic Calling

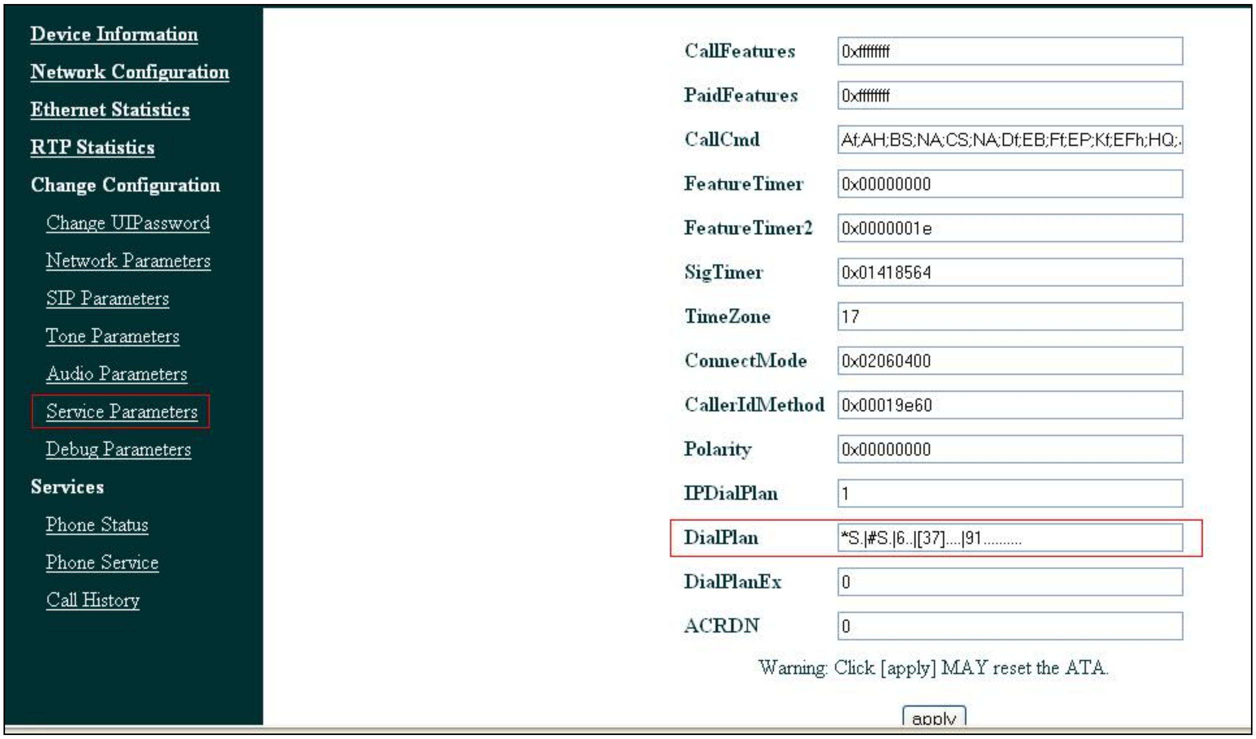
Cisco ATAs can be configured using three methods:

1. Configuration file downloaded from a TFTP server specified via DHCP at boot time. Two such files can be installed on the TFTP server. At boot time, the ATA will attempt to load a device-specific file of the form `ata`<*MAC-address*>`.cfg`, where <*MACaddress*> is the MAC address of the ATA. If this file is not present, the ATA will attempt to load `atadefault.cfg`, a default configuration file containing parameter settings that apply to all ATAs. Note that if the device-specific file is accessible, the ATA will not attempt to load the default file.
2. Web interface – for configuring a small number of adapters, or to inspect the current parameter settings on an adapter. Access http://*IP_address* using a web browser.
3. Manual configuration of the ATA telephone using voice configuration menu, accessible using the dial pad on an analog telephone connected to port 1. This is typically used to set the IP network parameters and to reset the gateway to its factory default parameter values.

Parameters that are manually changed using the web or voice configuration interfaces will revert back to the values in the configuration file when the telephone is re-booted, unless the DHCP and TFTP parameters have been manually changed. For the sample configuration, the IP address of the telephone and its TFTP server were manually entered at the telephone using the voice configuration menu. The remaining configuration was done via the configuration file. Excerpts of the configuration file are shown, including explanatory comments supplied with the product sample file.

1. Edit the default or device-specific configuration file, or use the web interface to configure ATA parameters. ATA software updates include a sample file `sip_example.txt`, annotated with explanations for each parameter (not included in these Application Notes). The screen below shows the relationship between the parameters that must be configured for the ATA and those administered in SIP Enablement Services. Sample values are shown for extension 40040 on the ATA 186. Parameter names in the configuration file are the same as in the web interface. Parameter names ending in 0 or 1 refer to telephone ports 1 and 2, respectively.

The following screen summarizes other important network and SIP related parameters that can be configured in the ATA, and the values used in the sample configuration. The ATA uses the **DialPlan** parameter to determine when enough digits have been pressed to complete dialing, so that the user need not press a **send** key such as * or # to launch the call. Note that **DialPlan** has been specified to support * and #, since they are used in feature access codes.



2. Continue editing the configuration file, specifying the audio codec and audio modes to be used for voice and fax. The table below summarizes the combination of values used in the sample configuration. The low bit rate codec parameter must be assigned the compressed codec to be used after the preferred codec (G.711 µ-law) in the preference list used by the ATA. The audio mode flags bit vector (**AudioMode**) was set to enable adding the compressed codec, silence suppression, fax CallED (CED) tone detection, and support for out of band DMTF signaling via RFC 2833 for both ATA ports. See **Appendix A** for the complete configuration file used for the ATA 186 in the sample configuration.
3. **Audio Parameter Definition Parameter Name Example Value**
   - Audio codec preference (receive) RxCodec 2 (G.711 µ-law)
   - Audio codec preference (transmit) TxCodec 2 (G.711 µ-law)
   - Low bit rate codec LBRCodec 3 (G.729A)
   - Audio mode flags (ports 1 and 2) AudioMode 0x00150015
4. The completed configuration file must be compiled into a form downloadable by the ATA via TFTP. Use the Cisco-supplied *cfgfmt* command line tool in a DOS window of the PC from which the configuration file will be downloaded. The command should be of the form: **cfgfmt -sip edited_configuration_file compiled_configuration_file** Move the file *compiled_configuration_file* to the default TFTP directory. As mentioned earlier

in this section, the name of this file should be either `ata`<*MAC-address*>`.cfg` (recommended) or `atadefault.cfg.`

5. Use the voice configuration menu to assign the initial network parameters. The voice configuration menu is accessed by connecting an analog telephone to the RJ-11 jack labeled **Phone1** on the back of the ATA. Lift the receiver and press the **function button** on top of the ATA as shown below.

A voice prompt will be played, requesting a command. Commands are entered using telephone key pad sequences, terminated by the # key. The basic parameters shown in the table below should be set using this method.

IVR Menu Number	Feature
1	IP Address (StaticIP)
2	Default Gateway (StaticRoute)
10	Subnet Mask (StaticNetMask)
20 (0 = Disable, 1 = Enable)	DHCP (Dhcp)
80	Check IP Address

After assigning the above parameters, press # to exit the voice configuration menu. The function button light should flicker, indicating that the parameters are being loaded and the ATA is resetting. The configuration file created in the previous step will be downloaded and the ATA will register with SIP Enablement Services. Successful registration will be indicated by the reception of dial tone when lifting the receiver on the analog telephone.

6. Local Calling Features on Cisco ATA 186

The following sections describe how to administer local telephone features supported by the ATA that are compatible with the Avaya SIP offer. These features are controlled by parameter values in the configuration file for the ATA. As described in **Section 5.1**, the procedure is to edit the configuration text file with the desired parameter values, compile the file, and reset the ATA so that the compiled configuration file will be downloaded. These Application Notes address only the parameter values that support the particular feature operations described.

6.1. Call Waiting

Call waiting allows a second call to be answered while a call is in progress. Switch hook flash can be used to switch between active and held calls. The user can also disconnect the active call by hanging up, after which the telephone will ring as a reminder that a held call still exists. Answering the telephone will resume the held call. To configure these call waiting capabilities, make sure that the following string sequence is included in the **CallCmd** parameter in Cisco ATA 186 configuration file.

Kf;EFh;HF

6.2. Fax

The ATA supports pass-through fax mode using G.711 μ-law if the **audiomode** and **connectmode** ATA configuration file parameters are set for pass-through. Both parameters are represented as bit strings, where the appropriate bits must be set to support fax pass-through. The bits are numbered 0-31 from right to left. In the sample configuration, **audiomode** was set to 0x00150015, where the low order and high order 16 bits apply to analog port 1 and port 2, respectively. The bits that pertain to fax for port 1 are:

Bit 0 = 1	Enables G.711 silence suppression (VAD).
Bit 2 = 1	Enables Fax CED tone detection and switchover upon detection.
Bit 4 = 1, Bit 5 = 0	DTMF transmission method = out-of-band through negotiation.
Bit 6 = Bit 7 = 0	Disable sending out switch-hook flash.

All 32 bits of the **connectmode** parameter bit string represent settings that are applied to both ports. In the sample configuration, this parameter was set to 0x02060400. The applicable bits for fax pass-through are bit 2 and bits 7-15:

Bit 2 = 0	Uses RTP payload number 126/127 for fax up-speed to G.711μ−law/ G.711A-law.
Bit 7 = 0	Disables fax pass-through redundancy.
Bits {12, 11, 10, 9, 8} = {0, 0, 1, 0, 0}	Sets the offset to NSE payload-type number 96 to 4. Setting the offset to 4 results in the Cisco ATA sending a Named Signaling Event (NSE) payload-type value of 100 by default.
Bit 13 = 0	Uses G.711μ−law for fax pass-through up-speed.
Bit 14 = Bit 15 = 0	Enables *fax pass-through mode* using the Cisco proprietary method, if supported at the far end.

7. Verification Steps

Features tested using the sample configuration included call bridging, voice mail with messaging waiting indicator, fax, codecs, and telephony features. The following steps can be used to verify and/or troubleshoot installations in the field.

1 After rebooting the ATA, use its web interface to verify that the parameters set in the compiled configuration file have been loaded. Lift the handset on the analog telephones and verify dial tone, indicating that registration has occurred. If there is no dial tone, check that the **Proxy** IP address is correct, and that **SIPRegOn** is set to **1**. The web interface can be used to change ATA parameters during troubleshooting, but remember that if a configuration file is normally used for administration, any changes must be reflected in the source and compiled files.

2 Verify basic feature set administration by lifting the handset and making calls to other telephones.

3 Call an ATA supported telephone that currently has no voice messages, and leave a message. Verify that stutter dial tone is heard when lifting the handset on the called telephone. Call the voice messaging system from that telephone. Use the voice messaging menus to retrieve and delete the voice message, verifying that DTMF is interpreted correctly by the system. Hang up, and then lift the handset again, verifying that normal dial tone is heard.

4 Connect analog fax machines to ATA ports and other analog ports supported on Avaya Media Gateways, as applicable. Confirm accurate fax transmissions among the ports.

8. Conclusion

The Cisco ATA 186 SIP gateway was successfully tested with the Avaya SIP offering. During testing the following observations were made:

- G.729AB is not supported by Cisco ATA 186
- Message Waiting Indication does not function when using the Cisco ATA 186.
- T.38 standard is not supported by Cisco ATA 186, used relay mode

9. Additional References

[1] *Installing and Administering SIP Enablement Services*, Doc ID 03-602508, available at http://support.avaya.com

[2] *Administrator Guide for Avaya Communication Manager*, Doc ID 03-300509, available at http://support.avaya.com

[3] *Cisco ATA 186 Analog Telephone Adaptor Administrator's Guide for SIP (version 3.0)*, available at http://www.cisco.com

Appendix A

ATA 186 configuration file for the sample configuration.

```
#txt
UIPassword:0
UseTftp:1
TftpURL:135.64.186.244
CfgInterval:3600
EncryptKey:0
upgradecode:0,0x301,0x0400,0x0200,135.64.186.244,69,0x050616a,ata030201SIP050
616a.zup
dhcp:1
DNS1IP:135.64.186.5
DNS2IP:0.0.0.0
NTPIP:0.0.0.0
AltNTPIP:0.0.0.0
VLANSetting:0x0000002b
PortsSetting:0x00000044
L2KeepAlive:0
Proxy:135.64.186.89
AltGk:0
SecProxy:0
AltGkTimeOut:0
LoginID0:0
LoginID1:0
UseLoginID:0
SIPPort:5060
SIPRegInterval:3600
SIPRegOn:1
MaxRedirect:5
SipOutBoundProxy:0
NatServer:0
NatTimer:0x00000000
MsgRetryLimits:0x00000000
SessionTimer:0x00000000
SessionInterval:1800
MinSessionInterval:1800
DisplayName0:ATA186R P1
DisplayName1:ATA186R P2
ACRDN:0
MediaPort:16384
RxCodec:2
TxCodec:2
LBRCodec:3
```

```
AudioMode:0x00150015
NumTxFrames:2
TOS:0x000068B8
PaidFeatures:0xffffffff
CallFeatures:0xffffffff
CallCmd:Af;AH;BS;NA;CS;NA;Df;EB;Ff;EP;Kf;EFh;HQ;Jf;AFh;HQ;I*67;gA*82;fA#90v#;
OI;H#72v#;bA#74v#;cA#75v#;dA#73;eA*67;gA*82;fA*70;iA*69;DA*99;xA;Uh;GQ;Af;AH;
R3;
FeatureTimer:0x00000000
FeatureTimer2:0x0000001e
SigTimer:0x01418564
ConnectMode:0x02060400
OpFlags:0x00000002
TimeZone:17
CallerIdMethod:0x00019e60
Polarity:0
FXSInputLevel:-1
FXSOutputLevel:-4
DialTone:2,31538,30831,1380,1740,1,0,0,1000,0,0
BusyTone:2,30467,28959,1191,1513,0,4000,4000,0,0,0
ReorderTone:2,30467,28959,1191,1513,0,2000,2000,0,0,0,0,0,0,0,0,0
RingBackTone:2,30831,30467,1943,2111,0,16000,32000,0,0,0
CallWaitTone:1,30831,0,5493,0,0,2400,2400,4800,0,0
AlertTone:1,30467,0,5970,0,0,480,480,1920,0,0
SITone:2,30467,28959,1191,1513,0,2000,2000,0,0,0,0,0,0,0,0,0
RingOnOffTime:2,4,25
DialPlan:*S.|#S.|6..|[37]....|91..........
IPDialPlan:1
NPrintf:0
TraceFlags:0x00000000
SyslogIP:0.0.0.0.514
SyslogCtrl:0x00000000
```

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 Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com