



Avaya Solution & Interoperability Test Lab

Configuring SIP IP Telephony Using Avaya SIP Enablement Services, Avaya Communication Manager, and Cisco ATA 186 and 188 Analog Telephone Adapters - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to connect Cisco ATA 186 and 188 Analog Telephone Adapters to a SIP infrastructure consisting of an Avaya SIP Enablement Services (SES) server and an Avaya S8300 Media Server with G700 Media Gateway running Avaya Communication Manager. Also described is how Avaya Communication Manager features can be made available in addition to the standard features supported in the ATA. The configuration steps described are also applicable to other Linux-based Avaya Media Servers and Media Gateways running Avaya Communication Manager.

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 Call Manager) with Avaya servers, but wish to re-use the existing non-Avaya gateways. In addition, the Off-PBX Station (OPS) feature set can be extended from Avaya 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 and 188 Analog Telephone Adapters with Avaya SIP Enablement Services, S8300 Media Server, and G700 Media Gateway. 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 Section 9. The configuration described should be applicable to other Linux-based Avaya Media Servers and Media Gateways running Avaya Communication Manager.

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 6408D Digital Telephone and the analog telephones and fax machines, all components are physically connected to a single Avaya C363T-PWR Converged Stackable Switch and are administered as a single subnet. Each Cisco ATA is configured to register to one of two SIP Enablement Services home servers. 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 OPS station on an Avaya S8300 Media Server with G700 Media Gateway.¹ The Avaya IA770 INTUITY™ AUDIX® Messaging Application resides on the S8300 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.

The main difference between the Cisco ATA 186 and 188 Analog Telephone Adapters is support for a LAN port on the ATA 188, to which a PC can be connected. The configuration steps described in these Application Notes apply to both models. **Table 1** profiles the network management capabilities of the ATA 186 and 188.

¹ The sample configuration uses multiple SIP Enablement Services servers for illustrative purposes. For installations of less than 6000 users, a single server configured as an Edge/Home combination would suffice.

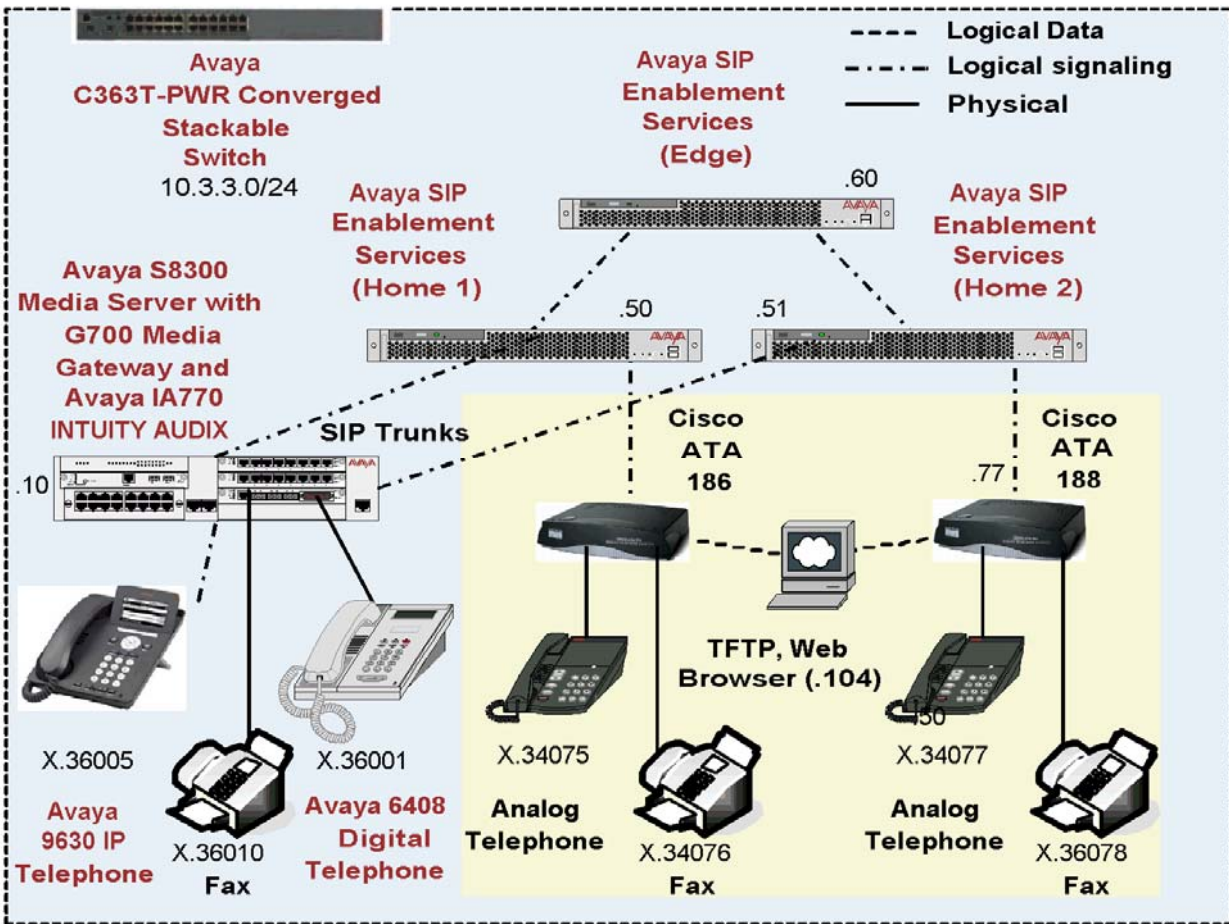


Figure 1: Avaya SIP Test Configuration with Cisco ATA 186/188 Analog Telephone Adapters

Administration mechanisms	Configuration files, DTMF and web interfaces
Administration levels	None
File transfer server	TFTP
Error logs	Via syslog messages (RFC 3164)
802.3af Power over Ethernet Support	No
SNMP support	None

Table 1: Network Management Capabilities of Cisco ATA 186/188 Analog Telephone Adapters

2. Equipment and Software Validated

The following equipment and software were used in the configuration.

Equipment	Software
Avaya SIP Enablement Services (SES) Server	3.1.1
Avaya C363T-PWR Converged Stackable Switch	4.3.10
Avaya S8300 Media Server with G700 Media Gateway	Avaya Communication Manager 3.1.2, Patch 12866
Avaya IA770 INTUITY®AUDIX™	N3.1-26.0
Avaya 6408 Digital Telephone	-
Avaya 9630 IP Telephone	S1.2.1
Cisco ATA 186 Analog Telephone Adapter	3.2.1
Cisco ATA 188 Analog Telephone Adapter	3.2.1
PC (HTTP, TFTP servers)	Microsoft Windows 2000 Professional Workstation, 5.00.2195, SP 4

Table 2: Equipment and Software Versions Used

3. Supported Features

3.1. Overview

Table 3 gives a summary of the features available on the Cisco ATA 186 and 188. Notes on specific feature operations are included in Section 3.2. Some features are supported locally in the adapter, while others are only available with Avaya SIP Enablement Services and Avaya Communication Manager with OPS. In addition to basic calling capabilities, the Internet Engineering Task Force (IETF) has defined a supplementary set of calling features, often referred to as the SIPPING-19 [6]. This provides a useful framework to describe product capabilities and compare features supported by various equipment vendors. OPS can support many of these features even though the ATA 186 and 188 may not locally support them. Additional features beyond the SIPPING-19 can be extended to the ATA using OPS.

Some OPS features shown in **Table 3** can be invoked by dialing a Feature Name Extension (FNE) or Feature Access Code (FAC). Or, a speed dial button on the analog telephone can be programmed to an FNE or FAC. Avaya Communication Manager automatically handles many other standard features via OPS, such as call coverage, trunk selection using Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS), Class Of Service/Class Of Restriction (COS/COR), and voice messaging. Details on operation and administration for OPS can be found in References [2] and [3]. The Avaya SIP solution requires all SIP telephones to be configured in Avaya Communication Manager as OPS.

FEATURE	Supported		COMMENTS
	Locally by Cisco ATA 186/188	With Avaya SIP Offer	
Basic Calling features			
Extension to extension call	YES	YES	
Basic call to legacy phones	NO	YES	
Intercept tones/displays	YES	YES	Reorder for all (announcements available with OPS)
Call Waiting	YES	YES	
Message Waiting Support	YES	YES	Via local stutter dial tone
Fax (Pass-through)	YES	YES	
Fax (T.38)	NO	NO	
SIPPING-19 Features			
Call Hold	YES	YES	
Consultation Hold	YES	YES	
Music on Hold	NO	YES	
Unattended Transfer	YES	YES	
Attended Transfer	YES	YES	
Transfer - Instant Messaging	NO	NO	
Call Forward Unconditional	YES	YES	Local access code or OPS FNE (See Section 3.2.1)
Call Forward Busy	YES	YES	Local access code or OPS FNE (See Section 3.2.1)
Call Forward No Answer	YES	YES	Local access code or OPS FNE (See Section 3.2.1)
3-way conference - 3rd party added	YES	YES	
3-way conference - 3rd party joins	NO	NO	Via OPS FNE
Single Line Extension	NO	NO	
Find-Me	NO	YES	Via bridged appearances
Incoming Call Screening	NO	YES	Via OPS Class Of Restriction (COR)
Outgoing Call Screening	NO	YES	Via OPS Class Of Restriction (COR)
Call Park/Unpark	NO	YES	Via OPS FNE
Call Pickup	NO	YES	Via OPS FNE
Automatic Redial	NO	YES	Via OPS FNE
Click to Dial	NO	NO	
OPS - Selected Additional Station-Side Features			
Conference on Answer	NO	YES	Via OPS FNE
Extended Group Call Pickup	NO	YES	Via OPS FNE
Directed Call Pick-Up	NO	YES	Via OPS FNE
Drop Last Added Party	NO	YES	Via OPS FNE
Last Number Dialed	YES	YES	Local access code or OPS FNE
Malicious Call Trace	NO	YES	Via OPS FNE
Malicious Call Trace Cancel	NO	YES	Via OPS FNE
Priority Call	NO	YES	Via OPS FNE (See Section 3.2.2)
Send All Calls	NO	YES	Via OPS FNE
Send All Calls Cancel	NO	YES	Via OPS FNE
Transfer to Voice Mail	NO	YES	Via OPS FNE
Whisper Page	NO	YES	Via OPS FNE

Table 3: SIP Telephony Feature Support

3.2. Operational Notes

3.2.1. Call Forward

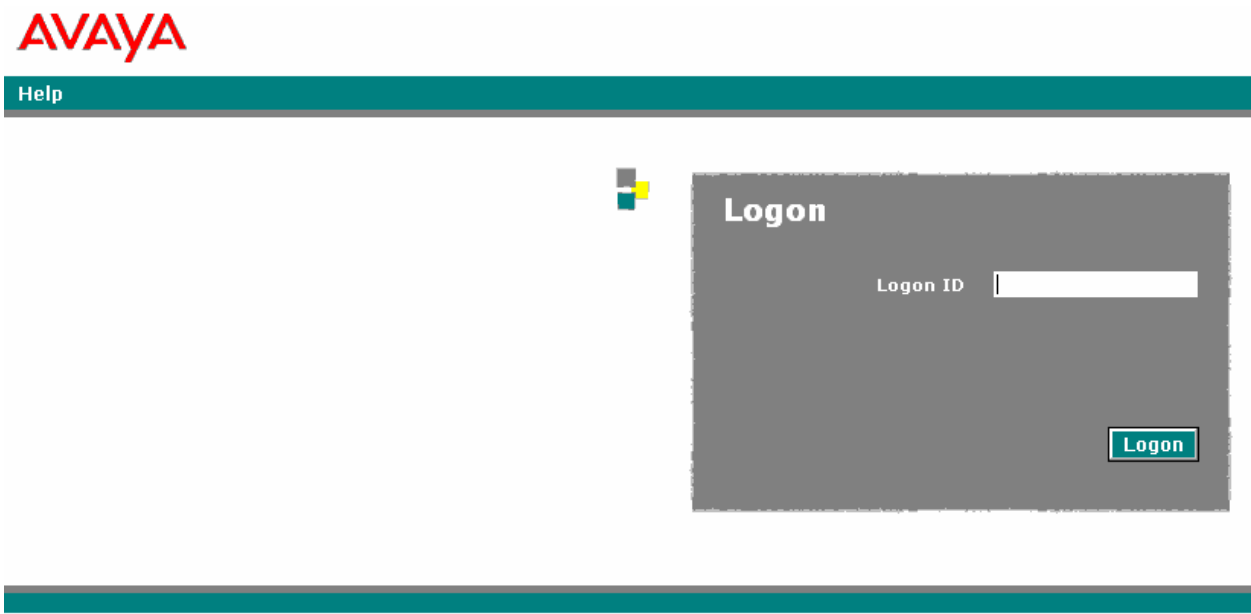
It is recommended that this feature be administered as an Avaya OPS FNE rather than using the call forward feature locally supported by the ATA. The user of local call forward will not benefit from any of the call coverage features available in Avaya Communication Manager with OPS, including coverage to voice messaging. Local call forwarding is covered in these Application Notes for completeness, but the customer should be aware of these limitations.

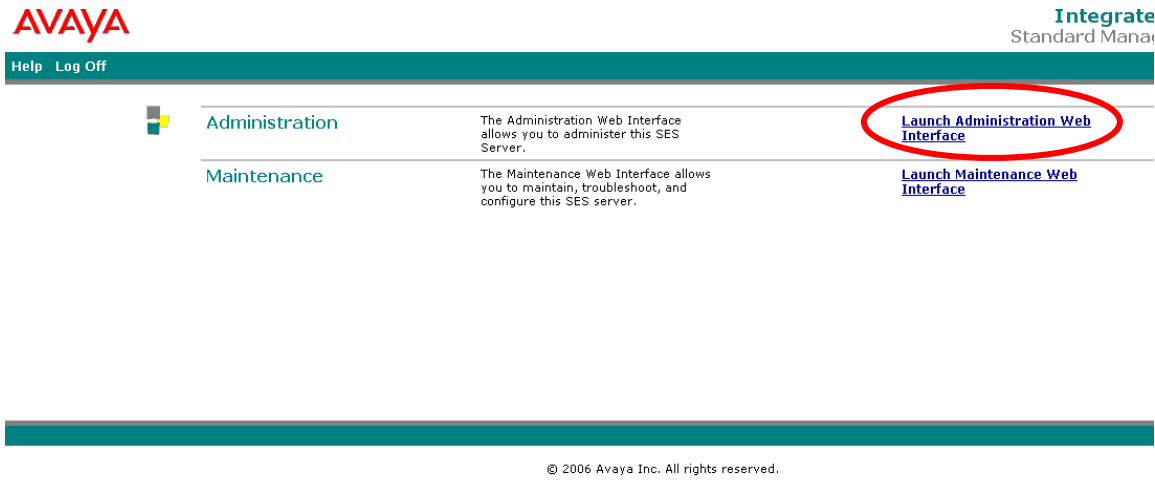
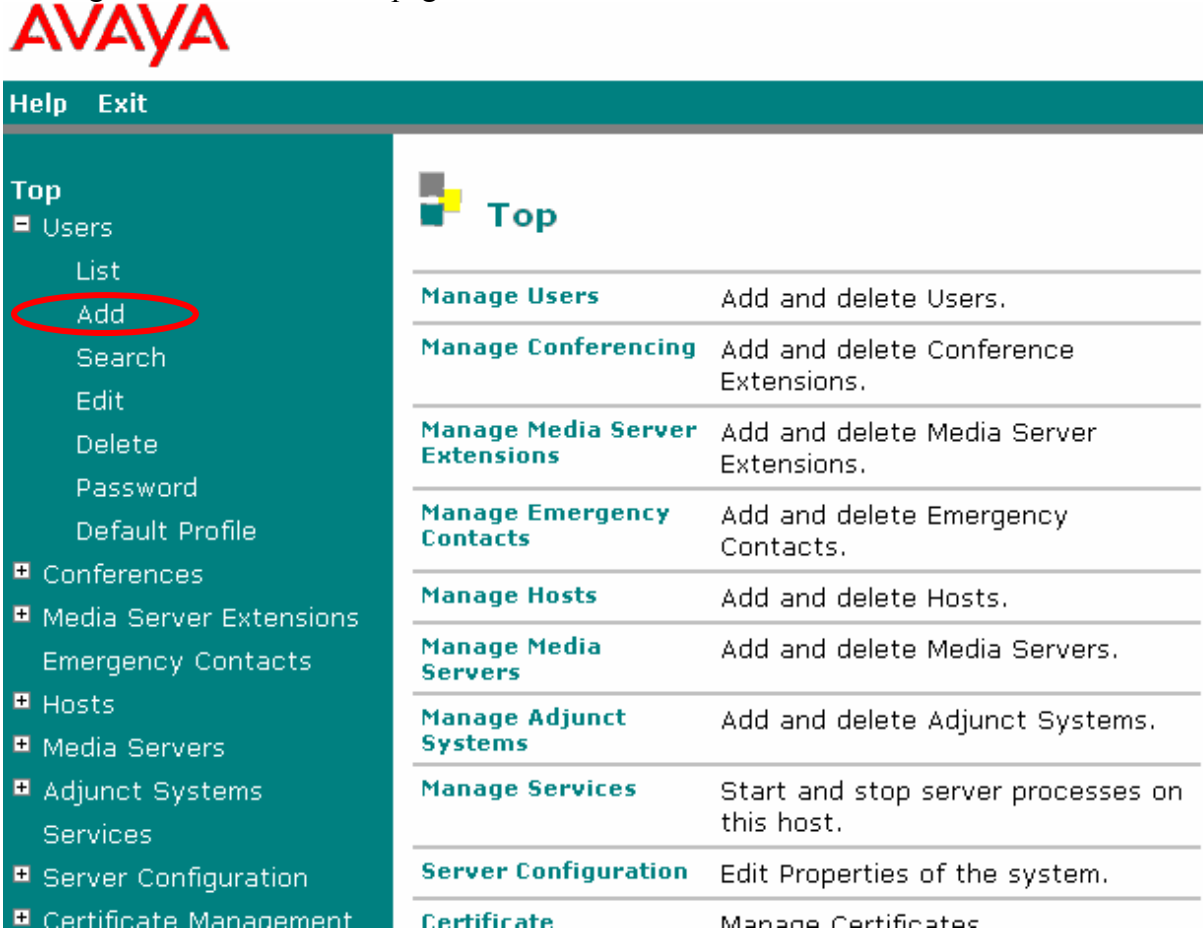
3.2.2. Priority Call

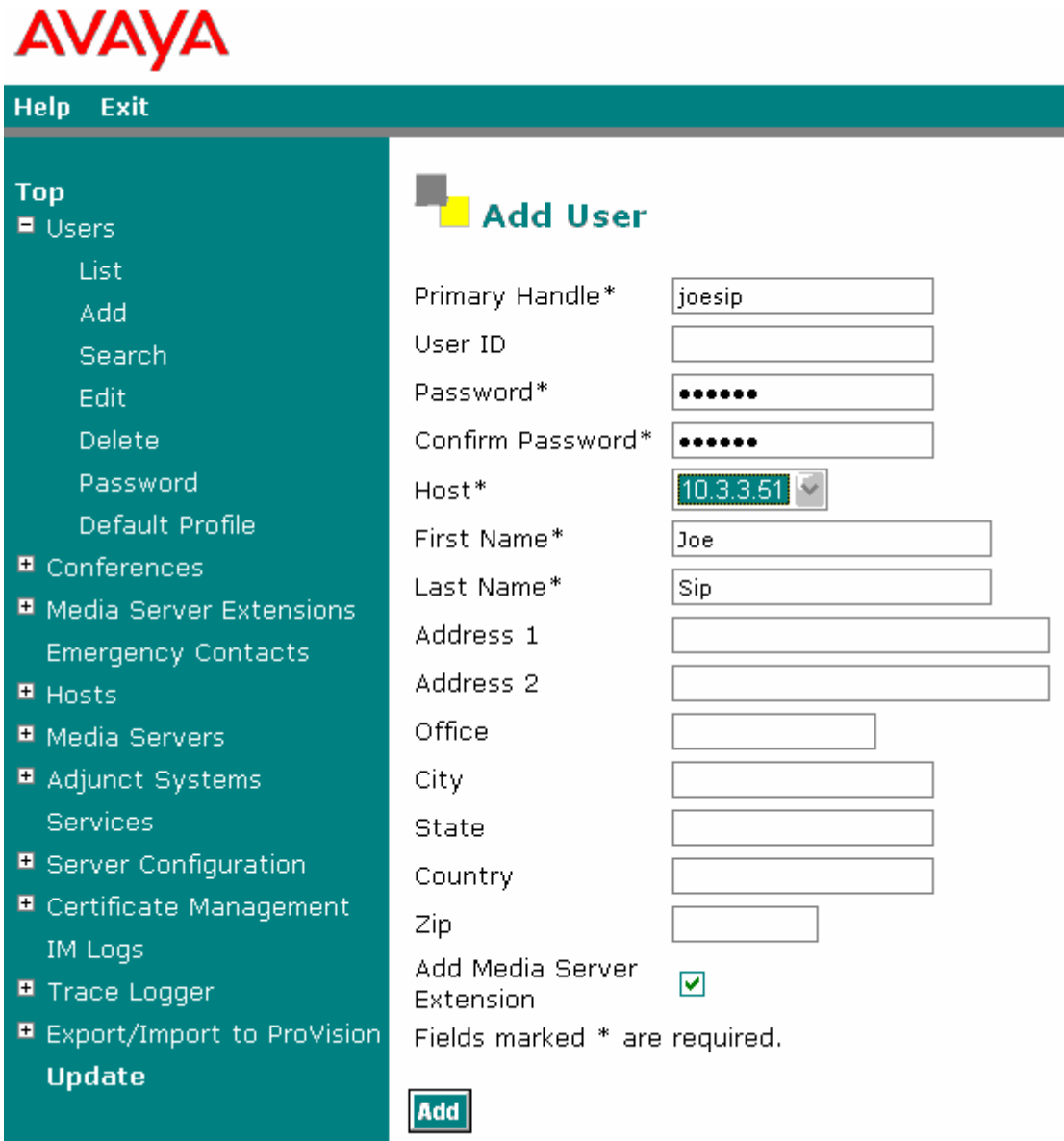
A telephone connected to an ATA may originate priority calls based on the class of service administered for it (see Section 5.6). Note however that since the ATA 186 and 188 do not support distinctive ring indication, telephones connected to them cannot properly indicate a received priority call.

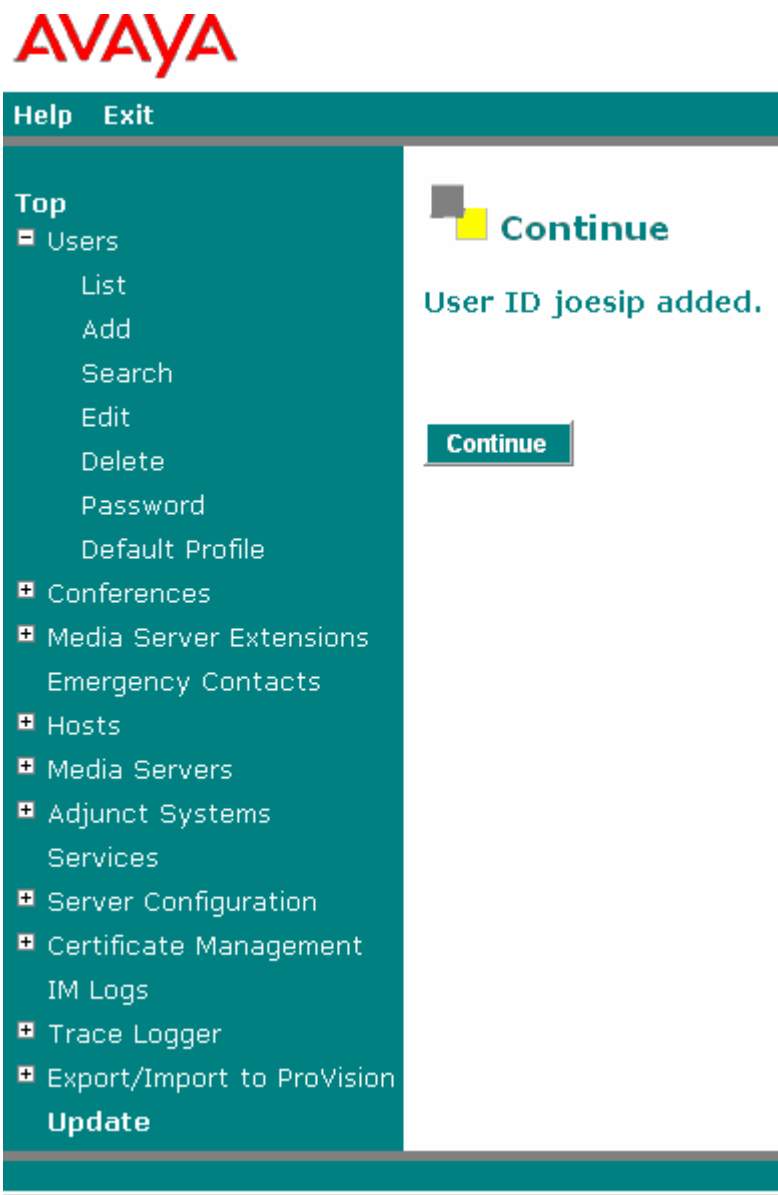
4. Administer SIP Enablement Services

The following steps describe configuration of Avaya SIP Enablement Services for use with the Cisco 186 and 188 ATA. Other standard administration functions are covered in Reference [1].

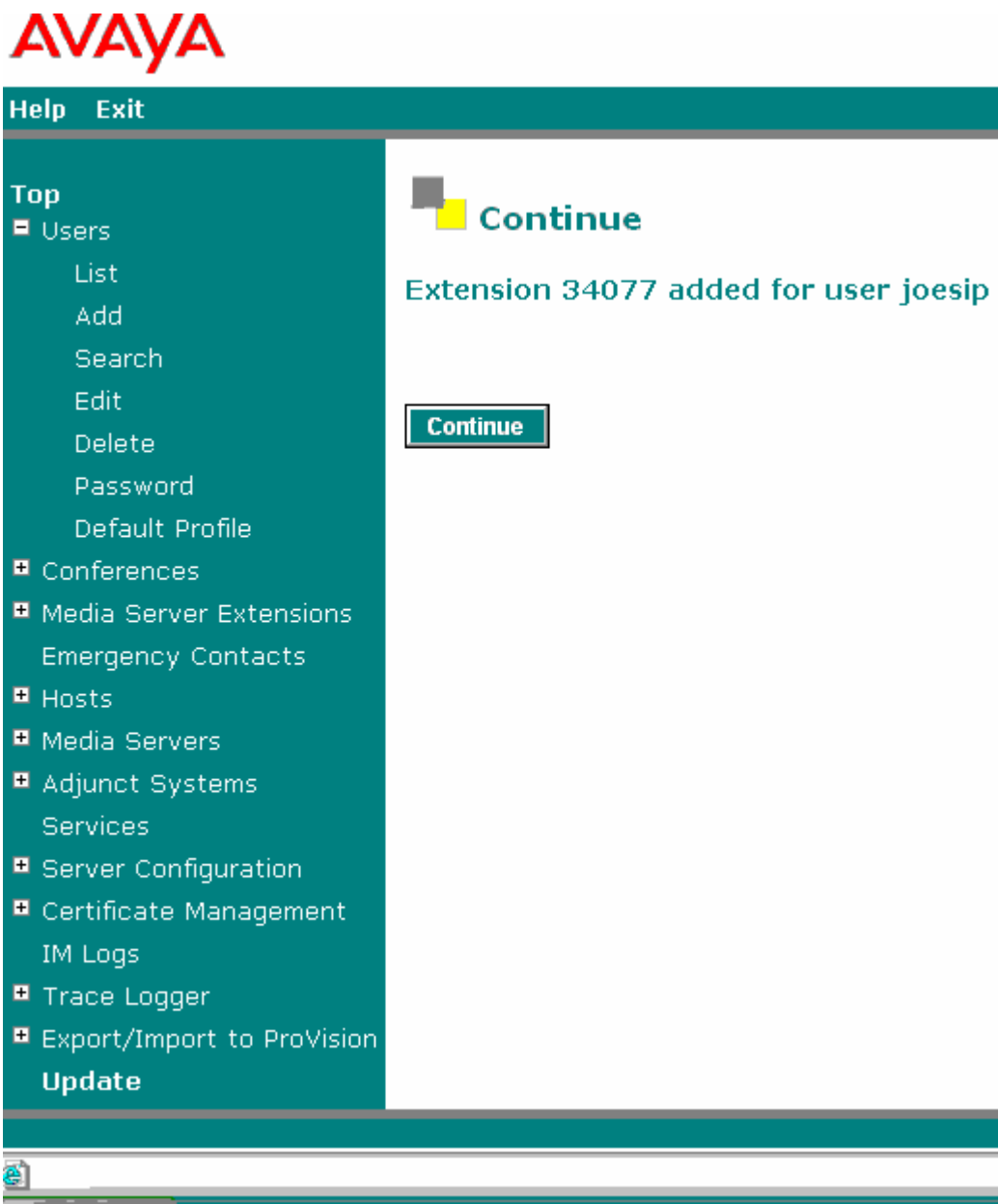
Steps	Description
1.	Avaya SIP Enablement Services is configured using a web browser. Set the URL of the browser to http://IP-address/admin, where <i>IP-address</i> is the IP address of the Avaya SIP Enablement Services Edge or Edge/Home Server. On the <i>Logon</i> screen, enter the appropriate Login ID and click on Logon. When prompted, enter the password. 

Steps	Description
2.	The main administration screen will be displayed after logging in. Click on Launch Administration Web Interface.  © 2006 Avaya Inc. All rights reserved.
3.	The SIP Enablement Services administration web interface will be displayed. Expand the Users heading on the left side of the page and click on Add. 

Steps	Description
4.	The <i>Add User</i> page will be displayed. Fill in the required fields (indicated by *). The screen below illustrates adding the user corresponding to the analog telephone connected to the ATA 188. Enter the user's handle in the Primary Handle field. The Host field should be set to the Avaya SIP Enablement Services Home or Home/Edge server to which the ATA will register. In this configuration, the ATA 188 will register to Home 2. Check the Add Media Server Extension checkbox. Click on Add. 

Steps	Description
5.	The confirmation page will be displayed. Click on Continue. 

Steps	Description
6.	The <i>Add Media Server Extension</i> page will be displayed. Enter the user's telephone extension in the Extension field. Click on Add. Since the user is being added to Home 2, the Media Server corresponding to the SIP trunk between the S8300 Media Server and Home 2 is selected automatically. The screenshot shows the Avaya web interface for adding a media server extension. The sidebar on the left lists various system management options. The main content area is titled 'Add Media Server Extension' and contains a form for adding a new extension for the user 'joesip'. The 'Extension*' field is populated with '34077' and the 'Media Server' dropdown is set to 's8300h2'. An 'Add' button is visible below the form. A note indicates that fields marked with an asterisk are required.

Steps	Description
7.	The confirmation page will be displayed. Click on Continue.  The screenshot shows the Avaya administration web interface. At the top is the Avaya logo. Below it is a teal header bar with 'Help' and 'Exit' links. A teal sidebar on the left contains a menu with categories like 'Top', 'Users', 'Conferences', 'Media Server Extensions', 'Hosts', 'Media Servers', 'Adjunct Systems', 'Services', 'Server Configuration', 'Certificate Management', 'IM Logs', 'Trace Logger', and 'Export/Import to ProVision'. The main content area is white and displays a confirmation message: 'Extension 34077 added for user joesip' in teal text, preceded by a small graphic of two overlapping squares (gray and yellow). Below the message is a teal button labeled 'Continue'.
8.	Repeat Steps 3-7 for each user to be added to the system.
9.	To apply the administration in the above steps, click on Update on the left side of the page. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administration performed to that point.

5. Configure Avaya Communication Manager

This section highlights the important commands for defining an analog telephone connected to an ATA as an Off-PBX Station (OPS) and administering support for the OPS features indicated in **Table 3**. As mentioned in Section 3.1, many other standard Avaya Communication Manager call features are available to these stations. For complete documentation on SIP administration, see References [2-3]. Use the System Access Terminal (SAT) interface to perform the following steps. Log in with the appropriate permissions.

5.1. Verify OPS Capacity

Use the **display system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones – OPS** 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 or 188.

display system-parameters customer-options		Page 1 of 10
OPTIONAL FEATURES		
G3 Version: V13		
Location: 1	RFA System ID (SID): 1	
Platform: 7	RFA Module ID (MID): 1	
		USED
Platform Maximum Ports: 900		116
Maximum Stations: 450		41
Maximum XMOBILE Stations: 0		0
Maximum Off-PBX Telephones - EC500: 0		0
Maximum Off-PBX Telephones - OPS: 50		26
Maximum Off-PBX Telephones - SCCAN: 0		0

5.2. Define System Features

Use the **change system-parameters features** command to administer system wide features for SIP endpoints. Those related to features listed in **Table 3** are shown in bold. These are all standard Avaya Communication Manager features that are also available to OPS stations.

```
change system-parameters features                               Page  4 of 17
      FEATURE-RELATED SYSTEM PARAMETERS
        Reserved Slots for Attendant Priority Queue: 5
          Time before Off-hook Alert: 10
        Emergency Access Redirection Extension:
        Number of Emergency Calls Allowed in Attendant Queue: 5

        Call Pickup on Intercom Calls? y          Call Pickup Alerting? n
        Temporary Bridged Appearance on Call Pickup? y    Directed Call Pickup? y
          Extended Group Call Pickup: simple

        Deluxe Paging and Call Park Timeout to Originator? n
        Controlled Outward Restriction Intercept Treatment: tone
        Controlled Termination Restriction (Do Not Disturb): tone
        Controlled Station to Station Restriction: tone
    AUTHORIZATION CODE PARAMETERS          Authorization Codes Enabled? n
          Controlled Toll Restriction Replaces: none
```

```
change system-parameters features                               Page 17 of 17
      FEATURE-RELATED SYSTEM PARAMETERS

    INTERCEPT TREATMENT PARAMETERS
      Invalid Number Dialed Intercept Treatment: announcement 35010
        Invalid Number Dialed Display:
      Restricted Number Dialed Intercept Treatment: announcement 35011
        Restricted Number Dialed Display:
      Intercept Treatment On Failed Trunk Transfers? n

    WHISPER PAGE
      Whisper Page Tone Given To: paged

    DIGITAL STATION LINE APPEARANCE LED SETTINGS
      Station Putting Call On Hold: green  wink
      Station When Call is Active: steady
      Other Stations When Call Is Put On Hold: green  wink
      Other Stations When Call Is Active: green
        Ringing: green  flash
        Idle: steady
      Display Information With Bridged Call? n
      Pickup On Transfer? y
```

5.3. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan formats used in the system. This includes all telephone extensions, OPS Feature Name Extensions (FNEs), and Feature Access Codes (FACs). To define the FNEs for the OPS features listed in **Table 3**, a Feature Access Code (FAC) must also be specified for the corresponding feature². In the sample configuration, telephone extensions are five digits long and begin with 2, 3, 4, or 5, FNEs are five digits beginning with 7, and the FACs have formats as indicated with **Call Type** “fac”.

change dialplan analysis						Page 1 of 12			
DIAL PLAN ANALYSIS TABLE									
						Percent Full: 0			
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	
0	3	fac							
1	3	dac							
2	5	ext							
3	5	ext							
4	5	ext							
5	5	ext							
6	3	fac							
7	5	ext							
8	1	fac							
9	1	fac							
*	2	fac							
#	2	fac							

5.4. Feature Access Codes (FACs)

Use **change feature-access-codes** to define the access codes for the OPS FNEs, shown in bold.

change feature-access-codes				Page 1 of 7	
FEATURE ACCESS CODE (FAC)					
Abbreviated Dialing List1 Access Code: 601					
Abbreviated Dialing List2 Access Code: 602					
Abbreviated Dialing List3 Access Code: 603					
Abbreviated Dial - Prgm Group List Access Code:					
Announcement Access Code: 606					
Answer Back Access Code: 605					
Attendant Access Code:					
Auto Alternate Routing (AAR) Access Code:					
Auto Route Selection (ARS) - Access Code 1: 9				Access Code 2:	
Automatic Callback Activation: *5				Deactivation: #5	
Call Forwarding Activation Busy/DA: *2 All: 612				Deactivation: #2	
Call Park Access Code: 604					
Call Pickup Access Code: *6					
CAS Remote Hold/Answer Hold-Unhold Access Code: #6					
CDR Account Code Access Code:					
Change COR Access Code:					
Change Coverage Access Code:					
Contact Closure Open Code:				Close Code:	
Contact Closure Pulse Code:					

² Note that if SIP Universal Resource Identifiers (URIs) can be programmed into the telephone, then Feature Name URIs (FNUs) can be used instead, and neither FACs nor FNEs need to be defined for these OPS features. See Reference [2] for more details.

change feature-access-codes	Page 2 of 7
FEATURE ACCESS CODE (FAC)	
Data Origination Access Code:	
Data Privacy Access Code:	
Directed Call Pickup Access Code: 654	
Emergency Access to Attendant Access Code:	
EC500 Self-Administration Access Code:	
Enhanced EC500 Activation: 660	Deactivation: 661
Enterprise Mobility User Activation:	Deactivation:
Extended Call Fwd Activate Busy D/A All:	Deactivation:
Extended Group Call Pickup Access Code: 621	
Facility Test Calls Access Code:	
Flash Access Code:	
Group Control Restrict Activation:	Deactivation:
Hunt Group Busy Activation: *8	Deactivation: #8
ISDN Access Code:	
Last Number Dialed Access Code: *9	
Leave Word Calling Message Retrieval Lock: *1	
Leave Word Calling Message Retrieval Unlock: #1	
Leave Word Calling Send A Message:	
Leave Word Calling Cancel A Message:	
Malicious Call Trace Activation: 613	Deactivation: 614

change feature-access-codes	Page 3 of 7
FEATURE ACCESS CODE (FAC)	
Meet-me Conference Access Code Change:	
PASTE (Display PBX data on Phone) Access Code:	
Personal Station Access (PSA) Associate Code:	Dissociate Code:
Per Call CPN Blocking Code Access Code: 615	
Per Call CPN Unblocking Code Access Code: 616	
Priority Calling Access Code: *7	
Program Access Code: *0	
Refresh Terminal Parameters Access Code: 694	
Remote Send All Calls Activation:	Deactivation:
Self Station Display Activation:	
Send All Calls Activation: *3	Deactivation: #3
Station Firmware Download Access Code:	
Station Lock Activation:	Deactivation:
Station Security Code Change Access Code: 699	
Station User Admin of FBI Assign:	Remove:
Station User Button Ring Control Access Code:	
Terminal Dial-Up Test Access Code: 695	

change feature-access-codes	Page 4 of 7
FEATURE ACCESS CODE (FAC)	
Terminal Translation Initialization Merge Code:	Separation Code:
Transfer to Voice Mail Access Code: #9	
Trunk Answer Any Station Access Code:	
User Control Restrict Activation: 691	Deactivation: 692
Voice Coverage Message Retrieval Access Code:	
Voice Principal Message Retrieval Access Code:	
Whisper Page Activation Access Code: 620	

5.5. Define Feature Name Extensions (FNEs)

The FNEs can be defined using the **change off-pbx-telephone feature-name-extensions** command. This command is used to support both OPS and Extension to Cellular. The fields that have been left blank correspond to those more appropriate for Extension to Cellular.

change off-pbx-telephone feature-name-extensions		Page	1	of	1
EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME					
Active Appearance Select:			Idle Appearance Select:		
Automatic Call Back: 70003			Last Number Dialed: 70019		
Automatic Call-Back Cancel: 70004			Malicious Call Trace: 70029		
Call Forward All: 70005			Malicious Call Trace Cancel: 70021		
Call Forward Busy/No Answer: 70006			Off-Pbx Call Enable:		
Call Forward Cancel: 70007			Off-Pbx Call Disable:		
Call Park: 70008			Priority Call: 70000		
Call Park Answer Back: 70009			Send All Calls: 70001		
Call Pick-Up: 70010			Send All Calls Cancel: 70002		
Conference on Answer: 70011			Transfer On Hang-Up:		
Calling Number Block: 70012			Transfer to Voice Mail: 70023		
Calling Number Unblock: 70013			Whisper Page Activation: 70026		
Directed Call Pick-Up: 70014					
Drop Last Added Party: 70015					
Exclusion (Toggle On/Off):					
Extended Group Call Pickup: 70025					
Held Appearance Select:					

5.6. Specify Class of Service (COS)

Use the **change class-of-service** command to set the appropriate service permissions to support OPS features (shown in bold). For the example, a COS of 1 was used. On Page 2, set the value of **VIP Caller** to “y” only if all calls made by telephones with this COS should be priority calls. Priority call indication (e.g., distinctive ring) is not supported on the Cisco ATA 186 or 188.

change cos										Page										1 of		2
										CLASS OF SERVICE												
	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15						
Auto Callback	n	y	y	n	y	n	y	n	y	n	y	n	y	n	y	n						
Call Fwd-All Calls	n	y	n	y	y	n	n	y	y	n	n	y	y	n	n	n						
Data Privacy	n	n	n	y	n	y	y	y	y	n	n	n	n	y	y	y						
Priority Calling	n	y	n	n	n	n	n	n	n	y	y	y	y	y	y	n						
Console Permissions	y	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n						
Off-hook Alert	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n						
Client Room	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n						
Restrict Call Fwd-Off Net	y	n	y	y	y	y	y	y	y	y	y	y	y	y	y	y						
Call Forwarding Busy/DA	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n						
Personal Station Access (PSA)	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n						
Extended Forwarding All	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n						
Extended Forwarding B/DA	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n						
Trk-to-Trk Transfer Override	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n						
QSIG Call Offer Originations	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n						
Contact Closure Activation	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n						

change cos											Page 2 of 2					
CLASS OF SERVICE																
0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15																
VIP Caller n n y n n n n n n n n n n n n n n																

5.7. Specify Class of Restriction (COR)

Use the **change class-of-restriction** command to enable applicable calling features. To use the Directed Call Pickup feature, the **Can Use Directed Call Pickup** and **Can Be Picked Up By Directed Call Pickup** fields must be set to "y" for the affected stations. In the sample configuration, the ATA 186 and 188 were assigned to COR 2. Note that Page 3 can be used to implement a form of centralized call screening for groups of stations and trunks.

change cor 2	Page 1 of 4	
CLASS OF RESTRICTION		
COR Number: 2		
COR Description: Stations		
FRL: 0	APLT? y	
Can Be Service Observed? n	Calling Party Restriction: none	
Can Be A Service Observer? n	Called Party Restriction: none	
Partitioned Group Number: 1	Forced Entry of Account Codes? n	
Priority Queuing? n	Direct Agent Calling? n	
Restriction Override: none	Facility Access Trunk Test? n	
Restricted Call List? n	Can Change Coverage? n	
Access to MCT? y	Fully Restricted Service? n	
Group II Category For MFC: 7		
Send ANI for MFE? n		
MF ANI Prefix:	Automatic Charge Display? n	
Hear System Music on Hold? y	PASTE (Display PBX Data on Phone)? n	
Can Be Picked Up By Directed Call Pickup? y		
Can Use Directed Call Pickup? y		
Group Controlled Restriction: inactive		

change cor 2	Page 3 of 4														
CLASS OF RESTRICTION															
CALLING PERMISSION (Enter "y" to grant permission to call specified COR)															
0? y	12? y	24? y	36? y	48? y	60? y	72? y	84? y								
1? y	13? y	25? y	37? y	49? y	61? y	73? y	85? y								
2? y	14? y	26? y	38? y	50? y	62? y	74? y	86? y								
3? y	15? y	27? y	39? y	51? y	63? y	75? y	87? y								
4? y	16? y	28? y	40? y	52? y	64? y	76? y	88? y								
5? y	17? y	29? y	41? y	53? y	65? y	77? y	89? y								
6? y	18? y	30? y	42? y	54? y	66? y	78? y	90? y								
7? y	19? y	31? y	43? y	55? y	67? y	79? y	91? y								
8? y	20? y	32? y	44? y	56? y	68? y	80? y	92? y								
9? y	21? y	33? y	45? y	57? y	69? y	81? y	93? y								
10? y	22? y	34? y	46? y	58? y	70? y	82? y	94? y								
11? y	23? y	35? y	47? y	59? y	71? y	83? y	95? y								

5.8. Add 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.

add coverage path 1		Page 1 of 1	
COVERAGE PATH			
Coverage Path Number: 1		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:	Point2:	Point3:
Point4:		Point5:	Point6:

5.9. Add Stations

Use the **add station** command to add a station for each ATA port to be supported. Assign the same extension as the media server extension administered in SIP Enablement Services. Use “4620” for the **Station Type**, “x” for the **Port**, and be sure to include the **Coverage Path** for voice messaging or other hunt group if applicable. Use the **COS** and **COR** values administered in the previous sections. 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 34077		Page 1 of 4	
STATION			
Extension: 34077	Lock Messages? n	BCC: 0	
Type: 4620	Security Code:	TN: 1	
Port: X	Coverage Path 1: 1	COR: 2	
Name: Joe SIP	Coverage Path 2:	COS: 1	
	Hunt-to Station:		
STATION OPTIONS			
Loss Group: 2	Personalized Ringing Pattern: 1		
Data Module? n	Message Lamp Ext: 34077		
Speakerphone: 2-way	Mute Button Enabled? y		
Display Language: english			
	Media Complex Ext:		
	IP SoftPhone? n		

On Page 2, note the following:

- 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 name of the voice messaging system administered for this system in **AUDIX Name**.

add station 34077		Page 2 of 4
STATION		
FEATURE OPTIONS		
LWC Reception: audix	Auto Select Any Idle Appearance? n	
LWC Activation? y	Coverage Msg Retrieval? y	
LWC Log External Calls? n	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Redirect Notification? y	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? y	Restrict Last Appearance? n	
Active Station Ringing: single	Conf/Trans on Primary Appearance? n	
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed		
Multimedia Mode: basic		
MWI Served User Type:	Display Client Redirection? n	
AUDIX Name: audix	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 34077	IP Audio Hairpinning? y	

On Page 3 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 and 188, the recommended number is 2.

add station 34077		Page 3 of 4
STATION		
SITE DATA		
Room:		Headset? n
Jack:		Speaker? n
Cable:		Mounting: d
Floor:		Cord Length: 0
Building:		Set Color:
ABBREVIATED DIALING		
List1: system	List2:	List3:
BUTTON ASSIGNMENTS		
1: call-appr	5:	
2: call-appr	6:	
3: no-hld-cnf	7:	
4: brdg-appr Btn:1 Ext:44072	8:	

Under the same heading, enter the function button names, if required, for OPS FNEs that will be used at the telephone. Only the FNEs shown in **Table 4** require the station to have a corresponding function button. Avaya Communication Manager features that do not require the user to dial an FNE, such as a bridged appearance, may require the appropriate function button, as shown above.

FNE Name	Function Button
Automatic Callback, Automatic Callback Cancel	auto-cback
Conference on Answer	no-hld-cnf

Table 4: Feature Name Extensions Requiring Station Buttons

In the sample configuration, two line appearances were administered for extension 34077. A bridged appearance was defined on station 44072 and the Conference on Answer FNE was included in the local speed dial button programming of the analog telephone.

Use the **change off-pbx-telephone station-mapping** command to map the Avaya Communication Manager extension (34077) to the same SIP Enablement Services media server 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 Avaya Communication Manager.

change off-pbx-telephone station-mapping 34077					Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION					
Station Extension	Application	Dial Prefix	Phone Number	Trunk Selection	Configuration Set
34077	OPS	-	34077	12	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 34077					Page 2 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION					
Station Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	
34077	2	both	all	both	

6. Configure the Cisco ATA 186/188 Analog Telephone Adapter

6.1. 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 `<MAC-address>` 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 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. See [5] for more details on this interface.

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. See Reference [5] for details on installing and maintaining Cisco ATAs using configuration files. 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.

Steps	Description																																										
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 <code>sip_example.txt</code>, annotated with explanations for each parameter (not included in these Application Notes). The table 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 34077 on the ATA 188. 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. <table><tr><th>Avaya SIP Enablement Services</th><th colspan="2">Cisco ATA 186/188</th></tr><tr><th></th><th>Parameter Name</th><th>Example Value</th></tr><tr><td><i>User Administration</i></td><td></td><td></td></tr><tr><td>User ID</td><td>UID0, UID1</td><td>34077</td></tr><tr><td>Password</td><td>PWD0, PWD1</td><td>123456</td></tr><tr><td><i>Proxy Administration</i></td><td></td><td></td></tr><tr><td>SES (proxy/registrar)</td><td>Proxy</td><td>10.3.3.51</td></tr><tr><td>Proxy Port</td><td>SIPPort</td><td>5060</td></tr></table> The following table 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 (see Section 5.4). See Appendix A for an excerpt from <code>sip_example.txt</code> that describes how to specify the DialPlan parameter. <table><tr><th>Parameter Definition</th><th>Parameter Name</th><th>Example Value</th></tr><tr><td>SNTP server address</td><td>NTPIP</td><td>10.3.3.10</td></tr><tr><td>Enable registration with SIP Proxy</td><td>SIPRegOn</td><td>1</td></tr><tr><td>User name displayed in caller ID</td><td>DisplayName0 DisplayName1</td><td>ATA188 P1 ATA188 P2</td></tr><tr><td>Dial Plan specification</td><td>DialPlan</td><td>*S. #S. 6.. [37].... 91.....</td></tr><tr><td>SIP registration expiration interval</td><td>SIPRegInterval</td><td>3600</td></tr></table>	Avaya SIP Enablement Services	Cisco ATA 186/188			Parameter Name	Example Value	<i>User Administration</i>			User ID	UID0, UID1	34077	Password	PWD0, PWD1	123456	<i>Proxy Administration</i>			SES (proxy/registrar)	Proxy	10.3.3.51	Proxy Port	SIPPort	5060	Parameter Definition	Parameter Name	Example Value	SNTP server address	NTPIP	10.3.3.10	Enable registration with SIP Proxy	SIPRegOn	1	User name displayed in caller ID	DisplayName0 DisplayName1	ATA188 P1 ATA188 P2	Dial Plan specification	DialPlan	*S. #S. 6.. [37].... 91.....	SIP registration expiration interval	SIPRegInterval	3600
Avaya SIP Enablement Services	Cisco ATA 186/188																																										
	Parameter Name	Example Value																																									
<i>User Administration</i>																																											
User ID	UID0, UID1	34077																																									
Password	PWD0, PWD1	123456																																									
<i>Proxy Administration</i>																																											
SES (proxy/registrar)	Proxy	10.3.3.51																																									
Proxy Port	SIPPort	5060																																									
Parameter Definition	Parameter Name	Example Value																																									
SNTP server address	NTPIP	10.3.3.10																																									
Enable registration with SIP Proxy	SIPRegOn	1																																									
User name displayed in caller ID	DisplayName0 DisplayName1	ATA188 P1 ATA188 P2																																									
Dial Plan specification	DialPlan	*S. #S. 6.. [37].... 91.....																																									
SIP registration expiration interval	SIPRegInterval	3600																																									

Steps	Description															
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 bitrate 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 B for the complete configuration file used for the ATA 188 in the sample configuration. <table><tr><th>Audio Parameter Definition</th><th>Parameter Name</th><th>Example Value</th></tr><tr><td>Audio codec preference (receive)</td><td>RxCodec</td><td>2 (G.711 μ-law)</td></tr><tr><td>Audio codec preference (transmit)</td><td>TxCodec</td><td>2 (G.711 μ-law)</td></tr><tr><td>Low bitrate codec</td><td>LBRCCodec</td><td>3 (G.729A)</td></tr><tr><td>Audio mode flags (ports 1 and 2)</td><td>AudioMode</td><td>0x00150015</td></tr></table>	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 bitrate codec	LBRCCodec	3 (G.729A)	Audio mode flags (ports 1 and 2)	AudioMode	0x00150015
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 bitrate codec	LBRCCodec	3 (G.729A)														
Audio mode flags (ports 1 and 2)	AudioMode	0x00150015														
3.	The completed configuration file must be compiled into a form downloadable by the ATA via TFTP. Use the Cisco-supplied <i>cfgfmt</i> 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: cfgtfmt -sip edited_configuration_file compiled_configuration_file Move the file <i>compiled_configuration_file</i> 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.															

Steps	Description																					
4.	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 “Phone 1” 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. <table><tr><th>Parameter Name</th><th>Voice Menu Code</th><th>Example Value</th></tr><tr><td>StaticIP</td><td>1</td><td>10.3.3.77</td></tr><tr><td>StaticNetMask</td><td>2</td><td>255.255.255.0</td></tr><tr><td>StaticRoute</td><td>10</td><td>10.3.3.1</td></tr><tr><td>dhcp</td><td>20</td><td>0</td></tr><tr><td>UseTftp</td><td>305</td><td>1</td></tr><tr><td>TftpURL</td><td>905</td><td>10.3.3.104</td></tr></table> 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 Avaya SIP Enablement Services. Successful registration will be indicated by the reception of dial tone when lifting the receiver on the analog telephone.	Parameter Name	Voice Menu Code	Example Value	StaticIP	1	10.3.3.77	StaticNetMask	2	255.255.255.0	StaticRoute	10	10.3.3.1	dhcp	20	0	UseTftp	305	1	TftpURL	905	10.3.3.104
Parameter Name	Voice Menu Code	Example Value																				
StaticIP	1	10.3.3.77																				
StaticNetMask	2	255.255.255.0																				
StaticRoute	10	10.3.3.1																				
dhcp	20	0																				
UseTftp	305	1																				
TftpURL	905	10.3.3.104																				

6.2. Local Calling Features

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 6.1, 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. See Reference [5] for details on variations in feature operation, parameter definitions, and formats.

6.2.1. 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:

Kf;EFh;HF

6.2.2. Call Forward

As described in Section 3.2, Avaya does not recommend using local call forward due to its limitations relative to call coverage and voice messaging features. However, the ATA 186 and 188 have the capability to perform local call forwarding to a specific extension, should it be desirable under certain circumstances. The **CallCmd** parameter in the sample configuration file provides for the following dial codes to access these features. Note that # must be pressed after entering the forward-to extension. Note also that these access codes would be unavailable for assignment to Avaya Communication Manager Features, since they are locally interpreted by the ATA.

Call Forward Feature	Access Code Sequence
Unconditional	#72 + extension + #
Busy	#74 + extension + #
Don't Answer	#75 + extension + #
Cancel	#73

6.3. The Avaya Extended Feature Set

To access any of the OPS features supported by Avaya Communication Manager, dial the corresponding FNE or FAC. For example, if the telephone has been defined in Avaya Communication Manager as part of a pickup group, then dial the Call Pickup FNE (70010) or the FAC (*6) to answer a call to any member of that group. OPS features that involve an existing call (e.g., conference on answer) will require putting that call on hold, and placing a new call using the appropriate FNE. This procedure can be streamlined by using the speed dialing capabilities of the analog telephone to access frequently used FNEs or FACs. In many cases, the * and # keys can be administered as part of an access code to activate and deactivate a feature in

Avaya Communication Manager (see Section 5.4). If so, the **Dialplan** parameter of the ATA configuration file must be modified to override their use as “send” keys. For instance, the following addition to the **Dialplan** parameter of Step 1 in Section 6.1 would allow * and # to be used to activate/deactivate a feature, based on the next digit entered:

*S.|#S.

In addition, ensure that these access codes do not overlap with any access codes locally administered for the ATA using the **CallCmd** parameter (e.g., *69, etc.).³

6.4. 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.711 μ -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.711 μ -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.

³ The typical default access codes included in the **CallCmd** parameter are:

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*77;lA*87

7. Verification Steps

All features shown in **Table 3** and Section 5.5 were tested using the sample configuration. 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. Additional diagnostic procedures are described in the *Troubleshooting* Chapter in [5].
2. Verify basic feature set administration by lifting the handset and making calls to other telephones. Test supported features according to **Table 3** and feature deployment plans at the site.
3. Verify extended OPS features by pressing the speed dial button for the feature, or lifting the handset and dialing the FNE or FAC. If busy or intercept tone is heard, check Avaya Communication Manager for the correct FNE/FAC, proper permissions under COS/COR, and the proper station button assignment to support the feature.
4. 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.
5. 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. Support

For technical support of Cisco products:

Internet: <http://www.cisco.com/en/US/support/index.html>

Email: tac@cisco.com

Telephone: 1-800-553-2447

9. Additional References

- [1] *Installing and Administering SIP Enablement Services R3.1*, Issue 1.4, Doc ID 03-600768, February, 2006, available at <http://support.avaya.com>.
- [2] *SIP Support in Release 3.1 of Communication Manager Running on the S8300, S8400, S8500, S8500B, S8700, and S8710 Media Server*, Issue 1.4, Doc ID 555-245-206, February, 2006, available at <http://support.avaya.com>.
- [3] *Avaya Extension to Cellular and OPS Installation and Administration Guide*, Version 6.0 Issue 9, DocID 210-100-500, June 2005, available at <http://support.avaya.com>.
- [4] *Administrator Guide for Avaya Communication Manager*, Issue 2, Doc ID 03-300509, February 2006, available at <http://support.avaya.com>.
- [5] *Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide for SIP (version 3.0)*, Cisco Systems, Inc., available at <http://www.cisco.com>.
- [6] *Session Initiation Protocol Service Examples - draft-ietf-sipping-service-examples-12*, SIPING Working Group, Internet-Draft, 1/23/2007, available at <http://tools.ietf.org/wg/sipping/draft-ietf-sipping-service-examples/draft-ietf-sipping-service-examples-12.txt>.

Appendix A

*Excerpt of sip_example.txt file describing how to configure the **Dialplan** parameter.*

```
# Parameter: DialPlan
# Access Code: 926
# Type: Alphanumeric string (199 characters maximum)
#
# Description: Dial plan rules.
#
# Note: No syntax check is performed by the actual implementation.
#       It is the responsibility of the provisioner to make sure that
#       the dial_plan is syntatically valid.
#
#       Programmable strings of dial plan that allow one to specify:
#       o special rule -- I{timeout} to control default inter-digit
#         timeout - specifying this rule also has the side effect
#         of preventing non-matching dial string from being sent out.
#       o optional send character to use (e.g. '#' or '*')
#       o how many digits before auto send
#       o send after timeout at any specified number of digits
#         (time out can be changed as digits are entered).
#         in the following:
#         o . means match any digits
#         o - means more digits can be entered, this (if needed) must
#           appear at the end of the individual rule
#           (i.e. e.g. 1408t5- is legal, but 1408t5-3...
#             is illegal).
#       o [] Range, means match any digit in the list. '_' indicate
#         a range of digits. For example, [135] matches the
#         digits 1, 3, and 5, [1_5] matches the digits 1, 2, 3,
#         4 and 5. No # or * is allowed in the range. Range
#         doesn't work with repeat, and range can't include
#         selection. (feature available after v3.0)
#       o (nnn|nnn) Selection, means match any strings in the list.
#         The string can be composed by any digit, #, *, ., -
#         and range. Selection is not used with any prefix or
#         suffix patterns. (feature available after v3.0)
#       o ># means terminating key to send is #, and termination
#         can be applied only after matching hits ># (So >*
#         means terminating char is *, i.e. terminating key
#         must follow >)
#       o rules applied in the order of listed (whichever matched
#         completely first will cause trigger the send).
#       o tn means timeout is n seconds (note: n is 0-9 and
#         a-z -- which ranges 0 to 26).
#       o more than one rules are separated by |.
#       o ^ Logical not, means match any character except the
character
#       immediately following the ^ command. ^ can be used as a
negation
#       before range and selection too.
#       o rn means repeat last pattern (except range) n times
```

```

#           (note:
#           1. ># or tn are modifier, they are not pattern;
#           2. n is 0-9 and a-z -- which ranges 0 to 26).
#           Use the repeat modifier to specify more rules in less
space.
#
#           You can also use the modifier 'S' to sieze the rule matching
#           (i.e. if a rule matches and the modifier 'S' is seen, all
other
#           rules after that matching rule will not be used for
matching).
#
#           Here is the summary of the dial plan rules:
#
#           o In: set the default inter-digit time out.
#           o Hnxxx: specify the hotline/warmline number. (since v2.14)
#           o Pnxxx: specify the prefix. (since v2.14)
#           o Bxxxx: specify the base number. (since v2.16)
#           o Rxxx(nnn|nnn): specify the prefix. (since v3.0)
#           o Cxxxx: specify the call blocking numbers. (since v3.0)
#           o Fxxxx: specify the call forwarding blocking numbers. (since
#                     v3.0)
#           o Xxxxx: specify the call blocking/call forwarding blocking
#                     numbers. (since v3.0)
#           o Dxxxx: displaying Caller ID. (since v3.0)
#
#           More details are available in the eng114487.doc.
#
# Examples 1: The set of dial plan rules:
#
#           ".t7>#.....t4-|911|1t7>#.....t1-|0t4>#.t7-"
#
#           or equivalently
#
#           ".t7>#r6t4-|911|1t7>#.r9t1-|0t4>#.t7-"
#
#           consists of the following rules:
#
#           .t7>#.....t2- -- at least one digit need to be
#                               entered, after that, time out is 7 seconds
#                               before send, and terminating char # can also
#                               be applied after the first digit is entered,
#                               and after 7 digits are entered, time out
#                               change to 2 seconds. * means further digits
#                               can be entered as long as not terminated by
#                               timeout or #.
#
#           911 -- send out immediately
#
#           1t7>#.....t1- -- at least one digit need to be
#                               entered, after that, time out is 7 seconds
#                               before send, and terminating char # can also
#                               be applied after the first digit is entered,
#                               and after 10 digits are entered, time out
#                               change to 1 second. * means further digits can be
#                               entered as long as not terminated by timeout

```

```

#           or #.
#
#           0t4>#.t7- -- after entering 0, if no other digit is
#           entered, it will timeout and send in 4 seconds,
#           otherwise, time out change to 7 seconds after
#           another key is entered. again # is terminating
#           digit.
# Examples 2: The set of dial plan rules:
#
#           "911|1>#.r9t3.t5-|0t411t9-"
#
#           if 911 entered, it will be sent out immediately.
#           if 14088713344 is entered, after 3 seconds, it will
#           be sent out but if another digit is entered (say
#           140887133445, the timeout changed to 5 seconds).
#           if 0 is entered, after 4 seconds, it will be sent out.
#           if 011 is entered, the time out changed to 9 seconds.

```

Appendix B

ATA 188 configuration file for the sample configuration.

#txt

UIPassword:0

UseTftp:1
TftpURL:10.3.3.104

CfgInterval:3600

EncryptKey:0

upgradeCode:0,0x301,0x0400,0x0200,10.3.3.104,69,0x050616a,ata030201SIP050616a
.zup

dhcp:0

StaticIp:10.3.3.77
StaticRoute:10.3.3.1
StaticNetMask:255.255.255.0

DNS1IP:0.0.0.0
DNS2IP:0.0.0.0

NTPIP:10.3.3.10
AltNTPIP:0.0.0.0

VLANSetting:0x0000002b

PortsSetting:0x00000044

L2KeepAlive:0

Proxy:10.3.3.51
AltGk:0
SecProxy:0
AltGkTimeOut:0

UID0:34077
UID1:34078

PWD0:123456
PWD1:123456

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:ATA188 P1
DisplayName1:ATA188 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;HF;Jf;AFh;EQ;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*77;LA*87;mA;
Uh;AQ;

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

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