

Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Cotell FG1088-IP (2S) SP SIP Phone with Avaya Aura® Session Manager R6.3 and Avaya Aura® Communication Manager R6.2 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Cotell FG1088-IP (2S) SP SIP Phone to interoperate with Avaya Aura® Session Manager R6.3 and Avaya Aura® Communication Manager R6.2.

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

The Cotell FG1088-IP (2S) SP SIP Phone integrates with Avaya SIP infrastructure consisting of Avaya Aura® Session Manager R6.3 and Avaya Aura® Communication Manager R6.2 as third-party SIP endpoints. These Application Notes describe the steps required to configure Cotell FG1088-IP (2S) SP SIP Phone to work as Avaya third party SIP endpoints.

2. General Test Approach and Test Results

To verify interoperability of FG1088-IP (2S) SP SIP Phone with Session Manager and Communication Manager, calls were made between Cotell telephones and Avaya SIP, H.323 and Digital telephones using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using speed-dial buttons.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of Cotell telephones with Session Manager.
- Calls between Cotell telephones and Avaya SIP, H.323, and digital telephones.
- G.711, G.722 and G729/B codec support and negotiation, with and without media shuffling.
- Basic features including phone display, mute/un-mute, answer, hang up, music on hold, DTMF transmission, Message Waiting Indicator (MWI) subscription and feature access code dialing.
- PBX features including Multiple Call Appearances, Hold, and Conference.
- Proper system recovery after a Cotell telephone after removal and reconnection of LAN cable.

2.2. Test Results

All test cases were executed and passed successfully. However, the following point was noted during the testing:

• Transfer feature whether attended or unattended is not supported.

This was intentional for Cotell FG1088-IP (2S) SP SIP Phone in Hospitality environment.

2.3. Support

Technical support from Cotell International Limited can be obtained through the following:

- Telephone: +86-755 2583 1520
- Website: www.cotell-international.com

3. Reference Configuration

The diagram illustrates an enterprise site with an Avaya SIP-based network, including Messaging, Session Manager, S8800 Server running Communication Manager with a G430 Media Gateway, and Avaya SIP, H.323 and Digital endpoints. System Manager is used for configuring Session Manager. The enterprise site also contains two FG1088-IP (2S) SP SIP Phones used in the compliance testing. The Cotell phones are registered with Session Manager and are configured as endpoint users.

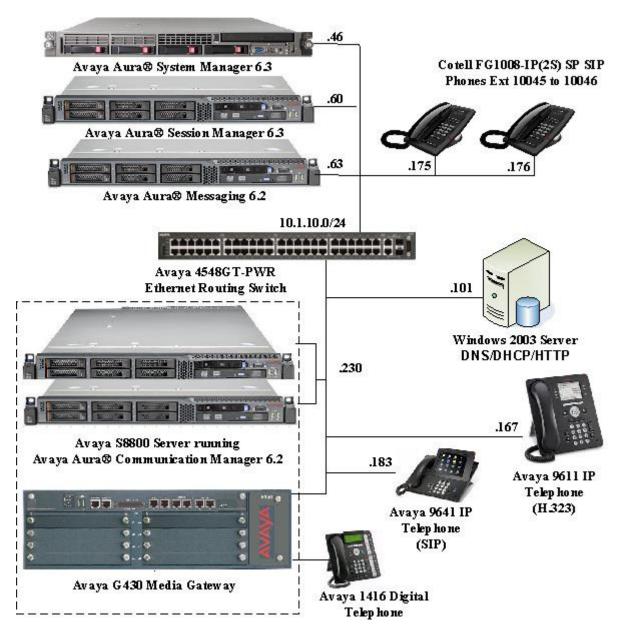


Figure 1: Cotell FG1088-IP(2S) SP SIP Phones with Avaya SIP Solution

Table 1 lists the extensions used for this testing.

Extension	Note
10001	Avaya 9611 IP Telephone (H.323)
10049	Avaya 9641 IP Telephone (SIP)
10016	Avaya 1416 Digital Telephone
10045 to 10046	Cotell FG1088-IP(2S) SP SIP Phones

Table 1 – Extension Setup

4. Equipment and Software Validated

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

Equipment/Software	Release/Version			
Avaya Aura® Communication	R6.2 SP5			
Manager on Avaya S8800 Duplex				
Servers				
Avaya Aura® Messaging on	R6.2 SP2			
DL360 G7 server				
Avaya G430 Media Gateway	32.26.0			
Avaya Aura® Session Manager on	R6.3 SP1			
Avaya S8800 Server				
Avaya Aura® System Manager	R6.3 SP1			
on DL360 G7 server				
Avaya 9641IP Telephone (SIP)	6.2.1			
Avaya 9611 IP Telephone (H.323)	6.2 SP3			
Avaya 1416 Digital Telephone	R4 SP1			
Avaya 4548GT-PWR Ethernet	V5.4.0.008			
Routing Switch				
Cotell FG1088-IP(2S) SP SIP	Firmware Version: Sep03, 2013 (2L)			
Phones	Hardware Version: v01A (2L base)			

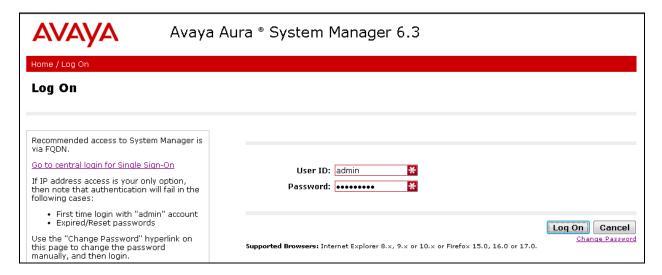
5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Avaya Aura® Session Manager. The procedures include the following areas:

- Launch System Manager
- Administer users

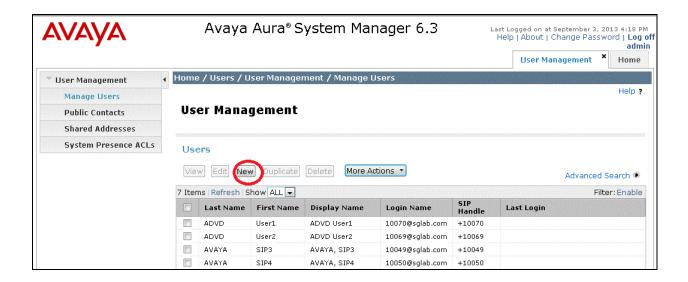
5.1. Launch System Manager

Access the System Manager Web interface by using the URL "https://ip-address" in an internet browser window, where "ip-address" is the IP address of the System Manager server. Log in using the appropriate credentials.



5.2. Administer Users

In the subsequent screen (not shown), select Users → User Management → Manage Users to display the User Management screen below. Click New to add a user.



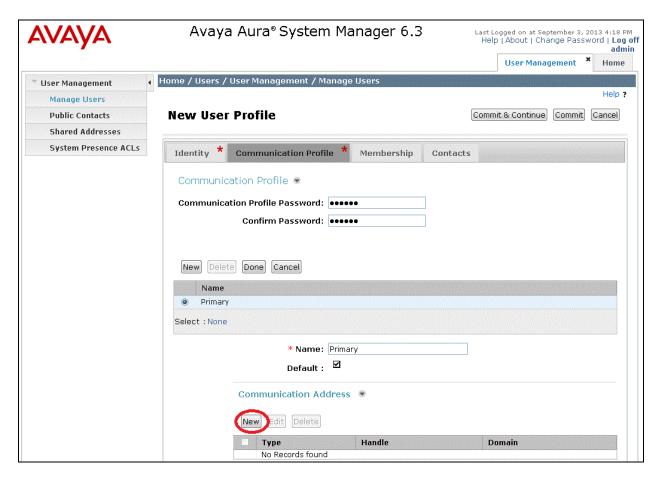
5.2.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter "n@z", where "n" is the Cotell FG1088-IP(2S) user extension and "z" is the domain name used for compliance testing, in this case "10046@sglab.com". For **Password** and **Confirm Password**, enter the appropriate credentials for SIP user for registration. Retain the default values in the remaining fields.



5.2.2. Communication Profile

Select Communication **Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration. Scroll down to Communication **Address** sub-section, and click **New** to add a new address.



For **Type**, retain "Avaya SIP". For **Fully Qualified Address**, enter and select the SIP user extension. The domain was pre-configured and is not shown in this document.



Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields. These settings are configured during the initial setup of Session Manager.

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Profile Type**, select **Endpoint**. For **Extension**, enter the SIP user extension from **Section 5.2.1**. For **Template**, select "**9630SIP_DEFAULT_CM_6_3**". For **Port**, select "**IP**". Retain the default values in the remaining fields.

Session Manage	r Profile 🏶					
SIP Registration	n					
	Primary Session Manager			Primary	Secondary 0	Maximum 7
Se	condary Session Manager	(None)		E		
Survivability Server		(None)	-			
Ma	ax. Simultaneous Devices	1				
	New Registration When Maximum trations Active?					
Application Sec	uences					
Origination Sequence		cm6-duplex-app-se	ea 💌			
Termir	nation Sequence	cm6-duplex-app-se	ea 💌			
Call Routing Se	ttings					
k	Home Location	Location1	-			
Confere	ence Factory Set	(None)				
CM Endpoint Pro		CM6-duplex				
	* Profile Type	Endpoint	•			
Use Ex	kisting Endpoints					
	* Extension	Q 10046	Endpoint Edi	tor		
	* Template	9630SIP DEFAULT	CM 6 3 💌			
	Set Type	9630SIP				
Security Code						
	Port	Q _{IP}				
Voice Mail Number				encencencental		
1	Preferred Handle	(None)	•			
Enhanced Callr-Info	display for 1-lin phones					
Delete Endpo	int on Unassign c	of 🔽				

Repeat Section 5.2 to add a user for the other Cotell FG1088-IP(2S) user. In the compliance testing, two users with extensions "10045" and "10046" were added.

6. Configure Avaya Aura® Communication Manager

The configuration for SIP trunk between Communication Manager and Session Manager was pre-configured and is not shown in this document. This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify Communication Manager license
- Administer IP codec set

6.1. Verify Communication Manager License

Log in to the System Access Terminal (SAT) to verify that Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command to verify that there is sufficient capacity for SIP stations by comparing the Maximum Off-PBX Telephones - OPS field value with the corresponding value in the USED column. The difference between the two values needs to be greater than or equal to the number of Cotell FG1088-IP(2S) SIP Phone extensions.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options
                                                                      1 of 11
                                                               Page
                               OPTIONAL FEATURES
    G3 Version: V16
                                                Software Package: Enterprise
      Location: 2
                                                 System ID (SID): 1
      Platform: 28
                                                 Module ID (MID): 1
                               Platform Maximum Ports: 65000 392
                                    Maximum Stations: 41000 200
                             Maximum XMOBILE Stations: 41000 0
                   Maximum Off-PBX Telephones - EC500: 41000 1
                   Maximum Off-PBX Telephones - OPS: 41000 35
                   Maximum Off-PBX Telephones - PBFMC: 41000 0
                   Maximum Off-PBX Telephones - PVFMC: 41000 0
                   Maximum Off-PBX Telephones - SCCAN: 0
                        Maximum Survivable Processors: 313
        (NOTE: You must logoff & login to effect the permission changes.)
```

6.2. Administer IP Codec Set

Use the "change ip-codec-set n" command, where n is the existing codec set number associated with the SIP trunk group to Session Manager. Update the audio codec types in the **Audio Codec** fields as necessary to include G.711MU, G.711A, G.729 and G.729B.

```
change ip-codec-set 6
                                                                Page 1 of
                                                                              2
                         IP Codec Set
   Codec Set: 6
Audio Silence Frames Packet Codec Suppression Per Pkt Size(ms)

1: G.711A n 2 20
                n
                             2
2: G.711MU
                     n
                              2
                                       20
                              2
3: G.729
                                        20
4: G.729B
                              2
                                        20
5:
 6:
7:
```

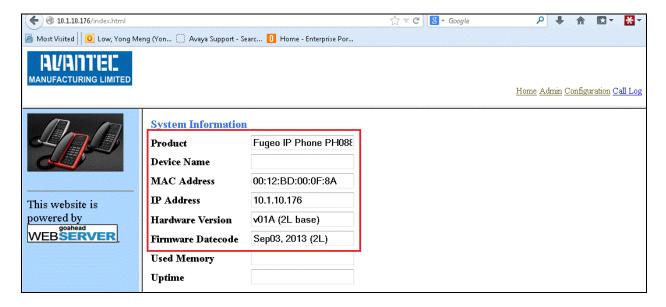
7. Configure Cotell FG1088-IP(2S) SIP Phones

This section provides the procedures for configuring Cotell FG1088-IP(2S) SIP Phones. The procedures include the following areas:

- Access Web Interface
- Configure Network Settings
- Configure SIP Settings
- Configure SIP Accounts
- Configure Audio Settings

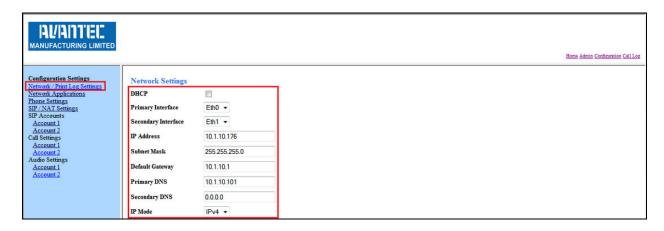
7.1. Access Web Interface

Enter <a href="http://<ip-addr>/">http://<ip-addr>/, where <ip-addr> is the IP address of the Cotell phone, into the address bar of web browser and log in using a valid account. The **System Information** screen is displayed on the Home page.



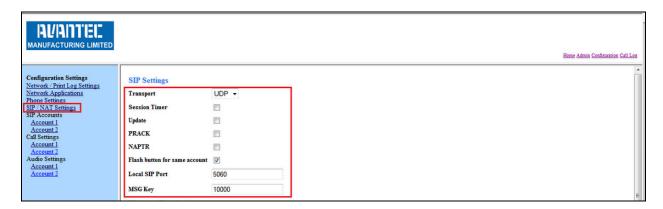
7.2. Configure Network Settings

Select Configuration menu (top right) and the left menu for Configuration Settings will be shown. Select the Network/Print Log Settings from the left menu. Uncheck the DHCP as Static IP is used in this testing. Enter the appropriate IP address, Subnet Mask, Default Gateway and Primary DNS. Select IPv4 for the IP Mode. Leave the rest as default. Click Save (not shown) to continue.



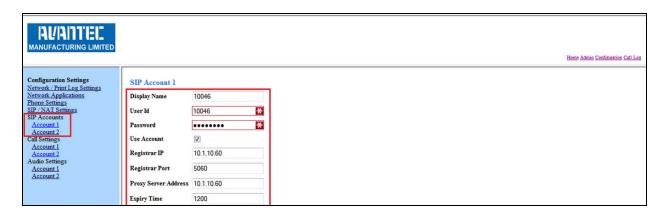
7.3. Configure SIP Settings

Select **Configuration** (top right) and the left menu for **Configuration Settings** will be shown. Select **SIP/NAT Settings** from the left menu. Select UDP as **Transport**. Check on the **Flash button for same account**. This is to allow making 3rd party call with same line using flash button. Enter the Voice Mail number as **10000** in **MSG Key**. This depends on the Voice Mail number setup in the configuration. Leave the rest as default. Click **Save** (not shown) to continue.



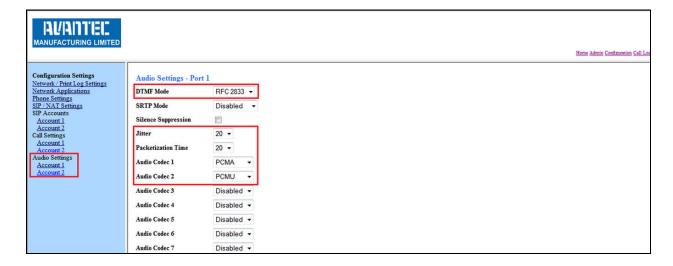
7.4. Configure SIP Accounts

Select Configuration menu (top right), the left menu for Configuration Settings will be shown. Select Accounts 1 under SIP Accounts from the left menu. Enter Display Name, User Id and Password with the account details as shown below to match the user settings in Session Manager added in Section 5.2. Check to activate the User Account. Enter the Registrar IP and Proxy Server Address with Session Manager address and the default port as 5060. Enter the Expiry Time as 1200. Click Save (not shown) to continue. Repeat with the same user settings for Account 2. For a different user account, a new user has to be created in Session Manager as in Section 5.2.



7.5. Configure Audio Settings

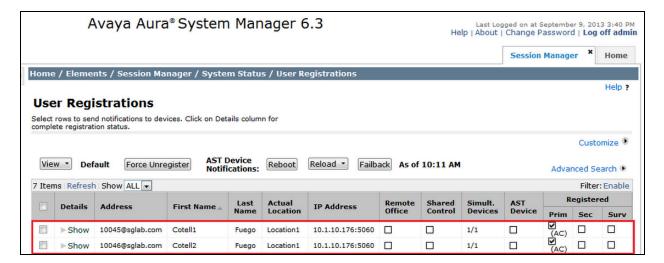
Select Configuration menu (top right), the left menu for Configuration Settings will be shown. Select Account 1 under Audio Settings in the left menu. Select RFC 2833 for DTMF Mode. Leave the Jitter and Packetization Time as default 20 ms. In codec preference, select the appropriate codec to be supported for the phone in the order listed. In the screen shown below, PCMA is set as first preference and PCMU is set as second preference. Click Save (not shown). Repeat the same for Account 2 under Audio Settings.



8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and Cotell FG1088-IP(2S) SIP Phones.

From the System Manager Web interface, select **Elements** → **Session Manager** → **System Status** → **User Registrations** to display the **User Registrations** screen. Verify that the user created is registered, as shown below with a check in the **Registered Prim** column.



9. Conclusion

These Application Notes describe the configuration steps required for Cotell FG1088-IP(2S) SP SIP Phones to successfully interoperate with Avaya Aura® Session Manager 6.3 and Avaya Aura® Communication Manager 6.2. All feature and serviceability test cases were completed successfully.

10. Additional References

This section references documentation relevant to these Application Notes. Avaya product documentation is available at http://support.avaya.com.

- [1] Administering Avaya Aura® Communication Manager Server Options, Release 6.2, Doc ID 03-300509, Issue 3, July 2012.
- [2] Administering Avaya Aura® Session Manager, Release 6.3, December 2012.
- [3] *Implementing Avaya Aura*® *Communication Manager Messaging*, Release 6.2, Issue 2, January 2013.

The following document is provided by Cotell International Limited.

- [1] Hospitality Telephones User Guide for Fuego 1000TM Desktop IP Telephone FG1088-IP (2S) SP 2-line IP Guestroom Telephone
- [2] Fuego SmartStation IP Datasheet.

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