Avaya Solution & Interoperability Test Lab

Application Notes for the Polycom 6020 Wireless Handsets, Polycom Master Control Unit and Polycom Base Stations with Avaya Communication Manager – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager and Polycom 6020 wireless phones, a Polycom Master Control Unit and a Polycom 6020 Base Station. The Polycom Master Control Unit is connected to an Avaya digital interface, capable of handling Avaya 8410D digital phones, on the Avaya Media Gateway. The Polycom 6020 wireless phones register to the Polycom Master Control Unit via the Polycom 6020 Base Station and are configured as digital stations in Avaya Communication Manager. The Polycom 6020 wireless phones are used to originate calls to, and receive calls from, digital, analog and IP phones and execute other features such as Call Waiting, Caller ID, Hold, Transfer, Call Forwarding, Voicemail, DTMF and MWI. Information in these notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Manager and Polycom 6020 wireless phones, a Polycom Master Control Unit and a Polycom 6020 Base Station. The Polycom Master Control Unit is connected to an Avaya digital interface, capable of handling Avaya 8410D digital phones, on the Avaya Media Gateway. The Polycom 6020 wireless phones register to the Polycom Master Control Unit via the Polycom 6020 Base Station and are configured as digital stations in Avaya Communication Manager. Polycom 6020 handsets are used to originate calls to, and receive calls from, digital, analog and IP phones and execute other features such as Call Waiting, Caller ID, Hold, Transfer, Call Forwarding, Voicemail, DTMF and MWI. The Polycom 6020 handsets are also tested for roaming between Polycom Base Stations.

Figure 1 illustrates a sample configuration consisting of an Avaya S8300B Server, an Avaya G700 Media Gateway, a Polycom Master Control Unit (Polycom MCU), Polycom 6020 Base Stations (Polycom Base Station) and Polycom 6020 handsets (Polycom handsets). Avaya Communication Manager is installed on the Avaya S8300B Server. The solution described herein is also extensible to other Avaya Servers and Media Gateways. For completeness, Avaya SIP Enablement Services (SES), Avaya IA770 Intuity Audix, Avaya 4600 Series IP Telephones, Avaya 9600 Series IP Telephones, and Avaya 2420 Digital Telephones, are included in **Figure 1** to demonstrate calls between Polycom handsets and Avaya SIP, H.323, digital telephones and to Avaya IA770 Intuity Audix.

Typical call flows in this configuration between Avaya Communication Manager and the Polycom MCU are as follows:

- Incoming calls into Polycom MCU
  - Calls originate from IP trunks/endpoints to a destination number configured in Avaya Communication Manager and associated with the Polycom handset.
  - The Polycom MCU delivers the call to Polycom handset.
- Outgoing calls from Polycom MCU
  - Polycom MCU receives an outbound call from Polycom handset registered with it and forwards it to Avaya Communication Manager over its digital link.
  - Avaya Communication Manager routes the call to an Avaya SIP, H.323 or digital telephone.

These Application Notes assume that Avaya Communication Manager is already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult **[1]** and **[2]**.

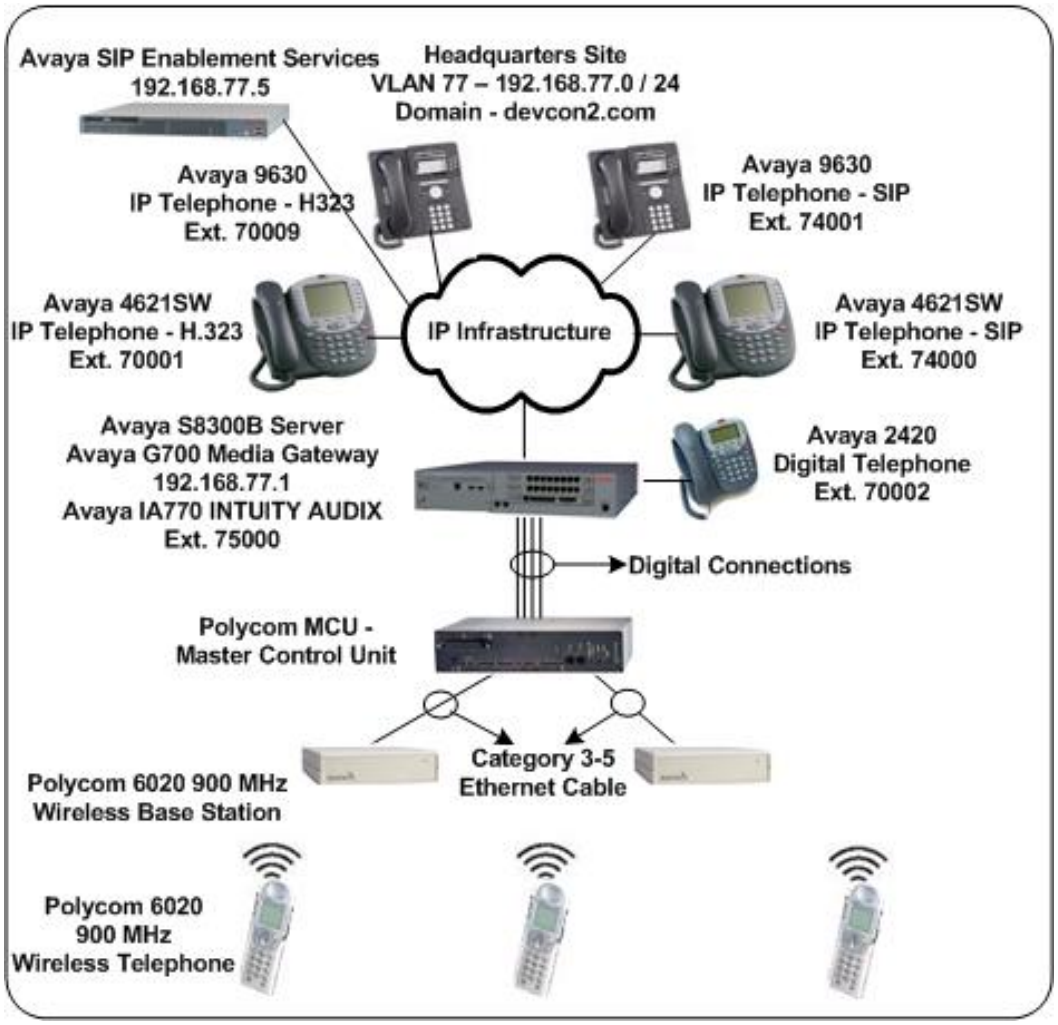

Figure 1: Sample Configuration

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8300B Server	Avaya Communication Manager 4.0 (R014x.00.730.5)
Avaya G700 Media Gateway	26.31.0
MM712AP Digital Card	HW 05 FW008
Avaya 9600 Series IP Telephones	1.0.1 (SIP) 1.5 (H.323)
Avaya 4600 Series IP Telephones	2.2.2 (SIP) 2.8 (H.323)
Avaya 2420 Digital Telephones	-
Polycom Master Control Unit	150/MOD 3
Polycom 6020 Base Station	N/A
Polycom 6020 Handset	110.016

3. Configure Avaya Communication Manager

Each Polycom handset corresponds to an Avaya 8410D Digital Telephone administered in Avaya Communication Manager. Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields. These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. Refer to [1] for additional details.

Step	Description
1.	Enter **add station s** where **s** is set to 71001, set **Type** field to **8410D** and set **Port** to a valid value.

```
add station 71001                                       Page   1 of   5
                                   STATION

Extension: 71001                        Lock Messages? n          BCC: 0
     Type: 8410D                        Security Code:             TN: 1
     Port: 02A0601                    Coverage Path 1:  3          COR: 1
     Name:                            Coverage Path 2:             COS: 1
                                      Hunt-to Station:

STATION OPTIONS
             Loss Group: 2            Personalized Ringing Pattern: 1
           Data Module? n                       Message Lamp Ext: 71001
         Speakerphone: 2-way                 Mute Button Enabled? y
       Display Language: english

         Survivable COR: internal                Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n


                                                   IP Video Phone? n
```

2.	On **Page 2** of the **STATION** form, set **Restrict Last Appearance** field to **n**.

```
add station 71001                                       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 Call Alerting? n                   Restrict Last Appearance? n
  Active Station Ringing: single

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

                                         Direct IP-IP Audio Connections? y
```

Step	Description
3.	On **Page 4** of the **STATION** form, enter the number of call appearances **(call-appr)** required.
	add station 71001 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** 6: **2: call-appr** 7: **3: call-appr** 8: **4: call-appr** 9: 5: 10:
4.	Save the station configuration. Repeat **Steps 1-3** to configure additional stations.

4. Configure Polycom Master Control Unit

This section describes the steps for configuring the Polycom MCU, which supports a variety of PBXs. Refer to [3] for additional information.

Step	Description
1.	Power on the Polycom MCU with the Operators Switch in Normal. After powering the system, it will alarm. The Error LED will flash along with Status LEDs 1,2,3,4 and 5, to indicate that the MCU is not configured. This should take less than 2 minutes.
2.	Power off the Polycom MCU.
3.	Move the Operators Switch to Admin and power on the Polycom MCU. At that time Status LEDs 2 and 4 will be lit. This should take less than 15 seconds.
4.	Press the STEP button 3 times. Status LEDs 1,2 and 4 must be lit. Line LEDs 1 through 8 should also be lit. If there is some other combination of Line LEDs lit, then a PBX type has already been selected. Go to **Step 5** to pick the desired PBX type.

Step	Description
5.	Press the DEL/ENTER button to pick the correct PBX type. With each press of the DEL/ENTER button, a different series of Line LEDs will be lit. Continue to hit DEL/ENTER button until the correct Line LEDs are lit. Use the following list to select the desired PBX integration. For this compliance test, **2wire Definity** configuration was selected.

PBX Type	Line LEDs	MCU Type	Notes
Unconfigured	1 through 8		
None	No Line LEDs		DO NOT USE
Analog	1	SCA516	The SCA will only support Analog
Norstar	2	SCU516	
Meridian	1,2	SCU516	
Comdial	3	SCU516	
Merlin Legend	1,3	SCF516	
Toshiba	2,3	SCU516	
Mitel	1,2,3	SCX516	The SCX will only support Mitel
Siemens / Rolm	4	SCU516	
2wire Definity	**1,4**	**SCU516**	
Fujitsu	2,4	SCU516	
NEC	1,2,4	SCU516	
ISDN-C	3,4	Not Available at this time	
ISDN-N	1,3,4	Not Available at this time	
Inter-Tel Access	2,3,4	SCU516	
Panasonic	5	SCP516	
4wire Definity	1,5	SCF516	
Executone	2,5	SCB516	
Siemens HiCom 150	1,2,5	SCH516	The SCH will only support HiCom
Siemens HiCom 300	3,5	SCH516	The SCH will only support HiCom
Inter-Tel Eclipse	1,3,5	SCU516	

6.	Move the Operator Switch back to Normal. At this time the Polycom MCU will go through the normal power up sequence and is properly configured. The system should come up in less than 2 minutes.

5. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment using the Polycom handsets registered with Polycom MCU and operations such as dialing methods (manual, re-dial), Hold, Mute, Transfer, Call Waiting, Caller ID and Conference.

5.1. General Test Approach

The general test approach was to place calls to and from the Polycom handsets and exercise basic telephone operations. The main objectives were as follows:

- The Polycom MCU successfully configures with Avaya Communication Manager.
- The Polycom handsets registered with the Polycom MCU can establish calls with Avaya SIP, H.323, and digital telephones associated with Avaya Communication Manager.
- The Polycom handsets successfully handle concurrent calls.
- The Polycom handsets display MWI, transmit DTMF during a call, and retrieve voice messages.
- The Polycom handsets successfully hold a call, transfer a call, establish a three party conference call, and display calling party number.

For serviceability testing, failures such as cable disconnect and hardware resets were introduced. For performance testing, a conference call involving two Polycom handsets and two Avaya telephones was formed as follows. A call was established between an Avaya telephone and a Polycom handset, which then established a call with another Polycom handset and bridged two calls together, forming a 3-party conference. The second Polycom handset then established a call with another Avaya telephone, and bridged the two calls together, effectively forming a 4-party conference.

5.2. Test Results

The test objectives of **Section 5.1** were verified. For serviceability testing, the Polycom solution worked properly after recovering from failures such as cable disconnects, and resets of the Polycom MCU, Polycom Base Stations, and Avaya Communication Manager. For performance testing, the conference call was successfully maintained for approximately two hours.

The following observations were made during testing:

- Polycom handsets were able to handle up to six call appearances only. The 7th and 8th calls show up on Polycom handsets but they could not be answered.

Polycom will address and resolve the above observation in a future release. Contact Polycom (www.polycom.com) for further updates.

6. Verification Steps

- Verify the Polycom MCU is configured properly by observing the LED on the base station. If the LED on the base station cycles between amber and green, then the Polycom MCU has been configured properly.

7. Support

For technical support on Polycom products in this solution, consult the support pages at http://www.polycom.com/support/ or contact technical support at:

- Telephone: 1-800-POLYCOM
- E-mail: support@polycom.com

8. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager and Polycom 6020 wireless phones, a Polycom Master Control Unit and Polycom 6020 Base Stations. The Polycom Master Control Unit is connected to an Avaya digital interface, capable of handling Avaya 8410D digital phones, on the Avaya Media Gateway. The Polycom 6020 wireless phones register to the Polycom Master Control Unit via the Polycom 6020 Base Station and are configured as digital stations in Avaya Communication Manager. The Polycom 6020 wireless phones are used to originate calls to, and receive calls from, digital, analog and IP phones and execute other feature such as Call Waiting, Caller ID, Hold, Transfer, Call Forwarding, Voicemail, DTMF and MWI. The compliance testing was successful with the exception of the issue noted in **Section 5.2**.

9. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Issue 3.1, February 2007, Document Number 03-300509
[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 12, February 2007, Document Number 555-233-504

Product documentation for Polycom products may be found at http://www.polycom.com.

[3] Polycom Master Control Unit Installation Guide

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 Dev*Connect* Program at devconnect@avaya.com.