Avaya Solution & Interoperability Test Lab

Application notes for Algo 8180 SIP Audio Alerter release 1.0.1 with Avaya Communication Server 1000 Release 7.0 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.0 and Algo 8180 SIP Audio Alerter. During the compliance testing, the Algo 8180 SIP Audio Alerter was able to register as a SIP client endpoint with the Communication Server 1000. The Algo 8180 SIP Audio Alerter was able to receive calls from Communication Server 1000 Release 7.0 non-SIP and SIP Line clients. The compliance tests focused on telephony features.

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

1. Introduction

These application notes provide detail configurations of Avaya Communication Server 1000 SIP Line release 7.0 (hereafter referred to as CS1000) and the Algo 8180 SIP Audio Alerter release 1.0.1 used during the compliance testing. The Algo 8180 SIP Audio Alerter was tested with the non-SIP and SIP clients of the CS1000 SIP line release 7.0. All the applicable SIP audio alerter feature test cases of release 7.0 SIP line were executed on the Algo 8180 SIP Audio Alerter , where applicable, to ensure the interoperability with CS 1000.

1.1. Interoperability Compliance Testing

The focus of this testing was to verify that the Algo 8180 SIP Audio Alerter was able to interoperate with the CS1000 SIP line system. The following areas were tested:

- The Algo 8180 SIP Audio Alerter must be able to be installed in the same local VLAN network as the CS1000 successfully.
- Registration of the Algo 8180 SIP Audio Alerter to the CS1000 SIP Line Gateway.
- Calls establishment of CS1000 SIP and non-SIP telephones with Algo 8180 SIP Audio Alerter.
- Telephony features: Ringing when receiving basic call, receiving conference call, transfer call as a targeted transfer, busy and paging.

1.2. Support

For technical support on Algo 8180 SIP Audio Alerter SIP endpoints, please contact Algo Communications technical support at website http://support.algosolutions.com/index.php or Toll Free: 1-877-884-2546 (Canada & USA only), Tel: 604-454-3792 or email support@algosolutions.com

2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the Algo 8180 SIP Audio Alerter.

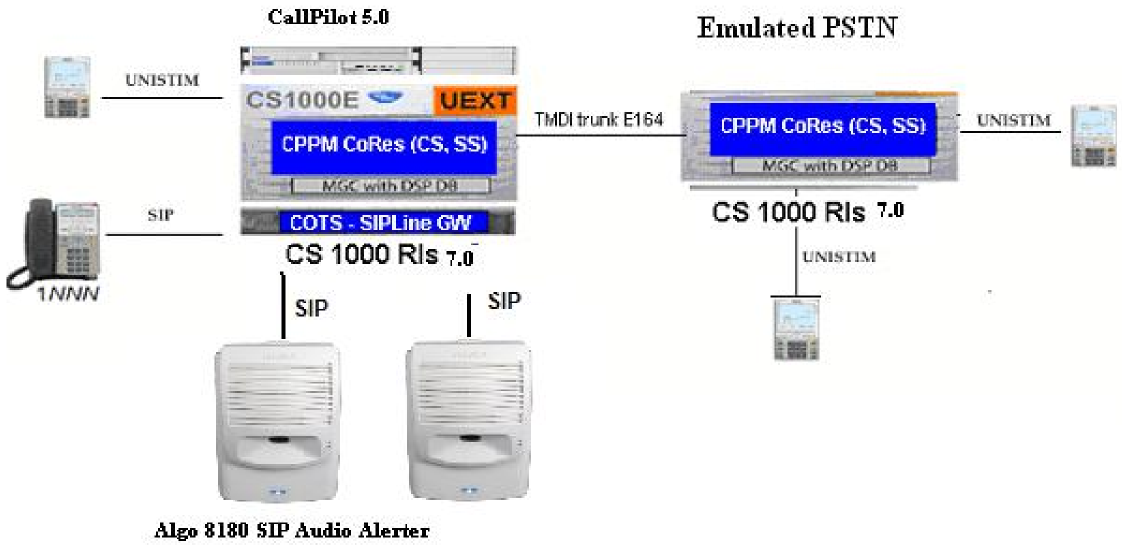

Figure 1: Test Bed Configuration

3. Equipment and Software Validated

System	Software Version
Avaya CS1000	• Call Server (CPPM): 7.00Q • Signaling Server (CPPM): 7.00.20 • SIP Line Gateway
Avaya voicemail system	• CallPilot 5.0 system
Avaya 11xx SIP client (Sigma) Avaya SIP soft-phones Avaya IP phones	• 02.02.16.00 • SMC3456: v2.6 Build 53715 • 2050PC: 3.02.0045
Algo 8180 SIP Audio Alerter	• 1.0.1

4. Configure Avaya CS 1000 - SIP LINE

This section describes the steps to configure the Avaya CS1000 SIP Line using CS 1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on CS 1000 system. For detailed information, see [1].

4.1. Prerequisite

- A CS1000 server which has been:
  - Installed with CS 1000 Release 7.0 Linux Base.
  - Joined CS 1000 Release 7.0 Security Domain.
  - Deployed with SIP Line Application.

  For more information, see [6].

- Following packages are enabled in the keycode. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

Package Mnemonic	Package Number	Package Description	Package Type (New or Existing or Dependency)	Applicable Market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_NORTEL	415	Nortel SIP Line package	Existing package	--
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	--

4.2. SIP Line Phones Configuration

Following is a configuration for a Third Party SIP Line endpoint. Depending on supported features and service access level of the user, this configuration can be adjusted accordingly. This configuration is using the Command Line Interface of CS1000. This can be done by login to Call Server of CS1000 and using overlay 11 as shown below. The bold text values are the changes required where others are at default values.
Note: SIPL on CS1K7.0 does not support a Paging configuration.
ALGO 8180 can page when detecting ringing tone of an extension.

>LD 11

REQ: prt
TYPE: tnb
TN 96 0 1 27
DATE
PAGE
DES

DES TELE
TN 096 0 01 27 VIRTUAL
TYPE **UEXT**

CDEN 8D
CTYP XDLC
CUST 0
UXTY **SIPL**
MCCL **YES**
SIPN 0← Set this to 1 and set SIP3 to 0 if this TN is reserved for Avaya SIP Phones
SIP3 **1** ← Set this to 1 and set SIPN to 0 if this TN is reserved for third party SIP Phones
FMCL 0
TLSV 0
SIPU **55573**
NDID **556**
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE **001**
CUR_ZONE 001
ERL
ECL 0
FDN **55576** ← If CLS FNA is equipped, call will be forwarded no answer to this number
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG **2** ← This field must be set first if call pickup is equipped (CLS PUA)
SCI 0
SSU
XLST
SCPW **1234**
SFLT NO
CAC_MFC 0
CLS UNR **FBA** WTA LPR **PUA** MTD **FNA HTA** TDD HFA CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD SLKD CCSD **SWD** LND **CNDA**
CFTD SFD MRD DDV **CNIA** CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUA DPUA **DNDA** CFXA ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTD AHA IPND **DDGA NAMA** MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3
MCBN
FDSD NOVD VOLA VOUD CDMR ICRD MCDD T87D MSNV FRA PKCH
CPND_LANG ENG
RCO 0

HUNT **55576** ← If CLS HTA/FBA is equipped, call will be forwarded busy to this number
LHK 0
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 **SCR 55573** 0 MARP
CPND
CPND_LANG ROMAN
NAME **Algo 55573**
XPLN 13
DISPLAY_FMT FIRST, LAST
01 HOT U **2655573** MARP 0
02 SCU **0004** ← Speed Call User
03
04 **MSB** ← This key can be different than key 04 to enable Make Set Busy (MBS) feature
05
.
.
16
17 TRN
18 AO6
19 CFW 16 55574
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN

5. Configure Algo 8180 SIP Audio Alerter

This section describes how to access the Algo 8180 SIP Audio Alerter SIP endpoint web interface and configure the Algo 8180 SIP Audio Alerter for testing.

ALGO 8180 is a SIP compliant PoE audio device that registers with a server in the same way as a SIP telephone. During the initialization, Algo 8180 will attempt to obtain an IP address from a DHCP server. The Blue Light Indicator will turn off after initialization is complete.

Configuring ALGO 8180 to register to User Agents for loud ringing and auto-answer voice paging:

- Access web interface of ALGO 8180 by entering the IP address of ALGO 8180 on Web Browser and using default password "algo" to login.
- Use the SIP Audio Alerter Control Panel to configure: SIP domain/ Proxy (IP address or FQDN), Ring Detect Extension, Page Audio Extension and password for registration. Then click on Save Settings button as shown in Figure 2.

Figure 2: Register ALGO 8180 to a User Agent.

For checking the status of registration, go to Status menu and check information as shown in Figure 3.

Figure 3: Register state of ALGO 8180

To change ringing tone or paging volume of ALGO 8180, go to Config page, scroll down to Features section and select desired ringing tone. Then click on Save Settings button as shown in Figures 4.

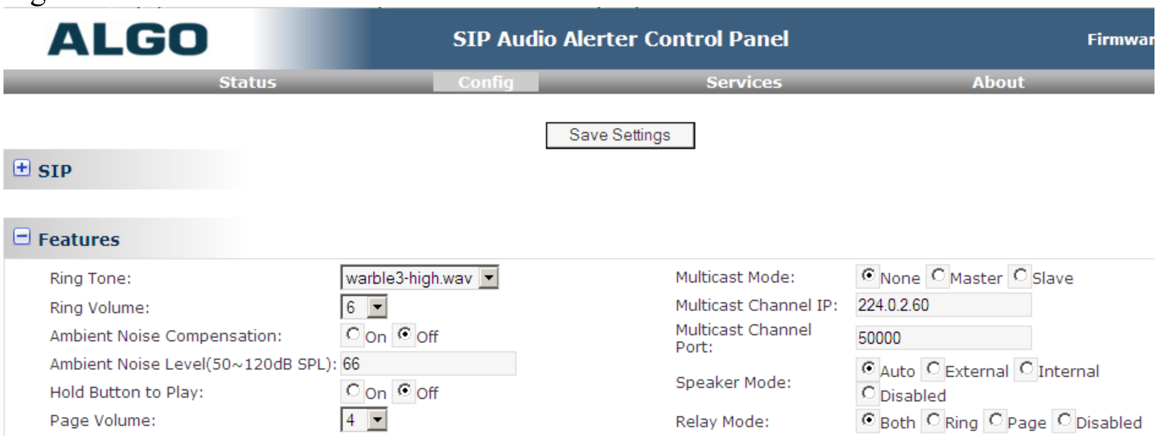

Figure 4: Configure ringing tone

6. General Test Approach and Test Results

The focus of this interoperability compliance testing was primarily to verify the call establishment between the Algo 8180 SIP Audio Alerter and the CS1000 telephones; SIP and non SIP. Other call features, receiving transfer call, receiving conference call, busy, were exercised.

6.1. General test approach

The general test approach was to have one of the CS1000 telephone clients/users to place a call to the Algo 8180 SIP Audio Alerter and to exercise other telephony features. The main objectives were to verify the Algo 8180 SIP Audio Alerter successfully performed the following:

- The Algo 8180 SIP Audio Alerter must be able to be installed in the same local VLAN network as the CS1000 successfully.
- Registration of the Algo 8180 SIP Audio Alerter to the CS1000 SIP Line Gateway.
- Calls establishment of CS1000 SIP and non-SIP telephones with Algo 8180 SIP Audio Alerter.
- Telephony features: Ringing when receiving basic call, receiving conference call, transfer call as a targeted transfer, busy and paging.

6.2. Test Results

The objectives outlined in section 6.1 were verified. The following observations were made during the compliance testing:

- SIPL on CS1K7.0 does not support Paging configuration.
- ALGO 8180 can page when detecting ringing tone of an extension

- Issue of wrong status display of the Algo 8180. After the successfully registering to CS1000 as a SIP line client, user go to Algo 8180 Config page, remove all the configuration details and save the settings. Then go to the Status page, the "Login successful" in SIP Registration Info is displayed instead of “No account”. This issue is fixed in the next release of Algo 8180 1.0.3.

Avaya has not performed audio performance testing or reviewed the SIP phones compliance to required industry standards.

7. Verification Steps

This section includes some steps that can be followed to verify the configuration.

- Verify that the SIP telephone registers successfully with the CS 1000 SIP Line Gateway server and Call Server by using CS 1000 Linux command line and CS 1000 Call Server overlay LD 32.
  - Login to the sipline server using Avaya account.
  - Issue command “slgSetShowByUID [userID]” where userID is SIP Line user’s ID being checked.

```
[nortel@sipl ~]$ slgSetShowByUID 55524
=== VTRK ===
UserID          TN               Clients  Calls  SetHandle
--------------- ---------------  -------  -----  ---------
          55524     096-00-02-24        1      0  0xb7c25e10
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = 0

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IP:Port:Trans = 47.248.100.56:5060:udp
         Type          = SIP3
         UserAgent     = PolycomVVX-VVX_1500-UA/3.2.2.0481
         x-nt-guid     = 9ad7a4871842718a35aeadf070608745
         RegDescrip    =
         RegStatus     = 1
         PbxReason     = OK
         SipCode       = 200
         Expire        = 300
         Contact       = sip:55524@47.248.100.56:5060
         Nonce         = cad64489bbd9bca62aa9a1f833052da4
         NonceCount    = 3
         hTimer        = 0x9c659d0
         TimeRemain    = 183
         Stale         = 0
         Outbound      = 0
         ClientGUID    = 0

        Key  Func  Lamp  Label
        0    3     0     55524
        1    126   0     2655524
```

```
2     3     0      55097
3     9     0
4     29    0
17    16    0
18    18    0
19    27    0
20    19    0
21    52    0
22    25    0
24    11    0
25    30    0
26    31    0
```

- Login to the call server using admin account.
- Load overlay 32 and then issue command "stat [TN]" where TN is the SIP Line user's TN being checked

```
>ld 32
NPR000
.stat 96 0 2 24
IDLE REGISTERED 00
```

- Place a call to Algo 8180 SIP Audio Alerter, the alerter will be ringing. There is one way voice path from Avaya telephone to the alerter.
- During the call, use pcap tool (ethereal/wireshark) at the SIPLine Gateway and clients to make sure that all SIP request/response messages are correct.

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 6.1**, with some exceptions outlined in **Section 6.2**. The outstanding issues are being investigated by Algo and Avaya design teams. Some of these issues are considered as exceptions. The Algo 8180 SIP Audio Alerter rel. 1.0.1 is considered in compliance with Avaya CS1000 SIP Line System Release 7.0.

9. Additional References

Product documentation for Avaya products may be found at:
http://support.nortel.com/go/main.jsp

[1] *Communication Server 1000 SIP Line Fundamentals, Release 7.0, Revision 02.03, August 2010, Document Number NN43001-508*

[2] *Communication Server 1000E Maintenance, Release 7.0, Revision 04.02, June 2010, Document Number NN43041-700*

[3] *Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning, Release 7.0, Revision 04.01, June 2010, Document Number NN43001-717*

[4] *Troubleshooting Guide for Distributors, Release 7.0, Revision 03.01, June 2010, Document Number NN43001-730*

[5] *Communication Server 1000E Installation and Commissioning, Release 7.0, Revision 04.02, June 2010, Document Number NN43041-310*

[6] *Communication Server 1000E Software Upgrades, Release 7.0, Revision 04.03, June 2010, Document Number NN43041-458*

[7] *Communication Server 1000E Linux Platform Base and Applications Installation and Commissioning, Release 7.0, Revision 04.03, September 2010, Document Number NN43001-315*

[8] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Release 7.0, Revision: 04.01, June 2010, Document Number NN43001-116*

Information for Algo 8180 SIP Audio Alerter products can be found at: www.algosolutions.com

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 DevConnect Program at devconnect@avaya.com.