Interoperability Certification

Algo 8180 SIP Audio Alerter / Asterisk 1.8.6
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Executive Summary

This document covers the setup and the tests used to validate the interoperability of Algo 8180 with Digium's Asterisk software. All relevant information is included in order to allow the replication of these scenarios.

Products Tested

The partner products listed below have been tested for interoperability with the Asterisk version(s) listed below. The software versions for all tested products are included.

Product	Version	Remarks
Asterisk	1.8.6.0	RPM from packages.asterisk.org
Algo 8180 SIP Audio Alerter	Firmware 1.3	

Algo 8180 SIP Audio Alerter

The Algo 8180 is a SIP compliant PoE network audio device for loud ring and voice paging applications using dual endpoints. When registered with a SIP server, one endpoint will play an audio file from internal memory upon ring detection. The second endpoint will auto-answer for voice paging, complete with two-way talkback.

Key features:

- Loudness - typically eight times louder than a telephone speaker
- Audio Files - plays pre-loaded audio
- Ambient Noise Compensation - automatically adjusts loud ring and paging volume
- Outputs for External Equipment and Devices - supports external speaker, slave amplifier, visual alerter
- Configuration & Provisioning - configured via web interface, by program buttons, or via TFTP download
- Paging Talkback - allows bidirectional talkback
- Multicasting - multicast simultaneously to other 8180s without requiring a separate paging server
- Blue Indicator Light - shows device state or message waiting
- Centralized Provisioning - supported via HTTP or FTP
- Carries CSA/UL and CE certifications

Summary of Test Focus

A summary of the capabilities validated is provided in the following table.

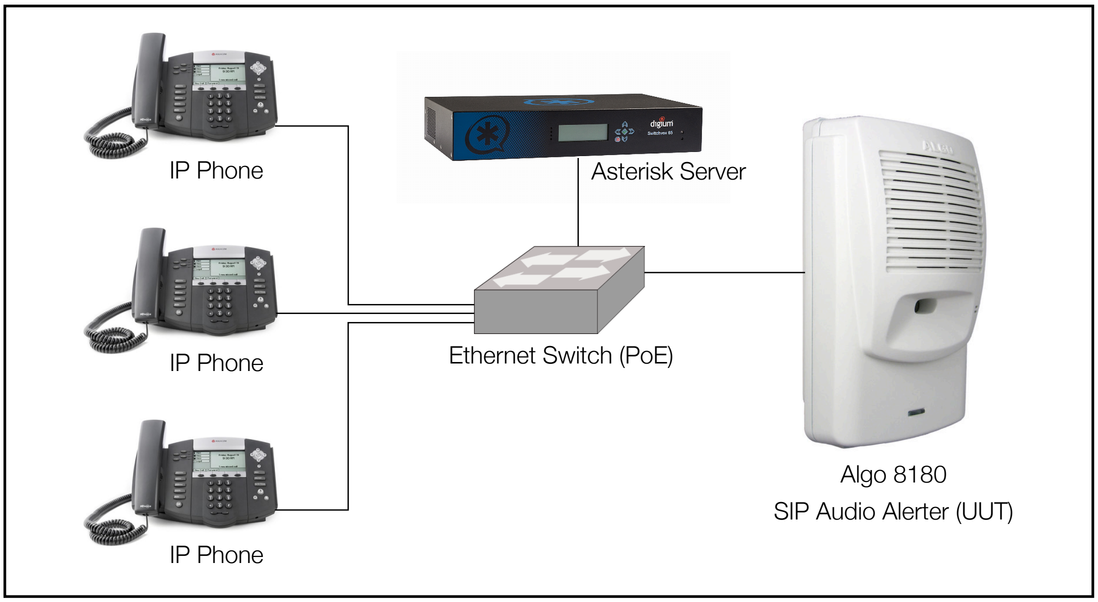
Feature	Algo 8180 SIP Audio Alerter
SIP Registration	✓
Inbound Call: Ring Extension	✓
Inbound Call: Page Extension	✓
Serviceability	✓

Test Configuration

This section describes the test configuration and setup. A diagram of the test setup is provided in Section 2.2.

Test Setup

As shown in the following diagram, an isolated network was assembled using an Adtran NetVanta PoE switch and a server running Asterisk 1.8.6 on CentOS Linux 5.5. The unit under test was connected to the network, and each feature test was conducted as described.



Configurations applied

The UUT was configured according to the included instructions. The particular settings used for the test were:

```
SIP Domain/Proxy: 10.19.11.250
Ring Detect Extension: alerterring
  Auth ID: alerterring
  Password: algotest123
Page Audio Extension: alerterpage
  Auth ID: alerterpage
  Password: algotest123
```

Asterisk was configured to accept SIP registrations and calls to/from the unit under test.

/etc/asterisk/sip.conf

```
[alerterring]
type=friend
context=default
secret=algotest123
host=dynamic

[alerterpage]
type=friend
context=default
secret=algotest123
host=dynamic

[phonea]
type=friend
context=default
secret=abc123
host=dynamic

[phoneb]
type=friend
context=default
secret=abc123
host=dynamic

[phonec]
type=friend
context=default
secret=abc123
host=dynamic
```

/etc/asterisk/extensions.conf

```
[default]
exten => 7001,1,Dial(SIP/alerterring)
exten => 7002,1,Dial(SIP/alerterpage)
exten => 7003,1,Dial(SIP/phonea)
exten => 7004,1,Dial(SIP/phoneb)
exten => 7005,1,Dial(SIP/phonec&SIP/alerterring)
```

Definitions

- **UUT** - Unit Under Test
- **Server** - Asterisk acting as a SIP back-to-back user agent
- **Phone A** - a SIP compatible endpoint used to call the UUT
- **Phone B** - a SIP compatible endpoint used to call Phone A & the UUT
- **Phone C** - a SIP compatible endpoint used to register with the same server as the UUT

- **Ring Extension** - This is the extension that will be called from Phone A or Phone B in order to trigger a “Ring” sound from the UUT. The UUT will register this extension with the Server, using the username & password entered in the Ring Extension fields via its web interface. The UUT will expect to receive INVITE messages for this extension, but will not answer the call.
- **Page Endpoint** - This is the extension that will be called from Phone A or Phone B in order to send “Page” audio to the UUT. The UUT will register this extension with the Server, using the username & password configured in the Page Extension fields via its web interface. The UUT will auto-answer any INVITE messages for this extension.

Tests Performed

each test performed is described below. The mandatory tests (Registration and Basic Call Function) validate functionality that is required for interoperability.

SIP Registration

The following test cases verify features related to the registration process with Asterisk.

Test Case	Actions	Result
Reg 1	**Attempt registration of UUT Ring Extension using incorrect password.** Verify that registration failure status is correctly displayed in web interface and UUT does not stall or hang	PASS
Reg 2	**Attempt registration of Ring Extension using incorrect username.** Verify that registration failure status is correctly displayed in web interface and UUT does not stall or hang	PASS
Reg 3	**Correctly register UUT Ring Extension** Verify that UUT registers properly and status is correctly displayed in web interface	PASS
Reg 4	**Register both UUT Ring and Page Extensions simultaneously** Verify that UUT registers properly and status is correctly displayed in web interface	PASS

Inbound Call - Ring Extension

The following test cases verify the Ring-detect feature of the UUT. Ensure Phone C is registered before beginning these tests. Through the server web interface, Phone C and ring extension has to be grouped together.

Test Case	Actions	Result
Ring 1	**Dial Phone C from Phone A and verify a simultaneous ring from UUT until canceled.** Verify that the UUT plays selected ring sound Verify that Phone C also rings Verify that the UUT continues to ring until the call attempt is canceled by hanging up Phone A	PASS
Ring 2	**Dial Phone C from Phone A and verify a simultaneous ring from UUT until answered.** Verify that the UUT plays selected ring sound Verify that Phone C also rings Verify that the UUT continues to ring until the call attempt is answered by Phone C	PASS

Test Case	Actions	Result
Ring 3	**Ring Phone C and verify simultaneous ringing from UUT. Do not answer call.** Verify that the UUT continues ringing until the call is dropped.	PASS
Ring 4	**Call Ring Extension by dialing Phone C and dial UUT Page Extension from Phone B.** Verify that the ringing stops and the paging call is established.	PASS

Inbound Call - Page Extension

The following test cases verify the inbound paging feature of the UUT. Phone C need not be registered for these tests.

Test Case	Actions	Result
Page 1	**Dial Page Extension from Phone A** Verify that UUT answers and a one-way audio page is established from Phone A to UUT. Verify that call is terminated by hanging up Phone A.	PASS
Page 2	**Dial Page Extension from Phone A and mute/un-mute call.** Verify that audio from Phone A is not heard through the UUT.	PASS
Page 3	**Dial Page Extension from Phone B while Phone A is already using the Page Extension.** Verify that the second call to the Page Extension receives Busy tone (UUT configured to allow only one simultaneous Page call)	PASS
Page 4	**Dial Page Extension from Phone A and maintain call for two minutes.** Verify that the call remains up after the Session Refresh (REINVITE) is sent.	PASS
Page 5	**Dial Page Extension from Phone A and answer, then Dial Ring Extension from Phone B.** Verify that the Paging call is continued without interruption (not dropped).	PASS

Serviceability

The following test cases verify the serviceability of the UUT. No other Phones need to be registered for these tests.

Test Case	Actions	Result
Serv 1	**Disconnect, then reconnect, the ethernet cable from the UUT.** Verify that the UUT registers with the SIP server after PoE-supplied power is restored.	PASS