Interoperability Certification

Algo 8028 SIP Doorphone / 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 8028 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 8028 SIP Doorphone	Firmware 1.3	

Algo 8028 SIP Doorphone

The Algo 8028 is a SIP compliant door phone. When registered with a SIP server, the doorphone can place a call to a pre-programmed extension, or receive (auto-answer) a call placed to the doorphone. The doorphone can also unlock a door using an integrated relay when a configured key is pressed.

Key features:

- SIP Support - Natively supports SIP and Voice over IP
- Single Pair Digital Station - a single pair 24 AWG connects the doorphone to the controller up to 1000ft (300m) away
- Security and Safety - relay is located securely inside the premise within the controller
- Door Sensor & External Call Button - programmable inputs and outputs available on controller and door station
- Residential or Commercial - FCC & CISPR22 Class B approval for residential use easy to retrofit
- 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.

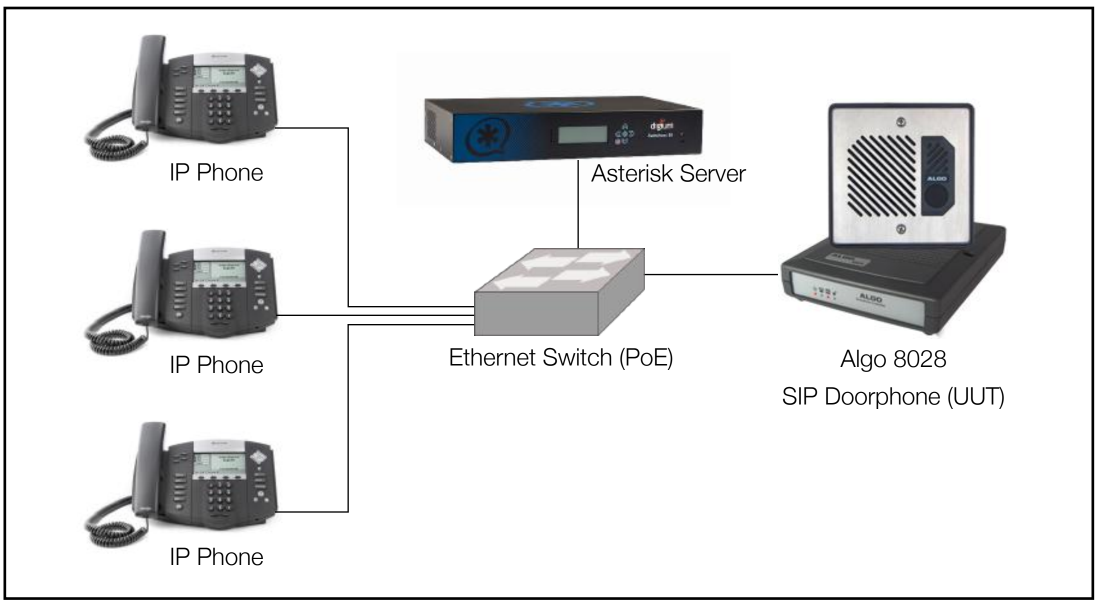
Feature	Algo 8028 SIP Doorphone
SIP Registration	✓
Outbound Call	✓
Inbound Call	✓
Serviceability	✓

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 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
Extension: doorphone
  Auth ID: doorphone
  Password: algotest123
Dialing Extension: 7003
```

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

/etc/asterisk/sip.conf

```
[doorphone]
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
```

/etc/asterisk/extensions.conf

```
[default]
exten => 7000,1,Dial(SIP/doorphone)
exten => 7003,1,Dial(SIP/phonea)
exten => 7004,1,Dial(SIP/phoneb)
```

Definitions

- **UUT** - Unit Under Test
- **Server** - Asterisk acting as a SIP back-to-back user agent
- **Phone A, Phone B** - a SIP compatible endpoint used to place and receive calls

Tests Performed

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 Extension using incorrect password.** Verify that registration failure status is correctly displayed in web interface and UUT does not stall or hang	PASS

Test Case	Actions	Result
Reg 2	**Attempt registration of 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 Extension** Verify that UUT registers properly and status is correctly displayed in web interface	PASS

Outbound Call

The following test cases verify the outbound calling capability of the UUT.

Test Case	Actions	Result
Out 1	**Press the call button on the UUT to call Phone A. Answer Phone A.** Verify that a two-way audio call is established	PASS
Out 2	**Press the call button on the UUT to call Phone A. Do not answer.** Verify that the call is canceled after 5 rings (30 seconds)	PASS
Out 3	**Press the call button on the UUT to call Phone A. Answer Phone A.** **While call is established, call the UUT from Phone B.** Verify that Phone B receives busy tone, while Phone A call continues uninterrupted.	PASS

Inbound Call

The following test cases verify the inbound calling capability of the UUT.

Test Case	Actions	Result
In 1	**From Phone A, call UUT.** Verify that a two-way audio call is established	PASS
In 2	**From Phone A, call UUT and mute/un-mute call.** Verify that audio from Phone A is not heard through the UUT.	PASS
In 3	**From Phone A, call UUT to establish a call.** **While call is established, call the UUT from Phone B.** Verify that Phone B receives busy tone, while Phone A call continues uninterrupted.	PASS
In 4	**From Phone A, call UUT and maintain call for two minutes.** Verify that the call remains up after the Session Refresh (REINVITE) is sent.	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 network connection is restored.	PASS
Serv 2	**Disconnect, then reconnect, the power cable from the UUT.** Verify that the UUT registers with the SIP server after PoE-supplied power is restored.	PASS

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