Interoperability Report

Panasonic
KX-UT670, KX-UT248

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1. STATEMENT OF LIABILITY

This document describes test setups, configurations, test plans, and test results that Outsource Testing, Inc. performed to determine the level of interoperability between the named Digium products (Asterisk 10) and two Panasonic SIP units (KX-UT670, KX-UT248). This document does not necessarily describe all features or usage scenarios of the products; only those which are essential for basic interoperability. These tests typically are of a base functional nature and in no way represent dynamic, stress, or performance tests. Thus, these tests may not be representative of "real-world" conditions you may encounter. Outsource Testing, Inc. has made reasonable efforts to ensure that the information contained in this document is accurate at the time of its release, for the versions of each product outlined in the Executive Summary. However, since products are often revised over time, Outsource Testing, Inc. cannot guarantee accuracy of the information contained herein after the date of release of this document. This document has been prepared for use by professional and properly trained personnel, and the user assumes full responsibility when using it.

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2. EXECUTIVE SUMMARY

This document covers the setup and the tests used to validate the interoperability of two Panasonic SIP Phones with Digium's Asterisk 10. All relevant information is included in order to allow the replication of these scenarios.

2.1. PRODUCTS TESTED

Asterisk Business Edition C has been thoroughly tested against the named devices below. The software versions for all tested products are included.

2.1.1. ASTERISK VERSION

Product	Version
Asterisk	**10**

2.1.1. UNIT UNDER TEST (UUTS)

Product
Panasonic - KX-UT670
Panasonic - KX-UT248

2.1.1.1. Panasonic – KX – UT670

Executive SIP Phone (Smart Desktop Phone)
Key Features

- Advanced 7 Inch Color Touch Screen Display
- High Quality Wideband Voice

- Camera View (H.264)
- 2 x GbE ports, PoE
- 6 SIP Accounts
- Full Duplex Speakerphone
- EHS

2.1.1.2. Panasonic – KX – UT248

Executive Desk Phone
Key Features

- 4.4 Inch Grayscale Graphical Screen
- High Quality Wideband Voice
- 3 Way Conference Call Support
- XML Application Interface
- Built in Bluetooth
- EHS
- 2 x GbE ports, PoE

2.2. SUMMARY OF TEST FOCUS

A summary of the test results is provided below. Detailed tests and results are available in Section 4.

2.2.1. FEATURE MATRIX

Feature	UT670	UT248
SIP Register	✓	✓
Outbound Call	✓	✓
Inbound Call	✓	✓
Call History	✓	✓
Hold and Resume	✓	✓
Conferencing	✓	✓
Codec G.729		
Codec G.722	✓	✓

3. TEST CONFIGURATION

This section describes the test configuration and setup. A diagram has been provided in section 3.2.

3.1. DESCRIPTION OF TEST SETUP

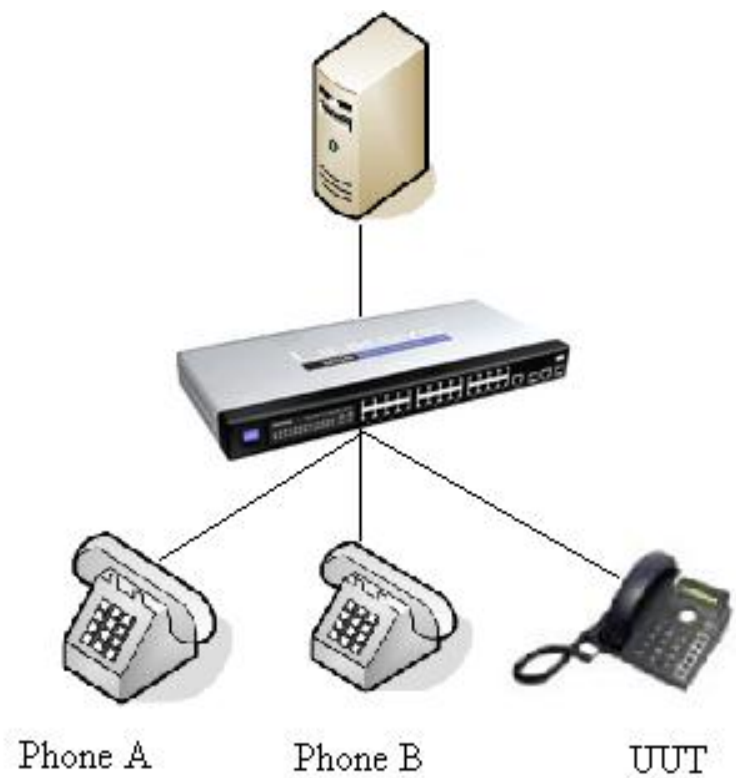
An isolated test network was created using a Linksys SR224G switch and a PC-based server running Asterisk 10. The unit under test was connected to the network via the Lynksys switch. Each feature listed in section 1.2 was tested according to the cases listed in section 4.

In addition to running the Asterisk 10, the server was also responsible for DNS and assigning DHCP.

3.1.1. SWITCH

Vendor	Product
Lynksys	SR224G

3.2. TEST SETUP DIAGRAM



4. TESTS PERFORMED

Included in this section are the tests performed for each device.

4.1. REGISTRATION

4.1.1. Test 1 - SIP Registration

Summary – This test will ensure the device will register successfully with the Asterisk Server.

Steps –

1 – Create profile in Asterisk GUI using MAC address
2 – Assign to Line 1
3 – Re-register phone

Expected Result –

1 – Phone should Provision
2 – IP Address should be assigned
3 – Phone should register

Phone should auto provision and register with the Asterisk Server.

Step	UT670	UT248
Profile Created and assigned to Line 1	Pass	Pass
IP Address Assigned	Pass	Pass
Registration Verified	Pass	Pass

4.2. BASIC INTEROPERABILITY

4.2.1. Test 2 - Outbound Calls

Summary – This test will verify the ability of each phone to make an outbound call

Steps –

1 – Dial an extension set for phone A
2 – Verify UUT receives ringback
3 – Verify phone A receives correct caller ID
4 – Repeat steps 1 – 3 for phone B

Expected Results –

1 – The UUT should receive ringback and the call should connect
2 – The line on the phone should display as busy.
3 – The call should be recorded in the call history as an outbound call.

Step	UT670	UT248
Dial Phone A		
Successful Connection	Pass	Pass
Verify Ringback	Pass	Pass
Verify Caller ID	Pass	Pass
Verify Call History	Pass	Pass
Dial Phone B		
Successful Connection	Pass	Pass
Verify Ringback	Pass	Pass
Verify Caller ID	Pass	Pass
Verify Call History	Pass	Pass

4.2.2. Test 3 – Inbound Calls

Summary – This test will verify the ability of each unit to receive an inbound call

Steps –

1 – Dial from Phone A to extension assigned to UUT
2 – Verify Ringback
3 – Verify correct Caller ID is displayed on UUT

Expected Results –

1 – Call will be received successfully
2 – The callers should receive full duplex Audio
3 – Caller ID will be received successfully
4 – Ringback will be given to the calling party

Step	UT670	UT248
Dial UUT from Phone A		
Successful Connection	Pass	Pass
Full Duplex Audio	Pass	Pass
Verify Caller ID	Pass	Pass
Verify Ringback	Pass	Pass
Verify Call History	Pass	Pass
Dial UUT from Phone B		
Successful Connection	Pass	Pass
Full Duplex Audio	Pass	Pass
Verify Caller ID	Pass	Pass

Step	UT670	UT248
Verify Ringback	Pass	Pass
Verify Call History	Pass	Pass

4.2.3. Test 4 – Hold and Resume

Summary – This test will verify the ability of a user to place a call on hold and then resume the call

Steps –

1 – Place a call from Phone A to the UUT
2 – Place the calling party on hold
3 – Place a call from the UUT to Phone B
4 – Disconnect the call to Phone B
5 – Resume call placed from Phone A

Expected Results –

1 – A full duplex voice path should be established between Phone A and UUT
2 – When placed on hold, Phone A should get MoH
3 – A full duplex voice path should be established between Phone B and UUT
4 – The new call is dropped and the original call is resumed
5 – Call history should show both the inbound and outbound calls in the test.

Step	UT670	UT248
Dial UUT from Phone A		
Successful Connection	Pass	Pass
Full Duplex Audio	Pass	Pass
Music on Hold Received by Phone A	Pass	Pass
Outbound Call Successful Connection to Phone B	Pass	Pass
Successful disconnect from Phone B	Pass	Pass
Successful resumption of call from Phone A	Pass	Pass
Call History Verified	Pass	Pass

4.2.4. Test 5 – Attended Call Transfer

Summary – This test verifies the functionality of an attended call transfer

Steps –

1 – Place a call from Phone A to the UUT
2 – Answer call from Phone A and place on hold
3 – Dial the Extension for Phone B
4 – Once connected, press transfer again

Expected Results –

1 – A voice channel is established between phone A and UUT
2 – A voice channel is established between phone B and UUT
3 – After the transfer, the Phone A and Phone B should have a voice channel established
4 – Call History for all three phones should register. An inbound call for UUT and Phone B and an outbound call for Phone A

Step	UT670	UT248
Dial UUT from Phone A		
Successful Connection	Pass	Pass
Phone A placed on Hold and receives MoH	Pass	Pass
Outbound Call Successful Connection to Phone B	Pass	Pass
Successful Transfer of Call from Phone A	Pass	Pass
Call Histories Verified	Pass	Pass

4.2.5. Test 6 – Unattended Call Transfer

Summary – This test verifies the functionality of an unattended call transfer

Steps –

1 – Place a call from Phone A to the UUT
2 – Before call is picked up, press transfer
3 – Dial the Extension for Phone B
4 – Once connected, press transfer again
5 – Verify connection between Phone A and Phone B

Expected Results –

1 – A voice channel is established between phone A and UUT
2 – Phone A is connected to Phone B
3 – After the transfer, the Phone A and Phone B should have a voice channel established
4 – All Lines on UUT show not in use after the transfer of the call.

5 – Call History for all three phones should register. A missed call for the UUT, and inbound call for Phone B and an outbound call for Phone A

Step	UT670	UT248
Dial UUT from Phone A		
Successful Connection	Pass	Pass
Call Transferred to Phone B Successfully	Pass	Pass
Call Histories Verified	Pass	Pass

4.2.6. Test 7 – Conferencing

Summary – This test verifies the functionality of a call with more then two parties connected

Steps –

1 – Place a call from UUT to Phone A
2 – After Phone A picks up, place call on Hold
3 – Dial the Extension for Phone B
4 – Once connected, press conference
5 – Verify connection between UUT, Phone A and Phone B

Expected Results –

1 – A voice channel is established between phone A and UUT
2 – A voice channel is established between phone B and UUT
3 – After pressing "Conference" the UUT, Phone A and Phone B should have a voice channel established
5 – Call History for all three phones should register. Two Outbound call from the UUT, and one inbound call from the UUT for Phone A and Phone B

Step	UT670	UT248
Dial UUT from Phone A		
Successful Connection to Phone A	Pass	Pass
Phone A successfully Placed on Hold	Pass	Pass
Successful Connection to Phone B	Pass	Pass

Step	UT670	UT248
UUT, Phone A and Phone B successfully Connected	Pass	Pass
Call Histories Verified	Pass	Pass

4.2.7. Test 8 – Call Forwarding

Summary – This test verifies the functionality of a call with more then two parties connected

Steps –

1 – Place a call from Phone A to UUT to verify the voice path
2 – On UUT, select forwarding from device menu and enter and enable to extension for Phone B
3 – Place a Call from Phone A to UUT
4 – On UUT, disable forwarding
5 – Re-verify voice path to UUT

Expected Results –

1 – UUT rings after step 1
2 – After Step 3, Phone B should ring directly
3 – After Step 5, Phone A connects to UUT
4 – Call History should show

Step	UT670	UT248
Dial from Phone A to UUT		
Successful Connection to UUT	Pass	Pass
Successful Connection to Phone B	Pass	Pass
Successful re-connection to UUT	Pass	Pass
Call Histories Verified	Pass	Pass

5. CONCLUSION

All basic functionality, including registration, was confirmed using the above configurations and tests. As a result of the tests completed, both model SIP phones met the specifications for basic functionality with the Asterisk Version 10.