THE JOINT FOIP TASK GROUP OF THE

INTERNATIONAL INTERCONNECTION FORUM FOR SERVICES OVER IP

(i3 FORUM)
(www.i3forum.org)

AND

THE SIP FORUM
(WWW.SIPFORUM.ORG)

Workstream "Technical Aspects"

Test Report
Fax over IP service

(October 2011)

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1 Background and Purpose

Fax over IP (FoIP) is now quite reliable when used in intra-enterprise applications and in carrier networks where the calls only transit the networks of IP-carriers with experience in handling FoIP calls. But FoIP, when used in international calls that use tandem-carrier connections and routes determined by existing technologies (SS-7 and voice-based least-cost routing (LCR)), exhibits error rates well above legacy fax over the PSTN. International FoIP must, therefore, have significant testing and improvements based on that testing before the transition to an all-IP global network can be completed.

The I3 Forum and SIP Forum FoIP Task Groups have joined forces to perform unprecedented testing of FoIP in international interconnect using existing routing technology to determine the success rate of such connections, to find routing and interoperability problems, describe and explain the most frequent errors, and offer solutions to the industry to bring FoIP transaction-success rates on a par with facsimile over the traditional PSTN. This report presents the results of that Phase I testing campaign. (Phase II testing is underway in 4Q2011.)

1.1 Voice and Fax Differences

When evaluating FoIP, it is helpful to keep in mind the differences between FoIP and VoIP. The global telecom community, seeing the enormous efficiencies and scale of moving TDM voice to VoIP, naturally began with voice. Fax was a subject left for a later chapter in the IP book and today, voice is the assumed medium for SIP calls, meaning that to go to a fax-specific protocol (e.g. T.38) the medium must transition away from voice.

Not only was fax deferred, compared to fax, voice was simple: agree on a codec, manage the jitter buffer, get rid of echo…you're done. Call set-up timing is not critical. After all, the usual endpoint is a very forgiving human ear and brain.

But with fax, timing is everything. Instead of a human, we have two computer-based terminals that must successfully execute T.30, a computer-to-computer protocol with relatively tight timing. Synchronous modems, which don't work well with dropped packets, are usually involved. And then there is the transition from voice to T.38. Timing is everything, and in tandem networks timing becomes looser with each network transited.

But during the 12-year FoIP-carrier incubation period, the global IP networks steadily improved. Dropped and out-of-order packets are rare in heavily industrialized routes. So now, the industry's carriers and equipment vendors are ready to begin the move to carrier-based FoIP.

The testing reported here is an important early step in this network evolution.

1.2 Phase I Testing

Phase I testing was performed by a total of 14 members of the two organizations. Two members of the SIP Forum FoIP Task Group provided and installed (for no charge) FoIP fax servers at 12 different i3 Forum member sites and two SIP Forum sites. The tests consisted of each tester sending faxes to all other testers. The destination addresses for each participant were the server, with a SIP: address, and a TDM-connected fax terminal with an E.164 address. In some cases, faxes were sent over the open Internet from server to server, bypassing carrier routing in order to establish a baseline of non-carrier-based routing performance.

Success rates varied from virtually 100% for the non-carrier Internet-based FoIP calls to 50% for many of the SS7-based international routes. The use of standard LCR appears to be the cause of most of the problems. For example, some initial calls from a test system in Chicago (USA) to Australia failed because the call was routed to be least cost for voice. The service provider rerouted the call to a different carrier resulting in 100-percent success for the route.

But, in general, the success rate varied considerably in different testing sessions and from country to country. Much of the variability appeared to be random, but it is likely that LCR algorithms routed the calls differently depending on the traffic load and intermediate carriers selected. The primary conclusion of Phase I testing was that the ability to determine and improve the call routing is required for success in international FoIP.

1.3 The Need for Phase II

But additional testing is required to determine the specific reasons for a chosen route to have problems, enabling the carriers to adjust their routing algorithms for FoIP success, and to, possibly, take remedial steps to improve the faulty routes. This requires that for Phase II testing, the test methodology must be changed from broadcasts of one-to-many to paired testing to determine the root cause of the identified problems.

2 Testing Configuration

For the Phase I testing campaign, a SIP Forum FoIP Task Group member installed its enterprise fax-server, which had a fax-broadcast feature that provides the reports needed to determine if the FoIP connection was successful and, if not, the nature of the error of a particular call. The server supports both T.38 and G.711 pass-through fax terminations, which is a required feature to support the contemplated tests.

Each tester was required to:

- Install the software on a Windows machine and,
- Connect the server to an IP network to enable international testing,
- Provide a PSTN-connected fax terminal
- Provide route-determination capability

Each participant published to the group two E.164 fax numbers, one for the testing fax server and one for the PSTN-connected fax terminal. The E.164 numbers were loaded into the fax-server phonebook so that the test could run automatically. Broadcast logs and Wireshark captures were collected and made available.

3 Testing Method

Tests were performed as follows:

- Prior to each session, the most up-to-date list of E.164 numbers and SIP URIs were sent to all participants as a phone book, together with the test image selected for that session. The test-image file contained information necessary for the server to setup connections, filename of facsimile pages that were to be sent, number of pages, identification information sent to the receiver etc.
- On "Test Tuesday" of each week of the testing campaign all tests participants automatically sent faxes to all other fax servers and to all PSTN fax machines according to the time schedule.
- The fax-broadcast log files and Wireshark capture files of all outgoing connections were saved and made available for analysis.
- After each testing session the analysis of the results were sent to all test participants.

The testing campaign extended from January 2011 through April 2011. During the campaign, there were 14 testing sessions, but not all participants participated in each session. In addition to the SIP Forum vendor testers, the testing group contained major carriers from Europe, USA, Canada, Australia, and Hong Kong.

4 Observations and Preliminary Recommendations

1. Eliminate all non-standard audio attributes (e.g. Cisco NSE).
2. Consider implementing V.152 for non-T.38 calls (see 10.2 of V.152) (echo cancellation, VAD switch off and jitter buffer set to fixed value).
3. Eliminate network elements that block RTP media when fax tones are detected.
4. A common problem is missing ACKs.
5. Re-INVITES changing an active fax session should be ignored.
6. Network elements in carrier interconnection networks should be transparent to the transmitted payload.
7. Receiving gateway should mute RTP channel in both direction as soon as it detects V.21 preamble to avoid T.30 negotiation in audio mode (ITU-T.38 2010/09 D.2.2.4.2.)
8. Attention must be paid to UDP redundancy.
9. PCM integrity was the cause of many failures. We suspect that much is due to peering-point problems. We recommend that this be investigated in Phase II.
10. Re-INVITE delay should be as short as possible.
11. Re-INVITES changing active fax session should be ignored.
12. Unexpected BYEs should be investigated in Phase II testing.
13. Media are sometimes blocked in interconnection segment in one or in two directions or terminals do not transition to T.38, even if T.38 is mutually negotiated. (Phase II investigation)
14. Sometimes strange RTP version 0 is sent by the receiving side and should be investigated in Phase II.
15. T.38 offer in initial INVITE seems to cause risk of failure in interconnection
16. Problems in initial T.38 calls appear also when audio is offered together with T.38.

5 Summary

To improve FoIP reliability, it is indispensable to find and eliminate the reason of the most frequent errors identified and described in the detailed part of the report. For each carrier that delivered log and PCAP files, there are summary files with descriptions and examples. For

Phase II testing, carriers will work in pairs tracking the FoIP calls with identified errors and determine the full routing information. By comparing the tracks from calling and called terminal and by tracking the route, it should be possible to find the responsible device in the network and to fix the failure.

Finally, the descriptions of the found reasons will be published in guidelines that will help to eliminate the problems with FoIP calls in international IP interconnect.