Preliminary

TsLinkNet vs. libpri Asterisk Bulk Call Test White Paper

Introduction

As use of IP-PBXs, both open source and commercial, has spread in recent years to provide PBX telephone services to small to medium size companies, there has been an increasing demand for gateway systems like TsLinkNet that work seamlessly with the SIP/RTP interface of the IP-PBX to provide multiple, reliable connections to the TDM/PSTN world. This is important because TDM/PSTN network interfaces like ISDN will continue for many years to be a key (today, in fact, the dominant) network interface for end-users, fixed & mobile.

Asterisk is an increasingly popular open source IP-PBX system, and there are many similar IP-PBX products, open source and commercial, that use DAHDI-compatible hardware (e.g., T1/E1/ISDN PCIe boards) to interface to the TDM/PSTN.

This white paper examines the performance and reliability differences between the "libpri" open source ISDN PRI add-on to Asterisk vs. the TsLinkNet from TeleSoft International, Inc. to provide the Asterisk connection to the TDM/PSTN world using two DAHDI internal PCI 4-span T1/E1 cards for a total of 8 T1/E1 spans under test. Although TsLinkNet is run on the same server as Asterisk in these tests, TsLinkNet can be run on a separate server allowing for the possibility of load sharing across servers as needed.

Glossary

Asterisk - An open source IP-PBX.

DAHDI – Digium Asterisk Hardware Driver Interface – also represents a family of DAHDI driver-compatible T1/E1 PCI cards, analog phone PCI cards, etc. available from several manufacturers and distributors.

IP-PBX – Internet Protocol-Private (Telephone) Branch eXchange – uses UDP/TCP/IP/SIP with ISDN/PSTN/TDM optional.

ISDN – Integrated Services Digital Network – comes in Basic Rate (BRI) and Primary Rate (PRI) flavors, with 2 voice channels for BRI and 23 to 30 voice channels per PRI line (also known as a span). Within PRI, there are two major variants: N. America/Japan with 23 voice channels per T1 (1.544 Mbit/sec) physical layer line and the rest of the world 30 voice channels per E1 (2.048 Mbit/sec) physical layer telephone line.

Libpri – An open source PRI software.

PSTN – Public Switched Telephone Network (e.g., analog, ISDN and T1/E1 telephone lines offered by AT&T, BT, NTT, etc.) PSTN also includes country-specific non-ISDN T1 and E1 lines which predate ISDN and offer 24 and 30 voice channels per line respectively.

SIP/RTP – Session Initiation Protocol/Real-Time Protocol – Voice-Over-IP (VoIP) Protocols.

TDM – Time-Division-Multiplexed communication. Connections that share the same physical telephone line by allowing each voice connection a fixed fraction of each second dedicated to communicate compressed voice data for that call.

TsLinkNet - TsLinkNet IP-PBX Gateway is a software solution that takes the packetized SIP messages from the IP-PBX and translates them to the TDM (Time Division Multiplexed) messages used on the worldwide digital and analog PSTN networks. While most applications will require PRI ISDN, other options include support for QSIG (the standard interface for PBX-to-PBX communications).

Test Configurations

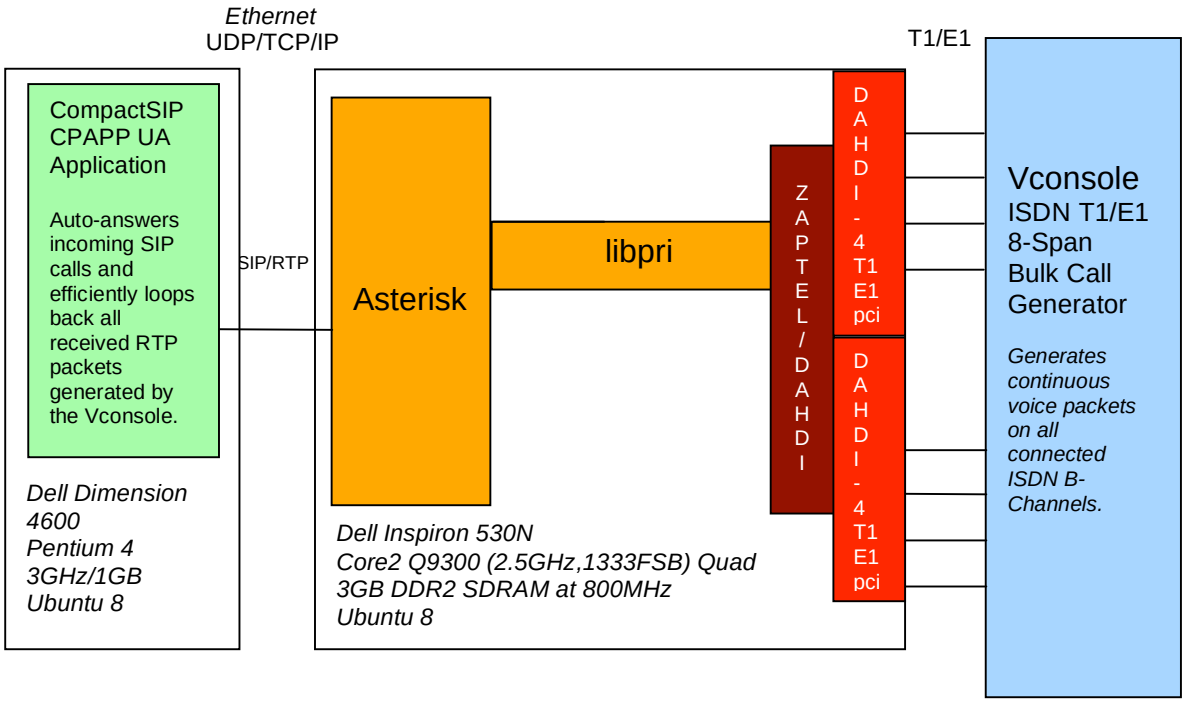

Figure 1 libpri Configuration

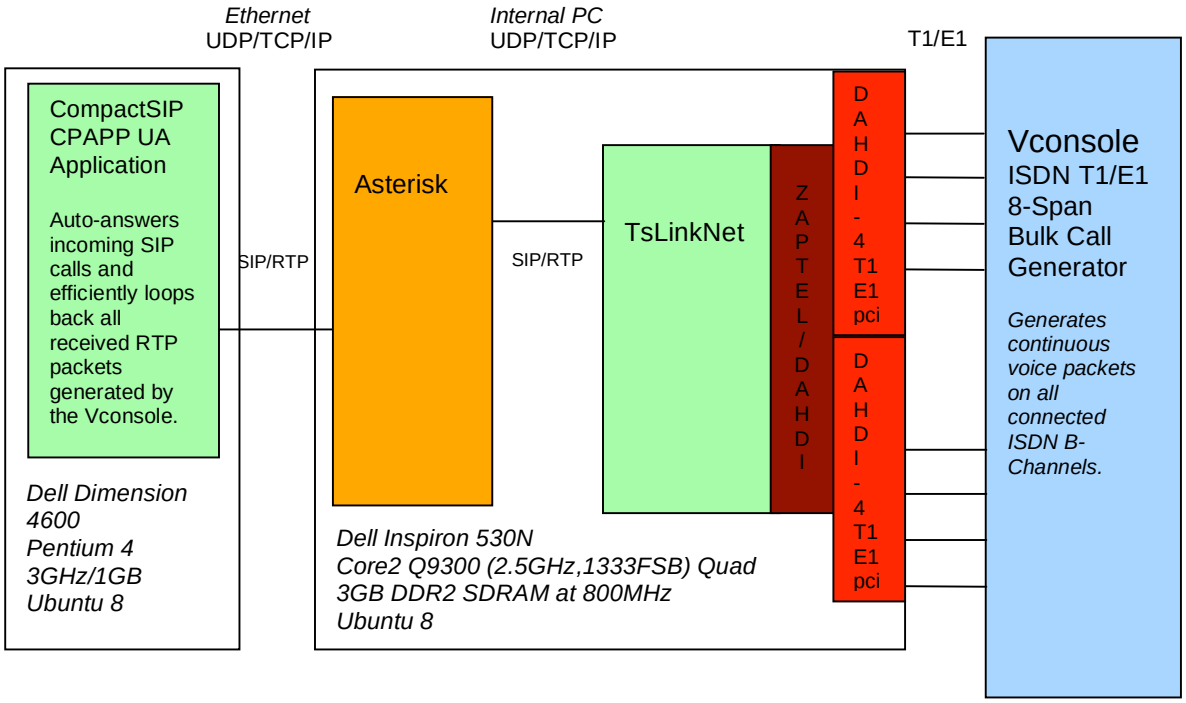

Figure 2 TsLinkNet Configuration

Test Configuration

The same Asterisk IP-PBX system is used in all tests covered by this white paper. Similar results are expected with many other IP-PBXs because TsLinkNet allows the total system to be partitioned cleanly between two processes which can be run efficiently on separate processors. In contrast, libpri complicates an already complex Asterisk runtime environment by running as part of Asterisk and makes Asterisk much more sensitive to the real-time latency requirements of DAHDI-compatible T1/E1 hardware.

For the tests described in this paper, the Vconsole Bulk Call Generator was selected since it can support 8 E1 spans for a total of 240 simultaneous calls. The Vconsole unit was configured to originate the number of calls per second indicated below until the maximum number of calls were connected simultaneously for the spans being tested (e.g., 240 calls for 8-spans). Then no new calls were set up for a few seconds before the Vconsole started the cycle again by tearing down the specified number of calls/sec as it also originated the specified number of new calls/sec in the same second.

The following tests were run:

Test No.	E1 Spans	ISDN Interface	Configuration	Calls/sec	Max calls possible
1	8	Libpri	Figure 1	3	240
2	8	TsLinkNet	Figure 2	3	240
3	8	Libpri	Figure 1	1	240

General Test Overview

Figure 1 applies shows the network configuration under test when libpri provides the connection to the ISDN side. Figure 2 shows the network configuration under test when TsLinkNet provides the connection to the ISDN side. The tests generate calls for both libpri and TsLinkNet from the ISDN side.

As shown in Figures 1 and 2, ISDN calls are generated by the Vconsole Bulk Call Generator on up to 8 E-1 spans. The Vconsole unit is widely used for ISDN testing.

On the other end of the test, the "cpapp" test application from TeleSoft was used. Cpapp is a simple command line application which was configured for all tests in this white paper to auto-answer any incoming SIP call from Asterisk and then echo any RTP voice packets that are sent by Asterisk during the length of each call. In these tests, cpapp does not teardown any calls, but rather waits for Asterisk to do so.

Asterisk kept each call alive until disconnected from the Vconsole side.

Voice quality (VQ) was graded on a scale of 1-5:

1	Unintelligible
2	Intelligible but badly distorted
3	Some degradation
4	Near toll quality
5	Toll quality

Test 1 – libpri 8-Spans – 3 calls/sec

For the 8-span test, the Vconsole unit was configured to continuously generate three (3) EuroISDN calls per second (each with an 88 second call duration) – with the expected result of building up to and then maintaining 234-240 calls active at any given time over a sustained period of time.

When libpri provided the ISDN interface as in Figure 1, the maximum calls reached (out of a possible 240) was 215 calls in repeated testing. During ISDN call generation/teardown by the Vconsole, libpri was unable to sustain a constant call volume and the number of active calls fluctuated by the second and varied anywhere from 215 to 177 calls over a minute or so.

VQ varied from 3 at 180 calls to 2 at 215 calls.

The Vconsole unit stopped generating new calls after about 500 total calls within a few minutes of test startup.

Overall, the libpri system behavior was unstable and erratic. The Vconsole unit eventually quit issuing new calls once it ran out of buffers because many buffers/calls had been lost/dropped by libpri. Many DAHDI "Bad FCS Received" and "Abort Received" were displayed in the Asterisk log, and cpapp reported multiple calls lost because ACK was never received in response to the SIP 200 message response sent by Asterisk.

Using Linux System Monitor, the total Asterisk/libpri processor usage was 18-20% to support the maximum of 215 calls.

Test 2 – TsLinkNet 8-Spans – 3 calls/sec

As before for libpri, for TsLinkNet for the first 8-span test, the Vconsole unit was configured to continuously generate 3 ISDN calls per second (each with an 88 second call duration) – with the expected result of building up to and then maintaining 234-240 calls active at any given time over a sustained period of time.

In contrast to libpri, when TsLinkNet provided the ISDN interface as in Figure 2, the maximum calls reached (out of a possible 240) was 240 calls in repeated testing. During ISDN call generation/teardown by the Vconsole, the number of active calls sustained was basically constant in a range from 234-236 for a period then to 240 for a period then back to 234-236 for a period with the same pattern repeated for days. The test was allowed to run over several days with over 1.6 million successful calls and no call failures.

VQ was 4 at 211 calls and 3 at 240 calls.

Overall, the behavior was stable, steady and predictable. The Vconsole unit never stopped issuing new calls (with no lost call/lost buffer problems).

No DAHDI "Bad FCS Received" and "Abort Received" were observed in the Asterisk log.

Using Linux System Monitor, the total Asterisk/TsLinkNet processor usage was 18-20% to support the maximum of 240 calls.

Test 3 – libpri 8-Spans – 1 call/sec

For this lower stress variation on the 8-span test, the Vconsole unit was configured to continuously generate 1 ISDN call per second (each with 340 second call duration) – with the expected result of building up to and then maintaining 239-240 calls active at any given time over a sustained period of time.

When libpri provided the ISDN interface as in Figure 1, the maximum calls reached (out of a possible 240) was 207 calls. During ISDN call generation/teardown by the Vconsole, libpri was unable to sustain a constant call volume and the number of active calls fluctuated by the second and varied from 207 to 167 over a few minutes.

VQ varied from 3 at 170 calls to 2 at 207 calls.

Overall, the behavior was again erratic, but the test was not run long enough for the Vconsole unit to quit issuing new calls. Many DAHDI "Bad FCS Received" and "Abort Received" were displayed in the Asterisk log, and cpapp reported several calls lost because ACK was never received in response to the SIP 200 message response sent by Asterisk.

Using Linux System Monitor, the total Asterisk/libpri processor usage was 18-20% to support the maximum of 207 calls.

Testing Summary

Results of the 8-span tests of [Asterisk + libpri] vs. [Asterisk + TsLinkNet]:

Test No.	E1 Spans	ISDN Interface	Calls/sec	Max calls possible	Medium number of calls (on way to max)	Max calls sustained	Call quality	Max calls b/f system reboot
1a	8	Libpri	3	240		215	2	0.5K
1b	8	Libpri	3	240	180		3	0.5K
2a	8	TsLinkNet	3	240		240	3	>1600K
2b	8	TsLinkNet	3	240	211		4	>1600K
3a	8	Libpri	1	240		207	2	0.5K
3b	8	Libpri	1	240	170		3	0.5K

Conclusions

In the Figure 1 test configuration, Asterisk running on the same processor as libpri is unable to utilize all 240 possible ISDN B-Channels even when the call setup rate was reduced to one call per second. Libpri's behavior is unstable, erratic and does not allow full utilization of available T1/E1 DAHDI hardware ISDN B-Channels.

In the libpri tests, the Vconsole unit quit issuing new calls once it ran out of buffers because many buffers/calls had been lost/dropped by libpri – one theory is that this is related to the inefficiency of libpri and the complexity of Asterisk trying to do its job with libpri's strong real-time demands added to the Asterisk load.

During libpri testing, many DAHDI "Bad FCS Received" and "Abort Received" errors were displayed in the Asterisk log, and cpapp reported multiple calls lost because ACK was never received in response to the SIP 200 message response sent by Asterisk.

In contrast, in the Figure 2 test configuration with Asterisk running on the same CPU as TsLinkNet testing showed none of these errors/exceptions.

One key to understanding the poor performance of libpri compared with TsLinkNet at moderate to high number of calls simultaneously active is to keep in mind that the load on the Asterisk system goes up rapidly as more voice calls are added. This is because each call generates 160 bytes of voice data every 20 milliseconds (50 times per second per active call) that must be read and written for each active channel whether Asterisk is busy doing other things or not.

Tests 1a and 2a show that TsLinkNet was able to support 33% more concurrent calls than libpri for the same call quality (3) and at the same CPU usage (18-20%). Similar differences are seen between TsLinkNet and libpri when comparing tests 1b vs. 2b, 2a vs. 3a and 2b vs. 3b when considering:

A) voice quality for about the same number of calls
B) maximum sustained number of calls.

It appears that once Asterisk reaches about 200 calls on 8 spans on the relatively powerful server used for this white paper, the complexity of Asterisk makes it difficult for the libpri subsystem to read/write the 160 bytes of voice data per call channel every 20 milliseconds on a consistent basis. This results in frequent losses of both voice data packets (worse voice quality at same number of calls as TsLinkNet), but also ISDN D-Channel packets exchanged with the ISDN network switch (e.g., the Vconsole) and lost calls.

By contrast, TsLinkNet allows full utilization of all 240 possible calls on all eight (8) spans tested, with acceptable voice quality under full load, and near toll voice quality under three-fourths load (i.e., 180 calls out of a max of 240). And, by allowing the ISDN-to-IP-PBX gateway function to be handled in a separate application, optimized for superior gateway performance, TsLinkNet provides a more efficient, more reliable and higher quality use of the T1/E1 hardware in Asterisk and similar IP-PBX systems than the libpri alternative.

Information in this document is preliminary and subject to change