IEEE TCSIM Newsletter

Issue 06 Sept 2010

Editorial Board

Chair's Message

By Dr. Dave Cavalcanti, Chair of TCSIM

Dear TCSIM colleagues,

Welcome to Q3 2010 Issue of the TCSIM newsletter, and once again I'd like to thank all the authors, reviewers and the editorial team for making the extra-effort in increasing the circulation frequency of the newsletter. As always we continue to encourage you to take advantage of this opportunity to share your work with other TCSIM members.

I'd like to take this opportunity to reinforce that we count on you to promote the TCSIM and attract more members. So please spread the word about TCSIM and the newsletter. Let us make this community more active, interesting and useful.

This issue of the TCSIM newsletter features one invited article about the implementation of wideband VoIP middleware in Embedded Systems.

Last, but not least, I'd like to remind you of the upcoming TCSIM elections. We have received several nominations, which are under review by the nominations committee. You'll find more details in this issue.

Hope you enjoy the reading and thanks for your collaboration!

Inside this issue:

New TCSIM mailing list

In order to facilitate communication and information sharing with TC members, a new TCSIM mailing list has been created using the IEEE Listserv system.

To send a message to the new TCSIM list, just send mail to TCSIM@LISTSERV.IEEE.ORG

If you are not currently subscribed to the list please send a message to *Prof. Kaushik Chowdhury* at krc@ece.neu.edu who is currently managing all the subscriptions to the new TCSIM list. You can also search for the TCSIM list at http://listserv.ieee.org/

We hope you can make use of this new list to share simulation related information with the TCSIM community.

Upcoming Events

IEEE SMARTGRIDCOMM 2010 – The 1st IEEE International Conference on Smart Grid Communications, October 4 – 6, 2010 National Institute of Standards and Technology (NIST), Gaithersburg, Maryland, USA
http://smartgrid.ieee.org/ieee-smartgrid-news/53-ieee-smartgridcomm-2010

(DS-RT 2010) 14th IEEE/ACM International Symposium on Distributed Simulation and Real Time Applications, October 17-20 2010, Fairfax, VA, USA
http://c4i.gmu.edu/events/conferences/2010/DS-RT/

(MASS 2010) IEEE The 7th International Conference on Mobile Adhoc and Sensor Systems, November 2010, - Intercontinental Mark Hopkins (tentative), San Francisco, CA (USA)
https://mass2010.soe.ucsc.edu/

TCSIM Elections

An election for TCSIM officers will occur sometime in Q4 2010. The candidates will be announced after nominations are closed (Sept 30). An electronic ballot will be set up and the TCSIM members will receive a URL to vote. **Note that you must be a CS and TCSIM member registered in the TECA system to vote.**

TCSIM Student Awards

"TCSIM supports students presenting ideas on exciting research frontiers through performance appreciation and travel awards."

TCSIM sponsors several Student Support Awards every year. The awards are intended to encourage and motivate student participation in conferences. The TCSIM awards are only given to students with accepted work in selected conferences. The awards selection process and distribution are fully managed by the organizing committees of the events.

Congratulations to the winners of the TCSIM sponsored student best paper award in SECON 2010:

- Lap Kong Law, paper: "On the Uplink Capacity of Hybrid Cellular Ad Hoc Networks"

Additional TCSIM awards in 2010 will be given during the following events:

- CGAMES 2010 (Student Best Paper Award)
- MASS 2010 (Student Travel Grants)

For more information on the 2010 TCSIM Student Awards, please visit the TCSIM webpage: http://tab.computer.org/tcsim

Implementation of Wideband VoIP Middleware in Embedded Systems

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Introduction

Although voice over IP (VoIP) has existed for several years, it has only recently begun to take off as a viable alternative to traditional voice systems and public switched telephone networks (PSTN). Interest in VoIP has grown in part because the technology can help both service providers and enterprises to reduce costs by using a single IP network for both data and voice applications. Cost is not the only factor driving VoIP's adoption. Growing experience and continued refinements in standards and technologies have led network managers to plan VoIP implementation as part of a more comprehensive communications transformation. In this context, VoIP can play a critical role in improving the effectiveness of a call center. Service providers are now leveraging VoIP technologies to introduce a wide variety of value-added applications and services for their customers.

In the early 70's Bell Systems invented and introduced the T1 digital trunk which employed an 8 bit uncompressed Pulse Code Modulation scheme with a sample rate of 8000 samples per second or 8 kHz. This allowed for a theoretical maximum voice bandwidth of 4000 Hz, slightly more than the PSTN (Public Switched Telephone Network) upper maximum of 3400 HZ. A T1 trunk carried 24 digital PCM (Pulse Code Modulation) channels multiplexed. The encoding scheme was later standardized as G.711 by the International Telecommunications Union (ITU) and is now commonly known in the United States as "U-law" and in Europe (where 30 channels are multiplexed) as "A-law" [1]. When the first Analog Telephone Adapters appeared, which allowed homes and small businesses to use the internet as the backbone for telephone communication, "U-law", "A-law" and the 8 KHz sampling rate became widely used in most implementations. Since G.711 can take up to 87 kbps of bandwidth which includes all the TCP/IP overhead many other codecs appeared which would compress the data rate to allow the possibility of many more calls on the backbone.

Asterisk is an open source private branch exchange (PBX) originally created by Mark Spencer of Digium. A PBX is a type of phone switch which allows multiple attached telephones to make calls to one another, and to connect to other telephone services including PSTN. Asterisk is widely supported in Linux, and it natively supports a wide range of telephony protocols, Voice over IP protocols including SIP, IAX and H.323. While most implementations of Asterisk are native to an ordinary computer, few have been ported to Analog Telephone Adapters or ATA. This implementation gives Asterisk functionality to a tiny, low cost, silent device, a very attractive choice for a small office or an individual. One of such implementations is the IP*x* family of IP-PBXs from Rowetel. The small *x* stands for the number of telephone ports supported either as interfaces to PSTN (FXO or foreign exchange office) or interfaces to an analog phone (FXS or foreign exchange subscriber).

In addition this implementation is open hardware, a similar concept to open source, where the hardware schematic can be obtained for free under the GPL license (GNU General Public License). While most dedicated ATAs are built on not too powerful embedded processors, the IP-01 is built on a low cost Blackfin DSP from Analog Devices capable of running at 400 Mhz which is enough processing power for processor intensive codecs, echo cancelation, and other PBX needed features. The Blackfin DSP is also supported by uClinux a derivative of Linux 2.x kernel intended for microcontrollers without Memory Management Units (MMUs).
When the IPx family was originally released, Asterisk was ported to this architecture by David Rowe of Rowetel with the support of a large open-source community dedicated to open source PBX software and Blackfin based uClinux [2]. While this port is very functional and stable it lacks some of the features that Asterisk has been evolving in the last few years. In particular, current port lacks support to:

- Built-in jitter buffer manipulation for hardware channels.
- Increased SIP performance
- Enhanced dial-plan functionality

- Integration to the G.722 (16Khz sampling capable) codec
- Improvements in the voicemail application
- Better support for call detail logging
- Support for the T.38 protocol designed for fax applications

Proposed Solution Approach

This work attempts to address those vital design factors and presents an approach that essentially creates new kernel level middleware. In particular, the approach is based on:

- Making a change in the IP01's open hardware by upgrading its telephony chipset to a newer version namely the Si3216. The Silicon Laboratories Si3216 ProSLIC is capable of wideband audio therefore a logical choice for upgrading to wideband ATA audio telephony devices.

- Writing a middleware layer of low level loadable kernel modules, which allow interfacing of Si3216 to the Asterisk PBX version 1.6.

The Si3216 contains a wideband codec which provides expanded audio band (50 Hz to 7 kHz). In comparison to other family member Si3210 or Si3215, the Si3216 PCM interface can be configured to support from 2 to 64 16 bit timeslots unlike the Si3210 or Si3215, which can only support 4 to 128 8-bit timeslots in each frame. The Si3216 is therefore capable to transmit a 16 bit PCM frame therefore doubling the rate of transmission. The schematic diagram for our proposed system is outlined in figure 1.

The proposed system has the following main components:

- Asterisk PBX software:
  - The PBX software, which allows through its interfaces (DAHDI and SIP or IAX2) communication with analog telephony system, and VoIP systems.

- uClinux loadable module for the FXS (foreign exchange subscriber):
  - Low level driver responsible with signaling between the Blackfin DSP and the telephony hardware. In this case the communication is performed through two channels: *SPI* (Serial Peripheral Interface) for controlling the device and *PCM* (Pulse Code Modulation) channel for audio data transfer. This module constitutes the first part of the proposed middleware layer.
- uClinux loadable module for DAHDI:
  - Generic interface between telephony devices and Asterisk. This interface was formerly known as Zaptel. This module constitutes the second part of the proposed middleware layer.

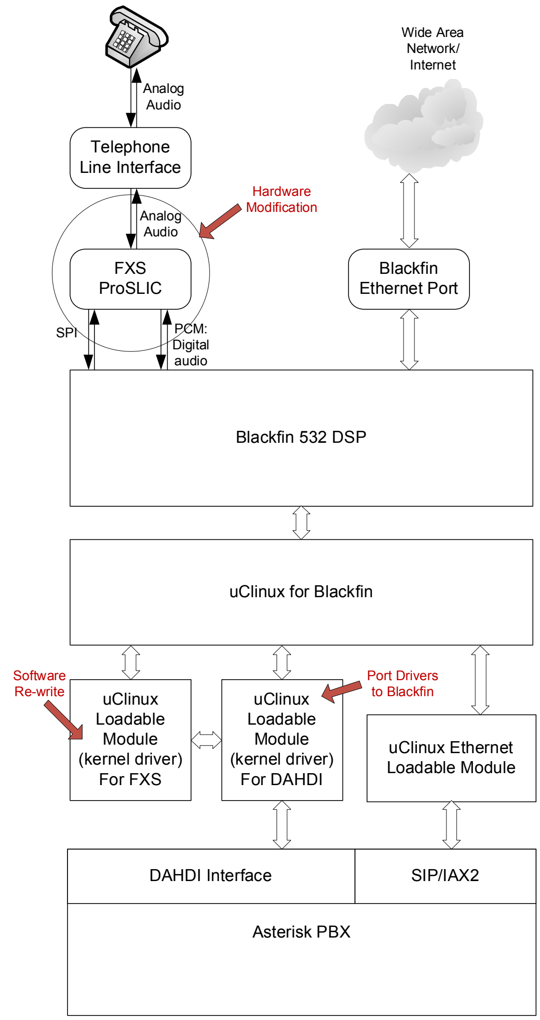

Figure 1. System Overview

This Zaptel driver was also ported to the Blackfin processor by Rowetel, this project is porting this driver it to the new signaling techniques available in the DAHDI interface. DAHDI stands for Digium Asterisk Hardware Device Interface. DAHDI is a high density kernel telephony interface for PSTN hardware which abstracts the hardware interface

- uClinux loadable module for Ethernet:
  - Kernel device driver for Ethernet communication. This is the standard Ethernet interface in uClinux

- uClinux module for Blackfin:
  - Embedded operating system customized for the Blackfin processor. uClinux is a Linux kernel specially designed for processors without a Memory Management Unit or MMU. It is a lightweight Operating System with very small binaries with built in connectivity and highly portable.

- Analog Devices Blackfin 532 Digital Signal Processor.
    - The Blackfin 532 Digital Signal Processor from Analog Devices is an advanced DSP capable of high performance of up to 600 Mhz speeds, which contain two 16 bits MACs, two 40 bit ALUs four 8 bit video ALU for faster video processing.
- FXS (foreign exchange subscriber) ProSLIC interface:
    - The interface is a daughter board that can be plugged in the IP 01 device. The interface was formerly based on the Silicon Labs Si3210/3215 chipset. In this paper, we modified this interface to the Silicon Labs Si3216 chipset.
- Blackfin Ethernet MAC.
    - The IP01 can be connected to Ethernet thru this port
- Telephone Line Interface:
    - Telephone Line Interface also present on the FXS device. The interface and isolation is accomplished by a Silicon Laboratories Si3210 chip

Experimental Testbed Environment

The experimental testbed is comprised of two parts hardware and software. The testbed's main hardware component is the IPx family hardware centers on the Analog Devices Blackfin 532 DSP. The IP01 is using uClinux as its operating system. Version 2008R1.5-RC3 has been chosen for the development environment due to its stability and its already existing support for the Blackfin family of processors. Asterisk 1.6.1.6 has been chosen as the PBX software due to its improved support to wideband and the numerous other features enhancements from Asterisk 1.4. Version 1.4 and below of Asterisk have used low level drivers for telephony devices called Zaptel.

In order for the driver upgrading to be successful the base DAHDI drivers had to be ported to uClinux Blackfin. The drivers can be compiled as either as part of the Linux kernel or as loadable modules. The later was chosen so that more control during development can be achieved. The DAHDI drivers chosen were dahdi-linux-2.2.0.2 and supporting DAHDI tools dahdi-tools-2.2.0. The drivers had to be compiled as part of the uClinux build image process so that all dependencies can be satisfied during compilation time.

Performance Analysis

An experiment was conducted to study the influence of the *jitter buffer* introduced with the DAHDI implementation while packets were lost and traffic performance was diminished. In this experiment, a Linux based tool called "wondershaper" has been used to shape the incoming TCP/IP traffic. Additionally, an automated SIP call-in server was setup by using the tool "sipomatic", which allows an automatic response when a SIP call is initiated. Few music files were re-encoded as 8000 samples/sec 16 bit wave file and configured as inputs to "sipomatic". This experiment is conducted using two stages. In the *first stage* of the experiment, the IP-01 was configured with Asterisk 1.4 and Zaptel. In the *second stage* of this experiment, the IP-01 was configured with Asterisk 1.6 and the DAHDI drivers. We noticed that at around 85 Kbits/sec, the wave file played with no hearable differences, further more the wave file played fine until the 75 Kbits/sec mark was reached.

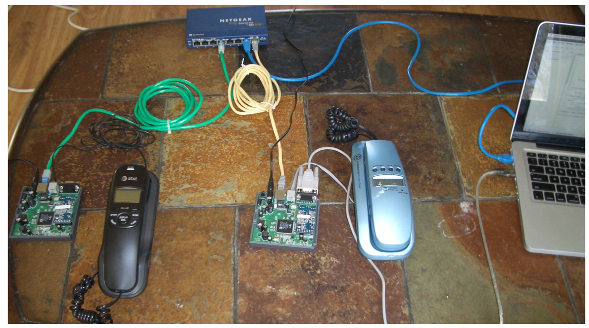

Figure 2 - Hardware Modified Test Bed for IP01

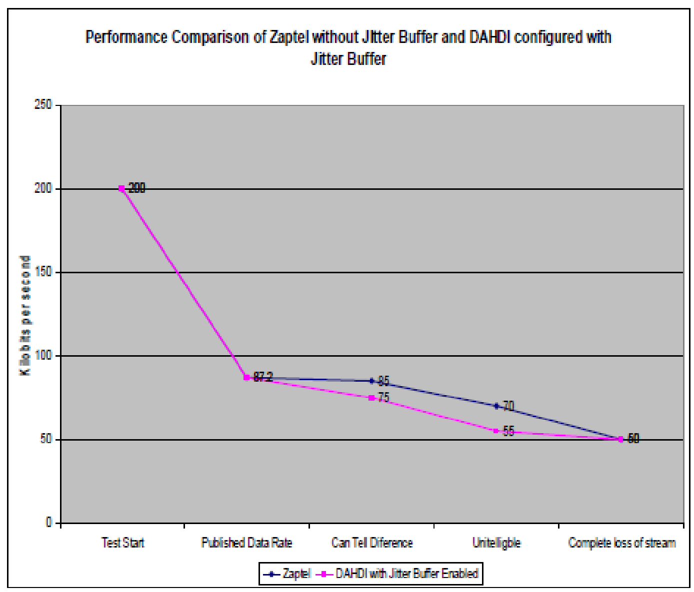

Figure 3. Jitter Buffer Comparison

From there short noticeable interruptions in the play of the wave file appeared. Finally, when the 55 Kbits/sec mark was reached, the stream could not keep up and the file became unintelligible. Figure 3 infers that Asterisk 1.6 with DAHDI drivers also outperforms Asterisk 1.4 with Zaptel drivers because of the added functionality, specifically in this example the jitter buffer capabilities.

References

[1] Voip-Info.org – A reference guide to all things VOIP (2008, June 5). Retrieved January 23, 2010 from http://www.voip-info.org/wiki/view/ITU+G.711.

[2] Free Telephony Project, Open Telephony Hardware, IP04 Four Port IP-PBX retrieved February 14, 2010 from http://rowetel.com/ucasterisk/ip04.html

Call for Papers – IEEE TCSIM Newsletter

The IEEE TCSIM Newsletters will publish short technical papers. The submissions should emphasize modeling, design, and analysis of computational methods for simulations and its applications in various areas, including, but not limited to, computer science, engineering, communications, and simulation applications.

Article submissions are encouraged throughout the year, though the deadline for the next quarterly newsletter is four weeks from its publication date. Submitted articles go through a quick peer review, and authors are notified of the result within three weeks.

The submissions are invited covering, but not limited to, the following topics:

- Simulation architecture modeling and prototyping
- Simulation algorithm design, implementation, and analysis
- Simulation complexity in computing
- Parallel and distributed simulation
- Design and usage of simulation tools
- Real-time simulation monitoring
- Simulation tools for communications and networks
- Simulation of computer systems and applications
- Agent-based simulation tools focus on the use of agents in engineering, human and social dynamics, military applications
- Simulation of ubiquitous networking and computing
- Simulation of transportation systems
- Automotive simulation applications
- Building and energy management simulations
- Machine learning
- Virtual reality systems
- Knowledge and data systems
- Systems optimization
- Web-based simulation and applications
- Department of Defense Architecture Framework (DoDAF)-based network simulations
- DoDAF-based vulnerability assessment

Submission

All papers must be submitted to elsaidm@gvsu.edu in four pages or fewer, including all figures, tables, and references. A manuscript submitted for publication should be original work that should not have been previously published and should not be under consideration for publication elsewhere. If an author uses charts, photographs, or other graphics from previously printed material, he/she is responsible for obtaining written permission from the publisher to use the material in his/her manuscript. The maximal number of figures and tables are five, and the number of reference is limited to ten.

Please submit electronically in DOC/PDF file, and ensure that the submitted file can be viewed in Acrobat Reader 9.0. A standard IEEE copyright release will also be required before full acceptance.

In the event that a particular cycle has large number of submissions, the editors reserve the right to schedule their publication in the subsequent editions of the newsletter.

All papers must include the authors' affiliation and e-mail addresses of all authors. All papers will be fully refereed for accuracy, technical content, and relevance. Contact Dr. El-Said at elsaidm@gvsu.edu with any questions concerning the paper submission and review process, or questions regarding the relevance of a paper to the IEEE TCSIM Newsletters.

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