USING DIGITAL SIGNAL PROCESSOR IN VOICE OVER IP COMMUNICATION

Marek Huczala

Department of Telecommunications, Brno University of Technology, Purkynova 118, 612 00 Brno, Czech Republic, e-mail: huczala@kn.vutbr.cz

The presented paper focuses mainly on the usage of the DSPs in the Voice over IP world. First chapter describes new evaluation methods of Quality of Services in Voice over IP communication. Radvision Voice over IP Toolkit as well as Telogy software arcitecture are discussed in later chapters. The described software packages and architectures are used in application development on different platforms including Texas Instruments and Intel Voice over IP devices. The last part of the paper will provide an outlook of the pros and cons of the presented solutions.

Keywords: DSP, VoIP, TCP/IP, quality of services.

1. INTRUDUCTION

Today, the Internet is a familiar term to millions of people all over the world. The popularity of the Internet and its pervasiveness in every day life has grown tremendously in the past several years. The most common and widely used of today's communication suite is TCP/IP (Transmission Control Protocol/Internet Protocol). Today's networks are carrying more data in the form of bandwidth-intensive, real-time voice, video and data, which stretch network capability and resources. The fast growth of the Ethernet communication protocol was a further stimulation element in even larger integration of voice and data networks.

I've lately been looking for new high-performance Ethernet hardware solutions for audio communication over the TCP/IP based network system. I've come accross a couple of solutions introduced by Texas Instruments, Intel and Radvision. I found out that some DSPs along with an appriopriate ethernet controller might be succesfully implemented in Voice over IP network architectures. It may eventually provide an excellent end-to-end solution for voice communication over an Ethernet network and might become an important player in the Voice over IP field.

However simple it may seem, the implementation of the DSP is not an easy task. The presented paper focuses mainly on the usage of the DSPs in the Voice over IP world. Radvision Voice over IP Toolkit as well as Telogy software arcitecture are discussed in later chapters. The described software packages and architectures are used in application development on different platforms including Texas Instruments and Intel Voice over IP devices. The last part of the paper will provide an outlook of the pros and cons of the presented solutions.

2. NEW VOICE OVER IP EVALUATION METHODS

In general, voice transmission over the Internet protocol (IP), or VoIP, means transmission of real-time voice signals and associated call control information over an IP-based network. Voice over IP communication could be affected in many ways. The main parameters that could affect the overall VoIP communication session are delay, delay variation, error and loss ratio as defined in ITU I.350.

Unfortunately, the metrics mentioned above are not reliable enough to describe the overall quality of end-to-end communication. This lead to the recent introduction of the rating factor (R-factor) evaluation. Unlike other evaluation methods, the R-factor also considers users' suggestive perception factors. The R-factor calculation has been built upon the E-model (G.107 specification) and it combines many parameters affecting the communication session, see (1). The R-factor conversion to MOS is also possible, but involves difficult mathemathic calculation.

The usual range of the R-factor is from 50 to 94 accordingly to the quality of communication.

$$\mathbf{R} = \mathbf{R_O} - \mathbf{I_S} - \mathbf{I_D} - \mathbf{I_{E\text{-}EFF}} + \mathbf{A} \tag{1}$$

R_0 – coefficient signal/noise.
I_S – the sum of all the devaluations during the transfer.
I_D – factor indicating the delay and jitter devaluations.
$I_{E\text{-}EEF}$ – factor indicating packet loss and equipment devaluations.
A – preferential factor.

A new management protocol RTCP XR (Real-Time Control Protocol), as recently published in RFC 3611, defines a set of metrics for evaluating end-to-end QoS in communication session. The RTCP XR is most like to be used in VoIP environments and it can be easily implemented as software integrated into IP phones and gateways. The protocol measures VoIP call quality using metrics such as packet loss and discard, delay, signal, noise and echo levels or call quality.

3. Radvision Voice over IP Toolkit

The Radvision Voice over IP Toolkit has been developed to help programmers in better development of VoIP applications on DSP pocessors at any kind. It contains RTP/RTCP and H.323 application programming sets.

The Real-Time Transport Protocol, or RTP, was designed to send real-time media such as voice and video over UDP/IP, though it can be used to transmit other types of data such as text and pointers. Since RTP can be configured for low latency, it is useful for interactive conversations as well as streaming media. RTP streams can be sent to unicast or multicast destinations.

The Real-Time Transport Control Protocol, or RTCP, is a companion protocol to RTP for gathering statistics on a media connection and information such as bytes sent, packets sent, lost packets, jitter, and round trip time. The application can use this information to judge the quality of its connections and make adjustments as required such as changing from a low compression codec to a high compression codec.

H.323 is the most mature and widely deployed IP communications protocol suite for real-time voice and/or video communication and is implemented in products ranging from infrastructure videoconferencing systems to IP phone chipsets. H.323 defines four primary entities and their functions in H.323 communication scheme. The set of H.323 entities consists of terminal, H.323 gateway, multiprotocol conferencing unit (MCU) and H.323 gatekeeper.

Implementations of Radvision VoIP Toolkit see later chapters.

4. TELOGY SOFTWARE ARCHITECTURE AND APPLICATIONS

The Telogy Software Architecture brings VoIP communication principles to the DSP world and TNEVx family of Texas Instruments processors respectivelly. The software provides a broad range of VoIP features available. Key Telogy Software capabilities include:

- Voice over IP
- Fax Relay
- Signaling
- Network Management

Voice over IP (VoIP) software processes voice samples for transmission over a data network. Its sub-components perform echo cancellation, voice compression (to conserve bandwidth), voice-activity detection (VAD), jitter removal and voice packetization {RTP → UDP (User Datagram Protocol) → IP, AAL (Asynchronous Transfer Mode, ATM) }.

Figure 1 describes the software architecture for gateway solutions. Each box represents a software component required to implement the features for voice, fax, modem, signaling, and network management functions.

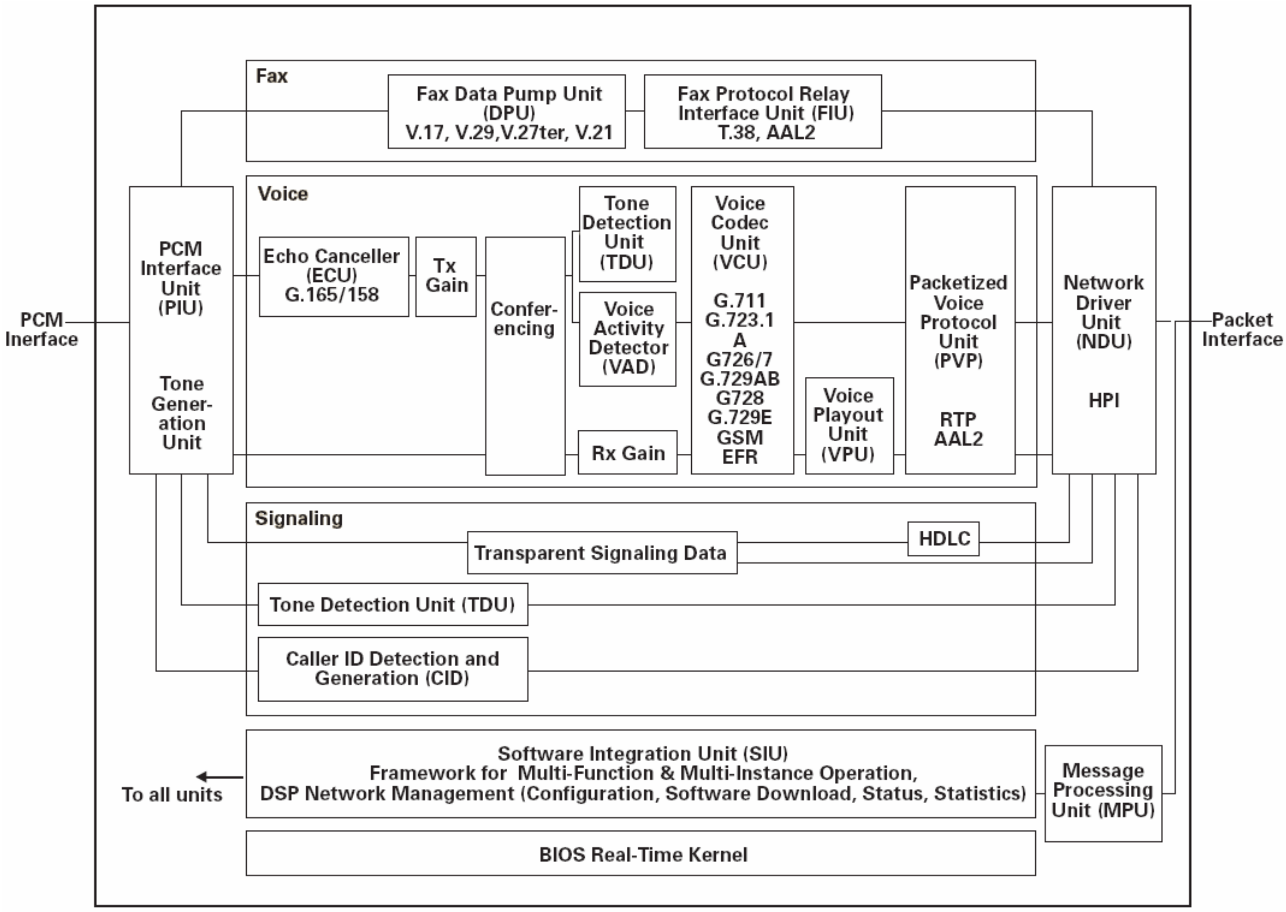

Fig. 1: Telogy Software Architecture.

Fax unit processes the incoming and outgoing fax connections between the clients. Fax relay provides real-time fax service between two analog fax machines over a packet network. Gateway platforms must support signaling for call establishment, in-band signaling and call termination. Both Channel Associated Signaling (CAS) and Common Channel Signaling (CCS) are supported.

4.1 Implementing Telogy Software Architecture

VoIP gateways solutions are available in two platform architectures to meet the needs of the devices. Texas Instruments offers several integrated solutions where voice processing and host processing are integrated into a single device (access communications processor): TNETV1010, TNETV1060 and TNETV2021.

These solutions should not only increase system integration and performance but also reduce power consumption and application space. When packet processing (host) is not needed on the same device as voice processing (DSP), stand-alone DSP-only solutions come into play: TNETV2402, TNETV2409, TNETV2510, TNETV2840.

Voice processing is usually done by C54x or C55x family of DSPs whereas the access comunications processor to integrate the device into existing network environment, is done by RISC processor architecture. All mentioned integrated solutions TNETV1010, TNETV1060 and TNETV2021 contain two internal ethernet media access controllers (EMAC) on board which provide as an interface to the outside world. Unfortunately, Physical layer devices (PHY) are most often put aside from the incorporated ethernet media access controller chips.

4.2 Connecting Physical Layer Device and Magnetics

Generally speaking physical layer device stands for an interface between MII (Media Independent Interface) and MDI (Media Dependent Interface) buses. The MII bus connects the PHY device to ethernet media access controller, namely to its EMAC module. The MDI bus links the PHY to the Magnetics and further to the RJ-45 socket.

The Intel LXT971A contains on board a single PHY supporting 10Base-T and 100Base-T, and the MII interface bus for connection to the EMAC controller. The XFMR block (magnetics) provides an interface between the LXT971A MDI bus and the RJ-45 socket.

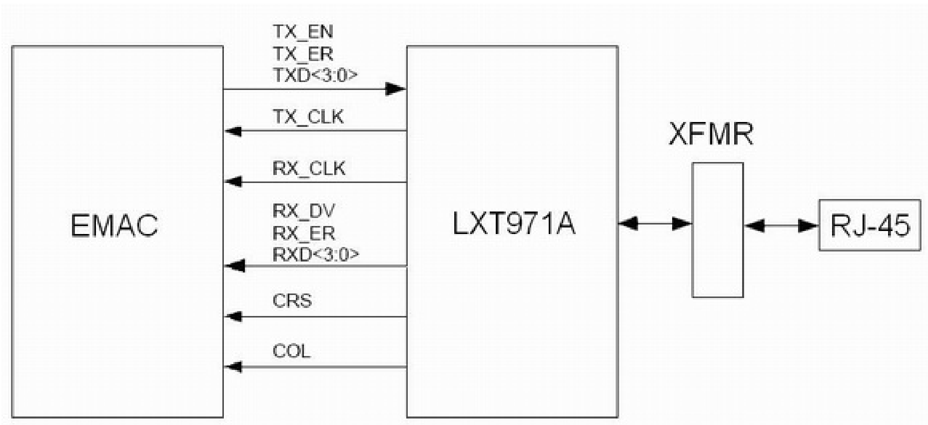

Fig. 2: Connecting the LXT971A PHY to the EMAC controller.

If necessary, more than one PHY can be accessed in the presented network system configuration. The TNETE2101 or TNET2004 devices might be of help is such cases.

5. Software Arcitecture for IP Phones

The IP phones software architecture is based on Radvision VoIP Toolkit and Telogy software architecture. Voice processing unit is in this case realized by Telogy software. Functions performed by the voice processing unit include: PCM reception, tone generation, acoustic echo cancellation, voice activity detection, voice playout, and compression options. All the specified functions are being performed on the DSP processor on board.

This software architecture is incorporated in TNETV1050 communication device. TNETV1050 was built upon two-processors core architecture and includes MIPS32 RISC processor and DSP C55x family processor. The integrated peripherals set includes all the units required for most of VoIP applications. The TNETV1050 peripheral set includes two EMACs, PHY, USB, UART and GPIO interfaces.

6. CONCLUSIONS

Digital signal processors along with apriopriate external devices might be succesfully implemented in Voice over IP network architectures. They may provide an excellent end-to-end solutions for voice communication over a TCP/IP based network.

Telogy software architecture and Radvision VoIP toolkit are incorporated in most of the discussed solutions. TNETV10xx family of processors from Texas Instruments are suitable for VoIP gateway solutions while TNETV1050 is most likely to be used within VoIP phone devices. Pros and cons of the presented solutions for both gateways and terminals are mentioned in previous chapters.

7. REFERENCES

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