VOICE OVER INTERNET PROTOCOL GATEWAY FUNCTIONALITY UPON THE ST100 DSP CORE FAMILY

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SUMMARY

Voice over Internet Protocol will probably be the next revolution step on the telecom market. Indeed, for more than one century, voice transport uses switched circuit network (SCN). Actually, switched packet network operators (like Internet provider), engaging huge resources on this market, offering voice transport at lower cost. However, the IP network, first designed to transport data only, must deal today with multimedia requirements. VoIP gateway will act like an interoperability bridge between IP network and conventional telecom network. This interoperability is required to guarantee the universal telephone service. Such gateway may be split into several independent functional blocks. Each block has specific needs in terms of hardware and software. STMicroelectronics is benchmarking the ST100 DSP core family to assume VoIP functionalities. Starting with analysis of overall core capabilities, we try to define in this paper the optimum topology of a ST100 VoIP gateway.

Keywords: *VoIP, gateway, ST100*

1. INDRODUCTION

Supporting voice transport and telecom signaling upon the Internet Protocol (IP) is one of the major challenges for telecom market in the next years. This new service called Voice over Internet Protocol (VoIP), allows a whole convergence between data and telecom networks. This meeting point of voice and data will help to boost multimedia services and market using for example videoconference, unified messaging and web-based call centers.

The Internet Protocol was originally created to transport data only, without any guarantee of service quality ("best effort"). Today, the Internet infrastructure must migrate to a complete multimedia world in order to fit with these new services requirements. This could be achieved, in current packet networks, using new mechanisms giving a different priority for each packet considering the real time needs of its contents (routing policies according to priority rules, bandwidth network management, "intelligent" real time network device, etc). Such differentiated services may add much complexity to both terminals and network nodes.

Since the Internet Protocol is not the standard for multimedia communication, it is necessary to build gateways between IP network and many different "worlds" like public switched telecom network (PSTN). A gateway, in this domain, is a network device allowing service translation (both logical and physical OSI layer) between heterogeneous networks. Fortunately, It is not mandatory for a gateway to support any kind of services translation upon each network. However, the quality of such device will be defined according to their overall capabilities (voice quality, services, real-time support, management, power consumption, scalability, cost…).

A voice over IP gateway is based on independent functions grouped in blocks. Each block has specific requirements in terms of hardware (DSP, microprocessor, interface, memory…) and software (processing load and type, memory size, data flow…). Many VoIP Gateway vendors (TI & Telogy, AudioCodes, NMS, Blue Wave Systems…) use architecture based on a cluster of DSP connected to a general-purpose microcontroller charged with all packet treatments. If the number of channels grows, the microcontroller becomes a bottleneck and the overall design efficiency decrease due to huge memory transfers between the different processing units. Consecutively, the voice quality is degraded and the number of simultaneous channel is also limited. The next study should allow to define a new structure reducing the bottleneck effect and to bring scalability.

STMicroelectronics is the 4th silicon manufacturer in the world (source: IC Insights ranking for first half 2001). ST120 is the first member of the ST100 DSP family. Running up to 800 MegaMAC/s at 400 MHz (1.2 Volt), it is a very suitable product for telecom applications. This DSP core supports two instruction sets (16/32 bits) and a SLIW mode allowing building scalable software between efficient DSP algorithms and small microcontroller RISC code. Currently, STMicroelectronics is benchmarking the capabilities of this core to implement different functional blocks of a VoIP Gateway. The result of this study will be

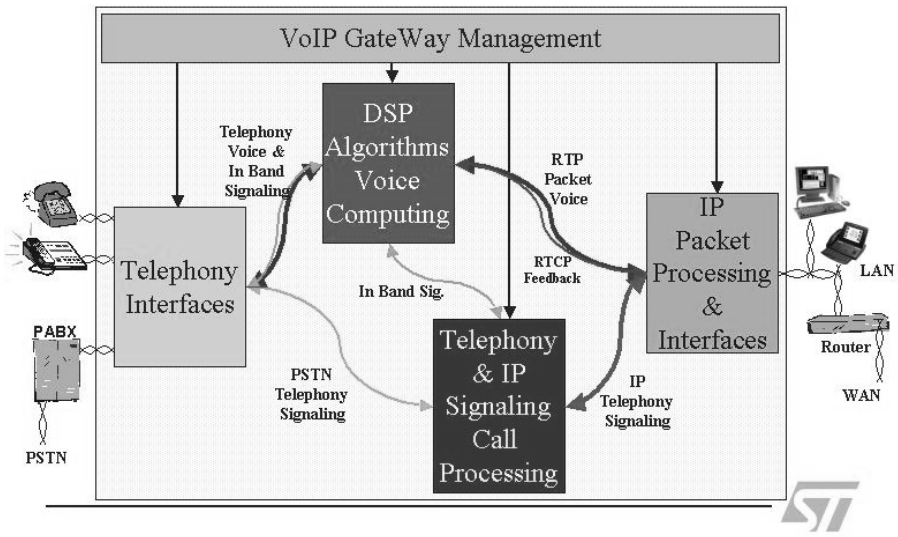

Fig. 1 VoIP Gateway Functional Blocks.

used to define the optimum architecture based on this DSP core. In this paper we will first present basic blocks of such gateway. Then we will evaluate ST100 capabilities to support expected functions and the corresponding benefits of such integration.

2. VoIP GATEWAY FUNCTIONAL BLOCKS

A VoIP Gateway could be divided into five basic blocks as shown in Fig.1.

Each block provides a set of functions required to build a complete voice over IP application [1, 2]. These functional blocks may integrate both hardware and software. The inter block communication represents either data flow (voice samples, HDLC frame…) or control message (on-off hook transition…). Blocks may also be distributed between different processing units (general-purpose microcontroller, bank of Digital Signal Processor, Personal Computer…) and hardware devices (HDLC controller, switching matrix, Ethernet interfaces, USB, ADSL interface…).

2.1. Management Block

The management block contains all software modules and hardware interfaces allowing to control and to monitor other blocks. This block is responsible of the automatic gateway configuration and software update at startup (DHCP/TFTP protocol) as well as the complete remote administration (SNMP agent, RS232…).

It is also used to provide inter block coordination permitting "high intelligence" functionality (automatic audio codec adaptation or jitter buffer size adjustment according to bandwidth monitoring). This block may be in a great extent realized by software, mostly using general-purpose microcontrollers. However, this function has no real time requirements and so it may be spread in the system to the lower priority level.

2.2. Telephony Block

The telephony interface is mainly a hardware module containing all devices needed to connect the gateway to the standard telephony world (corporate PABX or public Central Office) using analog and digital trunk termination or connecting directly analog and ISDN phone. It mainly consists of a panel of telecom components and related software in order to extract and generate voice channels (this includes signaling and voice samples). Some devices may be dedicated to manage only the signaling data from and to the PSTN (HDLC controller, embedded signaling gateway).

The telephony interface generates two independent flows. The voice samples flow including in-band signaling (like Dual Tone Multi Frequency: DTMF) and the signaling data flow, which are respectively forwarded to DSP algorithms module and telephony & IP signaling module. The telephony interface module is, in a common architecture, connected to the DSP block in order to realize channel treatments as fast as possible according to voice real time requirements.

2.3. DSP Block

The DSP algorithms & voice computing block is mainly used to compute the compression algorithms, for each voice channel, from the PSTN to the IP network in order to optimize the bandwidth consumption. While PSTN use the G.711 standard, IP Telephony use different algorithms like G.711, G.723.1, G.729…

This block principally based on high-optimized DSP algorithms is directly responsible of the number of channel supported and the overall voice quality.

Some other useful signal processing functionalities are added for voice and services improvement in Internet communication (echo canceling, voice activity detector and comfort noise generator, DTMF detection and generation, voice signal reconstruction, etc.).

This block, which is certainly the most important in a VoIP gateway, must have its hardware and software architecture carefully optimized. It must have full priority regarding all other gateway modules (real time processing of voice channels).

2.4. Signaling Block

The telephony signaling & call processing block is the key block to provide high-end interoperability and unified services between IP network and PSTN. It translates all signaling messages from the PSTN (using SS7, ISUP and in band messages) to the IP telephony protocols (ITU H.323 standard, SIP from IETF, or Megaco) [3].

This module contains also a call control function changing the current channel state (idle, ringing, accepting, open, closing, etc.) according to various incoming events (local off-hock, ring back tone detection, far end disconnect…) . It also negotiates all voice channel parameters with other terminals and components of the two different networks (gatekeeper, endpoint capabilities, billing, etc.).

This block, most of the time dedicated to high-end processor, must provide high-level services for multimedia communications and must be designed with a modular and upgradeable software structure. As Telephony signaling has no restricting real time requirements, it may be placed at a normal system priority.

2.5. IP Packet Block

The last block is the Internet packet processing and interfaces. It is close to the telephony interface functionalities for the packet network side. It is composed of hardware interfaces and associated software in order to be connected to any packet network like IP over Ethernet, or ATM over ADSL. In a Voice over Internet Protocol gateway we assume that IP is the packet network protocol (generally Ethernet is the underlying physical protocol).

IP protocol stack suite (TCP/UDP is commonly included) provides Internet packet services for transport and network layer processing (multiplexing, checkup of data and header integrity, encapsulation, routing, fragmentation…). Moreover, the gateway must also be able to access the physical network using lower level protocols (Ethernet/ARP, SDH…). It may be useful for network enhancement to add protocols enabling multicasting (IGMP, IPv6) to allow bandwidth savings during conference. Actually, the Internet network, first designed without any Quality of Services, migrates to the multimedia worlds. So, any VoIP gateway has now to support these network improvements (Type of Service in IP header, VLAN 802.1 p/q, RSVP protocol…).

This module is mainly based on software (protocol stacks) and specific network interfaces. Network packet processing has less real time constraints regarding to DSP block (voice packet contains most of the time 10ms of samples and far end jitter buffer may absorb some extra delay variations). However, you may avoid bottleneck to appear (high priority level) on the network interface of the VoIP gateway due to non real time processing like Telephony Signaling (especially to improve the voice packet throughput and limit the memory usage).

3. ST100 DSP CORE BENCHMARK

The ST100 DSP core should be used to build this kind of complete VoIP gateway solution. Currently, STMicroelectronics is benchmarking the capabilities of the ST100 to be in charge of some tasks of VoIP gateway blocks described above (DSP functions, Telephony Interface management, IP packet processing…). The goal of our study is to define the best hardware and software sharing between different elements of the gateway to optimize some features like throughput, delay and scalability.

Some management functions would be also included in the ST100 software in order to control some parts of the gateway. However, the DSP load for management and telephony signaling functions should be as small as possible. Thus, ST100 should be charged mostly with voice signal processing in the frame of the DSP block. This block must be optimized (assembly level programming and SLIW parallelism) in order to support as many channels as possible.

The signaling part should not be realized inside DSP processing units. Nevertheless, adding some telephony signaling software should be required to build a scalable VoIP gateway using only ST100 cluster. Basic signaling management for each channel may be done inside the ST100. So, it may relay all the signaling data to a signaling gateway or call agent for further specialized treatments (state machine update, Interactive Voice Response management: IVR). This kind of enhancement allows using one signaling gateway or call agent for an undefined number of ST100 standalone VoIP gateway (scalability).

Integrating the IP packet processing into the ST100 may also bring scalability and better performance to the overall system performance. However, it should be carefully optimized to be sure that the benefits (scalability, lower bottleneck and lower processing delays) wouldn't be at the expense of the supported numbers of channels.

Following some current results of our analysis we could mark the ST100 as a product suitable for VoIP applications. More complete tests should help us to define the best architecture using this core.

4. CONCLUSIONS

This paper has tried to define the optimum functional structure and optimum topology of a ST100 based VoIP Gateway. Using some benchmarks and ST100 load estimations, the tasks of the gateway have been split in several independent blocks (containing hardware and software). One of the next steps will be to evaluate IP protocol stack efficiency running on the ST100. TCP/IP will allow building a standalone VoIP processor based on ST100 core without any additional microcontroller based processing unit required to perform protocol stacks function. The result of this work will give an important information about tasks sharing between different processing units in the ST100 based VoIP Gateway. Using this knowledge we will be able to optimize the final topology.

REFERENCES

[1] Telogy Networks, "Voice over IP (VoIP)", www.techguide.com, White Paper

[2] Witowsky, W.E.: IP Telephone Design and Implementation Issues, Telogy Networks 31 Sep 1998, pp. 1-10.

[3] Liu, H., Mouchtaris, P.: Voice over IP Signaling H.323 and Beyond, IEEE Communications Magazine, IEEE, October 2000, pp. 142-148.

BIOGRAPHY

Damien Cottier received the Engineer diploma from ISTASE (Institute of advanced technologies of Saint-Etienne) in 2000. Currently, he is working on a Ph.D. degree in cooperation between STMicroelectronics (Grenoble) and the University Jean Monnet of Saint-Etienne. Its topic is mainly in system architecture (hardware/software), telecommunication, voice over Internet protocol, microprocessor & DSP and real time processing.

Viktor Fischer has received his master degree and engineer diploma in electronics from Technical University of Kosice, Slovak republic, in 1981. He has received the Ph. D. degree from the same university in 1991. From 1982 to 1991 he was Assistant Professor at the Department of Electronics of the University of Kosice. From 1991 he is working 6 month a year at Jean Monnet University of Saint-Etienne, France as Invited Professor in electronics and computer science.). His research interests include computer and communication security and embedded systems architecture. He is currently working also with MICRONIC, a small enterprise from Slovakia, oriented in development and production of computer security hardware and software.

Gerard Jacquet received the Engineer diploma from ENSERB (School of engineering in electronics and radio of Bordeaux France) in 1980 and the Ph.D degree in electrical engineering from University of Clermont-Ferrand (France) in 1985. In 1984, he was assistant professor in the department of electrical engineering at the University of Orleans (France). Currently, he is associate professor in ISTASE (Institut of advanced technologies of Saint-Etienne). His research interests include image processing, communication, system architecture, DSP and FPGA system design.