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FPGAs Driving Voice-Data Convergence
Part 1

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Introduction

Over the last few years, data exchange between people using the Internet has gained popularity, with over 6.9 trillion emails being exchanged last year. There is a big push to use the same backbone for voice traffic. This article gives an overview of voice data convergence technologies, the benefits to the users and some of the significant challenges facing the designers of these systems.

Voice-Data Convergence (Voice over Internet Protocol)

Convergence in networking refers to the ability to transfer data and voice (and/or video) traffic on a single network. Voice over IP, also known as IP telephony, and packet-voice, is the transmission of voice traffic in packets using the Internet Protocol (Internet backbone).

Current voice telephony is based on a circuit-switched infrastructure and uses the PSTN (public switched telephone network). When a call is placed, the PSTN reserves 64Kbps, end-to-end bandwidth for the duration of the call, on a fixed channel. A voice call generally does not utilize the full channel bandwidth. While, PSTN supports full duplex transfer, phone calls involve one person talking and the other listening and vice versa. There are many periods of silence where the network transmits no information, and hence wastes network bandwidth.

In VoIP networks, the packetization of voice happens in real-time. VoIP also decreases the bandwidth utilized significantly, since multiple packets can be transmitted simultaneously. The SS7 and TCP/IP networks are used together to setup and tear down the calls, along with Address Resolution Protocol (ARP).

Process of creating IP packets:

Step 1: An analog voice signal is converted to a linear pulse code modulation (PCM) digital stream (16 bits every 125msec).

Step 2: The line echo is removed from the PCM stream and is further analyzed for silence suppression and tone detection.

Step 3: The resulting PCM samples are converted to voice frames and a vocoder compresses the frames. G.729a creates a 10ms long frame with 10Bytes of speech. It compresses the 128kbps linear PCM stream to 8kbps.

Step 4: The voice frames are integrated into voice packets. First, a RTP packet with a 12-byte header is created. Then an 8-byte UDP packet with the source and destination address is added. Finally, a 20-byte IP header containing source and destination IP addresses is added.

Step 5: The packet is sent through the Internet where routers and switches examine the destination address, and route and deliver the packet appropriately to the destination. IP routing may require jumping between networks and pass through several nodes.

Step 6: When the destination receives the packet, the packet goes through the reverse process for playback.

The IP packets are numbered as they are created and sent to the destination address. The receiving end must reassemble the packets in their correct order (when they arrive out of order) to create voice. The IP addresses and telephone numbers must be mapped properly.

Description	PSTN	Internet
Designed for	Voice only	Packetized data, voice & video
Bandwidth Assignment	64Kbps (dedicated)	Full-line bandwidth over a period of time
Delivery	Guranteed	Not guaranteed
Delay	5-40ms (distance-dependent)	Not predictable (usually more than PSTN)
Cost for the Service	Per-minute charges: long distance Monthly flat rate: local access	Montly flat rate for access
Voice Quality	Toll quality	Depends on customer equipment
Connection Type	Telephone, PBX, switches with frame relay and ATM backbone	Modem, T1/E1, Gateway, Switches, ISDN, bridges, Routers, Backbone
Quality of Service	Real-time delivery	Not real-time delivery
Network Management	Homogeneous and interoperable at network and user level	Various styles with interoperability established at network layer only
Network Characteristics (Hardware)	Switching systems for assigned bandwidth	Routers & bridges for layer 3 and 2 switching
Network Characteristics (Software)	Homogeneous	Various interoperable software systems
Access Points	Telephones, PBX, PABX, ISDN, switches, high-speed trunks	Modem, ISDN, T1/E1 Gateway, high-speed DSL/cable modems

Figure 1: PSTN (circuit-switched) vs.
IP (packet-switched) networks

Motivation and Market

The integration of voice, video and data allows the use of a unified packet network, and thus reduces bandwidth consumption by 8:1 in favor of packet-based networks. By eliminating the voice infrastructure, the costs of maintaining both networks are eliminated. Web users are demanding free voice and video communications, as voice is the logical step from ubiquitous Internet mail and instant messengers. VoIP also provides enhanced features like flexible call routing and networked multimedia applications. The ability to use voice and video as part of the Web experience helps sell and support consumers, and improves site stickiness in portals, communities, directories and audio ads. Corporations can reduce costs using VoIP services for distance learning, customer support and remote sales presentations. Growing digital convergence and networking consumer devices in today's homes, requires a low-cost, integrated voice-data-video access to the Internet.

Cahners In-Stat Group projects that sales of VoIP equipment reached $61 million in 1998 and will exceed $3.8 billion in 2003. The VoIP market is expected to grow from 7.7 billion minutes in 2000, to 500 billion minutes by 2005, according to Probe Research. They also forecast the market for VoIP gateway equipment will increase from $1.2 billion in 2000 to $10 billion by 2005.