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FPGAs Driving Voice-Data Convergence
Part 2

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(View Part 1 of "FPGAs Driving Voice-Data Convergence")

VoIP Products

- Enterprise and Service Provider VoIP gateways are devices deployed between a PBX and WAN access device (router) to provide call set-up, call routing and to convert voice into IP packets. They aggregate incoming VoIP traffic and route the traffic accordingly; much like a traditional Class 5 CO switch.

- VoIP routers are standard routers with voice cards that perform packetization and compression, and the router then directs the packets to their ultimate destination.

- VoIP end stations (IP Phones) include telephone handsets, a VoIP gateway and a LAN interface.

VoIP Gateway Technologies

The VoIP gateway includes components such as:

- **Hardware Components**: Digital signal processor, controller, codec, analog front end, WAN interface

- **Software Components**:

    - Voice processing elements - Speech coders, echo cancellers, voice activity detection (VAD), comfort noise generator (CNG), telephony

    - Protocol stack related elements - H.323 and TCP/IP protocol stack

    - Voice and LAN/WAN interface management

    - RTOS

Figure 2 illustrates the functional architecture of a VoIP gateway and its three major functional blocks:

.- Voice Processing includes all functions required to encode voice data samples and packetize them for transmission.

-Telephony Signaling Gateway (TSG) subsystem performs the functions for establishing, maintaining and terminating a call.

- Network interface protocols include the TCP/IP protocol suite and the OSI Layer 2 protocols including ATM, Frame Relay, or Ethernet.

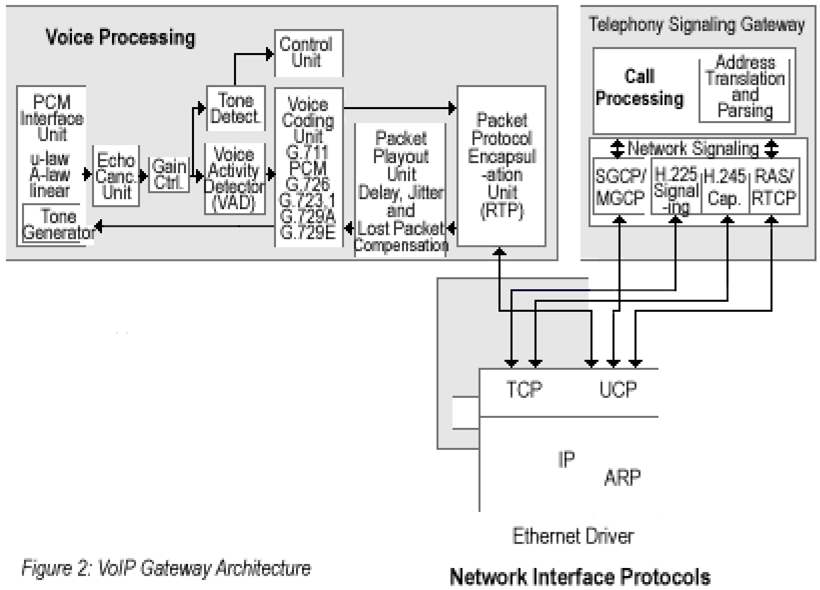

Figure 2: VoIP Gateway Architecture

VoIP Voice Processing

Voice processing functions include the following:

- **PCM Interface** conditions PCM data and includes functions such as companding and resampling. This block also includes the Tone Generator, which generates DTMF tones and call progress tones.

- **Echo Cancellation Unit** performs echo cancellation on sampled, full-duplex voice port signals in accordance with the ITU G.165 or G.168 standard.

- **Voice Activity Detector** suppresses packet transmission when voice signals are not present. If no activity is detected for a period of time, the voice encoder output will not be transported across the network. Idle noise levels are also measured and reported to the destination so that "comfort noise" can be inserted into the call.

- **Tone Detector** detects the reception of DTMF tones and discriminates between voice and facsimile signals.

- **Voice Coding Unit** compresses the voice data streams for transmission. There are several different codecs used to compress voice streams in VoIP applications. Each has been targeted at a different point in the tradeoff between data rate (compression ratio), its processing requirements, the processing delay, and audio quality. Table 1 compares the various ITU codecs with respect to these parameters. The MOS (mean opinion score) parameter is a measure of audio quality rated by a subjective testing process.

Standard	Description	Data Rate (Kbps)	Delay (mS)	MOS
G. 711	Pulse Code Modularion (PCM).	64	0.125	4.8
G. 721, G.723, G.726	Adaptive Differential PCM (ADP CM).	16, 24, 32, & 40	0.125	4.2
G.728	Low-Delay Codebook Excited Linear Predictors (LD-CELP).	16	2.5	4.2
G.729	Conjugate Structure Algebraic CELP (CS-ACELP).	8	10	4.2
G.723.1	Another CS-ACELP codec: Multi Pulse - Maximum Likelyhood Quantizer (MP-MLQ)	5.3 & 6.3	30	3.5 & 3.98

Table 1: Voice Coding Standards

- **Voice Play-out** buffers the packets that are received and forwards them to the voice codec for play-out. This module provides an adaptive jitter buffer and a measurement mechanism that allows buffer sizes to be adapted to the performance of the network.

- **Packet Voice Protocol** encapsulates the compressed voice data for transmission over the data network. Each packet includes a sequence number that allows the received packets to be delivered in the correct order. This also allows silence intervals to be reproduced properly and detection of lost packets.