Voice over IP Software

http://www.vocal.com

VOCAL Technologies voice over IP software libraries include components for proprietary and industry standard voice over public or private IP (internet protocol) networks. This system is a fully integrated audio processing system providing features such as voice coders and decoders, DTMF detection/generation, call progress detection/generation, echo cancellers, etc.

A system can be configured to support single or multiple voice over IP software coder algorithms. The typical technical criteria for selecting a particular algorithm is:

Perceived decoded signal quality
Compression data rate (expressed as bps or frame size/frames per second)
Frame size (contributes to the inherent algorithmic coding delay)
Look-ahead coding delay (also contributes to inherent algorithmic coding delay)
Memory Requirements (data memory and program memory for a particular processor)
Processing Requirements (expressed in MIPS for a particular processor)

For example, G.723 can be used to generate a 5 1/3k bps compressed stream. G.729A (an 8k bps compressed stream) requires roughly half of the number of instructions per second as the 5 1/3k bps G.723.

This system is compatible with VOCAL's H.323 multimedia system and VOCAL's V.70 DSVD (digital simultaneous voice and data) systems. Advanced features such as voice activity detection and comfort noise generation are available. Transmitting tokens for silence rather than the corresponding audio frames may further reduce the overall system data rate requirements.

Features:

- Fully integrated voice over IP/audio software processing system
- Full and half duplex modes of operation.
- Common compressed audio frame stream interface to support multiple speech coders systems
- Run-time selection of particular voice coder in multiple IP coder systems
- Multi-tasking environment compatible.

Voice over IP Software Coders (ordered by encoded stream data rate):

- G.723 (often referred to as G.723.1) - 5 1/3k and 6.4k bps ACELP/MP-MLQ
- G.729 - 8k bps CS-ACELP
- G.729A - reduced complexity version of G.729 - fewer MIPS at the expense of reduced perceived signal quality
- **GSM** 06.10 - 13k bps RPE-LTP
- G.728 - 16k bps LD-CELP
- G.726 - 16k, 24k, 32k and 40k bps ADPCM - normally not used in Voice-over-IP applications
- G.721 - 32k bps ADPCM - normally not used in audio applications
- G.711 - 64k bps PCM (A-Law or µ-Law format)

Configurations:

- Complete H.323 system available for industry standard and proprietary voIP networks
- Voice Activity Detection and Comfort Noise Generation available
- Direct interface to 8.0 kHz PCM data stream (A-law or µ-law), run-time time-slot selection.
- Analog DAA interface using linear codec at 8.0 kHz sample rate.
- North American/International Telephony supports DTMF generation, dialing procedures, call progress, Caller ID, etc.
- Line echo canceller (G.165 compliant) available.
- DTMF detector operation available - Zero hits on Mitel talkoff test tape, less than 150 hits on Bellcore test tape typical.
- Data/Facsimile/Voice Distinction and various startup procedures (V.8 /V.8bis) available.
- Complete data modem/facsimile systems available (through V.90/V.42bis and V.34fax/T.30)
- Multiple ports can be executed on a single DSP

Example System Features for ADSP-2181 at 40 MHz:

- Single executable image supports 2 channels of the following: G.729A encoder and decoder, DTMF detection, DTMF generation, call progress generation and G.165 compliant echo canceller.
- Multiple coder system: supports run-time selection of a single port of G.723/G.729/G.728/G.711 plus DTMF/Call Progress detect/generation, G.165 compliant echo canceller. Requires host support for servicing overlay requests.

VOCAL Technologies Ltd.

Custom Product Design Division
200 John James Audubon Parkway
Buffalo, New York 14228
716-688-4675