Telephony Software

http://www.vocal.com

VOCAL Technologies, Ltd. IP telephony software libraries include a complete range of ETSI / ITU / IEEE compliant modulations, optimized for execution on ANSI C and leading DSP architectures (ADI, ARM, DSP Group, LSI Logic ZSP, MIPS and TI). This software is modular and can be executed as a single task under a variety of operating systems or it can execute standalone with its own kernel.

VOCAL Technologies IP telephony 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.

VOCAL's Error Codes are used also to Optimized Power Saving in communications systems including Wireless LANs, ADSL modems, VDSL modems, satellite systems, etc.

G.729/G.729 Annex A/B Telephony Software

VOCAL Technologies modem/telephony software libraries include G.729 and G.729 Annex A speech coders optimized for execution on DSP architecture from leading silicon suppliers. This software is modular and can be executed as a single task under a variety of operating systems or it can be executed standalone with its own kernel.

G.729 is an 8 kbps Conjugate-Structure Algebraic-Code-Excited Linear Prediction (CS-ACELP) speech compression algorithm approved by ITU-T. G.729 Annex A is a reduced complexity version of the G.729 coder.

G.729 Annex A telephony software speech coder was developed for use in multimedia simultaneous voice and data applications like DSVD. The coder processes signals with 10 ms frames and has a 5 ms look-ahead which results in a total of 15 ms algorithmic delay. The input/output of this algorithm is 16 bit linear PCM samples that are converted from/to 8 kbps compressed data stream.

The line interface may be an analog front end (codec and DAA) or a digital interface such as T1/E1, switched 56 and ISDN. The upper end of this software offers a direct binary and speech frame interface.

G.728, G.726 and G.721 Telephony Software

The CCITT approved the telephony software G.711 recommendation on Pulse-Coded Modulation (PCM) µ-Law or A-Law in 1984. It is a 64 kbps compressed stream and is a common reference for speech compression quality. G.711 encoding/decoding is usually done within codec although a technique to convert linear samples to PCM (A-Law or µ-Law) is straightforward.

G.721 is a 32 kbps Adaptive Differential Pulse Code Modulation (ADPCM) speech compression algorithm. It produces toll quality speech. Under error free transmission, G.721 has a slightly worse perceived quality than G.711. The quality of G.721 significantly deteriorates when several such links are used in tandem. With transmission error rates higher than 10.4, the perceived quality of G.721 is better than G.711.

Telephony software G.726 extends the G.721 ADPCM to include 40, 24 and 16 kbps, as well as 32 kbps. G.726 at 40 kbps performs comparable to G.711.

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

G.728 Low-Delay Code Excited Linear Prediction (LD-CELP) compression is 16 kbps compression. This has an algorithmic coding delay of 0.625 ms. Compared with G.721, G.728 tends to score worse in objective, but better in subjective testes. A characteristic of CELP algorithms is that they tend to perform poorer than ADPCM in the presence of background noise.

G.723 Telephony Software

VOCAL Technologies modem/telephony software libraries include the G.723 dual rate speech coder optimized for execution on DSP architectures from leading silicon suppliers. This software is modular and can be executed as a single task under a variety of operating systems or it can be executed standalone with its own kernel. The G.723 speech coder recommendation was developed for use in multimedia platforms, in particular those specified by the H.32x series recommendations. It provides two compressed stream bit rates, 5 1/3k bps and 6.4k bps. The higher bit rate is of greater quality.

The coder processes signals with 30 ms frames and has a 7.5 ms look-ahead. Relative to the G.729/G.729A telephony software coders, the G.723 speech coders pass DTMF tones through with less distortion.

Both the 5 1/3k bps and 6.4k bps rates are mandatory for the encoder and decoder. A G.723 frame stream may switch between the two rates at any 30 ms frame boundary.

The line interface may be an analog front end (codec and DAA) or a digital interface such as T1/E1, switched 56 and ISDN. The upper end of this software offers a direct binary and speech frame interface.

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

http://www.vocal.com