G.728
Coding of Speech at 16 kbit/s using Low-Delay Code Excited Linear Prediction

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VOCAL Technologies, Ltd. software libraries include a complete range of ETSI / ITU / IEEE compliant and other standard and proprietary vocoder algorithms, optimized for execution on ANSI C and leading DSP architectures (ADI, AMD, ARM, DSP Group, LSI Logic ZSP, MIPS and TI).

G.728 contains the description of an algorithm for the coding of speech signals at 16 kbit/s using low-delay code excited linear prediction (LD-CELP). The essence of CELP techniques, which is an analysis-by-synthesis approach to codebook search, is retained in LD-CELP. The LD-CELP however, uses backward adaptation of predictors and gain to achieve an algorithmic delay of 0.625 ms. Only the index to the excitation codebook is transmitted. The predictor coefficients are updated through LPC analysis of previously quantized speech. The excitation gain is updated by using the gain information embedded in the previously quantized excitation. The block size for the excitation vector and gain adaptation is five samples only. A perceptual weighting filter is updated using LPC analysis of the unquantized speech

In the G.728 Encoder, after the conversion from A-law or mu-law PCM to uniform PCM, the input signal is partitioned into blocks of five-consecutive input signal samples. For each input block, the encoder passes each of 1024 candidate codebook vectors (stored in an excitation codebook) through a gain scaling unit and a synthesis filter. From the resulting 1024 candidate quantized signal vectors, the encoder identifies the one that minimizes a frequency-weighted mean-squared error measure with respect to the input signal vector. The 10-bit codebook index of the corresponding best codebook vector (or “codevector”), which gives rise to that best candidate quantized signal vector, is transmitted to the decoder. The best codevector is then passed through the gain scaling unit and the synthesis filter to establish the correct filter memory in preparation for the encoding of the next signal vector. The synthesis filter coefficients and the gain are updated periodically in a backward adaptive manner based on the previously quantized signal and gain-scaled excitation.

The decoding operation of the G.728 is also performed on a block-by-block basis. Upon receiving each 10-bit index, the decoder performs a table look-up to extract the corresponding codevector from the excitation codebook. The extracted codevector is then passed through a gain scaling unit and a synthesis filter to produce the current decoded signal vector. The synthesis filter coefficients and the gain are then updated in the same way as in the encoder. The decoded signal vector is then passed through an adaptive postfilter to enhance the perceptual quality. The postfilter coefficients are updated periodically using the information available at the decoder. The five samples of the postfilter signal vector are next converted to five A-law or m-law PCM output samples.

Features:

- Full or half duplex modes of operation.
- Pass ITU test vectors.
- Common compressed speech frame stream interface to support systems with multiple speech coders (G.728, G.723, G.729, et al.).
- Optimized for high performance on leading edge DSP architectures.
- Multi-tasking environment compatible.

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G.728 Configurations:

- DAA interface using linear codec at 8.0 kHz sample rate.
- Direct interface to 8.0 kHz PCM data stream (A-law or -law).
- North American/International Telephony (including caller ID) support available.
- Simultaneous DTMF detector operation available - (less than 150 hits on Bellcore test tape typical).
- MF tone detectors, general purpose programmable tone detectors/generators available.
- Line echo cancellation (G.165 compliant) available.
- Where multiple speech coders (G.723, G.729 et al.) are available, coder selection can occur at run-time.
- Data/Facsimile/Voice Distinction available.
- Data/Facsimile/Voice command sets available.
- Various startup procedures available (V.8 and V.8bis).
- Multiple ports can be executed on a single DSP.

Example Resource Requirements (ADSP-2181) for G.728:

- G.728 at 16 kbps requires 30 MIPS

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