G.727
5, 4, 3 and 2- bits sample Embedded Adaptive Differential Pulse Code Modulation (ADPCM)

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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.727 is an embedded Adaptive Differential Pulse Code Modulation (ADPCM) algorithms with 5-, 4-, 3- and 2-bits per sample (i.e., at rates of 40, 32, 24 and 16 kbit/s). G.727 defines the transcoding law when the source signal is a pulse-code modulation signal at a pulse rate of 64 kbit/s developed from voice frequency analog signals as fully specified by G.711.

Applications where the encoder is aware and the decoder is not aware of the way in which the ADPCM codeword bits have been altered, or when both the encoder and decoder are aware of the ways the codewords are altered, or where neither the encoder nor the decoder are aware of the ways in which the bits have been altered can benefit from other embedded ADPCM algorithms. The embedded ADPCM algorithms specified here are extensions of the ADPCM algorithms defined in G.726 and are recommended for use in packetized speech systems operating according to the Packetized Voice Protocol (PVP) specified in draft Recommendation G.764. PVP is able to relieve congestion by modifying the size of a speech packet when the need arises.

The four embedded ADPCM rates are 40, 32, 24 and 16 kbit/s, where the decision levels for the 32, 24 and 16 kbit/s quantizers are sub-sets of those for the 40 kbit/s quantizer. Embedded ADPCM algorithms are referred to by (x, y) pairs where x refers to the FF (enhancement and core) ADPCM bits and y refers to the FB (core) ADPCM bits. For example, if y is set to 2 bits, (5,2) will represent the 40 kbits/s embedded algorithm, (4,2) will represent the 32 kbit/s embedded algorithm, (3,2) will represent the 24 kbit/s embedded algorithm and (2,2) the 16 kbit/s algorithm. The bit rate is never less than 16 kbit/s because the minimum number of core bits is 2. G.727 provides coding rates of 40, 32, 24 and 16 kbit/s and core rates of 32, 24 and 16 kbit/s. This corresponds to the following pairs: (5,2), (4,2), (3,2), (2,2); (5,3), (4,3), (3,3); (5,4), (4,4).

G.727 Encoder:

- After the conversion of the A-law or mu-law PCM input signal to uniform PCM, a difference signal is obtained by subtracting an estimate of the input signal from the input signal itself.
- An adaptive 4-, 8-, 16- or 32-level quantizer is used to assign 2, 3, 4 or 5 binary digits to the value of the difference signal for transmission to the decoder. (Not all the bits necessarily arrive at the decoder since some of these bits can be dropped to relieve congestion in the packet network. For a given received sample, however, the core bits are guaranteed arrival if there are no transmission errors and the packets arrive at destination.)
- FB bits are fed to the inverse quantizer. The number of core bits depends on the embedded algorithm selected. For example, the (5,2) algorithm will always contain 2 core bits. The inverse quantizer produces a quantized difference signal from these binary digits.
- The signal estimate is added to this quantized difference signal to produce the reconstructed version of the input signal.
- Both the reconstructed signal and the quantized difference signal are operated upon by an adaptive predictor which produces the estimate of the input signal, thereby completing the feedback loop.

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G.727 Decoder:

- Includes a structure identical to the FB portion of the encoder.
- In addition, there is also an FF path that contains a uniform PCM to A-law or mu-law conversion.
- The core as well as the enhancement bits are used by the synchronous coding adjustment block to prevent cumulative distortion on synchronous tandem codings under certain conditions.
- The synchronous coding adjustment is achieved by adjusting the PCM output codes to eliminate quantizing distortion in the next ADPCM encoding stage.

Applications:

- WIFI phones VoWLAN
- Wireless GPRS EDGE systems.
- Personal Communications
- Wideband IP telephony
- Audio and Video Conferencing

Features:

- Full and half duplex modes of operation.
- Passes ITU test vectors.
- Common compressed speech frame stream interface to support systems with multiple speech coders (G.723, G.726, G.728, G.729 et al).
- Optimized for high performance on leading edge DSP architectures.
- Multi-tasking environment compatible.
- Can be integrated with G.168 and G.165 echo cancellers, and tone detection/regeneration.
- Supports Voice Activity detection and Comfort Noise Generation.
- Multi channel implementation.
- Complain with G.727 specification.
- Optimized implementation.

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