G.726
40, 32, 24, 16 kbps 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 vocoders algorithms, optimized for execution on ANSI C and leading DSP architectures (ADI, AMD, ARM, CEVA, LSI Logic ZSP, MIPS and TI)

G.726 makes a conversion of a 64 kbit/s A-law or mu-law pulse code modulation (PCM) channel to and from a 40, 32, 24 or 16 kbit/s channel. The principal application of 24 and 16 kbit/s channels is for overload channels carrying voice in Digital Circuit Multiplication Equipment (DCME). The principal application of 40 kbit/s channels is to carry data modem signals in DCME, especially for modems operating at greater than 4800 kbit/s.

In the ADPCM 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 31, 15, 7, or 4-level quantizer is used to assign five, four, three or two binary digits, respectively, to the value of the difference signal for transmission to the decoder. An inverse quantizer produces a quantized difference signal from these same five, four, three or two binary digits, respectively. 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.

The ADPCM decoder includes a structure identical to the feedback portion of the encoder, together with a uniform PCM to A-law or mu-law conversion and a synchronous coding adjustment. The synchronous coding adjustment prevents cumulative distortion occurring on synchronous tandem codings (ADPCM-PCM-ADPCM, etc., digital connections) under certain conditions. The synchronous coding adjustment is achieved by adjusting the PCM output codes in a manner that attempts 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.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.726 specification.
- Optimized implementation.

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Example Resource Requirements (ADSP-2181) for G.726:

- G.726 encoder requires 3.5 MIPS, decoder requires 3.7
- A popular non-standard configuration of G.726 encoder requires 3.2 MIPS, decoder requires 3.0 MIPS

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