G.729
Coding of Speech Signals at 8 kbit/s using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP)

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VOCAL Technologies G.729 software libraries include a complete range of ITU compliant modulations, optimized for execution on ANSI C and DSP architectures from leading silicon suppliers (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 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 microkernel. 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 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

G.729 Annex B provides a high level description of the Voice Activity Detection (VAD), Discontinuous Transmission (DTX), and Comfort Noise Generator (CNG) algorithms. These algorithms are used to reduce the transmission rate during silence periods of speech. They are designed and optimized to work in conjunction with Recommendation V.70. Recommendation V.70 mandates the use of Annex A/G.729 (G.729A) speech coding methods. However, when it is desirable, the full version of Recommendation G.729 can also be used to improve the quality of the speech

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.729 Configurations:

- DAA interface using linear codec at 8.0 kHz sample rate.
- Direct interface to 8.0 kHz PCM data stream (A-law or µ-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.729, G.723, G.728, G.726 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.

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Features:

- Passes ITU test vectors.
- Common compressed speech frame stream interface to support systems with multiple speech coders (G.729 Annex A/B, G.723, G.728, G.726 et al).
- Optimized for high performance on leading edge DSP architectures.
- Multi-tasking environment compatible.

Example Resource Requirements (ADSP-2181) for G.729:

- G.729 Annex A Encoder requires 8.7 MIPS, 6762 words of PM, and 1512 words of DM
- G.729 Annex A Decoder requires 2.1 MIPS, 3950 words of PM, and 725 words of DM
- G.729 Annex A Encoder/Decoder requires 10.8 MIPS, 7720 words of PM, and 1918 words of DM

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