G.723.1
Dual Rate Speech Coder for Multimedia Communications Transmitting at 5.3 and 6.3 kbit/s

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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 for execution on ANSI C and optimized for leading DSP architectures (ADI, AMD, ARM, DSP Group, LSI Logic ZSP, MIPS and TI).

G.723.1 specifies a coded representation that can be used for compressing the speech or other audio signal component of multimedia services at a very low bit rate. In the design of this coder, the principal application considered was very low bit rate visual telephony as part of the overall H.324 family of standards. G.723.1 has two bit rates associated with it. These are 5.3 and 6.3 kbit/s. The higher bit rate has greater quality. The lower bit rate gives good quality and provides system designers with additional flexibility. Both rates are a mandatory part of the encoder and decoder. It is possible to switch between the two rates at any 30 ms frame boundary. An option for variable rate operation using discontinuous transmission and noise fill during non-speech intervals is also possible.

G.723.1 coder is designed to operate with a digital signal obtained by first performing telephone bandwidth filtering (Recommendation G.712) of the analogue input, then sampling at 8000 Hz and then converting to 16-bit linear PCM for the input to the encoder. The output of the decoder should be converted back to analogue by similar means. Other input/output characteristics, such as those specified by Recommendation G.711 for 64 kbit/s PCM data, should be converted to 16-bit linear PCM before encoding or from 16-bit linear PCM to the appropriate format after decoding.

G.723.1 decoder operation is also performed on a frame-by-frame basis. First the quantized LPC indices are decoded, then the decoder constructs the LPC synthesis filter. For every subframe, both the adaptive codebook excitation and fixed codebook excitation are decoded and input to the synthesis filter. The adaptive postfilter consists of a formant and a forward-backward pitch postfilter. The excitation signal is input to the pitch postfilter, which in turn is input to the synthesis filter whose output is input to the formant postfilter. A gain scaling unit maintains the energy at the input level of the formant postfilter.

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.729, G.728, G.726 et al).
- Optimized for high performance on leading edge DSP architectures.
- Multi-tasking environment compatible.

Applications:

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

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Configurations:

- DAA interface using linear codec at 8.0 kHz sample rate.
- Direct interface to 8.0 kHz PCM data stream (A-law or mu-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 & G.168 compliant) available.
- Where multiple speech coders (G.729, G.728, G.726 et al.) are available, coder selection can occur at run-time.
- Data/Facsimile/Voice Distinction available.
- Various startup procedures available (V.8 and V.8bis).
- Multiple ports can be executed on a single DSP.

Example Resource Requirements (ADSP-2181):

- Encoder 5 1/3k bps requires 18 MIPS
- Encoder 6.4k bps requires 26 MIPS
- Decoder (5 1/3k bps or 6.4k bps) requires 2 MIPS

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