## G.722.2 Adaptive Multi-Rate Wideband AMR-WB Vocoder Algorithm

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, DSP Group, LSI Logic ZSP, MIPS and TI)

G.722.2 uses the audio band of 50 - 7000 Hz instead of 200 - 3400 Hz for traditional telephony. The increased bandwidth improves the intelligibility and naturalness of speech significantly.

G.722.2 describes the detailed mapping from input blocks of 320 speech samples in 16 bit uniform PCM format to encoded blocks of 132, 177, 253, 285, 317, 365, 397, 461 and 477 bits and from encoded blocks of 132, 177, 253, 285, 317, 365, 397, 461 and 477 bits to output blocks of 320 reconstructed speech samples. The sampling rate is 16 000 samples/s leading to a bit rate for the encoded bit stream of 6.60, 8.85, 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbit/s.

The coding scheme for the multi-rate coding modes is the so called Algebraic Code Excited Linear Prediction Coder, hereafter referred to as ACELP. The multi-rate wideband ACELP coder is referred to as AMR-WB. G.722.2 also utilizes an integrated Voice Activity Detector (VAD).

G.722 is used for Voice over IP (VoIP) and Internet (IP) applications, Mobile Communications, PSTN applications, ISDN wideband telephony, ISDN video-telephony and video-conferencing.

Annexes A and B and Appendix I provide supplemental functionalities allowing interoperability with GSM and 3GPP wireless systems. These functionalities have originally been developed for these systems, but their use is not limited to mobile applications. Two other Annexes D and E describe test vectors and frame structure respectively.

## 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.729, G.728, G.726 et al).
- Optimized for high performance on leading edge DSP architectures.
- Multi-tasking environment compatible.
- Codes 16 bit linear PCM sampled at 16kHz.
- Supports data 9 data rates (6.6 to 23.85 kbps).
- Supports Voice Activity detection and Comfort Noise Generation.
- Multi channel implementation.
- Complain with G.722.2 specification.
- Optimized implementation.



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http://www.vocal.com

Custom Product Design Division 200 John James Audubon Parkway Buffalo, New York 14228 716-688-4675