GSM 06.20
Half Rate (HR) Vocoder
Vector-Sum Excited Linear Prediction (VSELP)

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VOCAL Technologies, Ltd. ETSI GSM 06.20 software libraries include ANSI-C and optimized assembly code for 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.

GSM 06.20 GSM half rate codec uses the VSELP (Vector-Sum Excited Linear Prediction) algorithm. The VSELP algorithm is an analysis-by-synthesis coding technique and belongs to the class of speech coding algorithms known as CELP (Code Excited Linear Prediction).

GSM 06.20 GSM half rate codec's encoding process is performed on a 20 ms speech frame at a time. A speech frame of the sampled speech waveform is read and based on the current waveform and the past history of the waveform, the codec encoder derives 18 parameters that describe it. The parameters extracted are grouped into the following three general classes: Energy parameters (R0 and GSP0); Spectral parameters (LPC and INT_LPC); Excitation parameters (LAG and CODE).

GSM 06.20 half rate codec is an analysis-by-synthesis codec, therefore the speech decoder is primarily a subset of the speech encoder. The quantised parameters are decoded and a synthetic excitation is generated using the energy and excitation parameters. The synthetic excitation is then filtered to provide the spectral information resulting in the generation of the synthesized speech.

GSM 06.20 speech encoder takes its input as a 13 bit uniform Pulse Code Modulated (PCM) signal either from the audio part of the MS or on the network side, from the PSTN via an 8 bit/A-law or μ-law (PCS 1900) to 13 bit uniform PCM conversion. The encoded speech at the output of the speech encoder is delivered to the channel coding function as defined in GSM 05.03 [3] to produce an encoded block consisting of 228 bits leading to a gross bit rate of 11,4 kbit/s. In the RX direction, the inverse operations take place.

GSM 06.20 describes the detailed mapping between input blocks of 160 speech samples in 13 bit uniform PCM format into encoded blocks of 112 bits and from encoded blocks of 112 bits to output blocks of 160 reconstructed speech samples. The sampling rate is 8 000 sample/s leading to an average bit rate for the encoded bit stream of 5,6 kbit/s. The coding scheme is called Vector Sum Excited Linear Prediction (VSELP) coding.

GSM 06.20 Half-rate Encoder:

- The GSM half rate speech encoder uses an analysis by synthesis approach to determine the code to use to represent the excitation for each subframe.
- The codebook search procedure consists of trying each codevector as a possible excitation for the Code Excited Linear Predictive (CELP) synthesizer.
- The synthesized speech is compared against the input speech and a difference signal is generated.
- This difference signal is then filtered by a spectral weighting filter, to generate a weighted error signal.
- The codevector which generates the minimum weighted error power is chosen as the codevector for that subframe.
- The spectral weighting filter serves to weight the error spectrum based on perceptual considerations.
- This weighting filter is a function of the speech spectrum and can be expressed in terms of the a parameters of the short term (spectral) filter.

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GSM 06.20 Half-rate Decoder:

- The speech decoder creates the combined excitation signal from the long term filter state and the VSELP codevector.
- The long term filter state is replaced by another VSELP codebook and the pitch prefilter is not used.
- The combined excitation is then processed by an adaptive pitch prefilter and gain.
- The prefiltered excitation is applied to the LPC synthesis filter.
- After reconstructing the speech signal with the synthesis filter, an adaptive spectral postfilter is applied followed by an automatic gain control which is the final processing step in the speech decoder.

Applications:

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

Features:

- Full and half duplex modes of operation.
- Passes ETSI test vectors.
- Common compressed speech frame stream interface to support systems with multiple speech coders (GSM FR, GSM EFR, GSM HR, 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.
- Multi channel implementation
- Complain with GSM 06.20 Recommendation.
- Optimized implementation

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