



ETSI HR-GSM Speech Coder

SIGNALS+SOFTWARE



Processor

Motorola DSP56300 range.

Background

The algorithm implemented is the ETSI, Half-Rate (HR) GSM recommendation, digital mobile telephone system. The encoder compresses linear-PCM (Pulse Code Modulated), speech input data, at a sample rate of 8kHz, using a codebook search to processes 20ms speech frames into 112 transmission bits comprised of three general sets of parameters: energy, spectral and excitation. The decoder generates the synthesized speech using the energy and excitation parameters and weights the excitation to match the original input speech by filtering with the spectral parameters.

The HR-GSM algorithm implements silence compression techniques to reduce the transmitted bit rate during the silent intervals of speech. Systems allowing discontinuous transmission (DTX) are based on Voice Activity Detection (VAD) algorithms and Comfort Noise Generator (CNG) algorithms that allows the insertion of Silence Insertion Descriptor (SID) frames during the silence intervals. This also provides the additional advantage of using lower processing loads and DSP bandwidth resource during silence frames.

Features and Performance

- Passes all ETSI HR-GSM test vectors
- Less than 20K data memory required for 5 channels of HR-GSM

HR GSM	Program Memory		X Data Memory		Y Data Memory		Processing Load (MHz)
	Code (Kwords)	Tables (Kwords)	Variables (Kwords)	Tables (Kwords)	Variables (Kwords)	Tables (Kwords)	
Encoder	7.66	7.53	$0.78+n*1.0$	2.45	$0.78+n*0.31$	5.08	$n* 14.87$
Decoder	3.77	6.9	$0.34+n*0.48$	1.82	$0.34+n*0.24$	5.09	$n* 2.93$
Encoder + Decoder	10.21	7.78	$0.78+n*1.48$	2.49	$0.78+n*0.55$	5.29	$n*17.8$

Table 1 : DSP Requirements for HR-GSM

Note: Processing loads quote worst-case scenarios and n represents the number of channels. Program memory table values are initialisation values. 1 word equals 24 bits. Kwords equals 1024 words.

Technical Notes

The coding scheme is the Vector Sum Excited Linear Prediction (VSELP) algorithm. This is an analysis-by-synthesis technique and one of a class of speech coding algorithms called Code Excited Linear Prediction (CELP).

Interface Details

For convenience the individual software functions are supplied as a single library module. The library contains all the object code that is required to link in to a user's top- level application code.

Availability

Fully optimised code is available now for a one-off payment and/or royalties depending on the commercial application.

Also available for DSP56300™ are a full range of vocoders including AMR-GSM, FR-GSM, EFR-GSM as well as other communication algorithms.

SIGNALS+SOFTWARE

SIGNALS+SOFTWARE was founded in 1992 as a developer of high quality Digital Signal Processing application software for the communications industry. Supplying to a whole range of customers, including large blue chip corporations, **SIGNALS+SOFTWARE** has quickly established itself as the world leader in DSP software design and production.

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