

**EFR-GSM Speech Coder
(Incorporating AMR-GSM and IS-641A)**



Processor

Motorola StarCore™ MSC8101 & MSC8102 DSP.

Background

SIGNALS+SOFTWARE are developing a complete suite of communication software for the Motorola MSC8100 family of DSPs. The initial development platform, the MSC8101, utilizes the StarCore™ 140 four ALU (Algorithmic Logic Unit) DSP core. This device also has 512kb memory and a Communications Processor Module (CPM) making it a versatile device for communication applications.

The algorithms implemented are the ETSI Enhanced Full-Rate (EFR) GSM audio processing functions, including transcoding, discontinuous transmission and error concealment. The encoder compresses linear speech input data, at a sample rate of 8kHz, to a data rate of 12 200 bps.

The EFR-GSM algorithm uses silence compression or discontinuous transmission to reduce the transmitted bit rate during the silent intervals of speech. Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) algorithms are used to enable the transmission of Silence Descriptor (SID) frames during the periods of silence. This provides the additional advantage of using lower processing loads and DSP bandwidth resource during silence intervals.

Features and Performance

- Integrated AMR-GSM and IS-641A functionality.
- Both Subscriber and Infrastructure EFR-GSM solutions are available.

AMR-GSM	Program Memory (Kbytes)	Tables (Kbytes)	Stack (Kbytes)	Data Memory (Kbytes)	Processing Load (MHz)
Encoder	74.6	28.0	6.3	n * 3.3	n * 6.16
Decoder	35.7	26.9	1.5	n * 1.6	n * 1.10
Encoder + Decoder	104.5	29.1	6.3	n * 5.0	n * 7.26

Table 1 : DSP Requirements for EFR-GSM- Infrastructure Solution

Note: Processing loads quote worst-case scenarios and n represents the number of channels.
Program memory table values are initialisation values. Kbytes equals 1024 bytes.

Technical Notes

The EFR-GSM encoding scheme uses Algebraic Code Excited Linear Prediction Coder (ACELP). An excitation signal at the input of a short term linear predictor synthesis filter is constructed by adding two excitation vectors from adaptive and fixed codebooks.

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. The audio functions are either callable as C functions or as assembly functions.

Availability

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

Also available or in development for StarCore™ are a full range of vocoders including FR-GSM, AMR-GSM, IS-641A 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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