SIGNALS+SOFTWARE

Product Preview

ETSI AMR GSM Speech Coder
(incorporating EFR-GSM and IS-641A)

TI Third Party Network Member

DSP

TEXAS INSTRUMENTS

Processor

Texas Instruments TMS320C6000 DSP range.

Background

The algorithm implemented is the ETSI Adaptive Multi-Rate (AMR) GSM recommendation, digital cellular telecommunications system. The encoder compresses linear-PCM (Pulse Code Modulated) speech input data, at a sample rate of 8kHz, to one of eight data rate modes 12 200, 10 200, 7 950, 7 400, 6 700, 5 900, 5 500 and 4 750 bps. There is also one mode to handle non-speech frames. The data rate will be run-time selectable by the user.

The AMR-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

- TI eXpressDSP™ Compliant development
- 16 channels of AMR-GSM on 200MHz device
- Integrated IS-641-A and EFR-GSM functionality

AMR GSM	Program Memory		Data Memory			Processing Load (MHz)
	Code (Kbytes)	Tables (Kbytes)	Tables (Kbytes)	Stack Memory (Kbytes)	Static Memory (Kbytes)	
Encoder + Decoder	98	29.31	29.31	2.93	n * 4.7	n* 12.0

Table 1: DSP Requirements for AMR GSM

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

Technical Notes

The software will be written using only fixed-point instructions and will be compatible with both the TMS320C6000 fixed-point family and the TMS320C6700 floating-point family. It will operate in either big-endian or little-endian mode, with the selection being made at build time.

The AMR-GSM speech coding algorithm is a member of a subset of the linear predictive coders (LPC) known as the ACELP (Algebraic-Codebook-Excited Linear Predictive Coding) class of coders.

Interface Details

For convenience the individual software functions will be supplied as a single library module. The library will contain all the object code that is required to link in to a user's top-level application code. The audio functions will either be callable as C functions or as assembly functions.

AMR-GSM will also be available fully eXpressDSP compliant.

Availability

The code will be available shortly, for a one-off payment and/or royalties depending on the commercial application.

Software for the TMS320C6000 is available for a range of GSM vocoders including FR-GSM, EFR-GSM, and for 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.

For further information please contact:

SIGNALS+SOFTWARE Ltd.
The Heights,
Lowlands Road,
Harrow,
HA1 3AW
United Kingdom

Tel: +44 (0) 20 8872 9000
Fax: +44 (0) 20 8872 9001

www.signalsandsoftware.com

sales@signalsandsoftware.com