



ITU G.723.1 Speech Coder

SIGNALS+SOFTWARE

Processor

Texas Instruments TMS320C6000 DSP range.



Background

The algorithm implemented is the ITU-T recommendation G.723.1 dual rate speech coder for internet and multimedia communications. The encoder compresses linear narrowband speech input data, at a sample rate of 8kHz, to data rates of 5 300 or 6 400 bps. The encoder includes an optional High Pass Filter and the decoder includes an optional Post Filter.

This implementation also includes silence compression techniques specified in recommendation G.723.1 Annex A 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

- TI eXpressDSP™ Compliant software
- >20 channels of G.723.1 on 200MHz device
- Less than 62Kbytes program memory required
- Passes all ITU-T test vectors

G.723.1	Program Memory		Data Memory			Interrupt Latency (Cycles)	Typical call Period (ms)	Processing Load (MHz)
	Code (Kbytes)	Tables (Kbytes)	Static Memory		Stack Memory (Kbytes)			
			Heap (Kbytes)	Tables (Kbytes)				
Encoder	48.0	0.05	n * 1.4	18.5	5.1	44464	30	n* 8.6
Decoder	22.3	0.05	n * 0.5	17.5	2.9	9452	30	n* 0.8
Encoder + Decoder	61.9	0.10	n * 1.9	18.6	5.1	44464	30	n* 9.4

Table 1 : DSP Requirements for G.723.1

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 is written using only fixed-point instructions and is compatible with both the TMS320C6000 fixed-point family and the TMS320C6700 floating-point family. It is supplied in both big-endian and little-endian variants.

The G.723.1 specification encodes speech and audio signals using linear predictive analysis-by-synthesis coding. The higher 6 400 bps data rate requires the use of a Multiple Maximum Likelihood Quantisation (MP-MLQ) excitation signal. The lower 5 300 bps data rate uses Conjugate-Structure Algebraic-Code-Excited Linear Prediction (CS-ACELP).

Interface Details

The eXpressDSP™ G.723 software uses the IG723 interface specified by Texas Instruments in the eXpressDSP™ developers' kit.

The software is also available in a non-eXpressDSP version with a basic multi-channel interface. The DSP requirements for this version are similar to those given in Table 1.

Availability

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

Software for the TMS320C6000 is available for a full range of vocoders including G.711, G.722, G.726, G.728, G.729, G.729A, G.729B/G.729AB, and for other communication algorithms. G.723.1 is also under development for the TMS320C5000.

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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