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Product Preview

IS-641-A TDMA Speech Coder

TI Third Party Network Member
DSP
TEXAS INSTRUMENTS

Processor

Texas Instruments TMS320C6000 DSP range.

Background

The algorithm to be implemented is the TIA/EIA recommendation IS-641-A TDMA, a fixed rate speech coder. The encoder compresses speech input data in a 16-bit uniform Pulse Code Modulated (PCM) format, at a sample rate of 8k samples per second (128k bps) to a fixed bit rate of 7400 bps. The 20ms speech frames (160 samples) are processed by a Code Excited Linear Prediction (CELP) encoding process in which the fixed codebook employs the Algebraic Code Excited Linear Prediction (ACELP) codebook structure. IS-641-A also uses short-term post-filtering.

The IS-641-A TDMA 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) algorithm and a Comfort Noise Generator (CNG) algorithm that allows the insertion of Silence Insertion Descriptor (SID) frames during the silence intervals. This also gives the additional advantage of using lower processing loads and bandwidth resource of the DSP during silence frames.

Features and Performance

- TI eXpressDSP™ Compliant development
- Integrated solution with AMR-GSM and EFR-GSM functionality will be available

IS-641-A TDMA	Program Memory		Data Memory			Processing Load (MHz)
	Code (Kbytes)	Tables (Kbytes)	Tables (Kbytes)	Stack Memory (Kbytes)	Static Memory (Kbytes)	
Encoder + Decoder	58.62	0	9.8	2	n * 5	n* 10

Table 1 : DSP Requirements for IS-641-A

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 operation of the IS-641-A algorithm is essentially based on the analysis-by-synthesis method for minimising a residual signal. This is done by choosing a codebook vector which, when passed through a long-term pitch filter and a short-term linear predictive filter, will be as close to the original signal as possible. The term close, in this case, refers to perceptual audible difference rather than a mathematical subtraction. The parameters of the model are obtained by quantisation of the pitch filter coefficients, linear predictive filter coefficients (LPC) and codebook information. These are transmitted over the channel once per frame, each of which consists of 160 samples (20ms of speech).

Interface Details

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

IS-641-A will also be available fully eXpressDSP compliant.

Availability

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

Software for the TMS320C6000 is available for a range of vocoders including G.711, G.722, G.723.1, G.726, G.728, G.729, G.729A, G.729B, G.729AB, FR-GSM, AMR-GSM, EFR-GSM and 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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