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Universal Mobile Telecommunications System (UMTS);
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ANSI C code (floating-point)
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Contents

Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (3GPP).

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

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- z the third digit is incremented when editorial only changes have been incorporated in the document.

1 Scope

The present document contains an electronic copy of the ANSI C floating-point code for the Enhanced Voice Services (EVS) Codec. This ANSI C code is the unique alternative reference specification besides the ANSI-C fixed-point code for the EVS Codec (3GPP TS 26.442) for a standard compliant implementation of the EVS Codec (3GPP TS 26.445), Voice Activity Detection (VAD) (3GPP TS 26.451), Comfort Noise Generation (CNG) (3GPP TS 26.449), Discontinuous Transmission (DTX) (3GPP TS 26.450), Packet Loss Concealment (PLC) of Lost Packets (3GPP TS 26.447), Jitter Buffer Management (JBM) (3GPP TS 26.448), and AMR-WB Interoperable Function (3GPP TS 26.446).

Requirements for any implementation of the EVS codec to be standard compliant are specified in 3GPP TS 26.444 (Test sequences).

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
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- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[2] 3GPP TS 26.445: "Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description".

[3] 3GPP TS 26.451: "Codec for Enhanced Voice Services (EVS); Voice Activity Detection (VAD)".

[4] 3GPP TS 26.449: "Codec for Enhanced Voice Services (EVS); Comfort Noise Generation (CNG) Aspects".

[5] 3GPP TS 26.450: "Codec for Enhanced Voice Services (EVS); Discontinuous Transmission (DTX)".

[6] 3GPP TS 26.447: "Codec for Enhanced Voice Services (EVS); Error Concealment of Lost Packets".

[7] 3GPP TS 26.448: "Codec for Enhanced Voice Services (EVS); Jitter Buffer Management".

[8] 3GPP TS 26.446: "Codec for Enhanced Voice Services (EVS); AMR-WB Backward Compatible Functions".

[9] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications".

[10] Recommendation ITU-T G.191 (03/10): "Software tools for speech and audio coding standardization".

[11] Recommendation ITU-T G.192: "A common digital parallel interface for speech standardization activities".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the definitions given in TR 21.905 [1] apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in TR 21.905 [1].

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in TR 21.905 [1] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in TR 21.905 [1].

ACELP	Algebraic Code-Excited Linear Prediction
AMR-WB	Adaptive Multi Rate Wideband (codec)
CNG	Comfort Noise Generator
DTX	Discontinuous Transmission
EVS	Enhanced Voice Services
FB	Fullband
FEC	Frame Erasure Concealment
IP	Internet Protocol
JBM	Jitter Buffer Management
MSB	Most Significant Bit
MTSI	Multimedia Telephony Service for IMS
NB	Narrowband
PS	Packet Switched
PSTN	Public Switched Telephone Network
SAD	Sound Activity Detection
SC-VBR	Source Controlled - Variable Bit Rate
SID	Silence Insertion Descriptor
SWB	Super Wideband
VAD	Voice Activity Detection
WB	Wideband
WMOPS	Weighted Millions of Operations Per Second

4 C code structure

This clause gives an overview of the structure of the floating-point C code and provides an overview of the contents and organization of the C code attached to the present document.

The ANSI-C code has been verified on the following platforms:

- IBM PC compatible computers with Windows 7 operating systems and Microsoft Visual C++ 2010 compiler, 32-bit.

ANSI-C was selected as the programming language because portability was desirable.

4.1 Contents of the C source code

The C code is organized as follows:

Table 1: Source code directory structure

Directory	Description
README.txt	information on how to compile
Makefile	UNIX style encoder Makefile
Workspace_msvc/	Directory for the MSVC 2008 (or newer) project files
lib_com/	Source code files used both in encoder and decoder
lib_dec/	Source code files used solely in the decoder
lib_enc/	Source code files used solely in the encoder

The distributed files with suffix "c" contain the source code and the files with suffix "h" are the header files. The ROM data is contained in files named "rom_xxx" with suffix "c".

Makefiles are provided for the platforms in which the C code has been verified (listed above). Once the software is installed, this directory will have a compiled version of the encoder (named EVS_cod) and the decoder (named EVS_dec).

4.2 Program execution

The codec for Enhanced Voice Services is implemented in two programs:

- EVS_cod: encoder;
- EVS_dec: decoder.

The programs should be called like:

- EVS_cod [encoder options] <input file> <bitstream file>;
- EVS_dec [decoder options]<bitstream file> < output file>.

The input and output files contain 16-bit linear encoded PCM samples and the bitstream file contains encoded data.

The encoder and decoder options will be explained by running the programs without any input arguments. See the file readme.txt for more information on how to run the *encoder* and *decoder* programs.

5 File formats

This clause describes the file formats used by the encoder and decoder programs. The test sequences defined in [1] also use the file formats described here.

5.1 Input/output file format

Input files read by the encoder and output files written by the decoder consist of 16-bit integer words per each data sample. The byte order in each word depends on the host architecture (e.g. LSB first on PCs, etc.). Both the encoder and the decoder program process complete frames corresponding to multiples of 20 ms. The remaining samples are discarded.

This means that e.g. for 16 kHz the encoder will only process n*320 frames even if the length of the input file is n*320 + k words.In this case, the decoder will produce an output fileof n*320 words.

5.2 Rate switching profile

The encoder program can optionally read in a rate switching profile file which specifies the encoding bitrate for each frame of the input data. The rate switching profile is a binary file, generated by 'gen-rate-profile' tool, which is part of STL 2009, as contained in ITU-T G.191 [10]. The rate switching profile contains 32-bit integer words where each word represents the encoding bitrate for each particular frame. The rate switching profile is recycled if it contains less entries than the total number of frames in the input file.

5.3 Parameter bitstream file (encoder output / decoder input)

The files produced by the speech/audio encoder/expected by the speech decoder contain an arbitrary number of frames in the following available formats.

5.3.1 ITU-T G.192 compliant format

SYNC_WORD	DATA_LENGTH	B1	B2	…	Bnn

Each box corresponds to one `Word16` value in the bitstream file, for a total of 2+nn words or 4+2nn bytes per frame, where nn is the number of encoded bits in the frame. Each encoded bit is represented as follows: Bit 0 = 0x007f, Bit 1 = 0x0081. The fields have the following meaning:

- `SYNC_WORD:` Word to ensure correct frame synchronization between the encoder and the decoder. It is also used to indicate the occurrences of bad frames.

  In the encoder output: (0x6b21)

  In the decoder input: Good frames (0x6b21)

  Bad frames (0x6b20)

- `DATA_LENGTH:` Length of the speech data. Codec mode and frame type is extracted in the decoder using this parameter

5.4 VoIP parameter bitstream file (decoder input)

Packet size	Arrival time	RTP header	G.192 format (see 6.3.1)

The fields have the following size and meaning:

- Packet size: 32 bit unsigned integer (= 12 + 2 + DATA_LENGTH).

- Arrival time: 32 bit unsigned integer in ms.

- RTP header: 96 bits (see RFC 3550 [9]), including RTP timestamp and SSRC.

Annex A (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2014-12	66	SP-140729			Presented to TSG SA#66 for approval		1.0.0
2014-12	66					1.0.0	12.0.0

History

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