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TECHNICAL SPECIFICATION

Digital cellular telecommunications system (Phase 2+); Full rate speech; Comfort noise aspect for full rate speech traffic channels (3GPP TS 46.012 version 13.0.0 Release 13)

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GLOBAL SYSTEM FOR MOBILE COMMUNICATIONS

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Contents

Foreword

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y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

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1 Scope

The present document gives the detailed requirements for the correct operation of the background acoustic noise evaluation, noise parameter encoding/decoding and comfort noise generation in GSM Mobile Stations (MS)s and Base Station Systems (BSS)s during Discontinuous Transmission (DTX) on full rate speech traffic channels.

The requirements described in the present document are mandatory for implementation in all GSM MSs. The receiver requirements are mandatory for implementation in all GSM BSSs, the transmitter requirements only for those where downlink DTX will be used.

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.
- For a specific reference, subsequent revisions do not apply.
- 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] GSM 01.04: "Digital cellular telecommunications system (Phase 2+); Abbreviations and acronyms".

[2] GSM 05.03: "Digital cellular telecommunications system (Phase 2+); Channel coding".

[3] GSM 06.10: "Digital cellular telecommunications system (Phase 2+); Full rate speech; Transcoding".

[4] GSM 06.31: "Digital cellular telecommunications system (Phase 2+); Full rate speech; Discontinuous Transmission (DTX) for full rate speech traffic channel".

3 Definitions and abbreviations

Definitions and abbreviations used in the present document are listed in GSM 01.04 [1].

The definitions of terms used in this technical specification can be found in GSM 06.31 [4].

4 General

The overall operation of Discontinuous Transmission is described in GSM 06.31 [4].

A basic problem when using DTX is that the background acoustic noise, which is transmitted together with the speech, would disappear when the radio transmission is cut, resulting in a modulation of the background noise. Since the DTX switching can take place rapidly, it has been found that this effect can be very annoying for the listener - especially in a car environment with high background noise levels. In bad cases the speech may be hardly intelligible.

The present document specifies the way to overcome this problem by generating on the receive side synthetic noise similar to the transmit side background noise. The parameters of this so called comfort noise are estimated on the transmit side and transmitted to the receive side before the radio transmission is cut and at a regular low rate afterwards. This allows the comfort noise to adapt to the changes of the noise on the transmit side.

5 Functions on the transmit side

The comfort noise evaluation algorithm uses the unquantized block amplitude and Log Area Ratio (LAR) parameters of the full rate speech encoder, defined in subclauses 4.2.15 and 4.2.6 of GSM 06.10 [3]. These parameters give information on the level and the spectrum of the background noise, respectively.

The evaluated comfort noise parameters are encoded into a special frame, called a SID (Silence Descriptor) frame, for transmission to the receive side.

The SID frame also serves to initiate the comfort noise generation on the receive side, as a SID frame is always sent at the end of a speech burst, i.e. before the radio transmission is cut.

The scheduling of SID or speech frames on the radio path is described in GSM 06.31 [4].

5.1 Background acoustic noise evaluation

The comfort noise parameters to be encoded into a SID frame are calculated over N=4 consecutive frames marked with VAD=0, as follows:

The Log Area Ratio parameters shall be averaged according to the equation:

$$mean(LAR(i)) = \frac{1}{N}\sum_{n=1}^{N} LAR[j-n](i) \qquad i = 1,2..8$$

where LAR[j](i) is the i'th Log Area Ratio coefficient of the current frame j and j-n indicates the previous frames.

The block amplitude parameter shall be averaged according to the equation:

$$mean(x_{\max}) = \frac{1}{(4N)}\sum_{n=1}^{N}\sum_{i=1}^{4} x_{\max}[j-n](i)$$

where $x_{max}[j](i)$ is the block amplitude in sub-segment i of the current frame. The SID frame containing these averaged parameters is passed to the Radio Subsystem instead of frame number j.

5.2 SID-frame encoding

The SID-frame encoding algorithm exploits the fact that only some of the 260 bits in a frame are needed to code the comfort noise parameters. The other bits can then be used to mark the SID-frame by means of a fixed bit pattern, called the SID code word.

The log area ratio coefficients are replaced by the mean (LAR(i)) values defined above and encoded as described in GSM 06.10 [3].

The block amplitude values are replaced by the mean (x_{max}) value defined above, repeated four times inside the frame and encoded as described in GSM 06.10 [3].

The SID code word consists of 95 bits which are all zero. The bits of the SID code word are inserted in the SID field defined as the positions of those 95 bits of the encoded RPE-pulses Xmc, which are in the error protection class I (see GSM 05.03 [2], table 2).

The remaining bits in the SID frame are set to zero. The use of these bits is for further study.

6 Functions on the receive side

The situations in which comfort noise shall be generated on the receive side are defined in GSM 06.31 [4]. Generally speaking, the comfort noise generation is started or updated whenever a valid SID frame is received.

6.1 Comfort noise generation and updating

The comfort noise generation procedure uses the RPE-LTP speech decoder algorithm defined in GSM 06.10 [3].

When comfort noise is to be generated, then the various encoded parameters are set as follows.

The RPE pulses (X_{mcr}) are replaced by a locally generated random integer sequence, uniformly distributed between 1 and 6.

Also the grid position parameters (M_{cr}) are set to random integer values, uniformly distributed between 0 and 3.

The LTP gain values (b_{cr}) are set to 0.

The LTP lag values (N_{cr}) of the 4 sub-segments are set to 40, 120, 40 and 120 respectively.

The 4 block amplitude values (X_{maxcr}) used are those received in the SID frame.

The log area ratio parameters (LAR_{cr}) used are those received in the SID frame.

With these parameters, the speech decoder now performs the standard operations described in GSM 06.10 [3] and synthesizes comfort noise.

Updating of the comfort noise parameters occurs each time a valid SID frame is received, as described in GSM 06.31 [4].

When updating the comfort noise, the parameters above should preferably be interpolated over a few frames to obtain smooth transitions.

Annex A (informative): Change history

Change history					
SMG No.	TDoc. No.	CR. No.	Section affected	New version	Subject/Comments
SMG#07				4.0.4	ETSI Publication
SMG#20				5.0.1	Release 1996 version
SMG#27				6.0.0	Release 1997 version
SMG#29				7.0.0	Release 1998 version
				7.0.1	Version update to 7.0.1 for Publication
SMG#31				8.0.0	Release 1999 version

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
03-2001	11				Version for Release 4		4.0.0
06-2001	12	SP-010304	001		Corrections of the formula for averaging Xmax	4.0.0	4.1.0
06-2002	16				Version for Release 5	4.1.0	5.0.0
12-2006	26				Version for Release 6	5.0.0	6.0.0
06-2007	36				Version for Release 7	6.0.0	7.0.0
12-2008	42				Version for Release 8	7.0.0	8.0.0
12-2009	46				Version for Release 9	8.0.0	9.0.0
03-2011	51				Version for Release 10	9.0.0	10.0.0
09-2012	57				Version for Release 11	10.0.0	11.0.0
09-2014	65				Version for Release 12	11.0.0	12.0.0
12-2015	70				Version for Release 13	12.0.0	13.0.0

History

Document history		
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