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Digital cellular telecommunications system (Phase 2+) (GSM); Comfort noise aspects for Adaptive Multi-Rate (AMR) speech traffic channels (GSM 06.92 version 7.1.1 Release 1998)

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Foreword

This European Standard (Telecommunications series) has been produced by Special Mobile Group (SMG).

The present document introduces the Adaptive Multi-Rate (AMR) speech traffic channels within the digital cellular telecommunications system.

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

Version 7.x.y

where:

7 indicates Release 1998 of GSM Phase 2+

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

y the third digit is incremented when editorial only changes have been incorporated in the specification.

National transposition dates	
Date of adoption of this EN:	3 December 1999
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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 Mobile Stations (MSs) and Base Station Systems (BSSs) during Discontinuous Transmission (DTX) on adaptive multi-rate full rate and half rate speech traffic channels.

The requirements described in the present document are mandatory for implementation in all GSM MSs capable of supporting the adaptive multi-rate full rate and half tate speech traffic channels.

The receiver requirements are mandatory for implementation in all GSM BSSs capable of supporting the adaptive multi-rate full rate and half rate speech traffic channels, the transmitter requirements only for those where downlink DTX will be used.

In case of discrepancy between the requirements described in the present document and the fixed point computational description of these requirements contained in GSM 06.73 [2], the description in GSM 06.73 [2] will prevail.

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.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.
- For this Release 1998 document, references to GSM documents are for Release 1998 versions (version 7.x.y).

[1] GSM 01.04: "Digital cellular telecommunication system (Phase 2+); Abbreviations and acronyms".

[2] GSM 06.73: "Digital cellular telecommunications system (Phase 2+); ANSI-C code for the GSM Adaptive Multi-Rate speech codec".

[3] GSM 06.90: "Digital cellular telecommunications system (Phase 2+); Adaptive Multi-Rate speech transcoding".

[4] GSM 06.91: "Digital cellular telecommunications system (Phase 2+); Substitution and muting of lost frame for Adaptive Multi-Rate speech traffic channels".

[5] GSM 06.93: "Digital cellular telecommunications system (Phase 2+); Discontinuous transmission (DTX) for Adaptive Multi-Rate speech traffic channels".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purpose of the present document, the following terms and definitions apply.

Frame: time interval of 20 ms corresponding to the time segmentation of the adaptive multi-rate speech transcoder, also used as a short term for traffic frame.

SID frames: special SID (Silence Descriptor) frames. It may convey information on the acoustic background noise or inform the decoder that it should start generating background noise.

Speech frame: traffic frame that cannot be classified as a SID frame.

VAD flag: voice Activity Detection flag.

TX_TYPE: one of SPEECH, SID_FIRST, SID_UPD, NO_DATA (defined in GSM 06.93).

RX_TYPE: classification of the received traffic frame (defined in GSM 06.93).

Other definitions of terms used in the present document can be found in GSM 06.90 [3] and GSM 06.93 [5]. The overall operation of DTX is described in GSM 06.93 [5].

3.2 Symbols

For the purpose of the present document, the following symbols apply. Boldface symbols are used for vector variables.

$\mathbf{f}^T = [f_1 \ f_2 \ \ldots \ f_{10}]$	Unquantized LSF vector
$\hat{\mathbf{f}}^T = [\hat{f}_1 \ \hat{f}_2 \ \ldots \ \hat{f}_{10}]$	Quantized LSF vector
$\mathbf{f}^{(m)}$	Unquantized LSF vector of frame m
$\hat{\mathbf{f}}^{(m)}$	Quantized LSF vector of frame m
$\mathbf{f}^{mean}$	Averaged LSF parameter vector
$en_{\log}$	Logarithmic frame energy
$en_{\log}^{mean}$	Averaged logarithmic frame energy
$\hat{\mathbf{f}}^{ref}$	Reference vector for LSF quantization
$\mathbf{e}$	Computed LSF parameter prediction residual
$\hat{\mathbf{e}}$	Quantized LSF parameter prediction residual
$\sum_{n=a}^{b} x(n)$	$= x(a) + x(a+1) + \ldots + x(b-1) + x(b)$

3.3 Abbreviations

For the purpose of the present document, the following abbreviations apply.

AMR	Adaptive Multi-Rate
BSS	Base Station Subsystem
DTX	Discontinuous Transmission
MS	Mobile Station
SID	SIlence Descriptor
LP	Linear Prediction
LSP	Line Spectral Pair
LSF	Line Spectral Frequency
RX	Receive
TX	Transmit
VAD	Voice Activity Detector

For abbreviations not given in this subclause, see GSM 01.04 [1].

4 General

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 discontinuities 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 (RX) side synthetic noise similar to the transmit (TX) side background noise. The comfort noise parameters are estimated on the TX side and transmitted to the RX side before the radio transmission is switched off and at a regular rate afterwards. This allows the comfort noise to adapt to the changes of the noise on the TX side.

5 Functions on the transmit (TX) side

The comfort noise evaluation algorithm uses the following parameters of the AMR speech encoder, defined in GSM 06.90 [3]:

- the unquantized Linear Prediction (LP) parameters, using the Line Spectral Pair (LSP) representation, where the unquantized Line Spectral Frequency (LSF) vector is given by $\mathbf{f}^T = [f_1 \; f_2 \ldots f_{10}]$;
- the unquantized LSF vector for the 12.2 kbit/s mode is given by the second set of LSF parameters in the frame.

The algorithm computes the following parameters to assist in comfort noise generation:

- the averaged LSF parameter vector $\mathbf{f}^{mean}$ (average of the LSF parameters of the eight most recent frames);
- the averaged logarithmic frame energy $en_{\log}^{mean}$ (average of the logarithmic energy of the eight most recent frames).

These parameters give information on the level ($en_{\log}^{mean}$) and the spectrum ($\mathbf{f}^{mean}$) of the background noise.

The evaluated comfort noise parameters ($\mathbf{f}^{mean}$ and $en_{\log}^{mean}$) are encoded into a special frame, called a Silence Descriptor (SID) frame for transmission to the RX side.

A hangover logic is used to enhance the quality of the silence descriptor frames. A hangover of 7 frames is added to the VAD flag so that the coder waits with the switch from active to inactive mode for a period of 7 frames, during that time the decoder can compute a silence descriptor frame from the quantized LSFs and the logarithmic frame energy of the decoded speech signal. Therefore, no comfort noise description is transmitted in the first SID frame after active speech. If the background noise contains transients which will cause the coder to switch to active mode and then back to inactive mode in a very short timeperiod, no hangover is used. Instead the previously used comfort noise frames are used for comfort noise generation.

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

The scheduling of SID or speech frames on the radio path is described in GSM 06.93 [5].

5.1 LSF evaluation

The comfort noise parameters to be encoded into a SID frame are calculated over $N = 8$ consecutive frames marked with VAD=0, as follows:

The averaged LSF parameter vector $\mathbf{f}^{mean}(i)$ of the frame i shall be computed according to the equation: