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GLOBAL SYSTEM FOR
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Digital cellular telecommunications system; Comfort noise aspects for Enhanced Full Rate (EFR) speech traffic channels (GSM 06.62)

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Foreword

This draft European Telecommunication Standard (ETS) has been produced by the Special Mobile Group (SMG) Technical Committee of the European Telecommunications Standards Institute (ETSI), and is now submitted for the Public Enquiry phase of the ETSI standards approval procedure.

This draft ETS defines 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 enhanced full rate speech traffic channels within the digital cellular telecommunications system.

This draft ETS corresponds to GSM technical specification, GSM 06.62, version 5.0.0

Proposed transposition dates	
Date of latest announcement of this ETS (doa):	3 months after ETSI publication
Date of latest publication of new National Standard or endorsement of this ETS (dop/e):	6 months after doa
Date of withdrawal of any conflicting National Standard (dow):	6 months after doa

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1 Scope

This draft European Telecommunication Standard (ETS) 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 enhanced full rate speech traffic channels.

The requirements described in this ETS are mandatory for implementation in all GSM MSs capable of supporting the enhanced full rate speech traffic channel.

The receiver requirements are mandatory for implementation in all GSM BSSs capable of supporting the enhanced full rate speech traffic channel, the transmitter requirements only for those where downlink DTX will be used.

In case of discrepancy between the requirements described in this ETS and the fixed point computational description of these requirements contained in GSM 06.53, the description in GSM 06.53 will prevail.

2 Normative references

This ETS incorporates by dated and undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this ETS only when incorporated in it by amendment or revision. For undated references, the latest edition of the publication referred to applies.

[1] GSM 01.04 (ETR 100): "Digital cellular telecommunication system (Phase 2); Abbreviations and acronyms".

[2] GSM 06.53 (prETS 300 724): "Digital cellular telecommunications system; ANSI-C code for the GSM Enhanced Full Rate (EFR) speech codec".

[3] GSM 06.54 (Work item DE/SMG-020654 prETS 300 725): "Digital cellular telecommunications system (Phase 2); Test vectors for the GSM Enhanced Full Rate (EFR) speech codec".

[4] GSM 06.60 (prETS 300 726): "Digital cellular telecommunications system; Enhanced Full Rate (EFR) speech transcoding".

[5] GSM 06.61 (prETS 300 727): "Digital cellular telecommunications system; Substitution and muting of lost frame for Enhanced Full Rate (EFR) speech traffic channels".

[6] GSM 06.81 (prETS 300 729): "Digital cellular telecommunications system; Discontinuous transmission (DTX) for Enhanced Full Rate (EFR) speech traffic channels".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purpose of this ETS, the following definitions apply.

Frame: Time interval of 20 ms corresponding to the time segmentation of the enhanced full rate speech transcoder, also used as a short term traffic frame.

SID frame: Frame characterised by the SID (SIlence Descriptor) codeword. It conveys information on the acoustic background noise.

SID codeword: Fixed bit pattern for labelling a traffic frame as a SID frame.

SID field: The bit positions of the SID codeword within a SID frame.

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

VAD flag: Voice Activity Detection flag.

SP flag: SPeech flag.

$W(z)$**:** Spectral weighting filter of the GSM enhanced full rate speech codec.

$H(z)$**:** Combination of the short term (spectral) filter $A(z)$ and the spectral weighting filter $W(z)$.

Other definitions of terms used in this ETS can be found in GSM 06.60 and GSM 06.81. The overall operation of DTX is described in GSM 06.81.

3.2 Symbols

For the purpose of this ETS, the following symbols apply. Boldface symbols are used for vector variables.

$\mathbf{f}^T = [f_1\ f_2\ \dots\ f_{10}]$	Unquantized LSF vector
$\hat{\mathbf{f}}^T = [\hat{f}_1\ \hat{f}_2\ \dots\ \hat{f}_{10}]$	Quantized LSF vector
$\mathbf{f}^{(m)}$	*m*th unquantized LSF vector of the frame
$\hat{\mathbf{f}}^{(m)}$	*m*th quantized LSF vector of the frame
$\hat{\mathbf{f}}^{ref}$	Reference LSF parameter vector
$\mathbf{f}^{mean}$	Averaged LSF parameter vector
g_c	Unquantized fixed codebook gain
$\hat{g}_c$	Quantized fixed codebook gain
$\hat{g}_c^{ref}$	Reference fixed codebook gain
g_c^{mean}	Averaged fixed codebook gain
e_{LP}	Linear prediction residual signal

$\mathbf{e}$	Computed LSF parameter prediction residual
$\hat{\mathbf{e}}$	Quantized LSF parameter prediction residual
γ	Computed fixed codebook gain correction factor
$\hat{\gamma}$	Quantized fixed codebook gain correction factor
$\sum_{n=a}^{b} x(n)$	$= x(a) + x(a+1) + \ldots + x(b-1) + x(b)$

3.3 Abbreviations

For the purpose of this ETS, the following abbreviations apply.

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.

This ETS 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 low 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 GSM enhanced full rate speech encoder, defined in GSM 06.60:

- the unquantized and quantized Linear Prediction (LP) parameters, using the Line Spectral Pair (LSP) representation, where the unquantized Line Spectral Frequency (LSF) vector is given by $\mathbf{f}^T = \left[f_1 \ f_2 \ \ldots \ f_{10} \right]$, the quantized LSF vector is given by $\hat{\mathbf{f}}^T = \left[\hat{f}_1 \ \hat{f}_2 \ \ldots \ \hat{f}_{10} \right]$, and the two sets of unquantized and quantized LSF vectors (one for each half of a frame) are given by $\mathbf{f}^{(1)}$, $\mathbf{f}^{(2)}$, $\hat{\mathbf{f}}^{(1)}$ and $\hat{\mathbf{f}}^{(2)}$, respectively;

- the quantized fixed-codebook gain $\hat{g}_c$.

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

- the reference LSF parameter vector $\hat{\mathbf{f}}^{ref}$ (average of the quantized LSF parameters of the hangover period);

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

- the reference fixed codebook gain $\hat{g}_c^{ref}$ (average of the quantized fixed codebook gain values of the hangover period);

- the averaged fixed codebook gain g_c^{mean} (average of the fixed codebook gain values of the eight most recent frames);

- the unquantized fixed codebook gain g_c.

These parameters give information on the level (g_c, $\hat{g}_c$, $\hat{g}_c^{ref}$, g_c^{mean}) and the spectrum ($\mathbf{f}^{(1)}$, $\mathbf{f}^{(2)}$, $\hat{\mathbf{f}}^{(1)}$, $\hat{\mathbf{f}}^{(2)}$, $\hat{\mathbf{f}}^{ref}$, $\mathbf{f}^{mean}$) of the background noise.

Two of the evaluated comfort noise parameters ($\mathbf{f}^{mean}$ and g_c^{mean}) are encoded into a special frame, called a Silence Descriptor (SID) frame, for transmission to the RX side. Since the reference LSF parameter vector $\hat{\mathbf{f}}^{ref}$ and the reference fixed codebook gain $\hat{g}_c^{ref}$ can be evaluated in the same way in the encoder and decoder, as given in subclause 5.1, no transmission of these parameters is necessary.

The averaged LSF parameter and fixed codebook gain values, $\mathbf{f}^{mean}$ and g_c^{mean}, are computed in the encoder using both quantized and unquantized parameter values if the period of the eight most recent frames (the SID averaging period) is overlapping with the hangover period (the parameters from the frames overlapping with the hangover period have quantized values, while the parameters of the more recent frames of the SID averaging period have unquantized values). If the period of the eight most recent frames is non-overlapping with the hangover period, the averaged LSF parameter and fixed codebook gain values are computed using only unquantized parameter values.

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 terminated.

The scheduling of SID or speech frames on the radio path is described in GSM 06.81.

5.1 Background acoustic noise 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:

$$\mathbf{f}^{mean}(i) = \frac{1}{8}\sum_{n=0}^{7}\left(\frac{1}{2}\sum_{m=1}^{2}\mathbf{f}^{(m)}(i-n)\right) \tag{1}$$

where:

$\mathbf{f}^{(m)}(i)$ is the mth (unquantized) LSF parameter vector of the current frame i (n=0);

$\mathbf{f}^{(m)}(i-n)$ is the mth (quantized or unquantized) LSF parameter vector of one of the last frames (n = 1,..., 7);

n is the averaging period index (n = 0, 1,..., 7);

m is the LSF parameter vector index within a frame (1 or 2);

i is the frame index.

NOTE: When the averaging is performed at the end of the hangover period (first SID update), all of the LSF parameter vectors $\mathbf{f}^{(m)}(i-n)$ of the 7 previous frames (the hangover period) have quantized values, while the LSF parameter vectors $\mathbf{f}^{(m)}(i)$ of the current frame i have unquantized values. In the subsequent SID updates, the LSF parameter vectors of the SID averaging period in the frames overlapping with the hangover period have quantized values, while the parameter vectors of the more recent frames of the SID averaging period have unquantized values.

The averaged LSF parameter vector $\mathbf{f}^{mean}(i)$ of the frame i is encoded using the same encoding tables that are also used by the GSM enhanced full rate speech codec for the encoding of the non-averaged LSF parameter vectors in ordinary speech encoding mode, but the quantization algorithm is modified in order to support the quantization of comfort noise. The LSF parameter prediction residual to be quantized is obtained according to the following equation:

$$\mathbf{e}(i) = \mathbf{f}^{mean}(i) - \hat{\mathbf{f}}^{ref} \tag{2}$$

where:

$\mathbf{f}^{mean}(i)$ is the averaged LSF parameter vector at the current frame *I*

$\hat{\mathbf{f}}^{ref}$ is the reference LSF parameter vector

$\mathbf{e}(i)$ is the computed LSF parameter prediction residual at the current frame *I*

i is the frame index;

NOTE: This prediction residual is used for both halves of the frame in the quantization algorithm. The computation of the reference LSF parameter vector $\hat{\mathbf{f}}^{ref}$ is made on the basis of the quantized LSF parameters, $\hat{\mathbf{f}}^{(1)}$ and $\hat{\mathbf{f}}^{(2)}$, by averaging the parameters over the hangover period of 7 frames, according to the following equation:

$$\hat{\mathbf{f}}^{ref} = \frac{1}{7}\sum_{n=1}^{7}\left(\frac{1}{2}\sum_{m=1}^{2}\hat{\mathbf{f}}^{(m)}(i-n)\right) \tag{3}$$

where:

$\hat{\mathbf{f}}^{(m)}(i-n)$ is the mth quantized LSF parameter vector of one of the frames of the hangover period (n = 1,..., 7);

n is the hangover period frame index (n = 1,..., 7);

m is the LSF parameter index within a frame (1 or 2);

i is the frame index.

For each comfort noise insertion period, the computation of the reference LSF parameter vector $\hat{\mathbf{f}}^{ref}$ is done only once at the end of the hangover period and for the rest of the comfort noise insertion period $\hat{\mathbf{f}}^{ref}$ will be frozen. The reference LSF parameter vector $\hat{\mathbf{f}}^{ref}$ is evaluated in the decoder in the same way as in the encoder, because during the hangover period the same LSF parameter vectors $\hat{\mathbf{f}}^{(m)}$ are available at the encoder and decoder. An exception to this are the cases when transmission errors are severe enough to cause the parameters to become unusable, and the frame substitution procedure is activated (see GSM 06.61). In these cases, the modified parameters obtained from the frame substitution procedure are used instead of the received parameters.

The fixed codebook gain values shall be averaged and updated in every subframe according to the equation:

$$g_c^{mean} = \frac{1}{29}\sum_{n=0}^{28} g_c(-n) \tag{4}$$

where:

$g_c(0)$ is the (unquantized) fixed codebook gain in the current subframe (n=0);

$g_c(-n)$ is the (quantized or unquantized) fixed codebook gain in one of the past subframes (n = 1,..., 28);

n is the averaging period index (n = 0, 1,..., 28);

NOTE: When the averaging is started at the end of the hangover period (first SID update), all of the fixed codebook gains $g_c(-n)$ of the 28 (n=1,..., 28) previous subframes (the hangover period) have quantized values, while the fixed codebook gains $g_c(0)$ of the current subframe has an unquantized value. In the subsequent SID updates, the fixed codebook gain values of the SID averaging period in the subframes overlapping with the hangover period have quantized values, while the parameter vectors of the more recent subframes of the SID averaging period have unquantized values.

Since most parts of the subframe processing section in the encoder are switched off when the SP flag = "0" (to minimise the average complexity of the speech encoder algorithm), the unquantized fixed codebook gain is not directly available for gain averaging. Due to this, the unquantized fixed codebook gain is separately computed, based on the energy of the LP residual signal in each subframe, according to the following equation:

$$g_c(j) = \sqrt{\frac{\sum_{l=1}^{40} \left(e_{LP}(j)(l)\right)^2}{10}} \tag{5}$$

where:

$g_c(j)$ is the unquantized fixed codebook gain of the current subframe j;

$e_{LP}(j)(l)$ is the lth sample of the LP residual in the current subframe j;

j is the subframe index (j = 1,..., 4);

l is the sample index (l = 1,..., 40).

NOTE: The computed energy of the LP residual signal is divided by the value of 10 to yield the energy for one excitation pulse, since during comfort noise generation, the subframe excitation signal (pseudo noise) has 10 non-zero samples, whose amplitudes can take values of +1 or -1.

The averaged fixed codebook gain value g_c^{mean} of the current subframe is encoded using the non-averaged fixed codebook gain values in ordinary speech encoding mode, but the quantization algorithm is modified in order to support comfort noise quantization. The fixed codebook gain correction factor γ to be quantized is obtained according to the following equation:

$$\gamma = g_c^{mean} / \hat{g}_c^{ref} \tag{6}$$

where:

g_c^{mean} is the averaged fixed codebook gain value in the current subframe;

$\hat{g}_c^{ref}$ is the reference fixed codebook gain;

The computation of the reference fixed codebook gain $\hat{g}_c^{ref}$ is made on the basis of the quantized fixed codebook gain parameters $\hat{g}_c$, by averaging the parameter values over the hangover period of 7 frames according to the following equation:

$$\hat{g}_c^{ref}(i) = \frac{1}{7} \sum_{n=1}^{7} \left(\frac{1}{4} \sum_{j=1}^{4} \hat{g}_c(i-n)(j) \right) \tag{7}$$

where:

$\hat{g}_c(i-n)(j)$	is the quantized fixed codebook gain parameter value in subframe j of one of the frames of the hangover period (n = 1,..., 7);
n	is the hangover period frame index (n = 1, 2,..., 7);
i	is the frame index;
j	is the subframe index (j = 1,..., 4);

For each comfort noise insertion period, the computation of the reference fixed codebook gain $\hat{g}_c^{ref}$ is done only once at the end of the hangover period and for the rest of the comfort noise insertion period $\hat{g}_c^{ref}$ will be frozen. The reference fixed codebook gain $\hat{g}_c^{ref}$ can be evaluated in the decoder in the same way as in the encoder, because during the hangover period the same quantized fixed codebook gain values $\hat{g}_c$ are available at the encoder and decoder. An exception to this are the cases when transmission errors are severe enough to cause the parameters to become unusable, and the frame substitution procedure is activated (see GSM 06.61). In these cases, the modified parameters obtained from the frame substitution procedure are used instead of the received parameters.

The hangover period is defined in GSM 06.81. It is a period added at the end of a speech burst in which no voice activity is detected (VAD flag = "0"), but the speech encoder stays for the processing of 7 speech frames in speech encoding mode (SP flag = "1"). This hangover period and the first SID frame are used for averaging the comfort noise parameters contained in the first SID frame.

5.2 Modification of the speech encoding algorithm during SID frame generation

When the SP flag is equal to "0" the speech encoding algorithm is modified in the following way:

- The non-averaged LP parameters which are used to derive the filter coefficients of the filters $H(z)$ and $W(z)$ of the speech encoder are not quantized;

- The open loop pitch lag search is performed, but the closed loop pitch lag search is inactivated. The adaptive codebook gain is set to zero.

- No fixed codebook search is made. In each subframe the pulse positions and signs of the fixed codebook excitation are locally generated using uniformly distributed pseudo random numbers. The excitation pulses take values of +1 and -1 when comfort noise is generated. The fixed codebook comfort noise excitation generation algorithm is defined in subclause 6.2.

- The memory of weighting filter $W(z)$ is set to zero, i.e., the memory of $W(z)$ is not updated.

- The ordinary LP parameter quantization algorithm is inactive. At the end of the hangover period the reference LSF parameter vector $\hat{\mathbf{f}}^{ref}$ is calculated as defined in subclause 5.1. For the rest of the comfort noise insertion period $\hat{\mathbf{f}}^{ref}$ is frozen. The averaged LSF parameter vector $\mathbf{f}^{mean}$ is calculated each time a new SID frame is to be sent to the Radio Subsystem. This parameter vector is encoded into the SID frame as defined in subclause 5.1.

- The ordinary fixed codebook gain quantization algorithm is inactive. At the end of the hangover period the reference fixed codebook gain $\hat{g}_c^{ref}$ is calculated as defined in subclause 5.1. For the rest of the comfort noise insertion period $\hat{g}_c^{ref}$ is frozen. The averaged fixed codebook gain value g_c^{mean} is calculated each time a new SID frame is to be sent to the Radio Subsystem. This gain value is encoded into the SID frame as defined in subclause 5.1.

- The predictor memories of the ordinary LP parameter quantization and fixed codebook gain quantization algorithms are reset when SP flag = "0", so that the quantizers start from their initial states when the speech activity begins again.

- The computation of the unquantized fixed codebook gain is performed based on the energy of the LP residual signal.

5.3 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 quantization indices of the LP parameters are replaced by the quantization indices derived from the averaged LSF parameter vector $\mathbf{f}^{mean}$. The encoding of the quantization indices is defined in subclause 5.1.

The fixed codebook gain quantization indices are replaced by the quantization index derived from the averaged fixed codebook gain value g_c^{mean}, encoded as defined in subclause 5.1, repeated four times inside the frame.

The SID code word consists of 95 bits which are all set to one. The bits of the SID code word are inserted in the SID field as defined in table 1. All of the bits in the SID field are in the error protection Class I.

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

Table 1: SID codeword

Parameter	Number of bits	Bit positions (b0 = LSB)
LTP LAG 1	2	b0, b1
LTP LAG 2	3	b0, b1, b2
LTP LAG 3	2	b0, b1
LTP LAG 4	4	b0, b1, b2, b3
LTP GAIN 1	3	b0, b1, b2
LTP GAIN 2	3	b0, b1, b2
LTP GAIN 3	4	b0, b1, b2, b3
LTP GAIN 4	4	b0, b1, b2, b3
PULSE 1 of 1st subfr.	4	b0, b1, b2, b3
PULSE 2 of 1st subfr.	4	b0, b1, b2, b3
PULSE 3 of 1st subfr.	4	b0, b1, b2, b3
PULSE 4 of 1st subfr.	4	b0, b1, b2, b3
PULSE 5 of 1st subfr.	2	b4, b5
PULSE 1 of 2nd subfr.	4	b0, b1, b2, b3
PULSE 2 of 2nd subfr.	4	b0, b1, b2, b3
PULSE 3 of 2nd subfr.	4	b0, b1, b2, b3
PULSE 4 of 2nd subfr.	4	b0, b1, b2, b3
PULSE 5 of 2nd subfr.	2	b4, b5
PULSE 1 of 3rd subfr.	4	b0, b1, b2, b3
PULSE 2 of 3rd subfr.	4	b0, b1, b2, b3
PULSE 3 of 3rd subfr.	4	b0, b1, b2, b3
PULSE 4 of 3rd subfr.	4	b0, b1, b2, b3
PULSE 5 of 3rd subfr.	2	b4, b5
PULSE 1 of 4th subfr.	4	b0, b1, b2, b3
PULSE 2 of 4th subfr.	2	b2, b3
PULSE 3 of 4th subfr.	4	b0, b1, b2, b3
PULSE 4 of 4th subfr.	4	b0, b1, b2, b3
PULSE 5 of 4th subfr.	2	b4, b5

The parameters in table 1 are defined in GSM 06.60.

6 Functions on the receive (RX) side

The situations in which comfort noise shall be generated on the receive side are defined in GSM 06.81. Generally speaking, the comfort noise generation is started or updated whenever a valid SID frame is received.

6.1 Averaging and decoding of the LP and fixed codebook gain parameters

When speech frames are received by the decoder the LP and the fixed codebook gain parameters of the last seven speech frames shall be kept in memory. The decoder counts the number of frames elapsed since the last SID frame was updated and passed to the RSS by the encoder. Based on this count, the decoder determines whether or not there is a hangover period at the end of the speech burst (if at least 31 frames have elapsed since the last SID update when the first SID frame after a speech burst arrives, the hangover period has existed at the end of the speech burst).

As soon as a SID frame is received, and the hangover period is detected at the end of the speech burst, the stored LP and fixed codebook gain parameters shall be averaged to obtain the reference LSF parameter vector $\hat{\mathbf{f}}^{ref}$ and the reference fixed gain codebook value g_c^{ref}. The reference LSF parameter vector and the reference fixed codebook gain value shall be frozen and used for the actual comfort noise insertion period.

The averaging procedure for obtaining the reference parameters is as follows:

- when a speech frame is received, the LSF and fixed codebook gain parameters are decoded and stored in memory;

- when the first SID frame is received, and the hangover period is detected at the end of the speech burst, the stored LSF and fixed codebook gain parameters are averaged in the same way as in the speech encoder as follows (see also subclause 5.1):

$$\hat{\mathbf{f}}^{ref} = \frac{1}{7}\sum_{n=1}^{7}\left(\frac{1}{2}\sum_{m=1}^{2}\hat{\mathbf{f}}^{(m)}(i-n)\right) \tag{8}$$

where:

$\hat{\mathbf{f}}^{(m)}(i-n)$ is the mth quantized LSF parameter vector of one of the frames of the hangover period (n = 1,..., 7);

n is the hangover period frame index (n = 1, 2,..., 7);

m is the LSF parameter index within a frame (1 or 2)

i is the frame index.

and

$$\hat{g}_c^{ref} = \frac{1}{7}\sum_{n=1}^{7}\left(\frac{1}{4}\sum_{j=1}^{4}\hat{g}_c(i-n)(j)\right) \tag{9}$$

where:

$\hat{g}_c(i-n)(j)$ is the quantized fixed codebook gain parameter value in subframe j of one of the frames of the hangover period (n = 1,..., 7);

n is the hangover period frame index (n = 1,..., 7);

i is the frame index;

j is the subframe index (j = 1,..., 4).

Once the reference LSF parameter vector has been computed, the averaged LSF parameter vector $\hat{\mathbf{f}}^{mean}(i)$ of the frame i (encoded into the SID frame) can be reproduced at the decoder each time a SID update frame is received, according to the equation:

$$\hat{\mathbf{f}}^{mean}(i) = \hat{\mathbf{e}}(i) + \hat{\mathbf{f}}^{ref} \tag{10}$$

where:

$\hat{\mathbf{f}}^{mean}(i)$ is the quantized, averaged LSF parameter vector at the current frame i to be used for comfort noise generation;

$\hat{\mathbf{f}}^{ref}$ is the reference LSF parameter vector;

$\hat{\mathbf{e}}(i)$ is the received quantized LSF parameter prediction residual at the current frame I;

i is the frame index.

The averaged fixed codebook gain $\hat{g}_c^{mean}(i)$ of the frame i (encoded into the SID frame) can be similarly reproduced at the decoder each time a SID update frame is received, according to the following equation:

$$\hat{g}_c^{mean}(i) = \hat{g}_c^{ref} \cdot \hat{\gamma}(i) \tag{11}$$

where:

$\hat{g}_c^{mean}(i)$ is the averaged fixed codebook gain value in the current frame i to be used for comfort noise generation;

$\hat{g}_c^{ref}$ is the reference fixed codebook gain;

$\hat{\gamma}(i)$ is the received quantized fixed codebook gain correction factor in the current frame I;

i is the frame index.

6.2 Comfort noise generation and updating

The comfort noise generation procedure uses the GSM enhanced full rate speech decoder algorithm defined in GSM 06.60.

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

In each subframe, the pulse positions and signs of the fixed codebook excitation are locally generated using uniformly distributed pseudo random numbers. The excitation pulses take values of +1 and -1 when comfort noise is generated. The fixed codebook comfort noise excitation generation algorithm works as follows:

```
for (i = 0; i < 40; i++)          code[i] = 0;

for (i = 0; i < 10; i++) {

    j = random(4);

    idx = j * 10 + I;

    if (random(2) == 1) code[idx] =  1;

    else                code[idx] = -1;

}
```

where:

`code[0..39]` fixed codebook excitation buffer;

`random(4)` generates a random integer value, uniformly distributed between 0 and 3;

`random(2)` generates a random integer value, uniformly distributed between 0 and 1.

The fixed codebook gain values are those received in the SID frame.

The adaptive codebook gain values in each subframe are set to 0.

The pitch delay values in each subframe are set to 40.

The LP filter parameters used are those received in the SID frame.

The predictor memories of the ordinary LP parameter and fixed codebook gain quantization algorithms are reset when SP flag = "0", so that the quantizers start from their initial states when the speech activity begins again.

With these parameters, the speech decoder now performs the standard operations described in GSM 06.60 and synthesizes comfort noise.

Updating of the comfort noise parameters (fixed codebook gain and LP filter parameters) occurs each time a valid SID frame is received, as described in GSM 06.81.

When updating the comfort noise, the parameters above should preferably be interpolated over the SID update period to obtain smooth transitions.

7 Computational details

A low level description has been prepared in form of an ANSI C source code which is part of GSM 06.53.

History

Document history			
March 1996	Public Enquiry	PE 103:	1996-03-04 to 1996-06-28