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European Standard (Telecommunications series)

Digital cellular telecommunications system (Phase 2+); Substitution and muting of lost frames for Adaptive Multi Rate (AMR) speech traffic channels (GSM 06.91 version 7.1.0 Release 1998)

GLOBAL SYSTEM FOR MOBILE COMMUNICATIONS

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Contents

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Foreword

This European Standard (Telecommunications series) has been produced by ETSI Technical Committee Special Mobile Group (SMG), and is now submitted for the ETSI standards One-step Approval Procedure.

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.

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

1 Scope

The present document defines a frame substitution and muting procedure which shall be used by the Receive (RX) Discontinuous Transmission (DTX) handler when one or more lost speech or Silence Descriptor (SID) frames are received from the Radio Sub System (RSS).

The requirements of the present document are mandatory for implementation in all GSM Base Station Systems (BSS)s and Mobile Stations (MS)s capable of supporting the AMR speech traffic channel. It is not mandatory to follow the bit exact implementation outlined in the present document and the corresponding C source code.

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

[2] GSM 06.90: "Digital cellular telecommunications system (Phase 2+); AMR speech transcoding".

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

[4] GSM 08.60: "Digital cellular telecommunications system (Phase 2+); Inband control of remote transcoders and rate adaptors for AMR, Enhanced Full Rate (EFR) and Full Rate traffic channels".

[5] GSM 08.61: "Digital cellular telecommunications system (Phase 2+); Inband control of remote transcoders and rate adaptors for Half Rate traffic channels".

3 Definitions and abbreviations

3.1 Definitions

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

N-point median operation: Consists of sorting the N elements belonging to the set for which the median operation is to be performed in an ascending order according to their values, and selecting the (int (N/2) + 1) -the largest value of the sorted set as the median value.

Further definitions of terms used in the present document can be found in GSM 06.90 [2], GSM 06.93 [3], GSM 05.03 [1] and GSM 08.60 [4].

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

BFI	Bad Frame Indication from Radio Sub System
BSI_Abis	Bad Sub-block Indication obtained from A-bis CRC checks
prevBFI	Bad Frame Indication of previous frame
PDFI	Potentially Degraded Frame Indication from RSS
RSS	Radio Sub System
RX	Receive
DTX	Discontinuous Transmission
SID	SIlence Descriptor frame (Background descriptor)
CRC	Cyclic Redundancy Check
CCU	Channel Coding Unit
ECU	Error Concealment Unit
BFH	Bad Frame Handling
medianN	N-point median operation
TRAU	Transcoding Rate Adaptation Unit

4 General

The purpose of frame substitution is to conceal the effect of lost frames. The purpose of muting the output in the case of several lost frames is to indicate the breakdown of the channel to the user and to avoid generating possible annoying sounds as a result from the frame substitution procedure.

The RSS indicates lost speech or lost SID frames by setting its Bad Frame Indication flag (BFI) based on CRCs and possibly other error detection mechanisms. The TRAU calculates from the CRCs inserted by the CCU in the TRAU frames one BSI_Abis flag for every sub-block of speech parameters. If either one or both of these flags are set, the speech decoder shall perform parameter substitution to conceal errors.

The RSS also indicates potentially degraded frames using the flag PDFI. This flag is derived from the soft output of the channel decoder. It may be used by the speech decoder selectively depending on the estimated signal type.

The example solutions provided in paragraphs 6 and 7 apply only to bad frame handling on a complete speech frame basis. However some parts could be modified for substitution of only bad sub-blocks.

5 Requirements

5.1 Error detection

An error is detected and the BFI flag is set by the RSS.0. The PDFI flag is set appropriately using the soft output from the channel decoder.

5.2 Lost speech frames

Normal decoding of lost speech frames would result in very unpleasant noise effects. In order to improve the subjective quality, lost speech frames shall be substituted with either a repetition or an extrapolation of the previous good speech frame(s). This substitution is done so that it gradually will decrease the output level, resulting in silence at the output. Subclauses 6, and 7 provide example solutions.

5.3 First lost SID frame

A lost SID frame shall be substituted by using the SID information from earlier received valid SID frames and the procedure for valid SID frames be applied as described in GSM 06.93 [3].

5.4 Subsequent lost SID frames

For many subsequent lost SID frames, a muting technique shall be applied to the comfort noise that will gradually decrease the output level. For subsequent lost SID frames, the muting of the output shall be maintained. Subclauses 6 and 7 provide example solutions.

6 Example ECU/BFH Solution 1

The C code of the following example is embedded in the bit exact software of the codec. In the code the ECU is designed to allow subframe-by-subframe synthesis, thereby reducing the speech synthesis delay to a minimum.

6.1 State Machine

This example solution for substitution and muting is based on a state machine with seven states (Figure 1).

The system starts in state 0. Each time a bad frame is detected, the state counter is incremented by one and is saturated when it reaches 6. Each time a good speech frame is detected, the state counter is reset to zero, except when we are in state 6, where we set the state counter to 5. The state indicates the quality of the channel: the larger the value of the state counter, the worse the channel quality is. The control flow of the state machine can be described by the following C code (**BFI** = bad frame indicator, **State** = state variable):

```
if(BFI != 0 )
    State = State + 1;
else if(State == 6)
    State = 5;
else
    State = 0;
if(State > 6 )
    State = 6;
```

In addition to this state machine, the **Bad Frame Flag** from the previous frame is checked (**prevBFI**). The processing depends on the value of the **State**-variable. In states 0 and 5, the processing depends also on the two flags **BFI** and **prevBFI**.

The procedure can be described as follows:

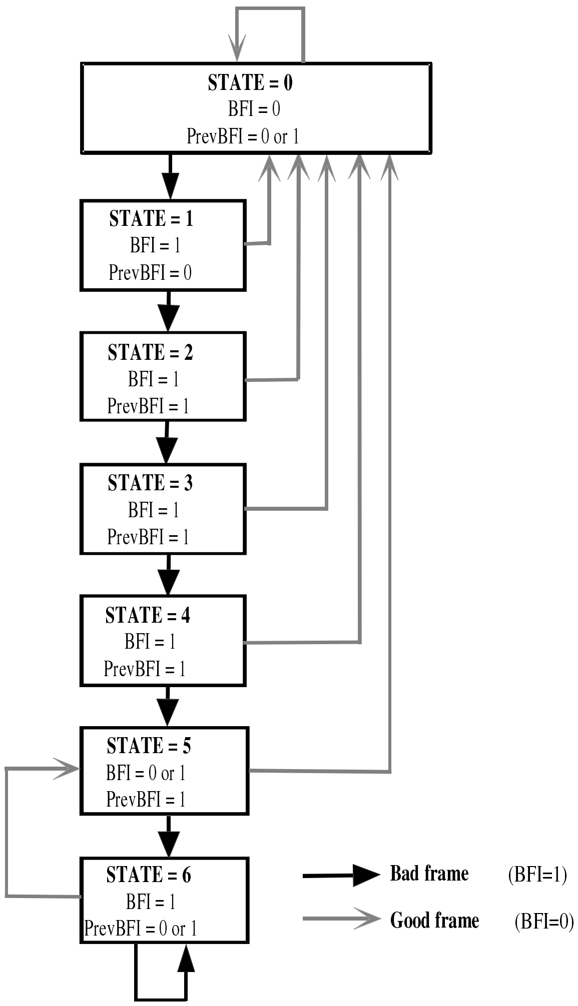

Figure 1: State machine for controlling the bad frame substitution

6.2 Assumed Active Speech Frame Error Concealment Unit Actions

6.2.1 BFI = 0, prevBFI = 0, State = 0

No error is detected in the received or in the previous received speech frame. The received speech parameters are used in the normal way in the speech synthesis. The current frame of speech parameters is saved.

6.2.2 BFI = 0, prevBFI = 1, State = 0 or 5

No error is detected in the received speech frame, but the previous received speech frame was bad. The LTP gain and fixed codebook gain are limited below the values used for the last received good subframe:

$$g^p = \begin{cases} g^p, & g^p \le g^p(-1) \\ g^p(-1), & g^p > g^p(-1) \end{cases} \tag{1}$$

where g^p = current decoded LTP gain, $g^p(-1)$ = LTP gain used for the last good subframe (BFI = 0), and

$$g^c = \begin{cases} g^c, & g^c \le g^c(-1) \\ g^c(-1), & g^c > g^c(-1) \end{cases} \tag{2}$$

where g^c = current decoded fixed codebook gain and $g^c(-1)$ = fixed codebook gain used for the last good subframe (BFI = 0).

The rest of the received speech parameters are used normally in the speech synthesis. The current frame of speech parameters is saved.

6.2.3 BFI = 1, prevBFI = 0 or 1, State = 1...6

An error is detected in the received speech frame and the substitution and muting procedure is started. The LTP gain and fixed codebook gain are replaced by attenuated values from the previous subframes:

$$g^p = \begin{cases} P(state)\ g^p(-1), & g^p(-1) \le median5(g^p(-1),\ldots,g^p(-5)) \\ P(state)\ median5(g^p(-1),\ldots,g^p(-5)), & g^p(-1) > median5(g^p(-1),\ldots,g^p(-5)) \end{cases} \tag{3}$$

where g^p = current decoded LTP gain, $g^p(-1),\ldots,g^p(-n)$ = LTP gains used for the last n subframes, *median5()* = 5-point median operation, *P(state)* = attenuation factor (*P(1)* = 0.98, *P(2)* = 0.98, *P(3)* = 0.8, *P(4)* = 0.3, *P(5)* = 0.2, *P(6)* = 0.2), *state* = state number, and

$$g^c = \begin{cases} C(state)\ g^c(-1), & g^c(-1) \le median5(g^c(-1),\ldots,g^c(-5)) \\ C(state)\ median5(g^c(-1),\ldots,g^c(-5)), & g^c(-1) > median5(g^c(-1),\ldots,g^c(-5)) \end{cases} \tag{4}$$

where g^c = current decoded fixed codebook gain, $g^c(-1),\ldots,g^c(-n)$ = fixed codebook gains used for the last n subframes, *median5()* = 5-point median operation, *C(state)* = attenuation factor (*C(1)* = 0.98, *C(2)* = 0.98, *C(3)* = 0.98, *C(4)* = 0.98, *C(5)* = 0.98, *C(6)* = 0.7), and *state* = state number.

The higher the state value is, the more the gains are attenuated. Also the memory of the predictive fixed codebook gain is updated by using the average value of the past four values in the memory:

$$ener(0) = \frac{1}{4}\sum_{i=1}^{4} ener(-i) \tag{5}$$

The past LSFs are shifted towards their mean:

$$lsf\_q1(i) = lsf\_q2(i) = \alpha\ past\_lsf\_q(i) + (1-\alpha)mean\_lsf(i), \quad i = 0\ldots9 \tag{6}$$

where α = 0.95, *lsf_q1* and *lsf_q2* are two sets of LSF-vectors for current frame, *past_lsf_q* is *lsf_q2* from the previous frame, and *mean_lsf* is the average LSF-vector. Note that two sets of LSFs are available only in the 12.2 mode.

6.2.3.1 LTP-lag update

The LTP-lag values are replaced by the past value from the 4th subframe of the previous frame (12.2 mode) or slightly modified values based on the last correctly received value (all other modes).

6.2.3.2 Innovation sequence

The received fixed codebook innovation pulses from the erroneous frame are always used in the state in which they were received when corrupted data are received . In the case when no data were received random fixed codebook indices should be employed.

6.3 Assumed Non-Active Speech Signal Error Concealment Unit Actions

6.3.1 General

The Non-Active Speech ECU is used to reduce the negative impact of amplitude variations and tonal artefacts when using the conventional Active Speech ECU in non-voiced signals such as background noise and unvoiced speech. The background ECU actions are only used for the lower rate Speech Coding modes of TCH-FS and TCH-HS.

The Non-Active Speech ECU actions are done as postprocessing actions of the Active Speech ECU, actions thus ensuring that the Active Speech ECU states are continuously updated. This will guarantee instant and seamless switching to the Active Speech ECU. The detectors and state updates have to be running continuously for all speech coding modes to avoid switching problems.

Only the differences to the Active Speech ECU are stated below.

6.3.2 Detectors

6.3.2.1 Background detector

An energy level and energy change detector is used to monitor the signal. If the signal is considered to contain background noise and only shows minor energy level changes, a flag is set. The resulting indicator is the **inBackgroundNoise** flag which indicates the signal state of the previous frame.

6.3.2.2 Voicing detector

The received LTP gain is monitored and used to prevent the use of the background ECU actions in possibly voiced segments. A median filtered LTP gain value with a varying filter memory length is thresholded to provide the correct voicing decision. Additionally, a counter **voicedHangover** is used to monitor the time since a frame was presumably voiced.

6.3.3 Background ECU Actions

The BFI, and DFI indications are used together with the flag **inBackgroundNoise** and the counter **voicedHangover** to adjust the LTP part and the innovation part of the excitation. The actions are only taken if the previous frame has been classified as background noise and sufficient time has passed since the last voiced frame was detected.

The background ECU actions are: energy control of the excitation signal, relaxed LTP lag control, stronger limitation of the LTP gain, adjusted adaptation of the Gain-Countour-Smoothing algorithm and modified adaptation of the Anti-Sparseness Procedure.

6.4 Substitution and muting of lost SID frames

In the speech decoder a single frame classified as SID_BAD shall be substituted by the last valid SID frame information and the procedure for valid SID frames be applied. If the time between SID information updates (updates are specified by SID_UPDATE arrivals and occasionally by SID_FIRST arrivals see 06.92) is greater than one second this shall lead to attenuation.

7 Example ECU/BFH Solution 2

This is an alternative example solution which is a simplified version of Example ECU/BFH Solution 1.

7.1 State Machine

This example solution for substitution and muting is based on a state machine with seven states (Figure 1, same state machine as in Example 1).

The system starts in state 0. Each time a bad frame is detected, the state counter is incremented by one and is saturated when it reaches 6. Each time a good speech frame is detected, the state counter is reset to zero, except when we are in state 6, where we set the state counter to 5. The state indicates the quality of the channel: the larger the state counter, the worse the channel quality is. The control flow of the state machine can be described by the following C code (**BFI** = bad frame indicator, **State** = state variable):

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else
    State = 0;
if(State > 6 )
    State = 6;
```

In addition to this state machine, the **Bad Frame Flag** from the previous frame is checked (**prevBFI**). The processing depends on the value of the **State**-variable. In states 0 and 5, the processing depends also on the two flags **BFI** and **prevBFI**.

7.2 Substitution and muting of lost speech frames

7.2.1 BFI = 0, prevBFI = 0, State = 0

No error is detected in the received or in the previous received speech frame. The received speech parameters are used normally in the speech synthesis. The current frame of speech parameters is saved.

7.2.2 BFI = 0, prevBFI = 1, State = 0 or 5

No error is detected in the received speech frame but the previous received speech frame was bad. The LTP gain and fixed codebook gain are limited below the values used for the last received good subframe:

$$g^p = \begin{cases} g^p, & g^p \le g^p(-1) \\ g^p(-1), & g^p > g^p(-1) \end{cases} \tag{7}$$

where g^p = current decoded LTP gain, $g^p(-1)$ = LTP gain used for the last good subframe (BFI = 0), and

$$g^c = \begin{cases} g^c, & g^c \le g^c(-1) \\ g^c(-1), & g^c > g^c(-1) \end{cases} \tag{8}$$

where g^c = current decoded fixed codebook-gain and $g^c(-1)$ = fixed codebook gain used for the last good subframe (BFI = 0).

The rest of the received speech parameters are used normally in the speech synthesis. The current frame of speech parameters is saved.

7.2.3 BFI = 1, prevBFI = 0 or 1, State = 1...6

An error is detected in the received speech frame and the substitution and muting procedure is started. The LTP gain and fixed codebook gain are replaced by attenuated values from the previous subframes:

$$g^p = \begin{cases} P(state)\ g^p(-1), & g^p(-1) \le median5(g^p(-1),\ldots,g^p(-5)) \\ P(state)\ median5(g^p(-1),\ldots,g^p(-5)), & g^p(-1) > median5(g^p(-1),\ldots,g^p(-5)) \end{cases} \quad (9)$$

where g^p = current decoded LTP gain, $g^p(-1),\ldots,g^p(-n)$ = LTP gains used for the last n subframes, *median5()* = 5-point median operation, *P(state)* = attenuation factor (*P(1)* = 0.98, *P(2)* = 0.98, *P(3)* = 0.8, *P(4)* = 0.3, *P(5)* = 0.2, *P(6)* = 0.2), *state* = state number, and

$$g^c = \begin{cases} C(state)\ g^c(-1), & g^c(-1) \le median5(g^c(-1),\ldots,g^c(-5)) \\ C(state)\ median5(g^c(-1),\ldots,g^c(-5)), & g^c(-1) > median5(g^c(-1),\ldots,g^c(-5)) \end{cases} \quad (10)$$

where g^c = current decoded fixed codebook gain, $g^c(-1),\ldots,g^c(-n)$ = fixed codebook gains used for the last n subframes, *median5()* = 5-point median operation, *C(state)* = attenuation factor (*C(1)* = 0.98, *C(2)* = 0.98, *C(3)* = 0.98, *C(4)* = 0.98, *C(5)* = 0.98, *C(6)* = 0.7), and *state* = state number.

The higher the state value is, the more the gains are attenuated. Also the memory of the predictive fixed codebook gain is updated by using the average value of the past four values in the memory:

$$ener(0) = \frac{1}{4}\sum_{i=1}^{4} ener(-i) \quad (11)$$

The past LSFs are used by shifting their values towards their mean:

$$lsf\_q1(i) = lsf\_q2(i) = \alpha\ past\_lsf\_q(i) + (1-\alpha)mean\_lsf(i), \quad i = 0\ldots9 \quad (12)$$

where α = 0.95, *lsf_q1* and *lsf_q2* are two sets of LSF-vectors for current frame, *past_lsf_q* is *lsf_q2* from the previous frame, and *mean_lsf* is the average LSF-vector. Note that two sets of LSFs are available only in the 12.2 mode.

7.2.3.1 LTP-lag update

The LTP-lag values are replaced by the past value from the 4th subframe of the previous frame (12.2 mode) or slightly modified values based on the last correctly received value (all other modes).

7.2.4 Innovation sequence

The received fixed codebook innovation pulses from the erroneous frame are always used in the state in which they were received when corrupted data are received. In the case when no data were received random fixed codebook indices should be employed.

7.3 Substitution and muting of lost SID frames

In the speech decoder a single frame classified as SID_BAD shall be substituted by the last valid SID frame information and the procedure for valid SID frames be applied. If the time between SID information updates (updates are specified by SID_UPDATE arrivals and occasionally by SID_FIRST arrivals see 06.92) is greater than one second this shall lead to attenuation.

Annex A (informative):
Document change history

SMG	SPEC	CR	PH	VERS	NEW_VE	SUBJECT
30	06.91	A001	R98	7.0.1	7.1.0	Use of random excitation when RX_NODATA and not in DTX

History

Document history		
V7.0.1	July 1999	One-step Approval Procedure OAP 9952: 1999-07-28 to 1999-11-26
V7.1.0	December 1999	One-step Approval Procedure OAP 200013: 1999-12-01 to 2000-03-31