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Digital cellular telecommunications system (Phase 2+) (GSM); Full rate speech; Transcoding (GSM 06.10 version 6.1.1 Release 1997)

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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).

The present document specifies the full rate speech transcoding within the digital cellular telecommunications system.

NOTE: The present document is a reproduction of recommendation T/L/03/11 "13 kbit/s Regular Pulse Excitation - Long Term Prediction - Linear Predictive Coder for use in the digital cellular telecommunications system".

Archive en_300961v060101p0.ZIP which accompanies the present document, contains test sequences, as described in clause 6 and annex A.3.

The archive contains the following:

Disk1.zip	Annex B: Test sequences for the GSM Full Rate speech codec; Test sequences SEQ01.xxx to SEQ05.xxx. (Disk1.zip contains LHA compressed files.)
Disk2.zip	Annex B: Test sequences for the GSM Full Rate speech codec with homing frames; Test sequences SEQ01H.* to SEQ02H.*.
Disk3.zip	Annex B: Test sequences for the GSM Full Rate speech codec with homing frames; Test sequences SEQ03H.* to SYNC159.COD.
Disk4.zip	Annex B: 8 bit A-law test sequences for the GSM Full Rate speech codec with and without homing frames (Disk4.zip contains self-extracting files).
Disk5.zip	Annex B: 8 bit μ-law test sequences for the GSM Full Rate speech codec with and without homing frames (Disk5.zip contains self-extracting files).

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Version 6.x.y

where:

6 indicates Release 1997 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:	24 November 2000
Date of latest announcement of this EN (doa):	28 February 2001
Date of latest publication of new National Standard or endorsement of this EN (dop/e):	31 August 2001
Date of withdrawal of any conflicting National Standard (dow):	31 August 2001

1 Scope

The transcoding procedure specified in the present document is applicable for the full-rate Traffic Channel (TCH) in the digital cellular telecommunications system. The use of this transcoding scheme for other applications has not been considered.

In GSM 06.01, a reference configuration for the speech transmission chain of the digital cellular telecommunications system is shown. According to this reference configuration, the speech encoder takes its input as a 13 bit uniform PCM signal either from the audio part of the mobile station or on the network side, from the PSTN via an 8 bit/A-law to 13 bit uniform PCM conversion. The encoded speech at the output of the speech encoder is delivered to a channel encoder unit which is specified in GSM 05.03. In the receive direction, the inverse operations take place.

The present document describes the detailed mapping between input blocks of 160 speech samples in 13 bit uniform PCM format to encoded blocks of 260 bits and from encoded blocks of 260 bits to output blocks of 160 reconstructed speech samples. The sampling rate is 8000 sample/s leading to an average bit rate for the encoded bit stream of 13 kbit/s. The coding scheme is the so-called Regular Pulse Excitation - Long Term prediction - Linear Predictive Coder, here-after referred to as RPE-LTP.

The present document also specifies the conversion between A-law PCM and 13 bit uniform PCM. Performance requirements for the audio input and output parts are included only to the extent that they affect the transcoder performance. The present document also describes the codec down to the bit level, thus enabling the verification of compliance to the present document to a high degree of confidence by use of a set of digital test sequences. These test sequences are described and are contained in archive en_300961v060101p0.ZIP which accompanies the present document.

1.1 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 1997 document, references to GSM documents are for Release 1997 versions (version 6.x.y).

[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 06.01: "Digital cellular telecommunications system (Phase 2+); Full rate speech; Processing functions".

[4] GSM 11.10: "Digital cellular telecommunications system (Phase 2+); Mobile Station (MS) conformity specification".

[5] ETS 300 085: "Integrated Services Digital Network (ISDN); 3,1kHz telephony teleservice; Attachment requirements for handset terminals (Candidate NET 33)".

[6] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".

[7] ITU-T Recommendation G.712: "Transmission performance characteristics of pulse code modulation".

[8] ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)".

[9] ITU-T Recommendation Q.35: "Technical characteristics of tones for the telephone service".

[10] ITU-T Recommendation V.21: "300 bits per second duplex modem standardized for use in the general switched telephone network".

[11] ITU-T Recommendation V.23: "600/1 200-band modem standardized for use in the general switched telephone network".

[12] GSM 06.32: "Digital cellular telecommunications system (Phase 2+); Voice Activity Detector (VAD)".

1.1.1 Abbreviations

Abbreviations used in the present document are listed in GSM 01.04.

1.2 Outline description

The present document is structured as follows:

Subclause 1.3 contains a functional description of the audio parts including the A/D and D/A functions. Subclause 1.4 describes the conversion between 13 bit uniform and 8 bit A-law samples. Subclauses 1.5 and 1.6 present a simplified description of the principles of the RPE-LTP encoding and decoding process respectively. In clause 1.7, the sequence and subjective importance of encoded parameters are given.

Clause 2 deals with the transmission characteristics of the audio parts that are relevant for the performance of the RPE-LTP codec.

Some transmission characteristics of the RPE-LTP codec are also specified in clause 2. Clause 3 presents the functional description of the RPE-LTP coding and decoding procedures, whereas clause 4 describes the computational details of the algorithm. Procedures for the verification of the correct functioning of the RPE-LTP are described in clause 5.

Performance and network aspects of the RPE-LTP codec are contained in annex A.

1.3 Functional description of audio parts

The analogue-to-digital and digital-to-analogue conversion will in principle comprise the following elements:

1) Analogue to uniform digital:

- microphone;
- input level adjustment device;
- input anti-aliasing filter;
- sample-hold device sampling at 8 kHz;
- analogue-to-uniform digital conversion to 13 bits representation.

The uniform format shall be represented in two's complement.

2) Uniform digital to analogue:

- conversion from 13 bit /8 kHz uniform PCM to analogue;
- a hold device;
- reconstruction filter including x/sin x correction;
- output level adjustment device;

- earphone or loudspeaker.

In the terminal equipment, the A/D function may be achieved either:

- by direct conversion to 13 bit uniform PCM format;
- or by conversion to 8 bit/A-law companded format, based on a standard A-law codec/filter according to ITU-T Recommendation G.711/714, followed by the 8-bit to 13-bit conversion according to the procedure specified in clause 1.4.

For the D/A operation, the inverse operations take place.

In the latter case it should be noted that the specifications in ITU-T recommendation G.714 (superseded by G.712) are concerned with PCM equipment located in the central parts of the network. When used in the terminal equipment, the present document does not on its own ensure sufficient out-of-band attenuation.

The specification of out-of-band signals is defined in clause 2 between the acoustic signal and the digital interface to take into account that the filtering in the terminal can be achieved both by electronic and acoustical design.

1.4 PCM Format conversion

The conversion between 8 bit A-law companded format and the 13-bit uniform format shall be as defined in ITU-T Recommendation G.721 (superseded by G.726), clause 4.2.1, sub-block EXPAND and clause 4.2.7, sub-block COMPRESS. The parameter LAW = 1 should be used.

1.5 Principles of the RPE-LTP encoder

A simplified block diagram of the RPE-LTP encoder is shown in figure 1.1. In this diagram the coding and quantization functions are not shown explicitly.

The input speech frame, consisting of 160 signal samples (uniform 13 bit PCM samples), is first pre-processed to produce an offset-free signal, which is then subjected to a first order pre-emphasis filter. The 160 samples obtained are then analysed to determine the coefficients for the short term analysis filter (LPC analysis). These parameters are then used for the filtering of the same 160 samples. The result is 160 samples of the short term residual signal. The filter parameters, termed reflection coefficients, are transformed to log.area ratios, LARs, before transmission.

For the following operations, the speech frame is divided into 4 sub-frames with 40 samples of the short term residual signal in each. Each sub-frame is processed blockwise by the subsequent functional elements.

Before the processing of each sub-block of 40 short term residual samples, the parameters of the long term analysis filter, the LTP lag and the LTP gain, are estimated and updated in the LTP analysis block, on the basis of the current sub-block of the present and a stored sequence of the 120 previous reconstructed short term residual samples.

A block of 40 long term residual signal samples is obtained by subtracting 40 estimates of the short term residual signal from the short term residual signal itself. The resulting block of 40 long term residual samples is fed to the Regular Pulse Excitation analysis which performs the basic compression function of the algorithm.

As a result of the RPE-analysis, the block of 40 input long term residual samples are represented by one of 4 candidate sub-sequences of 13 pulses each. The subsequence selected is identified by the RPE grid position (M). The 13 RPE pulses are encoded using Adaptive Pulse Code Modulation (APCM) with estimation of the sub-block amplitude which is transmitted to the decoder as side information.

The RPE parameters are also fed to a local RPE decoding and reconstruction module which produces a block of 40 samples of the quantized version of the long term residual signal.

By adding these 40 quantized samples of the long term residual to the previous block of short term residual signal estimates, a reconstructed version of the current short term residual signal is obtained.

The block of reconstructed short term residual signal samples is then fed to the long term analysis filter which produces the new block of 40 short term residual signal estimates to be used for the next sub-block thereby completing the feedback loop.

1.6 Principles of the RPE-LTP decoder

The simplified block diagram of the RPE-LTP decoder is shown in fig 1.2. The decoder includes the same structure as the feed-back loop of the encoder. In error-free transmission, the output of this stage will be the reconstructed short term residual samples. These samples are then applied to the short term synthesis filter followed by the de-emphasis filter resulting in the reconstructed speech signal samples.

1.7 Sequence and subjective importance of encoded parameters

As indicated in fig 1.1 the three different groups of data are produced by the encoder are:

- the short term filter parameters;
- the Long Term Prediction (LTP) parameters;
- the RPE parameters.

The encoder will produce this information in a unique sequence and format, and the decoder shall receive the same information in the same way. In table 1.1, the sequence of output bits b1 to b260 and the bit allocation for each parameter is shown.

The different parameters of the encoded speech and their individual bits have unequal importance with respect to subjective quality. Before being submitted to the channel encoding function the bits have to be rearranged in the sequence of importance as given in GSM 05.03. The ranking has been determined by subjective testing and the procedure used is described in annex A, clause A.2.

Table 1.1: Encoder output parameters in order of occurrence and bit allocation within the speech frame of 260 bits/20 ms

Parameter	Parameter number	Parameter name	Var. name	Number of bits	Bit no. (LSB-MSB)
	1		LAR 1	6	b1 - b6
	2		LAR 2	6	b7 - b12
FILTER	3	Log. Area	LAR 3	5	b13 - b17
	4	ratios	LAR 4	5	b18 - b22
PARAMETERS	5	1 - 8	LAR 5	4	b23 - b26
	6		LAR 6	4	b27 - b30
	7		LAR 7	3	b31 - b33
	8		LAR 8	3	b34 - b36

Sub-frame no.1

LTP	9	LTP lag	N1	7	b37 - b43
PARAMETERS	10	LTP gain	b1	2	b44 - b45
	11	RPE grid position	M1	2	b46 - b47
RPE	12	Block amplitude	Xmax1	6	b48 - b53
PARAMETERS	13	RPE-pulse no.1	x1(0)	3	b54 - b56
	14	RPE-pulse no.2	x1(1)	3	b57 - b59
	..	...			...
	25	RPE-pulse no.13	x1(12)	3	b90 - b92

Sub-frame no.2

LTP PARAMETERS	26	LTP lag	N2	7	b93 - b99
	27	LTP gain	b2	2	b100- b101
RPE PARAMETERS	28	RPE grid position	M2	2	b102- b103
	29	Block amplitude	Xmax2	6	b104- b109
	30	RPE-pulse no.1	x2(0)	3	b110- b112
	31	RPE-pulse no.2	x2(1)	3	b113- b115
	..	...			...
	42	RPE-pulse no.13	x2(12)	3	b146- b148

Table 1.1: Encoder output parameters in order of occurrence and bit allocation within the speech frame of 260 bits/20 ms

Sub-frame no.3

LTP PARAMETERS	43	LTP lag	N3	7	b149- b155
	44	LTP gain	b3	2	b156- b157
RPE PARAMETERS	45	RPE grid position	M3	2	b158- b159
	46	Block amplitude	Xmax3	6	b160- b165
	47	RPE-pulse no.1	x3(0)	3	b166- b168
	48	RPE-pulse no.2	x3(1)	3	b169- b171
	..	...			...
	59	RPE-pulse no.13	x3(12)	3	b202- b204

Sub-frame no.4

LTP PARAMETERS	60	LTP lag	N4	7	b205- b211
	61	LTP gain	b4	2	b212- b213
RPE PARAMETERS	62	RPE grid position	M4	2	b214- b215
	63	Block amplitude	Xmax4	6	b216- b221
	64	RPE-pulse no.1	x4(0)	3	b222- b224
	65	RPE-pulse no.2	x4(1)	3	b225- b227
	..	...			...
	76	RPE-pulse no.13	x4(12)	3	b258- b260

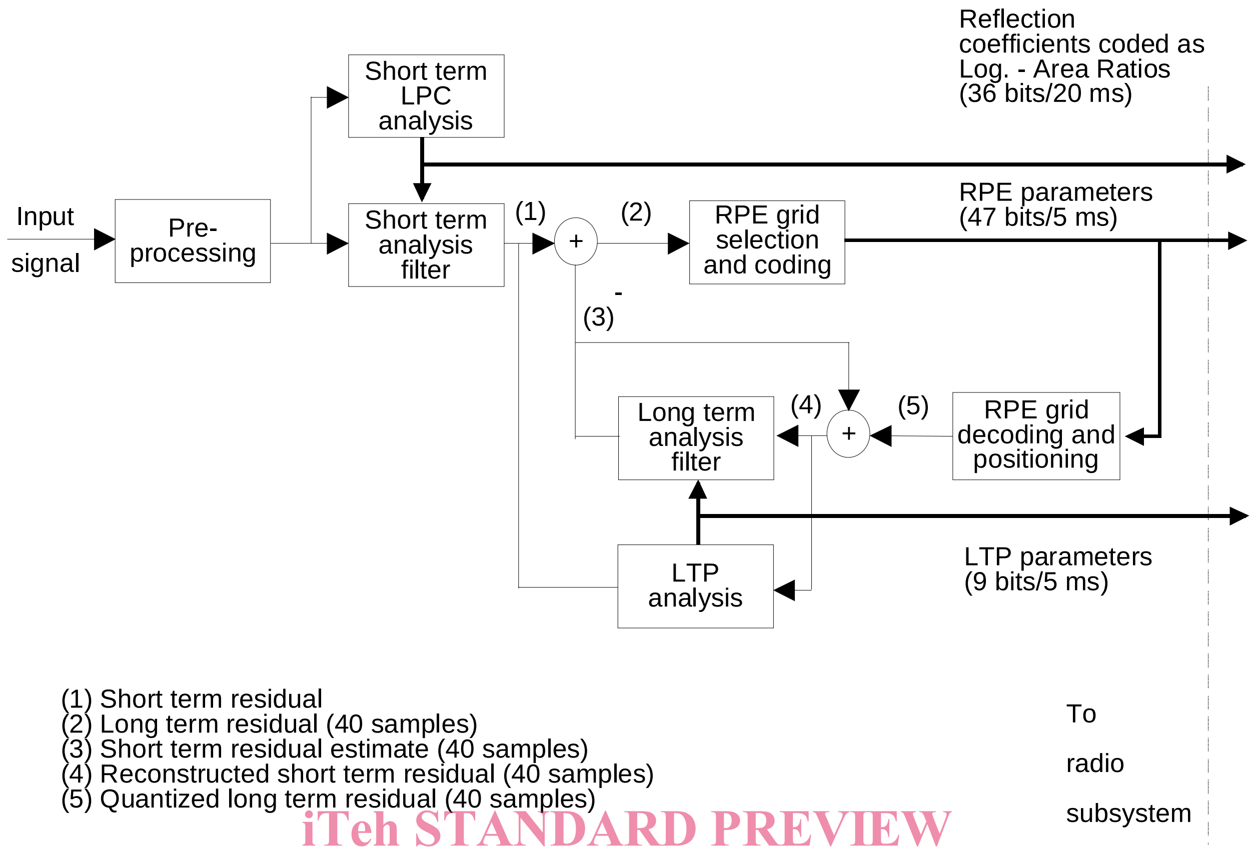

Figure 1.1: Simplified block diagram of the RPE - LTP encoder

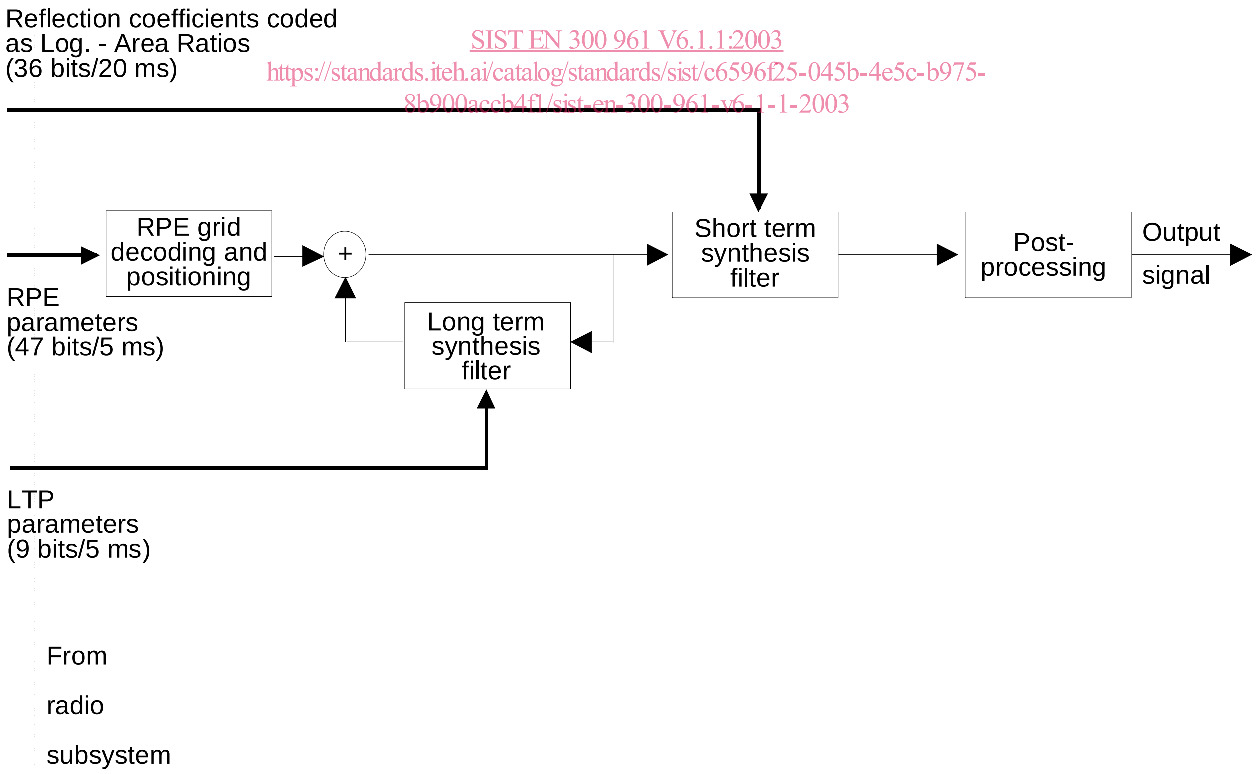

Figure 1.2: Simplified block diagram of the RPE - LTP decoder