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Contents

Intellectual Property Rights	2
Foreword.....	2
Modal verbs terminology	2
Foreword.....	6
1 Scope.....	7
1.1 References.....	7
1.1.1 Abbreviations	8
1.2 Outline description.....	8
1.3 Functional description of audio parts.....	8
1.4 PCM Format conversion.....	9
1.5 Principles of the RPE-LTP encoder.....	9
1.6 Principles of the RPE-LTP decoder.....	10
1.7 Sequence and subjective importance of encoded parameters	10
2 Transmission characteristics	13
2.1 Performance characteristics of the analogue/digital interfaces	13
2.2 Transcoder delay.....	13
3 Functional description of the RPE-LTP codec.....	13
3.1 Functional description of the RPE-LTP encoder	13
3.1.1 Offset compensation.....	14
3.1.2 Pre-emphasis	14
3.1.3 Segmentation	14
3.1.4 Autocorrelation	15
3.1.5 Schur Recursion	15
3.1.6 Transformation of reflection coefficients to Log.-Area Ratios	15
3.1.7 Quantization and coding of Log.-Area Ratios.....	15
3.1.8 Decoding of the quantized Log.-Area Ratios	16
3.1.9 Interpolation of Log.-Area Ratios	16
3.1.10 Transformation of Log.-Area Ratios into reflection coefficients	16
3.1.11 Short term analysis filtering	16
3.1.12 Sub-segmentation	17
3.1.13 Calculation of the LTP parameters.....	17
3.1.14 Coding/Decoding of the LTP lags.....	17
3.1.15 Coding/Decoding of the LTP gains.....	18
3.1.16 Long term analysis filtering	18
3.1.17 Long term synthesis filtering.....	18
3.1.18 Weighting Filter	19
3.1.19 Adaptive sample rate decimation by RPE grid selection.....	19
3.1.20 APCM quantization of the selected RPE sequence	19
3.1.21 APCM inverse quantization	20
3.1.22 RPE grid positioning	21
3.2 Decoder.....	21
3.2.1 RPE decoding clause.....	21
3.2.2 Long Term Prediction clause	21
3.2.3 Short term synthesis filtering clause	21
3.2.4 Post-processing	21
4 Codec homing	25
4.1 Functional description.....	25
4.2 Definitions	25
4.3 Encoder homing.....	26
4.4 Decoder homing.....	26
4.5 Encoder home state.....	27
4.6 Decoder home state.....	27
5 Computational details of the RPE-LTP codec	27

5.1	Data representation and arithmetic operations	27
5.2	Fixed point implementation of the RPE-LTP coder	29
5.2.0	Scaling of the input variable.....	30
5.2.1	Downscaling of the input signal.....	30
5.2.2	Offset compensation.....	30
5.2.3	Pre-emphasis	30
5.2.4	Autocorrelation	31
5.2.5	Computation of the reflection coefficients	31
5.2.6	Transformation of reflection coefficients to Log.-Area Ratios	32
5.2.7	Quantization and coding of the Log.-Area Ratios	33
5.2.8	Decoding of the coded Log.-Area Ratios	33
5.2.9	Computation of the quantized reflection coefficients.....	34
5.2.9.1	Interpolation of the LARpp[1..8] to get the LARp[1..8].....	34
5.2.9.2	Computation of the rp[1..8] from the interpolated LARp[1..8]	34
5.2.10	Short term analysis filtering	34
5.2.11	Calculation of the LTP parameters.....	35
5.2.12	Long term analysis filtering	36
5.2.13	Weighting filter	36
5.2.14	RPE grid selection.....	37
5.2.15	APCM quantization of the selected RPE sequence	37
5.2.16	APCM inverse quantization	38
5.2.17	RPE grid positioning	39
5.2.18	Update of the reconstructed short term residual signal dp[-120..-1]	39
5.3	Fixed point implementation of the RPE-LTP decoder.....	39
5.3.1	RPE decoding clause.....	39
5.3.2	Long term synthesis filtering.....	40
5.3.3	Computation of the decoded reflection coefficients	40
5.3.4	Short term synthesis filtering clause	40
5.3.5	De-emphasis filtering	41
5.3.6	Upscaling of the output signal.....	41
5.3.7	Truncation of the output variable	41
5.4	Tables used in the fixed point implementation of the RPE-LTP coder and decoder	42
6	Digital test sequences	43
6.1	Input and output signals.....	44
6.2	Configuration for the application of the test sequences	44
6.2.1	Configuration 1 (encoder only)	44
6.2.2	Configuration 2 (decoder only)	44
6.3	Test sequences	45
6.3.1	Test sequences for configuration 1	45
6.3.2	Test sequences for configuration 2.....	46
6.3.3	Additional Test sequences for Codec Homing	50
6.3.3.1	Codec homing frames.....	50
6.3.3.2	Sequence for an extensive test of the decoder homing	50
6.3.3.3	Sequences for finding the 20 ms framing of the GSM full rate speech encoder.....	50
6.3.3.4	Formats and sizes of the synchronization sequences	51
Annex A (informative): Codec performance.....		53
A.1	Performance of the RPE-LTP	53
A.1.1	Introduction.....	53
A.1.2	Speech performance.....	53
A.1.2.1	Single encoding.....	53
A.1.2.2	Speech performance when interconnected with coding systems on an analogue basis	54
A.1.2.2.1	Performance with 32 kbit/s ADPCM (G.721, superseded by G.726).....	54
A.1.2.2.2	Performance with another RPE-LTP codec.....	54
A.1.2.2.3	Performance with encoding other than RPE-LTP and 32 kbit/s ADPCM (G.721, superseded by G.726).....	54
A.1.3	Non-speech performance	55
A.1.3.1	Performance with single sine waves	55
A.1.3.2	Performance with DTMF tones	55
A.1.3.3	Performance with information tones	55
A.1.3.4	Performance with voice-band data	55

A.1.4	Delay.....	55
A.1.5	Bibliography	57
A.2	Subjective relevance of the speech coder output bits.....	57
A.3	Format for test sequence distribution	59
A.3.1	Type of files provided.....	59
A.3.2	File format description.....	60
Annex B (informative):	Test sequence disks	62
Annex C (informative):	Change history	63
History		64

Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (3GPP).

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_300961v080101p0.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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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- or μ -law (PCS 1900) 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- and μ -law (PCS 1900) 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_300961v080101p0.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.
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- [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] Void.
- [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".