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Digital cellular telecommunications system (Phase 2+) (GSM); Adaptive Multi-Rate (AMR) speech transcoding (GSM 06.90 version 7.2.1 Release 1998)

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Contents

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Foreword

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

The present document describes the detailed mapping from input blocks of 160 speech samples in 13-bit uniform PCM format to encoded blocks of 95, 103, 118, 134, 148, 159, 204, and 244 bits and from encoded blocks of 95, 103, 118, 134, 148, 159, 204, and 244 bits to output blocks of 160 reconstructed speech samples within the digital cellular telecommunications system.

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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:	31 March 2000
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1 Scope

The present document describes the detailed mapping from input blocks of 160 speech samples in 13-bit uniform PCM format to encoded blocks of 95, 103, 118, 134, 148, 159, 204, and 244 bits and from encoded blocks of 95, 103, 118, 134, 148, 159, 204, and 244 bits to output blocks of 160 reconstructed speech samples. The sampling rate is 8 000 samples/s leading to a bit rate for the encoded bit stream of 4.75, 5.15, 5.90, 6.70, 7.40, 7.95, 10.2 or 12.2 kbit/s. The coding scheme for the multi-rate coding modes is the so-called Algebraic Code Excited Linear Prediction Coder, hereafter referred to as ACELP. The multi-rate ACELP coder is referred to as MR-ACELP.

In the case of discrepancy between the requirements described in the present document and the fixed point computational description (ANSI-C code) of these requirements contained in GSM 06.73 [6], the description in GSM 06.73 [6] will prevail. The ANSI-C code is not described in the present document, see GSM 06.73 [6] for a description of the ANSI-C code.

The transcoding procedure specified in the present document is applicable for the adaptive multi-rate full rate and half rate speech traffic channels (TCH) in the GSM system.

In GSM 06.71 [5], a reference configuration for the speech transmission chain of the GSM adaptive multi-rate (AMR) 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 or μ-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 [3]. In the receive direction, the inverse operations take place.

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 telecommunications system (Phase 2+); Abbreviations and acronyms".

[2] GSM 03.50: "Digital cellular telecommunications system (Phase 2+); Transmission planning aspects of the speech service in the GSM Public Land Mobile Network (PLMN) system".

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

[4] GSM 06.94: "Digital cellular telecommunications system (Phase 2+); Voice Activity Detection (VAD) for Adaptive Multi-Rate speech traffic channels".

[5] GSM 06.71: "Digital cellular telecommunications system (Phase 2+); Adaptive Multi-Rate speech processing functions; General description".

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

[7] GSM 06.74: "Digital cellular telecommunications system (Phase 2+); Test sequences for the GSM Adaptive Multi-Rate speech codec".

[8] ITU-T Recommendation G.711 (1988): "Coding of analogue signals by pulse code modulation Pulse code modulation (PCM) of voice frequencies".

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

3 Definitions, symbols and abbreviations

3.1 Definitions

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

adaptive codebook: The adaptive codebook contains excitation vectors that are adapted for every subframe. The adaptive codebook is derived from the long-term filter state. The lag value can be viewed as an index into the adaptive codebook.

adaptive postfilter: This filter is applied to the output of the short-term synthesis filter to enhance the perceptual quality of the reconstructed speech. In the adaptive multi-rate codec, the adaptive postfilter is a cascade of two filters: a formant postfilter and a tilt compensation filter.

Adaptive Multi-Rate (AMR) codec: Speech and channel codec capable of operating at gross bit-rates of 11.4 kbit/s ("half-rate") and 22.8 kbit/s ("full-rate"). In addition, the codec may operate at various combinations of speech and channel coding (codec mode) bit-rates for each channel mode.

algebraic codebook: A fixed codebook where algebraic code is used to populate the excitation vectors (innovation vectors). The excitation contains a small number of nonzero pulses with predefined interlaced sets of positions.

AMR handover: Handover between the FR and HR channel modes to optimise AMR operation.

anti-sparseness processing: An adaptive post-processing procedure applied to the fixed codebook vector in order to reduce perceptual artifacts from a sparse fixed codebook vector.

channel mode: Half-rate or full-rate operation.

channel mode adaptation: The control and selection of the (FR or HR) channel mode.

channel repacking: Repacking of HR (and FR) radio channels of a given radio cell to achieve higher capacity within the cell.

closed-loop pitch analysis: This is the adaptive codebook search, i.e., a process of estimating the pitch (lag) value from the weighted input speech and the long term filter state. In the closed-loop search, the lag is searched using error minimization loop (analysis-by-synthesis). In the adaptive multi-rate codec, closed-loop pitch search is performed for every subframe.

codec mode: For a given channel mode, the bit partitioning between the speech and channel codecs.

codec mode adaptation: The control and selection of the codec mode bit-rates. Normally, implies no change to the channel mode.

direct form coefficients: One of the formats for storing the short term filter parameters. In the adaptive multi-rate codec, all filters which are used to modify speech samples use direct form coefficients.

fixed codebook: The fixed codebook contains excitation vectors for speech synthesis filters. The contents of the codebook are non-adaptive (i.e., fixed). In the adaptive multi-rate codec, the fixed codebook is implemented using an algebraic codebook.

fractional lags: A set of lag values having sub-sample resolution. In the adaptive multi-rate codec a sub-sample resolution of 1/6th or 1/3rd of a sample is used.

full-rate (FR): Full-rate channel or channel mode.

frame: A time interval equal to 20 ms (160 samples at an 8 kHz sampling rate).

gross bit-rate: The bit-rate of the channel mode selected (22.8 kbs or 11.4 kbs).

half-rate (HR): Half-rate channel or channel mode.

in-band signalling: Signalling for DTX, Link Control, Channel and codec mode modification, etc. carried within the traffic channel.

integer lags: A set of lag values having whole sample resolution.

interpolating filter: An FIR filter used to produce an estimate of subsample resolution samples, given an input sampled with integer sample resolution.

inverse filter: This filter removes the short term correlation from the speech signal. The filter models an inverse frequency response of the vocal tract.

lag: The long term filter delay. This is typically the true pitch period, or its multiple or sub-multiple.

Line Spectral Frequencies: (see Line Spectral Pair).

Line Spectral Pair: Transformation of LPC parameters. Line Spectral Pairs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The Line Spectral Pairs (also called as Line Spectral Frequencies) are the roots of these polynomials on the z-unit circle.

LP analysis window: For each frame, the short term filter coefficients are computed using the high pass filtered speech samples within the analysis window. In the adaptive multi-rate codec, the length of the analysis window is always 240 samples. For each frame, two asymmetric windows are used to generate two sets of LP coefficient in the 12,2 kbit/s mode. For the other modes, only a single asymmetric window is used to generate a single set of LP coefficients. In the 12,2 kbit/s mode, no samples of the future frames are used (no lookahead). The other modes use a 5 ms lookahead.

LP coefficients: Linear Prediction (LP) coefficients (also referred as Linear Predictive Coding (LPC) coefficients) is a generic descriptive term for the short term filter coefficients.

mode: When used alone, refers to the source codec mode, i.e., to one of the source codecs employed in the AMR codec. (See also codec mode and channel mode.)

open-loop pitch search: A process of estimating the near optimal lag directly from the weighted speech input. This is done to simplify the pitch analysis and confine the closed-loop pitch search to a small number of lags around the open-loop estimated lags. In the adaptive multi-rate codec, an open-loop pitch search is performed in every other subframe.

out-of-band signalling: Signalling on the GSM control channels to support link control.

residual: The output signal resulting from an inverse filtering operation.

short term synthesis filter: This filter introduces, into the excitation signal, short term correlation which models the impulse response of the vocal tract.

perceptual weighting filter: This filter is employed in the analysis-by-synthesis search of the codebooks. The filter exploits the noise masking properties of the formants (vocal tract resonances) by weighting the error less in regions near the formant frequencies and more in regions away from them.

subframe: A time interval equal to 5 ms (40 samples at 8 kHz sampling rate).

vector quantization: A method of grouping several parameters into a vector and quantizing them simultaneously.

zero input response: The output of a filter due to past inputs, i.e. due to the present state of the filter, given that an input of zeros is applied.

zero state response: The output of a filter due to the present input, given that no past inputs have been applied, i.e., given that the state information in the filter is all zeroes.

3.2 Symbols

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

$A(z)$	The inverse filter with unquantized coefficients
$\hat{A}(z)$	The inverse filter with quantized coefficients
$H(z) = \frac{1}{\hat{A}(z)}$	The speech synthesis filter with quantized coefficients
a_i	The unquantized linear prediction parameters (direct form coefficients)
$\hat{a}_i$	The quantified linear prediction parameters
m	The order of the LP model
$\frac{1}{B(z)}$	The long-term synthesis filter
$W(z)$	The perceptual weighting filter (unquantized coefficients)
γ_1, γ_2	The perceptual weighting factors
$F_E(z)$	Adaptive pre-filter
T	The integer pitch lag nearest to the closed-loop fractional pitch lag of the subframe
β	The adaptive pre-filter coefficient (the quantified pitch gain)
$H_f(z) = \frac{\hat{A}(z/\gamma_n)}{\hat{A}(z/\gamma_d)}$	The formant postfilter
γ_n	Control coefficient for the amount of the formant post-filtering
γ_d	Control coefficient for the amount of the formant post-filtering
$H_t(z)$	Tilt compensation filter
γ_t	Control coefficient for the amount of the tilt compensation filtering
$\mu = \gamma_t k_1'$	A tilt factor, with k_1' being the first reflection coefficient
$h_f(n)$	The truncated impulse response of the formant postfilter
L_h	The length of $h_f(n)$
$r_h(i)$	The auto-correlations of $h_f(n)$
$\hat{A}(z/\gamma_n)$	The inverse filter (numerator) part of the formant postfilter
$1/\hat{A}(z/\gamma_d)$	The synthesis filter (denominator) part of the formant postfilter
$\hat{r}(n)$	The residual signal of the inverse filter $\hat{A}(z/\gamma_n)$
$h_t(n)$	Impulse response of the tilt compensation filter
$\beta_{sc}(n)$	The AGC-controlled gain scaling factor of the adaptive postfilter
α	The AGC factor of the adaptive postfilter
$H_{h1}(z)$	Pre-processing high-pass filter
$w_I(n)$, $w_{II}(n)$	LP analysis windows
$L_1^{(I)}$	Length of the first part of the LP analysis window $w_I(n)$
$L_2^{(I)}$	Length of the second part of the LP analysis window $w_I(n)$
$L_1^{(II)}$	Length of the first part of the LP analysis window $w_{II}(n)$
$L_2^{(II)}$	Length of the second part of the LP analysis window $w_{II}(n)$
$r_{ac}(k)$	The auto-correlations of the windowed speech $s'(n)$
$w_{lag}(i)$	Lag window for the auto-correlations (60 Hz bandwidth expansion)

f_0	The bandwidth expansion in Hz
f_s	The sampling frequency in Hz
$r'_{ac}(k)$	The modified (bandwidth expanded) auto-correlations
$E_{LD}(i)$	The prediction error in the ith iteration of the Levinson algorithm
k_i	The ith reflection coefficient
$a_j^{(i)}$	The jth direct form coefficient in the ith iteration of the Levinson algorithm
$F_1'(z)$	Symmetric LSF polynomial
$F_2'(z)$	Antisymmetric LSF polynomial
$F_1(z)$	Polynomial $F_1'(z)$ with root $z=-1$ eliminated
$F_2(z)$	Polynomial $F_2'(z)$ with root $z=1$ eliminated
q_i	The line spectral pairs (LSPs) in the cosine domain
$\mathbf{q}$	An LSP vector in the cosine domain
$\hat{\mathbf{q}}_i^{(n)}$	The quantified LSP vector at the ith subframe of the frame n
ω_i	The line spectral frequencies (LSFs)
$T_m(x)$	A mth order Chebyshev polynomial
$f_1(i), f_2(i)$	The coefficients of the polynomials $F_1(z)$ and $F_2(z)$
$f_1'(i), f_2'(i)$	The coefficients of the polynomials $F_1'(z)$ and $F_2'(z)$
$f(i)$	The coefficients of either $F_1(z)$ or $F_2(z)$
$C(x)$	Sum polynomial of the Chebyshev polynomials
x	Cosine of angular frequency ω
λ_k	Recursion coefficients for the Chebyshev polynomial evaluation
f_i	The line spectral frequencies (LSFs) in Hz
$\mathbf{f}^t = [f_1\ f_2 \ldots f_{10}]$	The vector representation of the LSFs in Hz
$\mathbf{z}^{(1)}(n)$, $\mathbf{z}^{(2)}(n)$	The mean-removed LSF vectors at frame n
$\mathbf{r}^{(1)}(n)$, $\mathbf{r}^{(2)}(n)$	The LSF prediction residual vectors at frame n
$\mathbf{p}(n)$	The predicted LSF vector at frame n
$\hat{\mathbf{r}}^{(2)}(n-1)$	The quantified second residual vector at the past frame
$\hat{\mathbf{f}}^k$	The quantified LSF vector at quantization index k
E_{LSP}	The LSP quantization error
$w_i, i=1,\ldots,10,$	LSP-quantization weighting factors
d_i	The distance between the line spectral frequencies f_{i+1} and f_{i-1}
$h(n)$	The impulse response of the weighted synthesis filter
O_k	The correlation maximum of open-loop pitch analysis at delay k
$O_{t_i}, i=1,\ldots,3$	The correlation maxima at delays $t_i, i=1,\ldots,3$
$(M_i, t_i), i=1,\ldots,3$	The normalized correlation maxima M_i and the corresponding delays $t_i, i=1,\ldots,3$
$H(z)W(z) = \dfrac{A(z/\gamma_1)}{\hat{A}(z)A(z/\gamma_2)}$	The weighted synthesis filter
$A(z/\gamma_1)$	The numerator of the perceptual weighting filter
$1/A(z/\gamma_2)$	The denominator of the perceptual weighting filter

T_1	The integer nearest to the fractional pitch lag of the previous (1st or 3rd) subframe
$s'(n)$	The windowed speech signal
$s_w(n)$	The weighted speech signal
$\hat{s}(n)$	Reconstructed speech signal
$\hat{s}'(n)$	The gain-scaled post-filtered signal
$\hat{s}_f(n)$	Post-filtered speech signal (before scaling)
$x(n)$	The target signal for adaptive codebook search
$x_2(n)$, $\mathbf{x}_2^t$	The target signal for algebraic codebook search
$res_{LP}(n)$	The LP residual signal
$c(n)$	The fixed codebook vector
$v(n)$	The adaptive codebook vector
$y(n) = v(n) * h(n)$	The filtered adaptive codebook vector
$y_k(n)$	The past filtered excitation
$u(n)$	The excitation signal
$\hat{u}(n)$	The emphasized adaptive codebook vector
$\hat{u}'(n)$	The gain-scaled emphasized excitation signal
T_{op}	The best open-loop lag
t_{min}	Minimum lag search value
t_{max}	Maximum lag search value
$R(k)$	Correlation term to be maximized in the adaptive codebook search
b_{24}	The FIR filter for interpolating the normalized correlation term $R(k)$
$R(k)_t$	The interpolated value of $R(k)$ for the integer delay k and fraction t
b_{60}	The FIR filter for interpolating the past excitation signal $u(n)$ to yield the adaptive codebook vector $v(n)$
A_k	Correlation term to be maximized in the algebraic codebook search at index k
C_k	The correlation in the numerator of A_k at index k
E_{Dk}	The energy in the denominator of A_k at index k
$\mathbf{d} = \mathbf{H}^t \mathbf{x}_2$	The correlation between the target signal $x_2(n)$ and the impulse response $h(n)$, i.e., backward filtered target
$\mathbf{H}$	The lower triangular Toepliz convolution matrix with diagonal $h(0)$ and lower diagonals $h(1), \ldots, h(39)$
$\Phi = \mathbf{H}^t \mathbf{H}$	The matrix of correlations of $h(n)$
$d(n)$	The elements of the vector $\mathbf{d}$
$\phi(i, j)$	The elements of the symmetric matrix Φ
$\mathbf{c}_k$	The innovation vector
C	The correlation in the numerator of A_k
m_i	The position of the ith pulse
ϑ_i	The amplitude of the ith pulse
N_p	The number of pulses in the fixed codebook excitation
E_D	The energy in the denominator of A_k

$res_{LTP}(n)$	The normalized long-term prediction residual
$b(n)$	The signal used for presetting the signs in algebraic codebook search
$s_b(n)$	The sign signal for the algebraic codebook search
$d'(n)$	Sign extended backward filtered target
$\phi'(i,j)$	The modified elements of the matrix Φ, including sign information
$\mathbf{z}^t$, $z(n)$	The fixed codebook vector convolved with $h(n)$
$E(n)$	The mean-removed innovation energy (in dB)
$\overline{E}$	The mean of the innovation energy
$\widetilde{E}(n)$	The predicted energy
$[b_1\ b_2\ b_3\ b_4]$	The MA prediction coefficients
$\hat{R}(k)$	The quantified prediction error at subframe k
E_I	The mean innovation energy
$R(n)$	The prediction error of the fixed-codebook gain quantization
E_Q	The quantization error of the fixed-codebook gain quantization
$e(n)$	The states of the synthesis filter $1/\hat{A}(z)$
$e_w(n)$	The perceptually weighted error of the analysis-by-synthesis search
η	The gain scaling factor for the emphasized excitation
g_c	The fixed-codebook gain
g_c'	The predicted fixed-codebook gain
$\hat{g}_c$	The quantified fixed codebook gain
g_p	The adaptive codebook gain
$\hat{g}_p$	The quantified adaptive codebook gain
$\gamma_{gc} = g_c / g_c'$	A correction factor between the gain g_c and the estimated one g_c'
$\hat{\gamma}_{gc}$	The optimum value for γ_{gc}
γ_{sc}	Gain scaling factor

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply. Further GSM related abbreviations may be found in GSM 01.04 [1].

ACELP	Algebraic Code Excited Linear Prediction
AGC	Adaptive Gain Control
AMR	Adaptive Multi-Rate
CELP	Code Excited Linear Prediction
EFR	Enhanced Full Rate
FIR	Finite Impulse Response
FR	Full Rate
HR	Half Rate
ISPP	Interleaved Single-Pulse Permutation
LP	Linear Prediction
LPC	Linear Predictive Coding
LSF	Line Spectral Frequency
LSP	Line Spectral Pair
LTP	Long Term Predictor (or Long Term Prediction)
MA	Moving Average

4 Outline description

The present document is structured as follows:

Section 4.1 contains a functional description of the audio parts including the A/D and D/A functions. Section 4.2 describes the conversion between 13-bit uniform and 8-bit A-law or μ -law samples. Sections 4.3 and 4.4 present a simplified description of the principles of the AMR codec encoding and decoding process respectively. In subclause 4.5, the sequence and subjective importance of encoded parameters are given.

Section 5 presents the functional description of the AMR codec encoding, whereas clause 6 describes the decoding procedures. In section 7, the detailed bit allocation of the AMR codec is tabulated.

4.1 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 PCM

   – microphone;

   – input level adjustment device;

   – input anti-aliasing filter;

   – sample-hold device sampling at 8 kHz;

   – analogue–to–uniform digital conversion to 13–bit representation.

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

2) Uniform digital PCM 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 or μ -law compounded format, based on a standard A-law or μ -law codec/filter according to ITU-T Recommendations G.711 [8] and G.714, followed by the 8-bit to 13-bit conversion as specified in subclause 4.2.1.

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

In the latter case it should be noted that the specifications in ITU-T 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 GSM 03.50 [2] in clause 2.