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**Prizemni snopovni radio (TETRA) – Govorni kodek za kanal s polno hitrostjo – 2.
del: Kodek TETRA**

Terrestrial Trunked Radio (TETRA); Speech codec for full-rate traffic channel; Part 2:
TETRA codec

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(TETRA)

Terrestrial Trunked Radio
(TETRA)

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Terrestrial Trunked Radio (TETRA); Speech codec for full-rate traffic channel; Part 2: TETRA codec

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Foreword

This European Standard (Telecommunications series) has been produced by ETSI Project Terrestrial Trunked Radio (TETRA).

The present document is part 2 of a multi-part deliverable covering speech codec for full-rate traffic channel, as identified below:

Part 1: "General description of speech functions";

Part 2: "TETRA codec";

Part 3: "Specific operating features";

Part 4: "Codec conformance testing";

Clause 4 provides a complete description of the full rate speech source encoder and decoder, whilst clause 5 describes the speech channel encoder and clause 6 the speech channel decoder.

Clause 7 describes the codec performance.

Clause 8 introduces the bit exact description of the codec. This description is given as an ANSI C code, fixed point, bit exact. The whole C code corresponding to the TETRA codec is given in computer files attached to the present document, and are an integral part of this multi-part deliverable.

Clause 9 describes the optional AMR codec.

Clause 10 describes the AMR speech channel encoder.

Clause 11 describes the AMR speech channel decoder.

Clause 12 introduces the AMR speech channel encoder and decoder. This description is given as an ANSI C code.

In addition to these clauses, five informative annexes are provided.

Annex A describes a possible implementation of the speech channel decoding function.

Annex B provides comprehensive indexes of all the routines and files included in the C code associated with the present document.

Annex C describes the actual quality, performance and complexity aspects of the codec.

Annex D reports detailed results from codec characterization listening and complexity tests.

Annex E contains instructions for the use of the attached electronic files.

Annex F lists informative references relevant to the speech codec.

National transposition dates	
Date of adoption of this EN:	21 January 2005
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1 Scope

The present document contains the full specification of the speech codecs for use in the Terrestrial Trunked Radio (TETRA) system.

The TETRA codec specified in clauses 4 to 8 is mandatory for all TETRA mobiles and networks. The AMR codec specified in clauses 9 to 12 is optional. If the AMR codec is implemented, the equipment shall conform to the whole of clause 9 to 12.

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 and/or edition number or version number) or non-specific.
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- [1] ETSI EN 300 392-2: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 2: Air Interface (AI)".
 - [2] ETSI TS 126 073: "Universal Mobile Telecommunications System (UMTS); ANSI-C code for the Adaptive Multi Rate speech codec (3GPP TS 26.073 Release 4)".
 - [3] ETSI TS 126 074: "Universal Mobile Telecommunications System (UMTS); Mandatory speech codec speech processing functions; AMR speech codec test sequences (3GPP TS 26.074 Release 4)".
 - [4] ETSI TS 126 090: "Universal Mobile Telecommunications System (UMTS); Mandatory Speech Codec speech processing functions AMR Speech Codec - Transcoding functions (3GPP TS 26.090 Release 4)".

3 Abbreviations

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

ACELP	Algebraic CELP
AMR	Adaptive Multi-Rate
ANSI	American National Standards Institute
BER	Bit Error Ratio
BFI	Bad Frame Indicator
BS	Base Station
CELP	Code-Excited Linear Predictive
CRC	Cyclic Redundancy Code
DSP	Digital Signal Processor
DTMF	Dual Tone Multiple Frequency
EP	Error Pattern
EQ	EQualizer test
FIR	Finite Impulse Response
GSM	Global System for Mobile communications
HT	Hilly Terrain
IRS	Intermediate Reference System

LP	Linear Prediction
LPC	Linear Predictive Coding
LSF	Line Spectral Frequency
LSP	Line Spectral Pair
MER	Message Error Rate
MNRU	Multiplicative Noise Reference Unit
MOPS	Million of Operations per Second
MOS	Mean Opinion Score
MS	Mobile Station
MSE	Mean Square Error
PDF	Probability Density Function
PUEM	Probability of Undetected Erroneous Message
RAM	Random Access Memory
RCPC	Rate-Compatible Punctured Convolutional
RF	Radio Frequency
ROM	Read-Only Memory
SCR	Source Controlled Rate
STCH	STealing CHannel
TDM	Time Division Multiplex
TU	Typical Urban
VQ	Vector Quantization
V+D	Voice + Data

4 Full rate codec

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4.1 Structure of the codec (standards.iteh.ai)

The TETRA speech codec is based on the Code-Excited Linear Predictive (CELP) coding model. In this model, a block of N speech samples is synthesized by filtering an appropriate innovation sequence from a codebook, scaled by a gain factor g_c , through two time varying filters. A simplified high level block diagram of this synthesis process, as implemented in the TETRA codec, is shown in figure 1.

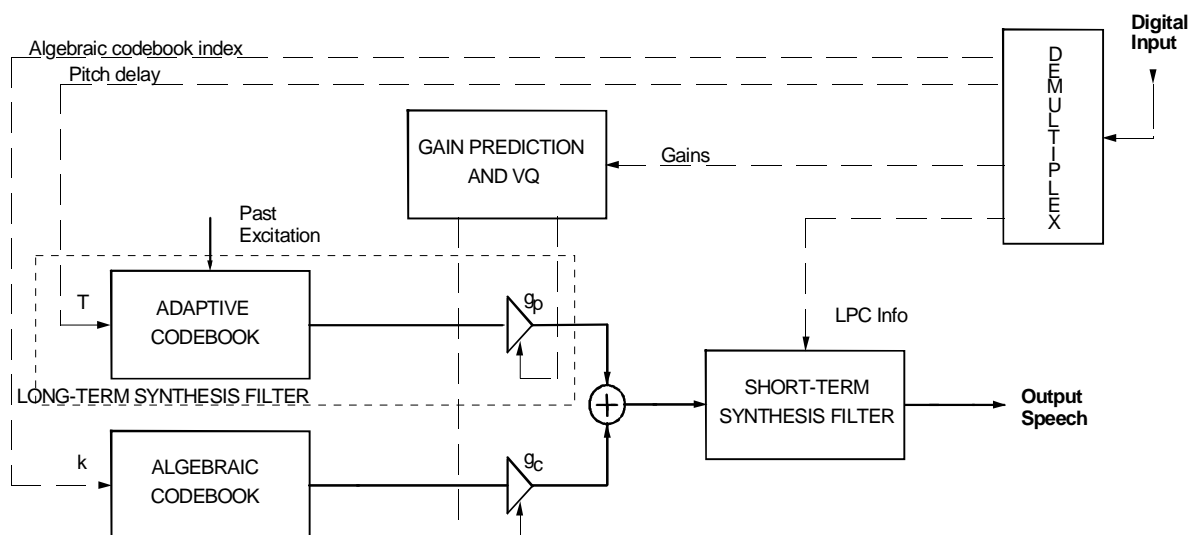


Figure 1: High level block diagram of the TETRA speech synthesizer

The first filter is a long-term prediction filter (pitch filter) aiming at modelling the pseudo-periodicity in the speech signal and the second is a short-term prediction filter modelling the speech spectral envelope.

The long-term or pitch, synthesis filter is given by:

$$\frac{1}{B(z)} = \frac{1}{1 - g_p z^{-T}} \quad (1)$$

where T is the pitch delay and g_p is the pitch gain. The pitch synthesis filter is implemented as an adaptive codebook, where for delays less than the sub-frame length the past excitation is repeated.

The short-term synthesis filter is given by:

$$H(z) = \frac{1}{A(z)} = \frac{1}{1 + \sum_{i=1}^p a_i z^{-i}} \quad (2)$$

where $a_i, i = 1, \dots, p$, are the Linear Prediction (LP) parameters and p is the predictor order. In the TETRA codec p shall be 10.

The TETRA encoder uses an analysis-by-synthesis technique to determine the pitch and excitation codebook parameters. The simplified block diagram of the TETRA encoder is shown in figure 2.

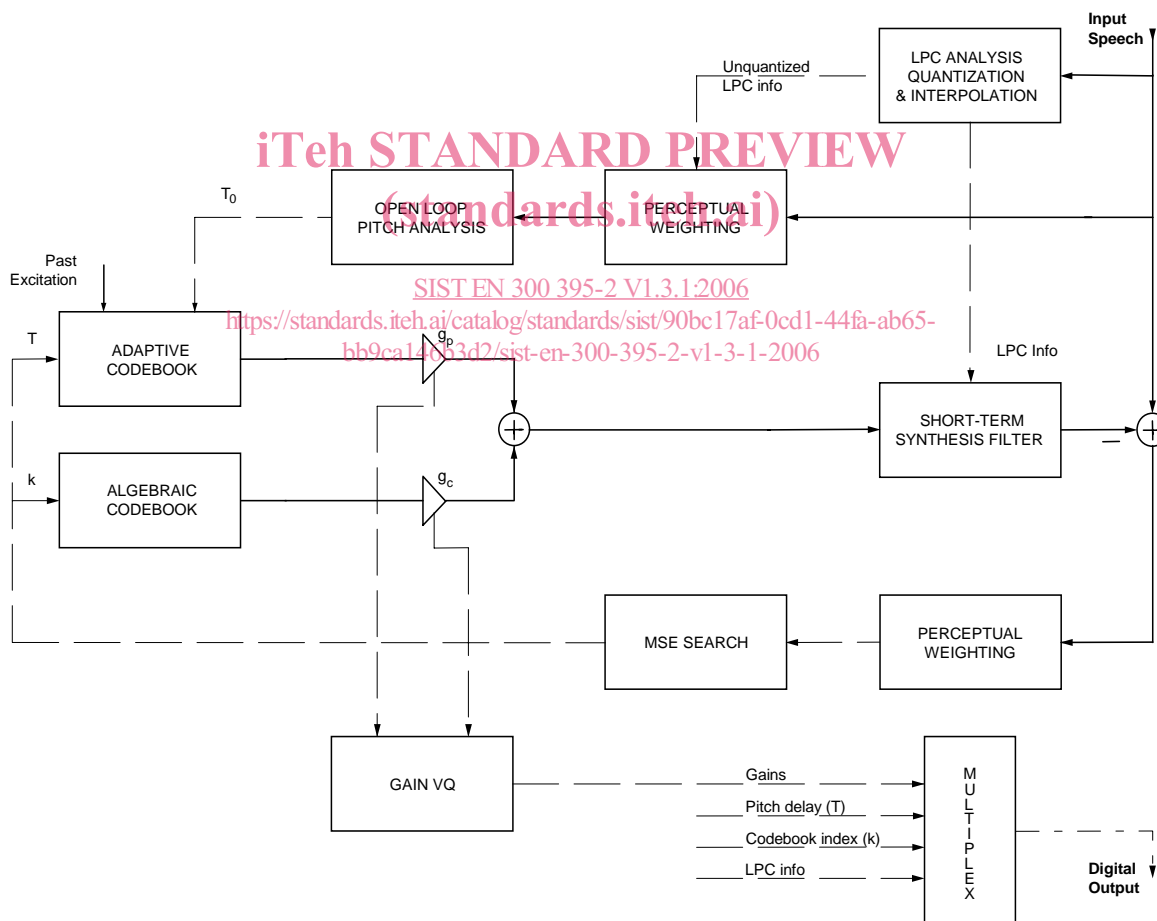


Figure 2: High level block diagram of the TETRA speech encoder

In this analysis-by-synthesis technique, the synthetic speech is computed for all candidate innovation sequences retaining the particular sequence that produces the output closer to the original signal according to a perceptually weighted distortion measure. The perceptual weighting filter de-emphasizes the error at the formant regions of the speech spectrum and is given by:

$$W(z) = \frac{A(z)}{A(z/\gamma)} \quad (3)$$

where $A(z)$ is the LP inverse filter (as in Equation (2)) and $0 < \gamma \leq 1$. The value $\gamma_1 = 0,85$ shall be used.

Both the weighting filter, $W(z)$, and formant synthesis filter, $H(z)$, shall use the quantized LP parameters.

In the Algebraic CELP (ACELP) technique, special innovation codebooks having an algebraic structure are used. This algebraic structure has several advantages in terms of storage, search complexity, and robustness. The TETRA codec shall use a specific dynamic algebraic excitation codebook whereby the fixed excitation vectors are shaped by a dynamic shaping matrix (see annex F). The shaping matrix is a function of the LP model $A(z)$, and its main role is to shape the excitation vectors in the frequency domain so that their energies are concentrated in the important frequency bands. The shaping matrix used is a Toeplitz lower triangular matrix constructed from the impulse response of the filter:

$$F(z) = \frac{A(z/\gamma_1)}{A(z/\gamma_2)} \quad (4)$$

where $A(z)$ is the LP inverse filter. The values $\gamma_1 = 0,75$ and $\gamma_2 = 0,85$ shall be used.

In the TETRA codec, 30 ms speech frames shall be used. It is required that the short-term prediction parameters (or LP parameters) are computed and transmitted every speech frame. The speech frame shall be divided into 4 sub-frames of 7,5 ms (60 samples). The pitch and algebraic codebook parameters have also to be transmitted every sub-frame.

Table 1 gives the bit allocation for the TETRA codec. 137 bits shall be produced for each frame of 30 ms resulting in a bit rate of 4 567 bit/s.

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Table 1: Bit allocation for the TETRA codec

Parameter	1 st subframe	2 nd subframe	3 rd subframe	4 th subframe	Total per frame
LP filter					26
Pitch delay	8	5	5	5	23
Algebraic code	16	16	16	16	64
VQ of 2 gains	6	6	6	6	24
Total					137

More details about the sequence of bits within the speech frame of 137 bits per 30 ms, with reference to the speech parameters, can be found in clause 4.2.2.7, table 3.

4.2 Functional description of the codec

4.2.1 Pre- and post-processing

Before starting the encoding process, the speech signal shall be pre-processed using the offset compensation filter:

$$H_p(z) = \frac{1}{2} \left(\frac{1 - z^{-1}}{1 - \alpha z^{-1}} \right) \quad (5)$$

where $\alpha = 32\,735/32\,768$. In the time domain, this filter corresponds to:

$$s'(n) = s(n)/2 - s(n-1)/2 + \alpha s'(n-1) \quad (6)$$

where $s(n)$ is the input signal and $s'(n)$ is the pre-processed signal. The purpose of this pre-processing is firstly to remove the dc from the signal (offset compensation), and secondly, to scale down the input signal in order to avoid saturation of the synthesis filtering.

At the decoder, the post-processing consists of scaling up the reconstructed signal (multiplication by 2 with saturation control).

4.2.2 Encoder

Figure 3 presents a detailed block diagram of the TETRA encoder illustrating the major parts of the codec as well as signal flow. On this figure, names appearing at the bottom of the various building blocks correspond to the C code routines associated with the present document.

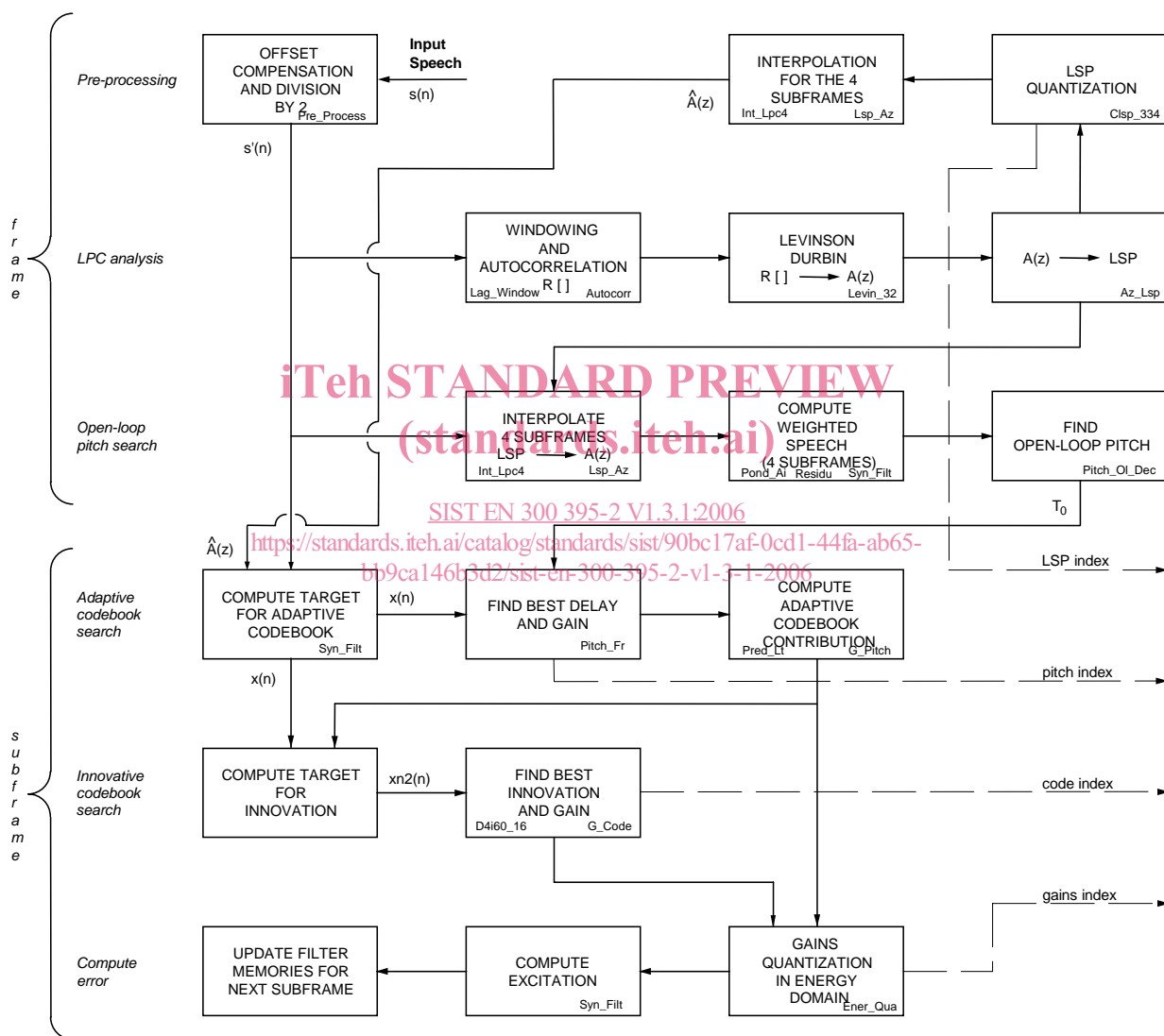


Figure 3: Signal flow at the encoder