**SLOVENSKI STANDARD**
**SIST ETS 300 395-2 E1:2003**

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**Radijska oprema in sistemi (RES) – Vseevropski snopovni radio – Govorni kodek za prometni kanal s polno hitrostjo – 2. del: Kodek TETRA**

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



**Ta slovenski standard je istoveten z:** **ETS 300 395-2 Edition 1**

**ICS:**

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| 33.070.10 | Prizemni snopovni radio (TETRA) | Terrestrial Trunked Radio (TETRA) |

**SIST ETS 300 395-2 E1:2003** **en**

# EUROPEAN TELECOMMUNICATION STANDARD

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# Radio Equipment and Systems (RES); Trans-European Trunked Radio (TETRA); Speech codec for full-rate traffic channel; Part 2: TETRA codec

**ETSI**

European Telecommunications Standards Institute

**ETSI Secretariat**

**Postal address:** F-06921 Sophia Antipolis CEDEX - FRANCE
**Office address:** 650 Route des Lucioles - Sophia Antipolis - Valbonne - FRANCE
**X.400:** c=fr, a=atlas, p=etsi, s=secretariat - **Internet:** secretariat@etsi.fr

Tel.: +33 4 92 94 42 00 - Fax: +33 4 93 65 47 16

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# Foreword

This European Telecommunication Standard (ETS) has been produced by the Radio Equipment and Systems (RES) Technical Committee of the European Telecommunications Standards Institute (ETSI).

This ETS consists of four parts as follows:

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.

Finally, 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 this ETS, and are an integral part of this ETS.

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 this ETS.



Annex C lists informative references relevant to the speech codec.



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

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

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

| **Transposition dates** | |
|---|---|
| Date of adoption | 22 November 1996 |
| Date of latest announcement of this ETS (doa): | 31 March 1997 |
| Date of latest publication of new National Standard or endorsement of this ETS (dop/e): | 30 September 1997 |
| Date of withdrawal of any conflicting National Standard (dow): | 30 September 1997 |

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# 1 Scope

This European Telecommunication Standard (ETS) contains the full specification of the speech codec for use in the Trans-European Trunked Radio (TETRA) system.

# 2 Normative references

This ETS incorporates by dated and undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referred to applies.

[1] ETS 300 392-2: "Radio Equipment and Systems (RES); Trans-European Trunked Radio (TETRA) system; Voice plus Data; Part 2: Air Interface".

[2] CCITT Recommendation P.48 (1988): "Specifications for an Intermediate Reference System".

# 3 Abbreviations

For the purposes of this ETS, the following abbreviations apply:

| | |
|---|---|
| ACELP | Algebraic CELP |
| 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 |
| FIR | Finite Impulse Response |
| 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 |
| MOS | Mean Opinion Score |
| MS | Mobile Station |
| MSE | Mean Square Error |
| PDF | Probability Density Function |
| PUEM | Probability of Undetected Erroneous Message |
| RCPC | Rate-Compatible Punctured Convolutional |
| RF | Radio Frequency |
| VQ | Vector Quantization |



# 4 Full rate codec

## 4.1 Structure of the codec

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.

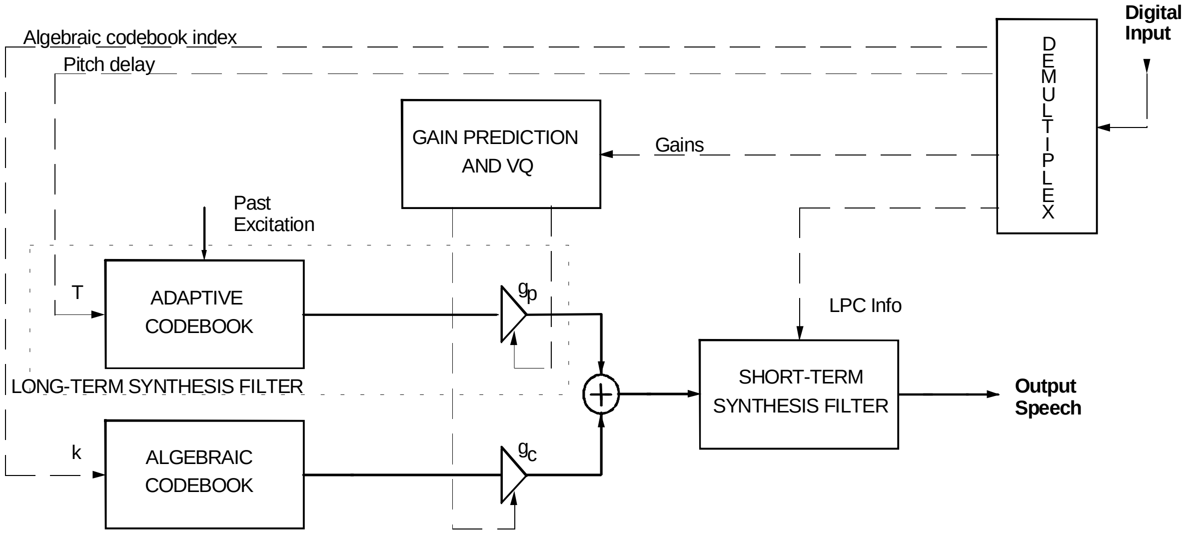


**Figure 1: High level block diagram of the TETRA speech synthesizer**

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:

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



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:

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

where $a_i, i = 1, \ldots, p$, 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.

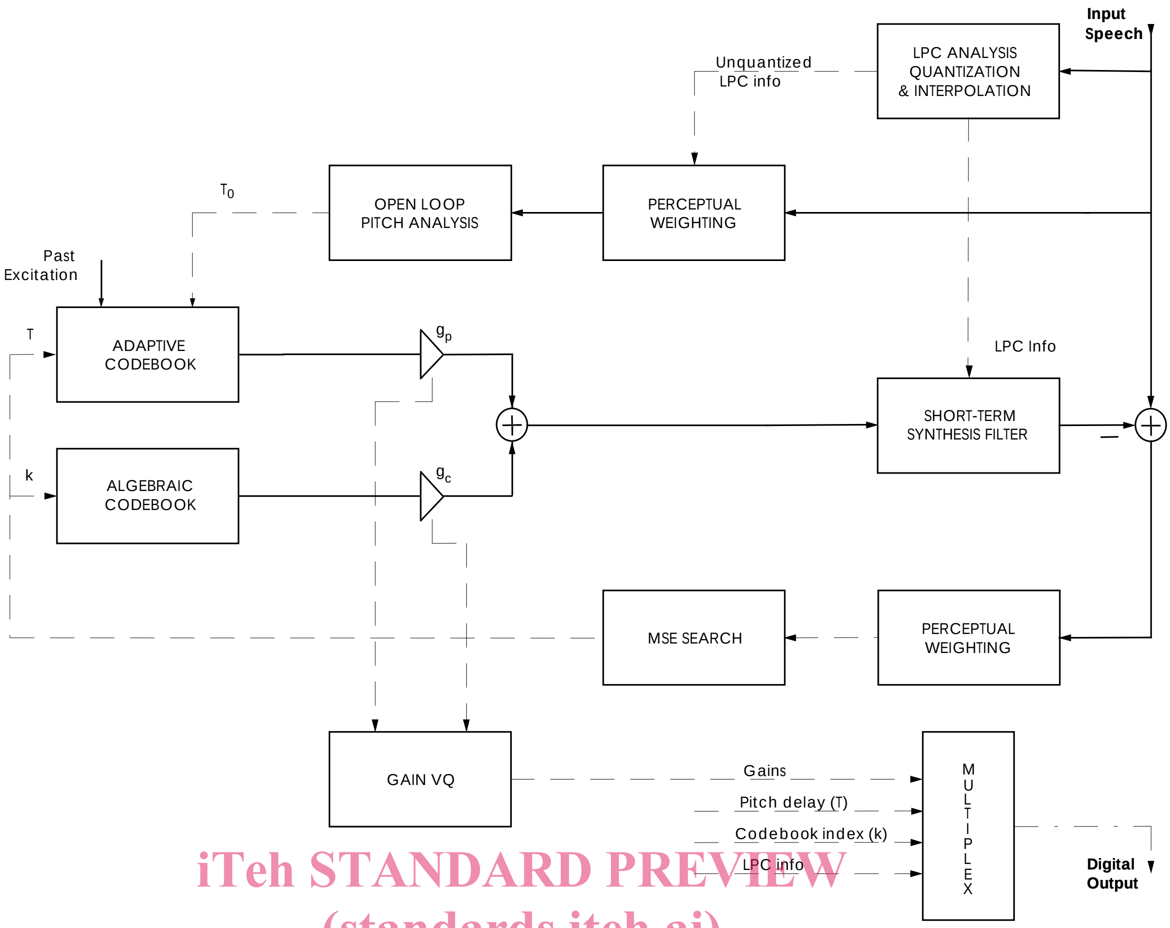




**Figure 2: High level block diagram of the TETRA speech encoder**



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:

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

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 C {1}). 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:

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

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.

**Table 1: Bit allocation for the TETRA codec**

| Parameter | 1st subframe | 2nd subframe | 3rd subframe | 4th 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 subclause 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:



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

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



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

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 this ETS.

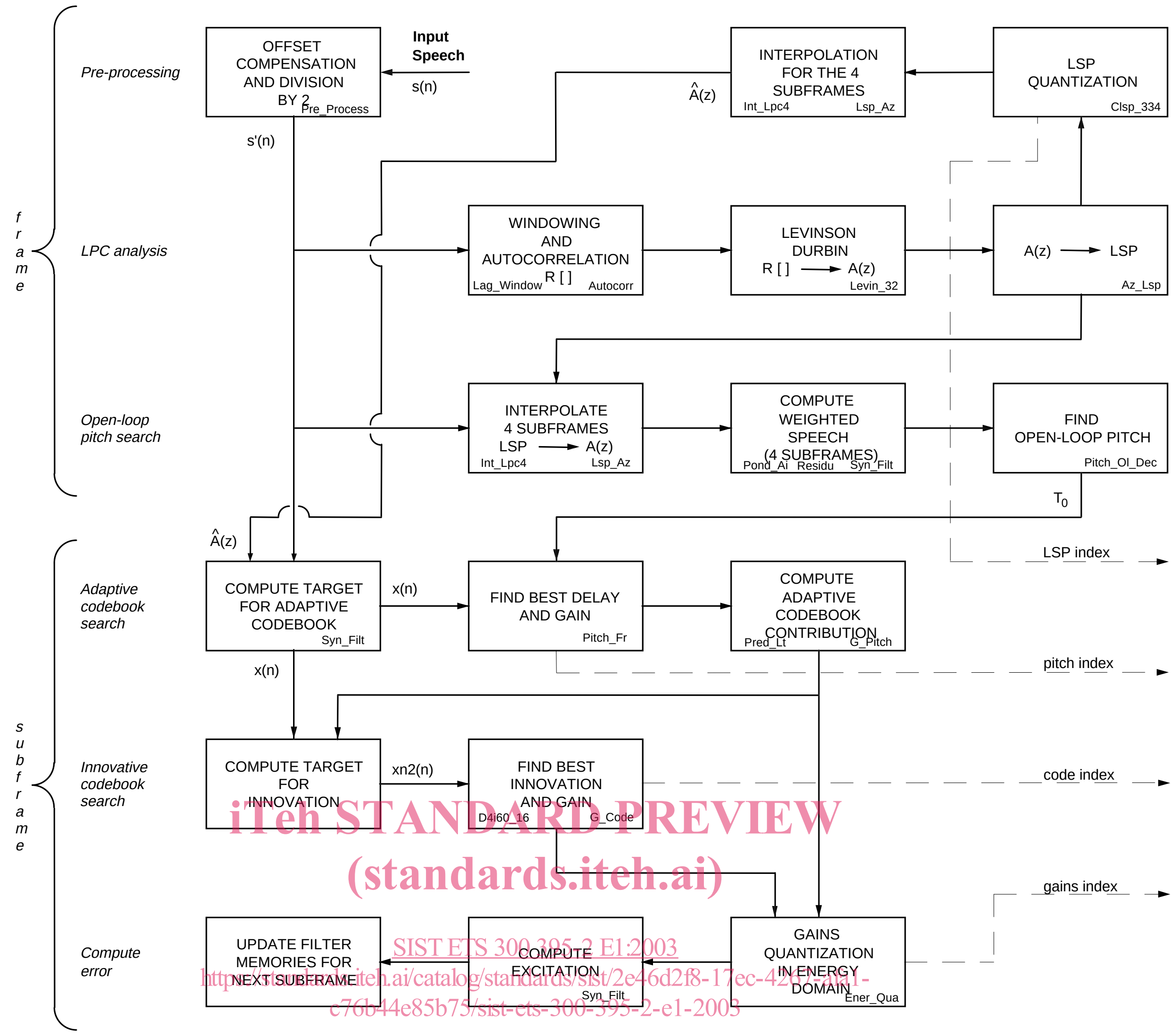




**Figure 3: Signal flow at the encoder**

### 4.2.2.1 Short-term prediction

Short-term prediction (LP or LPC analysis) shall be performed every 30 ms. The auto-correlation approach shall be used with an asymmetric analysis window. The LP analysis window consists of two halves of Hamming windows with different lengths. This window is given by:

$$\begin{aligned} w(n) &= 0{,}54 - 0{,}46\cos\left(\frac{\pi n}{L_1 - 1}\right), & n &= 0,\ldots,L_1 - 1 \\ &= 0{,}54 + 0{,}46\cos\left(\frac{\pi(n - L_1)}{L_2 - 1}\right), & n &= L_1,\ldots,L_1 + L_2 - 1 \end{aligned} \tag{7}$$

A 32 ms analysis window (corresponding to 256 samples with the sampling frequency of 8 kHz) shall be used with values $L_1 = 216$ and $L_2 = 40$. The window shall be positioned such that 40 samples are taken from the future frame (look-ahead of 40 samples).