



SLOVENSKI STANDARD

SIST ETS 300 395-2:1999

01-julij-1999

**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

iTeh STANDARD PREVIEW
(standards.iteh.ai)

Ta slovenski standard je istoveten z: **ETS 300 395-2 Edition 2**
<https://standards.iteh.ai/catalog/standards/sist/ca2651ab-711f-4116-ad04-3245bd98f38/sist-ets-300-395-2-1999>

ICS:

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

SIST ETS 300 395-2:1999

en

iTeh STANDARD PREVIEW
(standards.iteh.ai)

[SIST ETS 300 395-2:1999](#)

<https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff411b-ad04-3245bd98f38/sist-ets-300-395-2-1999>



EUROPEAN
TELECOMMUNICATION
STANDARD

ETS 300 395-2

February 1998

Second Edition

Source: TETRA

Reference: RE/TETRA-05032

ICS: 33.020

Key words: TETRA, codec

iTeh STANDARD PREVIEW
Terrestrial Trunked Radio (TETRA);
(standards.iteh.ai)
Speech codec for full-rate traffic channel;
Part 2: TETRA codec
SIST ETS 300 395-2:1999
<https://standards.iteh.ai/catalog/standards/sist/300-395-2-1999/ets-300-395-2-1999>
3245bd98f38/sist-ets-300-395-2-1999

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

Copyright Notification: No part may be reproduced except as authorized by written permission. The copyright and the foregoing restriction extend to reproduction in all media.

© European Telecommunications Standards Institute 1998. All rights reserved.

iTeh STANDARD PREVIEW (standards.iteh.ai)

[SIST ETS 300 395-2:1999](https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff411b-ad04-3245bd98f38/sist-ets-300-395-2-1999)

<https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff411b-ad04-3245bd98f38/sist-ets-300-395-2-1999>

Contents

Foreword	7
1 Scope	10
2 Normative references	10
3 Abbreviations	10
4 Full rate codec	11
4.1 Structure of the codec	11
4.2 Functional description of the codec	13
4.2.1 Pre- and post-processing	13
4.2.2 Encoder	13
4.2.2.1 Short-term prediction	14
4.2.2.2 LP to LSP and LSP to LP conversion	15
4.2.2.3 Quantization and interpolation of LP parameters	17
4.2.2.4 Long-term prediction analysis	18
4.2.2.5 Algebraic codebook: structure and search	19
4.2.2.6 Quantization of the gains	22
4.2.2.7 Detailed bit allocation	24
4.2.3 Decoder	24
4.2.3.1 Decoding process	25
4.2.3.1.1 Decoding of LP filter parameters	25
4.2.3.1.2 Decoding of the adaptive codebook vector	25
4.2.3.1.3 Decoding of the innovation vector	26
4.2.3.1.4 Decoding of the adaptive and innovative codebook gains	26
4.2.3.1.5 Computation of the reconstructed speech	26
4.2.3.2 Error concealment	26
5 Channel coding for speech	27
5.1 General	27
5.2 Interfaces in the error control structure	27
5.3 Notations	29
5.4 Definition of sensitivity classes and error control codes	29
5.4.1 Sensitivity classes	29
5.4.2 CRC codes	29
5.4.3 16-state RCPC codes	31
5.4.3.1 Encoding by the 16-state mother code of rate 1/3	31
5.4.3.2 Puncturing of the mother code	31
5.5 Error control scheme for normal speech traffic channel	32
5.5.1 CRC code	32
5.5.2 RCPC codes	32
5.5.2.1 Puncturing scheme of the RCPC code of rate 8/12 (equal to 2/3)	32
5.5.2.2 Puncturing scheme of the RCPC code of rate 8/18	32
5.5.3 Matrix Interleaving	33
5.6 Error control scheme for speech traffic channel with frame stealing activated	34
5.6.1 CRC code	34
5.6.2 RCPC codes	35
5.6.2.1 Puncturing scheme of the RCPC code of rate 8/17	36
5.6.3 Interleaving	36
6 Channel decoding for speech	36
6.1 General	36

6.2	Error control structure	36
7	Codec performance	37
8	Bit exact description of the TETRA codec	37
Annex A (informative): Implementation of speech channel decoding		39
A.1	Algorithmic description of speech channel decoding	39
A.1.1	Definition of error control codes	39
A.1.1.1	16-state RCPC codes	39
A.1.1.1.1	Obtaining the mother code from punctured code	39
A.1.1.1.2	Viterbi decoding of the 16-state mother code of the rate 1/3	39
A.1.1.2	CRC codes	40
A.1.1.3	Type-4 bits	40
A.1.2	Error control scheme for normal speech traffic channel	40
A.1.2.1	Matrix Interleaving	40
A.1.2.2	RCPC codes	40
A.1.2.2.1	Puncturing scheme of the RCPC code of rate 8/12 (equal to 2/3)	41
A.1.2.2.2	Puncturing scheme of the RCPC code of rate 8/18	41
A.1.2.3	CRC code	41
A.1.2.4	Speech parameters	41
A.1.3	Error control scheme for speech traffic channel with frame stealing activated	41
A.1.3.1	Interleaving	41
A.1.3.2	RCPC codes	41
A.1.3.2.1	Puncturing scheme of the RCPC code of rate 8/17	42
A.1.3.3	CRC code	42
A.1.3.4	Speech parameters	42
A.2	C Code for speech channel decoding	42
Annex B (informative): Indexes		43
B.1	Index of C code routines	43
B.2	Index of files	46
Annex C (informative): Bibliography		47
Annex D (informative): Codec performance		48
D.1	General	48
D.2	Quality	48
D.2.1	Subjective speech quality	48
D.2.1.1	Description of characterization tests	48
D.2.1.2	Absolute speech quality	48
D.2.1.3	Effect of input level	48
D.2.1.4	Effect of input frequency characteristic	48
D.2.1.5	Effect of transmission errors	48
D.2.1.6	Effect of tandeming	49
D.2.1.7	Effect of acoustic background noise	49
D.2.1.8	Effect of vocal effort	49
D.2.1.9	Effect of frame stealing	49
D.2.1.10	Speaker and language dependency	49
D.2.2	Comparison with analogue FM	49
D.2.2.1	Analogue and digital systems results	49
D.2.2.2	All conditions	50
D.2.2.3	Input level	50
D.2.2.4	Error patterns	51
D.2.2.5	Background noise	51

D.2.3	Additional tests.....	51
D.2.3.1	Types of signals	51
D.2.3.2	Codec behaviour	51
D.3	Performance of the channel coding/decoding for speech.....	52
D.3.1	Classes of simulation environment conditions.....	52
D.3.2	Classes of equipment	52
D.3.3	Classes of bits	53
D.3.4	Channel conditions	53
D.3.5	Results for normal case	53
D.4	Complexity.....	54
D.4.1	Complexity analysis	54
D.4.1.1	Measurement methodology.....	54
D.4.1.2	TETRA basic operators	54
D.4.1.3	Worst case path for speech encoder	56
D.4.1.4	Worst case path for speech decoder	57
D.4.1.5	Condensed complexity values for encoder and decoder	58
D.4.2	DSP independence	59
D.4.2.1	Program control structure.....	59
D.4.2.2	Basic operator implementation.....	59
D.4.2.3	Additional operator implementation.....	59
D.5	Delay	59
Annex E (informative): Results of the TETRA codec characterization listening and complexity tests....		60
E.1	Characterization listening test	60
E.1.1	Experimental conditions.....	60
E.1.2	Tables of results	61
E.2	TETRA codec complexity study	75
E.2.1	Computational analysis results.....	75
E.2.1.1	TETRA speech encoder.....	75
E.2.1.2	TETRA speech decoder.....	83
E.2.1.3	TETRA channel encoder and decoder.....	86
E.2.2	Memory requirements analysis results	88
E.2.2.1	TETRA speech encoder.....	88
E.2.2.2	TETRA speech decoder.....	89
E.2.2.3	TETRA speech channel encoder	89
E.2.2.4	TETRA speech channel decoder	90
Annex F (informative): Description of attached computer files		91
F.1	Directory C-WORD.....	91
F.2	Directory C-CODE.....	91
History.....		92

Blank page

iTeh STANDARD PREVIEW
(standards.iteh.ai)

[SIST ETS 300 395-2:1999](https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff411b-ad04-3245bd98f38/sist-ets-300-395-2-1999)

<https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff411b-ad04-3245bd98f38/sist-ets-300-395-2-1999>

Foreword

This European Telecommunication Standard (ETS) has been produced by Terrestrial Trunked Radio (TETRA) Project of the European Telecommunications Standards Institute (ETSI).

The sole purpose of the copyright statement below is to protect the documentation of the standard itself and not the technology which is described therein.

ETSI states that the technology described herein is in the sole ownership of THOMSON-CSF (subject to the right of it's associated partner) and the disclosure of the standard documentation cannot be construed as granting any right whatsoever by licence or otherwise on the said technology.

THOMSON-CSF has undertaken to grant licences for the technology described in the standard, on fair, reasonable and non-discriminatory terms and conditions to the users of the present standard in accordance with the ETSI IPR Policy. Such licences are subject to a licence agreement to be agreed upon and entered into with THOMSON-CSF.

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 of this ETS:	23 January 1998
Date of latest announcement of this ETS (doa):	31 May 1998
Date of latest publication of new National Standard or endorsement of this ETS (dop/e):	30 November 1998
Date of withdrawal of any conflicting National Standard (dow):	30 November 1998

iTeh STANDARD PREVIEW (standards.iteh.ai)

[SIST ETS 300 395-2:1999](https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff411b-ad04-3245bd98f38/sist-ets-300-395-2-1999)

<https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff411b-ad04-3245bd98f38/sist-ets-300-395-2-1999>

Blank page

iTeh STANDARD PREVIEW
(standards.iteh.ai)

[SIST ETS 300 395-2:1999](https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff411b-ad04-3245bd98f38/sist-ets-300-395-2-1999)

<https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff411b-ad04-3245bd98f38/sist-ets-300-395-2-1999>

1 Scope

This European Telecommunication Standard (ETS) contains the full specification of the speech codec for use in the Terrestrial 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
EQ	Equalizer test
EP	Error Pattern
FIR	Finite Impulse Response
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
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
TDM	Time Division Multiplex
TU	Typical Urban
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.

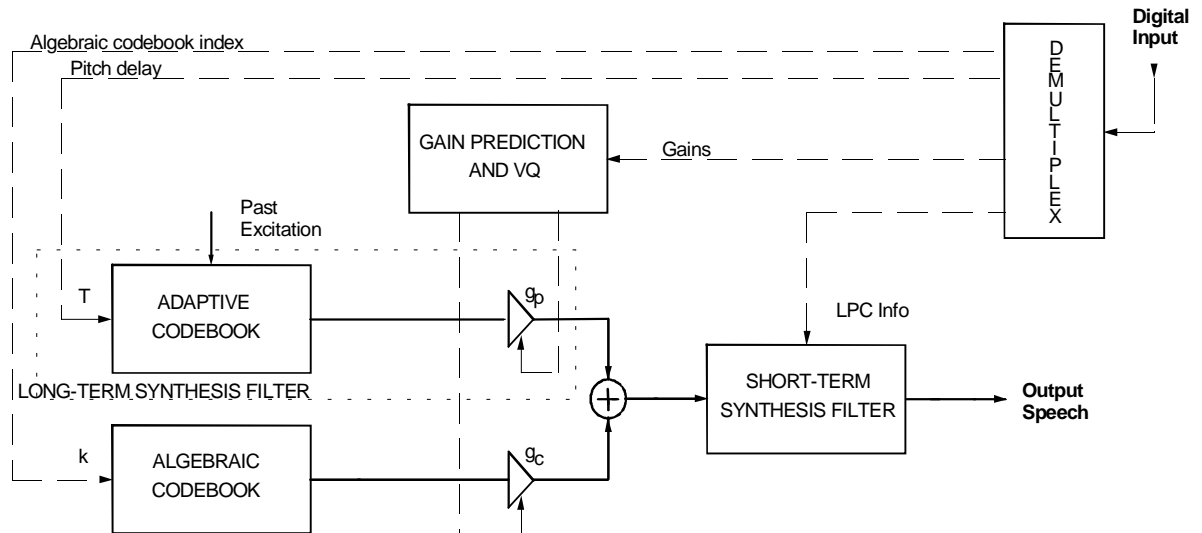


Figure 1: High level block diagram of the TETRA speech synthesizer
(standards.iteh.ai)

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.

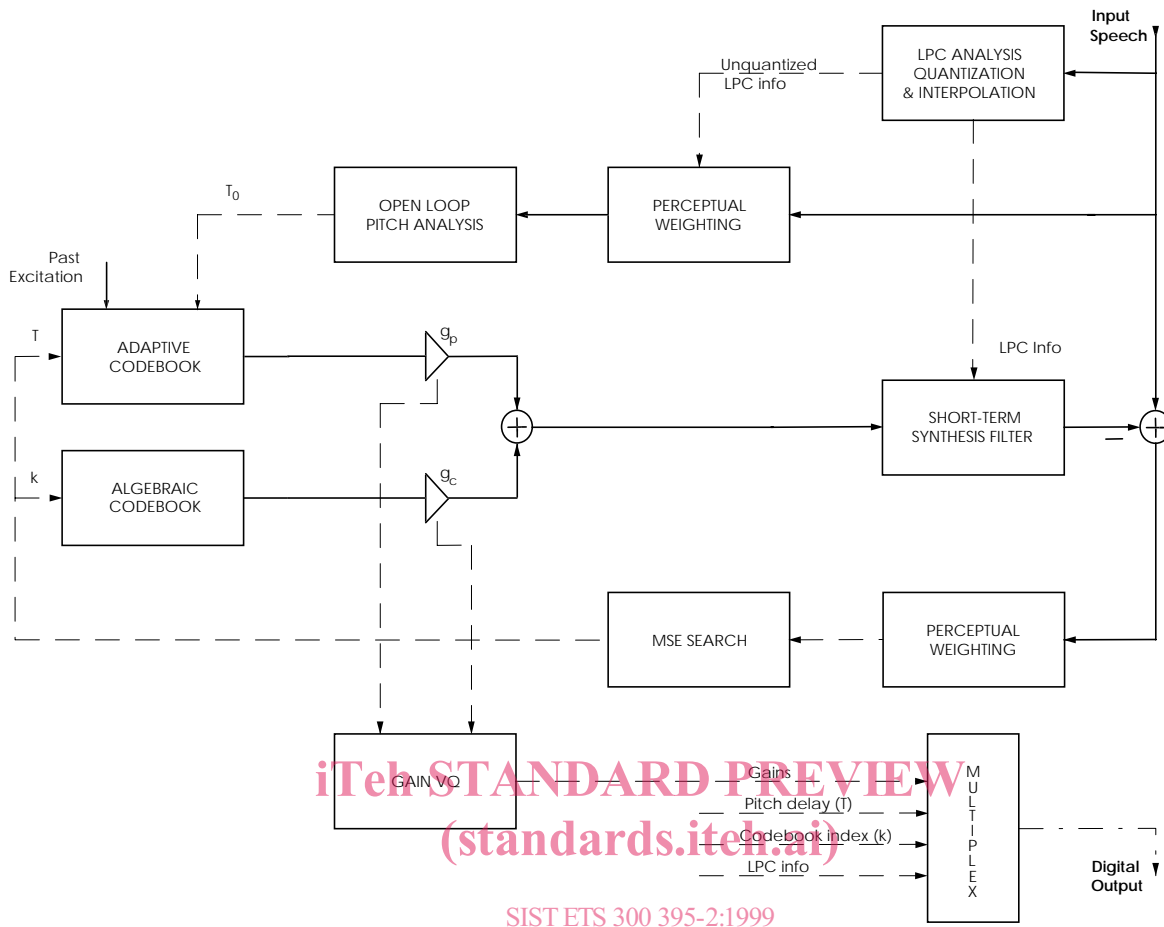


Figure 2: High level block diagram of the TETRA speech encoder

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 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:

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

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.

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:

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

(standards.iteh.ai)
SIST ETS 300 395-2:1999
<https://standards.iteh.ai/catalog/standards/sist/ca2b5fab-7ff-f411b-ad04-3245bd98f38/sist-ets-300-395-2-1999>

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 this ETS.