



INTERNATIONAL TELECOMMUNICATION UNION

# ITU-T

TELECOMMUNICATION  
STANDARDIZATION SECTOR  
OF ITU

# G.729

## Annex I

(02/00)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA,  
DIGITAL SYSTEMS AND NETWORKS

Digital transmission systems – Terminal equipments –  
Coding of analogue signals by methods other than PCM

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Coding of speech at 8 kbit/s using Conjugate  
Structure Algebraic Code-Excited Linear Prediction  
(CS-ACELP)

**Annex I: Reference fixed-point implementation  
for integrating G.729 CS-ACELP speech coding  
main body with Annexes B, D and E**

ITU-T Recommendation G.729 – Annex I

(Previously CCITT Recommendation)

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# **ITU-T RECOMMENDATION G.729**

## **CODING OF SPEECH AT 8 kbit/s USING CONJUGATE STRUCTURE ALGEBRAIC CODE-EXCITED LINEAR PREDICTION (CS-ACELP)**

### **ANNEX I**

#### **Reference fixed-point implementation for integrating G.729 CS-ACELP speech coding main body with Annexes B, D and E**

#### **Summary**

This annex describes the integration of G.729 main body with Annexes B, D and E.

This annex includes an electronic attachment containing version 1.1 of reference C code and test vectors for fixed-point implementation of CS-ACELP at 6.4 kbit/s, 8 kbit/s and 11.8 kbit/s with DTX functionality.

#### **Source**

Annex I to ITU-T Recommendation G.729 was prepared by ITU-T Study Group 16 (1997-2000) and was approved under the WTSC Resolution No. 1 procedure on 17 February 2000.

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The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1.

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### Electronic attachment:

- reference C code implementation
- test vectors



## Recommendation G.729

### CODING OF SPEECH AT 8 kbit/s USING CONJUGATE STRUCTURE ALGEBRAIC CODE-EXCITED LINEAR PREDICTION (CS-ACELP)

#### ANNEX I

#### Reference fixed-point implementation for integrating G.729 CS-ACELP speech coding main body with Annexes B, D and E<sup>1</sup>

(Geneva, 2000)

#### I.1 Scope

This annex provides a description of integrating the G.729 main body with Annexes B, D and E, hereby defining the integrated C code. It presents a standard way of performing this integration and expansion of the functionality thereby guiding the industry and ensuring a standard speech quality and compatibility worldwide. The integration has been performed with focus on several constraints in order to satisfy the need of the industry:

- 1) Bit-exactness with the main body and individual annexes.
- 2) Minimum additional program code, memory, and complexity usage.
- 3) Stringent quality requirements to new functionality inline with quality and application areas of the according standard annexes.

#### I.2 Normative references

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

- [1] ITU-T Recommendation G.729 (1996), *Coding of speech at 8 kbit/s using conjugate structure algebraic code-excited linear prediction (CS-ACELP)*.
- [2] ITU-T Recommendation G.729 Annex B (1996), *A silence compression scheme for G.729 optimized for terminals conforming to Recommendation V.70*.
- [3] ITU-T Recommendation G.729 Annex D (1998), *6.4 kbit/s CS-ACELP speech coding algorithm*.
- [4] ITU-T Recommendation G.729 Annex E (1998), *11.8 kbit/s CS-ACELP speech coding algorithm*.

#### I.3 Overview

Recommendation G.729 main body [1] and Annexes B [2], D [3] and E [4] provide a bit-exact, fixed-point specification of a CS-ACELP coder at 8 kbit/s, with DTX functionality, lower and higher bit-rate extension capability at 6.4 kbit/s and 11.8 kbit/s. Exact details of these specifications are

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<sup>1</sup> This annex includes an electronic attachment containing version 1.1 of reference C code and test vectors for fixed-point implementation of CS-ACELP at 6.4 kbit/s, 8 kbit/s and 11.8 kbit/s with DTX functionality.

given in bit-exact, fixed-point C code in an electronic attachment to this annex. This annex describes and defines the integration of the G.729 main body with Annexes B, D and E.

## **I.4 New functionality**

This subclause presents a brief overview of the modifications/additions to the algorithms in order to facilitate the integration of the main body and Annexes B, D and E. Also certain additions have been found necessary in order to accommodate the application area of the different modules.

### **I.4.1 Annex B DTX operation with Annex D**

Integrating Annexes B and D functionality in order to provide DTX operation with Annex D is straightforward. The VAD (Voice Activity Detection), SID coding (Silence Description), and CNG (Comfort Noise Generation) of Annex B are reused without any modifications. Care is taken to update the parameters for the phase dispersion for the postfilter in Annex D during discontinued transmission (see I.5.2).

### **I.4.2 Annex B DTX operation with Annex E**

Integrating Annexes B and E functionality in order to provide DTX operation with Annex E is slightly more involved. Since the DTX operation of Annex B is based on the 10th order LPC analysis, the VAD function of Annex B is performed after the 10th order forward adaptive LPC analysis and before the backward adaptive LPC analysis of Annex E. In case the VAD function detects "non-speech", the LPC mode of Annex E is forced to forward adaptive LPC and the backward adaptive LPC analysis is skipped. Furthermore, it has been found necessary to add a correctional module after the VAD in order to detect music and accommodate the somewhat expanded application area of Annex E – one of the purposes of Annex E is to provide transmission capability of music with a certain quality. Accordingly, during the development of Annex E there were strict requirements to the performance with music signals. On the other hand, for the main body and Annexes B and D there were no strict requirements to the performance with music signals. In order to guarantee the quality with music signals of Annex E during Annex B DTX operation, the music detection function forces the VAD to "speech" during music segments, hereby ensuring that the music segments are coded with the 11.8 kbit/s of Annex E. The SID coding and the CNG of Annex B are reused without any modifications. Furthermore, care is taken to appropriately update the parameters of the LPC mode selection algorithm of Annex E during discontinued transmission (see I.5.3).

## **I.5 Algorithm description**

This subclause presents the algorithm description of the necessary additions to the algorithms of the individual annexes in order to facilitate the integration. All remaining modules originate from the main body, Annex B, D or E.

### **I.5.1 Music detection**

The music detection is a new function. It is performed immediately following the VAD and forces the VAD to "speech" during music segments. It is active only during Annex E operation, though its parameters are updated continuously independently of bit-rate mode during DTX operation of the integrated G.729.

The music detection algorithm corrects the decision from the Voice Activity Detection (VAD) in the presence of music signals. It is used in conjunction with Annex E during Annex B DTX operation, i.e. in Discontinuous Transmission mode. The music detection is based on the following parameters:

- *Vad\_dec* : VAD decision of the current frame.
- *PVad\_dec* : VAD decision of the previous frame.



- *Lpc\_mod*: flag indicator of either forward or backward adaptive LPC of the previous frame.
- *Rc*: reflection coefficients from LPC analysis.
- *Lag\_buf*: buffer of corrected open loop pitch lags of last 5 frames.
- *Pgain\_buf*: buffer of closed loop pitch gain of last 5 subframes.
- *Energy*: first autocorrelation coefficient  $R(0)$  from LPC analysis.
- *LEnergy*: normalized log energy from VAD module.
- *Frm\_count*: counter of the number of processed signal frames.
- *Rate*: selection of speech coder.

The algorithm has two main parts:

- 1) Computation of relevant parameters.
- 2) Classification based on parameters.

#### I.5.1.1 Computation of Relevant Parameters

This subclause describes the computation of the parameters used by the decision module.

##### Partial Normalized Residual Energy

$$LEnergy = 10 \log_{10} \left( \prod_{i=1}^4 (1 - Rc(i)^2) \frac{Energy}{240} \right)$$

##### Spectral Difference and Running Mean of Partial Normalized Residual Energy of Background Noise

A spectral difference measure between the current frame reflection coefficients  $Rc$  and the running mean reflection coefficients of the background noise  $mRc$  is given by:

$$SD = \sum_{i=1}^{10} (Rc(i) - mRc(i))^2$$

The running means  $\overline{mrc}$  and  $mLEnergy$  are updated as follows using the VAD decision  $Vad\_deci$  that was generated by the VAD module.

```

if  $Vad\_deci == NOISE$  {
     $\overline{mrc} = 0.9\overline{mrc} + 0.1rc$ 
     $mLEnergy = 0.9mLEnergy + 0.1LEnergy$ 
}

```

##### Open loop Pitch Lag Correction for Pitch Lag Buffer Update

The open loop pitch lag  $T_{op}$  is corrected to prevent pitch doubling or tripling as follows:

$$avg\_lag = \sum_{i=1}^4 \frac{Lag\_buf(i)}{4}$$

$$\begin{aligned}
& \text{if} \left( \text{abs} \left( \frac{T_{op}}{2} - \text{avg\_lag} \right) \leq 2 \right) \\
& \quad \text{Lag\_buf}(5) = \frac{T_{op}}{2} \\
& \text{else if} \left( \text{abs} \left( \frac{T_{op}}{3} - \text{avg\_lag} \right) \leq 2 \right) \\
& \quad \text{Lag\_buf}(5) = \frac{T_{op}}{3} \\
& \text{else} \\
& \quad \text{Lag\_buf}(5) = T_{op}
\end{aligned}$$

It should be noted that the open loop pitch lag  $T_{op}$  is not modified and is the same as derived by the open loop analysis.

### Pitch Lag Standard Deviation

$$std = \sqrt{\frac{Var}{4}}$$

where:

$$Var = \sum_{i=1}^{i=5} (\text{Lag\_buf}(i) - \mu)^2 \quad \text{and} \quad \mu = \sum_{i=1}^{i=5} \left[ \frac{\text{Lag\_buf}(i)}{5} \right]$$

### Running Mean of Pitch Gain

$$mPgain = 0.8mPgain + 0.2\theta, \quad \text{where} \quad \theta = \sum_{i=1}^{i=5} \left[ \frac{Pgain\_buf(i)}{5} \right]$$

The pitch gain buffer  $Pgain\_buf$  is updated after the subframe processing with a pitch gain value of 0.5 if  $Vad\_deci = NOISE$ , and otherwise with the quantized pitch gain.

### Pitch Lag Smoothness and Voicing Strength Indicator

A pitch lag smoothness and voicing strength indicator  $Pflag$  is generated using the following logical steps:

First, two intermediary logical flags  $Pflag1$  and  $Pflag2$  are obtained as:

$$\begin{aligned}
& \text{if} (std < 1.3 \text{ and } mPgain > 0.45) \text{ set } Pflag1 = 1 \text{ else } 0 \\
& \text{if} (mPgain > Thres) \text{ set } Pflag2 = 1 \text{ else } 0, \\
& \text{where } Thres = 0.73 \text{ if } Rate = G729D, \text{ otherwise } Thres = 0.63
\end{aligned}$$

Finally,  $Pflag$  is determined from the following:

$$\begin{aligned}
& \text{if} ((PVad\_dec == VOICE \text{ and } (Pflag1 == 1 \text{ or } Pflag2 == 1)) \text{ or } (Pflag2 == 1)) \\
& \quad \text{set } Pflag = 1 \text{ else } 0
\end{aligned}$$

## Stationarity counters

A set of counters are defined and updated as follows:

- a) *count\_consc\_rflag* tracks the number of consecutive frames where the 2nd reflection coefficient and the running mean of the pitch gain satisfy the following condition:

if ( $R_c(2) < 0.45$  and  $R_c(2) > 0$  and  $mP_{gain} < 0.5$ )

$count\_consc\_rflag = count\_consc\_rflag + 1$

else

$count\_consc\_rflag = 0$

- b) *count\_music* tracks the number of frames where the previous frame uses backward adaptive LPC and the current frame is "speech" (according to the VAD) within a window of 64 frames.

if ( $Lpc\_mod == 1$  and  $Vad\_deci == VOICE$ )

$count\_music = count\_music + 1$

Every 64 frames, a running mean of *count\_music*, *mcount\_music* is updated and reset to zero as described below:

if ( $(Frm\_count \bmod 64) == 0$ ) {

if ( $Frm\_count == 64$ )

$mcount\_music = count\_music$

else

$mcount\_music = 0.9mcount\_music + 0.1count\_music$

}

- c) *count\_consc* tracks the number of consecutive frames where the *count\_music* remains zero:

if ( $count\_music == 0$ )

$count\_consc = count\_consc + 1$

else

$count\_consc = 0$

if ( $count\_consc > 500$  or  $count\_consc\_rflag > 150$ ) set  $mcount\_music = 0$

*count\_music* in b) is reset to zero every 64 frames after the update of the relevant counters.

The logic in c) is used to reset the running mean *count\_music*.

- d) *count\_pflag* tracks the number of frames where  $Pflag = 1$ , within a window of 64 frames.

if ( $Pflag == 1$ )

$count\_pflag = count\_pflag + 1$

Every 64 frames, a running mean of *count\_pflag* , *mcount\_pflag* , is updated and reset to zero as described below:

```
if ((Frm_count mod 64) == 0){
    if (Frm_count == 64)
        mcount_pflag = count_pflag
    else{
        if (count_pflag > 25)
            mcount_pflag = 0.98mcount_pflag + 0.02count_pflag
        else (count_pflag > 20)
            mcount_pflag = 0.95mcount_pflag + 0.05count_pflag
        else
            mcount_pflag = 0.9mcount_pflag + 0.1count_pflag
    }
}
```

- e) *count\_consc\_pflag* tracks the number of consecutive frames satisfying the following condition:

```
if (count_pflag == 0)
    count_consc_pflag = count_consc_pflag + 1
else
    count_consc_pflag = 0
```

if (*count\_consc\_pflag* > 100 or *count\_consc\_rflag* > 150) set *mcount\_pflag* = 0

*count\_pflag* is reset to zero every 64 frames. The logic in e) is used to reset the running mean of *count\_pflag* .

### I.5.1.2 Classification

Based on the estimation of the above parameters, the VAD decision *Vad\_deci* from the VAD module is reverted if the following conditions are satisfied:

```
if (Rate = G729E){
    if (SD > 0.15 and (Lenergy - mLenergy) > 4 and LLenergy > 50)
        Vad_deci = VOICE
    else if ((SD > 0.38 or (Lenergy - mLenergy) > 4) and LLenergy > 50)
        Vad_deci = VOICE
    else if ((mcount_pflag >= 10 or mcount_music >= 1.0938 or Frm_count < 64)
        and LLenergy > 7)
        Vad_deci = VOICE
}
```

Note that the music detection function is called all the time regardless of the operational coding mode in order to keep the memories current. However, the VAD decision *Vad\_deci* is altered only if the integrated G.729 is operating at 11.8 kbit/s (Annex E). It should be noted that the music detection only has the capability to change the decision from "non-speech" to "speech" and not vice versa.

## **I.5.2 Update of state variables specific to Annex D during discontinued transmission**

The only state variables specific to Annex D are the state variables of the phase dispersion module (see D.6.2 of ITU-T G.729 – Annex D [3]) at the decoder. In case of inactive frames, the same update procedure as in case of nominal bit rate (8 kbit/s) is followed using as adaptive and ACELP gain estimations the gain values computed by the comfort noise excitation generator (see B.4.4 of ITU-T G.729 – Annex B [2]). Note also that the update for the higher rate is identical to the update for the nominal bit rate.

## **I.5.3 Update of state variables specific to Annex E during discontinued transmission**

### **I.5.3.1 Update of encoder state variables specific to Annex E**

At the encoder in case of inactive frames, the update of state variables is identical to the update performed in G.729 Annex E [4] in case of switch to the nominal bit rate 8 kbit/s. The update procedure is the following: the LP mode is set to 0, the global stationarity indicator is decreased and the high stationarity indicator is reset to 0 (see E.3.2.7.2 of ITU-T G.729 – Annex E [4]), the interpolation factor used to smoothly switch from LP forward filter to backward LP filter is reset to its maximum value (see E.3.2.7.1 of ITU-T G.729 – Annex E [4]). Note that this update is also performed in case of switch to the lower bit rate 6.4 kbit/s.

### **I.5.3.2 Update of decoder state variables specific to Annex E during discontinued transmission**

At the decoder in case of inactive frames, the update of state variables is almost identical to the update performed in G.729 Annex E [4] in case of switch to the forward mode only rates (8 kbit/s and 6.4 kbit/s) except that the pitch delay stationary indicator is reset to 0 instead of being computed by the pitch tracking procedure (see E.4.4.5 of ITU-T G.729 – Annex E [4]).

## **I.6 Description of C source code**

Annex I of G.729, integrating the G.729 main body with Annexes B, D and E, is simulated in 16-bit fixed-point ANSI-C code using the same types of fixed-point data and the same set of fixed-point basic operators as in the G.729 software. The ANSI-C code represents the normative specification of this annex. The algorithmic description given by the C code shall take precedence over the texts contained in the main body of Recommendation G.729 and in Annexes B, D, E and I. The following subclauses summarize the use of this simulation code, and how the software is organized.

### **I.6.1 Use of the simulation software**

The C code consists of two main programs **coderi.c** and **decoderi.c**, which simulate encoder and decoder, respectively. The encoder is run as follows:

**coderi inputfile bitstreamfile dtx\_option rate\_option**

The decoder is run as follows:

**decoderi bitstreamfile outputfile**

The **inputfile** and **outputfile** are 8 kHz sampled data files containing 16-bit PCM signals. The **bitstreamfile** is a binary file containing the bit stream; the mapping table of the encoded bit stream is contained in the simulation software. The two parameters are used for the encoder: **dtx\_option** and **rate\_option** where:

**dtx\_option** = 1: DTX enabled 0: DTX disabled, the default is 0 (DTX disabled).

**rate\_option** = 0 to select the lower rate (6.4 kbit/s); = 1 to select the main Recommendation G.729 (8 kbit/s); = 2 is to select the higher rate (11.8 kbit/s) or a **file\_rate\_name**: a binary file of 16-bit word containing either 0, 1, 2 to select the rate on a frame-by-frame basis; the default is 1 (8 kbit/s).

## I.6.2 Organization of the simulation software

The files can be classified into four groups:

- 1) Files identical to software files of G.729 main body [1], Annex B [2], Annex D [3] or Annex E [4], listed in Table I.1.
- 2) Files adapted from software files of G.729 Annex B, Annex D or Annex E, listed in Table I.2, some minor modifications have been introduced to cope with the integration. Most modifications come from the integration of annexes routine prototype declaration files in one file (ld8cp.h) or to the integration of extern ROM declaration annexes files into one file (tabld8cp.h). Some were introduced to deal with the update of the annexes state variables.
- 3) Files integrating G.729 software files of Annex B, Annex D or Annex E, listed in Table I.3.
- 4) Files specific to this integrated G.729 (new files) listed in Table I.4.

**Table I.1/G.729 – List of software files identical to software files of G.729 main body and Annex B, D or E**

File name	Description	Identical to
Basic_op.c	Basic operators	Main
Oper_32b.c	Extended basic operators	Main
Dspfunc.c	Mathematical functions	Main
Gainpred.c	Gain predictor	Main
lpcfunc.c	Miscellaneous routines related to LP filter	Main
Pre_proc.c	Pre-processing (HP filtering and scaling)	Main
P_parity.c	Compute pitch parity	Main
pwf.c	Computation of perceptual weighting coefficients (8 kbit/s)	Main
Pred_lt3.c	Generation of adaptive codebook	Main
Post_pro.c	Post processing (HP filtering and scaling)	Main
Tab_ld8k.c	ROM tables	Main
Basic_op.h	Basic operators prototypes	Main
Ld8k.h	Function prototypes	Main
Oper_32b.h	Extended basic operators prototypes	Main
Tab_ld8k.h	Extern ROM table declarations	Main
Typedef.h	Data type definition (machine dependent)	Main
Taming.c	Pitch instability control	B
Qsidgain.c	SID Gain Quantization	B
QsidLSF.c	SID-LSF Quantization	B
Tab_dtx.c	ROM tables	B
Sid.h	Prototype and Constants	B
Octet.h	Octet transmission mode definition	B
Tab_dtx.h	Extern ROM table declarations	B
Pwfe.c	Computation of perceptual weighting coefficients (11.8 kbit/s)	E

**Table I.2/G.729 – List of software files adapted from software files of G.729  
main body and Annex B, D or E**

<b>File name</b>	<b>Description</b>	<b>Adapted from</b>
Vad.c	VAD	B
Dtx.c	DTX Decision	B
Vad.h	Prototype and Constants	B
Dtx.h	Prototype and Constants	B
Calcexc.c	CNG Excitation Calculation	B
Dec_sid.c	Decode SID Information	B
Utilcp.c	Utility functions	B
Phdisp.c	Phase dispersion	D
Bwfw.c	Backward/forward switch selection	E
Bwfwfunc.c	Miscellaneous routines related to backward/forward switch selection	E
Filtere.c	Filter functions	E
Lpccp.c	LP analysis	E
Lspcdece.c	LSP decoding routines	E
Lspgetqe.c	LSP quantizer	E
Qua_lspe.c	LSP quantizer	E
Track_pi.c	Pitch tracking	E

**Table I.3/G.729 – List of software files integrating software files from G.729 main body and Annex B, D or E**

<b>File name</b>	<b>Description</b>	<b>Integrated from</b>
Coderi.c	Main encoder routine	B+D+E
Codld8i.c	Encoder routine	B+D+E
Decodi.c	Main decoder routine	B+D+E
Decld8i.c	Decoder routine	B+D+E
Acelpcp.c	Search ACELP fixed codebook (6.4, 8, 11.8 kbit/s)	D+E
Dacelpcp.c	Decode algebraic codebook (6.4, 8, 11.8 kbit/s)	D+E
Pitchcp.c	Pitch search	D+E
Declagcp.c	Decode adaptive-codebook index	D+E
Q_gaincp.c	Gain quantizer	D+E
Degaincp.c	Decode gain	D+E
Pstpcp.c	Postfilter routines	B+E
Bitscp.c	Bit manipulation routines	B+D+E
Tabld8cp.c	ROM tables for G.729 at 6.4 and 11.8 kbit/s	D+E
Tabld8cp.h	Extern ROM declarations for G.729 at 6.4 and 11.8 kbit/s	D+E
Ld8cp.h	Constant and Function prototypes for G.729 at 6.4 and 11.8 kbit/s	D+E

**Table I.4/G.729 – List of software files specific to integrated G.729 Annexes B, D and E**

<b>File name</b>	<b>Description</b>
Mus_dtct.c	Music detection module



## ITU-T RECOMMENDATIONS SERIES

Series A	Organization of the work of the ITU-T
Series B	Means of expression: definitions, symbols, classification
Series C	General telecommunication statistics
Series D	General tariff principles
Series E	Overall network operation, telephone service, service operation and human factors
Series F	Non-telephone telecommunication services
<b>Series G</b>	<b>Transmission systems and media, digital systems and networks</b>
Series H	Audiovisual and multimedia systems
Series I	Integrated services digital network
Series J	Transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Construction, installation and protection of cables and other elements of outside plant
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