Chapter 8

G.722: The ITU-T 64, 56, and 48 kbit/s wideband speech coding algorithm

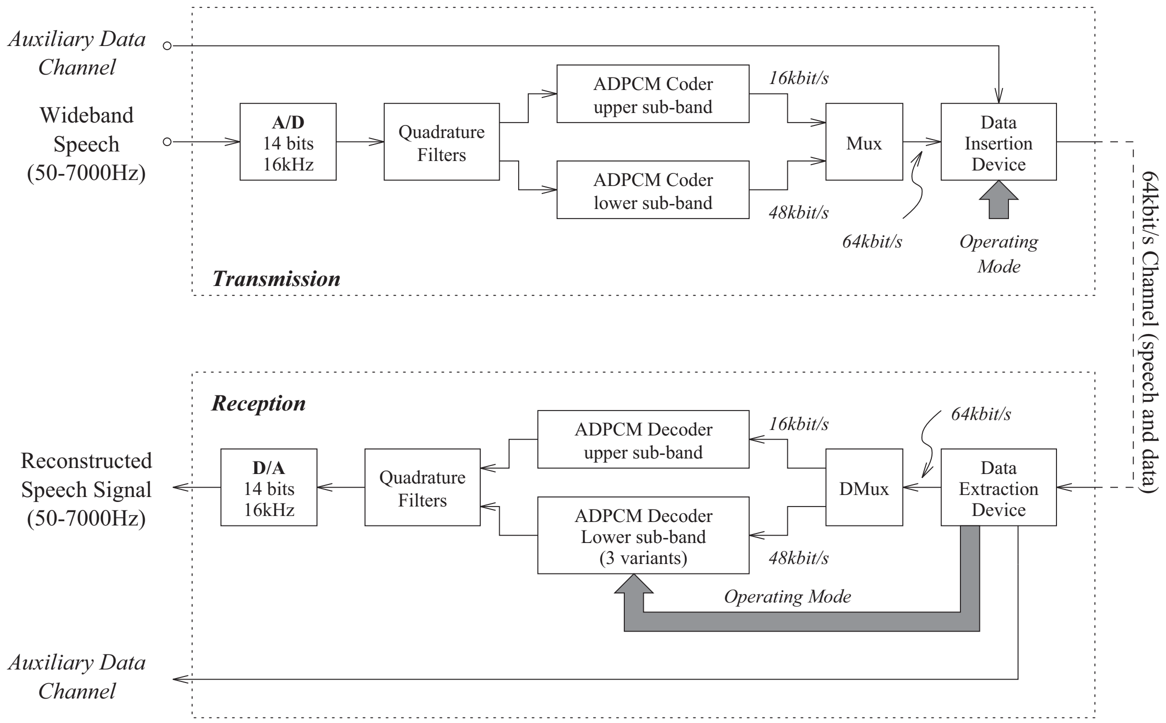
With the emergence of ISDN networks offering digital connectivity at 64 kbit/s between subscribers, the possibility was given to improve the standard telephone quality by increasing the transmitted bandwidth. A bandwidth of 50-7000 Hz corresponding to a sampling of 16 kHz was chosen because it provides a substantial improvement of the quality for applications where the speech is to be heard through high quality loudspeakers e.g. for audio or video conference services, commentary broadcasting, and high quality handsfree phones.

An expert group was created in November 1983 whose mandate was to define a standard for 7 kHz speech coding within 64 kbit/s. After many contributions received from several organisations, it has been decided to choose a coder which combined subband filtering and adaptive differential pulse-code modulation algorithms (SB-ADPCM). The final recommendation was produced in March 1986 and approved in July 1986 by the then CCITT SG XVIII as Recommendation G.722 [41].

The full description on the implementation of the G.722 algorithm is found in [41], and network aspects related to its operation are found in [42]. Figure 8.1 summarizes some systemic aspects of the G.722 algorithm. Overview and notes on the development of the G.722 algorithm can be found in several papers [43, 44, 45, 46, 47, 48]. The following description of the G.722 algorithm is based on the text in [49].

In 2006, upon ETSI DECT request, packet loss concealment (PLC) procedures for G.722 were standardized to ensure a sufficient robustness over the DECT wireless interface. ITU-T G.722 at 64 kbit/s codec is the mandatory coder in ETSI New Generation DECT (NG-DECT) standards (ETSI TS 102 527-1 and -3) [50, 51]. These standards are intended for wideband audio enabled devices to be connected on VoIP networks.

In the above mentioned PLC standardization, G.722 software tool were updated. Originally, the ITU-T G.722 algorithm used its own binary bit stream format without synchronism headers; this made almost impossible to apply frame errors. In the STL 2009, G.722 codec software tool were made compliant with G.192 bitstream format and EID-XOR/G.192 style of frame and bit error application to G.722 is enabled. At the same time, STL basic operators and complexity counters were introduced. Furthermore, the standalone decoder tool now includes basic reference Packet Loss Concealment functionality.



Notes:

* Operating modes:

Mode 1:	64kbit/s for speech and 0kbit/s for auxiliary data
Mode 1-bis:	56kbit/s for speech and 0kbit/s for auxiliary data
Mode 2:	56kbit/s for speech and 8kbit/s for auxiliary data
Mode 3:	48kbit/s for speech and 16kbit/s for auxiliary data
Mode 3-bis:	48kbit/s for speech, 6.4kbit/s for auxiliary data and 1.6kbit/s for service channel framing and mode control

* Operating modes 1-bis and 3-bis are applicable only to US national 56 kbit/s networks

* The signal in the 64kbit/s channel comprises 64, 56 or 48 kbit/s for speech and 0, 8 or 16 kbit/s for data, depending on the operating mode.

Figure 8.1: G.722 encoder and decoder block diagrams

8.1 Description of the 64, 56, and 48 kbit/s G.722 algorithm

In order to improve the transmitted speech quality, the input signal has to be converted after antialiasing filtering by an analog-to-digital (A/D) converter operating at 16 kHz sampling rate and with a resolution of at least 14 uniform PCM bits. Similarly, at the receive side, a digital-to-analog (D/A) converter operating at 16 kHz sampling rate and with a resolution of at least 14 uniform PCM bits should be used. The specifications of the transmission characteristics of the audio parts suited for the G.722 algorithm are described

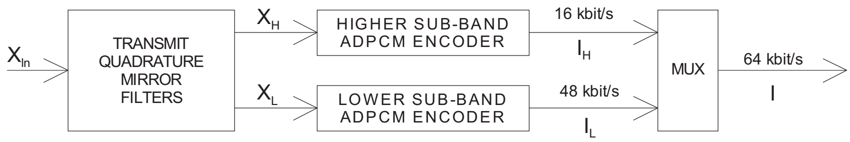


Figure 8.2: Block diagram of the SB-ADPCM encoder

in the Recommendation. Some flexibility of the output bit rate was implemented to allow the opening of an auxiliary data channel within the 64 kbit/s channel.

8.1.1 Functional description of the SB-ADPCM encoder

Figure 8.2 shows block diagram of the SB-ADPCM encoder which comprises the following main blocks.

Transmit quadrature mirror filters

The input signal X_{in} is first filtered by two quadrature mirror filters (QMF) which split the frequency band [0, 8000 Hz] into two equal subbands. The outputs X_L and X_H of the lower and higher subbands are downsampled at 8 kHz by the filtering procedure.

Lower subband ADPCM encoder

Figure 8.3 gives block diagram of the lower subband ADPCM encoder. To transmit the lower band, the encoder was designed to operate at 6, 5 or 4 bits per sample, corresponding to 48, 40 or 32 kbit/s, respectively. The ADPCM algorithm is very similar to the embedded ADPCM algorithm of ITU-T Recommendation G.727 [13]. It is an embedded ADPCM with 4 core bits and 2 additional bits. The embedded property was introduced to prevent degradation in speech quality when the encoder and the decoder operate during short intervals in different modes.

Adaptive quantizer A 60-level non-uniform adaptive quantizer is used to quantize the difference eL between the input signal X_L and the estimated signal S_L. The output of the quantizer I_L is the ADPCM codeword for the lower subband. The 4 forbidden output codewords were primarily introduced to prevent the generation of all zero codes at all modes, but have also later be used to recover the 8 kHz frame used by the coder.

Inverse adaptive quantizer In the feedback loop the two least significant bits of I_L are deleted to produce a 4 bit signal I_{Lt} which is used for the adaptation of the quantizer scale factor and applied to a 15-level inverse adaptive quantizer to produce the quantized difference signal d_{Lt}.

Quantizer adaptation In order to maintain a wide dynamic range and minimize complexity, the quantizer scale factor adaptation is performed in the base 2 logarithmic do-

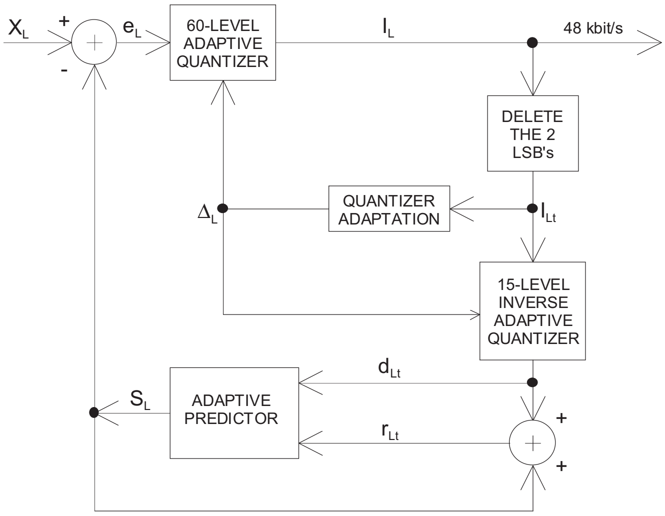


Figure 8.3: Block diagram of the lower subband ADPCM encoder

main. The log-to-linear conversion is accomplished using a lookup table. There is no adaptation of the speed control parameter as in 32 kbit/s ADPCM [6] because the encoder is designed to transmit more than voiceband data.

Adaptive predictor and reconstructed signal computation The adaptive predictor structure is similar to the one used for G.727 ADPCM standard: 2 poles and 6 zeroes. The two sets of coefficients (one for the poles and the other for the zeroes section) are updated using a simplified gradient algorithm. Stability constraints are applied to the poles in order to prevent possible unstable conditions. However, no predictor reset is applied for some specifics inputs conditions as it is done in G.726 algorithm. The reconstructed signal r_{Lt} is computed by adding the quantized difference signal d_{Lt} to the signal estimate S_L produced by the adaptive predictor. The use of a 4-bit operation instead of a 6-bit operation in the feedback loops of the lower band ADPCM encoder and decoder allows for the insertion of data in the two least significant bits without causing mistracking in the decoder.

Higher subband ADPCM encoder

Figure 8.4 shows block diagram of the higher subband ADPCM encoder. This encoder is designed to operate at 2 bits per sample, corresponding to a fixed bit rate of 16 kbit/s. The encoder algorithm is very similar to the lower band one but with the following main differences. The quantizer is a 4-level non-linear adaptive quantizer. The higher subband ADPCM encoder is not embedded, hence the inverse quantizer uses the 2 bits in the feedback loop.

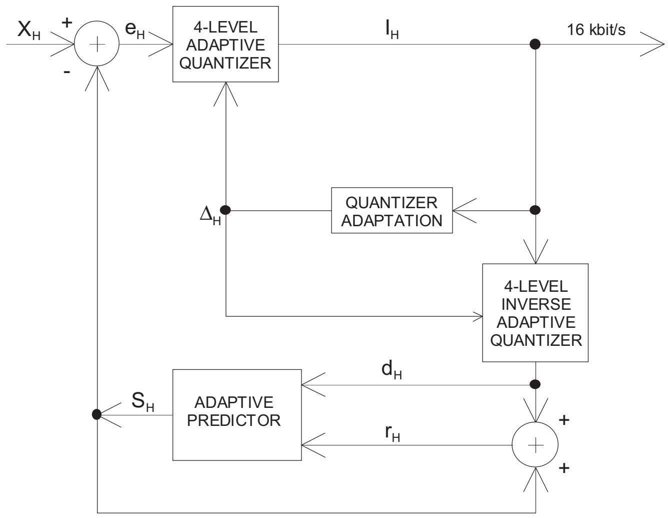


Figure 8.4: Block diagram of the higher subband ADPCM encoder

Multiplexer

The resulting codewords from the higher and lower subbands I_H and I_L are combined to obtain the output codeword I with an octet format for transmission every 8 kHz frame resulting in 64 kbit/s. Note that the 8 kHz clock may be provided by the network as it is always done for 64 kbit/s A-law or μ-law log-PCM (G.711) systems.

8.1.2 Functional description of the SB-ADPCM decoder

Figure 8.5 shows block diagram of the SB-ADPCM decoder.

Demultiplexer

The demultiplexer decomposes the received 64 kbit/s octet formatted signal I_r into two signals I_{Lr} and I_{Hr} which form the codeword inputs for the lower and higher subband ADPCM decoders, respectively.

Lower subband ADPCM decoder

Figure 8.6 shows a block diagram of the lower subband decoder. This decoder operates in three different modes depending on the received mode indication: 64, 56 and 48 kbit/s. The block which produces the estimate signal is identical to the feedback portion of the lower subband ADPCM encoder. The reconstructed signal r_L is produced by adding the signal estimate to the relevant quantized difference signals $d_{L,6}$, $d_{L,5}$ or $d_{L,4}$, which are selected according to the received indication of the mode of operation.

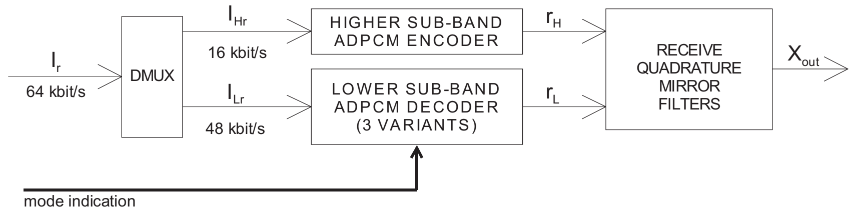


Figure 8.5: Block diagram of the SB-ADPCM decoder

Higher subband ADPCM decoder

This decoder (see Figure 8.7) is identical to the feedback portion of the higher subband ADPCM encoder described in the Section 8.1.1. Here, the output is reconstructed signal r_H.

Receive QMF

The receive QMF are two reconstruction filters which interpolate the ouputs of the lower and higher subband ADPCM decoders from 8 to 16 kHz (r_H and r_L) and generate 16 kHz sampling reconstructed output X_{out}. Signal X_{out} is converted to analog by the digital to analog converter of the receiving side.

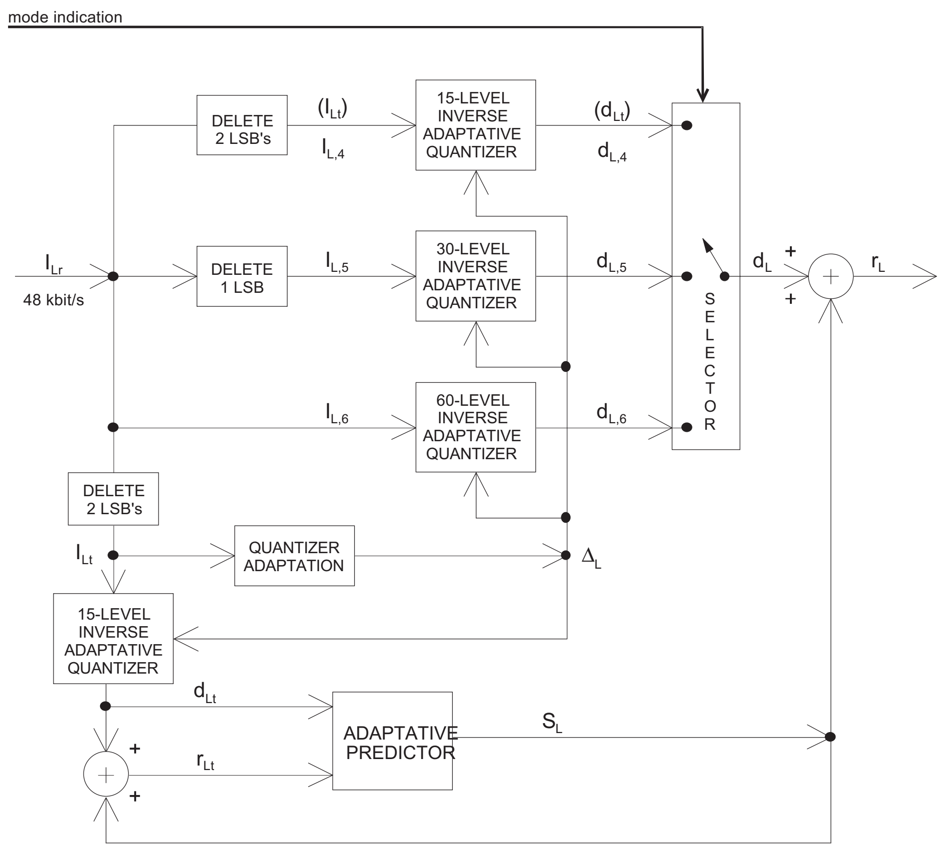


Figure 8.6: Block diagram of the lower subband ADPCM decoder

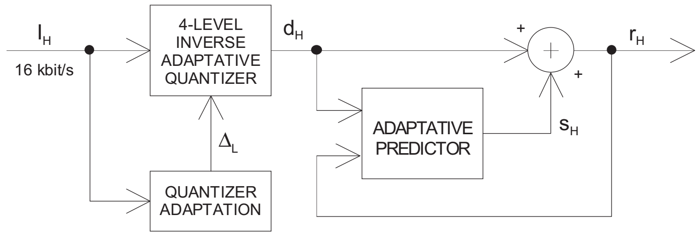


Figure 8.7: Block diagram of the higher subband ADPCM decoder

8.1.3 Functional description of the basic Packet Loss Concealment functionality

In 2006, basic index domain PLC functionality (zero index and frame repeat algorithms) for detected frame errors were included in the standalone decoder tool. The numbering and description of the four basic algorithms provided can be seen in Table 8.1.

PLC algorithm number	Description of concealment action
0 (default)	All bits in the erroneous frame are set to "1". (every 8 kHz octet index is set to 0xFF)
1	same as PLC(0), but includes a decoder reset after the frame decoding and synthesis operation.
2	The bits from the previous frame(s) are repeated. After four erroneous frames, algorithm PLC(0) is used.
3	as PLC(2), but the first (0.625 ms) bits of the first good frame after an erroneous frame are set to the value "1".

Table 8.1: Basic PLC algorithms supported by the G.722 standalone decoder.

The PLC actions in the G.722 standalone decoder are activated by incoming G.192 frames that do not have a valid G192_SYNC header tag. Note that in November 2006, ITU-T SG16 approved two new Appendices to G.722 for Packet Loss Concealment (PLC). These two Appendices (Appendices III and IV [52, 53]) provide better quality over the basic PLC functionality included in STL.

8.2 ITU-T STL G.722 Implementation

This implementation of the G.722 algorithm is composed of several source files. The interface routines are in file `g722.c`, with prototypes in `g722.h`. The original code of the STL G.722 was provided by CNET/France and its user interface was modified to be consistent with the other software modules of the STL. The update to make G.722 tool compliant with G.192 bit stream format and include basic PLC functionality was performed by Ericsson. The basic operators and complexity counters were introduced by France Telecom.

The problem of storing the state variables was solved by defining a structure called `g722_state` which containing all the necessary state variables. By means of this approach, several streams may be processed in parallel[1], provided that one structure is assigned (and that one call to the encoding/decoding routines is done) for each data stream (this can be advantageous for machines with support for parallel processing). The G.722 state structure has the following fields (which are all `shorts`):

[1] This feature was not possible with the original code provided by CNET and was added in the modifications of the user interface.

ah, al	Second-order pole section coefficient buffer for higher and lower band, respectively
bh, bl	Seventh-order zero section coefficient buffer for higher and lower band, respectively
deth, detl	Delayed quantizer scale factor for higher and lower band, respectively
dh	Quantizer difference signal memory
dlt	Quantizer difference signal for the adaptive predictor
init_qmf_rx	Flag indicating the need to initialize the QMF filters on the reception (decoder) side
init_qmf_tx	Flag indicating the need to initialize the QMF filters on the transmission (encoder) side
nbh, nbl	Delayed logarithmic quantizer factor for higher and lower band, respectively
ph, plt	Partially reconstructed signal memory for higher and lower band, respectively
qmf_rx_delayx	Memory of past 24 received (decoded) samples
qmf_tx_delayx	Memory of past 24 transmitted (encoded) samples
rh[3]	Quantized reconstructed signal
rlt[3]	Reconstructed signal memory for the adaptive predictor
sh, sl	Predictor output value for higher and lower band, respectively
sph, spl	Pole section output signal for higher and lower band, respectively
szh, szl	Zero section output signal for higher and lower band, respectively

The default bitstream generated by the STL G.722 encoder is a G.192 compatible 16 bit output file and it is provided using the frame size specified by the -fsize command. Bits in a G.192 bitstream frame are ordered so that the frame can be truncated at 56 and 48 kbps. If the number of input samples are non divisible by the frame size, the output codeword stream is truncated at the last frame boundary.

A legacy g722 octet stream can be obtained using the option -byte. The legacy bitstream generated by the STL G.722 encoder has 8 valid bits for each encoded sample, saved in right-justified `shorts`. i.e., the codewords are located in the lower 8-bits of the encoded bitstream file. The MSB is always 0 for the 16 bit bitstream file. The lower 6 bits are the lower-subband encoded bits, and the upper two bits of the 8 valid bits are the upper-subband encoded bits. When the decoder is not in operation mode 1, the decoder will discard 1 or 2 of the lower bits of the lower-subband. It should be noted that, when bit errors are inserted in this bitstream and the operation mode is not mode 1, the actual bit error rate seen by the decoder may not be the one actually desired. One may consider that, in simulating a system where auxiliary data channels are used, such as modes 2 and 3, this is actually the desired behaviour, because errors hitting the auxiliary data will not affect the decoded speech quality. However,if simulation of modes 1-bis or 3-bis is intended, then the some of the errors hitting the lower 1 (mode 1-bis) or 2 bits (mode 3-bis) will not be seen by the decoder, and the overall bit error rate will actually be smaller than the desired one. There are two possible approaches to circumvent this problem:

- the use of an external program to shift the bitstream samples one or two bits (respectively for modes 1-bis or 3-bis) to the right before the bitstream serialization process for use with the STL EID module, and an external program to left-shift the bitream samples by one or two bits after error insertion and before using the STL G.722 decoder. This solution is valid for both random and burst bit errors.

- to increase proportionally the bit error rate by 1/8 (mode 1-bis) or 1/4 (mode 3-bis), to statistically compensate for errors hitting unused bits. This solution is valid only for random bit errors.

From the users' perspective, the encoding function is `g722_encode`, and the decoding function is `g722_decode`. Before using these functions, state variables for the encoder and the decoder must be initialized respectively by `g722_reset_encoder` and `g722_reset_decoder`. It should be noted that encoder and decoder need individual state variables to work properly.

In the following part a summary of calls to the three entry functions is found.

8.2.1 g722_encode

Syntax:

```
#include "g722.h"
long g722_encode (short *inp_buf, short *g722_frame, long smpno,
                  g722_state *g722_encode);
```

Prototype: g722.h

Description:

Simulation of the ITU-T G.722 64 kbit/s encoder. Takes the linear (16-bit, left-justified) input array of shorts *inp_buf* (16 bit, right-justified, without sign extension) with *smpno* samples, and saves the encoded bit-stream in the array of shorts *g722_frame*.

The state variables are saved in the structure pointed by *g722_encode*, and the reset can be established by making a call to `g722_reset_encoder`.

Variables:

inp_buf .... Is the input samples' buffer with *smpno* left-justified 16-bit linear `short` speech samples.

g722_frame .... Is the encoded samples' buffer; each `short` sample will contain the encoded parameters as right-justified 8-bit samples.

smpno .... Is a long with the number of samples to be encoded from the input buffer *inp_buf*.

g722_encode.............. A pointer to the state variable structure; all the variables here are for internal use of the G.722 algorithm, and should not be changed by the user. Fields of this structure are described above.

Return value:

Returns the number of speech samples encoded.

8.2.2 g722_decode

Syntax:

```
#include "g722.h"
short g722_decode (short *g722_frame, short *out_buf, int mode, long
                    smpno, g722_state *g722_decoder, );
```

Prototype: g722.h

Description:

Simulation of the ITU-T 64 kbit/s G.722 decoder. Reconstructs a linear (16-bit, left-justified) array of shorts *inp_buf* (16 bit, right-justified, without sign extension) with *smpno* samples from the encoded bit-stream in the array of shorts *g722_frame*. Include a basic Packet Loss Concealment functionality.

The state variables are saved in the structure pointed by *g722_decoder*, and the reset can be established by making a call to *g722_reset_decoder*.

Variables:

g722_frame .... Is the encoded samples' buffer; each `short` sample will contain the encoded parameters as right-justified 8-bit samples.

out_buf .... Is the output samples' buffer with *smpno* left-justified 16-bit linear `short` speech samples.

mode .... Is an `int` which indicates the operation mode for the G.722 decoder. If equal to 1, the decoder will operate at 64 kbit/s. If equal to 2, the decoder will operate at 56 kbit/s, discarding the least significant bit of the lower-band ADPCM. If equal to 3, the decoder will discard the two least significant bits of the lower band ADPCM, being equivalent to the 48 kbit/s operation of the G.722 algorithm. It should be noted that, for this implementation of the G.722 algorithm, mode 1-bis is identical to mode 2, and mode 3-bis is identical to mode 3.

smpno .... Is a long with the number of samples in the input encoded sample buffer *g722_frame* to be decoded.

g722_decoder............. A pointer to the state variable structure; all the variables here are for internal use of the G.722 algorithm, and should not be changed by the user. Fields of this structure are described above.

Return value:

Returns the number of speech samples encoded.

8.2.3 g722_reset_encoder

Syntax:

```
#include "g722.h"
void g722_reset_encoder (g722_state *g722_encoder);
```

Prototype: g722.h

Description:

Initializes the state variables for the G.722 encoder or decoder. Coder and decoder require each a different state variable.

Variables:

g722_encoder . . . . A pointer to the G.722 encoder state variable structure which is to be initialized.

Return value: None.

8.2.4 g722_reset_decoder

Syntax:

```
#include "g722.h"
void g722_reset_decoder (g722_state *g722_decoder);
```

Prototype: g722.h

Description:

Initializes the state variables for the G.722 decoder. Coder and decoder require each a different state variable.

Variables:

g722_decoder . . . . A pointer to the G.722 decoder state variable structure which is to be initialized.

Return value: None.

8.3 Standalone G.192 compatible G.722 encoder tool and standalone G.192 compatible G.722 decoder tool

8.3.1 G.192 bit stream format for standalone G.722 encoder and decoder

As described previously, G.722 is an embedded algorithm which supports network scaling of the bitstream in three bitrates: 48, 56 and 64 kbit/s. To support this functionality within the G.192 file format the G.722 encoder writes the 16 kbit/s embedded bits in the end of every G.192 bitstream frame.

For every 16 kHz sample, there are eight G.722 encoded bits (an octet index) to be transported in the G.192 frame. Bits b1 and b0 are the embedded bits and bits b2-b7 are the non-embedded bits. In a G.192 frame, the b2 bits are written first in chunk, followed by that of b3, until b7 bits. After all b7 bits are written, b1 bits are written and then b0 bits. By sorting bits in this manner, truncation of a G.192 type G.722 frame bitstream from 64 kbit/s to 48 or 56 kbit/s is made easy. Figure 8.8, Figure 8.9 and Figure 8.10 below show how the G.192 frames are composed for various bit rates (64, 56, 48 kbit/s) and example frame sizes (10, 5, 20 ms). G.192 format is useful for enhanced simulation of G.722, however actual application transport formats may be different (e.g. as defined in [41] and as used in [54]).

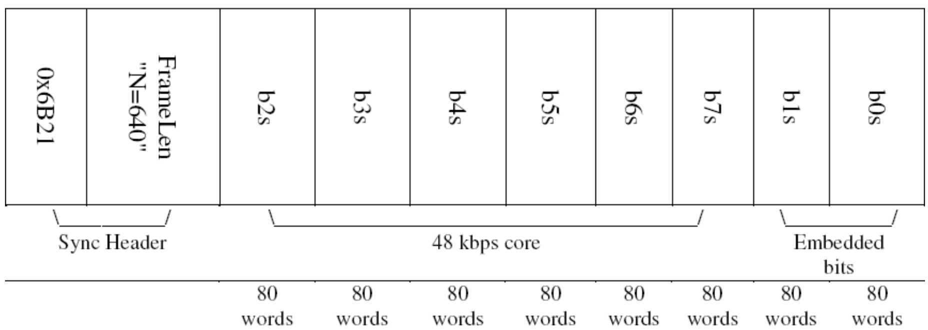


Figure 8.8: Example of a G.192 compatible G.722 encoder output frame for a 10 ms input frame size. This frame has a length of 640 bits and a G192_SYNC (0x6B21) header tag, in the G.192 file each g722 bit is stored as a 16 bit word.

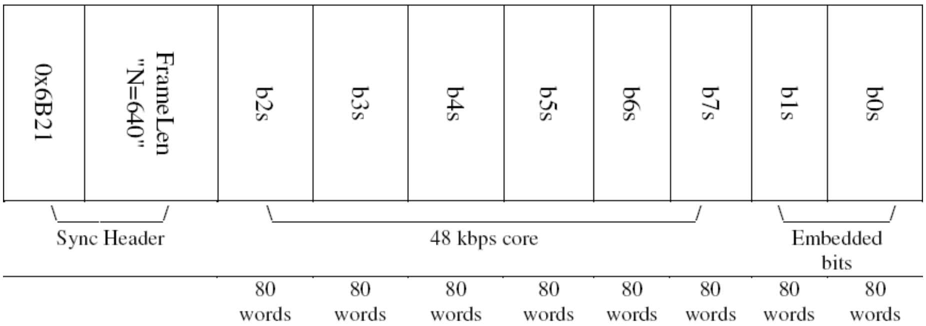


Figure 8.9: Example of a G.192 compatible G.722 56 kbit/s encoder output frame for a 5 ms input frame size. This frame has a length of 280 bits and a G192_SYNC (0x6B21) header tag.

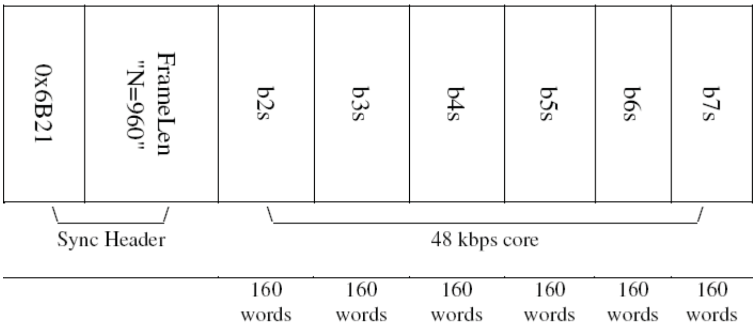


Figure 8.10: Example of G.192 compatible G.722 48 kbit/s encoder output frame for a 20 ms input frame size. This frame has a length of 960 bits and a G192_SYNC (0x6B21) header tag.

8.3.2 Standalone G.722 Encoder specific operation

If the encoder can not read a complete input frame it stops its processing, this means that the final decoder output file may be up to a frame shorter than the input speech file.

8.4 Portability and compliance

The portability test for these routines has been performed using the test sequences designed by the ITU-T for the G.722 algorithm[2]. It should be noted that the G.722 test sequences are not designed to test the QMF filters, but only to exercise the upper and lower band encoder and decoder ADPCM algorithms. Therefore, testing of the codec with the test sequences was done with a special set of test programs that used the core G.722 upper- and lower-band ADPCM coding and decoding functions. All test sequences were correctly processed.

This module has been compiled and tested on PC platform with Cygwin (CYGWIN_NT-5.0), using gcc(3.3.3), and with Microsoft Visual Studio 8.

Please note that the 16 bit oriented G.192 files require correct octet swapping of inputs and outputs on big/little-endian machines.

8.5 Encoder(encg722) tool command line options

```
Usage:
encg722  [-options] InpFile OutFile
where:
InpFile     is the name of the speech file to be processed;
OutFile     is the name with the processed bitstream;

options:
-fsize  #   Number of 16 kHz input samples per frame (must be an even number).
            Default is 160 samples (16 kHz) (10 ms)
-mode   #   Operating mode (1,2,3) (or rate 64, 56, 48 in kbps).
            Default is mode 1 (= 64 kbps)
-frames #   number of frames to process
            (values -1 or 0 processes the whole file )
-byte       Provide encoder output data in legacy octet format.
            (default is g192).
-h/-help    print help message
```

8.6 Decoder(decg722) tool command line options

```
Usage:

decg722 [-options] InpFile OutFile

where:
InpFile     is the name of the bit stream input file;
```

[2] The G.722 test sequences are freely downloadable from http://www.itu.int/rec/T-REC-G.722-198703-I!AppII/en.

```
OutFile      is the name of the file with synthesized speech;

options:

-mode #      is the operation mode for the G.722 decoder.
             Default is mode 1/64 kbit/s.
             (others are mode 2/56 kbps and mode 3/48 kbps)

-fsize #     Number of samples per frame
             Default is 160, 16 kHz samples (or 10ms)
             (NB! must be the same as on the encoder side.)

-frames #    Number of frames to process

-plc #       Packet Loss Concealment algorithm number
             (0 = zero index insertion, no decoder state reset)
             (1 = zero index insertion, decoder state reset)
             (2 = previous frame repetition, no decoder state reset)
             (3 = previous frame repetition, a few zero_indeces in first good
              frame, no decoder state reset)

-byte        Use legacy nonG192 G.722 format (byte oriented) without
             frame/synch headers.

-h/-help     print help message
```

8.7 Example code

8.7.1 Description of the demonstration programs

One demonstration program is provided for the G.722 module, g722demo.c. In addition, two programs are provided in the distribution when compliance testing of the encoder and decoder is necessary, tstcg722.c and tstdg722.c[3].

Program `g722demo.c` accepts 16-bit, linear PCM samples sampled at 16 kHz as encoder input. The decoder also produces files in the same format. The bitstream signals out of the encoder are always organized in 16-bit, right-justified words that use the lower 8 bits (i.e., 64 kbit/s). According to the user-specified mode, the decoder will decode the G.722-encoded bitstream using 64, 56, or 48 kbit/s (i.e. full 8 bits, discard 1 bit of the lower band, or discard 2 bits of the lower band). It should be noted that the demonstration programs g722demo.c, tstcg722.c and tstdg722.c only produce G.722 legacy bitstream. To produce bitstream compliant with G.192, please use encg722.c and decg722.c. Similarly, basic PLC functionnality is not supported the demonstration programs g722demo.cand tstdg722.c. Please use decg722.c for basic PLC functionnality.

[3]. The demonstration program g722demo.c cannot be used for compliance verification because the test vectors for G.722 do not foresee processing through the quadrature mirror filters.

8.7.2 Simple example

The following C code gives an example of G.722 coding and decoding using as input wideband speech which is encoded and decoded at either 64, 56, or 48 kbit/s, according to the user-specified parameter *mode*.

```
#include <stdio.h>
#include "ugstdemo.h"
#include "g722.h"
#define BLK_LEN 256

void main(argc, argv)
  int             argc;
  char           *argv[];
{
  g722_state      encoder_state, decoder_state;
  int             mode;
  char            FileIn[180], FileOut[180];
  short           smpno, tmp_buf[BLK_LEN], inp_buf[BLK_LEN], out_buf[BLK_LEN];
  FILE           *Fi, *Fo;

  /* Get parameters for processing */
  GET_PAR_S(1, "_Input File: .................. ", FileIn);
  GET_PAR_S(2, "_Output File: ................. ", FileOut);
  GET_PAR_I(3, "_Mode: ........................ ", mode);

  /* Initialize state structures */
  g722_reset_encoder(&encoder_state);
  g722_reset_decoder(&decoder_state);

  /* Opening input and output 16-bit linear PCM speech files */
  Fi = fopen(FileIn, RB);
  Fo = fopen(FileOut, WB);

 /* File processing */
  while (fread(inp_buf, BLK_LEN, sizeof(short), Fi) == BLK_LEN)
  {
    /* Encode input samples in blocks of length BLK_LEN */
    smpno = g722_encode(inp_buf, tmp_buf, BLK_LEN, &encoder_state);

    /* Decode G.722-coded samples in blocks of length BLK_LEN */
    smpno = g722_decode(tmp_buf, out_buf, mode, smpno, &decoder_state);

    /* Write 16-bit linear PCM output decoded samples */
    fwrite(out_buf, smpno, sizeof(short), Fo);
  }

  /* Close input and output files */
  fclose(Fi); fclose(Fo);
}
```

8.7.3 Example operation of encoder (encg722)

```
foreach mode ( 1 2 3 ) # (corresponds to 64,56,48 kbps)
   foreach fsize ( 160 320 640) # (corresponds to 10,20,40 ms)
      encg722  -mode $mode -fsize $fsize inpsp.smp outp.g722.$fsize.$mode.g192
   end #fsize
end #mode
```

8.7.4 Example operation of decoder (decg722)

```
foreach mode ( 1 2 3 ) # (corresponds to 64,56,48 kbps)
   foreach fsize ( 160 320 640) # (corresponds to 10,20,40 ms)
      decg722 -plc 0 -fsize $fsize outp.g722.$fsize.$mode.g192  outsp.smp
      mv outpsp.smp outpsp.g722.$fsize.$mode.plc.0.smp
   end #fsize
end #mode
```