available bits "cb", the number of bits needed for the header "bhdr" (32 bits), the CRC checkword "bcrc" if used (16 bits), the bit allocation "bbal", and the number of bits "banc" required for ancillary data:

adb = cb - (bhdr + bcrc + bbal + banc)

The resulting number can be used to code the subband samples and the scalefactors. The principle used in the allocation procedure is minimization of the total noise-to-mask ratio over the frame with the constraint that the number of bits used does not exceed the number of bits available for that frame. Use is made of table B.2, "Layer II Possible Quantization per subband" that indicates for every subband the number of steps that may be used to quantize the samples. The number of bits required to represent these quantized samples can be derived from table B.4, "Layer II Classes of Quantization".

The allocation procedure is an iterative procedure where, in each iteration step the number of levels of the subband that has the greatest benefit is increased.

First the mask-to-noise ratio "MNR" for each subband is calculated by subtracting from the signal-to-noise-ratio "SNR" the signal-to-mask-ratio "SMR":

MNR = SNR - SMR

The signal-to-noise-ratio can be found in table C.5 "Layer II Signal-to-Noise Ratios". The signal-to-mask-ratio is the output of the psychoacoustic model.

Then zero bits are allocated to the samples and the scalefactors. The number of bits for the samples "bspl" and the number of bits for the scalefactors "bscf" are set to zero. Next an iterative procedure is started. Each iteration loop contains the following steps :

- Determination of the minimal MNR of all subbands.
- The accuracy of the quantization of the subband with the minimal MNR is increased by using the next higher entry in the relevant table B.2, "Layer II Possible Quantization per Subband".
- The new MNR of this subband is calculated.
- bspl is updated according to the additional number of bits required. If a non-zero number of bits is assigned to a subband for the first time, bsel has to be updated, and bscf has to be updated according to the number of scalefactors required for this subband. Then adb is calculated again using the formula :

  adb = cb - (bhdr + bcrc + bbal + bsel + bscf + bspl + banc)

The iterative procedure is repeated as long as adb is not less than any possible increase of bspl, bsel and bscf within one loop.

C.1.5.2.8 Quantization and encoding of subband samples

Each of the 12 subband samples is normalized by dividing its value by the scalefactor to obtain X and quantized using the following formula:

- Calculate A * X + B
- Take the N most significant bits.
- Invert the MSB

A and B can be found in the table C.6, "Layer II Quantization Coefficients". N represents the necessary number of bits to encode the number of steps. The inversion of the MSB is done in order to avoid the all '1' code that is used for the synchronization word.

Given the number of steps that the samples will be quantized to, table B.4, "Layer II Classes of Quantization" shows whether grouping will be used. If grouping is not required, the three samples are coded with individual codewords.

If grouping is required, three consecutive samples are coded as one codeword. Only one value v_m, MSB first, is transmitted for this triplet. The relationships between the coded value v_m (m=3,5,9) and the three consecutive subband samples x, y, z are:

$$v_3 = 9z + 3y + x \qquad (v_3 \text{ in } 0\ldots 26)$$
$$v_5 = 25z + 5y + x \qquad (v_5 \text{ in } 0\ldots 124)$$
$$v_9 = 81z + 9y + x \qquad (v_9 \text{ in } 0\ldots 728)$$

C.1.5.2.9 Coding of bit allocation

For the purpose of a more efficient coding, only a limited number of possible quantizations, which may be different for each subband, are allowed. Only the index with wordlength "nbal" in the relevant table B.2, "Layer II Possible Quantizations per Subband" is transmitted, MSB first.

C.1.5.2.10 Ancillary data

The Audio standard provides a number of bits for the inclusion and transmission of variable length ancillary data with the audio bitstream. The ancillary data will reduce the number of bits available for audio, which may result in a degradation of audio quality.

The presence of a bit pattern in the ancillary data matching the syncword may hamper synchronization. This problem is more likely to occur when the free format is used.

C.1.5.2.11 Formatting

An overview of the Layer II format can be seen in figure C.3.

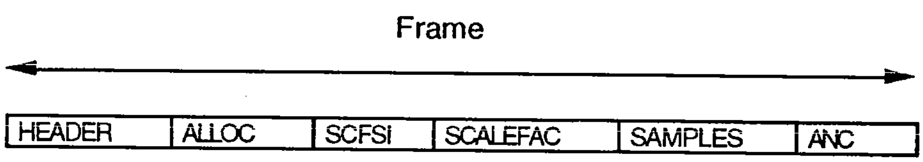

Figure C.3 -- Layer II Format

The differences compared to the Layer I format are:

- The length of a slot equals 8 bits.
- A new block scfsi containing the scalefactor selection information has been introduced.
- The bit allocation information, scalefactors and samples have been subject to further coding (see the relateds).

The details can be found in 2.4.1.

Table C.4 -- Layer II scalefactor transmission patterns

$Class_1$	$Class_2$	Scalefactors used in encoder	Transmission pattern	Selection Information
1	1	1 2 3	1 2 3	0
1	2	1 2 2	1 2	3
1	3	1 2 2	1 2	3
1	4	1 3 3	1 3	3
1	5	1 2 3	1 2 3	0
2	1	1 1 3	1 3	1
2	2	1 1 1	1	2
2	3	1 1 1	1	2
2	4	4 4 4	4	2
2	5	1 1 3	1 3	1
3	1	1 1 1	1	2
3	2	1 1 1	1	2
3	3	1 1 1	1	2
3	4	3 3 3	3	2
3	5	1 1 3	1 3	1
4	1	2 2 2	2	2
4	2	2 2 2	2	2
4	3	2 2 2	2	2
4	4	3 3 3	3	2
4	5	1 2 3	1 2 3	0
5	1	1 2 3	1 2 3	0
5	2	1 2 2	1 2	3
5	3	1 2 2	1 2	3
5	4	1 3 3	1 3	3
5	5	1 2 3	1 2 3	0

Table C.5 -- Layer II Signal-to-Noise Ratios

No. of steps	SNR (dB)
0	0,00
3	7,00
5	11,00
7	16,00
9	20,84
15	25,28
31	31,59
63	37,75
127	43,84
255	49,89
511	55,93
1 023	61,96
2 047	67,98
4 095	74,01
8 191	80,03
16 383	86,05
32 767	92,01
65 535	98,01

Table C.6 -- Layer II quantization coefficients

No. of steps	A	B
3	0,750000000	-0,250000000
5	0,625000000	-0,375000000
7	0,875000000	-0,125000000
9	0,562500000	-0,437500000
15	0,937500000	-0,062500000
31	0,968750000	-0,031250000
63	0,984375000	-0,015625000
127	0,992187500	-0,007812500
255	0,996093750	-0,003906250
511	0,998046875	-0,001953125
1 023	0,999023438	-0,000976563
2 047	0,999511719	-0,000488281
4 095	0,999755859	-0,000244141
8 191	0,999877930	-0,000122070
16 383	0,999938965	-0,000061035
32 767	0,999969482	-0,000030518
65 535	0,999984741	-0,000015259

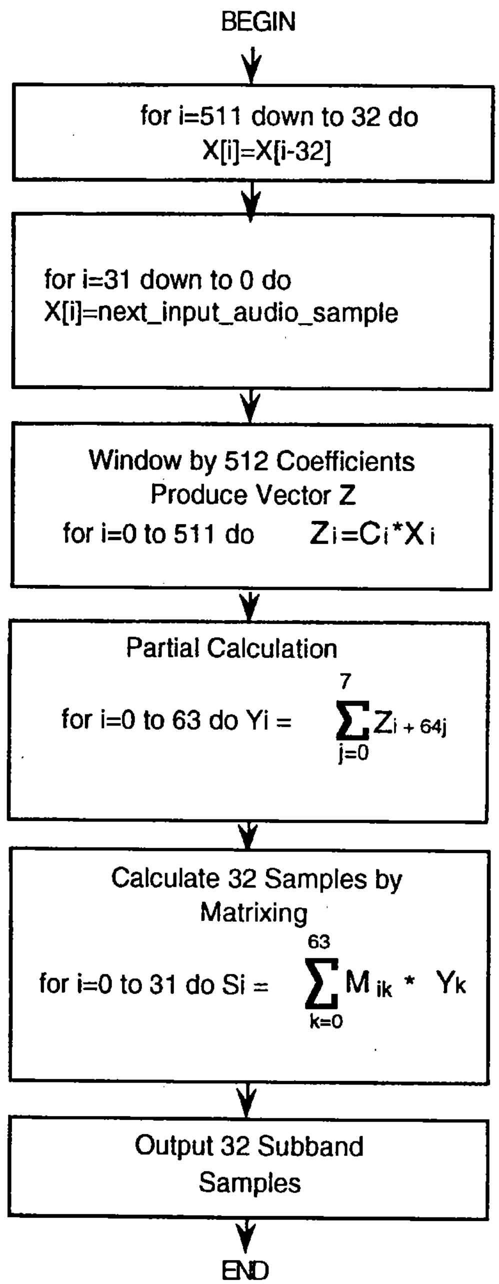

Figure C.4 -- Analysis subband filter flow chart

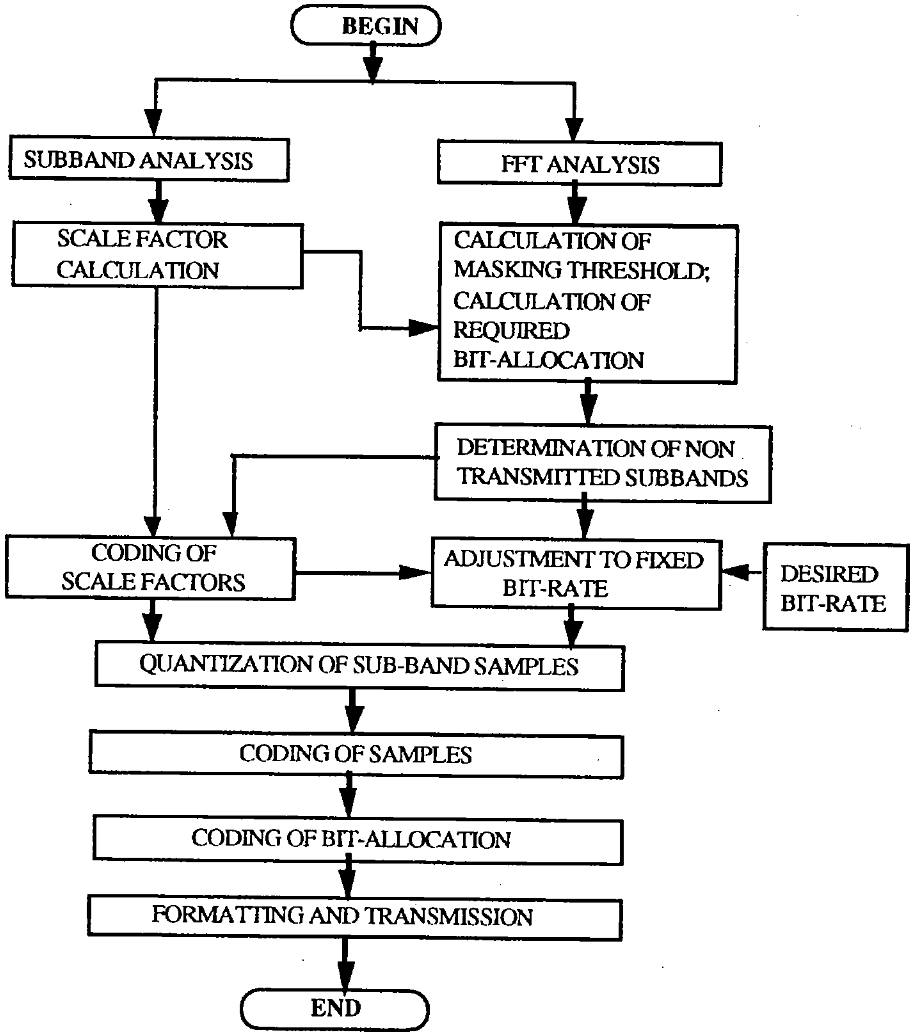

Figure C.5 -- Layer I, II Encoder flow chart

C.1.5.3 Layer III encoding

C.1.5.3.1 Introduction

This clause describes a possible Layer III encoding method. The basic data flow is described by the general psychoacoustic coder block diagram. The basic blocks are described in more detail and below.

C.1.5.3.2 Psychoacoustic model

The calculation of the psychoacoustic parameters can be done either with Psychoacoustic Model I described in clause D.1. or with Psychoacoustic Model 2 described in clause D.2. A description of modifications to Psychoacoustic Model 2 for use with Layer III can be found below. The model is run twice per block, using a shift length of 576 samples. A signal-to-mask-ratio is provided for every scalefactor band.

C.1.5.3.2.1 Adaptation of psychoacoustic model II for Layer III

Psychoacoustic Model 2 (clause D.2) is modified as described below for the use with Layer III encoding.

General considerations:

The model is calculated twice in parallel. One computation is done with a shift length **iblen** of 192 samples (to be used with short blocks), the other is done with a shift length of 576 samples. For the shift length of 192 samples the block length of the FFT is changed to 256, and the parameters changed accordingly.

Change to unpredictability calculation:

The calculation of the unpredictability metric in Psychoacoustic Model 2 is changed.

- Calculation of the unpredictability:
  The unpredictability cw is calculated for the first 206 spectral lines. For the other spectral lines, the unpredictability is set to 0,4.
  The unpredictability for the first 6 lines is calculated from the long FFT (window length = 1024, shiftlen = 576). For the spectral lines 6 up to 205, the unpredictability is calculated from the short FFT (window length 256, shiftlen = 192):

$$cw(w) = \begin{cases} cw\_l(w) & \text{for} \quad 0 \leq w < 6 \\ cw\_s((w+2)\,\mathrm{DIV}\,4) & \text{for} \quad 6 \leq w < 206 \\ 0{,}4 & \text{for} \quad w \geq 206 \end{cases}$$

  cw_l is the unpredictability calculated from the long FFT, cw_s is the unpredictability calculated from the second short block out of three short blocks within one granule.

- The spreading function has been replaced:

  If $j \geq i$ tmpy= 3,0 (j -i)
  else tmpy= 1,5(j-i) is used.

  Only values of the spreading function greater than 10^{-6} are used. All other values are set to zero.

- For converting the unpredictability the parameters

  conv1 = -0,299
  conv2 = - 0,43

  are used.

- The parameter NMT (noise masking tone) is set to 6,0 dB for all threshold calculation partions. The parameter TMN (tone masking noise) is set to 29,0 dB for all partitions.
  For minval see table "threshold calculation partitions" (table C.7).

- The psychoacoustic entropy is estimated from the ratio thr/eb, where thr is the threshold and eb is the energy:

  $pe = - \sum (cbwidth_k \cdot \log(thr_k/(eb_k+1.)))$

  where k indexes the threshold calculation partitions and cbwidth is the width of the threshold calculation partition (see tables).

- pre-echo control

  The following constants are used for the control of pre-echo's (see block diagram):
  rpelev = 2
  rpelev2 = 16

- The threshold is not spread over the FFT lines. The threshold calculation partitions are converted directly to scalefactor bands. The first partition which is added to the scalefactor band is weighted with w1, the last with w2 (see table C.8 "Converting Threshold Calculation Partitions to Scalefactor Bands"). The table contains also the number of partitions (cbw) converted to one scalefactor band (excluding the first and the last partition).

  The parameters bo and bu are shown in table C.8. They are used for converting threshold calculation partitions to scalefactor bands.

- For short blocks a simplified version of the threshold calculation (constant signal to noise ratio) is used. The constants can be found in the columns labelled "SNR (dB)" in table C.7(def) below.

Table C.7 -- Threshold calculation partitions with following parameters width, minval, threshold in quiet, norm and bval:

Table C.7.a -- Sampling_frequency = 48 kHz long blocks

no.	FFT-lines	minval	qthr	norm	bval
0	1	24,5	4,532	0,970	0,000
1	1	24,5	4,532	0,755	0,469
2	1	24,5	4,532	0,738	0,937
3	1	24,5	0,904	0,730	1,406
4	1	24,5	0,904	0,724	1,875
5	1	20	0,090	0,723	2,344
6	1	20	0,090	0,723	2,812
7	1	20	0,029	0,723	3,281
8	1	20	0,029	0,718	3,750
9	1	20	0,009	0,690	4,199
10	1	20	0,009	0,660	4,625
11	1	18	0,009	0,641	5,047
12	1	18	0,009	0,600	5,437
13	1	18	0,009	0,584	5,828
14	1	12	0,009	0,531	6,187
15	1	12	0,009	0,537	6,522
16	2	6	0,018	0,857	7,174
17	2	6	0,018	0,858	7,800
18	2	3	0,018	0,853	8,402
19	2	3	0,018	0,824	8,966
20	2	3	0,018	0,778	9,483
21	2	3	0,018	0,740	9,966
22	2	0	0,018	0,709	10,426
23	2	0	0,018	0,676	10,866
24	2	0	0,018	0,632	11,279
25	2	0	0,018	0,592	11,669
26	2	0	0,018	0,553	12,042
27	2	0	0,018	0,510	12,386
28	2	0	0,018	0,513	12,721
29	3	0	0,027	0,608	13,115
30	3	0	0,027	0,673	13,561
31	3	0	0,027	0,636	13,983
32	3	0	0,027	0,586	14,371
33	3	0	0,027	0,571	14,741
34	4	0	0,036	0,616	15,140
35	4	0	0,036	0,640	15,562
36	4	0	0,036	0,597	15,962
37	4	0	0,036	0,538	16,324
38	4	0	0,036	0,512	16,665
39	5	0	0,045	0,528	17,020
40	5	0	0,045	0,516	17,373
41	5	0	0,045	0,493	17,708
42	6	0	0,054	0,499	18,045
43	7	0	0,063	0,525	18,398
44	7	0	0,063	0,541	18,762
45	8	0	0,072	0,528	19,120
46	8	0	0,072	0,510	19,466
47	8	0	0,072	0,506	19,807
48	10	0	0,180	0,525	20,159
49	10	0	0,180	0,536	20,522
50	10	0	0,180	0,518	20,873
51	13	0	0,372	0,501	21,214
52	13	0	0,372	0,496	21,553
53	14	0	0,400	0,497	21,892
54	18	0	1,628	0,495	22,231
55	18	0	1,628	0,494	22,569
56	20	0	1,808	0,497	22,909
57	25	0	22,607	0,494	23,248
58	25	0	22,607	0,487	23,583
59	35	0	31,650	0,483	23,915
60	67	0	605,867	0,482	24,246
61	67	0	605,867	0,524	24,576

Table C.7.b Sampling_frequency = 44,1 kHz long blocks

no.	FFT-lines	minval	qthr	norm	bval
0	1	24,5	4,532	0,951	0,000
1	1	24,5	4,532	0,700	0,431
2	1	24,5	4,532	0,681	0,861
3	1	24,5	0,904	0,675	1,292
4	1	24,5	0,904	0,667	1,723
5	1	20	0,090	0,665	2,153
6	1	20	0,090	0,664	2,584
7	1	20	0,029	0,664	3,015
8	1	20	0,029	0,664	3,445
9	1	20	0,029	0,655	3,876
10	1	20	0,009	0,616	4,279
11	1	20	0,009	0,597	4,670
12	1	18	0,009	0,578	5,057
13	1	18	0,009	0,541	5,415
14	1	18	0,009	0,575	5,774
15	2	12	0,018	0,856	6,422
16	2	6	0,018	0,846	7,026
17	2	6	0,018	0,840	7,609
18	2	3	0,018	0,822	8,168
19	2	3	0,018	0,800	8,710
20	2	3	0,018	0,753	9,207
21	2	3	0,018	0,704	9,662
22	2	0	0,018	0,674	10,099
23	2	0	0,018	0,640	10,515
24	2	0	0,018	0,609	10,917
25	2	0	0,018	0,566	11,293
26	2	0	0,018	0,535	11,652
27	2	0	0,018	0,531	11,997
28	3	0	0,027	0,615	12,394
29	3	0	0,027	0,686	12,850
30	3	0	0,027	0,650	13,277
31	3	0	0,027	0,611	13,681
32	3	0	0,027	0,567	14,062
33	3	0	0,027	0,520	14,411
34	3	0	0,027	0,513	14,751
35	4	0	0,036	0.557	15,119
36	4	0	0,036	0,584	15,508
37	4	0	0,036	0,570	15,883
38	5	0	0,045	0,579	16,263
39	5	0	0,045	0,585	16,654
40	5	0	0,045	0,548	17,020
41	6	0	0,054	0,536	17,374
42	6	0	0,054	0,550	17,744
43	7	0	0,063	0,532	18,104
44	7	0	0,063	0,504	18,447
45	7	0	0,063	0,496	18,781
46	9	0	0,081	0,516	19,130
47	9	0	0,081	0,527	19,487
48	9	0	0,081	0,516	19,838
49	10	0	0,180	0,497	20,179
50	10	0	0,180	0,489	20,510
51	11	0	0,198	0,502	20,852
52	14	0	0,400	0,502	21,196
53	14	0	0,400	0,491	21,531
54	15	0	0,429	0,497	21,870
55	20	0	1.808	0.504	22,214
56	20	0	1,808	0,504	22,558
57	21	0	1,899	0,495	22,898
58	27	0	24,415	0,486	23,232
59	27	0	24,415	0,484	23,564
60	36	0	32,554	0,483	23,897
61	73	0	660,124	0,475	24,229
62	18	0	162,770	0,515	24,542

Table C.7.c -- Sampling_frequency = 32 kHz long blocks

no.	FFT-lines	minval	qthr	norm	bval
0	2	24,5	9,064	0,997	0,312
1	2	24,5	9,064	0,893	0,937
2	2	24,5	1,808	0,881	1,562
3	2	20	0,181	0,873	2,187
4	2	20	0,181	0,872	2,812
5	2	20	0,057	0,871	3,437
6	2	20	0,018	0,860	4,045
7	2	20	0,018	0,839	4,625
8	2	18	0,018	0,812	5,173
9	2	18	0,018	0,784	5,698
10	2	12	0,018	0,741	6,184
11	2	12	0,018	0,697	6,634
12	2	6	0,018	0,674	7,070
13	2	6	0,018	0,651	7,492
14	2	6	0,018	0,633	7,905
15	2	3	0,018	0,611	8,305
16	2	3	0,018	0,589	8,695
17	2	3	0,018	0,575	9,064
18	3	3	0,027	0,654	9,483
19	3	3	0,027	0,724	9,966
20	3	0	0,027	0,701	10,425
21	3	0	0.027	0,673	10,866
22	3	0	0,027	0,631	11,279
23	3	0	0,027	0,592	11,669
24	3	0	0,027	0,553	12,042
25	3	0	0,027	0,510	12,386
26	3	0	0,027	0,505	12,721
27	4	0	0,036	0,562	13,091
28	4	0	0,036	0,598	13,488
29	4	0	0,036	0,589	13,873
30	5	0	0,045	0,607	14,268
31	5	0	0,045	0,620	14,679
32	5	0	0,045	0,580	15,067
33	5	0	0,045	0,532	15,424
34	5	0	0,045	0,517	15,771
35	6	0	0,054	0,517	16,120
36	6	0	0,054	0,509	16,466
37	6	0	0,054	0,506	16,807
38	8	0	0,072	0,522	17,158
39	8	0	0,072	0,531	17,518
40	8	0	0,072	0,519	17,869
41	10	0	0,090	0,512	18,215
42	10	0	0,090	0,509	18,562
43	10	0	0,090	0,497	18,902
44	12	0	0,108	0,494	19,239
45	12	0	0,108	0,501	19,579
46	13	0	0,117	0,507	19,925
47	14	0	0,252	0,502	20,269
48	14	0	0,252	0,493	20,606
49	16	0	0,289	0,497	20,944
50	20	0	0,572	0,506	21,288
51	20	0	0,572	0,510	21,635
52	23	0	0,658	0,504	21,979
53	27	0	2,441	0,496	22,319
54	27	0	2,441	0,493	22,656
55	32	0	2,894	0,490	22,993
56	37	0	33,458	0,483	23,326
57	37	0	33,458	0,458	23,656
58	12	0	10,851	0,500	23,937

Table C.7.d -- Sampling_frequency = 48 kHz short blocks

no.	FFT-lines	qthr	norm	SNR (db)	bval
0	1	4,532	0,970	-8,240	0,000
1	1	0,904	0,755	-8,240	1,875
2	1	0,029	0,738	-8,240	3,750
3	1	0,009	0,730	-8,240	5,437
4	1	0,009	0,724	-8,240	6,857
5	1	0,009	0,723	-8,240	8,109
6	1	0,009	0,723	-8,240	9,237
7	1	0,009	0,723	-8,240	10,202
8	1	0,009	0,718	-8,240	11,083
9	1	0,009	0,690	-8,240	11,864
10	1	0,009	0,660	-7,447	12,553
11	1	0,009	0,641	-7,447	13,195
12	1	0,009	0,600	-7,447	13,781
13	1	0,009	0,584	-7,447	14,309
14	1	0,009	0,532	-7,447	14,803
15	1	0,009	0,537	-7,447	15,250
16	1	0,009	0,857	-7,447	15,667
17	1	0,009	0,858	-7,447	16,068
18	1	0,009	0,853	-7,447	16,409
19	2	0,018	0,824	-7,447	17,044
20	2	0,018	0,778	-6,990	17,607
21	2	0,018	0,740	-6,990	18,097
22	2	0,018	0,709	-6,990	18,528
23	2	0,018	0,676	-6,990	18,930
24	2	0,018	0,632	-6,990	19,295
25	2	0,018	0,592	-6,990	19,636
26	3	0,054	0,553	-6,990	20,038
27	3	0,054	0,510	-6,990	20,486
28	3	0,054	0,513	-6,990	20,900
29	4	0,114	0,608	-6,990	21,305
30	4	0,114	0,673	-6,020	21,722
31	5	0,452	0,637	-6,020	22,128
32	5	0,452	0,586	-6,020	22,512
33	5	0,452	0,571	-6,020	22,877
34	7	6,330	0,616	-5,229	23,241
35	7	6,330	0,640	-5,229	23,616
36	11	9,947	0,597	-5,229	23,974
37	17	153,727	0,538	-5,229	24,312

Table C.7.e -- Sampling_frequency = 44,1 kHz short blocks

no.	FFT-lines	qthr	norm	SNR (db)	bval
0	1	4,532	0,952	-8,240	0,000
1	1	0,904	0,700	-8,240	1,723
2	1	0,029	0,681	-8,240	3,445
3	1	0,009	0,675	-8,240	5,057
4	1	0,009	0,667	-8,240	6,422
5	1	0,009	0,665	-8,240	7,609
6	1	0,009	0,664	-8,240	8,710
7	1	0,009	0,664	-8,240	9,662
8	1	0,009	0,664	-8,240	10,515
9	1	0,009	0,655	-8,240	11,293
10	1	0,009	0,616	-7,447	12,009
11	1	0,009	0,597	-7,447	12,625
12	1	0,009	0,578	-7,447	13,210
13	1	0,009	0,541	-7,447	13,748
14	1	0,009	0,575	-7,447	14,241
15	1	0,009	0,856	-7,447	14,695
16	1	0,009	0,846	-7,447	15,125
17	1	0,009	0,840	-7,447	15,508
18	1	0,009	0,822	-7,447	15,891
19	2	0,018	0,800	-7,447	16,537
20	2	0,018	0,753	-6,990	17,112
21	2	0,018	0,704	-6,990	17,620
22	2	0,018	0,674	-6,990	18,073
23	2	0,018	0,640	-6,990	18,470
24	2	0,018	0,609	-6,990	18,849
25	3	0,027	0,566	-6,990	19,271
26	3	0,027	0,535	-6,990	19,741
27	3	0,054	0,531	-6,990	20,177
28	3	0,054	0,615	-6,990	20,576
29	3	0,054	0,686	-6,990	20,950
30	4	0,114	0,650	-6,020	21,316
31	4	0,114	0,612	-6,020	21,699
32	5	0,452	0,567	-6,020	22,078
33	5	0,452	0,520	-6,020	22,438
34	5	0,452	0,513	-5,229	22,782
35	7	6,330	0,557	-5,229	23,133
36	7	6,330	0,584	-5,229	23,484
37	7	6,330	0,570	-5,229	23,828
38	19	171,813	0,578	-4,559	24,173

Table C.7.f -- Sampling_frequency = 32 kHz short blocks

no.	FFT-lines	qthr	norm	SNR (db)	bval
0	1	4,532	0,997	-8,240	0,000
1	1	0,904	0,893	-8,240	1,250
2	1	0,090	0,881	-8,240	2,500
3	1	0,029	0,873	-8,240	3,750
4	1	0,009	0,872	-8,240	4,909
5	1	0,009	0,871	-8,240	5,958
6	1	0,009	0,860	-8,240	6,857
7	1	0,009	0,839	-8,240	7,700
8	1	0,009	0,812	-8,240	8,500
9	1	0,009	0,784	-8,240	9,237
10	1	0,009	0,741	-7,447	9,895
11	1	0,009	0,697	-7,447	10,500
12	1	0,009	0,674	-7,447	11,083
13	1	0,009	0,651	-7,447	11,604
14	1	0,009	0,633	-7,447	12,107
15	1	0,009	0,611	-7,447	12,554
16	1	0,009	0,589	-7,447	13,000
17	1	0,009	0,575	-7,447	13,391
18	1	0,009	0,654	-7,447	13,781
19	2	0,018	0,724	-7,447	14,474
20	2	0,018	0,701	-6,990	15,096
21	2	0,018	0,673	-6,990	15,667
22	2	0,018	0,631	-6,990	16,177
23	2	0,018	0,592	-6,990	16,636
24	2	0,018	0,553	-6,990	17,057
25	2	0,018	0,510	-6,990	17,429
26	2	0,018	0,506	-6,990	17,786
27	3	0,027	0,562	-6,990	18,177
28	3	0,027	0,598	-6,990	18,597
29	3	0,027	0,589	-6,990	18,994
30	3	0,027	0,607	-6,020	19,352
31	3	0,027	0,620	-6,020	19,693
32	4	0,072	0,580	-6,020	20,066
33	4	0,072	0,532	-6,020	20,461
34	4	0,072	0,517	-5,229	20,841
35	5	0,143	0,517	-5,229	21,201
36	5	0,143	0,509	-5,229	21,549
37	6	0,172	0,506	-5,229	21,911
38	7	0,633	0,522	-4,559	22,275
39	7	0,633	0,531	-4,559	22,625
40	8	0,723	0,519	-3,980	22,971
41	10	9,043	0,512	-3,980	23,321

Table C.8 -- Tables for converting threshold calculation partitions to scalefactor bands

Table C.8.a -- Sampling_frequency = 48 kHz long blocks

no. sb	cbw	bu	bo	w1	w2
0	3	0	4	1,000	0,056
1	3	4	7	0,944	0,611
2	4	7	11	0,389	0,167
3	3	11	14	0,833	0,722
4	3	14	17	0,278	0,639
5	2	17	19	0,361	0,417
6	3	19	22	0,583	0,083
7	2	22	24	0,917	0,750
8	3	24	27	0,250	0,417
9	3	27	30	0,583	0,648
10	3	30	33	0,352	0,611
11	3	33	36	0,389	0,625
12	4	36	40	0,375	0,144
13	3	40	43	0,856	0,389
14	3	43	46	0,611	0,160
15	3	46	49	0,840	0,217
16	3	49	52	0,783	0,184
17	2	52	54	0,816	0,886
18	3	54	57	0,114	0,313
19	2	57	59	0,687	0,452
20	1	59	60	0,548	0,908

Table C.8.b -- Sampling_frequency = 44,1 kHz long blocks

no. sb	cbw	bu	bo	w1	w2
0	3	0	4	1,000	0,056
1	3	4	7	0,944	0,611
2	4	7	11	0,389	0,167
3	3	11	14	0,833	0,722
4	3	14	17	0,278	0,139
5	1	17	18	0,861	0,917
6	3	18	21	0,083	0,583
7	3	21	24	0,417	0,250
8	3	24	27	0,750	0,805
9	3	27	30	0,194	0,574
10	3	30	33	0,426	0,537
11	3	33	36	0,463	0,819
12	4	36	40	0,180	0,100
13	3	40	43	0,900	0,468
14	3	43	46	0,532	0,623
15	3	46	49	0,376	0,450
16	3	49	52	0,550	0,552
17	3	52	55	0,448	0,403
18	2	55	57	0,597	0,643
19	2	57	59	0,357	0,722
20	2	59	61	0,278	0,960

Table C.8.c -- Sampling_frequency = 32 kHz long blocks

no. sb	cbw	bu	bo	w1	w2
0	1	0	2	1,000	0,528
1	2	2	4	0,472	0,305
2	2	4	6	0,694	0,083
3	1	6	7	0,917	0,861
4	2	7	9	0,139	0,639
5	2	9	11	0,361	0,417
6	3	11	14	0,583	0,083
7	2	14	16	0,917	0,750
8	3	16	19	0,250	0,870
9	3	19	22	0,130	0,833
10	4	22	26	0,167	0,389
11	4	26	30	0,611	0,478
12	4	30	34	0,522	0,033
13	3	34	37	0,967	0,917
14	4	37	41	0,083	0,617
15	3	41	44	0,383	0,995
16	4	44	48	0,005	0,274
17	3	48	51	0,726	0,480
18	3	51	54	0,519	0,261
19	2	54	56	0,739	0,884
20	2	56	58	0,116	1,000

Table C.8.d -- Sampling_frequency = 48 kHz short blocks

no. sb	cbw	bu	bo	w1	w2
0	2	0	3	1,000	0,167
1	2	3	5	0,833	0,833
2	3	5	8	0,167	0,500
3	3	8	11	0,500	0,167
4	4	11	15	0,833	0,167
5	4	15	19	0,833	0,583
6	3	19	22	0,417	0,917
7	4	22	26	0,083	0,944
8	4	26	30	0,055	0,042
9	2	30	32	0,958	0,567
10	3	32	35	0,433	0,167
11	2	35	37	0,833	0,618

Table C.8.e -- Sampling_frequency = 44,1 kHz short blocks

no. sb	cbw	bu	bo	w1	w2
0	2	0	3	1,000	0,167
1	2	3	5	0,833	0,833
2	3	5	8	0,167	0,500
3	3	8	11	0,500	0,167
4	4	11	15	0,833	0,167
5	5	15	20	0,833	0,250
6	3	20	23	0,750	0,583
7	4	23	27	0,417	0,055
8	3	27	30	0,944	0,375
9	3	30	33	0,625	0,300
10	3	33	36	0,700	0,167
11	2	36	38	0,833	1,000

Table C.8.f -- Sampling_frequency = 32 kHz short blocks

no. sb	cbw	bu	bo	w1	w2
0	2	0	3	1,000	0,167
1	2	3	5	0,833	0,833
2	3	5	8	0,167	0,500
3	3	8	11	0,500	0,167
4	4	11	15	0,833	0,167
5	5	15	20	0,833	0,250
6	4	20	24	0,750	0,250
7	5	24	29	0,750	0,055
8	4	29	33	0,944	0,375
9	4	33	37	0,625	0,472
10	3	37	40	0,528	0,937
11	1	40	41	0,062	1,000

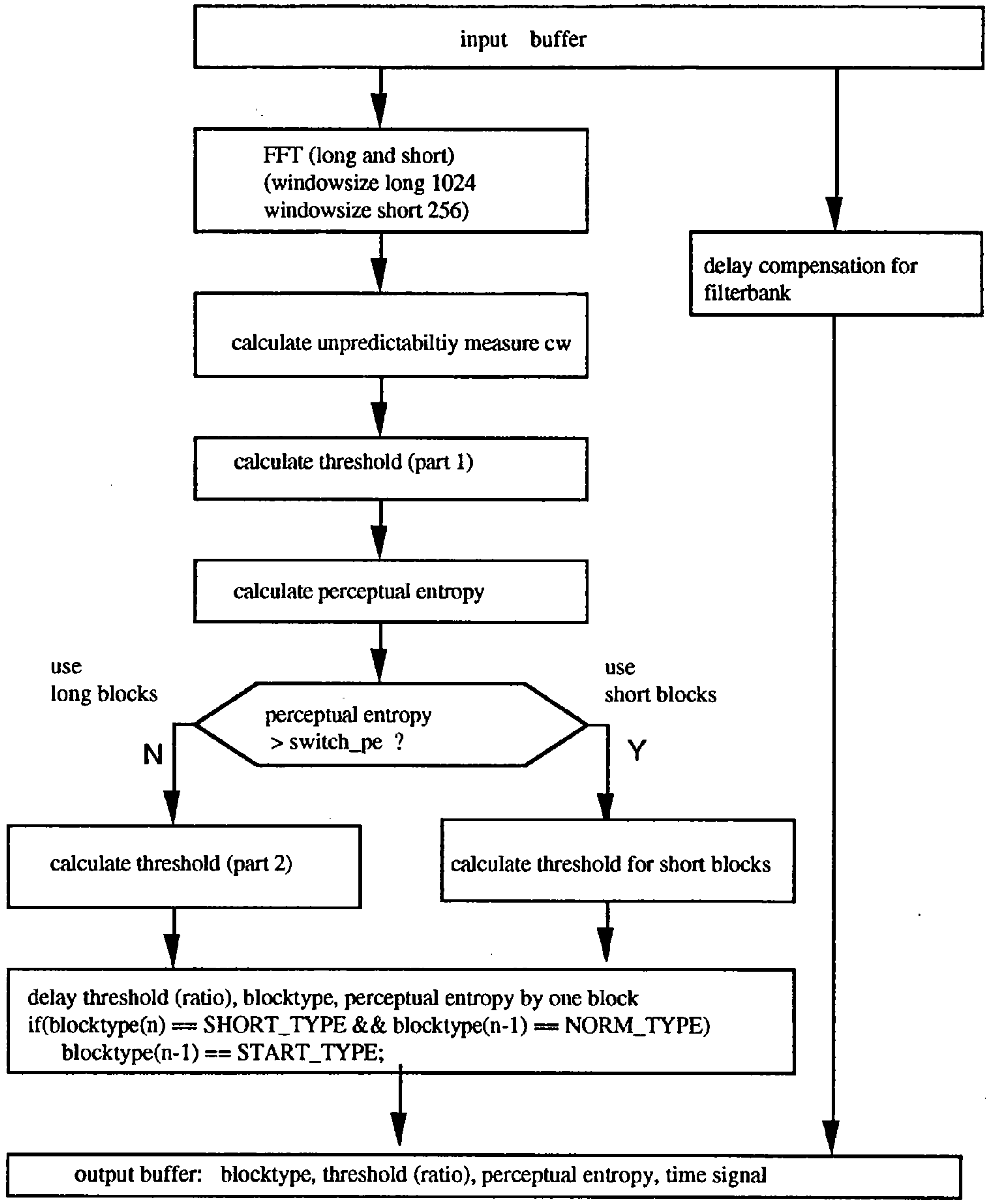

Figure C.6.a -- Block diagram psychoacoustic model 2, Layer III: calculate threshold

calculate the energy and the unpredictability in the threshold calculation partitions

$eb(b) = \sum r(w)^2$

$cb(b) = \sum cw(w) \cdot r(w)^2$

convolve the partitioned energy and unpredictability with the spreading function

$ecbb(b) = eb * sprdngf$

$ctb(b) = cb * sprdngf$

calculate the tonality of each threshold calculation partition

$tbb(b) = conv1 + conv2 \cdot \log_e\left(\frac{ctb(b)}{ecbb(b)}\right)$

calculate the SNR in each threshhold calculation partition

$SNR(b) = MAX\,(minval(b),\ TMN(b) \cdot tbb(b) + NMT(b) \cdot (1 - tbb(b)))$

calculate the threshold for each partition

$nbb(b) = ecbb(b) \cdot norm(b) \cdot 10^{SNR(b)/10}$

norm(b) is a normalizing constant for each partition

pre-echo control:
compare the threshold with the last threshold and the threshold in quiet and take the maximum:

$thr(b) = MAX(qthr(b), nbb(b), nbb\_l(b), nbb\_ll(b))$

$nbb\_l(b) = rpelev \cdot nbb(b)$ from the last block

$nbb\_ll(b) = rpelev2 \cdot nbb(b)$ from the block before the last block

Figure C.6.b -- Block diagram psychoacoustic model 2, Layer III: calculate threshold (part 1)

convert from threshold calculation partitions to codec partitions (scalefactor bands):

energy in each scalefactor band:

$$en(sb) = w1 \bullet eb(bu) + \sum_{b=bu+1}^{b=bo-1} eb(b) + w2 \bullet eb(bo)$$

threshold in each scalefactor band:

$$thrn(sb) = w1 \bullet thr(bu) + \sum_{b=bu+1}^{b=bo-1} thr(b) + w2 \bullet thr(bo)$$

Calculate ratios for each scalefactor band:

$$ratio(sb) = \frac{thrn(sb)}{en(sb)}$$

Figure C.6.c -- Block diagram psychoacoustic model 2, Layer III: calculate threshold (part 2)

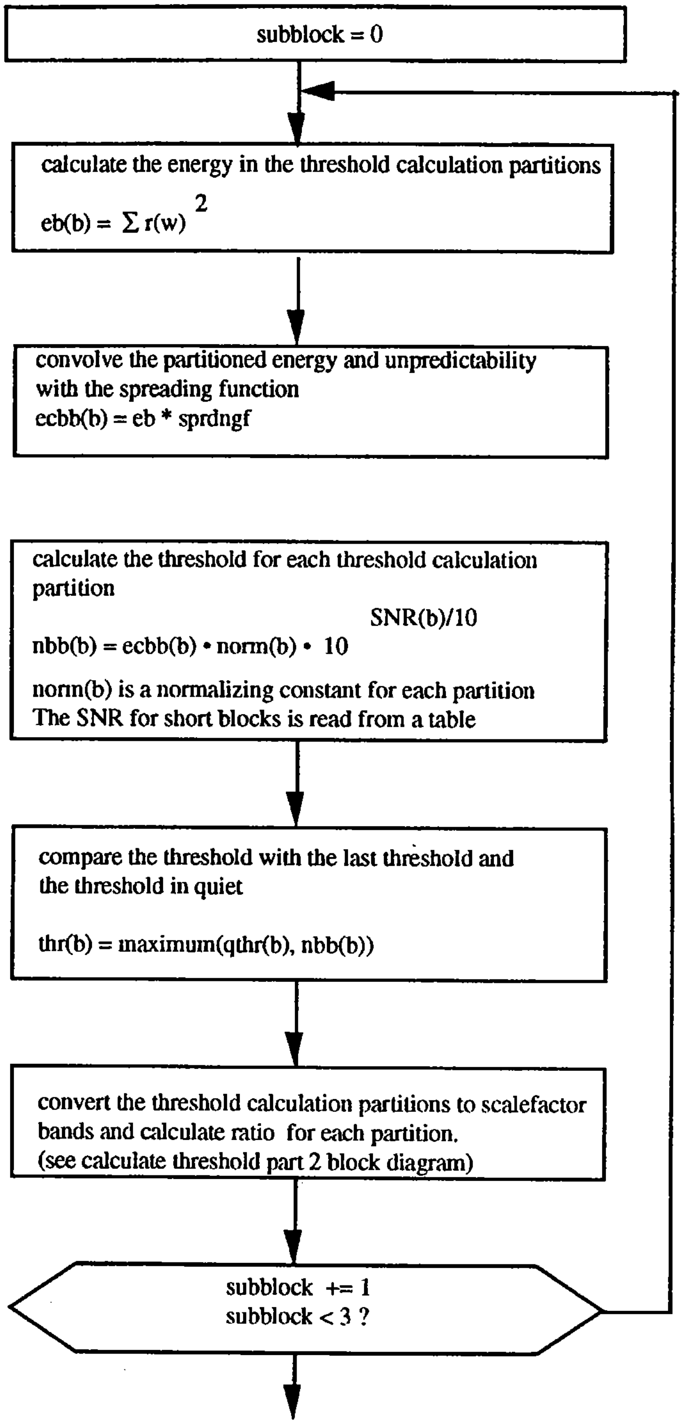

Figure C.6.d -- Block diagram psychoacoustic model 2, Layer III: calculate threshold for short blocks

Window switching decision:

The decision whether the filterbank should be switched to short windows is derived from the calculation of the masking threshold by calculating the estimate of the psychoacoustic entropy (PE) and switching when the PE exceeds the value 1800. If this condition is met, the sequence start (block_type=1), short (block_type=2), short, stop (block_type=3) is started. Figure C.7 shows the possible state changes for the window switching logic.

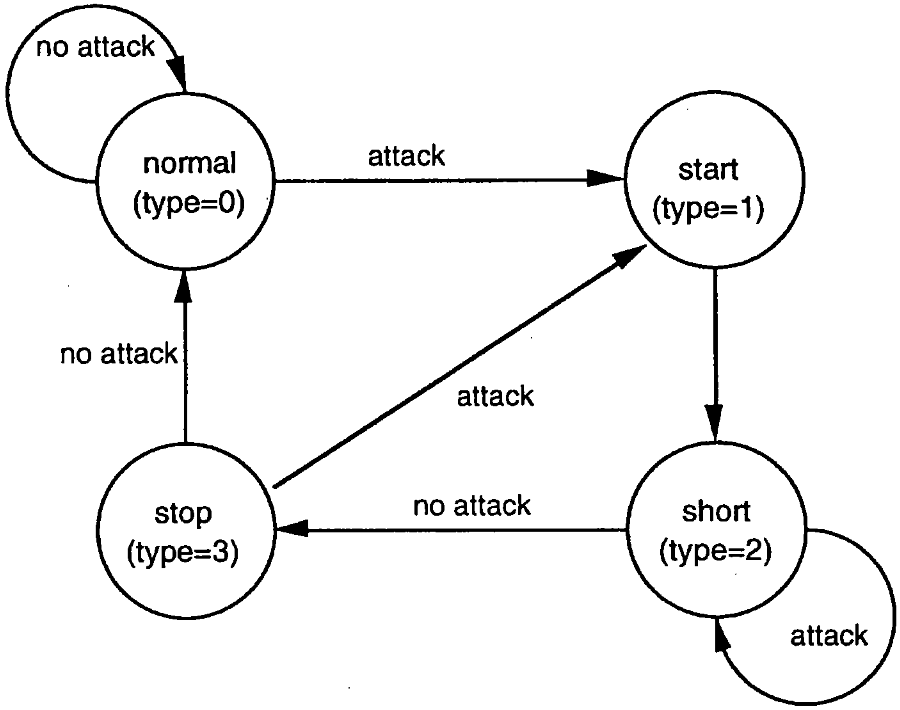

Figure C.7 -- Window Switching State Diagram

C.1.5.3.3 Analysis part of the hybrid filterbank

The subband analysis of the polyphase filterbank is described in clause C.1.3, "Subband analysis filter". The output of the polyphase filterbank is the input to the subdivision using the MDCT. According to the output of the psychoacoustic model (variables **blocksplit_flag** and **block_type**) the window and transform types **normal, start, short** or **stop** are used.

18 consecutive output values of one granule and 18 output values of the granule before are assembled to one block of 36 samples.

Block type "**normal**"

$$z_i = x'_i \sin\left(\frac{\pi}{36}\left(i + \frac{1}{2}\right)\right) \qquad \text{for } i=0 \text{ to } 35$$

Block type "**start**"

$$z_i = \begin{cases} x'_i \sin\left(\frac{\pi}{36}\left(i+\frac{1}{2}\right)\right) & \text{for } i=0 \text{ to } 17 \\ x'_i & \text{for } i=18 \text{ to} 23 \\ x'_i \sin\left(\frac{\pi}{12}\left(i-18+\frac{1}{2}\right)\right) & \text{for } i=24 \text{ to } 29 \\ 0 & \text{for } i=30 \text{ to } 35 \end{cases}$$

Block type "**stop**"

$$z_i = \begin{cases} 0 & \text{for } i=0 \text{ to } 5 \\ x'_i \sin\left(\frac{\pi}{12}\left(i-6+\frac{1}{2}\right)\right) & \text{for } i=6 \text{ to } 11 \\ x'_i & \text{for } i=12 \text{ to } 17 \\ x'_i \sin\left(\frac{\pi}{36}\left(i+\frac{1}{2}\right)\right) & \text{for } i=18 \text{ to } 35 \end{cases}$$

Block type "**short**"

The block of 36 samples is divided into three overlapping blocks:

$$y_i^{(0)} = x'_{i+6} \quad \text{for } i=0 \text{ to } 11$$
$$y_i^{(1)} = x'_{i+12} \quad \text{for } i=0 \text{ to } 11$$
$$y_i^{(2)} = x'_{i+18} \quad \text{for } i=0 \text{ to } 11$$

Each of the three small blocks is windowed separately:

$$z_i^{(k)} = y_i^{(k)} \sin\left(\frac{\pi}{12}\left(i+\frac{1}{2}\right)\right) \quad \text{for } i=0 \text{ to } 11, \text{ for } k=0 \text{ to } 2$$

MDCT:

In the following n is the number of windowed samples. For short blocks n is 12, for long blocks n is 36. The analytical expression of the MDCT is:

$$x_i = \sum_{k=0}^{n-1} z_k \cos\left(\frac{\pi}{2n}\left(2k+1+\frac{n}{2}\right)(2i+1)\right) \qquad \text{for } i=0 \text{ to } \frac{n}{2}-1$$

Aliasing-Butterfly, Encoder:

The calculation of aliasing reduction in the encoder is performed as in the decoder. The general procedure is shown in figure A.5. The butterfly definition to be used in the encoder is shown in figure C.8. The coefficients ca_i and cs_i can be found in table B.9.

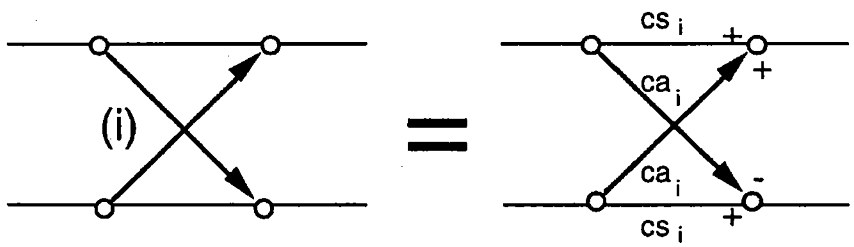

Figure C.8 -- Encoder Butterfly Definition

C.1.5.3.4 Calculation of average available bits

The average number of bits per granule is calculated from the frame size. The bitrate 64 kbits/s is used as an example. At bitrate 64 kbits/s at 48 000 samples per second,

(64 000 * (1 152/48 000) bits per frame) / (2 granules per frame) = 768 bits per granule.

As the header takes 32 bits and the side information takes 17 bytes (136 bits) in single_channel mode, the average amount of available bits for the main_data for a granule is given by

mean_bits = 768 bits per granule - (32+136 bits per frame)/(2 granules per frame) = 684 bits per granule.

Bit reservoir:

The bit reservoir can provide additional bits which may be used for the granule. The number of bits which are provided is determined within the iteration loops.

C.1.5.3.5 Quantization and encoding of frequency domain samples

The frequency domain data are quantized and coded within two nested iteration loops. Sublause C.1.5.4 contains a detailed description of these iteration loops.

C.1.5.3.6 Ancillary data

The Audio Standard provides a number of bits for the inclusion and transmission of variable length ancillary data with the audio bitstream. The ancillary data will reduce the number of bits available for audio, which may result in a degradation of audio quality.

The presence of a bit pattern in the ancillary data matching the syncword may hamper synchronization. This problem is more likely to occur when the free format is used.

C.1.5.3.7 Formatting

The details about the Layer III bitstream format can be found in 2.4.4. The formatting of the Huffman code words is described below:

The Huffman code words are in sequence from low to high frequencies. In the iteration loops the following variables have been calculated and are used in encoding the Huffman code words:

is(i), i=0...575	quantized frequency domain values
table_select[region]	Huffman code table used for regions (region = 0, 1, 2)
region_adress1	defines the border between region 0 and 1
region_adress2	defines the border between region 1 and 2
max_value[region]	maximum absolute value of quantized data in regions (region = 0, 1, 2)

The data are written to the bitstream according to the Huffman code syntax described in 2.4.2.7

The actual assembly of the Huffman code for the big_values part is described in a pseudo high level language:

```
for region number from 0 to 2
 if table_select for this region is 0
  nothing to do, all values in region are zero
 else
  if table_select for this region is > 15
   an ESC-table is used: look up linbits value connected to the table used
   for i = begin of region to end of region, count in pairs
    x = is(i), y = is(i+1)
    if x > 14
     linbitsx = x - 15, x = 15
    end if
    signx = sign(x), x = abs(x)
    if y > 14
     linbitsy = y - 15, y = 15
    end if
    signy = sign(y), y = abs(y)
    look for codeword = hcod([x][y]]) in table table_select
    write hcod([x][y]), beginning with the leftmost bit, number of bits is hlen([x][y])
    if x > 14
     write linbitsx to the bitstream, number of bits is linbits
    end if
    if x != 0
     write signx to bitstream
    end if
    if y > 14
     write linbitsy to the bitstream, number of bits is linbits
    end if
    if y != 0
     write signy to bitstream
    end if
   end do
  else
   no ESC-words are used in this region:
   for i = beginning of region to end of region, count in pairs
    x = is(i), y = is(i+1)
    signx = sign(x), x = abs(x)
    signy = sign(y), y = abs(y)
    look for codeword = hcod([x][y]) in table table_seletct
    write hcod([x][y]), beginning with the leftmost bit, number of bits is hlen([x][y])
    if x != 0
     write signx to bitstream
    end if
    if y != 0
     write signy to bitstream
    end if
   end do
  end if
 end if
end for
```

A possible application for the private_bits is to use them as frame counter.

C.1.5.4 Layer III iteration loops

C.1.5.4.1 Introduction

The description of the Layer III loop module is subdivided into three levels. The top level is called "loops frame program". The loops frame program calls a subroutine named "outer iteration loop" which calls the subroutine "inner iteration loop". For each level a corresponding flow diagram is shown.

The loops module quantizes an input vector of spectral data in an iterative process according to several demands. The inner loop quantizes the input vector and increases the quantizer step size until the output vector can be coded with the available number of bits. After completion of the inner loop an outer loop checks the distortion of each scalefactor band and, if the allowed distortion is exceeded, amplifies the scalefactor band and calls the inner loop again.

Layer III loops module input:

(1) vector of the magnitudes of the spectral values xr(0..575).
(2) xmin(sb), the allowed distortion of the scalefactor bands. xmin = ratio(sb) * en(sb) / bw(sb).
(3) window_switching_flag which, in conjunction with mixed_block_flag and block_type, determines the number of scalefactor bands.
(4) mean_bits (bit available for the Huffman coding and the coding of the scalefactors).
(5) more_bits, the number of bits in addition to the average number of bits, as demanded by the value of the psychoacoustic entropy for the granule:
more_bits = 3.1 * PE - (average number of bits)

Layer III loops module output:

(1) vector of quantized values ix(0..575).
(2) scalefac_l(sb) or scalefac_s(sb) depending on window_switching_flag, block_type and mixed_block_flag.
(3) global_gain (quantizer step size information)
global_gain = qquant + system_constant.
system_constant includes all the scaling operations of the encoder and an offset to achieve the correct output with the decoding process described in the main part.
(4) number of unused bits available for later use.
(5) preflag (loops preemphasis on/off).
(6) Huffman code related side information
- big_values (number of pairs of Huffman coded values, excluding "count1")
- count1table_select (Huffman code table of absolut values <= 1 at the upper end of the spectrum
- table_select[0..2](Huffman code table of regions)
- region0_count, region1_count (used to calculate boundaries between regions)
- part2_3_length

C.1.5.4.2 Preparatory steps

C.1.5.4.2.1 Reset of all iteration variables

The scalefactors of the coder partitions, scalefac_l[sb] or scalefac_s[sb], are respectively set to zero.

The counter qquant for the quantizer step size is reset to zero.

Preflag is reset to zero.

Scalefac_scale is reset to zero.

The inital value of quantanf is set as follows:

quantanf = system_const * $\log_e$(sfm),

where sfm is the spectral flatness measure and quantanf depends on the computational implementation of the encoder.

The spectral flatness measure sfm is given by

$$\mathrm{sfm} = \frac{e^{\frac{1}{n}\left(\sum_{i=0}^{n-1} \log \mathrm{xr}(i)^2\right)}}{\frac{1}{n}\sum_{i=0}^{n-1} \mathrm{xr}(i)^2}$$

The value of system_const is chosen so that for all signals the first iteration of the inner loop for all signals comes out with a bit sum higher than the desired bitsum. By that it is ensured that the first call of the inner loop results in the solution which uses as many of the available bits as possible. In order to spare computing time it is desirable to minimize the number of iterations by adapting the value of quantanf to the bitrate and the signal statistics.

C.1.5.4.2.2 Bit reservoir control

Bits are saved to the reservoir when fewer than the mean_bits are used to code one granule. If bits are saved for a frame, the value of main_data_end is increased accordingly. See figure A.7.a.

The number of bits which are made available for the main_data (called "max_bits") is derived from the actual estimated threshold (the PE as calculated by the psychoacoustic model), the average number of bits (mean_bits) and the actual content of the bit reservoir. The number of bytes in the bit reservoir is given by main_data_end.

The actual rules for the control of the bit reservoir are given below:

- If a number of bytes available to the inner iteration loop is not used for the Huffman encoding or other main_data, the number is added to the bit reservoir.
- If the bit reservoir contains more than 0,8 times the maximum allowed content of the bit reservoir, all bytes exceeding this number are made available for main_data (in addition to mean_bits)
- If more_bits is greater than 100 bits, then max(more_bits/8, 0,6*main_data_end) bytes are taken from the bit reservoir and made available for main_data (in addition to mean_bits).
- After the actual loops computations have been completed, the number of bytes not used for main_data is added to the bit reservoir.
- If after the step above the number of bytes in the bit reservoir exceeds the maximum allowed content, stuffing bits are written to the bitstream and the content of the bit reservoir is adjusted accordingly.

C.1.5.4.2.3 Calculation of the scalefactor selection information (scfsi)

The scfsi contains the information, which scalefactors (grouped in the scfsi_bands) of the first granule can also be used for the second granule. These scalefactors are therefore not transmitted, the bits gained can be used for the Huffman coding.

To determine the usage of the scfsi, the following information of each granule must be stored:

a) The block type

b) The total energy of the granule:

$$\text{en\_tot} = \text{int}\left\{ \log 2\left(\sum_{i=1}^{n} |\ \text{xr}(i)|^2 \right) \right\}$$

where n is the total number of spectral values.

c) The energy of each scalefactor band:

$$\text{en(sb)} = \text{int}\left\{ \log_2\left(\sum_{i=\text{lbl(sb)}}^{\text{lbl(sb)+bw(sb)-1}} |\ \text{xr}(i)\ |^2 \right) \right\}$$

where lbl(sb) is the number of the first coefficient belonging to scalefactor band sb and bw(sb) is the number of coefficients within scalefactor band sb.

d) The allowed distortion of each scalefactor band:

$$xm(sb) = \text{int}\left\{ \log_2 \left(xmin(i) \right) \right\}$$

xmin(sb) is calculated by the psychoacoustic model.

The scalefactors of the first granule are always transmitted. When coding the second granule, the information of the two granules is compared. There are four criteria to determine if the scfsi can be used in general. If one of the four is not fulfilled, the scfsi is disabled (that means it is set to 0 in all scfsi_bands). The criteria are (index 0 means first, index 1 second granule):

a) The spectral values are not all zero

b) None of the granules contains short blocks

c)

$$\left| en\_tot_0 - en\_tot_1 \right| < en\_tot_{krit}$$

d)

$$\sum_{\text{all scalefactor bands}} \left| en(sb)_0 - en(sb)_1 \right| < en\_dif_{krit}$$

If the scfsi is not disabled after the tests above, there are two criteria for each scfsi_band, which have both to be fulfilled to enable scfsi (that means to set it to 1 in this scfsi_band):

a)

$$\sum_{\text{all cr. bds in scfsi\_band}} \left| en(sb)_0 - en(sb)_1 \right| < en(scfsi\_band)_{krit}$$

b)

$$\sum_{\text{all cr. bds in scfsi\_band}} \left| xm(sb)_0 - xm(sb)_1 \right| < xm(scfsi\_band)_{krit}$$

The constants (with the index *krit*) have to be chosen so, that the scfsi is only enabled in case of similar energy/distortion.

Suggested values are:

en_tot_{krit}	=	10	
en_dif_{krit}	=	100	
$en(scfsi\_band)_{krit}$	=	10	for each scfsi_band
$xm(scfsi\_band)_{krit}$	=	10	for each scfsi_band

C.1.5.4.3 Outer iteration loop (distortion control loop)

The outer iteration loop controls the quantization noise which is produced by the quantization of the frequency domain lines within the inner iteration loop. The colouring of the noise is done by multiplication of the lines within scalefactor bands with the actual scalefactors before doing the quantization. The following pseudo-code illustrates the multiplication.

```
do for each scalefactor band:
    do from lower index to upper index of scale factor band
        xr(i) = xr(i) * √(2) ^ ((1 + scalefac_scale) *  scalefac(sb) )
    end do
end do
```

Where scalefac is either scalefac_l or scalefac_s as appropriate.

In the actual system the multiplication is done incrementally with just the increase of the scalefactors applied in each distortion control loop. This is described in C.1.5.4.3.5 below.

The distortion loop is always started with scalefac_scale = 0. If after some iterations the maximum length of the scalefactors would be exceeded (see scalefac_compress table in 2.4.2.7 and C.1.5.4.3.5 below), then scalefac_scale is increased to the value 1 thus increasing the possible dynamic range of the scalefactors. In this case the actual scalefactors and frequency lines have to be corrected accordingly.

C.1.5.4.3.1 Saving of the scalefactors

The scalefactors of all scalefactor bands, scalefac_1(sb) or scalefac_s(sb), as well as the quantizer step size qquant are saved. If the computation of the outer loop is cancelled without having reached a proper result, this value together with the quantized spectrum give an approximation and can be transmitted.

C.1.5.4.3.2 Call of inner iteration loop

For each outer iteration loop (distortion control loop) the inner iteration loop (rate control loop) is called. The parameters are the frequency domain values (hybrid filterbank output) with the scalefactors applied to the values within the scalefactor bands and the number of bits which are available to the rate control loop. The result is the number of bits actually used and the quantized frequency lines ix(i).

C.1.5.4.3.3 Calculation of the distortion of the scalefactor bands

For each scalefactor band the actual distortion is calculated according to:

$$xfsf(sb) = \sum_{lbl(sb)}^{i=lbl(sb)+bw(sb)-1} \frac{\left(|xr(i)| - ix(i)^{4/3} * \sqrt[4]{2}^{\,qquant+quantanf} \right)^2}{bandwidth(sb)}$$

where lbl(sb) is the number of the coefficient representing the lowest frequency in a scalefactor band and bw(sb) is the number of coefficients within this band.

C.1.5.4.3.4 Preemphasis

The preemphasis option (switched on by setting preflag to a value of 1) provides the possibility to amplifiy the upper part of the spectrum according to the preemphasis tables, table B.6.

```
if (preflag==1) {
        ifqstep = 2 ^ (0,5*(1+scalefac_scale) )
        xmin(j) = xmin(j) *ifqstep ^ (2*prefact(j))
        for (i=lower limit of scalefactor band j; i <=upper limit of scalefactor band j; i++)  {
                xr(i) = xr(i) * ifqstep^prefact(j)
        }
}
```

The condition to switch on the preemphasis is up to the implementation. For example preemphasis could be switched on if in all of the upper 4 scalefactor bands the actual distortion exceeds the threshold after the first call of the inner loop.

If the second granule is being coded and scfsi is active in at least one scfsi_band, the preemphasis in the second granule is set equal to the setting in first granule.

C.1.5.4.3.5 Amplification of scalefactor bands which violate the masking threshold

All spectral values of the scalefactor bands which have a distortion that exceeds the allowed distortion are amplified by a factor of ifqstep. The value of ifqstep is transmitted by scalefac_scale.

```
if ( (xmin - xfsf) of scalefactor band j < 0 ) {
        xmin(j) = xmin(j) * ifqstep ^ 2
        ifq(j) = ifq(j) + 1
        for (i=lower limit of scalefactor band; i <=upper limit of scalefactor band; i++) {
                xr(i) = xr(i) * ifqstep
        }
}
```

If the second granule is being coded and scfsi is active in at least one scfsi_band, the following steps have to be done:

a) ifqstep has to be set similar to the first granule

b) If it is the first iteration, the scalefactors of scalefactor bands in which scfsi is enabled have to be taken over from the first granule. The corresponding spectral values have to be amplified:

```
if ( scfsi according to scalefactor band j == 1) {
    ifq(j) = ifq(j)first granule
    for (i=lower limit of scalefactor band; i <=upper limit of scalefactor band; i++) {
        xr(i) = xr(i) * ifqstep ^ scalefac(j)
    }
}
```

Where scalefac is either scalefac_1 () or scalefac_s() as appropriate.

c) If it is not the first iteration, the amplification must be prevented for scalefactor bands in which scfsi is enabled.

C.1.5.4.3.6 Conditions for the termination of the loops processing

Normally the loops processing terminates if there is no scalefactor band with more than the allowed distortion. However this is not always possible to obtain. In this case there are other conditions to terminate the outer loop. If

a) All scalefactor bands are already amplified, or
b) The amplification of at least one band exceeds the upper limit which is determined by the transmission format of the scalefactors. The upper limit is a scalefactor of 15 for scalefactor bands 0 through 10 and 7 for scalefactors 11 through 20. In the case of block_type == 2 and mixed_block_flag == 0, the upper limit is 15 for scalefactors 0 through 18. In the case of block_type == 2 and mixed_block_flag == 1, the upper limit is 15 for scalefactors 0 through 17. The upper limit is 7 for other scalefactors.

The loop processing stops, and by restoring the saved sacalefac_1(sb) or scalefac_s(sb) a useful output is available. For realtime implementation, there might be a third condition added which terminates the loops in case of a lack of computing time.

C.1.5.4.4 Inner iteration loop (rate control loop)

The inner iteration loop does the actual quantization of the frequency domain data and prepares the formatting. The table selection, subdivision of the big_values range into regions and the selection of the quantizer step size takes place here.

C.1.5.4.4.1 Quantization

The quantization of the complete vector of spectral values is done according to

$$ix(i) = \mathrm{nint}\left(\left(\frac{|xr(i)|}{\sqrt[4]{2}^{\,qquant+quantanf}}\right)^{0,75} - 0,0946\right)$$

C.1.5.4.4.2 Test of the maximum of the quantized values

The maximum allowed quantized value is limited. This limit is set to constraint the table size if a table-lookup is used to requantize the quantized frequency lines. The limit is given by the possible values of the length identifier, "linbits", of values flagged with an ESC-code. Therefore before any bit counting is done the quantizer stepsize is increased by

qquant = qquant+1

until the maximum of the quantized values is within the range of the largest Huffman code table.

C.1.5.4.4.3 Calculation of the run length of zeros

The run length rzero of pairs of spectral coefficients quantized to zero on the upper end of the spectrum is counted and called "rzero".

C.1.5.4.4.4 Calculation of the run length of values less or equal one

The run length of quadrupels of spectral coefficients quantized to one or zero, following the rzero pairs of zeros, is calculated and called "count1".

C.1.5.4.4.5 Counting the bits necessary to code the values less or equal one

One Huffman code word is used to code one of the "count1" quadrupels. There are two different Huffman code books with corresponding code length tables (table A and table B in clause B.7). The number of bits to code all the count1 quadrupels is given by:

bitsum_count1 = min(bitsum_table0 , bitsum_table1)

where count1table_0 is used to point to table A

$$\text{bitsum\_table0} = \sum_{k=\text{firstcount1}}^{k=\text{firstcount1}+\text{count1}-1} \text{count1table\_0}\ (\ ix(4k)+2*ix(4k+1)+4*ix(4k+2)+8*ix(4k+3)\ )$$

and count1table_1 is used to point to table B

$$\text{bitsum\_table1} = \sum_{k=\text{firstcount1}}^{k=\text{firstcount1}+\text{count1}-1} \text{count1table\_1}\ (\ ix(4k)+2*ix(4k+1)+4*ix(4k+2)+8*ix(4k+3)\ )$$

Count1table_0 as well as count1table_1 have to include the number of bits necessary to encode the sign bits.

The information which table is used is transmitted by count1table_select, which is "0" for table A or "1" for table B, respectively.

C.1.5.4.4.6 Call of subroutine SUBDIVIDE

The number of pairs of quantized values not counted in "count1" or "rzero" is called bigvalues. SUBDIVIDE splits the scalefactor bands corresponding to this values into three groups. The last one, incomplete generally, counts as a complete one. The number of scalefactor bands in the first and second regions are contained in (region0_count+1) and (region1_count+1) respectively. The number of scalefactor bands in the third region can be calculated using bigvalues. The split strategy is up to the implementation. A very simple one for instance is to assign 1/3 of the scalefactor bands to the first and 1/4 to the last region.

Subdivide in case of blocksplit is done analoguously but there are only two subregions. Region1_count is set to a default in this case. This default is 8 in the case of split_point=0 and 9 in the case of split_point=1. Both these values point to the same absolute frequency.

C.1.5.4.4.7 Calculation of the code book for each subregion

There are 32 different Huffman code tables available for the coding of pairs of quantized values. They differ from each other in the maximum value that can be coded and in the signal statistics for which they are optimized. Only codes for values < 16 are in the table. For values >=16 there are two tables provided, where the largest value 15 is an escape character. In this case the value 15 is coded in an additional word using a linear PCM code with a word length called linbits.

A simple way to choose a table is to use the maximum of the quantized values in a subregion. tables which have the same size are optimized for different signal statistics. Therefore additional coding gain can be achieved for example by trying all of these tables.

C.1.5.4.4.8 Counting of the bits necessary to code the values in the subregions

The number of bits necessary to code the quantized values of a subregion is given by:

$$\text{bitsum}(j) = \sum_{k=0}^{k=np(j)-1} \text{bitz}\big(\text{tableselect}(j), \min(15, ix(2k+fe(j))), \min(15, ix(2k+fe(j)+1))\big)$$

$$+ \sum_{k=0}^{k=np(j)-1} \big(s(ix(2k+fe(j)) - 15) + s(ix(2k+fe(j)+1) - 15)\big) * \text{linbits}(j)$$

np(j): number of pairs in a sub region
fe(j): number of the first quantized value in a sub-region
bitz: table with Huffman code length

s(...)	step function:	if x >= 0	s(x) = 1
		if x < 0	s(x) = 0

Note that the Huffman code length tables have to include the number of bits necessary to encode the sign bits.

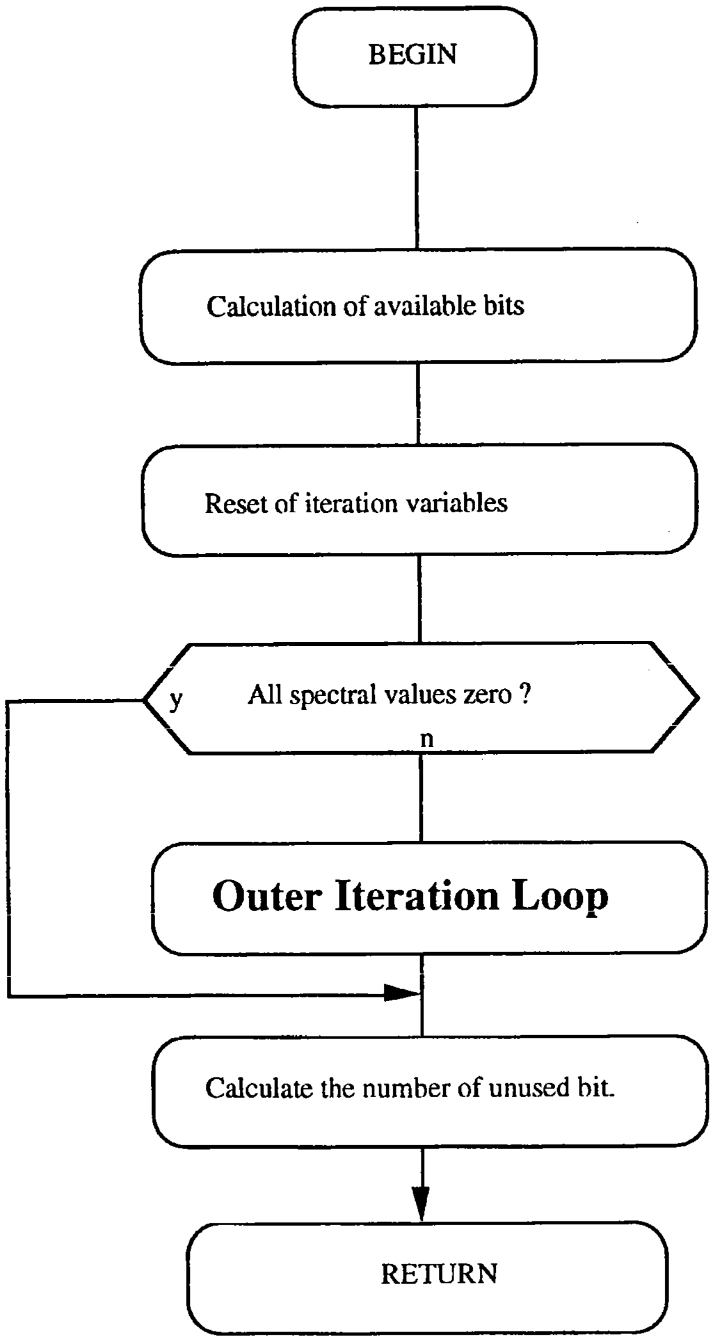

Figure C.9.a -- Layer III iteration loop

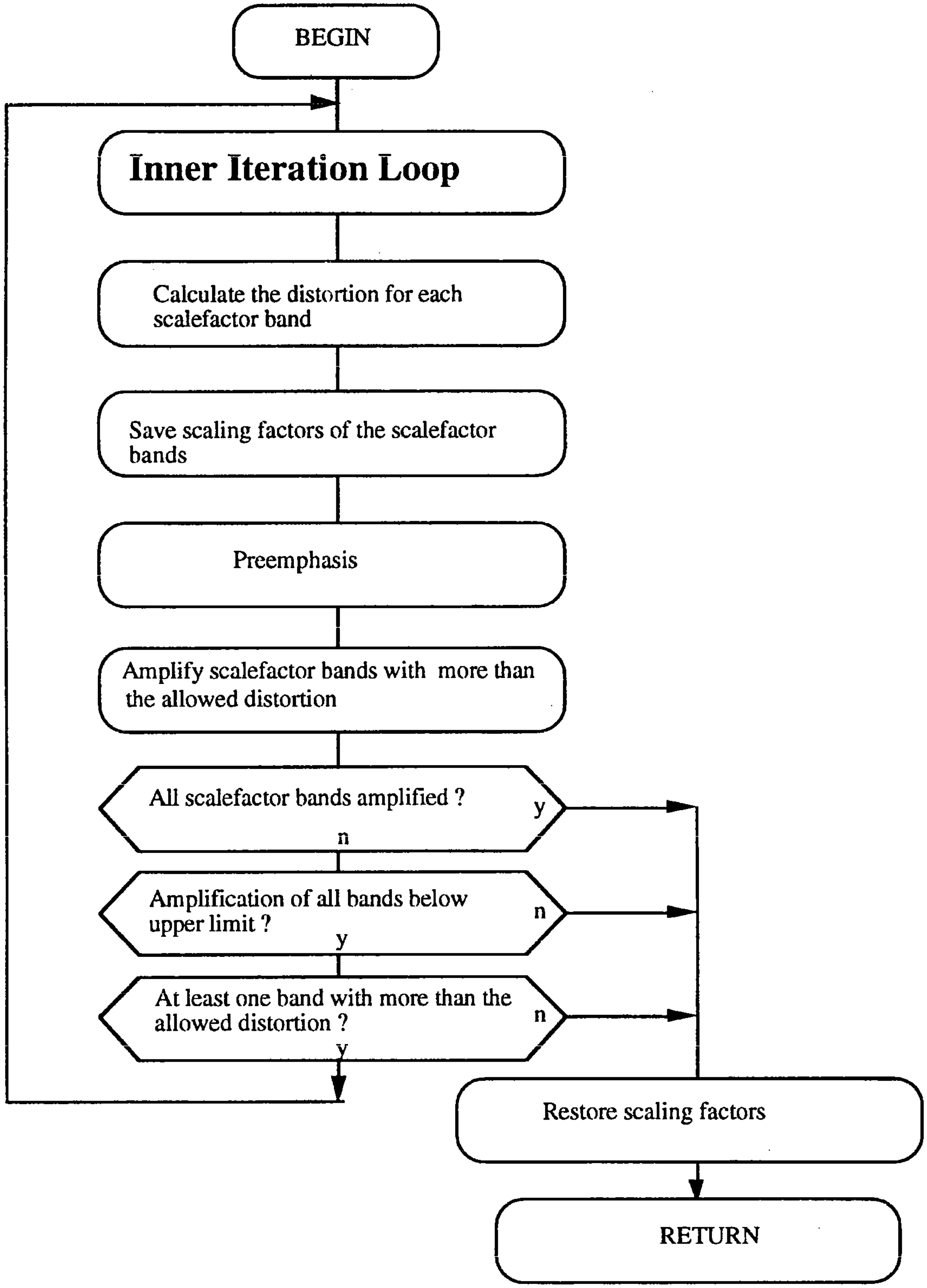

Figure C.9.b -- Layer III outer iteration loop

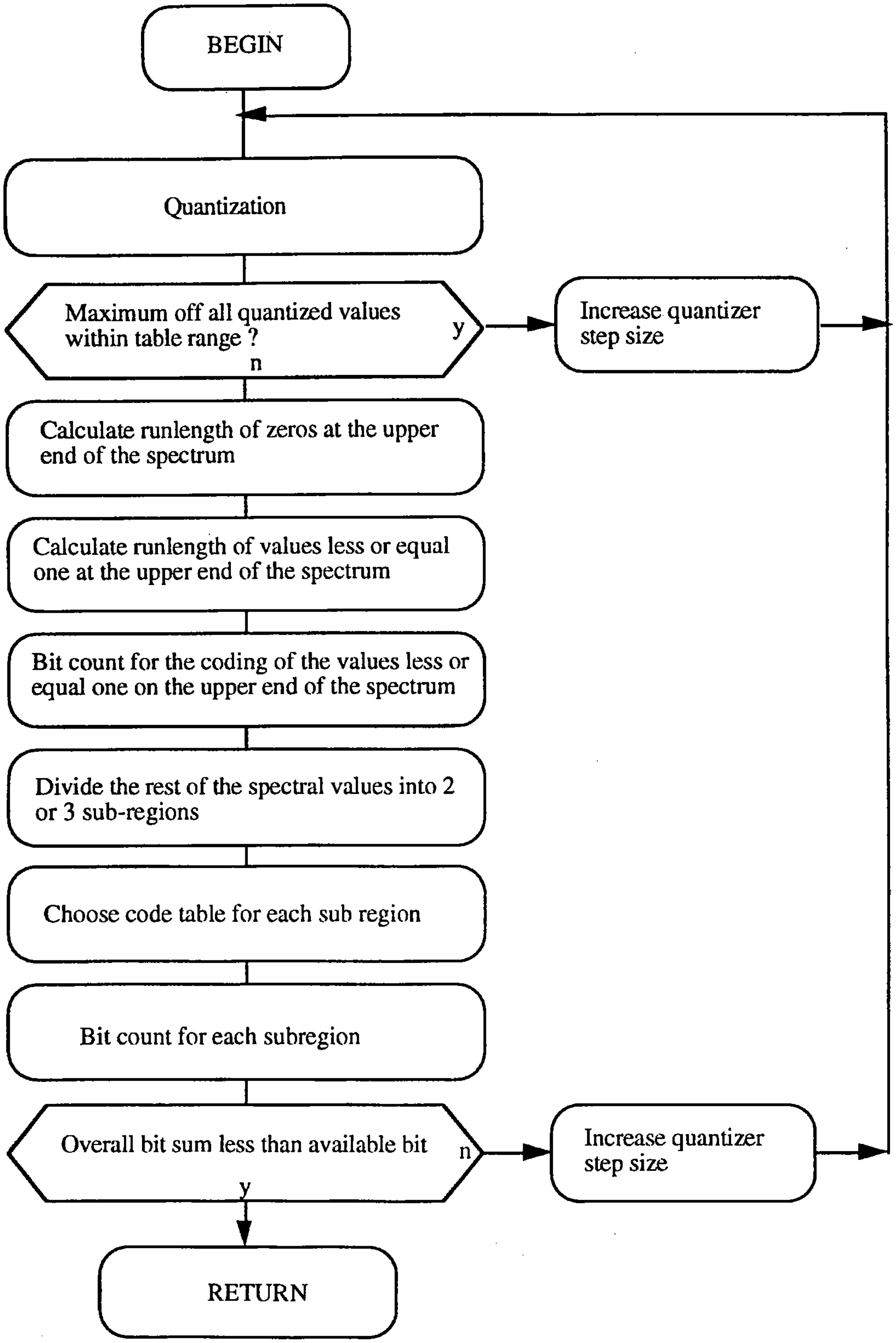

Figure C.9.c -- Layer III inner iteration loop

Annex D

(informative)

Psychoacoustic models

D.1. Psychoacoustic model 1

The calculation of the psychoacoustic model has to be adapted to the corresponding layer. This example is valid for Layers I and II. The model can be adapted to Layer III.

There is no principal difference in the application of psychoacoustic model 1 to Layer I or II.

Layer I: A new bit allocation is calculated for each block of 12 subband or 384 input PCM samples.

Layer II: A new bit allocation is calculated for three blocks totaling 36 subband samples corresponding to 3*384 (1 152) input PCM samples.

The bit allocation of the 32 subbands is calculated on the basis of the signal-to-mask ratios of all the subbands. Therefore, it is necessary to determine for each subband, the maximum signal level and the minimum masking threshold. The minimum masking threshold is derived from an FFT of the input PCM signal, followed by a psychoacoustic model calculation.

The FFT in parallel with the subband filter compensates for the lack of spectral selectivity obtained at low frequencies by the subband filterbank. This technique provides both a sufficient time resolution for the coded audio signal (Polyphase filter with optimized window for minimal pre-echoes) and a sufficient spectral resolution for the calculation of the masking thresholds. The frequencies and levels of aliasing distortions can be calculated. This is necessary for calculating a minimum bitrate for those subbands which need some bits to cancel the aliasing components in the decoder. The additional complexity to calculate the better frequency resolution is necessary only in the encoder, and introduces no additional delay or complexity in the decoder.

The calculation of the signal-to-mask-ratio is based on the following steps:

Step 1
- Calculation of the FFT for time to frequency conversion.

Step 2
- Determination of the sound pressure level in each subband.

Step 3
- Determination of the threshold in quiet (absolute threshold).

Step 4
- Finding of the tonal (more sinusoid-like) and non-tonal (more noise-like) components of the audio signal.

Step 5
- Decimation of the maskers, to obtain only the relevant maskers.

Step 6
- Calculation of the individual masking thresholds.

Step 7
- Determination of the global masking threshold.

Step 8
- Determination of the minimum masking threshold in each subband.

Step 9
- Calculation of the signal-to-mask ratio in each subband.

These steps will be further discussed. A sampling frequency of 48 kHz is assumed. For the other two sampling frequencies all frequencies mentioned should be scaled accordingly.

Step 1: FFT Analysis

The masking threshold is derived from an estimate of the power density spectrum that is calculated by a 512-point FFT for Layer I, or by a 1 024-point FFT for Layer II. The FFT is calculated directly from the input PCM signal, windowed by a Hann window.

For a coincidence in time between the bit allocation and the corresponding subband samples, the PCM-samples entering the FFT have to be delayed:

a) The delay of the analysis subband filter is 256 samples, corresponding to 5,3 ms at the 48 kHz sampling rate. A window shift of 256 samples is required to compensate for the delay in the analysis subband filter.

b) The Hann window must coincide with the subband samples of the frame. For Layer I this amounts to an additional window shift of 64 samples. For Layer II an additional window shift of minus 64 samples is required.

Technical data of the FFT:

		Layer I	Layer II
-	transform length	512 samples	1 024 samples
	Window size if fs = 48 kHz	10,67 ms	21,3 ms
	Window size if fs = 44,1 kHz	11,6 ms	23,2 ms
	Window size if fs = 32 kHz	16 ms	32 ms
-	Frequency resolution	sampling_frequency / 512	sampling_frequency / 1024

- Hann window, h(i):

$$h(i) = \sqrt{8/3} * 0{,}5 * \{1 - \cos[2 * \pi * (i)/N]\} \qquad 0 <= i <= N-1$$

- power density spectrum X(k):

$$X(k) = 10 * \log_{10} \left| \frac{1}{N} \sum_{l=0}^{N-1} h(l) * s(l) * e^{(-j*k*l*2*\pi/N)} \right|^2 \text{ dB} \qquad k = 0...N/2,$$

where s(l) is the input signal.

A normalization to the reference level of 96 dB SPL (Sound Pressure Level) has to be done in such a way that the maximum value corresponds to 96 dB.

Step 2: Determination of the sound pressure level

The sound pressure level L_{sb} in subband n is computed by:

$$L_{sb}(n) = \underset{X(k) \text{ in subband } n}{\mathrm{MAX}}[\, X(k),\ 20*\log(scf_{max}(n)*32\ 768)-10\,] \text{ dB}$$

where X(k) is the sound pressure level of the spectral line with index k of the FFT with the maximum amplitude in the frequency range corresponding to subband n. The expression $scf_{max}(n)$ is in Layer I the scalefactor, and in Layer II the maximum of the three scalefactors of subband n within a frame. The "-10 dB" term corrects for the difference between peak and RMS level. The sound pressure level $L_{sb}(n)$ is computed for every subband n.

The following alternative method of calculating $L_{sb}(n)$ offers a potential for better encoder performance, but this technique has not been subjected to a formal audio quality test.

The alternative sound pressure level L_{sb} in subband n is computed by:

$$L_{sb}(n) = \quad MAX[\, X_{spl}(n),\ 20*\log(scf_{max}(n)*32\,768)-10\,]\ \text{dB}$$

with

$$X_{spl}(n)=10*\log_{10}\Big(\sum_{\substack{k \\ k \text{ in subband } n}} 10^{X(k)/10}\Big)\ \text{dB}$$

where $X_{spl}(n)$ is the alternative sound pressure level corresponding to subband n.

Step 3: Considering the threshold in quiet

The threshold in quiet $LT_q(k)$, also called absolute threshold, is available in the tables "Frequencies, critical band rates and absolute threshold" (tables D.1a, D.1b, D.1c for Layer I; tables D.1d, D.1e, D.1f for Layer II). These tables depend on the sampling rate of the input PCM signal. Values are available for each sample in the frequency domain where the masking threshold is calculated. An offset depending on the overall bit rate is used for the absolute threshold. This offset is -12 dB for bit rates >= 96 kbits/s and 0 dB for bit rates < 96 kbits/s per channel.

Step 4: Finding of tonal and non-tonal components

The tonality of a masking component has an influence on the masking threshold. For this reason, it is worthwhile to discriminate between tonal and non-tonal components. For calculating the global masking threshold it is necessary to derive the tonal and the non-tonal components from the FFT spectrum.

This step starts with the determination of local maxima, then extracts tonal components (sinusoids) and calculates the intensity of the non-tonal components within a bandwidth of a critical band. The boundaries of the critical bands are given in the tables "Critical band boundaries" (tables D.2a, D.2b, D.2c for Layer I; tables D.2d, D.2e, D.2f for Layer II).

The bandwidth of the critical bands varies with the center frequency with a bandwidth of about only 0,1 kHz at low frequencies and with a bandwidth of about 4 kHz at high frequencies. It is known from psychoacoustic experiments that the ear has a better frequency resolution in the lower than in the higher frequency region. To determine if a local maximum may be a tonal component, a frequency range df around the local maximum is examined. The frequency range df is given by:

Sampling rate: 32 kHz

Layer I:	df = 125 Hz	0 kHz < f <= 4,0kHz
	df = 187,5 Hz	4,0 kHz < f <= 8,0 kHz
	df = 375 Hz	8,0 kHz < f <= 15,0kHz
Layer II:	df = 62,5 Hz	0 kHz < f <= 3,0 kHz
	df = 93,75 Hz	3,0 kHz < f <= 6,0 kHz
	df = 187,5 Hz	6,0 kHz < f <= 12,0 kHz
	df = 375 Hz	12,0 kHz < f <= 24,0 kHz

Sampling rate: 44,1kHz

Layer I:	df = 172,266 Hz	0 kHz < f <= 5,512kHz
	df = 281,25 Hz	5,512 kHz < f <= 11,024 kHz
	df = 562,50 Hz	11,024 kHz < f <= 19,982kHz
Layer II:	df = 86,133 Hz	0 kHz < f <= 2,756 kHz
	df = 129,199 Hz	2,756 kHz < f <= 5,512kHz
	df = 258,398 Hz	5,512 kHz < f <= 11,024 kHz
	df = 516,797 Hz	11,024 kHz < f <= 19,982kHz

Sampling rate: 48 kHz

Layer I:	df = 187,5 Hz	0 kHz	< f <=	6,0 kHz
	df = 281,25 Hz	6,0 kHz	< f <=	12,0 kHz
	df = 562,50 Hz	12,0 kHz	< f <=	24,0 kHz

Layer II:	df = 93,750 Hz	0 kHz	< f <=	3,0 kHz
	df = 140,63 Hz	3,0 kHz	< f <=	6,0 kHz
	df = 281,25 Hz	6,0 kHz	< f <=	12,0 kHz
	df = 562,50 Hz	12,0 kHz	< f <=	24,0 kHz

To make lists of the spectral lines X(k) that are tonal or non-tonal, the following three operations are performed:

a) Labelling of local maxima

A spectral line X(k) is labelled as a local maximum if

$$X(k) > X(k-1) \quad \text{and} \quad X(k) >= X(k+1)$$

b) Listing of tonal components and calculation of the sound pressure level

A local maximum is put in the list of tonal components if

$$X(k) - X(k+j) >= 7 \text{ dB},$$

where j is chosen according to

Layer I:

j = -2, +2	for	2 < k < 63
j = -3, -2, +2, +3	for	63 <= k < 127
j = -6,..., -2, +2,..., +6	for	127<= k <= 250

Layer II:

j = -2, +2	for	2 < k < 63
j = -3, -2, +2, +3	for	63 <= k < 127
j = -6,..., -2, +2,..., +6	for	127 <= k < 255
j = -12,..., -2, +2,..., +12	for	255<= k <= 500

If X(k) is found to be a tonal component, then the following parameters are listed:

- Index number k of the spectral line.
- Sound pressure level $X_{tm}(k) = 10 * \log_{10} \{ 10^{\frac{X(k-1)}{10}} + 10^{\frac{X(k)}{10}} + 10^{\frac{X(k+1)}{10}} \}$, in dB
- Tonal flag.

Next, all spectral lines within the examined frequency range are set to $-\infty$ dB.

c) Listing of non-tonal components and calculation of the power

The non-tonal (noise) components are calculated from the remaining spectral lines. To calculate the non-tonal components from these spectral lines X(k), the critical bands z(k) are determined using the tables, "Critical band boundaries" (tables D.2a, D.2b, D.2c for Layer I; tables D.2d, D.2e, D.2f for Layer II). In Layer I, 23 critical bands are used for the sampling rate of 32 kHz, 24 critical bands for 44,1 kHz and 25 critical bands are used for 48 kHz. In Layer II, 24 critical bands are used for 32 kHz sampling rate, and 26 critical bands are used for 44,1 kHz and 48 kHz sampling rate. Within each critical band, the power of the spectral lines (remaining after the tonal components have been zeroed) are summed to form the sound pressure level of the new non-tonal component $X_{nm}(k)$ corresponding to that critical band.

The following parameters are listed:
- Index number k of the spectral line nearest to the geometric mean of the critical band.
- Sound pressure level $X_{nm}(k)$ in dB.
- Non-tonal flag.

Step 5: Decimation of tonal and non-tonal masking components

Decimation is a procedure that is used to reduce the number of maskers which are considered for the calculation of the global masking threshold.

a) Tonal $X_{tm}(k)$ or non-tonal components $X_{nm}(k)$ are considered for the calculation of the masking threshold only if:

$$X_{tm}(k) >= LT_q(k) \quad \text{or} \quad X_{nm}(k) >= LT_q(k)$$

In this expression, $LT_q(k)$ is the absolute threshold (or threshold in quiet) at the frequency of index k. These values are given in tables D.1a, D.1b, D.1c for Layer I; tables D.1d, D.1e, D.1f for Layer II.

b) Decimation of two or more tonal components within a distance of less then 0,5 Bark: Keep the component with the highest power, and remove the smaller component(s) from the list of tonal components. For this operation, a sliding window in the critical band domain is used with a width of 0,5 Bark.

In the following, the index j is used to indicate the relevant tonal or non-tonal masking components from the combined decimated list.

Step 6: Calculation of individual masking thresholds

Of the original N/2 frequency domain samples, indexed by k, only a subset of the samples, indexed by i, are considered for the global masking threshold calculation. The samples used are shown in tables D.1a, D.1b, D.1c for Layer I; tables D.1d, D.1e, D.1f for Layer II.

Layer I:
For the frequency lines corresponding to the frequency region which is covered by the first six subbands no subsampling is used. For the frequency region corresponding to the next six subbands every second spectral line is considered. Finally, in the case of 44,1 kHz and 48 kHz sampling rates, in the frequency region corresponding to the remaining subbands, every fourth spectral line is considered up to 20 kHz. In the case of 32 kHz sampling rate, in the frequency region corresponding to the remaining subbands, every fourth spectral line is considered up to 15 kHz (see also tables D.1a, D.1b, D.1c for Layer I).

Layer II:
For the frequency lines corresponding to the frequency region which is covered by the first three subbands no subsampling is used. For the frequency region which is covered by next three subbands every second spectral line is considered. For the frequency region corresponding to the next six subbands every fourth spectral line is considered. Finally, in the case of 44,1 kHz and 48 kHz sampling rates, in the remaining subbands every eighth spectral line is considered up to 20 kHz. In the case of 32 kHz sampling rate, in the frequency region corresponding to the remaining subbands, every eighth spectral line is considered up to 15 kHz. (See also tables D.1d, D.1e, D.1f for Layer II).

The number of samples, n, in the subsampled frequency domain is different depending on the sampling rates and layers.

32 kHz sampling rate:	n = 108 for Layer I	and	n = 132 for Layer II
44,1 kHz sampling rate:	n = 106 for Layer I	and	n = 130 for Layer II
48 kHz sampling rate:	n = 102 for Layer I	and	n = 126 for Layer II

Every tonal and non-tonal component is assigned the value of the index i that most closely corresponds to the frequency of the original spectral line X(k). This index i is given in tables D.1a, D.1b, D.1c for Layer I; tables D.1d, D.1e, D.1f for Layer II.

The individual masking thresholds of both tonal and non-tonal components are given by the following expression:

$$LT_{tm}[z(j),z(i)] = X_{tm}[z(j)] + av_{tm}[z(j)] + vf[z(j),z(i)] \text{ dB}$$
$$LT_{nm}[z(j),z(i)] = X_{nm}[z(j)]+ av_{nm}[z(j)] + vf[z(j),z(i)] \text{ dB}$$

In this formula, LT_{tm} and LT_{nm} are the individual masking thresholds at critical band rate z in Bark of the masking component at the critical band rate of the masker z_m in Bark. The values in dB can be either positive or negative. The term $X_{tm}[z(j)]$ is the sound pressure level of the masking component with the index number j at the corresponding critical band rate z(j). The term av is called the masking index and vf the masking function of the masking component $X_{tm}[z(j)]$. The masking index av is different for tonal and non-tonal maskers (av_{tm} and av_{nm}).

For tonal maskers it is given by

$$av_{tm} = - 1{,}525 - 0{,}275 * z(j) - 4{,}5 \text{ dB},$$

and for non-tonal maskers

$$av_{nm} = - 1{,}525 - 0{,}175 * z(j) - 0{,}5 \text{ dB}.$$

The masking function vf of a masker is characterized by different lower and upper slopes, which depend on the distance in Bark dz = z(i)- z(j) to the masker. In this expression i is the index of the spectral line at which the masking function is calculated and j that of the masker. The critical band rates z(j) and z(i) can be found in tables D.1a, D.1b, D.1c for Layer I; tables D.1d, D.1e, D.1f for Layer II. The masking function, which is the same for tonal and non-tonal maskers, is given by:

$$vf = 17 * (dz + 1) - (0{,}4 * X[z(j)] + 6) \text{ dB} \qquad \text{for } -3 <= dz < -1 \text{ Bark}$$

$$vf = (0{,}4 * X[z(j)] + 6) * dz \text{ dB} \qquad \text{for } -1 <= dz < 0 \text{ Bark}$$

$$vf = - 17 * dz \text{ dB} \qquad \text{for } 0 <= dz < 1 \text{ Bark}$$

$$vf = - (dz -1) * (17 - 0{,}15 * X[z(j)]) - 17 \text{ dB} \qquad \text{for } 1 <= dz < 8 \text{ Bark}$$

In these expressions X[z(j)] is the sound pressure level of the j'th masking component in dB. For reasons of implementation complexity, the masking is no longer considered (LT_{tm} and LT_{nm} are set to -∞ dB outside this range) if dz < -3 Bark, or dz >= 8 Bark.

Step 7: Calculation of the global masking threshold LT_g

The global masking threshold $LT_g(i)$ at the i'th frequency sample is derived from the upper and lower slopes of the individual masking thresholds of each of the j tonal and non-tonal maskers and from the threshold in quiet $LT_q(i)$. This is also given in tables D.1a, D.1b, D.1c for Layer I; tables D.1d, D.1e, D.1f for Layer II. The global masking threshold is found by summing the powers corresponding to the individual masking thresholds and the threshold in quiet.

$$LT_g(i) = 10 \log_{10} \left(10^{LT_q(i)/10} + \sum_{j=1}^{m} 10^{LT_{tm}(z(j),z(i))/10} + \sum_{j=1}^{n} 10^{LT_{nm}(z(j),z(i))/10} \right)$$

The total number of tonal maskers is given by m, and the total number of non-tonal maskers is given by n. For a given i, the range of j can be reduced to just encompass those masking components that are within -8 to +3 Bark from i. Outside of this range LT_{tm} and LT_{nm} are -∞ dB.

Step 8: Determination of the minimum masking threshold

The minimum masking level LTmin(n) in subband n is determined by the following expression:

$$LT_{min}(n) = \underset{f(i) \text{ in subband } n}{\mathrm{MIN}} [\, LT_g(i) \,] \text{ dB}$$

where f(i) is the frequency of the i'th frequency sample. The f(i) are tabulated in tables D.1a, D.1b, D.1c for Layer I; tables D.1d, D.1e, D.1f for Layer II. A minimum masking level $LT_{min}(n)$ is computed for every subband.

Step 9: Calculation of the signal-to-mask-ratio

The signal-to-mask ratio

$$SMR_{sb}(n) = L_{sb}(n) - LT_{min}(n) \quad dB$$

is computed for every subband n.

Table D.1a. -- Frequencies, critical band rates and absolute threshold
Table is valid for Layer I at a sampling rate of 32 kHz.

Index Number i	Frequency [Hz]	Crit.Band Rate [z]	Absolute Thresh. [dB]
1	62,50	0,617	33,44
2	125,00	1,232	19,20
3	187,50	1,842	13,87
4	250,00	2,445	11,01
5	312,50	3,037	9,20
6	375,00	3,618	7,94
7	437,50	4,185	7,00
8	500,00	4,736	6,28
9	562,50	5,272	5,70
10	625,00	5,789	5,21
11	687,50	6,289	4,80
12	750,00	6,770	4,45
13	812,50	7,233	4,14
14	875,00	7,677	3,86
15	937,50	8,103	3,61
16	1 000,00	8,511	3,37
17	1 062,50	8,901	3,15
18	1 125,00	9,275	2,93
19	1 187,50	9,632	2,73
20	1 250,00	9,974	2,53
21	1 312,50	10,301	2,32
22	1 375,00	10,614	2,12
23	1 437,50	10,913	1,92
24	1 500,00	11,199	1,71
25	1 562,50	11,474	1,49
26	1 625,00	11,736	1,27
27	1 687,50	11,988	1,04
28	1 750,00	12,230	0,80
29	1 812,50	12,461	0,55
30	1 875,00	12,684	0,29
31	1 937,50	12,898	0,02
32	2 000,00	13,104	-0,25
33	2 062,50	13,302	-0,54
34	2 125,00	13,493	-0,83
35	2 187,50	13,678	-1,12
36	2 250,00	13,855	-1,43
37	2 312,50	14,027	-1,73
38	2 375,00	14,193	-2,04
39	2 437,50	14,354	-2,34
40	2 500,00	14,509	-2,64
41	2 562,50	14,660	-2,93
42	2 625,00	14,807	-3,22
43	2 687,50	14,949	-3,49
44	2 750,00	15,087	-3,74
45	2 812,50	15,221	-3,98
46	2 875,00	15,351	-4,20
47	2 937,50	15,478	-4,40
48	3 000,00	15,602	-4,57
49	3 125,00	15,841	-4,82
50	3 250,00	16,069	-4,96
51	3 375,00	16,287	-4,97
52	3 500,00	16,496	-4,86
53	3 625,00	16,697	-4,63
54	3 750,00	16,891	-4,29
55	3 875,00	17,078	-3,87
56	4 000,00	17,259	-3,39
57	4 125,00	17,434	-2,86
58	4 250,00	17,605	-2,31
59	4 375,00	17,770	-1,77
60	4 500,00	17,932	-1,24
61	4 625,00	18,089	-0,74
62	4 750,00	18,242	-0,29
63	4 875,00	18,392	0,12
64	5 000,00	18,539	0,48
65	5 125,00	18,682	0,79
66	5 250,00	18,823	1,06
67	5 375,00	18,960	1,29
68	5 500,00	19,095	1,49
69	5 625,00	19,226	1,66
70	5 750,00	19,356	1,81
71	5 875,00	19,482	1,95
72	6 000,00	19,606	2,08
73	6 250,00	19,847	2,33
74	6 500,00	20,079	2,59
75	6 750,00	20,300	2,86
76	7 000,00	20,513	3,17
77	7 250,00	20,717	3,51
78	7 500,00	20,912	3,89
79	7 750,00	21,098	4,31
80	8 000,00	21,275	4,79
81	8 250,00	21,445	5,31
82	8 500,00	21,606	5,88
83	8 750,00	21,760	6,50
84	9 000,00	21,906	7,19
85	9 250,00	22,046	7,93
86	9 500,00	22,178	8,75
87	9 750,00	22,304	9,63
88	10 000,00	22,424	10,58
89	10 250,00	22,538	11,60
90	10 500,00	22,646	12,71
91	10 750,00	22,749	13,90
92	11 000,00	22,847	15,18
93	11 250,00	22,941	16,54
94	11 500,00	23,030	18,01
95	11 750,00	23,114	19,57
96	12 000,00	23,195	21,23
97	12 250,00	23,272	23,01
98	12 500,00	23,345	24,90
99	12 750,00	23,415	26,90
100	13 000,00	23,482	29,03
101	13 250,00	23,546	31,28
102	13 500,00	23,607	33,67
103	13 750,00	23,666	36,19
104	14 000,00	23,722	38,86
105	14 250,00	23,775	41,67
106	14 500,00	23,827	44,63
107	14 750,00	23,876	47,76
108	15 000,00	23,923	51,04

Table D.1b. -- Frequencies, critical band rates and absolute threshold
Table is valid for Layer I at a sampling rate of 44,1 kHz.

Index Number i	Frequency [Hz]	Crit.Band Rate [z]	Absolute Thresh. [dB]
1	86,13	0,850	25,87
2	172,27	1,694	14,85
3	258,40	2,525	10,72
4	344,53	3,337	8,50
5	430,66	4,124	7,10
6	516,80	4,882	6,11
7	602,93	5,608	5,37
8	689,06	6,301	4,79
9	775,20	6,959	4,32
10	861,33	7,581	3,92
11	947,46	8,169	3,57
12	1 033,59	8,723	3,25
13	1 119,73	9,244	2,95
14	1 205,86	9,734	2,67
15	1 291,99	10,195	2,39
16	1 378,13	10,629	2,11
17	1 464,26	11,037	1,83
18	1 550,39	11,421	1,53
19	1 636,52	11,783	1,23
20	1 722,66	12,125	0,90
21	1 808,79	12,448	0,56
22	1 894,92	12,753	0,21
23	1 981,05	13,042	-0,17
24	2 067,19	13,317	-0,56
25	2 153,32	13,578	-0,96
26	2 239,45	13,826	-1,38
27	2 325,59	14,062	-1,79
28	2 411,72	14,288	-2,21
29	2 497,85	14,504	-2,63
30	2 583,98	14,711	-3,03
31	2 670,12	14,909	-3,41
32	2 756,25	15,100	-3,77
33	2 842,38	15,284	-4,09
34	2 928,52	15,460	-4,37
35	3 014,65	15,631	-4,60
36	3 100,78	15,796	-4,78
37	3 186,91	15,955	-4,91
38	3 273,05	16,110	-4,97
39	3 359,18	16,260	-4,98
40	3 445,31	16,406	-4,92
41	3 531,45	16,547	-4,81
42	3 617,58	16,685	-4,65
43	3 703,71	16,820	-4,43
44	3 789,84	16,951	-4,17
45	3 875,98	17,079	-3,87
46	3 962,11	17,205	-3,54
47	4 048,24	17,327	-3,19
48	4 134,38	17,447	-2,82
49	4 306,64	17,680	-2,06
50	4 478,91	17,905	-1,32
51	4 651,17	18,121	-0,64
52	4 823,44	18,331	-0,04
53	4 995,70	18,534	0,47
54	5 167,97	18,731	0,89
55	5 340,23	18,922	1,23
56	5 512,50	19,108	1,51
57	5 684,77	19,289	1,74
58	5 857,03	19,464	1,93
59	6 029,30	19,635	2,11
60	6 201,56	19,801	2,28
61	6 373,83	19,963	2,46
62	6 546,09	20,120	2,63
63	6 718,36	20,273	2,82
64	6 890,63	20,421	3,03
65	7 062,89	20,565	3,25
66	7 235,16	20,705	3,49
67	7 407,42	20,840	3,74
68	7 579,69	20,972	4,02
69	7 751,95	21,099	4,32
70	7 924,22	21,222	4,64
71	8 096,48	21,342	4,98
72	8 268,75	21,457	5,35
73	8 613,28	21,677	6,15
74	8 957,81	21,882	7,07
75	9 302,34	22,074	8,10
76	9 646,88	22,253	9,25
77	9 991,41	22,420	10,54
78	10 335,94	22,576	11,97
79	10 680,47	22,721	13,56
80	11 025,00	22,857	15,31
81	11 369,53	22,984	17,23
82	11 714,06	23,102	19,34
83	12 058,59	23,213	21,64
84	12 403,13	23,317	24,15
85	12 747,66	23,415	26,88
86	13 092,19	23,506	29,84
87	13 436,72	23,592	33,05
88	13 781,25	23,673	36,52
89	14 125,78	23,749	40,25
90	14 470,31	23,821	44,27
91	14 814,84	23,888	48,59
92	15 159,38	23,952	53,22
93	15 503,91	24,013	58,18
94	15 848,44	24,070	63,49
95	16 192,97	24,125	68,00
96	16 537,50	24,176	68,00
97	16 882,03	24,225	68,00
98	17 226,56	24,271	68,00
99	17 571,09	24,316	68,00
100	17 915,63	24,358	68,00
101	18 260,16	24,398	68,00
102	18 604,69	24,436	68,00
103	18 949,22	24,473	68,00
104	19 293,75	24,508	68,00
105	19 638,28	24,542	68,00
106	19 982,81	24,574	68,00

Table D.1c. -- Frequencies, critical band rates and absolute threshold
Table is valid for Layer I at a sampling rate of 48 kHz.

Index Number i	Frequency [Hz]	Crit.Band Rate [z]	Absolute Thresh. [dB]
1	93,75	0,925	24,17
2	187,50	1,842	13,87
3	281,25	2,742	10,01
4	375,00	3,618	7,94
5	468,75	4,463	6,62
6	562,50	5,272	5,70
7	656,25	6,041	5,00
8	750,00	6,770	4,45
9	843,75	7,457	4,00
10	937,50	8,103	3,61
11	1 031,25	8,708	3,26
12	1 125,00	9,275	2,93
13	1 218,75	9,805	2,63
14	1 312,50	10,301	2,32
15	1 406,25	10,765	2,02
16	1 500,00	11,199	1,71
17	1 593,75	11,606	1,38
18	1 687,50	11,988	1,04
19	1 781,25	12,347	0,67
20	1 875,00	12,684	0,29
21	1 968,75	13,002	-0,11
22	2 062,50	13,302	-0,54
23	2 156,25	13,586	-0,97
24	2 250,00	13,855	-1,43
25	2 343,75	14,111	-1,88
26	2 437,50	14,354	-2,34
27	2 531,25	14,585	-2,79
28	2 625,00	14,807	-3,22
29	2 718,75	15,018	-3,62
30	2 812,50	15,221	-3,98
31	2 906,25	15,415	-4,30
32	3 000,00	15,602	-4,57
33	3 093,75	15,783	-4,77
34	3 187,50	15,956	-4,91
35	3 281,25	16,124	-4,98
36	3 375,00	16,287	-4,97
37	3 468,75	16,445	-4,90
38	3 562,50	16,598	-4,76
39	3 656,25	16,746	-4,55
40	3 750,00	16,891	-4,29
41	3 843,75	17,032	-3,99
42	3 937,50	17,169	-3,64
43	4 031,25	17,303	-3,26
44	4 125,00	17,434	-2,86
45	4 218,75	17,563	-2,45
46	4 312,50	17,688	-2,04
47	4 406,25	17,811	-1,63
48	4 500,00	17,932	-1,24
49	4 687,50	18,166	-0,51
50	4 875,00	18,392	0,12
51	5 062,50	18,611	0,64
52	5 250,00	18,823	1,06
53	5 437,50	19,028	1,39
54	5 625,00	19,226	1,66
55	5 812,50	19,419	1,88
56	6 000,00	19,606	2,08
57	6 187,50	19,788	2,27
58	6 375,00	19,964	2,46
59	6 562,50	20,135	2,65
60	6 750,00	20,300	2,86
61	6 937,50	20,461	3,09
62	7 125,00	20,616	3,33
63	7 312,50	20,766	3,60
64	7 500,00	20,912	3,89
65	7 687,50	21,052	4,20
66	7 875,00	21,188	4,54
67	8 062,50	21,318	4,91
68	8 250,00	21,445	5,31
69	8 437,50	21,567	5,73
70	8 625,00	21,684	6,18
71	8 812,50	21,797	6,67
72	9 000,00	21,906	7,19
73	9 375,00	22,113	8,33
74	9 750,00	22,304	9,63
75	10 125,00	22,482	11,08
76	10 500,00	22,646	12,71
77	10 875,00	22,799	14,53
78	11 250,00	22,941	16,54
79	11 625,00	23,072	18,77
80	12 000,00	23,195	21,23
81	12 375,00	23,309	23,94
82	12 750,00	23,415	26,90
83	13 125,00	23,515	30,14
84	13 500,00	23,607	33,67
85	13 875,00	23,694	37,51
86	14 250,00	23,775	41,67
87	14 625,00	23,852	46,17
88	15 000,00	23,923	51,04
89	15 375,00	23,991	56,29
90	15 750,00	24,054	61,94
91	16 125,00	24,114	68,00
92	16 500,00	24,171	68,00
93	16 875,00	24,224	68,00
94	17 250,00	24,275	68,00
95	17 625,00	24,322	68,00
96	18 000,00	24,368	68,00
97	18 375,00	24,411	68,00
98	18 750,00	24,452	68,00
99	19 125,00	24,491	68,00
100	19 500,00	24,528	68,00
101	19 875,00	24,564	68,00
102	20 250,00	24,597	68,00

Table D.1d. -- Frequencies, critical band rates and absolute threshold
Table is valid for Layer II at a sampling rate of 32 kHz.

Index Number i	Frequency [Hz]	Crit.Band Rate [z]	Absolute Thresh. [dB]
1	31,25	0,309	58,23
2	62,50	0,617	33,44
3	93,75	0,925	24,17
4	125,00	1,232	19,20
5	156,25	1,538	16,05
6	187,50	1,842	13,87
7	218,75	2,145	12,26
8	250,00	2,445	11,01
9	281,25	2,742	10,01
10	312,50	3,037	9,20
11	343,75	3,329	8,52
12	375,00	3,618	7,94
13	406,25	3,903	7,44
14	437,50	4,185	7,00
15	468,75	4,463	6,62
16	500,00	4,736	6,28
17	531,25	5,006	5,97
18	562,50	5,272	5,70
19	593,75	5,533	5,44
20	625,00	5,789	5,21
21	656,25	6,041	5,00
22	687,50	6,289	4,80
23	718,75	6,532	4,62
24	750,00	6,770	4,45
25	781,25	7,004	4,29
26	812,50	7,233	4,14
27	843,75	7,457	4,00
28	875,00	7,677	3,86
29	906,25	7,892	3,73
30	937,50	8,103	3,61
31	968,75	8,309	3,49
32	1 000,00	8,511	3,37
33	1 031,25	8,708	3,26
34	1 062,50	8,901	3,15
35	1 093,75	9,090	3,04
36	1 125,00	9,275	2,93
37	1 156,25	9,456	2,83
38	1 187,50	9,632	2,73
39	1 218,75	9,805	2,63
40	1 250,00	9,974	2,53
41	1 281,25	10,139	2,42
42	1 312,50	10,301	2,32
43	1 343,75	10,459	2,22
44	1 375,00	10,614	2,12
45	1 406,25	10,765	2,02
46	1 437,50	10,913	1,92
47	1 468,75	11,058	1,81
48	1 500,00	11,199	1,71
49	1 562,50	11,474	1,49
50	1 625,00	11,736	1,27
51	1 687,50	11,988	1,04
52	1 750,00	12,230	0,80
53	1 812,50	12,461	0,55
54	1 875,00	12,684	0,29
55	1 937,50	12,898	0,02
56	2 000,00	13,104	-0,25
57	2 062,50	13,302	-0,54
58	2 125,00	13,493	-0,83
59	2 187,50	13,678	-1,12
60	2 250,00	13,855	-1,43
61	2 312,50	14,027	-1,73
62	2 375,00	14,193	-2,04
63	2 437,50	14,354	-2,34
64	2 500,00	14,509	-2,64
65	2 562,50	14,660	-2,93
66	2 625,00	14,807	-3,22
67	2 687,50	14,949	-3,49
68	2 750,00	15,087	-3,74
69	2 812,50	15,221	-3,98
70	2 875,00	15,351	-4,20
71	2 937,50	15,478	-4,40
72	3 000,00	15,602	-4,57
73	3 125,00	15,841	-4,82
74	3 250,00	16,069	-4,96
75	3 375,00	16,287	-4,97
76	3 500,00	16,496	-4,86
77	3 625,00	16,697	-4,63
78	3 750,00	16,891	-4,29
79	3 875,00	17,078	-3,87
80	4 000,00	17,259	-3,39
81	4 125,00	17,434	-2,86
82	4 250,00	17,605	-2,31
83	4 375,00	17,770	-1,77
84	4 500,00	17,932	-1,24
85	4 625,00	18,089	-0,74
86	4 750,00	18,242	-0,29
87	4 875,00	18,392	0,12
88	5 000,00	18,539	0,48
89	5 125,00	18,682	0,79
90	5 250,00	18,823	1,06
91	5 375,00	18,960	1,29
92	5 500,00	19,095	1,49
93	5 625,00	19,226	1,66
94	5 750,00	19,356	1,81
95	5 875,00	19,482	1,95
96	6 000,00	19,606	2,08
97	6 250,00	19,847	2,33
98	6 500,00	20,079	2,59
99	6 750,00	20,300	2,86
100	7 000,00	20,513	3,17
101	7 250,00	20,717	3,51
102	7 500,00	20,912	3,89
103	7 750,00	21,098	4,31
104	8 000,00	21,275	4,79
105	8 250,00	21,445	5,31
106	8 500,00	21,606	5,88
107	8 750,00	21,760	6,50
108	9 000,00	21,906	7,19
109	9 250,00	22,046	7,93
110	9 500,00	22,178	8,75
111	9 750,00	22,304	9,63
112	10 000,00	22,424	10,58
113	10 250,00	22,538	11,60
114	10 500,00	22,646	12,71
115	10 750,00	22,749	13,90
116	11 000,00	22,847	15,18
117	11 250,00	22,941	16,54
118	11 500,00	23,030	18,01
119	11 750,00	23,114	19,57
120	12 000,00	23,195	21,23
121	12 250,00	23,272	23,01
122	12 500,00	23,345	24,90
123	12 750,00	23,415	26,90
124	13 000,00	23,482	29,03
125	13 250,00	23,546	31,28
126	13 500,00	23,607	33,67
127	13 750,00	23,666	36,19
128	14 000,00	23,722	38,86
129	14 250,00	23,775	41,67
130	14 500,00	23,827	44,63
131	14 750,00	23,876	47,76
132	15 000,00	23,923	51,04

Table D.1e. -- Frequencies, Critical Band Rates and Absolute Threshold
Table is valid for Layer II at a sampling rate of 44,1 kHz.

Index Number i	Frequency [Hz]	Crit.Band Rate [z]	Absolute Thresh. [dB]
1	43,07	0,425	45,05
2	86,13	0,850	25,87
3	129,20	1,273	18,70
4	172,27	1,694	14,85
5	215,33	2,112	12,41
6	258,40	2,525	10,72
7	301,46	2,934	9,47
8	344,53	3,337	8,50
9	387,60	3,733	7,73
10	430,66	4,124	7,10
11	473,73	4,507	6,56
12	516,80	4,882	6,11
13	559,86	5,249	5,72
14	602,93	5,608	5,37
15	646,00	5,959	5,07
16	689,06	6,301	4,79
17	732,13	6,634	4,55
18	775,20	6,959	4,32
19	818,26	7,274	4,11
20	861,33	7,581	3,92
21	904,39	7,879	3,74
22	947,46	8,169	3,57
23	990,53	8,450	3,40
24	1 033,59	8,723	3,25
25	1 076,66	8,987	3,10
26	1 119,73	9,244	2,95
27	1 162,79	9,493	2,81
28	1 205,86	9,734	2,67
29	1 248,93	9,968	2,53
30	1 291,99	10,195	2,39
31	1 335,06	10,416	2,25
32	1 378,13	10,629	2,11
33	1 421,19	10,836	1,97
34	1 464,26	11,037	1,83
35	1 507,32	11,232	1,68
36	1 550,39	11,421	1,53
37	1 593,46	11,605	1,38
38	1 636,52	11,783	1,23
39	1 679,59	11,957	1,07
40	1 722,66	12,125	0,90
41	1 765,72	12,289	0,74
42	1 808,79	12,448	0,56
43	1 851,86	12,603	0,39
44	1 894,92	12,753	0,21
45	1 937,99	12,900	0,02
46	1 981,05	13,042	-0,17
47	2 024,12	13,181	-0,36
48	2 067,19	13,317	-0,56
49	2 153,32	13,578	-0,96
50	2 239,45	13,826	-1,38
51	2 325,59	14,062	-1,79
52	2 411,72	14,288	-2,21
53	2 497,85	14,504	-2,63
54	2 583,98	14,711	-3,03
55	2 670,12	14,909	-3,41
56	2 756,25	15,100	-3,77
57	2 842,38	15,284	-4,09
58	2 928,52	15,460	-4,37
59	3 014,65	15,631	-4,60
60	3 100,78	15,796	-4,78
61	3 186,91	15,955	-4,91
62	3 273,05	16,110	-4,97
63	3 359,18	16,260	-4,98
64	3 445,31	16,406	-4,92
65	3 531,45	16,547	-4,81
66	3 617,58	16,685	-4,65
67	3 703,71	16,820	-4,43
68	3 789,84	16,951	-4,17
69	3 875,98	17,079	-3,87
70	3 962,11	17,205	-3,54
71	4 048,24	17,327	-3,19
72	4 134,38	17,447	-2,82
73	4 306,64	17,680	-2,06
74	4 478,91	17,905	-1,32
75	4 651,17	18,121	-0,64
76	4 823,44	18,331	-0,04
77	4 995,70	18,534	0,47
78	5 167,97	18,731	0,89
79	5 340,23	18,922	1,23
80	5 512,50	19,108	1,51
81	5 684,77	19,289	1,74
82	5 857,03	19,464	1,93
83	6 029,30	19,635	2,11
84	6 201,56	19,801	2,28
85	6 373,83	19,963	2,46
86	6 546,09	20,120	2,63
87	6 718,36	20,273	2,82
88	6 890,63	20,421	3,03
89	7 062,89	20,565	3,25
90	7 235,16	20,705	3,49
91	7 407,42	20,840	3,74
92	7 579,69	20,972	4,02
93	7 751,95	21,099	4,32
94	7 924,22	21,222	4,64
95	8 096,48	21,342	4,98
96	8 268,75	21,457	5,35
97	8 613,28	21,677	6,15
98	8 957,81	21,882	7,07
99	9 302,34	22,074	8,10
100	9 646,88	22,253	9,25
101	9 991,41	22,420	10,54
102	10 335,94	22,576	11,97
103	10 680,47	22,721	13,56
104	11 025,00	22,857	15,31
105	11 369,53	22,984	17,23
106	11 714,06	23,102	19,34
107	12 058,59	23,213	21,64
108	12 403,13	23,317	24,15
109	12 747,66	23,415	26,88
110	13 092,19	23,506	29,84
111	13 436,72	23,592	33,05
112	13 781,25	23,673	36,52
113	14 125,78	23,749	40,25
114	14 470,31	23,821	44,27
115	14 814,84	23,888	48,59
116	15 159,38	23,952	53,22
117	15 503,91	24,013	58,18
118	15 848,44	24,070	63,49
119	16 192,97	24,125	68,00
120	16 537,50	24,176	68,00
121	16 882,03	24,225	68,00
122	17 226,56	24,271	68,00
123	17 571,09	24,316	68,00
124	17 915,63	24,358	68,00
125	18 260,16	24,398	68,00
126	18 604,69	24,436	68,00
127	18 949,22	24,473	68,00
128	19 293,75	24,508	68,00
129	19 638,28	24,542	68,00
130	19 982,81	24,574	68,00

Table D.1f. -- Frequencies, critical band rates and absolute threshold
Table is valid for Layer II at a sampling rate of 48 kHz

Index Number i	Frequency [Hz]	Crit.Band Rate [z]	Absolute Thresh. [dB]
1	46,88	0,463	42,10
2	93,75	0,925	24,17
3	140,63	1,385	17,47
4	187,50	1,842	13,87
5	234,38	2,295	11,60
6	281,25	2,742	10,01
7	328,13	3,184	8,84
8	375,00	3,618	7,94
9	421,88	4,045	7,22
10	468,75	4,463	6,62
11	515,63	4,872	6,12
12	562,50	5,272	5,70
13	609,38	5,661	5,33
14	656,25	6,041	5,00
15	703,13	6,411	4,71
16	750,00	6,770	4,45
17	796,88	7,119	4,21
18	843,75	7,457	4,00
19	890,63	7,785	3,79
20	937,50	8,103	3,61
21	984,38	8,410	3,43
22	1 031,25	8,708	3,26
23	1 078,13	8,996	3,09
24	1 125,00	9,275	2,93
25	1 171,88	9,544	2,78
26	1 218,75	9,805	2,63
27	1 265,63	10,057	2,47
28	1 312,50	10,301	2,32
29	1 359,38	10,537	2,17
30	1 406,25	10,765	2,02
31	1 453,13	10,986	1,86
32	1 500,00	11,199	1,71
33	1 546,88	11,406	1,55
34	1 593,75	11,606	1,38
35	1 640,63	11,800	1,21
36	1 687,50	11,988	1,04
37	1 734,38	12,170	0,86
38	1 781,25	12,347	0,67
39	1 828,13	12,518	0,49
40	1 875,00	12,684	0,29
41	1 921,88	12,845	0,09
42	1 968,75	13,002	-0,11
43	2 015,63	13,154	-0,32
44	2 062,50	13,302	-0,54
45	2 109,38	13,446	-0,75
46	2 156,25	13,586	-0,97
47	2 203,13	13,723	-1,20
48	2 250,00	13,855	-1,43
49	2 343,75	14,111	-1,88
50	2 437,50	14,354	-2,34
51	2 531,25	14,585	-2,79
52	2 625,00	14,807	-3,22
53	2 718,75	15,018	-3,62
54	2 812,50	15,221	-3,98
55	2 906,25	15,415	-4,30
56	3 000,00	15,602	-4,57
57	3 093,75	15,783	-4,77
58	3 187,50	15,956	-4,91
59	3 281,25	16,124	-4,98
60	3 375,00	16,287	-4,97
61	3 468,75	16,445	-4,90
62	3 562,50	16,598	-4,76
63	3 656,25	16,746	-4,55
64	3 750,00	16,891	-4,29
65	3 843,75	17,032	-3,99
66	3 937,50	17,169	-3,64
67	4 031,25	17,303	-3,26
68	4 125,00	17,434	-2,86
69	4 218,75	17,563	-2,45
70	4 312,50	17,688	-2,04
71	4 406,25	17,811	-1,63
72	4 500,00	17,932	-1,24
73	4 687,50	18,166	-0,51
74	4 875,00	18,392	0,12
75	5 062,50	18,611	0,64
76	5 250,00	18,823	1,06
77	5 437,50	19,028	1,39
78	5 625,00	19,226	1,66
79	5 812,50	19,419	1,88
80	6 000,00	19,606	2,08
81	6 187,50	19,788	2,27
82	6 375,00	19,964	2,46
83	6 562,50	20,135	2,65
84	6 750,00	20,300	2,86
85	6 937,50	20,461	3,09
86	7 125,00	20,616	3,33
87	7 312,50	20,766	3,60
88	7 500,00	20,912	3,89
89	7 687,50	21,052	4,20
90	7 875,00	21,188	4,54
91	8 062,50	21,318	4,91
92	8 250,00	21,445	5,31
93	8 437,50	21,567	5,73
94	8 625,00	21,684	6,18
95	8 812,50	21,797	6,67
96	9 000,00	21,906	7,19
97	9 375,00	22,113	8,33
98	9 750,00	22,304	9,63
99	10 125,00	22,482	11,08
100	10 500,00	22,646	12,71
101	10 875,00	22,799	14,53
102	11 250,00	22,941	16,54
103	11 625,00	23,072	18,77
104	12 000,00	23,195	21,23
105	12 375,00	23,309	23,94
106	12 750,00	23,415	26,90
107	13 125,00	23,515	30,14
108	13 500,00	23,607	33,67
109	13 875,00	23,694	37,51
110	14 250,00	23,775	41,67
111	14 625,00	23,852	46,17
112	15 000,00	23,923	51,04
113	15 375,00	23,991	56,29
114	15 750,00	24,054	61,94
115	16 125,00	24,114	68,00
116	16 500,00	24,171	68,00
117	16 875,00	24,224	68,00
118	17 250,00	24,275	68,00
119	17 625,00	24,322	68,00
120	18 000,00	24,368	68,00
121	18 375,00	24,411	68,00
122	18 750,00	24,452	68,00
123	19 125,00	24,491	68,00
124	19 500,00	24,528	68,00
125	19 875,00	24,564	68,00
126	20 250,00	24,597	68,00

Table D.2a. -- Critical band boundaries
This table is valid for Layer I at a sampling rate of 32 kHz.
The frequencies represent the top end of each critical band.

no	index of Table F&CB	frequency [Hz]	Bark [z]
0	1	62,500	0,617
1	3	187,500	1,842
2	5	312,500	3,037
3	7	437,500	4,185
4	9	562,500	5,272
5	11	687,500	6,289
6	13	812,500	7,233
7	15	937,500	8,103
8	18	1 125,000	9,275
9	21	1 312,500	10,301
10	24	1 500,000	11,199
11	27	1 687,500	11,988
12	32	2 000,000	13,104
13	37	2 312,500	14,027
14	44	2 750,000	15,087
15	50	3 250,000	16,069
16	55	3 875,000	17,078
17	61	4 625,000	18,089
18	68	5 500,000	19,095
19	74	6 500,000	20,079
20	79	7 750,000	21,098
21	85	9 250,000	22,046
22	94	11 500,000	23,030
23	108	15 000,000	23,923

Table D.2b. -- Critical band boundaries
This table is valid for Layer I at a sampling rate of 44,1 kHz.
The frequencies represent the top end of each critical band.

no	index of Table F&CB	frequency [Hz]	Bark [z]
0	1	86,133	0,850
1	2	172,266	1,694
2	3	258,398	2,525
3	5	430,664	4,124
4	6	516,797	4,882
5	8	689,063	6,301
6	9	775,195	6,959
7	11	947,461	8,169
8	13	1 119,727	9,244
9	15	1 291,992	10,195
10	17	1 464,258	11,037
11	20	1 722,656	12,125
12	23	1 981,055	13,042
13	27	2 325,586	14,062
14	32	2 756,250	15,100
15	37	3 186,914	15,955
16	45	3 875,977	17,079
17	50	4 478,906	17,904
18	55	5 340,234	18,922
19	61	6 373,828	19,963
20	68	7 579,688	20,971
21	75	9 302,344	22,074
22	81	11 369,531	22,984
23	93	15 503,906	24,013
24	106	19 982,813	24,573

Table D.2c. -- Critical band boundaries
This table is valid for Layer I at a sampling rate of 48 kHz.
The frequencies represent the top end of each critical band.

no	indexof TableF&CB	frequency[Hz]	Bark[z]
0	1	93,750	0,925
1	2	187,500	1,842
2	3	281,250	2,742
3	4	375,000	3,618
4	5	468,750	4,463
5	6	562,500	5,272
6	7	656,250	6,041
7	9	843,750	7,457
8	10	937,500	8,103
9	12	1 125,000	9,275
10	14	1 312,500	10,301
11	16	1 500,000	11,199
12	19	1 781,250	12,347
13	21	1 968,750	13,002
14	25	2 343,750	14,111
15	29	2 718,750	15,018
16	35	3 281,250	16,124
17	41	3 843,750	17,032
18	49	4 687,500	18,166
19	53	5 437,500	19,028
20	58	6 375,000	19,964
21	65	7 687,500	21,052
22	73	9 375,000	22,113
23	79	11 625,000	23,072
24	89	15 375,000	23,991
25	102	20 250,000	24,597

Table D.2d. -- Critical band boundaries
This tablc is valid for Layer II at a sampling rate of 32 kHz.
The frequencies represent the top end of each critical band.

no	indexof TableF&CB	frequency[Hz]	Bark[z]
0	1	31,250	0,309
1	3	93,750	0,925
2	6	187,500	1,842
3	10	312,500	3,037
4	13	406,250	3,903
5	17	531,250	5,006
6	21	656,250	6,041
7	25	781,250	7,004
8	30	937,500	8,103
9	35	1 093,750	9,090
10	41	1 281,250	10,139
11	47	1 468,750	11,058
12	51	1 687,500	11,988
13	56	2 000,000	13,104
14	61	2 312,500	14,027
15	68	2 750,000	15,087
16	74	3 250,000	16,069
17	79	3 875,000	17,078
18	85	4 625,000	18,089
19	92	5 500,000	19,095
20	98	6 500,000	20,079
21	103	7 750,000	21,098
22	109	9 250,000	22,046
23	118	11 500,000	23,030
24	132	15 000,000	23,923

Table D.2e. -- Critical band boundaries

This table is valid for Layer II at a sampling rate of 44,1 kHz.
The frequencies represent the top end of each critical band.

no	indexof TableF&CB	frequency[Hz]	Bark[z]
0	1	43,066	0,425
1	2	86,133	0,850
2	3	129,199	1,273
3	5	215,332	2,112
4	7	301,465	2,934
5	10	430,664	4,124
6	13	559,863	5,249
7	16	689,063	6,301
8	19	818,262	7,274
9	22	947,461	8,169
10	26	1 119,727	9,244
11	30	1 291,992	10,195
12	35	1 507,324	11,232
13	40	1 722,656	12,125
14	46	1 981,055	13,042
15	51	2 325,586	14,062
16	56	2 756,250	15,100
17	62	3 273,047	16,11
18	69	3 875,977	17,079
19	74	4 478,906	17,904
20	79	5 340,234	18,922
21	85	6 373,828	19,963
22	92	7 579,688	20,971
23	99	9 302,344	22,074
24	105	11 369,531	22,984
25	117	15 503,906	24,013
26	130	19 982,813	24,573

Table D.2f. -- Critical band boundaries
This table is valid for Layer II at a sampling rate of 48 kHz.
The frequencies represent the top end of each critical band.

no	index of Table F&CB	frequency [Hz]	Bark [z]
0	1	46,875	0,463
1	2	93,750	0,925
2	3	140,625	1,385
3	5	234,375	2,295
4	7	328,125	3,184
5	9	421,875	4,045
6	12	562,500	5,272
7	14	656,250	6,041
8	17	796,875	7,119
9	20	937,500	8,103
10	24	1 125,000	9,275
11	27	1 265,625	10,057
12	32	1 500,000	11,199
13	37	1 734,375	12,170
14	42	1 968,750	13,002
15	49	2 343,750	14,111
16	53	2 718,750	15,018
17	59	3 281,250	16,124
18	65	3 843,750	17,032
19	73	4 687,500	18,166
20	77	5 437,500	19,028
21	82	6 375,000	19,964
22	89	7 687,500	21,052
23	97	9 375,000	22,113
24	103	11 625,000	23,072
25	113	15 375,000	23,991
26	126	20 250,000	24,597

D.2 Psychoacoustic model 2

D.2.1 General

Psychoacoustic Model 2 is an independent psychoacoustic model that can be adjusted and adapted to any ISO/IEC 11172-3 layer. This annex presents the general Psychoacoustic Model 2, and provides sufficient information for implementation of Model 2 with Layers I and II. The Layer III psychoacoustic model is based on this implementation, with adaptations as described in the Layer III encoder.

The threshold generation process has three inputs. They are:

a) The shift length for the threshold calculation process, *iblen*, where 384<*iblen*<640. This *iblen* must remain constant over any particular application of the threshold calculation process. If (as in Layer III), it is necessary to calculate thresholds for two different shift lengths, two processes, each running with a fixed shift length, will be necessary. In the case of *iblen* outside the range of 384 to 640 it may be necessary to calculate the psychoacoustic thresholds with a different window length as well as shift length. There are two ways to do this:

   - Use a different length transform, and recalculate the startup coefficients for the model, or

   - Use the same length transform, but a substantially shorter Hann window, appropriate to the data and problem at hand.

   The choice of these is left to the implementation.

b) The newest *iblen* samples of the signal, with the samples delayed (either in the filter bank or psychoacoustic calculation) such that the window of the psychoacoustic calculation is centered in the time-window of application.

c) The sampling rate. There are sets of tables provided for the standard sampling rates. Sampling rate, like *iblen*, must necessarily remain constant over one implementation of the threshold calculation process.

There is one output from Psychoacoustic Model 2, a set of Signal-to-Masking Ratios, SMR_n, which are adapted to the layers as described below.

Before running the model initially, the array used to hold the preceding FFT source data window and the arrays used to hold r and f should be zeroed to provide a known starting point.

In Layer II, the psychoacoustic masking ratios must be calculated twice during each coder frame. The more stringent of each pair of ratios is used for bit allocation as shown in the software simulation model for Layers I and II with Psychoacoustic Model 2.

D.2.2 Comments on notation

Throughout this threshold calculation process, three indices for data values are used. These are:

ω - indicates that the calculation is indexed by frequency in the FFT spectral line domain. An index of 1 corresponds to the DC term and an index of 513 corresponds to the spectral line at the Nyquist frequency.

b - indicates that the calculation is indexed in the threshold calculation partition domain. In the case where the calculation includes a convolution or sum in the threshold calculation partition domain, *bb* will be used as the summation variable. Partition numbering starts at 1.

n - indicates that the calculation is indexed in the coder bit (or codebook) allocation domain. An index of 1 corresponds to the lowest band in the subband filter bank.

D.2.3 The "spreading function"

Several points in the following description refer to the "spreading function". It is calculated by the following method:

$$tmpx = 1{,}05\ (j-i),$$

Where i is the Bark value of the signal being spread, j is the Bark value of the band being spread into, and $tmpx$ is a temporary variable.

$$x = 8\ minimum\ ((tmpx-0{,}5)^2-2(tmpx-0{,}5),0)$$

Where x is a temporary variable, and minimum (a,b) is a function returning the more negative of a or b.

$$tmpy = 15{,}811389+7{,}5(tmpx+0{,}474)-17{,}5(1{,}0+(tmpx+0{,}474)^2)^{0{,}5}$$

where $tmpy$ is another temporary variable.

$$\text{if } (tmpy<-100) \text{ then } \{sprdngf(i,j)=0\} \text{ else } \{sprdngf(i,j)=10^{\frac{(x+tmpy)}{10}}\}$$

D.2.4 Steps in threshold calculation

The following are the necessary steps for calculation of the SMR_n used in the coder.

a) Reconstruct 1 024 samples of the input signal.

iblen new samples are made available at every call to the threshold generator. The threshold generator must store 1 024-*iblen* samples, and concatenate those samples to accurately reconstruct 1 024 consecutive samples of the input signal, s_i, where i represents the index, $1 <= i <= 1\ 024$ of the current input stream.

b) Calculate the complex spectrum of the input signal.

First, s_i is windowed by a 1 024 point Hann window, i.e. $sw_i = s_i * (0{,}5-0{,}5\cos(\frac{2\pi(i-0{,}5)}{1024}))$. Note that in Layer III, a shorter window may be used when window switching is active, with appropriate centering of the window, per the Layer III encoder description.

Second, a standard forward FFT of sw_i is calculated.

Third, the polar representation of the transform is calculated. r_ω and f_ω represent the magnitude and phase components of the transformed sw_i, respectively.

c) Calculate a predicted r and f.

A predicted magnitude, $\hat{r}_\omega$, and phase, $\hat{f}_\omega$ are calculated from the preceding two threshold calculation blocks' r and f:

$$\hat{r}_\omega = 2{,}0 r_\omega(t-1) - r_\omega(t-2)$$

$$\hat{f}_\omega = 2{,}0 f_\omega(t-1) - f_\omega(t-2)$$

where t represents the current block number, $t-1$ indexes the previous block's data, and $t-2$ indexes the data from the threshold calculation block before that.

d) Calculate the unpredictability measure c_ω

c_ω, the unpredictability measure, is:

$$c_\omega = \frac{((r_\omega \cos f_\omega - \hat{r}_\omega \cos \hat{f}_\omega)^2 + (r_\omega \sin f_\omega - \hat{r}_\omega \sin \hat{f}_\omega)^2)^{0,5}}{r_\omega + \mathrm{abs}(\hat{r}_\omega)}$$

By sacrificing performance, this measure can be calculated on only a lower portion of the frequency lines. Calculations should be done from DC to at least 3 kHz and preferably to 7kHz. An upper limit of less than 5,5kHz may considerably reduce performance from that obtained during the subjective testing of the audio algorithm. The c_ω values above this limit should be set to 0,3. Best results will be obtained by calculating c_ω up to 20 kHz.

e) Calculate the energy and unpredictability in the threshold calculation partitions.

The energy in each partition, e_b, is:

$$e_b = \sum_{\omega=\omega low_b}^{\omega high_b} r_\omega^2$$

and the weighted unpredictability, c_b, is:

$$c_b = \sum_{\omega=\omega low_b}^{\omega high_b} r_\omega^2 c_\omega$$

The threshold calculation partitions provide a resolution of approximately either one FFT line or $\frac{1}{3}$ critical band, whichever is wider. At low frequencies, a single line of the FFT will constitute a calculation partition. At high frequencies, many lines will be combined into one calculation partition. A set of partition values is provided for each of the three sampling rates in table D.3."Calculation partition tables". These table elements will be used in the threshold calculation process. There are several elements in each table entry:

1. The index of the calculation partition, b.

2. The lowest frequency line in the partition, ωlow_b.

3. The highest frequency line in the partition, $\omega high_b$.

4. The median bark value of the partition, $bval_b$.

5. A lower limit for the SNR in the partition that controls stereo unmasking effects, $minval_b$.

6. The value for tone masking noise (in dB) for the partition, TMN_b.

A largest value of b, $bmax$, equal to the largest index, exists for each sampling rate.

f) Convolve the partitioned energy and unpredictability with the spreading function.

$$ecb_b = \sum_{bb=1}^{bmax} e_{bb} * sprdngf(bval_{bb}, bval_b)$$

$$ct_b = \sum_{bb=1}^{bmax} c_{bb} * sprdngf(bval_{bb}, bval_b)$$

Because ct_b. is weighted by the signal energy, it must be renormalized to cb_b.

$$cb_b = \frac{ct_b}{ecb_b}$$

At the same time, due to the non-normalized nature of the spreading function, ecb_b should be renormalized and the normalized energy en_b, calculated.

$$en_b = ecb_b * rnorm_b$$

The normalization coefficient, $rnorm_b$. is:

$$rnorm_b = \frac{1}{\sum_{bb=0}^{bmax} sprdngf(bval_{bb}, bval_b)}$$

g) Convert cb_b to tb_b, the tonality index.

$$tb_b = -0{,}299 - 0{,}43 \log_e (cb_b)$$

Each tb_b is limited to the range of $0<tb_b<1$.

h) Calculate the required SNR in each partition.

$NMT_b = 5{,}5$dB for all b. NMT_b is the value for noise masking tone (in dB) for the partition. The required signal to noise ratio, SNR_b, is:

$$SNR_b = \text{maximum}(minval_b,\ tb_b * TMN_b + (1-tb_b) * NMT_b)$$

Where maximum (a,b) is a function returning the least negative of a or b.

i) Calculate the power ratio.

The power ratio, bc_b, is:

$$bc_b = 10^{\frac{-SNR_b}{10}}$$

j) Calculation of actual energy threshold, nb_b.

$$nb_b = en_b\, bc_b$$

k) Spread the threshold energy over FFT lines, yielding nb_ω.

$$nb_\omega = \frac{nb_b}{\omega high_b - \omega low_b + 1}$$

l) Include absolute thresholds, yielding the final energy threshold of audibility, thr_ω

$$thr_\omega = \max(nb_\omega, absthr_\omega)$$

The dB values of *absthr* shown in tables D.4. "Absolute threshold tables" are relative to the level that a sine wave of $\pm\frac{1}{2}$ lsb has in the FFT used for threshold calculation. The dB values must be converted into the energy domain after considering the FFT normalization actually used.

m) Pre-echo control

For Layer III, pre-echo control occurs at this point. The actual control is described as part of the Layer III encoder specification. This step is omitted for Layers I and II.

n) Calculate the signal-to-mask ratios, SMR_n.

Table D.5. "Layer I and II coder partition table" shows:

1. The index, n, of the coder partition.

2. The lower index ωlow_n, of the coder partition.

3. The upper index, $\omega high_n$ of the coder partition.

4. The width index, $width_n$, where $width_n$=1 for a psychoacoustically narrow scalefactor band, and $width_n$=0 for a psychoacoustically wide scalefactor band. A psychoacoustically narrow scalefactor band is one whose width is less than approximately $\frac{1}{3}$ critical band.

The energy in the scalefactor band, $epart_n$, is:

$$epart_n = \sum_{\omega=\omega low_n}^{\omega high_n} r_\omega^2$$

Then, if ($width_n$ = 1), the noise level in the scalefactor band, $npart_n$ is calculated as:

$$npart_n = \sum_{\omega=\omega low_n}^{\omega high_n} thr_\omega$$

else,

$$npart_n = \text{minimum}(thr_{\omega low_n},...,thr_{\omega high_n}) * (\omega high_n - \omega low_n + 1)$$

Where, in this case, minimum (a,...,z) is a function returning the smallest positive argument of the arguments a...z.

The ratios to be sent to the coder, SMR_n, are calculated as:

$$SMR_n = 10 \log_{10} \left(\frac{epart_n}{npart_n} \right)$$

Table D.3a. -- Calculation partition table
This table is valid at a sampling rate of 32 kHz.

Index	ωlow	ωhigh	bval	minval	TMN
1	1	1	0,00	0,0	24,5
2	2	4	0,63	0,0	24,5
3	5	7	1,56	20,0	24,5
4	8	10	2,50	20,0	24,5
5	11	13	3,44	20,0	24,5
6	14	16	4,34	20,0	24,5
7	17	19	5,17	20,0	24,5
8	20	22	5,94	20,0	24,5
9	23	25	6,63	17,0	24,5
10	26	28	7,28	15,0	24,5
11	29	31	7,90	15,0	24,5
12	32	34	8,50	10,0	24,5
13	35	37	9,06	7,0	24,5
14	38	41	9,65	7,0	24,5
15	42	45	10,28	4,4	24,8
16	46	49	10,87	4,4	25,4
17	50	53	11,41	4,5	25,9
18	54	57	11,92	4,5	26,4
19	58	61	12,39	4,5	26,9
20	62	65	12,83	4,5	27,3
21	66	70	13,29	4,5	27,8
22	71	75	13,78	4,5	28,3
23	76	81	14,27	4,5	28,8
24	82	87	14,76	4,5	29,3
25	88	93	15,22	4,5	29,7
26	94	99	15,63	4,5	30,1
27	100	106	16,06	4,5	30,6
28	107	113	16,47	4,5	31,0
29	114	120	16,86	4,5	31,4
30	121	129	17,25	4,5	31,8
31	130	138	17,65	4,5	32,2
32	139	148	18,05	4,5	32,5
33	149	159	18,42	4,5	32,9
34	160	170	18,81	4,5	33,3
35	171	183	19,18	4,5	33,7
36	184	196	19,55	4,5	34,1
37	197	210	19,93	4,5	34,4
38	211	225	20,29	4,5	34,8
39	226	240	20,65	4,5	35,2
40	241	258	21,02	4,5	35,5
41	259	279	21,38	4,5	35,9
42	280	300	21,74	4,5	36,2
43	301	326	22,10	4,5	36,6
44	327	354	22,44	4,5	36,9
45	355	382	22,79	4,5	37,3
46	383	420	23,14	4,5	37,6
47	421	458	23,49	4,5	38,0
48	459	496	23,83	4,5	38,3
49	497	513	24,07	4,5	38,6

Table D.3b. -- Calculation partition table

This table is valid at a sampling rate of 44,1 kHz.

Index	ωlow	ωhigh	bval	minval	TMN
1	1	1	0,00	0,0	24,5
2	2	2	0,43	0,0	24,5
3	3	3	0,86	0,0	24,5
4	4	4	1,29	20,0	24,5
5	5	5	1,72	20,0	24,5
6	6	6	2,15	20,0	24,5
7	7	7	2,58	20,0	24,5
8	8	8	3,01	20,0	24,5
9	9	9	3,45	20,0	24,5
10	10	10	3,88	20,0	24,5
11	11	11	4,28	20,0	24,5
12	12	12	4,67	20,0	24,5
13	13	13	5,06	20,0	24,5
14	14	14	5,42	20,0	24,5
15	15	15	5,77	20,0	24,5
16	16	16	6,11	17,0	24,5
17	17	19	6,73	17,0	24,5
18	20	22	7,61	15,0	24,5
19	23	25	8,44	10,0	24,5
20	26	28	9,21	7,0	24,5
21	29	31	9,88	7,0	24,5
22	32	34	10,51	4,4	25,0
23	35	37	11,11	4,5	25,6
24	38	40	11,65	4,5	26,2
25	41	44	12,24	4,5	26,7
26	45	48	12,85	4,5	27,4
27	49	52	13,41	4,5	27,9
28	53	56	13,94	4,5	28,4
29	57	60	14,42	4,5	28,9
30	61	64	14,86	4,5	29,4
31	65	69	15,32	4,5	29,8
32	70	74	15,79	4,5	30,3
33	75	80	16,26	4,5	30,8
34	81	86	16,73	4,5	31,2
35	87	93	17,19	4,5	31,7
36	94	100	17,62	4,5	32,1
37	101	108	18,05	4,5	32,5
38	109	116	18,45	4,5	32,9
39	117	124	18,83	4,5	33,3
40	125	134	19,21	4,5	33,7
41	135	144	19,60	4,5	34,1
42	145	155	20,00	4,5	34,5
43	156	166	20,38	4,5	34,9
44	167	177	20,74	4,5	35,2
45	178	192	21,12	4,5	35,6
46	193	207	21,48	4,5	36,0
47	208	222	21,84	4,5	36,3
48	223	243	22,20	4,5	36,7
49	244	264	22,56	4,5	37,1
50	265	286	22,91	4,5	37,4
51	287	314	23,26	4,5	37,8
52	315	342	23,60	4,5	38,1
53	343	371	23,95	4,5	38,4
54	372	401	24,30	4,5	38,8
55	402	431	24,65	4,5	39,1
56	432	469	25,00	4,5	39,5
57	470	513	25,33	3,5	39,8

Table D.3c. -- Calculation partition table
This table is valid at a sampling rate of 48 kHz.

Index	ωlow	ωhigh	bval	minval	TMN
1	1	1	0,00	0,0	24,5
2	2	2	0,47	0,0	24,5
3	3	3	0,94	0,0	24,5
4	4	4	1,41	20,0	24,5
5	5	5	1,88	20,0	24,5
6	6	6	2,34	20,0	24,5
7	7	7	2,81	20,0	24,5
8	8	8	3,28	20,0	24,5
9	9	9	3,75	20,0	24,5
10	10	10	4,20	20,0	24,5
11	11	11	4,63	20,0	24,5
12	12	12	5,05	20,0	24,5
13	13	13	5,44	20,0	24,5
14	14	14	5,83	20,0	24,5
15	15	15	6,19	20,0	24,5
16	16	16	6,52	17,0	24,5
17	17	17	6,86	17,0	24,5
18	18	20	7,49	15,0	24,5
19	21	23	8,40	10,0	24,5
20	24	26	9,24	7,0	24,5
21	27	29	9,97	7,0	24,5
22	30	32	10,65	4,4	25,1
23	33	35	11,28	4,5	25,8
24	36	38	11,86	4,5	26,4
25	39	41	12,39	4,5	26,9
26	42	45	12,96	4,5	27,5
27	46	49	13,56	4,5	28,1
28	50	53	14,12	4,5	28,6
29	54	57	14,62	4,5	29,1
30	58	62	15,14	4,5	29,6
31	63	67	15,67	4,5	30,2
32	68	72	16,15	4,5	30,7
33	73	77	16,58	4,5	31,1
34	78	83	17,02	4,5	31,5
35	84	89	17,44	4,5	31,9
36	90	95	17,84	4,5	32,3
37	96	103	18,24	4,5	32,7
38	104	111	18,66	4,5	33,2
39	112	120	19,07	4,5	33,6
40	121	129	19,47	4,5	34,0
41	130	138	19,85	4,5	34,3
42	139	149	20,23	4,5	34,7
43	150	160	20,63	4,5	35,1
44	161	173	21,02	4,5	35,5
45	174	187	21,40	4,5	35,9
46	188	201	21,76	4,5	36,3
47	202	219	22,12	4,5	36,6
48	220	238	22,47	4,5	37,0
49	239	257	22,83	4,5	37,3
50	258	283	23,18	4,5	37,7
51	284	309	23,53	4,5	38,0
52	310	335	23,88	4,5	38,4
53	336	363	24,23	4,5	38,7
54	364	391	24,58	4,5	39,1
55	392	423	24,93	4,5	39,4
56	424	465	25,27	4,5	39,8
57	466	507	25,61	3,5	40,1
58	508	513	25,81	3,5	40,3

Table D.4a. -- Absolute threshold table

This table is valid at a sampling rate of 32 kHz.

A value of 0 dB represents a level in the absolute threshold calculation of 96 dB below the energy of a sine wave of amplitude +-32 760.

index [line]		abstthr
lower	higher	[dB]
1	1	58,23
2	2	33,44
3	3	24,17
4	4	19,20
5	5	16,05
6	6	13,87
7	7	12,26
8	8	11,01
9	9	10,01
10	10	9,20
11	11	8,52
12	12	7,94
13	13	7,44
14	14	7,00
15	15	6,62
16	16	6,28
17	17	5,97
18	18	5,70
19	19	5,44
20	20	5,21
21	21	5,00
22	22	4,80
23	23	4,62
24	24	4,45
25	25	4,29
26	26	4,14
27	27	4,00
28	28	3,86
29	29	3,73
30	30	3,61
31	31	3,49
32	32	3,37
33	33	3,26
34	34	3,15
35	35	3,04
36	36	2,93
37	37	2,83
38	38	2,73
39	39	2,63
40	40	2,53
41	41	2,42
42	42	2,32
43	43	2,22
44	44	2,12
45	45	2,02
46	46	1,92
47	47	1,81
48	48	1,71
49	50	1,49
51	52	1,27
53	54	1,04
55	56	0,80
57	57	0,55
59	60	0,29
61	62	0,02
63	64	-0,25
65	66	-0,54
67	68	-0,83
69	70	-1,12
71	72	-1,43
73	74	-1,73
75	76	-2,04
77	78	-2,34
79	80	-2,64
81	82	-2,93
83	84	-3,22
85	86	-3,49
87	88	-3,74
89	90	-3,98
91	92	-4,20
93	94	-4,40
95	96	-4,57
97	100	-4,82
101	104	-4,96
105	108	-4,97
109	112	-4,86
113	116	-4,63
117	120	-4,29
121	124	-3,87
125	128	-3,39
129	132	-2,86
133	136	-2,31
137	140	-1,77
141	144	-1,24
145	148	-0,74
149	152	-0,29
153	156	0,12
157	160	0,48
161	164	0,79
165	168	1,06
169	172	1,29
173	176	1,49
177	180	1,66
181	184	1,81
185	188	1,95
189	192	2,08
193	200	2,33
201	208	2,59
209	216	2,86
217	224	3,17
225	232	3,51
233	240	3,89
241	248	4,31
249	256	4,79
257	264	5,31
265	272	5,88
273	280	6,50
281	288	7,19
289	296	7,93
297	304	8,75
305	312	9,63
313	320	10,58
321	328	11,60
329	336	12,71
337	344	13,90
345	352	15,18
353	360	16,54
361	368	18,01
369	376	19,57
377	384	21,23
385	392	23,01
393	400	24,90
401	408	26,90
409	416	29,03
417	424	31,28
425	432	33,67
433	440	36,19
441	448	38,86
449	456	41,67
457	464	44,63
465	472	47,76
473	480	51,03

Table D.4b -- Absolute threshold table

This table is valid at a sampling rate of 44,1kHz.

A value of 0 dB represents a level in the absolute threshold calculation of 96dB below the energy of a sine wave of amplitude +-32 760.

index [line]		absthr
lower	higher	[dB]
1	1	45,05
2	2	25,87
3	3	18,70
4	4	14,85
5	5	12,41
6	6	10,72
7	7	9,47
8	8	8,50
9	9	7,73
10	10	7,10
11	11	6,56
12	12	6,11
13	13	5,72
14	14	5,37
15	15	5,07
16	16	4,79
17	17	4,55
18	18	4,32
19	19	4,11
20	20	3,92
21	21	3,74
22	22	3,57
23	23	3,40
24	24	3,25
25	25	3,10
26	26	2,95
27	27	2,81
28	28	2,67
29	29	2,53
30	30	2,39
31	31	2,25
32	32	2,11
33	33	1,97
34	34	1,83
35	35	1,68
36	36	1,53
37	37	1,38
38	38	1,23
39	39	1,07
40	40	0,90
41	41	0,74
42	42	0,56
43	43	0,39
44	44	0,21
45	45	0,02
46	46	-0,17
47	47	-0,36
48	48	-0,56
49	50	-0,96
51	52	-1,37
53	54	-1,79
55	56	-2,21
57	58	-2,63
59	60	-3,03
61	62	-3,41
63	64	-3,77
65	66	-4,09
67	68	-4,37
69	70	-4,60
71	72	-4,78
73	74	-4,91
75	76	-4,97
77	78	-4,98
79	80	-4,92
81	82	-4,81
83	84	-4,65
85	86	-4,43
87	88	-4,17
89	90	-3,87
91	92	-3,54
93	94	-3,19
95	96	-2,82
97	100	-2,06
101	104	-1,33
105	108	-0,64
109	112	-0,04
113	116	0,47
117	120	0,89
121	124	1,23
125	128	1,51
129	132	1,74
133	136	1,93
137	140	2,11
141	144	2,28
145	148	2,45
149	152	2,63
153	156	2,82
157	160	3,03
161	164	3,25
165	168	3,49
169	172	3,74
173	176	4,02
177	180	4,32
181	184	4,64
185	188	4,98
189	192	5,35
193	200	6,15
201	208	7,07
209	216	8,10
217	224	9,25
225	232	10,54
233	240	11,97
241	248	13,56
249	256	15,30
257	264	17,23
265	272	19,33
273	280	21,64
281	288	24,15
289	296	26,88
297	304	29,84
305	312	33,04
313	320	36,51
321	328	40,24
329	336	44,26
337	344	48,58
345	352	53,21
353	360	58,17
361	368	63,48
369	376	69,13
377	384	69,13
385	392	69,13
393	400	69,13
401	408	69,13
409	416	69,13
417	424	69,13
425	432	69,13
433	440	69,13
441	448	69,13
449	456	69,13
457	464	69,13

Table D.4c -- Absolute threshold table

This table is valid at a sampling rate of 48 kHz.

A value of 0 dB represents a level in the absolute threshold calculation of 96dB below the energy of a sine wave of amplitude +-32 760.

index [line]		absthr
lower	higher	[dB]
1	1	42,10
2	2	24,17
3	3	17,47
4	4	13,87
5	5	11,60
6	6	10,01
7	7	8,84
8	8	7,94
9	9	7,22
10	10	6,62
11	11	6,12
12	12	5,70
13	13	5,33
14	14	5,00
15	15	4,71
16	16	4,45
17	17	4,21
18	18	4,00
19	19	3,79
20	20	3,61
21	21	3,43
22	22	3,26
23	23	3,09
24	24	2,93
25	25	2,78
26	26	2,63
27	27	2,47
28	28	2,32
29	29	2,17
30	30	2,02
31	31	1,86
32	32	1,71
33	33	1,55
34	34	1,38
35	35	1,21
36	36	1,04
37	37	0,86
38	38	0,67
39	39	0,49
40	40	0,29
41	41	0,09
42	42	-0,11
43	43	-0,32
44	44	-0,54
45	45	-0,75
46	46	-0,97
47	47	-1,20
48	48	-1,43
49	50	-1,88
51	52	-2,34
53	54	-2,79
55	56	-3,22
57	58	-3,62
59	60	-3,98
61	62	-4,30
63	64	-4,57
65	66	-4,77
67	68	-4,91
69	70	-4,98
71	72	-4,97
73	74	-4,90
75	76	-4,76
77	78	-4,55
79	80	-4,29
81	82	-3,99
83	84	-3,64
85	86	-3,26
87	88	-2,86
89	90	-2,45
91	92	-2,04
93	94	-1,63
95	96	-1,24
97	100	-0,51
101	104	0,12
105	108	0,64
109	112	1,06
113	116	1,39
117	120	1,66
121	124	1,88
125	128	2,08
129	132	2,27
133	136	2,46
137	140	2,65
141	144	2,86
145	148	3,09
149	152	3,33
153	156	3,60
157	160	3,89
161	164	4,20
165	168	4,54
169	172	4,91
173	176	5,31
177	180	5,73
181	184	6,18
185	188	6,67
189	192	7,19
193	200	8,33
201	208	9,63
209	216	11,08
217	224	12,71
225	232	14,53
233	240	16,54
241	248	18,77
249	256	21,23
257	264	23,94
265	272	26,90
273	280	30,14
281	288	33,67
289	296	37,51
297	304	41,67
305	312	46,17
313	320	51,04
321	328	56,29
329	332	61,94
333	340	68,00
341	348	68,00
349	356	68,00
357	364	68,00
365	372	68,00
373	380	68,00
381	388	68,00
389	396	68,00
397	404	68,00
405	412	68,00
413	420	68,00
421	428	68,00

Table D.5 -- Layer I and Layer II coder partition table

Index	ωlow_{n+1} $\omega high_n$	$width_n$
0	1	0
1	17	0
2	33	0
3	49	0
4	65	0
5	81	0
6	97	0
7	113	0
8	129	0
9	145	0
10	161	0
11	177	0
12	193	0
13	209	1
14	225	1
15	241	1
16	257	1
17	273	1
18	289	1
19	305	1
20	321	1
21	337	1
22	353	1
23	369	1
24	385	1
25	401	1
26	417	1
27	433	1
28	449	1
29	465	1
30	481	1
31	497	1
32	513	1

Annex E

(informative)

Bit sensitivity to errors

E.1. General

This annex indicates the sensitivity of individual bits to random errors if application specific error protection is needed.This sensitivity is given for each bit by a value from 0 to 5, indicating the amount of degradation resulting from one isolated error :

5 catastrophic
4 very annoying
3 annoying
2 slightly annoying
1 audible
0 insensitive

The values are not the results of precise measurements, rather they rely upon knowledge of the codec. They assume the error detection scheme is not in use.

Some fields in the bit stream do not have a fixed length. All bits in these fields are rated for error sensitivity, even if not in use.

For all layers, the header and error check information defined in 2.4.1.3 and 2.4.1.4 are considered to have the highest sensitivity.

E.2. Layers I and II

Parameters	#bit	sensitivity
Bit allocation	all bits	5
Scalefactors select information	all bits	5
Scalefactors	5 (msb)	4
	4	4
	3	4
	2	3
	1	2
	0 (lsb)	1
Subband samples (*)	8-16(msb)	3
	5-7	2
	3,4	1
	(lsb)0-2	0

(*) according to the bit allocation

E.3. Layer III

Parameters	#bit	sensitivity
scfsi	all bits	5
part2_3_length	all bits	4
big_values	all bits	3
global_gain	all bits	5
scalefac_compress	all bits	5
window_switching_flag	0	5
block_type	all bits	4
mixed_block_flag	0	4
table_select	all bits	5
region0_count	all bits	3
region1_count	all bits	3
preflag	0	2
scalefac_scale	0	2
count1table_select	0	3
Subblock_gain	2 (msb)	4
	1	3
	0 (lsb)	2
scale_fac (**)	3 (msb)	3(2)
	2	3(2)
	1	2(1)
	0 (lsb)	2(1)
Huffmancodebits () (***)	0...n-1	3 - 0

(**) the scalefac length depends on scalefac_compress.
The bit sensitivity values refer to the scalefac_scale value 1 (if 0 the value is in parenthesis).
(***) If n is the number of bits for Huffman coding in one block the bit sensivity decreases linearly from 3 to 0 as the bit number varies from 0 up to n, (from low to high frequency).

Note:
Rearrangement of the Huffman coded values:
To get better implicit error robustness for the low frequency part of the spectrum the Huffman coded values can be transmitted not in their logical order, but in an interleaved fashion.

If max_hlen is the maximum length of a Huffman codeword over the tables which are used to code the particular block and n is the number of bits used for Huffman coding of data in the block (not frame), then int(n/max_hlen) slots are filled with the first codewords, beginning from low frequencies. The remaining codewords are filled into the remaining place, again arranged from low to high frequencies.

After bit interleaving, the bit sensivity of bit k+i*int(n/max_hlen) decreases linearly from 3 to 0 as k varies from 0 up to int(n/max_hlen)-1, where i=0,...,max_hlen-1, and n is the number of bits for Huffman coding in one block.

This is the recommended practice for Layer III data for all channels where error robustness is important.

Annex F

(informative)

Error concealment

An optional feature of the coded bit stream is the CRC word which provides some error detection facility to the decoder. The Hamming distance of this error detection code is d=4, which allows for the detection of up to 3 single bit errors or for the detection of one error burst of up to 16 bit length. The amount and the position of the protected bits within one encoded audio frame generally depends on the layer, the mode, data rate, and sampling frequency.

This can be used to control an error concealment strategy in order to avoid severe impairments of the reconstructed signal due to errors in the most sensitive information.

Some basic techniques can be used for concealment, for instance information substitution, or muting. A simple substitution technique consists, when an erroneous frame occurs, of replacing it by the previous one (if error free).

Annex G

(informative)

Joint stereo coding

G.1. Intensity stereo coding Layer I, II

An optional joint stereo coding method used in Layers I and II is intensity stereo coding. Intensity stereo coding can be used to increase the audio quality and/or reduce the bitrate for stereophonic signals. The gain in bitrate is typically about 10 to 30 kbits/s. It requires negligible additional decoder complexity. The increase of encoder complexity is small. The encoder and decoder delay is not affected.

Psychoacoustic results indicate that at high frequencies (above about 2 kHz) the localization of the stereophonic image within a critical band is determined by the temporal envelope and not by the temporal fine structure of the audio signal.

The basic idea for intensity stereo coding is that for some subbands, instead of transmitting separate left and right subband samples, only the sum-signal is transmitted, but with scalefactors for both the left and right channels, thus preserving the stereophonic image.

Flow diagrams of a stereo encoder and decoder, including intensity stereo mode, are shown in figure G.1 "General stereo encoder flow-chart" and figure G.2 "General stereo decoder flow-chart". First, an estimation is made of the required bitrate for both left and right channel. If the required bitrate exceeds the available bitrate, the required bitrate can be decreased by setting a number of subbands to intensity stereo mode. Depending on the bitrate needed, subbands

16 to 31,
12 to 31,
8 to 31, or
4 to 31

can be set to intensity stereo mode. For the quantization of such combined subbands, the higher of the bit allocations for left and right channel is used.

The left and right subband signals of the subbands in joint stereo mode are added. These new subband signals are scaled in the normal way, but the originally determined scalefactors of the left and right subband signals are transmitted according to the bitstream syntax. Quantization of common subband samples, coding of common samples, and coding of common bit allocation are performed in the same way as in independent coding.

G.2. MS_Stereo and intensity stereo coding Layer III

In Layer III a combination of ms_stereo mode (sum/difference) and intensity stereo mode can be used.

a) MS_stereo switching

MS_stereo mode is switched on if in joint stereo mode condition

$$\sum_{i=0}^{511} [\, rl_i^2 - rr_i^2 \,] < 0.8 * \sum_{i=0}^{511} [\, rl_i^2 + rr_i^2 \,]$$

is true. The values rl_i and rr_i correspond to the energies of the FFT line spectrum of the left and right channel calculated within the psychoacoustic model.

b) MS_stereo processing

- MS matrix

In MS_stereo mode the values of the normalized middle/side channel M_i/S_i are transmitted instead of the left/right channel values L_i/R_i:

$$M_i = \frac{R_i + L_i}{\sqrt{2}} \quad \text{and} \quad S_i = \frac{L_i - R_i}{\sqrt{2}}$$

- Limitation of S_i channel bandwith

All S_i values above the highest scalefactor band are set to zero.

- Sparsing of S_i channel

In every scalefactor band sb all pairs of small values (S_i, S_{i+1}) are set to zero:

$$\text{if } (S_i^2 + S_{i+1}^2) < s_{sb} * (L_i^2 + L_{i+1}^2 + R_i^2 + R_{i+1}^2) \ \{ \quad S_i = 0; \quad S_{i+1} = 0; \ \}$$

The following difference channel threshold coefficients apply to the scalefactor bands for block type != 2 ('long MDCT transforms'):

sb	0	1	2	3	4	5	6	7	8	9
s_{sb}	0,0	0,0	0,0	0,0	0,0	0,10	0,10	0,10	0,10	0,10

sb	10	11	12	13	14	15	16	17	18	19	20
s_{sb}	0,10	0,20	0,30	0,40	0,50	0,60	0,70	0,80	0,90	1,00	1,50

c) Intensity stereo processing

- Calculation of intensity stereo position

For each scalefactor band sb coded in intensity stereo the following steps are executed:

- $is\_pos_{sb} = \text{NINT}\left(\frac{12}{\pi} * \arctan\left(\sqrt{\frac{L\_Energy_{sb}}{R\_Energy_{sb}}}\right)\right)$
- $L_i = L_i + R_i$ for all indices i within the actual scalefactor band sb
- $R_i = 0$ for all indices i within the actual scalefactor band sb
- the intensity stereo position is_pos_{sb} is transmitted instead of the scalefactor of the right channel (3 bits always, stereo positions 0..6, 7=illegal stereo position)

where $L\_Energy_{sb}/R\_Energy_{sb}$ denote the signal energies of the left/right channel within the actual scalefactor band and L_i/R_i are the transformed values.

Scalefactor bands of the right/difference channel containing only zeros after coding which do not belong to the intensity coded part should be transmitted with the scalefactor '7' to prevent intensity stereo decoding.

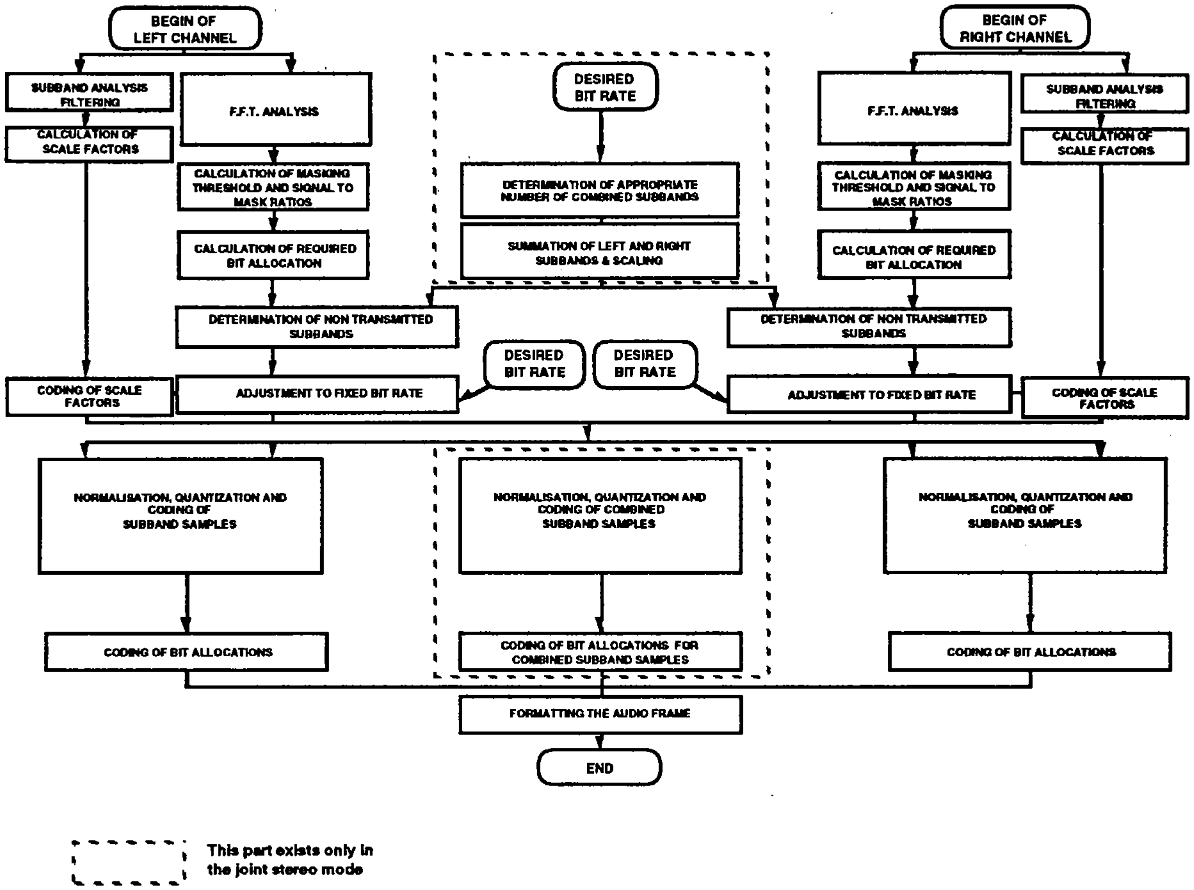

Figure G.1 -- General stereo encoder flow chart

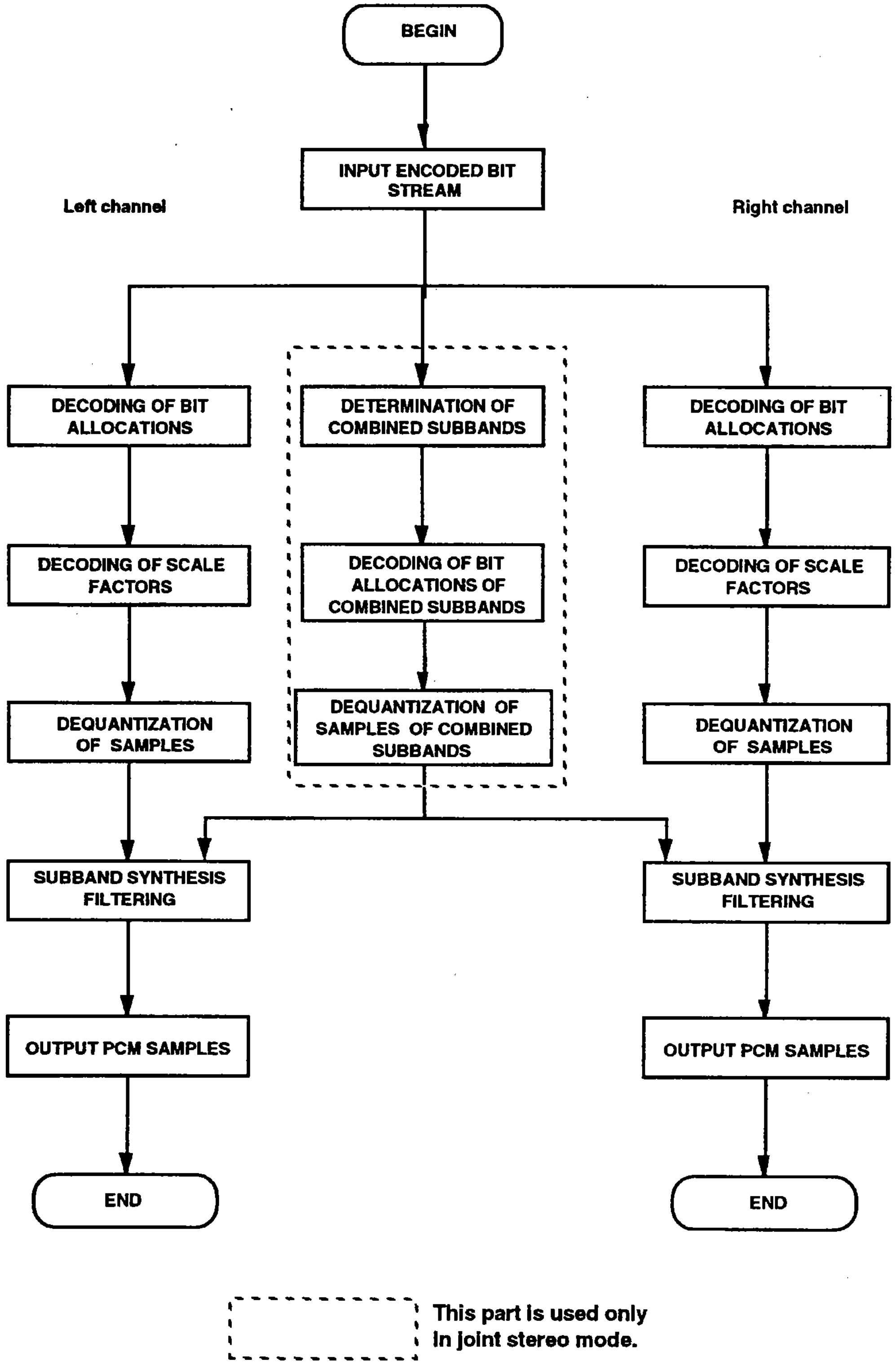

Figure G.2 -- General stereo decoder flow chart

Annex H

(informative)

List of patent holders

The user's attention is called to the possibility that - for some of the processes specified in this part of ISO/IEC 11172 - compliance with this International Standard may require use of an invention covered by patent rights.

By publication of this part of ISO/IEC 11172, no position is taken with respect to the validity of this claim or of any patent rights in connection therewith. However, each company listed in this annex has filed with the Information Technology Task Force (ITTF) a statement of willingness to grant a license under such rights that they hold on reasonable and nondiscriminatory terms and conditions to applicants desiring to obtain such a license.

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UDC 681.3.04(084.14)

Descriptors: data processing, moving pictures, audio data, video recording, data storage devices, digital storage, coded representation, coding (data conversion).

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