13. The decision, whether long or short block type is used for encoding is made according to this pseudo code.

```
if PE for long block is greater than switch_pe then
   coding_block_type = short_block_type
else
   coding_block_type = long_block_type
end if
if (coding_block_type == short_block_type) and
   (last_coding_block_type == long_type) then
      last coding block type = start_type
else
   last_coding_block_type = short_type
```

The last four lines are necessary since there is no combined stop/start block type in AAC. *switch_pe* is a implementation dependend constant

14. Calculate the signal-to-mask ratios, *SMR(n)* and the codec threshold *xmin(n)*.

Table 45 to Table 57 shows:

1. The index, *swb*, of the coder partition called scalefactor band.

2. The offset of mdct line for the scalefactor band *swb_offset_long/short_window*.

we define the following variable :

```
n = swb
w_low(n) = swb_offset_long/short_window(n)
w_high(n) = swb_offset_long/short_window(n+1) - 1
```

The FFT energy in the scalefactor band, *epart(n)*, is:

```
do for each scalefactor band n
   epart(n) = 0
   do for w = lower index w_low(n) to n = upper index w_high(n)
       epart(n) = epart ( n) + r(w)^2
   end do
end do
```

the threshold for one line of the spectrum is calculated according to:

```
do for each threshold partition b
   thr(all line_indices in this partition b )=
     thr (w_low(b),...,w_high(b)) = nb(b) / (w_high(b)+1-w_low(b))
end do
```

the noise level in the scalefactor band on FFT level , *npart(n)* is calculated as:

```
do for each scalefactor band n
   npart(n) = minimum( thr(w_low(n)),..., thr(w_high(n)) )
        * (w_high(n)+1-w_low(n))
end do
```

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 quantization module, *SMR(n)*, are calculated as:

```
SMR(n) = epart(n) / npart(n)
```

For the calculation of coder thresholds xmin(n) the MDCT energy for each scalefactor band is calculated:

```
do for all scalefactor bands n
   codec_e(n) = 0
   do for lower index i to higher index i of this scalefactor band
     codec_e(n) = codec_e(n) +( mdct_line(i))^2
   end do
end do
```

Then *xmin(n)*, the maximum allowed error energy on MDCT level, can be calculated according to this formula:

```
xmin (n) = npart(n) * codec_e(n) / epart (n)
```

15. Calculate the bit allocation out of the psychoacoustic entropy (PE).
bit_allocation = pew1 PE + pew2*sqrt(PE);*

for long blocks the constants are defined as:

```
pew1 = 0.3 , pew2 = 6.0
```

for short blocks the PE of the eight short blocks is summed up and the constants are :

```
pew1 = 0.6 , pew2 = 24
```

then bit_allocation is limited to 0<bit_allocation < 3000 and more_bits is calculated :

```
more_bits = bit_allocation - (mean_bits - side_info_bits)
```

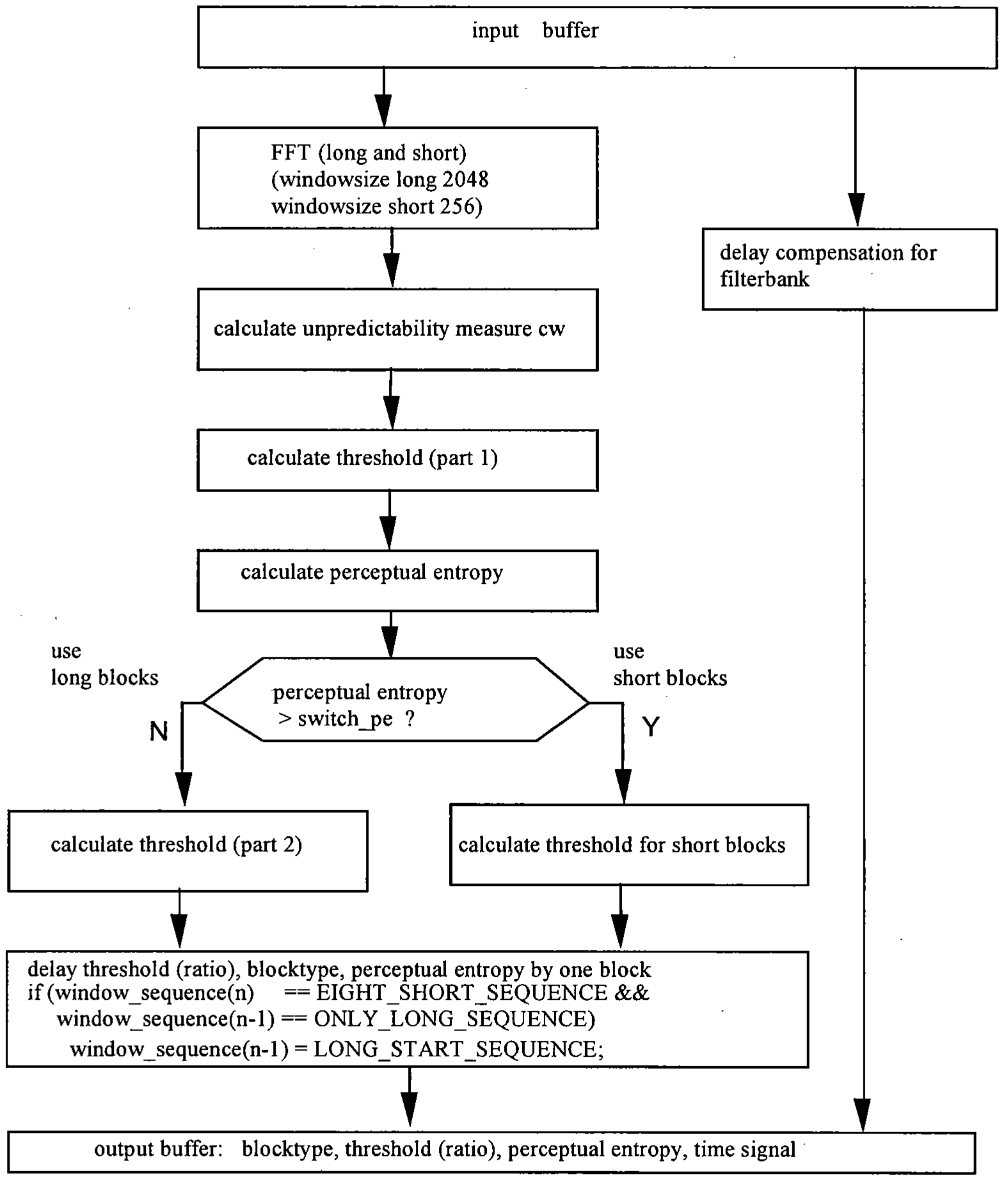


Figure C.1 — Block diagram psychoacoustic model

Table C.1 — Psychoacoustic parameters for 8 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	8	9	0.18	46.82
1	9	17	9	0.53	46.82
2	18	26	9	0.89	46.82
3	27	35	9	1.24	41.82
4	36	44	9	1.59	41.82
5	45	53	9	1.94	41.82
6	54	62	9	2.29	38.82
7	63	71	9	2.63	38.82
8	72	80	9	2.98	38.82
9	81	89	9	3.31	33.82
10	90	98	9	3.65	33.82
11	99	108	10	3.99	34.28
12	109	118	10	4.35	32.28
13	119	128	10	4.71	32.28
14	129	138	10	5.05	32.28
15	139	148	10	5.39	32.28
16	149	159	11	5.74	32.69
17	160	170	11	6.10	32.69
18	171	181	11	6.45	32.69
19	182	192	11	6.79	32.69
20	193	204	12	7.13	33.07
21	205	216	12	7.48	33.07
22	217	228	12	7.82	33.07
23	229	241	13	8.17	33.42
24	242	254	13	8.51	33.42
25	255	268	14	8.85	33.74
26	269	282	14	9.20	33.74
27	283	297	15	9.54	34.04
28	298	312	15	9.88	34.04
29	313	328	16	10.22	34.32
30	329	345	17	10.56	34.58
31	346	363	18	10.91	34.83
32	364	381	18	11.25	34.83
33	382	400	19	11.58	35.06
34	401	420	20	11.91	35.29
35	421	441	21	12.24	35.50
36	442	464	23	12.58	35.89
37	465	488	24	12.92	36.08
38	489	514	26	13.26	36.43
39	515	541	27	13.59	36.59
40	542	570	29	13.93	36.90
41	571	601	31	14.26	37.19
42	602	634	33	14.60	37.46
43	635	670	36	14.93	37.84
44	671	708	38	15.27	38.07
45	709	749	41	15.60	38.40
46	750	793	44	15.93	38.71
47	794	841	48	16.26	39.09
48	842	893	52	16.60	39.44
49	894	949	56	16.93	39.76
50	950	1009	60	17.26	40.06
51	1010	1023	14	17.47	33.74

Table C.2 — Psychoacoustic parameters for 8 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	1	2	0.32	30.29
1	2	3	2	0.95	30.29
2	4	5	2	1.57	25.29
3	6	7	2	2.19	22.29
4	8	9	2	2.80	22.29
5	10	11	2	3.40	17.29
6	12	13	2	3.99	17.29
7	14	15	2	4.56	15.29
8	16	17	2	5.12	15.29
9	18	19	2	5.66	15.29
10	20	21	2	6.18	15.29
11	22	23	2	6.68	15.29
12	24	25	2	7.16	15.29
13	26	27	2	7.63	15.29
14	28	29	2	8.07	15.29
15	30	31	2	8.50	15.29
16	32	33	2	8.90	15.29
17	34	35	2	9.29	15.29
18	36	37	2	9.67	15.29
19	38	39	2	10.03	15.29
20	40	41	2	10.37	15.29
21	42	44	3	10.77	17.05
22	45	47	3	11.23	17.05
23	48	50	3	11.66	17.05
24	51	53	3	12.06	17.05
25	54	56	3	12.44	17.05
26	57	59	3	12.79	17.05
27	60	63	4	13.18	18.30
28	64	67	4	13.59	18.30
29	68	71	4	13.97	18.30
30	72	75	4	14.32	18.30
31	76	80	5	14.69	19.27
32	81	85	5	15.07	19.27
33	86	90	5	15.42	19.27
34	91	96	6	15.77	20.06
35	97	102	6	16.13	20.06
36	103	109	7	16.49	20.73
37	110	116	7	16.85	20.73
38	117	124	8	17.20	21.31
39	125	127	3	17.44	17.05

Table C.3 — Psychoacoustic parameters for 11.025 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	6	7	0.19	45.73
1	7	13	7	0.57	45.73
2	14	20	7	0.95	45.73
3	21	27	7	1.33	40.73
4	28	34	7	1.71	40.73
5	35	41	7	2.08	37.73
6	42	48	7	2.45	37.73
7	49	55	7	2.82	37.73
8	56	62	7	3.18	32.73
9	63	69	7	3.54	32.73
10	70	76	7	3.89	32.73
11	77	83	7	4.24	30.73
12	84	90	7	4.59	30.73
13	91	97	7	4.92	30.73
14	98	105	8	5.28	31.31
15	106	113	8	5.65	31.31
16	114	121	8	6.01	31.31
17	122	129	8	6.36	31.31
18	130	137	8	6.70	31.31
19	138	146	9	7.06	31.82
20	147	155	9	7.42	31.82
21	156	164	9	7.77	31.82
22	165	173	9	8.11	31.82
23	174	183	10	8.46	32.28
24	184	193	10	8.82	32.28
25	194	203	10	9.16	32.28
26	204	214	11	9.50	32.69
27	215	225	11	9.85	32.69
28	226	237	12	10.19	33.07
29	238	249	12	10.54	33.07
30	250	262	13	10.88	33.42
31	263	275	13	11.22	33.42
32	276	289	14	11.56	33.74
33	290	304	15	11.90	34.04
34	305	320	16	12.24	34.32
35	321	337	17	12.59	34.58
36	338	355	18	12.94	34.83
37	356	374	19	13.28	35.06
38	375	394	20	13.62	35.29
39	395	415	21	13.96	35.50
40	416	438	23	14.29	35.89
41	439	462	24	14.63	36.08
42	463	488	26	14.96	36.43
43	489	516	28	15.29	36.75
44	517	546	30	15.63	37.05
45	547	579	33	15.96	37.46
46	580	614	35	16.30	37.72
47	615	652	38	16.63	38.07
48	653	693	41	16.97	38.40
49	694	737	44	17.30	38.71
50	738	785	48	17.64	39.09
51	786	836	51	17.97	39.35
52	837	891	55	18.30	39.68
53	892	950	59	18.64	39.98
54	951	1014	64	18.97	40.34
55	1015	1023	9	19.16	31.82

Table C.4 — Psychoacoustic parameters for 11.025 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	0.44	27.28
2	2	2	1	0.87	27.28
3	3	3	1	1.30	22.28
4	4	4	1	1.73	22.28
5	5	5	1	2.16	19.28
6	6	6	1	2.58	19.28
7	7	7	1	3.00	14.28
8	8	8	1	3.41	14.28
9	9	9	1	3.82	14.28
10	10	10	1	4.22	12.28
11	11	11	1	4.61	12.28
12	12	12	1	4.99	12.28
13	13	13	1	5.37	12.28
14	14	14	1	5.74	12.28
15	15	15	1	6.10	12.28
16	16	16	1	6.45	12.28
17	17	17	1	6.79	12.28
18	18	19	2	7.44	15.29
19	20	21	2	8.05	15.29
20	22	23	2	8.64	15.29
21	24	25	2	9.19	15.29
22	26	27	2	9.70	15.29
23	28	29	2	10.19	15.29
24	30	31	2	10.65	15.29
25	32	33	2	11.08	15.29
26	34	35	2	11.48	15.29
27	36	37	2	11.86	15.29
28	38	39	2	12.22	15.29
29	40	42	3	12.64	17.05
30	43	45	3	13.10	17.05
31	46	48	3	13.53	17.05
32	49	51	3	13.93	17.05
33	52	54	3	14.30	17.05
34	55	58	4	14.69	18.30
35	59	62	4	15.11	18.30
36	63	66	4	15.49	18.30
37	67	70	4	15.84	18.30
38	71	75	5	16.21	19.27
39	76	80	5	16.58	19.27
40	81	85	5	16.92	19.27
41	86	91	6	17.27	20.06
42	92	97	6	17.62	20.06
43	98	104	7	17.97	20.73
44	105	111	7	18.32	20.73
45	112	119	8	18.67	21.31
46	120	127	8	19.02	21.31

Table C.5 — Psychoacoustic parameters for 12 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	5	6	0.18	45.06
1	6	11	6	0.53	45.06
2	12	17	6	0.89	45.06
3	18	23	6	1.24	40.06
4	24	29	6	1.59	40.06
5	30	35	6	1.94	40.06
6	36	41	6	2.29	37.06
7	42	47	6	2.63	37.06
8	48	53	6	2.98	37.06
9	54	59	6	3.31	32.06
10	60	65	6	3.65	32.06
11	66	72	7	4.00	30.73
12	73	79	7	4.38	30.73
13	80	86	7	4.75	30.73
14	87	93	7	5.11	30.73
15	94	100	7	5.47	30.73
16	101	107	7	5.82	30.73
17	108	114	7	6.15	30.73
18	115	122	8	6.51	31.31
19	123	130	8	6.88	31.31
20	131	138	8	7.24	31.31
21	139	146	8	7.58	31.31
22	147	154	8	7.92	31.31
23	155	163	9	8.27	31.82
24	164	172	9	8.62	31.82
25	173	181	9	8.96	31.82
26	182	191	10	9.31	32.28
27	192	201	10	9.66	32.28
28	202	212	11	10.01	32.69
29	213	223	11	10.36	32.69
30	224	235	12	10.71	33.07
31	236	247	12	11.06	33.07
32	248	260	13	11.41	33.42
33	261	273	13	11.75	33.42
34	274	287	14	12.09	33.74
35	288	302	15	12.43	34.04
36	303	318	16	12.77	34.32
37	319	335	17	13.11	34.58
38	336	353	18	13.46	34.83
39	354	372	19	13.80	35.06
40	373	392	20	14.13	35.29
41	393	414	22	14.47	35.70
42	415	437	23	14.81	35.89
43	438	462	25	15.14	36.26
44	463	489	27	15.48	36.59
45	490	518	29	15.81	36.90
46	519	549	31	16.15	37.19
47	550	583	34	16.48	37.59
48	584	619	36	16.82	37.84
49	620	658	39	17.15	38.19
50	659	700	42	17.48	38.51
51	701	745	45	17.81	38.81
52	746	794	49	18.14	39.18
53	795	847	53	18.48	39.52
54	848	904	57	18.81	39.83
55	905	965	61	19.15	40.13
56	966	1023	58	19.47	39.91

Table C.6 — Psychoacoustic parameters for 12 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	0.47	27.28
2	2	2	1	0.95	27.28
3	3	3	1	1.42	22.28
4	4	4	1	1.88	22.28
5	5	5	1	2.35	19.28
6	6	6	1	2.81	19.28
7	7	7	1	3.26	14.28
8	8	8	1	3.70	14.28
9	9	9	1	4.14	12.28
10	10	10	1	4.57	12.28
11	11	11	1	4.98	12.28
12	12	12	1	5.39	12.28
13	13	13	1	5.79	12.28
14	14	14	1	6.18	12.28
15	15	15	1	6.56	12.28
16	16	16	1	6.93	12.28
17	17	17	1	7.28	12.28
18	18	18	1	7.63	12.28
19	19	20	2	8.28	15.29
20	21	22	2	8.90	15.29
21	23	24	2	9.48	15.29
22	25	26	2	10.02	15.29
23	27	28	2	10.53	15.29
24	29	30	2	11.00	15.29
25	31	32	2	11.45	15.29
26	33	34	2	11.86	15.29
27	35	36	2	12.25	15.29
28	37	38	2	12.62	15.29
29	39	40	2	12.96	15.29
30	41	43	3	13.36	17.05
31	44	46	3	13.80	17.05
32	47	49	3	14.21	17.05
33	50	52	3	14.59	17.05
34	53	55	3	14.94	17.05
35	56	59	4	15.32	18.30
36	60	63	4	15.71	18.30
37	64	67	4	16.08	18.30
38	68	72	5	16.45	19.27
39	73	77	5	16.83	19.27
40	78	82	5	17.19	19.27
41	83	88	6	17.54	20.06
42	89	94	6	17.90	20.06
43	95	101	7	18.26	20.73
44	102	108	7	18.62	20.73
45	109	116	8	18.97	21.31
46	117	124	8	19.32	21.31
47	125	127	3	19.55	17.05

Table C.7 — Psychoacoustic parameters for 16 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	4	5	0.20	43.30
1	5	9	5	0.59	43.10
2	10	14	5	0.99	38.30
3	15	19	5	1.38	38.10
4	20	24	5	1.77	38.00
5	25	29	5	2.16	35.10
6	30	34	5	2.54	35.30
7	35	39	5	2.92	30.00
8	40	44	5	3.29	30.00
9	45	49	5	3.66	28.30
10	50	54	5	4.03	28.30
11	55	59	5	4.39	28.30
12	60	64	5	4.74	28.30
13	65	69	5	5.09	28.30
14	70	74	5	5.43	28.30
15	75	80	6	5.79	28.30
16	81	86	6	6.18	28.30
17	87	92	6	6.56	28.00
18	93	98	6	6.92	29.27
19	99	104	6	7.28	29.27
20	105	110	6	7.63	29.27
21	111	116	6	7.96	29.27
22	117	123	7	8.31	29.27
23	124	130	7	8.68	29.06
24	131	137	7	9.03	30.06
25	138	144	7	9.37	30.06
26	145	152	8	9.71	30.06
27	153	160	8	10.07	30.73
28	161	168	8	10.41	30.73
29	169	177	9	10.75	30.73
30	178	186	9	11.10	31.31
31	187	196	10	11.45	31.31
32	197	206	10	11.80	31.82
33	207	217	11	12.14	31.82
34	218	228	11	12.48	32.28
35	229	240	12	12.82	32.28
36	241	253	13	13.16	32.69
37	254	267	14	13.51	32.69
38	268	282	15	13.86	33.07
39	283	298	16	14.21	33.46
40	299	315	17	14.56	33.82
41	316	333	18	14.90	34.12
42	334	352	19	15.24	34.42
43	353	373	21	15.58	34.68
44	374	395	22	15.91	35.15
45	396	419	24	16.25	35.32
46	420	445	26	16.58	35.73
47	446	473	28	16.92	35.91
48	474	503	30	17.25	36.42
49	504	536	33	17.59	36.75
50	537	571	35	17.93	37.11
51	572	609	38	18.26	37.34
52	610	650	41	18.60	37.63
53	651	694	44	18.94	38.12

54	695	741	47	19.27	38.17
55	742	791	50	19.60	41.52
56	792	845	54	19.94	41.84
57	846	903	58	20.27	42.13
58	904	965	62	20.61	44.41
59	966	1023	58	20.92	44.87

Table C.8 — Psychoacoustic parameters for 16 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	0.63	27.28
2	2	2	1	1.26	22.28
3	3	3	1	1.88	22.28
4	4	4	1	2.50	19.28
5	5	5	1	3.11	14.28
6	6	6	1	3.70	14.28
7	7	7	1	4.28	12.28
8	8	8	1	4.85	12.28
9	9	9	1	5.39	12.28
10	10	10	1	5.92	12.28
11	11	11	1	6.43	12.28
12	12	12	1	6.93	12.28
13	13	13	1	7.40	12.28
14	14	14	1	7.85	12.28
15	15	15	1	8.29	12.28
16	16	16	1	8.70	12.28
17	17	17	1	9.10	12.28
18	18	18	1	9.49	12.28
19	19	19	1	9.85	12.28
20	20	20	1	10.20	12.28
21	21	22	2	10.85	15.29
22	23	24	2	11.44	15.29
23	25	26	2	11.99	15.29
24	27	28	2	12.50	15.29
25	29	30	2	12.96	15.29
26	31	32	2	13.39	15.29
27	33	34	2	13.78	15.29
28	35	36	2	14.15	15.29
29	37	39	3	14.57	17.05
30	40	42	3	15.03	17.05
31	43	45	3	15.45	17.05
32	46	48	3	15.84	17.05
33	49	51	3	16.19	17.05
34	52	55	4	16.57	18.30
35	56	59	4	16.97	18.30
36	60	63	4	17.33	18.30
37	64	68	5	17.71	19.27
38	69	73	5	18.09	19.27
39	74	78	5	18.44	19.27
40	79	84	6	18.80	20.06
41	85	90	6	19.17	20.06
42	91	97	7	19.53	20.73
43	98	104	7	19.89	20.73
44	105	112	8	20.25	24.31
45	113	120	8	20.61	24.31
46	121	127	7	20.92	23.73

Table C.9 — Psychoacoustic parameters for 22.05 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	3	4	0.22	43.30
1	4	7	4	0.65	43.30
2	8	11	4	1.09	38.30
3	12	15	4	1.52	38.30
4	16	19	4	1.95	38.30
5	20	23	4	2.37	35.30
6	24	27	4	2.79	35.30
7	28	31	4	3.21	30.30
8	32	35	4	3.62	30.30
9	36	39	4	4.02	28.30
10	40	43	4	4.41	28.30
11	44	47	4	4.80	28.30
12	48	51	4	5.18	28.30
13	52	55	4	5.55	28.30
14	56	59	4	5.92	28.30
15	60	63	4	6.27	28.30
16	64	67	4	6.62	28.30
17	68	71	4	6.95	28.30
18	72	76	5	7.32	29.27
19	77	81	5	7.71	29.27
20	82	86	5	8.10	29.27
21	87	91	5	8.46	29.27
22	92	96	5	8.82	29.27
23	97	101	5	9.16	29.27
24	102	107	6	9.52	30.06
25	108	113	6	9.89	30.06
26	114	119	6	10.25	30.06
27	120	125	6	10.59	30.06
28	126	132	7	10.95	30.73
29	133	139	7	11.31	30.73
30	140	146	7	11.65	30.73
31	147	154	8	12.00	31.31
32	155	162	8	12.35	31.31
33	163	171	9	12.70	31.82
34	172	180	9	13.05	31.82
35	181	190	10	13.40	32.28
36	191	200	10	13.74	32.28
37	201	211	11	14.07	32.69
38	212	223	12	14.41	33.07
39	224	236	13	14.76	33.42
40	237	250	14	15.11	33.74
41	251	265	15	15.46	34.04
42	266	281	16	15.80	34.32
43	282	298	17	16.14	34.58
44	299	317	19	16.48	35.06
45	318	337	20	16.82	35.29
46	338	359	22	17.16	35.70
47	360	382	23	17.50	35.89
48	383	407	25	17.84	36.26
49	408	434	27	18.17	36.59
50	435	463	29	18.51	36.90
51	464	494	31	18.84	37.19
52	495	527	33	19.17	37.46
53	528	563	36	19.51	37.84
54	564	601	38	19.84	38.07
55	602	642	41	20.17	41.40
56	643	686	44	20.50	41.71
57	687	733	47	20.84	42.00
58	734	784	51	21.17	44.35

59	785	839	55	21.50	44.68
60	840	898	59	21.84	44.98
61	899	962	64	22.17	50.34
62	963	1023	61	22.48	50.13

Table C.10 — Psychoacoustic parameters for 22.05 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	0.87	27.28
2	2	2	1	1.73	22.28
3	3	3	1	2.58	19.28
4	4	4	1	3.41	14.28
5	5	5	1	4.22	12.28
6	6	6	1	4.99	12.28
7	7	7	1	5.74	12.28
8	8	8	1	6.45	12.28
9	9	9	1	7.12	12.28
10	10	10	1	7.75	12.28
11	11	11	1	8.36	12.28
12	12	12	1	8.92	12.28
13	13	13	1	9.45	12.28
14	14	14	1	9.96	12.28
15	15	15	1	10.43	12.28
16	16	16	1	10.87	12.28
17	17	17	1	11.29	12.28
18	18	18	1	11.68	12.28
19	19	19	1	12.05	12.28
20	20	21	2	12.71	15.29
21	22	23	2	13.32	15.29
22	24	25	2	13.86	15.29
23	26	27	2	14.35	15.29
24	28	29	2	14.80	15.29
25	30	31	2	15.21	15.29
26	32	33	2	15.58	15.29
27	34	35	2	15.93	15.29
28	36	38	3	16.32	17.05
29	39	41	3	16.75	17.05
30	42	44	3	17.15	17.05
31	45	47	3	17.51	17.05
32	48	51	4	17.89	18.30
33	52	55	4	18.30	18.30
34	56	59	4	18.67	18.30
35	60	63	4	19.02	18.30
36	64	68	5	19.37	19.27
37	69	73	5	19.74	19.27
38	74	78	5	20.09	22.27
39	79	84	6	20.44	23.06
40	85	90	6	20.79	23.06
41	91	97	7	21.15	25.73
42	98	104	7	21.50	25.73
43	105	112	8	21.85	26.31
44	113	120	8	22.20	31.31
45	121	127	7	22.49	30.73

Table C.11 — Psychoacoustic parameters for 24 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	2	3	0.18	42.05
1	3	5	3	0.53	42.05
2	6	8	3	0.89	42.05
3	9	11	3	1.24	37.05
4	12	14	3	1.59	37.05
5	15	17	3	1.94	37.05
6	18	20	3	2.29	34.05
7	21	23	3	2.63	34.05
8	24	26	3	2.98	34.05
9	27	29	3	3.31	29.05
10	30	32	3	3.65	29.05
11	33	36	4	4.03	28.30
12	37	40	4	4.46	28.30
13	41	44	4	4.88	28.30
14	45	48	4	5.29	28.30
15	49	52	4	5.69	28.30
16	53	56	4	6.08	28.30
17	57	60	4	6.46	28.30
18	61	64	4	6.83	28.30
19	65	68	4	7.19	28.30
20	69	72	4	7.54	28.30
21	73	76	4	7.88	28.30
22	77	81	5	8.25	29.27
23	82	86	5	8.64	29.27
24	87	91	5	9.02	29.27
25	92	96	5	9.38	29.27
26	97	101	5	9.73	29.27
27	102	107	6	10.09	30.06
28	108	113	6	10.47	30.06
29	114	119	6	10.83	30.06
30	120	125	6	11.18	30.06
31	126	132	7	11.53	30.73
32	133	139	7	11.89	30.73
33	140	146	7	12.23	30.73
34	147	154	8	12.57	31.31
35	155	162	8	12.92	31.31
36	163	171	9	13.26	31.82
37	172	180	9	13.61	31.82
38	181	190	10	13.95	32.28
39	191	201	11	14.29	32.69
40	202	213	12	14.65	33.07
41	214	225	12	15.00	33.07
42	226	238	13	15.33	33.42
43	239	252	14	15.66	33.74
44	253	267	15	16.00	34.04
45	268	284	17	16.34	34.58
46	285	302	18	16.69	34.83
47	303	321	19	17.02	35.06
48	322	342	21	17.36	35.50
49	343	364	22	17.70	35.70
50	365	388	24	18.03	36.08
51	389	414	26	18.37	36.43
52	415	442	28	18.70	36.75
53	443	472	30	19.04	37.05

54	473	504	32	19.38	37.33
55	505	538	34	19.71	37.59
56	539	575	37	20.04	40.96
57	576	614	39	20.38	41.19
58	615	656	42	20.71	41.51
59	657	701	45	21.04	43.81
60	702	750	49	21.37	44.18
61	751	803	53	21.70	44.52
62	804	860	57	22.04	49.83
63	861	922	62	22.37	50.20
64	923	989	67	22.70	50.54
65	990	1023	34	22.95	47.59

Table C.12 — Psychoacoustic parameters for 24 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	0.95	27.28
2	2	2	1	1.88	22.28
3	3	3	1	2.81	19.28
4	4	4	1	3.70	14.28
5	5	5	1	4.57	12.28
6	6	6	1	5.39	12.28
7	7	7	1	6.18	12.28
8	8	8	1	6.93	12.28
9	9	9	1	7.63	12.28
10	10	10	1	8.29	12.28
11	11	11	1	8.91	12.28
12	12	12	1	9.49	12.28
13	13	13	1	10.03	12.28
14	14	14	1	10.53	12.28
15	15	15	1	11.01	12.28
16	16	16	1	11.45	12.28
17	17	17	1	11.87	12.28
18	18	18	1	12.26	12.28
19	19	19	1	12.62	12.28
20	20	21	2	13.28	15.29
21	22	23	2	13.87	15.29
22	24	25	2	14.40	15.29
23	26	27	2	14.88	15.29
24	28	29	2	15.32	15.29
25	30	31	2	15.71	15.29
26	32	33	2	16.08	15.29
27	34	36	3	16.49	17.05
28	37	39	3	16.94	17.05
29	40	42	3	17.35	17.05
30	43	45	3	17.73	17.05
31	46	48	3	18.07	17.05
32	49	52	4	18.44	18.30
33	53	56	4	18.83	18.30
34	57	60	4	19.20	18.30
35	61	65	5	19.57	19.27
36	66	70	5	19.96	19.27
37	71	75	5	20.31	22.27
38	76	81	6	20.67	23.06
39	82	87	6	21.04	25.06
40	88	94	7	21.41	25.73
41	95	101	7	21.77	25.73
42	102	109	8	22.13	31.31
43	110	117	8	22.48	31.31
44	118	126	9	22.82	31.82
45	127	127	1	23.01	32.28

Table C.13 — Psychoacoustic parameters for 32 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	2	3	0.24	42.05
1	3	5	3	0.71	42.05
2	6	8	3	1.18	37.05
3	9	11	3	1.65	37.05
4	12	14	3	2.12	34.05
5	15	17	3	2.58	34.05
6	18	20	3	3.03	29.05
7	21	23	3	3.48	29.05
8	24	26	3	3.92	29.05
9	27	29	3	4.35	27.05
10	30	32	3	4.77	27.05
11	33	35	3	5.19	27.05
12	36	38	3	5.59	27.05
13	39	41	3	5.99	27.05
14	42	44	3	6.37	27.05
15	45	47	3	6.74	27.05
16	48	50	3	7.10	27.05
17	51	53	3	7.45	27.05
18	54	56	3	7.80	27.05
19	57	60	4	8.18	28.30
20	61	64	4	8.60	28.30
21	65	68	4	9.00	28.30
22	69	72	4	9.39	28.30
23	73	76	4	9.76	28.30
24	77	80	4	10.11	28.30
25	81	84	4	10.45	28.30
26	85	89	5	10.81	29.27
27	90	94	5	11.19	29.27
28	95	99	5	11.55	29.27
29	100	104	5	11.90	29.27
30	105	110	6	12.25	30.06
31	111	116	6	12.62	30.06
32	117	122	6	12.96	30.06
33	123	129	7	13.31	30.73
34	130	136	7	13.66	30.73
35	137	144	8	14.01	31.31
36	145	152	8	14.36	31.31
37	153	161	9	14.71	31.82
38	162	171	10	15.07	32.28
39	172	181	10	15.42	32.28
40	182	192	11	15.76	32.69
41	193	204	12	16.10	33.07
42	205	217	13	16.45	33.42
43	218	231	14	16.80	33.74
44	232	246	15	17.14	34.04
45	247	262	16	17.48	34.32
46	263	279	17	17.82	34.58
47	280	298	19	18.15	35.06
48	299	318	20	18.49	35.29
49	319	340	22	18.84	35.70
50	341	363	23	19.17	35.89
51	364	388	25	19.51	36.26
52	389	415	27	19.85	36.59
53	416	444	29	20.19	39.90
54	445	475	31	20.53	40.19
55	476	508	33	20.87	40.46
56	509	543	35	21.20	42.72
57	544	581	38	21.53	43.07
58	582	622	41	21.86	43.40

59	623	667	45	22.20	48.81
60	668	715	48	22.53	49.09
61	716	768	53	22.86	49.52
62	769	826	58	23.20	59.91
63	827	890	64	23.53	60.34
64	891	961	71	23.86	60.79
65	962	1023	62	24.00	65.89

Table C.14 — Psychoacoustic parameters for 32 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	1.26	22.28
2	2	2	1	2.50	19.28
3	3	3	1	3.70	14.28
4	4	4	1	4.85	12.28
5	5	5	1	5.92	12.28
6	6	6	1	6.93	12.28
7	7	7	1	7.85	12.28
8	8	8	1	8.70	12.28
9	9	9	1	9.49	12.28
10	10	10	1	10.20	12.28
11	11	11	1	10.85	12.28
12	12	12	1	11.45	12.28
13	13	13	1	12.00	12.28
14	14	14	1	12.50	12.28
15	15	15	1	12.96	12.28
16	16	16	1	13.39	12.28
17	17	17	1	13.78	12.28
18	18	18	1	14.15	12.28
19	19	20	2	14.80	15.29
20	21	22	2	15.38	15.29
21	23	24	2	15.89	15.29
22	25	26	2	16.36	15.29
23	27	28	2	16.77	15.29
24	29	30	2	17.15	15.29
25	31	32	2	17.50	15.29
26	33	35	3	17.90	17.05
27	36	38	3	18.34	17.05
28	39	41	3	18.74	17.05
29	42	44	3	19.11	17.05
30	45	48	4	19.50	18.30
31	49	52	4	19.92	18.30
32	53	56	4	20.30	21.30
33	57	60	4	20.65	21.30
34	61	65	5	21.02	24.27
35	66	70	5	21.40	24.27
36	71	75	5	21.75	24.27
37	76	81	6	22.10	30.06
38	82	87	6	22.45	30.06
39	88	94	7	22.80	30.73
40	95	102	8	23.16	41.31
41	103	110	8	23.51	41.31
42	111	119	9	23.85	41.82
43	120	127	8	24.00	60.47

Table C.15 – Psychoacoustic parameters for 44.1 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	1	2	0.22	40.29
1	2	3	2	0.65	40.29
2	4	5	2	1.09	35.29
3	6	7	2	1.52	35.29
4	8	9	2	1.95	35.29
5	10	11	2	2.37	32.29
6	12	13	2	2.79	32.29
7	14	15	2	3.21	27.29
8	16	17	2	3.62	27.29
9	18	19	2	4.02	25.29
10	20	21	2	4.41	25.29
11	22	23	2	4.80	25.29
12	24	25	2	5.18	25.29
13	26	27	2	5.55	25.29
14	28	29	2	5.92	25.29
15	30	31	2	6.27	25.29
16	32	33	2	6.62	25.29
17	34	35	2	6.95	25.29
18	36	38	3	7.36	27.05
19	39	41	3	7.83	27.05
20	42	44	3	8.28	27.05
21	45	47	3	8.71	27.05
22	48	50	3	9.12	27.05
23	51	53	3	9.52	27.05
24	54	56	3	9.89	27.05
25	57	59	3	10.25	27.05
26	60	62	3	10.59	27.05
27	63	66	4	10.97	28.30
28	67	70	4	11.38	28.30
29	71	74	4	11.77	28.30
30	75	78	4	12.13	28.30
31	79	82	4	12.48	28.30
32	83	87	5	12.84	29.27
33	88	92	5	13.22	29.27
34	93	97	5	13.57	29.27
35	98	103	6	13.93	30.06
36	104	109	6	14.30	30.06
37	110	116	7	14.67	30.73
38	117	123	7	15.03	30.73
39	124	131	8	15.40	31.31
40	132	139	8	15.76	31.31
41	140	148	9	16.11	31.82
42	149	157	9	16.45	31.82
43	158	167	10	16.79	32.28
44	168	178	11	17.13	32.69
45	179	190	12	17.48	33.07
46	191	203	13	17.83	33.42
47	204	217	14	18.18	33.74
48	218	232	15	18.52	34.04
49	233	248	16	18.87	34.32
50	249	265	17	19.21	34.58
51	266	283	18	19.54	34.83
52	284	303	20	19.88	35.29
53	304	324	21	20.22	38.50

54	325	347	23	20.56	38.89
55	348	371	24	20.90	39.08
56	372	397	26	21.24	41.43
57	398	425	28	21.57	41.75
58	426	455	30	21.91	42.05
59	456	488	33	22.24	47.46
60	489	524	36	22.58	47.84
61	525	563	39	22.91	48.19
62	564	606	43	23.25	58.61
63	607	653	47	23.58	59.00
64	654	706	53	23.91	59.52
65	707	765	59	24.00	69.98
66	766	832	67	24.00	70.54
67	833	908	76	24.00	71.08
68	909	996	88	24.00	71.72
69	997	1023	27	24.00	72.09

Table C.16 — Psychoacoustic parameters for 44.1 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	1.73	22.28
2	2	2	1	3.41	14.28
3	3	3	1	4.99	12.28
4	4	4	1	6.45	12.28
5	5	5	1	7.75	12.28
6	6	6	1	8.92	12.28
7	7	7	1	9.96	12.28
8	8	8	1	10.87	12.28
9	9	9	1	11.68	12.28
10	10	10	1	12.39	12.28
11	11	11	1	13.03	12.28
12	12	12	1	13.61	12.28
13	13	13	1	14.12	12.28
14	14	14	1	14.59	12.28
15	15	15	1	15.01	12.28
16	16	16	1	15.40	12.28
17	17	17	1	15.76	12.28
18	18	19	2	16.39	15.29
19	20	21	2	16.95	15.29
20	22	23	2	17.45	15.29
21	24	25	2	17.89	15.29
22	26	27	2	18.30	15.29
23	28	29	2	18.67	15.29
24	30	31	2	19.02	15.29
25	32	34	3	19.41	17.05
26	35	37	3	19.85	17.05
27	38	40	3	20.25	20.05
28	41	43	3	20.62	20.05
29	44	47	4	21.01	23.30
30	48	51	4	21.43	23.30
31	52	55	4	21.81	23.30
32	56	59	4	22.15	28.30
33	60	64	5	22.51	29.27
34	65	69	5	22.87	29.27
35	70	75	6	23.23	40.06
36	76	81	6	23.59	40.06
37	82	88	7	23.93	40.73
38	89	96	8	24.00	51.31
39	97	105	9	24.00	51.82
40	106	115	10	24.00	52.28
41	116	127	12	24.00	53.07

Table C.17 — Psychoacoustic parameters for 48 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	1	2	0.24	40.29
1	2	3	2	0.71	40.29
2	4	5	2	1.18	35.29
3	6	7	2	1.65	35.29
4	8	9	2	2.12	32.29
5	10	11	2	2.58	32.29
6	12	13	2	3.03	27.29
7	14	15	2	3.48	27.29
8	16	17	2	3.92	27.29
9	18	19	2	4.35	25.29
10	20	21	2	4.77	25.29
11	22	23	2	5.19	25.29
12	24	25	2	5.59	25.29
13	26	27	2	5.99	25.29
14	28	29	2	6.37	25.29
15	30	31	2	6.74	25.29
16	32	33	2	7.10	25.29
17	34	35	2	7.45	25.29
18	36	37	2	7.80	25.29
19	38	40	3	8.20	27.05
20	41	43	3	8.68	27.05
21	44	46	3	9.13	27.05
22	47	49	3	9.55	27.05
23	50	52	3	9.96	27.05
24	53	55	3	10.35	27.05
25	56	58	3	10.71	27.05
26	59	61	3	11.06	27.05
27	62	65	4	11.45	28.30
28	66	69	4	11.86	28.30
29	70	73	4	12.25	28.30
30	74	77	4	12.62	28.30
31	78	81	4	12.96	28.30
32	82	86	5	13.32	29.27
33	87	91	5	13.70	29.27
34	92	96	5	14.05	29.27
35	97	102	6	14.41	30.06
36	103	108	6	14.77	30.06
37	109	115	7	15.13	30.73
38	116	122	7	15.49	30.73
39	123	130	8	15.85	31.31
40	131	138	8	16.20	31.31
41	139	147	9	16.55	31.82
42	148	157	10	16.91	32.28
43	158	167	10	17.25	32.28
44	168	178	11	17.59	32.69
45	179	190	12	17.93	33.07
46	191	203	13	18.28	33.42
47	204	217	14	18.62	33.74
48	218	232	15	18.96	34.04
49	233	248	16	19.30	34.32
50	249	265	17	19.64	34.58
51	266	283	18	19.97	34.83
52	284	303	20	20.31	38.29
53	304	324	21	20.65	38.50

54	325	347	23	20.99	38.89
55	348	371	24	21.33	41.08
56	372	397	26	21.66	41.43
57	398	425	28	21.99	41.75
58	426	456	31	22.32	47.19
59	457	490	34	22.66	47.59
60	491	527	37	23.00	47.96
61	528	567	40	23.33	58.30
62	568	612	45	23.67	58.81
63	613	662	50	24.00	69.27
64	663	718	56	24.00	69.76
65	719	781	63	24.00	70.27
66	782	853	72	24.00	70.85
67	854	937	84	24.00	71.52
68	938	1023	86	24.00	70.20

Table C.18 — Psychoacoustic parameters for 48 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	1.88	22.28
2	2	2	1	3.70	14.28
3	3	3	1	5.39	12.28
4	4	4	1	6.93	12.28
5	5	5	1	8.29	12.28
6	6	6	1	9.49	12.28
7	7	7	1	10.53	12.28
8	8	8	1	11.45	12.28
9	9	9	1	12.26	12.28
10	10	10	1	12.96	12.28
11	11	11	1	13.59	12.28
12	12	12	1	14.15	12.28
13	13	13	1	14.65	12.28
14	14	14	1	15.11	12.28
15	15	15	1	15.52	12.28
16	16	16	1	15.90	12.28
17	17	18	2	16.56	15.29
18	19	20	2	17.15	15.29
19	21	22	2	17.66	15.29
20	23	24	2	18.13	15.29
21	25	26	2	18.54	15.29
22	27	28	2	18.93	15.29
23	29	30	2	19.28	15.29
24	31	33	3	19.69	17.05
25	34	36	3	20.14	20.05
26	37	39	3	20.54	20.05
27	40	42	3	20.92	20.05
28	43	45	3	21.27	22.05
29	46	49	4	21.64	23.30
30	50	53	4	22.03	28.30
31	54	57	4	22.39	28.30
32	58	62	5	22.76	29.27
33	63	67	5	23.13	39.27
34	68	73	6	23.49	40.06
35	74	79	6	23.85	40.06
36	80	86	7	24.00	50.73
37	87	94	8	24.00	51.31
38	95	103	9	24.00	51.82
39	104	113	10	24.00	52.28
40	114	125	12	24.00	53.07
41	126	127	1	24.00	53.07

Table C.19 — Psychoacoustic parameters for 64 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	1	2	0.32	40.29
1	2	3	2	0.95	40.29
2	4	5	2	1.57	35.29
3	6	7	2	2.19	32.29
4	8	9	2	2.80	32.29
5	10	11	2	3.40	27.29
6	12	13	2	3.99	27.29
7	14	15	2	4.56	25.29
8	16	17	2	5.12	25.29
9	18	19	2	5.66	25.29
10	20	21	2	6.18	25.29
11	22	23	2	6.68	25.29
12	24	25	2	7.16	25.29
13	26	27	2	7.63	25.29
14	28	29	2	8.07	25.29
15	30	31	2	8.50	25.29
16	32	33	2	8.90	25.29
17	34	35	2	9.29	25.29
18	36	37	2	9.67	25.29
19	38	39	2	10.03	25.29
20	40	41	2	10.37	25.29
21	42	44	3	10.77	27.05
22	45	47	3	11.23	27.05
23	48	50	3	11.66	27.05
24	51	53	3	12.06	27.05
25	54	56	3	12.44	27.05
26	57	59	3	12.79	27.05
27	60	63	4	13.18	28.30
28	64	67	4	13.59	28.30
29	68	71	4	13.97	28.30
30	72	75	4	14.32	28.30
31	76	80	5	14.69	29.27
32	81	85	5	15.07	29.27
33	86	90	5	15.42	29.27
34	91	96	6	15.77	30.06
35	97	102	6	16.13	30.06
36	103	109	7	16.49	30.73
37	110	116	7	16.85	30.73
38	117	124	8	17.20	31.31
39	125	132	8	17.54	31.31
40	133	141	9	17.88	31.82
41	142	151	10	18.23	32.28
42	152	161	10	18.58	32.28
43	162	172	11	18.91	32.69
44	173	184	12	19.25	33.07
45	185	197	13	19.60	33.42
46	198	211	14	19.94	33.74
47	212	226	15	20.29	37.04
48	227	242	16	20.63	37.32
49	243	259	17	20.97	37.58
50	260	277	18	21.31	39.83
51	278	297	20	21.64	40.29
52	298	318	21	21.98	40.50
53	319	341	23	22.31	45.89

54	342	366	25	22.65	46.26
55	367	394	28	22.98	46.75
56	395	424	30	23.32	57.05
57	425	458	34	23.66	57.59
58	459	495	37	23.99	57.96
59	496	537	42	24.00	68.51
60	538	584	47	24.00	69.00
61	585	638	54	24.00	69.60
62	639	701	63	24.00	70.27
63	702	774	73	24.00	70.91
64	775	861	87	24.00	71.67
65	862	966	105	24.00	72.49
66	967	1023	57	24.00	69.83

Table C.20 – Psychoacoustic parameters for 64 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	2.50	19.28
2	2	2	1	4.85	12.28
3	3	3	1	6.93	12.28
4	4	4	1	8.70	12.28
5	5	5	1	10.20	12.28
6	6	6	1	11.45	12.28
7	7	7	1	12.50	12.28
8	8	8	1	13.39	12.28
9	9	9	1	14.15	12.28
10	10	10	1	14.81	12.28
11	11	11	1	15.39	12.28
12	12	12	1	15.90	12.28
13	13	13	1	16.36	12.28
14	14	14	1	16.78	12.28
15	15	15	1	17.16	12.28
16	16	17	2	17.82	15.29
17	18	19	2	18.40	15.29
18	20	21	2	18.92	15.29
19	22	23	2	19.39	15.29
20	24	25	2	19.82	15.29
21	26	27	2	20.21	18.29
22	28	29	2	20.57	18.29
23	30	32	3	20.98	20.05
24	33	35	3	21.43	22.05
25	36	38	3	21.84	22.05
26	39	41	3	22.22	27.05
27	42	45	4	22.61	28.30
28	46	49	4	23.02	38.30
29	50	53	4	23.39	38.30
30	54	58	5	23.75	39.27
31	59	63	5	24.00	49.27
32	64	69	6	24.00	50.06
33	70	76	7	24.00	50.73
34	77	84	8	24.00	51.31
35	85	93	9	24.00	51.82
36	94	104	11	24.00	52.69
37	105	117	13	24.00	53.42
38	118	127	10	24.00	52.28

Table C.21 — Psychoacoustic parameters for 88.2 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	37.28
1	1	1	1	0.44	37.28
2	2	2	1	0.87	37.28
3	3	3	1	1.30	32.28
4	4	4	1	1.73	32.28
5	5	5	1	2.16	29.28
6	6	6	1	2.58	29.28
7	7	7	1	3.00	24.28
8	8	8	1	3.41	24.28
9	9	9	1	3.82	24.28
10	10	10	1	4.22	22.28
11	11	11	1	4.61	22.28
12	12	12	1	4.99	22.28
13	13	13	1	5.37	22.28
14	14	14	1	5.74	22.28
15	15	15	1	6.10	22.28
16	16	16	1	6.45	22.28
17	17	17	1	6.79	22.28
18	18	19	2	7.44	25.29
19	20	21	2	8.05	25.29
20	22	23	2	8.64	25.29
21	24	25	2	9.19	25.29
22	26	27	2	9.70	25.29
23	28	29	2	10.19	25.29
24	30	31	2	10.65	25.29
25	32	33	2	11.08	25.29
26	34	35	2	11.48	25.29
27	36	37	2	11.86	25.29
28	38	39	2	12.22	25.29
29	40	42	3	12.64	27.05
30	43	45	3	13.10	27.05
31	46	48	3	13.53	27.05
32	49	51	3	13.93	27.05
33	52	54	3	14.30	27.05
34	55	58	4	14.69	28.30
35	59	62	4	15.11	28.30
36	63	66	4	15.49	28.30
37	67	70	4	15.84	28.30
38	71	75	5	16.21	29.27
39	76	80	5	16.58	29.27
40	81	85	5	16.92	29.27
41	86	91	6	17.27	30.06
42	92	97	6	17.62	30.06
43	98	104	7	17.97	30.73
44	105	111	7	18.32	30.73
45	112	119	8	18.67	31.31
46	120	127	8	19.02	31.31
47	128	136	9	19.35	31.82
48	137	146	10	19.71	32.28
49	147	156	10	20.05	35.28
50	157	167	11	20.39	35.69
51	168	179	12	20.73	36.07
52	180	192	13	21.08	38.42
53	193	206	14	21.43	38.74

54	207	221	15	21.77	39.04
55	222	237	16	22.11	44.32
56	238	255	18	22.45	44.83
57	256	274	19	22.80	45.06
58	275	295	21	23.13	55.50
59	296	318	23	23.47	55.89
60	319	344	26	23.81	56.43
61	345	373	29	24.00	66.90
62	374	405	32	24.00	67.33
63	406	442	37	24.00	67.96
64	443	484	42	24.00	68.51
65	485	533	49	24.00	69.18
66	534	591	58	24.00	69.91
67	592	660	69	24.00	70.66
68	661	745	85	24.00	71.57
69	746	851	106	24.00	72.53
70	852	988	137	24.00	73.64
71	989	1023	35	24.00	67.72

Table C.22 — Psychoacoustic parameters for 88.2 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	3.41	14.28
2	2	2	1	6.45	12.28
3	3	3	1	8.92	12.28
4	4	4	1	10.87	12.28
5	5	5	1	12.39	12.28
6	6	6	1	13.61	12.28
7	7	7	1	14.59	12.28
8	8	8	1	15.40	12.28
9	9	9	1	16.09	12.28
10	10	10	1	16.69	12.28
11	11	11	1	17.21	12.28
12	12	12	1	17.68	12.28
13	13	13	1	18.11	12.28
14	14	14	1	18.49	12.28
15	15	15	1	18.85	12.28
16	16	17	2	19.48	15.29
17	18	19	2	20.05	18.29
18	20	21	2	20.55	18.29
19	22	23	2	21.01	20.29
20	24	25	2	21.43	20.29
21	26	27	2	21.81	20.29
22	28	29	2	22.15	25.29
23	30	32	3	22.55	27.05
24	33	35	3	22.98	27.05
25	36	38	3	23.36	37.05
26	39	42	4	23.75	38.30
27	43	46	4	24.00	48.30
28	47	51	5	24.00	49.27
29	52	56	5	24.00	49.27
30	57	62	6	24.00	50.06
31	63	69	7	24.00	50.73
32	70	77	8	24.00	51.31
33	78	87	10	24.00	52.28
34	88	99	12	24.00	53.07
35	100	115	16	24.00	54.32
36	116	127	12	24.00	53.07

Table C.23 — Psychoacoustic parameters for 96 kHz long FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	37.28
1	1	1	1	0.47	37.28
2	2	2	1	0.95	37.28
3	3	3	1	1.42	32.28
4	4	4	1	1.88	32.28
5	5	5	1	2.35	29.28
6	6	6	1	2.81	29.28
7	7	7	1	3.26	24.28
8	8	8	1	3.70	24.28
9	9	9	1	4.14	22.28
10	10	10	1	4.57	22.28
11	11	11	1	4.98	22.28
12	12	12	1	5.39	22.28
13	13	13	1	5.79	22.28
14	14	14	1	6.18	22.28
15	15	15	1	6.56	22.28
16	16	16	1	6.93	22.28
17	17	17	1	7.28	22.28
18	18	18	1	7.63	22.28
19	19	20	2	8.28	25.29
20	21	22	2	8.90	25.29
21	23	24	2	9.48	25.29
22	25	26	2	10.02	25.29
23	27	28	2	10.53	25.29
24	29	30	2	11.00	25.29
25	31	32	2	11.45	25.29
26	33	34	2	11.86	25.29
27	35	36	2	12.25	25.29
28	37	38	2	12.62	25.29
29	39	40	2	12.96	25.29
30	41	43	3	13.36	27.05
31	44	46	3	13.80	27.05
32	47	49	3	14.21	27.05
33	50	52	3	14.59	27.05
34	53	55	3	14.94	27.05
35	56	59	4	15.32	28.30
36	60	63	4	15.71	28.30
37	64	67	4	16.08	28.30
38	68	72	5	16.45	29.27
39	73	77	5	16.83	29.27
40	78	82	5	17.19	29.27
41	83	88	6	17.54	30.06
42	89	94	6	17.90	30.06
43	95	101	7	18.26	30.73
44	102	108	7	18.62	30.73
45	109	116	8	18.97	31.31
46	117	124	8	19.32	31.31
47	125	133	9	19.67	31.82
48	134	143	10	20.03	35.28
49	144	153	10	20.38	35.28
50	154	164	11	20.72	35.69
51	165	176	12	21.07	38.07
52	177	189	13	21.42	38.42
53	190	203	14	21.77	38.74

54	204	218	15	22.12	44.04
55	219	234	16	22.46	44.32
56	235	252	18	22.80	44.83
57	253	271	19	23.14	55.06
58	272	292	21	23.47	55.50
59	293	316	24	23.81	56.08
60	317	342	26	24.00	66.43
61	343	372	30	24.00	67.05
62	373	406	34	24.00	67.59
63	407	445	39	24.00	68.19
64	446	490	45	24.00	68.81
65	491	543	53	24.00	69.52
66	544	607	64	24.00	70.34
67	608	685	78	24.00	71.20
68	686	783	98	24.00	72.19
69	784	910	127	24.00	73.31
70	911	1023	113	24.00	72.81

Table C.24 — Psychoacoustic parameters for 96 kHz short FFT

index	w_low	w_high	width	bval	qsthr
0	0	0	1	0.00	27.28
1	1	1	1	3.70	14.28
2	2	2	1	6.93	12.28
3	3	3	1	9.49	12.28
4	4	4	1	11.45	12.28
5	5	5	1	12.96	12.28
6	6	6	1	14.15	12.28
7	7	7	1	15.11	12.28
8	8	8	1	15.90	12.28
9	9	9	1	16.57	12.28
10	10	10	1	17.16	12.28
11	11	11	1	17.67	12.28
12	12	12	1	18.13	12.28
13	13	13	1	18.55	12.28
14	14	14	1	18.93	12.28
15	15	16	2	19.60	15.29
16	17	18	2	20.20	18.29
17	19	20	2	20.73	18.29
18	21	22	2	21.21	20.29
19	23	24	2	21.64	20.29
20	25	26	2	22.03	25.29
21	27	28	2	22.39	25.29
22	29	31	3	22.79	27.05
23	32	34	3	23.23	37.05
24	35	37	3	23.62	37.05
25	38	41	4	24.00	48.30
26	42	45	4	24.00	48.30
27	46	50	5	24.00	49.27
28	51	55	5	24.00	49.27
29	56	61	6	24.00	50.06
30	62	68	7	24.00	50.73
31	69	77	9	24.00	51.82
32	78	88	11	24.00	52.69
33	89	102	14	24.00	53.74
34	103	120	18	24.00	54.83
35	121	127	7	24.00	50.73

C.2 Gain Control

C.2.1 Encoding Process

The gain control tool consists of a PQF (Polyphase Quadrature Filter), gain detectors and gain modifiers. This tool receives the input time-domain signals and **window_sequence**, and then outputs gain_control_data and a gain controlled signal whose length is equal to the length of the MDCT window. The block diagram for the gain control tool is shown in Figure C.2 .

Due to the characteristics of the PQF filterbank, the order of the MDCT coefficients in each even PQF band needs to be reversed. This is done by reversing the spectral order of the MDCT coefficients, i.e. exchanging the higher frequency MDCT coefficients with the lower frequency MDCT coefficients.

If the gain control tool is used, the configuration of the filterbank tool is changed as follows. In the case of an EIGHT_SHORT_SEQUENCE window_sequence, the number of coefficients for the MDCT is 32 instead of

128 and eight MDCTs are carried out. In the case of other window_sequence values, the number of coefficients for the MDCT is 256 instead of 1024 and one MDCT is carried out. In all cases, the filter bank tool receives a total of 2048 gain controlled signal values per frame, because the input samples have been overlapped.

C.2.1.1 PQF

The input signal is divided by a PQF into four equal width frequency bands. The coefficients of each band PQF are given as follows.

$$h_i(n) = \frac{1}{4}\cos\left(\frac{(2i+1)(2n+5)\pi}{16}\right)Q(n), 0 \le n \le 95, 0 \le i \le 3$$

where

$$Q(n) = Q(95-n), 48 \le n \le 95$$

and the values of *Q(n)* are the same values as those of the decoder.

C.2.1.2 Gain Detector

The gain detectors produce gain control data which satisfies the bitstream syntax. This information consists of the number of gain changes, the index of gain change positions and the index of gain change level. Note that the output gain control data applies to the previous input time signal. This means that the gain detector has a one frame delay.

The detection of the gain change point is done in the second half of the MDCT window region and in the non-overlapped region (of LONG_START_SEQUENCE and LONG_STOP_SEQUENCE). Thus the number of regions are one for ONLY_LONG_SEQUENCE, two for LONG_START_SEQUENCE and LONG_STOP_SEQUNCE, and eight for EIGHT_SHORT_SEQUENCE.

The samples in each region are divided into subregions, each having eight-tuple samples. Then one value (e.g. peak value of samples) is selected in these subregions. The ratios between the values of subregions and the value of the last subregion are calculated. If the ratio is greater or less than the value of 2 ^ n where n is an integer between -4 to 11, those subregions can be detected as the gain change points of the signals. The subregion number which is detected as the gain change point is set to be the position data. The exponent of the ratio is set to be the gain data. The time resolution of the gain control is approximately 0.7 ms at 48 kHz sampling rate.

C.2.1.3 Gain Modifier

The gain modifier for each PQF band controls the gain of each signal band. The complementary gain control process in the decoder decreases the pre-echo and reconstructs the original signal. A window function for gain control, the Gain Modification Function (GMF), which is defined in the decoding process, is derived from the gains and the gain-changed positions. The gain controlled signals are derived by applying the GMF to the corresponding band signals.

C.2.2 Diagrams

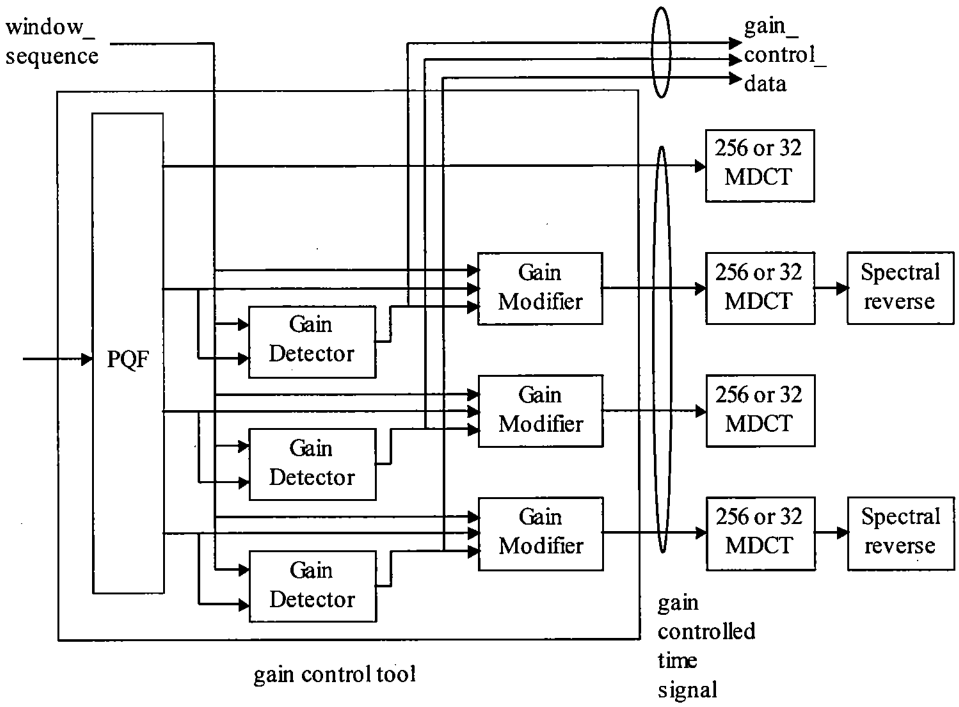


Figure C.2 — Block diagram of gain control tool for encoder

C.3 Filterbank and Block Switching

A fundamental component in the audio coding process is the conversion of the time domain signals into a time-frequency representation. This conversion is done by a forward modified discrete cosine transform (MDCT).

C.3.1 Encoding Process

In the encoder the filterbank takes the appropriate block of time samples, modulates them by an appropriate window function, and performs the MDCT. Each block of input samples is overlapped by 50% with the immediately preceding block and the following block. The transform input block length N can be set to either 2048 or 256 samples. Since the window function has a significant effect on the filterbank frequency response, the filterbank has been designed to allow a change in window shape to best adapt to input signal conditions. The shape of the window is varied simultaneously in the encoder and decoder to allow the filterbank to efficiently separate spectral components of the input for a wider variety of input signals.

C.3.1.1 Windowing and Block Switching

The adaptation of the time-frequency resolution of the filterbank to the characteristics of the input signal is done by shifting between transforms whose input lengths are either 2048 or 256 samples. The meaningfull transitions are described in subclause 15.3.1.

Window shape decisions are made by the encoder on a frame-by-frame-basis. The window selected is applicable to the second half of the window function only, since the first half is constrained to use the appropriate window shape from the preceding frame. Figure C.3 shows the sequence of blocks for the transition (D-E-F) to and from a frame employing the sine function window. The window shape selector generally produces window shape run-lengths greater than that shown in the figure.

The 2048 time-domain values $x'_{i,n}$ to be windowed are the last 1024 values of the previous window_sequence concatenated with 1024 values of the current block. The formula below shows this fact:

$$x'_{i,n} = \begin{cases} x_{(i-1),(n+1024)}, & \text{for } 0 \le \mathrm{n} < 1024 \\ x_{i,n}, & \text{for } 1024 \le \mathrm{n} < 2048 \end{cases}$$

Where i is the block index and n is the sample index within a block. Once the window shape is selected, the **window_shape** syntax element is initialized. Together with the chosen **window_sequence** all information needed for windowing exist.

With the window halves described in subclause 15.3.2, all **window_sequence**'s can be assembled.

C.3.1.2 MDCT

The spectral coefficient, $X_{i,k}$, are defined as follows:

$$X_{i,k} = 2 \cdot \sum_{n=0}^{N-1} z_{i,n} \cos\left(\frac{2\pi}{N}(n+n_0)\left(k+\frac{1}{2}\right)\right) \text{ for } 0 \le k < N/2 .$$

where :

z_{in} = windowed input sequence

n = sample index

k = spectral coefficient index

i = block index

N = window length of the one transform window based on the window_sequence value

n_0 = (N/2 + 1)/2

The analysis window length N of one transform window of the mdct is a function of the syntax element **window_sequence** and is defined as follows:

$$\mathrm{N} = \begin{cases} 2048, & \text{if ONLY\_LONG\_SEQUENCE (0x0)} \\ 2048, & \text{if LONG\_START\_SEQUENCE (0x1)} \\ 256, & \text{if EIGHT\_SHORT\_SEQUENCE (0x2) (8 times)} \\ 2048, & \text{if LONG\_STOP\_SEQUENCE (0x3)} \end{cases}$$

C.3.2 Diagrams

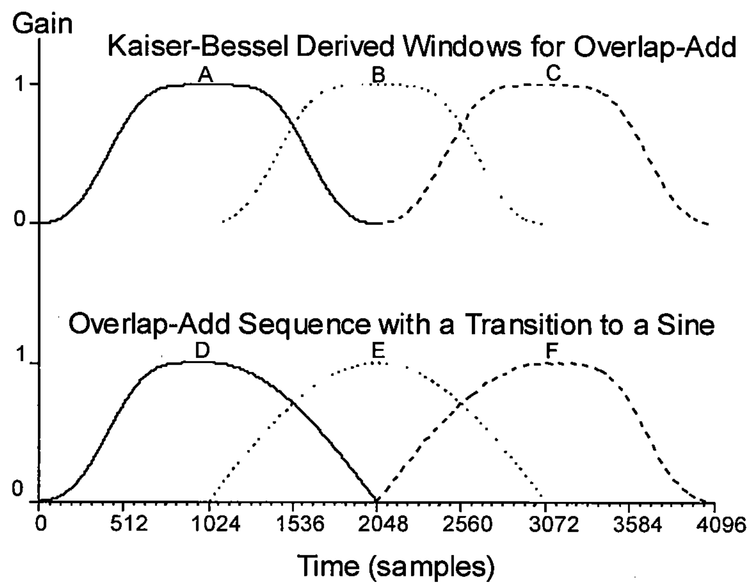


Figure C.3 — Example of the Window Shape Adaptation Process.

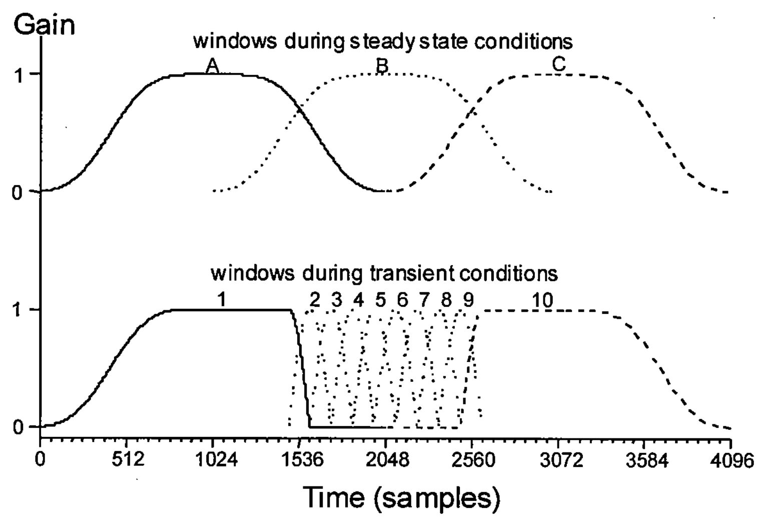


Figure C.4 — Example of Block Switching During Transient Signal Conditions

C.4 Prediction

C.4.1 Tool Description

Since each predictor itself is identical on both, the encoder and decoder side, all descriptions and definitions as specified for the decoder in clause 13 are also valid here.

Prediction is used for an improved redundancy reduction and is especially effective in case of more or less stationary parts of a signal which belong to the most demanding parts in terms of required bitrate. Prediction can be applied to every channel using an intra channel (or mono) predictor which exploits the auto-correlation between the spectral components of consecutive frames. Because a window_sequence of type EIGHT_SHORT_SEQUENCE indicates signal changes, i.e. non-stationary signal characteristic, prediction is only used if window_sequence is of type ONLY_LONG_SEQUENCE, LONG_START_SEQUENCE or LONG_STOP_SEQUENCE.

For each channel prediction is applied to the spectral components resulting from the spectral decomposition of the filterbank. For each spectral component up to limit specified by PRED_SFB_MAX, there is one corresponding predictor resulting in a bank of predictors, where each predictor exploits the auto-correlation between the spectral component values of consecutive frames.

The overall coding structure using a filterbank with high spectral resolution implies the use of backward adaptive predictors to achieve high coding efficiency. In this case, the predictor coefficients are calculated from preceding quantized spectral components in the encoder as well as in the decoder and no additional side information is needed for the transmission of predictor coefficients - as would be required for forward adaptive predictors. A second order backward-adaptive lattice structure predictor is used for each spectral component, so that each predictor is working on the spectral component values of the two preceding frames. The predictor parameters are adapted to the current signal statistics on a frame by frame base, using an LMS based adaptation algorithm. If prediction is activated, the quantizer is fed with a prediction error instead of the original spectral component, resulting in a coding gain.

C.4.2 Encoding Process

For each spectral component up to the limit specified by PRED_SFB_MAX of each channel there is one predictor. The following description is valid for one single predictor and has to be applied to each predictor. As said above, each predictor is identical on both, the encoder and decoder side. Therefore, the predictor structure is the same as shown in Figure 4 and the calculations of the estimate $x_{est}(n)$ of the current spectral component $x(n)$ as well as the calculation and adaptation of the predictor coefficients are exactly the same as those described for the decoder in subclause 8.3.2.

The only difference on the encoder side is that the prediction error has to be calculated according to

$$e(n) = x(n) - x_{est}(n)$$

to be fed to the quantizer. In this case the quantized prediction error is transmitted instead of the quantized spectral component.

C.4.2.1 Predictor Control

In order to guarantee that prediction is only used if this results in a coding gain, an appropriate predictor control is required and a small amount of predictor control information has to be transmitted to the decoder. For the predictor control, the predictors are grouped into scalefactor bands.

The following description is valid for either one single_channel_element() or one channel_pair_element() and has to be applied to each such element. Since prediction is only used if window_sequence is of type ONLY_LONG_SEQUENCE, LONG_START_SEQUENCE or LONG_STOP_SEQUENCE for the channel associated with the single_channel_element() or for both channels associated with the channel_pair_element(), the following applies only in these cases.

The predictor control information for each frame, which has to be transmitted as side information, is determined in two steps. First, it is determined for each scalefactor band whether or not prediction leads to a coding gain and if yes, the **prediction_used** bit for that scalefactor band is set to one. After this has been done for all scalefactor bands up to PRED_SFB_MAX, it is determined whether the overall coding gain by prediction in this frame compensates at least the additional bit need for the predictor side information. If yes, the **predictor_data_present** bit is set to 1, the complete side information including that needed for predictor reset (see below) has to be transmitted and the prediction error value is fed to the quantizer. Otherwise, the **predictor_data_present** bit is set to 0, the **prediction_used** bits are all reset to zero and are not transmitted. In this case, the spectral component value is fed to the quantizer. Figure C.5 shows a block diagram of the prediction unit for one scalefactor band. As described above, the predictor control first operates on all predictors of one scalefactor band and is then followed by a second step over all scalefactor bands.

In case of a single_channel_element() or a channel_pair_element() with **common_window** = 0 the control information is calculated and valid for the predictor bank(s) of the channel(s) associated with that element. In case of a channel_pair_element() with **common_window** = 1 the control information is calculated considering both channels associated with that element together. In this case the control information is valid for both predictor banks of the two channels in common.

C.4.2.2 Reconstruction of the Quantized Spectral Component

Since the reconstructed value of the quantized spectral component is needed as predictor input signal, it has to be calculated in the encoder, see also Figure 8 and Figure C.5. Depending on the value of the **prediction_used** bit, the reconstructed value is either the quantized spectral component or the quantized prediction error. Therefore, the following steps are necessary:

- If the bit is set (1), then the quantized prediction error, reconstructed from data to be transmitted, is added to the estimate $x_{est}(n)$, calculated by the predictor, resulting in the reconstructed value of the quantized spectral component, i.e. $x_{rec}(n) = x_{est}(n) + e_q(n)$
- If the bit is not set (0), then the quantized value of the spectral component is identical to the value reconstructed directly from the data to be transmitted.

C.4.3 Diagrams

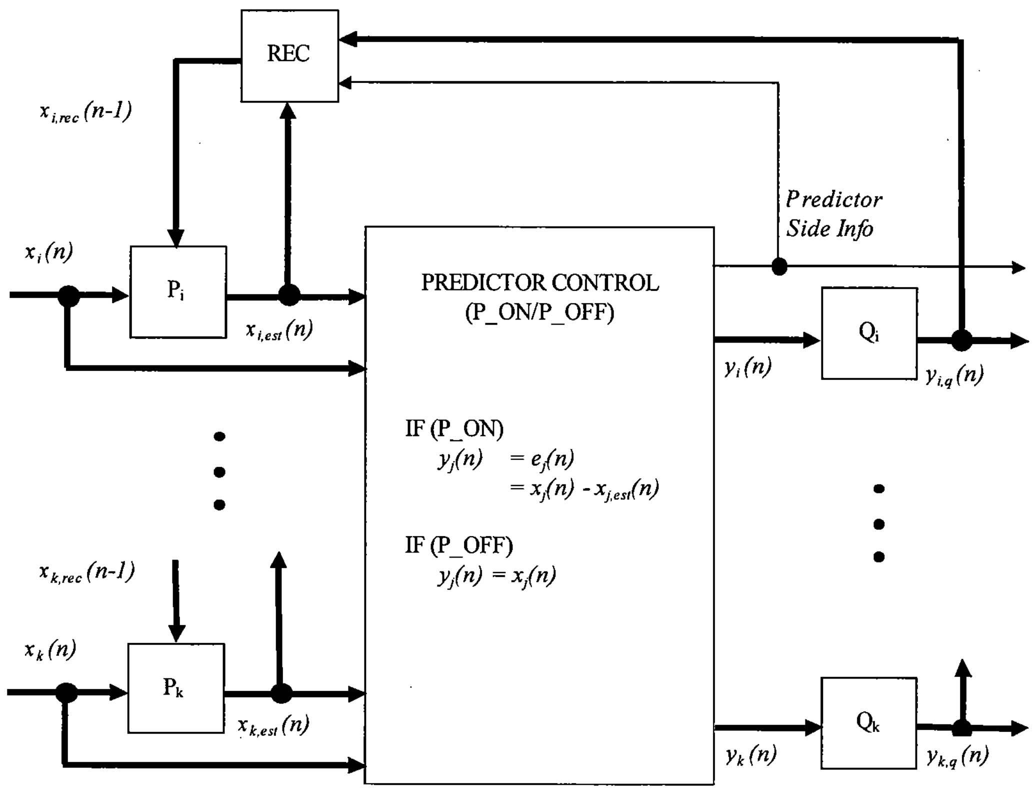


Figure C.5 — Block diagram of prediction unit for one scalefactor band. The complete processing is only shown for predictor P_i (Q - quantizer, REC - reconstruction of last quantized value). Note that the predictor control operates on all predictors P_i ... P_j ... P_k of a scalefactor band and is followed by a second control over all scalefactor bands.

C.5 Temporal Noise Shaping (TNS)

Temporal Noise Shaping is used to control the temporal shape of the quantization noise within each window of the transform. This is done by applying a filtering process to parts of the spectral data of each channel.

Encoding is done on a window basis. The following steps are carried out to apply the Temporal Noise Shaping tool to one window of spectral data:

- A target frequency range for the TNS tool is chosen. A suitable choice is to cover a frequency range from 1.5 kHz to the uppermost possible scalefactor band with one filter. Please note that this parameter (TNS_MAX_BANDS) depends on profile and sampling rate as indicated in the normative part.
- Next, a linear predictive coding (LPC) calculation is carried out on the spectral MDCT coefficients corresponding to the chosen target frequency range. For better stability, coefficients corresponding to frequencies below 2.5 kHz may be excluded from this process. Standard LPC procedures as known from speech processing can be used for the LPC calculation, e.g. the well-known Levinson-Durbin algorithm. The calculation is carried out for the maximum permitted order of the noise shaping filter (TNS_MAX_ORDER). Please note that this value depends on the profile as indicated in the normative part.

- As a result of the LPC calculation, the expected prediction gain g_p is known as well as the TNS_MAX_ORDER reflection coefficients r[] (so-called PARCOR coefficients).
- If the prediction gain g_p does not exceed a certain threshold t, no temporal noise shaping is used. In this case, the tns_data_present bit is set to zero and TNS processing is finished. A suitable threshold value is t = 1.4.
- If the prediction gain g_p exceeds the threshold t, temporal noise shaping is used.
- In a next step the reflection coefficients are quantized using coef_res bits. An appropriate choice for coef_res is 4 bits. The following pseudo code describes the conversion of the reflection coefficients r[] to index values index[] and back to quantized reflection coefficients rq[].

```
iqfac = ((1 << (coef_res-1)) - 0.5) / (π/2.0);
iqfac_m = ((1 << (coef_res-1)) + 0.5) / (π/2.0);

/* Reflection coefficient quantization */
for (i = 0; i < TNS_MAX_ORDER; i++) {
  index[i] = NINT(arcsin( r[i] ) * ((r[i] >= 0) ? iqfac : iqfac_m));
}
/* Inverse quantization */
for (i = 0; i < TNS_MAX_ORDER; i++) {
  rq[i] = sin( index[i] / ((index[i] >= 0) ? iqfac : iqfac_m) );
}
```

  where arcsin() denotes the inverse sin() function.

- The order of the used noise shaping filter is determined by subsequently removing all reflection coefficients with an absolute value smaller than a threshold p from the "tail" of the reflection coefficient array. The number of the remaining reflection coefficients is the order of the noise shaping filter. A suitable threshold for truncation is p = 0.1.
- The remaining reflection coefficients rq[] are converted into order+1 linear prediction coefficients a[] (known as "step-up procedure"). A description of this procedure is provided in the normative part as a part of the tool description (see "/* Conversion to LPC coefficients */").
- The computed LPC coefficients a[] are used as the encoder noise shaping filter coefficients. This FIR filter is slid across the specified target frequency range exactly the way it is described in the normative part for the decoding process (tool description). The difference between the decoding and encoding filtering is that the all-pole (auto-regressive) filter used for decoding is replaced by its inverse all-zero (moving-average) filter, i.e. replacing the decoder filter equation

```
y[n] = x[n] - a[1]*y[n-1] - ... - a[order]*y[n-order]
```

by the inverse (encoder) filter equation

```
y[n] = x[n] + a[1]*x[n-1] + ... + a[order]*x[n-order]
```

By default, an upward direction of the filtering is appropriate.

- Finally, the side information for Temporal Noise Shaping is transmitted:

Table C.25 — TNS side information

Data Element	Algorithmic Variable or Value
n_filt	1
coef_res	coef_res-3
coef_compress	0
length	Number of processed scalefactor bands
direction	0 (upwards)
order	Order of noise shaping filter
coef[]	index[]

Optionally, the use of the coef_compress field allows saving 1 bit per transmitted reflection coefficient if none of the reflection coefficients use more than half of their full range. Specifically, if the two most significant bits of each quantized reflection coefficient are either '00' or '11', coeff_compress may be set to a value of one and the size of the transmitted quantized reflection coefficients decreased by one.

C.6 Joint Coding

C.6.1 M/S Stereo

The decison to code left and right coefficients as either left + right (L/R) or mid/side (M/S) is made on a noiseless coding band by noiseless coding band basis for all spectral coefficients in the current block. For each noiseless coding band the following decison process is used:

1. For each noiseless coding band, not only L and R raw thresholds, but also M = (L+R)/2 and S = (L-R)/2 raw thresholds are calculated. For the raw M and S thresholds, rather than using the tonality for the M or S threshold, one uses the more tonal value from the L or R calculation in each threshold calculation band, and proceed with the psychoacoustic model for M and S from the M and S energies and the minimum of the L or R values for $C(\omega)$ in each threshold calculation band. The values that are provided to the imaging control process are identified in the psychoacoustic model information section as en(b) (the spread normalized energy) and nb(b), the raw threshold.
2. The raw thresholds for M, S, L and R, and the spread energy for M, S, L and R, are all brought into an "imaging control process". The resulting adjusted thresholds are inserted as the values for cb(b) into step 11 of the psychoacoustic model for further processing.
3. The final, protected and adapted to coder-band thresholds for all of M,S,L and R are directly applied to the appropriate spectrum by quantizing the actual L, R, M and S spectral values with the appropriate calculated and quantized threshold.
4. The number of bits actually required to code M/S, and the number of bits required to code L/R are calculated.
5. The method that uses the least bits is used in each given noiseless coding band, and the stereo mask is set accordingly.

With these definitions

Mthr,Sthr,Rthr, Lthr	raw thresholds. (the nb(b) from step 10 of the psychoacoustic model)
Mengy,Sengy,Rengy,Sengy	spread energy.(en(b) from step 6 of the psychoacoustic model)
Mfthr, Sfthr, Rfthr, Lfthr	final (output) thresholds. (returned as nb(b) in step 11 of the psychoacoustic model)
bmax(b)	BMLD protection ratio, as can be calculated from

$$bmax(b) = 10^{-3\cdot\left[0.5+0.5\cdot\cos\left(\pi\cdot\frac{\min(bval(b),\,15.5)}{15.5}\right)\right]}$$

the imaging control process for each noiseless coding band is as follows:

```
t = Mthr/Sthr
if (t > 1)
   t = 1/t
Rfthr = max(Rthr*t, min (Rthr, bmax*Rengy)
```


Lfthr = max(Lthr*t, min (Lthr, bmax) *Lengy)

t = min(Lthr, Rthr)

Mfthr = min(t, max(Mthr, min(Sengy*bmax,Sthr))

Sfthr = min(t, max(Sthr, min(Mengy*bmax,Mthr))

C.6.2 Intensity Stereo Coding

Intensity stereo coding is used to exploit irrelevance in the between both channels of a channel pair in the high frequency region. The following procedure describes one possible implementation while several different implementations are possible within the framework of the defined bitstream syntax.

Encoding is done on a window group basis. The following steps are carried out to apply the intensity stereo coding tool to one window group of spectral data:

- A suitable approach is to code a consecutive region of scalefactor bands in intensity stereo technique starting above a lower border frequency f_0. An average value of f_0 = 6 kHz is appropriate for most types of signals.
- For each scalefactor band, the energy of the left, right and the sum channel is calculated by summing the squared spectral coefficients, resulting in values E_l[sfb], E_r[sfb], E_s[sfb]. If the window group comprises several windows, the energies of the included windows are added.
- For each scalefactor band, the corresponding intensity position value is computed as

$$is\_position[sfb] = NINT\left(2 \cdot \log_2\left(\frac{E_l[sfb]}{E_r[sfb]}\right)\right)$$

- Next, the intensity signal spectral coefficients $spec_i$[i] are calculated for each scalefactor bands by adding spectral samples from the left and right channel ($spec_l$[i] and $spec_r$[i]) and rescaling the resulting values like

$$spec_i[i] = (spec_l[i] + spec_r[i]) \cdot \sqrt{\frac{E_l[sfb]}{E_s[sfb]}}$$

- The intensity signal spectral components are used to replace the corresponding left channel spectral coefficients. The corresponding spectral coefficients of the right channel are set to zero.

Then, the standard process for quantization and encoding is performed on the spectral data of both channels. However, the prediction status of the right channel predictors is forced to "off" for the scalefactor bands coded in intensity stereo. These predictors are updated by using an intensity decoded version of the quantized spectral coefficients. The procedure for this is described in the tool description for the intensity stereo decoding process in the normative part.

Finally, before transmission the Huffman codebook INTENSITY_HCB (15) is set in the sectioning information for all scalefactor bands that are coded in intensity stereo.

C.7 Quantization

C.7.1 Introduction

The description of the AAC quantization 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, attenuates the scalefactor band and calls the inner loop again.

AAC loops module input:

1. vector of the magnitudes of the spectral values mdct_line(0..1023).
2. xmin(sb) (see subclause C.1.4, step 0)
3. mean_bits (average number of bits available for encoding the bitstream).
4. more_bits, the number of bits in addition to the average number of bits, calculated by the psychoacoustic module out of the perceptual entropy (PE).
5. the number and width of the scalefactor bands (see Table 45 to Table 57)
6. for short block grouping the spectral values have to be interleaved so that spectral lines that belong to the same scalefactor band but to different block types which shall be quantized with the same scalefactors are put together in one (bigger) scalefactor band (for a full description of grouping see subclause 8.3.4)

AAC loops module output:

1. vector of quantized values x_quant(0..1023).
2. a scalefactor for each scalefactor band (sb)
3. common_scalefac (quantizer step size information for all scalefactor bands)
4. number of unused bits available for later use.

C.7.2 Preparatory Steps

C.7.2.1 Reset of all Iteration Variables

1. The start value of common_scalefac for the quantizer is calculated so that all quantized MDCT values can be encoded in the bitstream :

start_common_scalefac = ceiling(16/3(log$_2$((max_mdct_line ^ (3/4))/MAX_QUANT)))*

max_mdct_line is the largest absolute MDCT coefficient and ceiling() is the function which rounds to the nearest integer in the direction of positive infinity. MAX_QUANT is the maximum quantized value which can be encoded in the bitstream, defined as 8191. During the iteration process, the common_scalefac must not become less than start_common_scalefac.

2. Scalefactor[sb] is set to zero for all values of sb.

C.7.3 Bit Reservoir Control

Bits are saved to the reservoir when fewer than the mean_bits are used to code one frame.

*mean_bits = bit_rate * 1024 / sampling_rate.*

The number of bits which can be saved in the bit reservoir at maximum is called 'max_bit_reservoir' which is calculated using the procedure outlined in subclause 8.2.3. If the reservoir is full, unused bits have to be encoded in the bitstream as fillbits.

The maximum amount of bits available for a frame is the sum of mean_bits and bits saved in the bit reservoir.

The number of bits that should be used for encoding a frame depends on the more_bits value which is calculated by the psychoacoustic model and the maximum available bits. The simplest way to control bit reservoir is :

```
if more_bits > 0 :
   available_bits = mean_bits + min ( more_bits, bit_reservoir_state[frame])
if more_bits < 0 :
   available_bits = mean_bits + max ( more_bits, bit_reservoir_state[frame]
   - max_bit_reservoir)
```

C.7.4 Quantization of MDCT Coefficients

The formula for the quantization in the encoder is the inverse of the decoder dequantization formula (see also the decoder description) :

*x_quant = int ((abs(mdct_line) * (2^(- ¼ * (sf_decoder - SF_OFFSET))))^(3/4) + MAGIC_NUMBER)*

MAGIC_NUMBER is defined to 0.4054, SF_OFFSET is defined as 100 and mdct_line is one of spectral values, which is calculated from the MDCT. These values are also called 'coefficients'. The scalefactor 'sf_decoder' is the same as 'sf[g][sfb]' defined in clause 11.

For use in the iteration loops, the scalefactor 'sf_decoder' is split in two variables:

sf_decoder = common_scalefac - scalefactor + SF_OFFSET

It follows from this, that the formula used in the distortion control loop is:

x_quant = int((abs(mdct_line) * (2^(-¼ * (scalefactor - common_scalefac))))^(3/4) + MAGIC_NUMBER)

The signs of scalefactor is such that a *positive* change *increases* the magnitude of x_quant, and so *decreases* the distortion and *increases* the number of bits used.

The sign of the *mdct_line* is saved separately and added again only for counting the bits and encoding the bitstream.

C.7.4.1 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 coloring 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 **sb**:*

*do from lower index to upper index **i** of scalefactor band*

*mdct_scaled(i) = abs(mdct_line(i))^(3/4) * 2^(3/16 * scalefactor(sb))*

end do

end do

C.7.4.2 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 with the scalefactors applied to the values within the scalefactor bands (mdct_scaled(0..1023)), a start value for common_scalefac, 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 x_quant(i), and a new common_scalefac.

The formula to calculate the quantized MDCT coefficients is:

*x_quant(i) = int ((mdct_scaled (i) * 2^(-3/16 * common_scalefac)) + MAGIC_NUMBER)*

The bits, that would be needed to encode the quantized values and the side information (scalefactors etc.) are counted according to the bitstream syntax, described in clause 9.

C.7.4.3 Attenuation of Scalefactor Bands which Violate the Masking Threshold

The calculation of the distortion (*error_energy(sb)*) of the scalefactor band is done as follows:

```
   do for each scalefactor band sb:
      error_energy(sb)=0
      do from lower index to upper index i of scalefactor band
         error_energy(sb) = error_energy(sb) + (abs( mdct_line(i))
               - (x_quant(i) ^(4/3) * 2^(¼ * (scalefactor(sb) -common_scalefac
))))^2
      end do
   end do
```

All spectral values of the scalefactor bands which have a distortion that exceeds the allowed distortion (*xmin(sb)*) are attenuated according to formula in subclause C.7.4.1, the new scalefactors can be calculated according to this pseudocode:

```
   do for each scalefactor band sb
      if ( error_energy(sb) > xmin(sb) ) then
         scalefactor(sb) = scalefactor(sb) - 1
      end if
   end do
```

C.7.4.4 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

- All scalefactor bands with an energy exceeding xmin(sb) are already attenuated, or
- The difference between two consecutive scalefactors is greater than 60

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

The procedure described above is only valid in the case the number of available bits is equal to the number of required bits corresponding to the perceptual entropy. In the case the number of available bits is higher or lower than the number of required bits, it is the objective of the loops module to create a constant ratio between the quantisation noise and the masked threshold over all scale factor bands (constant Noise to Mask Ratio (NMR)). This can be realised by applying an offset to the target allowed distortion *xmin(sb)*, that is the same for all scale factor bands, prior to starting the loops module.

C.7.4.5 Inner Iteration Loop (Rate Control Loop)

The inner iteration loop calculates the actual quantization of the frequency domain data (*mdct_scaled*) with the following function, which uses the formula from subclause C.7.4.2:

```
quantize_spectrum(x_quant[] , mdct_scaled[] , common_scalefac):
   do for all MDCT coefficients i :
      x_quant(i) = int ((mdct_scaled (i) * 2^(-3/16 * common_scalefac))
                      + MAGIC_NUMBER)
   end do
```

and then calls a function *bit_count()*. This function counts the number of bits that would be neccessary to encode a bitstream frame according to clause 6.

The inner iteration loop can be implemented using successive approximation:

```
inner_loop():
   if (outer_loop_count == 0)
      common_scalefac = start_common_scalefac;
      quantizer_change = 32;
   else
      quantizer_change = 1;
   end if
   do
      quantize_spectrum();
      counted_bits = bit_count();
      if (counted_bits > available_bits) then
         common_scalefac = common_scalefac + quantizer_change;
      else
         common_scalefac = common_scalefac - quantizer_change;
      end if
      quantizer_change = int (quantizer_change / 2) ;
      if (quantizer_change == 0) && (counted_bits > available_bits)
         quantizer_change = 1;
      end if
   while (quantizer_change != 0)
```

Due to the choice of *start_common_scalefac* calculated from subclause C.7.2.1, after the first run through the inner loop the number of needed bits is usually greater than the available bits , and therefore *common_scalefac* will be increased by the *quantizer_change.*

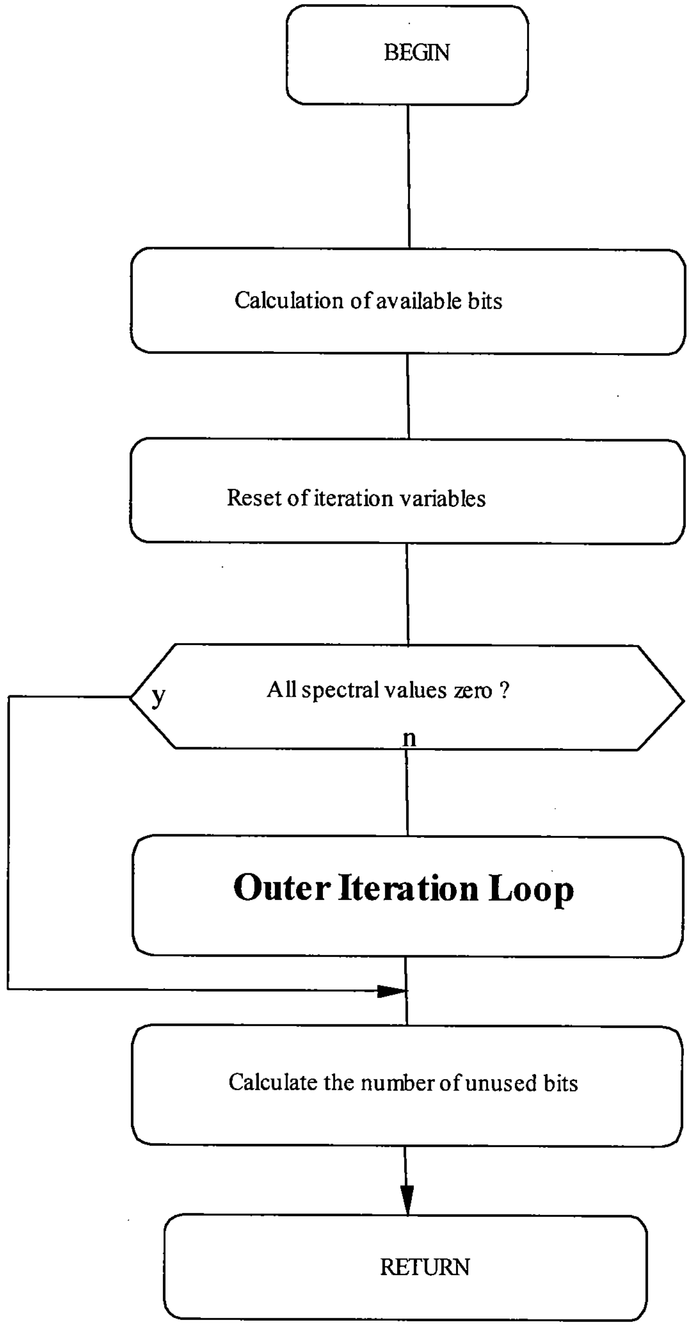


Figure C.6 — AAC iteration loop

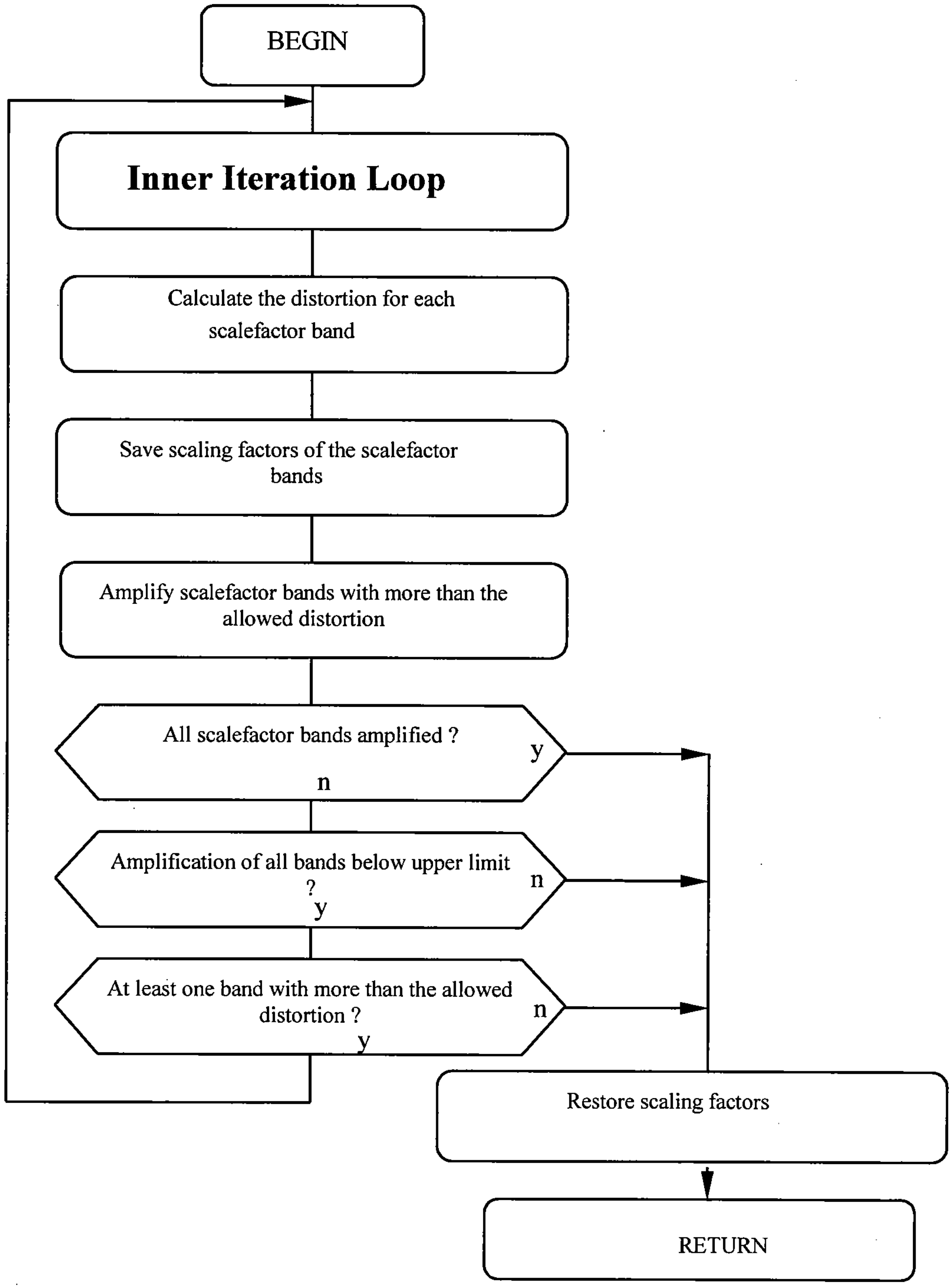


Figure C.7 — AAC outer iteration loop

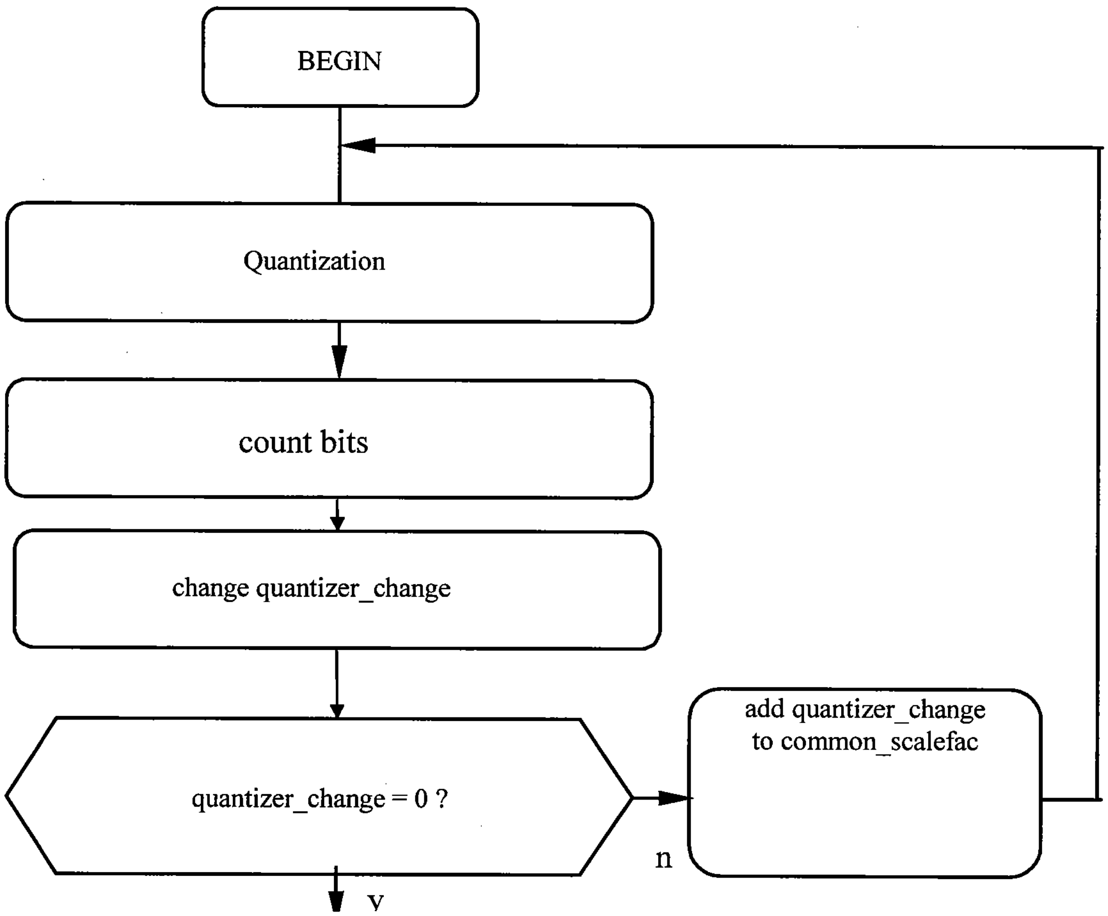


Figure C.8 — AAC inner iteration loop

C.8 Noiseless Coding

C.8.1 Introduction

In the AAC encoder the input to the noiseless coding module is the set of 1024 quantized spectral coefficients. Since the noiseless coding is done inside the quantizer inner loop, it is part of an iterative process that converges when the total bit count (of which the noiseless coding is the vast majority) is within some interval surrounding the allocated bit count. This section will describe the encoding process for a single call to the noiseless coding module.

Noiseless coding is done via the following steps:

- Spectrum clipping
- Preliminary Huffman coding using maximum number of sections
- Section merging to achieve lowest bit count

C.8.2 Spectrum Clipping

As a first step a method of noiseless dynamic range limiting may be applied to the spectrum. Up to four coefficients can be coded separately as magnitudes in excess of one, with a value of +-1 left in the quantized coefficient array to carry the sign. The index of the scalefactor band containing the lowest-frequency "clipped" coefficients is sent in the bitstream. Each of the "clipped" coefficients is coded as a magnitude (in excess of 1) and an offset from the base of the previously indicated scalefactor band. For this the long block scalefactor bands and coefficient ordering within those bands are used regardless of the window sequence. One strategy for applying spectrum clipping is to clip high-frequency coefficients whose absolute amplitudes are larger than one. Since the side information for carrying the clipped coefficients costs some bits, this noiseless compression is applied only if it results in a net savings of bits.

C.8.3 Sectioning

The noiseless coding segments the set of 1024 quantized spectral coefficients into *sections*, such that a single Huffman codebook is used to code each section (the method of Huffman coding is explained in a later section). For reasons of coding efficiency, section boundaries can only be at scalefactor band boundaries so that for each section of the spectrum one must transmit the length of the section, in scalefactor bands, and the Huffman codebook number used for the section.

Sectioning is dynamic and typically varies from block to block, such that the number of bits needed to represent the full set of quantized spectral coefficients is minimized. This is done using a greedy merge algorithm starting with the maximum possible number of sections each of which uses the Huffman codebook with the smallest possible index. Sections are merged if the resulting merged section results in a lower total bit count, with merges that yield the greatest bit count reduction done first. If the sections to be merged do not use the same Huffman codebook then the codebook with the higher index must be used.

Sections often contain only coefficients whose value is zero. For example, if the audio input is band limited to 20 kHz or lower, then the highest coefficients are zero. Such sections are coded with Huffman codebook zero, which is an escape mechanism that indicates that all coefficients are zero and it does not require that any Huffman codewords be sent for that section.

C.8.4 Grouping and Interleaving

If the window sequence is eight short windows then the set of 1024 coefficients is actually a matrix of 8 by 128 frequency coefficients representing the time-frequency evolution of the signal over the duration of the eight short windows. Although the sectioning mechanism is flexible enough to efficiently represent the 8 zero sections, *grouping* and *interleaving* provide for greater coding efficiency. As explained earlier, the coefficients associated with contiguous short windows can be grouped such that they share scalefactors amongst all scalefactor bands within the group. In addition, the coefficients within a group are interleaved by interchanging the order of scalefactor bands and windows. To be specific, assume that before interleaving the set of 1024 coefficients *c* are indexed as

c[g][w][b][k]

where

g is the index on groups

w is the index on windows within a group

b is the index on scalefactor bands within a window

k is the index on coefficients within a scalefactor band

and the right-most index varies most rapidly.

After interleaving the coefficients are indexed as

c[g][b][w][k]

This has the advantage of combining all zero sections due to band-limiting within each group.

C.8.5 Scalefactors

The coded spectrum uses one quantizer per scalefactor band. The step sizes of each of these quantizers is specified as a set of scalefactors and a global gain which normalizes these scalefactors. In order to increase compression, scalefactors associated with scalefactor bands that have only zero-valued coefficients are ignored in the coding process and therefore do not have to be transmitted. Both the global gain and scalefactors are quantized in 1.5 dB steps. The global gain is coded as an 8-bit unsigned integer and the scalefactors are differentially encoded relative to the previous scalefactor (or global gain for the first scalefactor) and then Huffman coded. The dynamic range of the global gain is sufficient to represent full-scale values from a 24-bit PCM audio source.

C.8.6 Huffman Coding

Huffman coding is used to represent n-tuples of quantized coefficients, with the Huffman code drawn from one of 11 codebooks. The spectral coefficients within n-tuples are ordered (low to high) and the n-tuple size is two or four coefficients. The maximum absolute value of the quantized coefficients that can be represented by each Huffman codebook and the number of coefficients in each n-tuple for each codebook is shown in Table C.26. There are two codebooks for each maximum absolute value, with each representing a distinct probability distribution function. The best fit is always chosen. In order to save on codebook storage (an important consideration in a mass-produced decoder), most codebooks represent unsigned values. For these codebooks the magnitude of the coefficients is Huffman coded and the sign bit of each non-zero coefficient is appended to the codeword.

Table C.26 — Huffman Codebooks

Codebook index	n-Tuple size	Maximum absolute value	Signed values
0		0	
1	4	1	yes
2	4	1	yes
3	4	2	no
4	4	2	no
5	2	4	yes
6	2	4	yes
7	2	7	no
8	2	7	no
9	2	12	no
10	2	12	no
11	2	16 (ESC)	no

Two codebooks require special note: codebook 0 and codebook 11. As mentioned previously, codebook 0 indicates that all coefficients within a section are zero. Codebook 11 can represent quantized coefficients that have an absolute value greater than or equal to 16. If the magnitude of one or both coefficients is greater than or equal to 16, a special *escape coding* mechanism is used to represent those values. The magnitude of the coefficients is limited to no greater than 16 and the corresponding 2-tuple is Huffman coded. The sign bits, as needed, are appended to the codeword. For each coefficient magnitude greater or equal to 16, an *escape sequence* is also appended, as follows:

escape sequence = <escape_prefix><escape_separator><escape_word>

where

<escape_prefix> is a sequence of N binary "1's"

<escape_separator> is a binary "0"

<escape_word> is an N+4 bit unsigned integer, msb first

and N is a count that is just large enough so that the magnitude of the quantized coefficient is equal to

2^(N+4) + <escape_word>

C.9 Features of AAC dynamic range control

In order to handle source material with variable peak levels, mean levels and dynamic range in a manner that minimizes the variability for the consumer, it is necessary to control the reproduced level such that, for instance, dialogue level or mean music level is set to a consumer controlled level at reproduction, regardless of how the programme was originated. Additionally, not all consumers will be able to audition the programmes in a good (i.e. low noise) environment, with no constraint on how loud they make the sound. The car environment, for instance, has a high ambient noise level and it can therefore be expected that the listener will want to reduce the range of levels that would otherwise be reproduced.

For both of these reasons, dynamic range control has to be available within the specification of AAC. To achieve this, it is necessary to accompany the bit-rate reduced audio with data used to set and control the dynamic range of the programme items. This control has to be specified relative to a reference level and in relationship to the important programme elements, e.g. the dialogue.

The features of the dynamic range control are as follows:

1. Dynamic Range Control is entirely optional. Therefore, with correct syntax, there is no change in complexity for those not wishing to invoke DRC.

2. The bit-rate reduced audio data is transmitted with the full dynamic range of the source material, with supporting data to assist in dynamic range control.

3. The dynamic range control data can be sent every frame to reduce to a minimum the latency in setting replay gains.

4. The dynamic range control data is sent using the 'fill_element' feature of AAC.

5. The *Reference Level* is defined as Full-scale.

6. The *Programme Reference Level* is transmitted to permit level parity between the replay levels of different sources and to provide a reference about which the dynamic range control may be applied. It is that feature of the source signal that is most relevant to the subjective impression of the loudness of a programme, such as the level of the dialogue content of a programme or the average level of a music programme.

7. The *Programme Reference Level* represents that level of programme that may be_reproduced at a set level relative to the *Reference Level* in the consumer hardware to achieve replay level parity. Relative to this, the quieter portions of the programme may be increased in level and the louder portions of the programme may be reduced in level.

8. *Programme Reference Level* is specified within the range 0 to -31.75 dB relative to *Reference Level*.

9. *Programme Reference Level* uses a 7 bit field with 0.25 dB steps.

10. The dynamic range control is specified within the range ±31.75 dB.

11. The dynamic range control uses an 8 bit field (1 sign, 7 magnitude) with 0.25 dB steps.

12. The dynamic range control can be applied to all of an audio channel's spectral coefficientsfrequency bands as a single entity or the coefficients can be split intowith different scalefactor bands, each being controlled separately by separate sets of dynamic range control data.

13. The dynamic range control can be applied to all channels (of a stereo or multichannel bitstream) as a single entity or can be split, with sets of channelsChannels being controlled separately by separate sets of dynamic range control data.

14. If an expected set of dynamic range control data is missing, the last received valid values should be used.

15. Not all elements of the dynamic range control data are sent every time. For instance, *Programme Reference Level* may only be sent on average once every 200 ms.

16. Where necessary, error detection/protection is provided by the Transport Layer.

17. The user shall be given the means to alter the amount of dynamic range control, present in the bitstream, that is applied to the level of the signal.

Annex D
(informative)

Patent Holders

D.1 List of Patent Holders

The International Organization for Standardization and the International Electrotechnical Commission (IEC) draw attention to the fact that it is claimed that compliance with this part of ISO/IEC 13818 may involve the use of patents.

ISO and IEC take no position concerning the evidence, validity and scope of these patent rights.

The holders of these patent rights have assured the ISO and IEC that they are willing to negotiate licences under reasonable and non-discriminatory terms and conditions with applicants throughout the world. In this respect, the statements of the holders of these patents right are registered with ISO and IEC. Information may be obtained from the companies listed in Table D.1.

Attention is drawn to the possibility that some of the elements of this part of ISO/IEC 13818 may be the subject of patent rights other than those identified in this annex. ISO and IEC shall not be held responsible for identifying any or all such patent rights.

Table D.1 — Companies who supplied patent statements

AT&T
BOSCH
Dolby Laboratories, Inc.
Fraunhofer Gesellschaft
GCL
Lucent Technologies
NEC Corporation
Philips Electronics N.V.
Sony Corporation
Thomson Multimedia

Annex E
(informative)

Registration Procedure

E.1 Procedure for the Request of a Registered Identifier (RID)

Requesters of a RID shall apply to the Registration Authority. Registration forms shall be available from the Registration Authority. Information which the requester shall provide is given in subclause E.3. Companies and organizations are eligible to apply.

E.2 Responsibilities of the Registration Authority

The primary responsibilities of the Registration Authority administrating the registration of copyright_identifiers is outlined in this clause ; certain other responsibilities may be found in the JTC 1 Directives. The Registration Authority shall :

a) implement a registration procedure for application for a unique RID in accordance with Annex H of the JTC 1 Directives ;

b) receive and process the applications for allocation of the work type code identifier from Copyright Registration Authority ;

c) ascertain which applications received are in accordance with this registration procedure, and to inform the requester within 30 days of receipt of the application of their assigned RID ;

d) inform application providers whose request is denied in writing within 30 days of receipt of the application, and also inform the requesting party of the appeals process ;

e) maintain an accurate register of the allocated RID. Revisions to the contact information and technical specifications shall be accepted and maintained by the Registration Authority ;

f) make the contents of this register available upon request to any interested party ;

g) maintain a data base of RID request forms, granted and denied. Parties seeking technical information on the format of private data which has a copyright_identifier shall have access to such information which is part of the data base maintained by the Registration Authority ;

h) report its activities to JTC 1, the ITTF, and the JTC 1/SC 29 Secretariat, or their respective assignees, annually on a schedule mutually agreed upon.

E.3 Contact Information of the Registration Authority

Organization Name:

Address:

Telephone:

Fax:

E.4 Responsibilities of Parties Requesting a RID

The party requesting a RID for the purpose of copyright identification shall :

a) apply using the Form and procedures supplied by the Registration Authority ;

b) provide contact information describing how a complete description of the copyright organization can be obtained on a non-discriminatory basis;

c) include technical details of the syntax and semantics of the data format used to describe the audio-visual works or other copyrighted works within the additional_copyright_info field. Once registered, the syntax used for the additional copyright information shall not change;

d) agree to institute the intended use of the granted copyright_identifier within a reasonable time frame;

e) to maintain a permanent record of the application form and the notification received from the Registration Authority of each granted copyright_identifier.

E.5 Appeal procedure for Denied Applications

The Registration Management Group is formed to have jurisdiction over appeals relating to a denied request for a RID. The RMG shall have a membership who are nominated by P and L members of the ISO technical body responsible for this part of ISO/IEC 13818. It shall have a convenor and secretariat nominated from its members. The Registration Authority is entitled to nominate one non-voting observing member.

The responsibilities of the RMG shall be :

a) To review and act on all appeals within a reasonable time frame ;

b) to inform, in writing, organisations which make an appeal for reconsideration of its petition of the RMGs disposition of the matter;

c) to review the annual report of the Registration Authority summary of activities;

d) to supply ISO member bodies with information concerning the scope of operation of the Registration Authority.

Annex F
(informative)

Registration Application Form

Contact information of organization requesting a Registered Identifier (RID)

Organization Name :

Address :

Telephone :

Fax :

E-mail :

Statement of an intention to apply the assigned RID

RID application domain : using guidelines to be provided by the Registration Authority

Date of intended implementation of the RID

Authorized representative

Name :

Title :

Address :

Signature ___________________________________

For official use only of the Registration Authority

Registration Rejected _____

Registration Granted ______ Registration Value _______

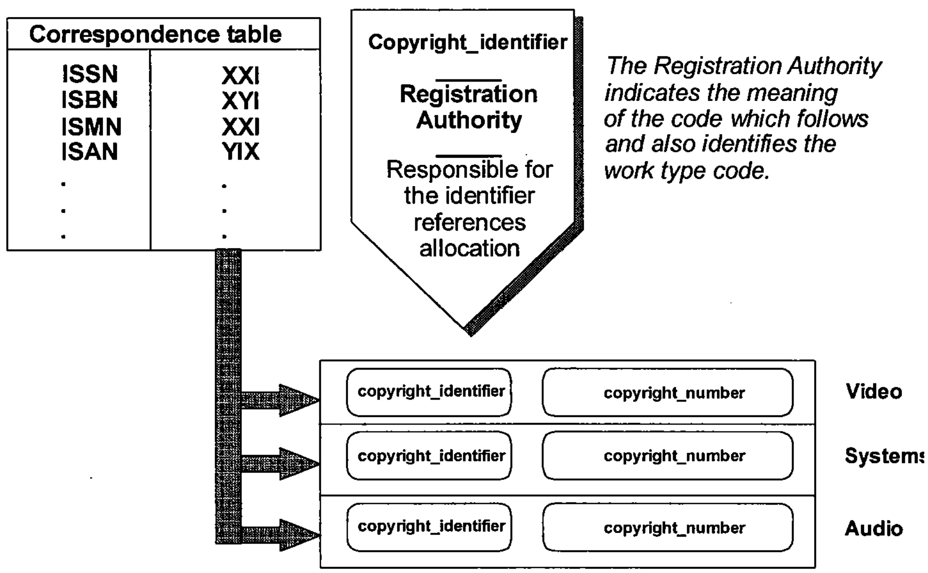
Attachment 1: Attachment of technical details of the registered data format

Attachment 2: Attachment of notification of appeal procedure for rejected applications

Annex G
(informative)

Registration Authority

Registration Authority
Diagramm of administration structure



Examples

copyright_identifier	copyright_number
I.S.B.N. (for books)	2-11- 0725 575 (ISBN Number)
I.S.A.N. (for audiovisual works)	1234567890123456 (ISAN Number)

All the copyright_identifiers are registered by the Registration Authority, uniquely for copyright_numbers standardized by ISO.
Each organization which allocates copyright_numbers requests a specific copyright_identifier from the Registration Authority. e.g. Staatsbibliothek Preussischer Kulturbesitz, designated by ISO to manage I.S.B.N., asks for a specific copyright_identifier from the R.A. for book numbering.

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[2] ITU-R Document TG10-2/3- E only, *Basic Audio Quality Requirements for Digital Audio Bit-Rate Reduction Systems for Broadcast Emission and Primary Distribution*, 28 October 1991.

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