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Four-channel Sub-band coding with combination of [8 8 4 2] for digital audio signal processing

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Abstract: This paper presents the research result for the four-channel sub-band coding (SBC) with a combination of share [8 8 4 2]. The research method is based on the Wavelet multi-resolution analysis using ATRAC3 algorithm using filters designed by a novel technique of filter bank called "transit-window". The advantages of the method are as: reduction of ripple of output amplitude response and information lossless at the interface region between two filters. Especially, under wavelet analysis with the change in the base-2 exponential partition factor from SBC [4 4 2] is to SBC [8 8 4 2] and it may be used to record/read audio and digitally broadcast on internet. The result is obtained by using the Matlab/Simulink modeling for the method.

Keywords: Sub-Band Coding, Digital Signal Processing, Digital Filter, ATRAC3, APT-X

1. Introduction

Sub-band coding (SBC) was invented in 1980 by researchers at Bell Laboratory. This technique offers the convenience to the speech signal compression since the spectral energy of speech signals is unequally distributed and mostly in the low frequency band [1, 9]. SBC has been widely used to encode audio, image and video signals [2, 3, 4, 5, 8, 14], etc. SBC has been effectively used for saving a signal bandwidth [10, 11, 12] because of rejection of information on frequencies called screened by mask, a phenomenon in the organ of auditory.

In the SBC technique, the larger number of channels will result the higher frequency resolution. This means that the bit allocation for sub-bands is more accurate and we can reject sub-bands with the energy lower or equal to masked threshold. Consequently, the larger compression factor can be obtained. However, the increase in the number of channels will require the larger number of filters and more complicated design. In a specific application, the balance between the number of channels, compression factor and quality must be optimally chosen. Nowadays, SBC has been used for digital audio processing with the number of 32 channels (MPEG/audio, PASC), of 3 channels (ATRAC1) and of 4 channels (ATRAC3, APT-X) [6, 11]. In the present paper, we study four-channel SBC propose a design method for digital filters using ATRAC3, APT-X algorithms and the technique of "transit-window", that can be used to record/read audio and digitally broadcast on the internet.

2. Four-Channel Sub-Band Coding with Combination of [8 8 4 2]

2.1. Characters of SBC (8842)

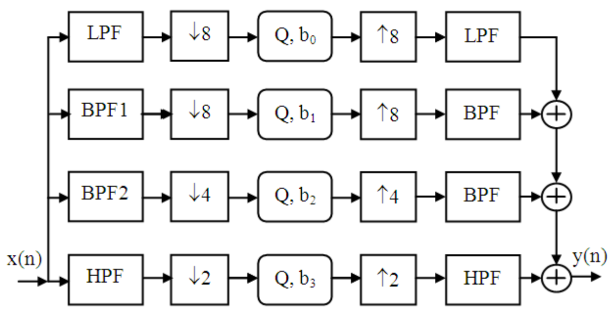


Figure 1. *Block diagram of SBC (8842)*

SBC with combination of [8 8 4 2] is relatively single-resolution sub-band coding with four channels. SBC (8842) is based on the Wavelet to separate bands of input signal. In which, there are two equal sub-bands with the partition factor of 8. The block diagram of four-channel sub-band coding with combination of [8 8 4 2] is illustrated in Figure 1. The signal of sub-bands is quantized and coded with different bits, b_0, b_1, b_2, b_3. The ATRAC algorithm employs quadrature mirror filter (QMF) [6, 12, 14] banks to separate the signal band into sub-bands. Suppose that input signal x(n) is samples from an analog signal with sampling frequency f_s which is equal to the Nyquist frequency f_{Ny}. The frequency spectrum of signal x(n) is presented in Figure 2.

The analyzing filter bank divides the input bandwidth into four sub-bands with the width of 1/8, 1/8, 1/4, and 1/2, respectively. The frequency spectrums of sub-bands are respectively $X_0(e^{j\omega})$, $X_1(e^{j\omega})$, $X_2(e^{j\omega})$, and $X_3(e^{j\omega})$, as shown in Figure 3. Signals of sub-bands are input to dividers with partition factors of 8, 8, 4, and 2, respectively. These signals are diminished in the time domain but expanded in the frequency domain on the same partition factors. Consequently, the frequency spectrum of four sub-band signals at the outputs of dividers is given in Figure 4. After the separation of frequency spectrum, the signal consists of a useful signal (the solid line) and reactive signal components (the dash line). The non-useful components must be rejected in the processing of data reconstruction.

Next, the sub-band signals are processed for transmission or recording. The processing consists of quantization and encoding. The encoding process will allocate the number of bits to sub-bands dependent on the energy levels in those sub-bands. The bit rate is considerably reduced in compared with that at the input. As a result, the required bandwidth of transmission channel or data volume will be decreased.

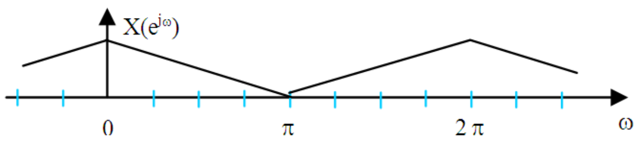


Figure 2. *Frequency spectrum of input signal x(n)*

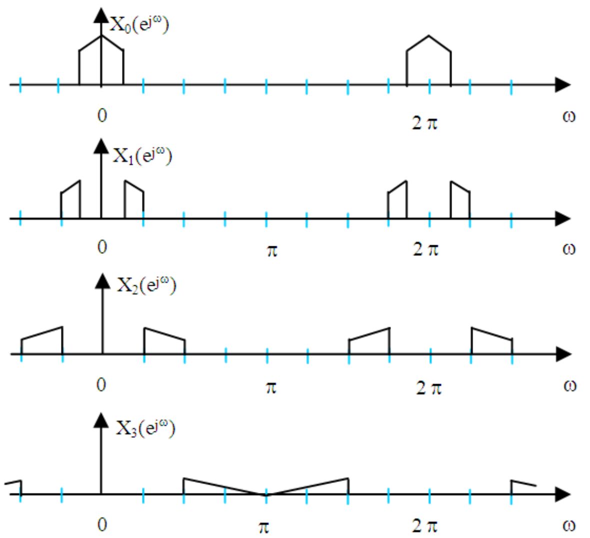


Figure 3. *Frequency spectrum of four sub-band signals at filter's output*

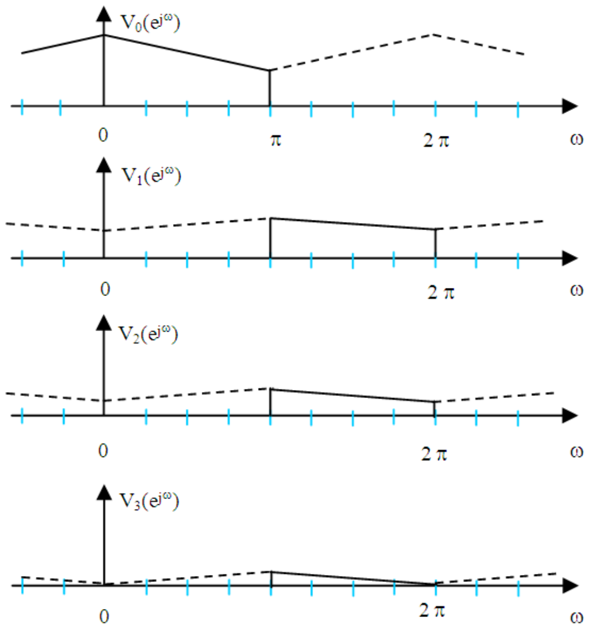


Figure 4. *Frequency spectrum of four sub-band signals at divider's output*

However, the quality of restored signal will be decreased. Compression ratio should be chosen suitably to meet a required audio quality for a particular application.

Restoration of signal is in the reverse process of analysis one. Firstly, the signal is input into interpolators with interpolation factors corresponding to divider's factors of 8, 8, 4, and 2. Then, the signals at the output of interpolators are stretched in the time domain and diminished in the frequency domain on the same interpolation factors. The interpolation factors of four filters are 8, 8, 4, and 2, respectively. The frequency spectrum of four sub-band signals at the input of interpolator is as presented in Figure 5.

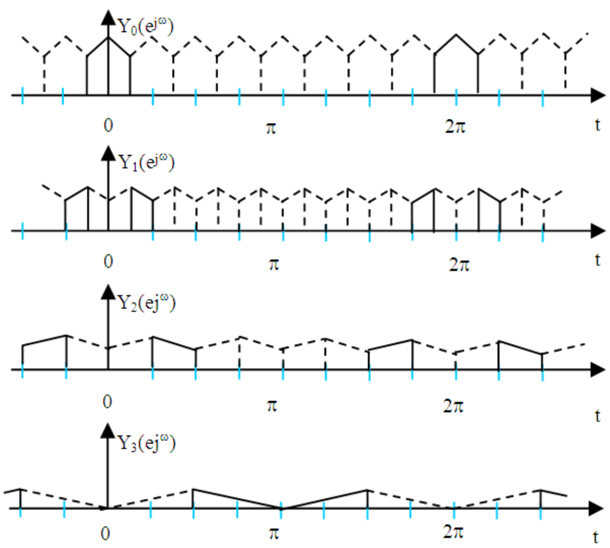


Figure 5. *Frequency spectrum of four sub-band signals at the output of interpolators*

There are imaginary components (the dash line) appearing beside active components (the solid line). The outputs of the first and second interpolators have seven imaginary components and there are three imaginary components and one imaginary component at the output of the third and fourth interpolators, respectively. In order to ensure that signals are completely restored, the imaginary components have to be removed in the signal restoration process. Further, to remove imaginary components and anonymous components, the sub-band signals are input to filters in synthesis filter bank and only useful signals are retrieved. The frequency spectrums of signals at the output of filters in the synthesis filter banks are the same as the frequency spectrums of signals at the output of filters in the analysis filter banks. Finally, the frequency spectrums ofsub-band signals are combined together to fully restore frequency band of the original signal.

2.2. Bit allocation in SBC

In general, there are many algorithms what can be used for bit allocation in SBC [7, 10, 12]. In the present study, bit allocation is described by

$$b_{aver_i} = b_{aver} + \frac{1}{2}\log_2 \frac{\sigma_i^2}{\prod_{i=0}^{M-1}(\sigma_i^2)^{\frac{1}{M}}} \quad (1)$$

Sub-band coding with partition factors of [8 8 4 2] is with four channels. The average number of bits coded for sub-band signal i is

$$b_{aver_i} = b_{aver} + \frac{1}{2}\log_2 \frac{\sigma_i^2}{\left(\sigma_0^2.\sigma_1^2.\sigma_2^2.\sigma_3^2\right)^{\frac{1}{4}}}; \ (i = 0,1,2,3) \quad (2)$$

where b_{aver} is the average number of bits of SBC (8 8 4 2) in the unit of bit/sample and is computed as

$$b_{aver} = \sum_{i=0}^{M-1} \frac{b_{aver_i}}{n_i} \quad (3)$$

where σ_i^2 is variance of sub-band signal i (i = 0,1,2,3).

SBC (8 8 4 2) that uses partition combination [8 8 4 2] is relative four-channel multi resolution sub-band coding, in which Wavelet analysis is used to divide the input frequency bandwidth. There are two sub-bands which are with the same partition factor of 8; it means that two sub-bands are with the same width.

2.3. Comparison of SBC (8 8 4 2) and SBC (4 4 2)

Suppose that the input signal x(n) is sampled at f_s = 44,100 Hz; coding 16 bit/sample PCM; the bit rate of two stereo channels is 1,411,200 bps (44,100 sample/sec×16 bit/sample×2 channels) [12, 13]. In the SBC (8 8 4 2), the sampling frequency of sub-bands signal at the output of signal divider is computed as

- Sample frequency of sub-band 0:

$$\frac{f_s}{8} = \frac{44{,}100\text{Hz}}{8} = 5512.5\text{Hz}$$

- Sample frequency of sub-band 1:

$$\frac{f_s}{8} = \frac{44{,}100\text{Hz}}{8} = 5512.5\text{Hz}$$

- Sample frequency of sub-band 2:

$$\frac{f_s}{4} = \frac{44{,}100\text{Hz}}{4} = 11025\text{Hz}$$

- Sample frequency of sub-band 3:

$$\frac{f_s}{2} = \frac{44{,}100\text{Hz}}{2} = 22050\text{Hz}$$

The average of bit rate in SBC (8 8 4 2) is R_{aver}(8 8 4 2) = 256kbps, R_{aver}(4 4 2) = 292kbps. The average number of bits is

$$b_{aver}(8\ 8\ 4\ 2) = \frac{R_{aver}(4\ 4\ 2)}{\frac{f_s}{8}+\frac{f_s}{8}+\frac{f_s}{4}+\frac{f_s}{2}} = \frac{256{,}000}{44{,}100} = 5.805\,\text{bit/sample}$$

$$b_{aver}(4\ 4\ 2) = \frac{R_{aver}(4\ 4\ 2)}{\frac{f_s}{4}+\frac{f_s}{4}+\frac{f_s}{2}} = \frac{292{,}000}{44{,}100} = 6.62\,\text{bit / sample}$$

The data compression factor of SBC (8 8 4 2) is approximately 1.14 times greater than that of SBC (4 4 2), respectively. And the compression is

$$\eta_{aver}(8\ 8\ 4\ 2) = \frac{1{,}411{,}200}{256{,}000} = 5.5125$$

and

$$\eta_{aver}(4\ 4\ 2) = \frac{1{,}411{,}200}{292{,}000} = 4.8329$$

It is clear that compression ratio in SBC (8 8 4 2) is 1.14 higher in compared with that of SBC (4 4 2).

3. Discussion and Result

In order to verify the quality of compression for digital audio in the four-channel SBC model with combination of [8 8 4 2], we make a program modeling for the filters in SBC (8 8 4 2) using "transit-window" method using Matlab/Simulink [6, 11]. The simulation result for four filters in the filter bank is as given in Tables 1 to 4.

Table 1. *The factors, LBK_n, of filter LPF in SBC (8842) with n =20 and α=82.5^0*

n	Lbk_n	n	Lbk_n	n	Lbk_n
0	0.1460	7	-0.0022	14	0.0007
1	0.1400	8	-0.0130	15	0.0022
2	0.1231	9	-0.0170	16	0.0024
3	0.0982	10	-0.0158	17	0.0020
4	0.0693	11	-0.0117	18	0.0013
5	0.0406	12	-0.0067	19	0.0007
6	0.0159	13	-0.0023	20	0.0002

Table 2. *The factors, Bbk_n, of filter BPF1 in SBC (8842) with n=20 and α=82.5^0*

n	Bbk_n	n	Bbk_n	n	Bbk_n
0	0.1669	7	-0.0352	14	0.0020
1	0.1363	8	-0.0000	15	0.0046
2	0.0594	9	0.0160	16	0.0048
3	-0.0277	10	0.0147	17	0.0032
4	-0.0880	11	0.0062	18	0.0011
5	-0.1025	12	-0.0001	19	-0.0004
6	-0.0774	13	-0.0007	20	-0.0011

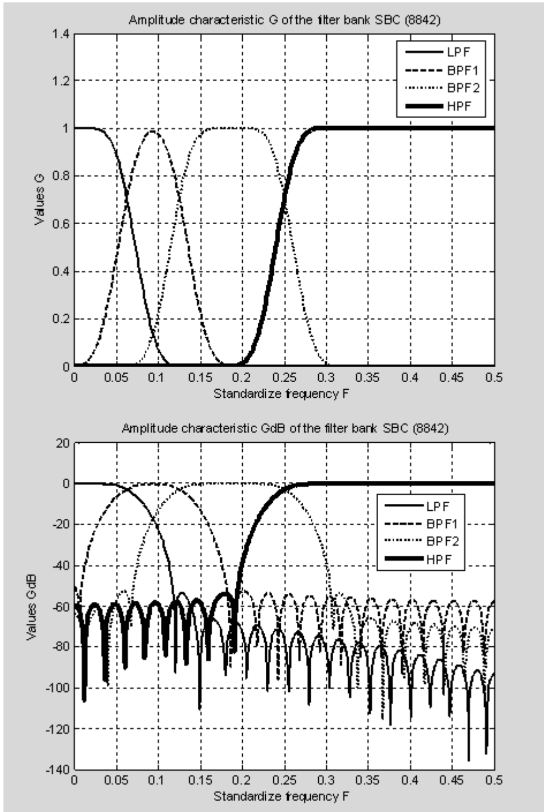
Table 3. *The factors, Bbk_n, of filter BPF2 in SBC (8842) with n=20 and α=82.5^0*

n	Bbk_n	n	Bbk_n	n	Bbk_n
0	0.2919	7	0.0017	14	-0.0010
1	0.1071	8	0.0260	15	0.0017
2	-0.1741	9	0.0130	16	0.0048
3	-0.1815	10	-0.0224	17	0.0015
4	-0.0000	11	-0.0216	18	-0.0018
5	0.0750	12	-0.0000	19	-0.0013
6	0.0225	13	0.0042	20	-0.0000

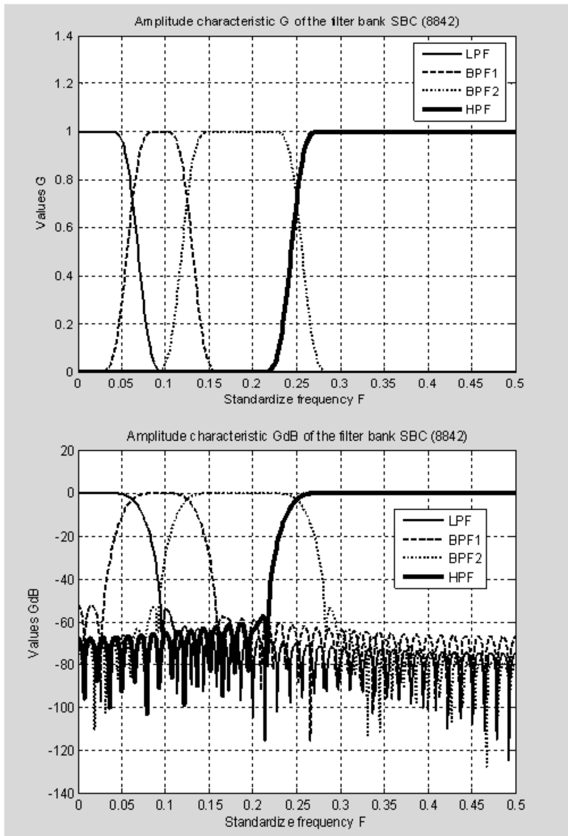
Table 4. *The factors, Hbk_n, of filter HPF in SBC (8842) with n=20 and α=82.5^0*

n	Hbk_n	n	Hbk_n	n	Hbk_n
0	0.5210	7	0.0294	14	-0.0042
1	-0.3156	8	0.0130	15	0.0021
2	-0.0204	9	-0.0169	16	0.0024
3	0.0982	10	-0.0098	17	-0.0009
4	0.0187	11	0.0093	18	-0.0013
5	-0.0512	12	0.0067	19	0.0003
6	-0.0161	13	-0.0047	20	0.0009

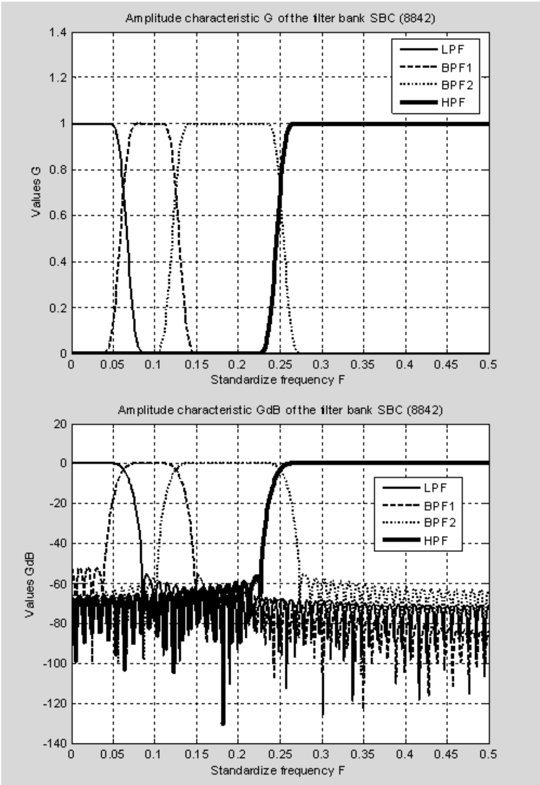
By observing the results given in Tables for 4 filters in the filter bank of SBC (8 8 4 2), the bandwidth ripple and width of transition band are correspondingly different with different orders of n and angles of α. Specifically for the case of n < 20 and α < 82.5^0, the ripple is unsatisfied in compared with the fact of $\delta_P \le 0.1$ [11]. We, therefore, choose the order of filter of n = 20 and α = 82.5^0 to calculate characteristics of G(F) and GdB(F). Note that, the width of transition band is more narrow as the order of filter increases. We determine the maximum order which is satisfied actual requirements. In the this study, the maximum order of filter is n = 100 for the "transit-window" method. Amplitude responses, G(F) and GbB(F), of four filters in the filter bank of SBC (8 8 4 2) are shown in Figure 6.



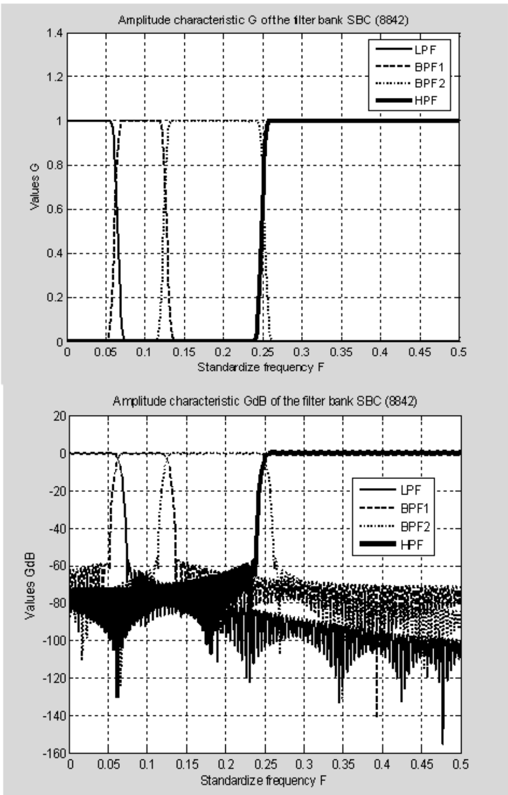
a. Amplitude characteristic G(F) and GdB(F) of four filters in SBC(8 8 4 2) with n=20, α=82.5^0



b. Amplitude characteristic G(F) and GdB(F) of four filters in SBC(8 8 4 2) with n=35, α=82.5^0



c. Amplitude characteristic G(F) and GdB(F) of four filters in SBC(8 8 4 2) with n=50, α=82.5⁰



d. Amplitude characteristic G(F) and GdB(F) of four filters in SBC(8 8 4 2) with n=100, α=82.5⁰

Figure 6. *Amplitude responses, G(F) and GdB(F), of four filters in the filter bank of SBC(8 8 4 2)*

In Figure 6, the bandwidth ripple and rejection band are very low, where the ripple in the bandwidth $\delta_P \le 0.02$ and rejection band $\delta_S \le -20$ dB are corresponding to n = 20, 35, and 50 (Figure 6.a, 6.b, 6.c). And, the ripple in bandwidth $\delta_P \le 0.01$ and rejection band $\delta_S \le -20$ dB are corresponding to n = 100 (Figure 6.d). Concurrently, the magnitude response is satisfied to the condition GdB $\ge$ -3dB = 0.707 at the cut frequency [11]. The width of transition band $B_{tr} = 0.13$ and 0.02 corresponding to n = 20 and 100, respectively. The result is met the practical requirements and could avoid the loss of information in the contact region of two sub-bands.

4. Conclusion

The computation and simulation results show the advantages in adjusting the value of dividing factor in SBC based on the Wavelet multi resolution analysis, with base 2 exponent, exactly SBC (8 8 4 2) offers higher compression ratio in compared with SBC (4 4 2). The method of designing filters in the filter bank using “transit-window” has been proposed by relying on characteristics of filter, that is different from traditional method. That is, firstly the width of transition band is defined, then the order of filter is computed so that the other parameters are met the requirements and avoid of information lost because of the contact between two filters. The simulation result shows the advantage, i.e. reducing the ripple, which allows to have higher quality of audio compression in compared with SBC (4 4 2) (used in minidisc). SBC (8 8 4 2) can be used in recording and broadcasting in the internet. This technology could also apply to sound digital broadcasting since digital audio broadcasting is now using MPEC-layer II with 32 sub-bands in a complex filter bank.

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