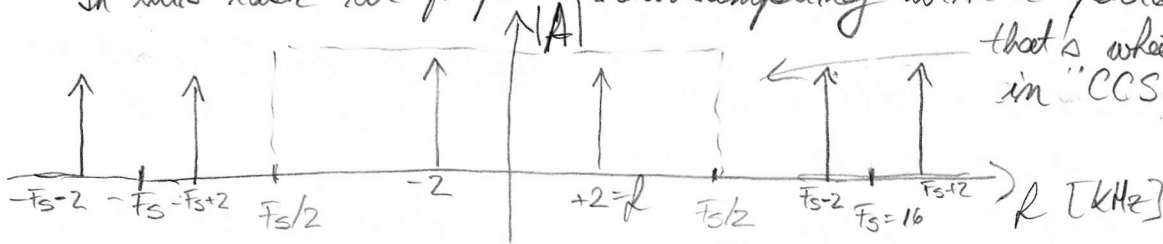


Lab 8: Two-channel filter bank

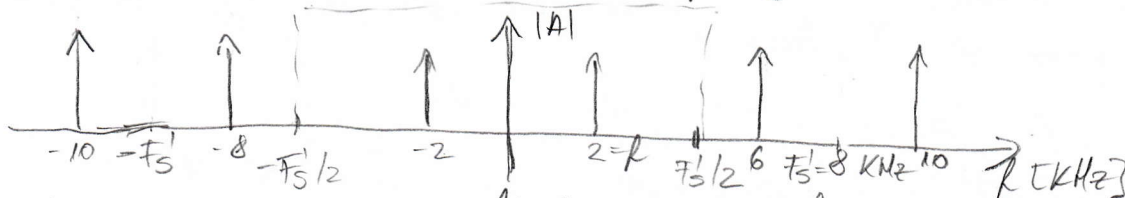
1) Upsampling and downsampling

a) $f = 2 \text{ kHz}$; $F_s = 16 \text{ kHz}$

In this task we perform downsampling with a factor of 2. that's what we see in CCS

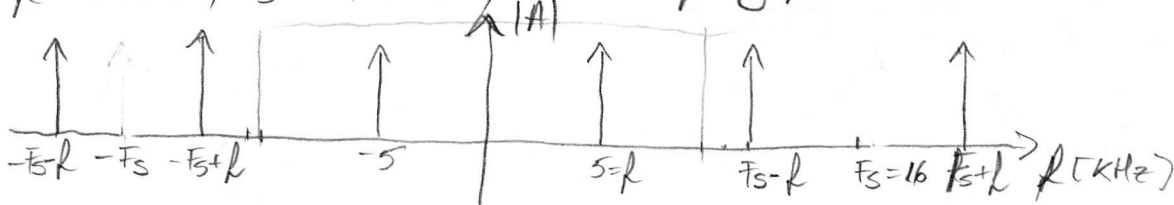


↓ downsampling with factor of 2 $\Rightarrow F_s' = \frac{F_s}{2} = 8 \text{ kHz}$

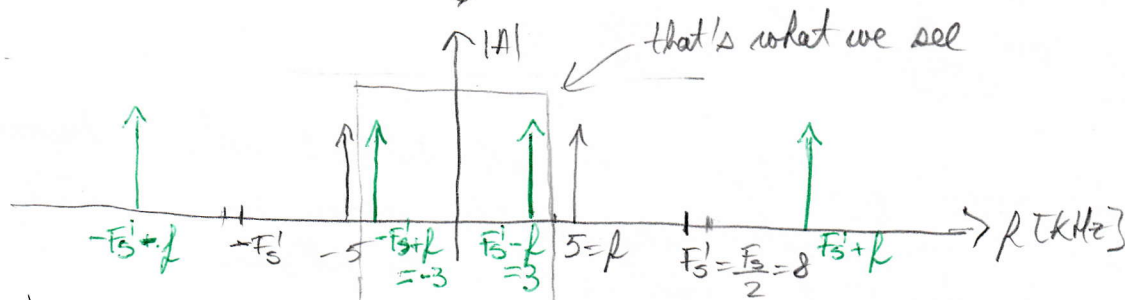


As we can see, no aliasing occurs, because $f < \frac{F_s'}{2}$, so the Nyquist criterion is obeyed also after downsampling.

b) $f = 5 \text{ kHz}$, $F_s = 16 \text{ kHz}$; downsampling factor: 2

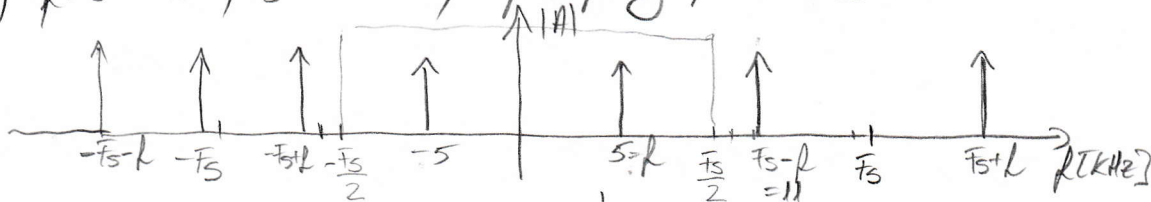


↓ $\downarrow 2$

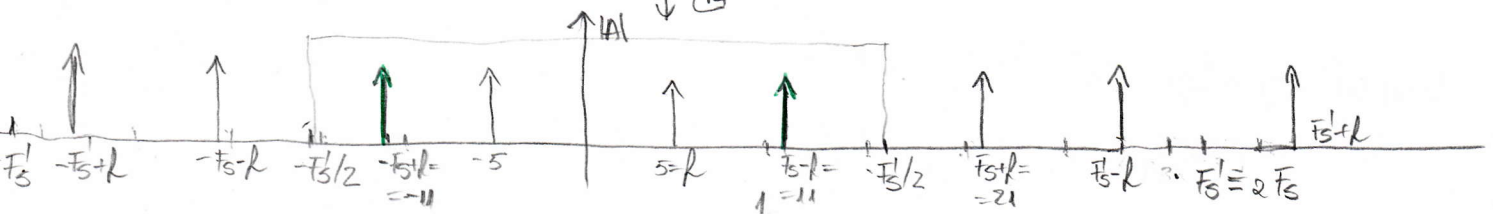


Here, aliasing occurs because $f > \frac{F_s'}{2}$. The imaged spectra are shown in the above image with green.

c) $f = 5 \text{ kHz}$; $F_s = 16 \text{ kHz}$; upsampling factor: 2



↓ $\uparrow 2$



cause by the fact that at the initial $F_s = 16 \text{ kHz}$ we had a component at $f = F_s - f = 11 \text{ kHz}$, which appears now in the observable spectrum due to the doubling of the sampling frequency.

d) In this exercise I designed a LPF with:

$$\omega_c = \frac{\pi}{L} = \frac{\pi}{2} ; \omega_c = 2\pi \frac{f_c}{F_s} = \frac{\pi}{2} \Rightarrow f_c = \frac{F_s}{4} = \frac{32000 \text{ Hz}}{4} = 8 \text{ kHz}$$

I chose $f_p = 5 \text{ kHz}$ because this is the highest frequency which we want to pass unattenuated and $f_{\text{stopband}} = 11 \text{ kHz}$ because we want to eliminate this frequency.

I chose the ripple to be the same in the stopband and passband, namely $\delta_p = \delta_s = 0.001$.

Then I proceeded with the design of the LPF with the Kaiser window

$$\omega_c = \frac{\omega_p + \omega_s}{2} \Rightarrow f_c = \frac{f_p + f_s}{2} = \frac{5 + 11}{2} \text{ kHz} = 8 \text{ kHz}$$

$$\Delta\omega = \omega_s - \omega_p = \frac{2\pi}{32 \text{ kHz}} (11 \text{ kHz} - 5 \text{ kHz}) = \frac{6\pi}{16} = \frac{3\pi}{8} = 0.375\pi$$

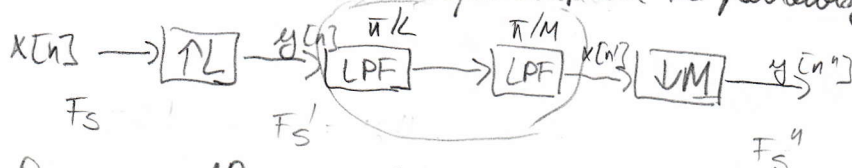
$$A = -20 \log_{10} \delta = \frac{16}{60}$$

$$\beta = 0.1102 (A - 8.7) = 5.65326$$

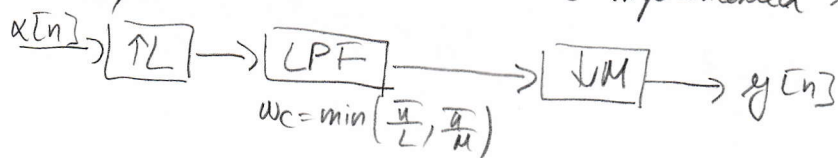
$$M = \frac{A - 8}{2.2854\beta} = 19.31 \Rightarrow M = 20$$

2) Sampling rate conversion

In this exercise I started from the following schematic:



Because the cascading of two LPF with different cutoff-frequencies is impractical, I implemented only one LPF whose $\omega_c = \min(\frac{\pi}{L}, \frac{\pi}{M})$. Thus, the schematic that I implemented is the following:



conversion ratio 1.5 $\Rightarrow L=3, M=2 ; \omega_c = \frac{\pi}{3} = \frac{2\pi f_c}{F_s} \Rightarrow f_c = \frac{F_s}{6} = \frac{8 \text{ kHz} \cdot 3}{6} = 4 \text{ kHz}$

conversion ratio 0.75 $\Rightarrow L=3, M=4 ; \omega_c = \frac{\pi}{4} = \frac{2\pi f_c}{F_s} \Rightarrow f_c = \frac{F_s}{8} = \frac{8 \text{ kHz} \cdot 3}{8} = 3 \text{ kHz}$

Initial sampling rate is $F_s = 8 \text{ kHz}$

3) Analysis filter bank and synthesis filter bank coefficients Prototype filter design 1

In this exercise we are given h_0 , which is the impulse response of the analysis FIR LPF and f_0 , which is the impulse response of the synthesis FIR LPF and we are required to find h_1 , the impulse response of the analysis HPF and f_1 , the impulse response of the synthesis HPF.

I did this by using the following condition:

$$\begin{cases} F_0(z) = H_1(-z) \\ F_1(z) = -H_0(-z) \end{cases}$$

$$\begin{cases} F_0(z) = H_1(-z) \\ F_1(z) = -H_0(-z) \end{cases}$$

Knowing $h_0 = [-1 \ 2 \ 6 \ 2 \ -1] / 8 \Rightarrow H_0(z) = (-1 + 2z^{-1} + 6z^{-2} + 2z^{-3} - z^{-4}) / 8$

$$-H_0(-z) = -(-1 - 2z + 6z^2 - 2z^3 - z^4) / 8 = (1 + 2z - 6z^2 + 2z^3 + z^4) / 8$$

$$\Rightarrow F_1(z) = -H_0(-z) \Rightarrow f_1 = [1 \ 2 \ -6 \ 2 \ 1] / 8$$

Knowing $f_0 = [1 \ 2 \ 1] / 2 \Rightarrow F_0(z) = (1 + 2z^{-1} + z^{-2}) / 2$

$$H_1(-z) = F_0(z) \Rightarrow H_1(z) = F_0(-z) = (1 - 2z^{-1} + z^{-2}) / 2$$

$$\Rightarrow h_1 = [1 \ -2 \ 1] / 2$$

Prototape filter design 2

In this exercise we are given only h_0 and we are required to find h_1 , f_0 and f_1 . To do so, we need to use both of the following conditions:

(I) $\begin{cases} F_0(z) = H_1(-z) \\ F_1(z) = -H_0(-z) \end{cases}$

(II) $P_0(z) - P_0(-z) = z^{-l}$, where $P_0(z) = H_0(z)F_0(z)$

$$H_0 = \frac{1}{4\sqrt{2}} [(1+\sqrt{3}) + (3+\sqrt{3})z^{-1} + (3-\sqrt{3})z^{-2} + (1-\sqrt{3})z^{-3}]$$

Assume $F_0 = a_0 + a_1 z^{-1} + a_2 z^{-2} + a_3 z^{-3}$

$$\Rightarrow [a_0 + a_1 z^{-1} + a_2 z^{-2} + a_3 z^{-3}] \cdot \frac{1}{4\sqrt{2}} [(1+\sqrt{3}) + (3+\sqrt{3})z^{-1} + (3-\sqrt{3})z^{-2} + (1-\sqrt{3})z^{-3}]$$

$$- [a_0 - a_1 z^{-1} + a_2 z^{-2} - a_3 z^{-3}] \cdot \frac{1}{4\sqrt{2}} [(1+\sqrt{3}) - (3+\sqrt{3})z^{-1} + (3-\sqrt{3})z^{-2} - (1-\sqrt{3})z^{-3}]$$

$= 2z^{-3}$, we choose this value for the delay l because all filters that we considered have the maximum delay of 3, so the cascading of such filters cannot yield a lower delay.

$$\Rightarrow \frac{1}{4\sqrt{2}} \left[\begin{aligned} & a_0(1+\sqrt{3}) + a_0(3+\sqrt{3})z^{-1} + a_0(3-\sqrt{3})z^{-2} + a_0(1-\sqrt{3})z^{-3} + \\ & a_1(1+\sqrt{3})z^{-1} + a_1(3+\sqrt{3})z^{-2} + a_1(3-\sqrt{3})z^{-3} + a_1(1-\sqrt{3})z^{-4} + \\ & a_2(1+\sqrt{3})z^{-2} + a_2(3+\sqrt{3})z^{-3} + a_2(3-\sqrt{3})z^{-4} + a_2(1-\sqrt{3})z^{-5} + \\ & a_3(1+\sqrt{3})z^{-3} + a_3(3+\sqrt{3})z^{-4} + a_3(3-\sqrt{3})z^{-5} + a_3(1-\sqrt{3})z^{-6} - \\ & a_0(1+\sqrt{3}) - a_0(3+\sqrt{3})z^{-1} - a_0(3-\sqrt{3})z^{-2} - a_0(1-\sqrt{3})z^{-3} + \\ & a_1(1+\sqrt{3})z^{-1} - a_1(3+\sqrt{3})z^{-2} + a_1(3-\sqrt{3})z^{-3} - a_1(1-\sqrt{3})z^{-4} - \\ & a_2(1+\sqrt{3})z^{-2} + a_2(3+\sqrt{3})z^{-3} - a_2(3-\sqrt{3})z^{-4} + a_2(1-\sqrt{3})z^{-5} + \\ & a_3(1+\sqrt{3})z^{-3} - a_3(3+\sqrt{3})z^{-4} + a_3(3-\sqrt{3})z^{-5} - a_3(1-\sqrt{3})z^{-6} \end{aligned} \right] = 2z^{-3}$$

$$\begin{cases} \frac{1}{4\sqrt{2}} \cdot 22^{-1} [a_0(3+\sqrt{3}) + a_1(1+\sqrt{3})] = 0 \\ \frac{1}{4\sqrt{2}} \cdot \cancel{22^{-3}} [a_0(1-\sqrt{3}) + a_1(3-\sqrt{3}) + a_2(3+\sqrt{3}) + a_3(1+\sqrt{3})] = \cancel{22^{-3}} 1 \\ \frac{1}{4\sqrt{2}} \cdot 22^{-5} [a_2(1-\sqrt{3}) + a_3(3-\sqrt{3})] = 0 \end{cases}$$

$$\Rightarrow 1) a_0(3+\sqrt{3}) + a_1(1+\sqrt{3}) = 0 \Rightarrow a_1 = -\frac{a_0\sqrt{3}(1+\sqrt{3})}{1+\sqrt{3}} = -a_0\sqrt{3}$$

$$2) a_0(1-\sqrt{3}) + a_1(3-\sqrt{3}) + a_2(3+\sqrt{3}) + a_3(1+\sqrt{3}) = 4\sqrt{2}$$

$$3) a_2(1-\sqrt{3}) + a_3(3-\sqrt{3}) = 0 \Rightarrow a_3 = a_2 \frac{\sqrt{3}-1}{\sqrt{3}(\sqrt{3}-1)} = \frac{a_2}{\sqrt{3}}$$

$$\text{Choose } a_0 = \frac{1-\sqrt{3}}{4\sqrt{2}} \Rightarrow a_1 = \frac{-\sqrt{3}+3}{4\sqrt{2}}$$

$$2) \frac{(1-\sqrt{3})^2}{4\sqrt{2}} + \frac{(3-\sqrt{3})^2}{4\sqrt{2}} + a_2(3+\sqrt{3}) + a_3(1+\sqrt{3}) = 4\sqrt{2}$$

$$\Rightarrow \frac{1-2\sqrt{3}+3}{4\sqrt{2}} + \frac{9-6\sqrt{3}+3}{4\sqrt{2}} + a_2(3+\sqrt{3}) + a_3(1+\sqrt{3}) = 4\sqrt{2}$$

$$\Rightarrow \frac{16-8\sqrt{3}}{4\sqrt{2}} + a_2(3+\sqrt{3}) + a_3(1+\sqrt{3}) = 4\sqrt{2} \Rightarrow$$

$$\Rightarrow a_2(3+\sqrt{3}) + a_3(1+\sqrt{3}) = \frac{32-16+8\sqrt{3}}{4\sqrt{2}} = \frac{16+8\sqrt{3}}{4\sqrt{2}}$$

$$\left. \begin{aligned} \sqrt{3}a_2(1+\sqrt{3}) + a_3(1+\sqrt{3}) &= \frac{16+8\sqrt{3}}{4\sqrt{2}} \\ a_3 &= \frac{a_2}{\sqrt{3}} \end{aligned} \right\} \Rightarrow$$

$$\Rightarrow \sqrt{3}a_2(1+\sqrt{3}) + a_2 \frac{\sqrt{3}+1}{\sqrt{3}} = \frac{16+8\sqrt{3}}{4\sqrt{2}} \Rightarrow \frac{3a_2(1+\sqrt{3}) + a_2(1+\sqrt{3})}{\sqrt{3}} = \frac{16+8\sqrt{3}}{4\sqrt{2}}$$

$$\Rightarrow \frac{4(1+\sqrt{3})a_2}{\sqrt{3}} = \frac{8(\sqrt{3}+2)}{4\sqrt{2}} \Rightarrow (1+\sqrt{3})a_2 = \frac{6+4\sqrt{3}}{4\sqrt{2}}$$

$$\Rightarrow a_2 = \frac{3+\sqrt{3}}{4\sqrt{2}} \Rightarrow a_3 = \frac{\sqrt{3}+1}{4\sqrt{2}}$$

$$h_0 = \frac{1}{4\sqrt{2}} [(1+\sqrt{3}), (3+\sqrt{3}), (3-\sqrt{3}), (1-\sqrt{3})]$$

$$h_1 = \frac{1}{4\sqrt{2}} [(1-\sqrt{3}), (\sqrt{3}-3), (3+\sqrt{3}), -(1+\sqrt{3})]$$

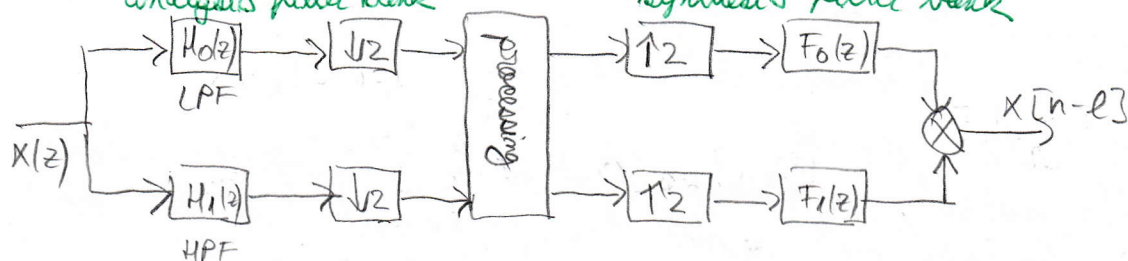
$$h_0 = \frac{1}{4\sqrt{2}} [(1-\sqrt{3}), (3-\sqrt{3}), (3+\sqrt{3}), (1+\sqrt{3})]$$

$$h_1 = \frac{1}{4\sqrt{2}} [-(1+\sqrt{3}), 3+\sqrt{3}, -(3-\sqrt{3}), 1-\sqrt{3}]$$

Having a look on the output yielded by ftool, we can see that the perfect reconstruction condition is verified (Th 4.1 / pag 105): the magnitude response is a straight line at around 6dB, and the impulse response is a Dirac delayed with 3 samples, as desired.

4) Non-efficient implementation of two-channel filter bank

In this exercise I implemented the following scheme:



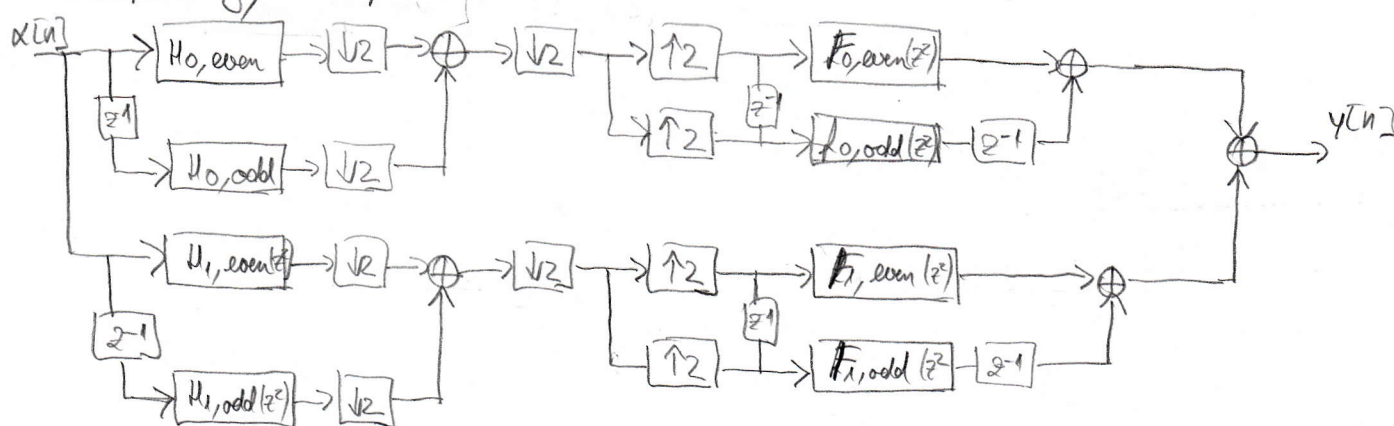
I chose for h_0, h_1, f_0, f_1 the values obtained a 3.1.

At the output we observe an impulse which has a delay of 3 with respect to the input.

5) Efficient-implementation of two-channel filter bank

The main problem in the previous implementation is that we downsample after the LPF/HPF and upsample before it. This is of course not desirable: it is better to first downsample and then filter the signal. Thus, moving the downsampling before the filter and the upsampling after the filter, we obtain the following schematics:

1° firstly, we split each filter in an odd and even part:



$$H(z) = h_0 + h_1 z^{-1} + h_2 z^{-2} + h_3 z^{-3} = (h_0 + h_2 z^{-2}) + (h_1 + h_3 z^{-2}) z^{-1}$$

$$H_{\text{even}}(z^2) = h_0 + h_2 z^{-2}$$

$$H_{\text{odd}}(z^2) = h_1 + h_3 z^{-2}$$

then,

$$h_{0, \text{even}} = [-1 \ 6 \ -1]/8 ; \quad h_{0, \text{odd}} = [2 \ 2]/8$$

$$h_{1, \text{even}} = [1 \ 1]/2 , \quad h_{1, \text{odd}} = [-2]/2 ;$$

$$f_{0, \text{even}} = [1 \ 1]/2 , \quad f_{0, \text{odd}} = [2]/2 ;$$

$$f_{1, \text{even}} = [1 \ -6 \ 1]/8 , \quad f_{1, \text{odd}} = [2 \ 2]/8$$

Looking at the output, we observe that it is an impulse which is delayed with 4 samples with respect to the input impulse we provide to the filter bank.