date: 10 September 2019

## Ex. 1 (Pt.14)

A signal, $x(t)=\cos \left(2 \pi f_{1} t\right)+\cos \left(2 \pi f_{2} t\right)$, with $f_{1}=5 \mathrm{kHz}$ and $f_{2}=12 \mathrm{kHz}$ is sampled at 60 kHz and then, its digital version $x[n]$, is downsampled of an order of 3 .

In order to perform the downsampling we just have a single digital IIR filter whose Discrete Difference Equation is the following: $y[n]=A(x[n]+\sqrt{2} x[n-1]+x[n-2])+\frac{1}{2} y[n-1]$
With $A=\frac{1}{2(2+\sqrt{2})}$.

- [5pt] Write the z-transform $H(z)$ of the filter and draw the pole-zero plot.
- [2pt] Represent the amplitude response of the filter in the $0-\pi$ normalized frequency range.
- [3pt] Represent the signal in the normalized frequencies domain before and after the downsampling, highlighting the aliased components.
- [4pt] Evaluate, in the final signal, the relative amplitude of the aliased components with respect to the main ones.


## Ex. 2 (Pt.8)

A signal $x[n]=\{-1,-1,2,0,0,1,-2,1,0,1,1,-1\}$ shall be filtered with this FIR filter: $h[n]=\{1,-1,2\}$.
We need to process blocks of 4 samples of the original signal for real time processing and we want to get the result depicting all the intermediate steps with these methods:

- [4pt] Overlap and Add
- [4pt] Overlap and Save


## Ex. 3 (Pt. 11 - MATLAB code)

1. [2pt.] Generate five cosine tones with the following parameters:

|  | Amplitude | Frequency $[\mathrm{Hz}]$ | Phase $[\mathrm{deg}]$ |
| :--- | :--- | :--- | :--- |
| $x 1$ | 1.0 | 220 | 0 |
| $x 2$ | 0.75 | 440 | 45 |
| $x 3$ | 0.5 | 660 | 90 |
| $x 4$ | 0.25 | 880 | 135 |
| $x 5$ | 0.125 | 1100 | 180 |

All five signals have a duration of 1 second and a sampling rate of 8000 Hz
2. [1pt.]Generate the signal $x 6$ as the sum of the five signal generated in point 1
3. [2pt] Compute y 6 as $\times 6$ filtered with a low-pass filter with normalized cut-off frequecy of 0.4 and 64 weights
4. [2pt.] Apply a Hanning window to select the first 512 samples of the signals $x 6$ and $y 6$.
5. [1pt.] Compute the DFT of the two windowed $x 6$ and $y 6$ using the MATLAB function fft
6. [1pt.] Plot the magnitude of the two DFTs computed at step 5 , with frequency in Hz in the x axis.
7. [2pt.] Comment on what you expect from this result.

## Solutions

Ex. 1
$H(z)=A \frac{1+\sqrt{2} z^{-1}+z^{-2}}{1-\frac{1}{2} z^{-1}}$



Normalized frequencies and pulsations of the signal before downsampling


Normalized pulsations after downsampling: in red and green the spectral replicas of the signal due to downsampling. The aliased component is also indicated.


The amplitude of the output for the first sinusoid will be:
$\left|H\left(z=e^{j \pi / 2}\right)\right|=\left|A \frac{1+\sqrt{2} z^{-1}+z^{-2}}{1-\frac{1}{2} z^{-1}}\right|=\left|A \frac{1-\sqrt{2} j-1}{1+j / 2}\right|=0.185$
While for the aliased replica at $2 \pi-\frac{6}{5} \pi=\frac{4}{5} \pi$ the amplitude is:
$\left|H\left(z=e^{j 4 \pi / 5}\right)\right|=\left|A \frac{1+\sqrt{2} e^{-j 4 \pi / 5}+e^{-j 8 \pi / 5}}{1-\frac{1}{2} e^{-j 4 \pi / 5}}\right|=0.02$
Ex. 2
Subdividing the signal in 3 blocks for the Overlap and Add we get:
$x_{1}[n]=\{-1,-1,2,0\}$
$x_{2}[n]=\{0,1,-2,1\}$
$x_{3}[n]=\{0,1,1,-1\}$
Filtering each block we get:
$y_{1}[n]=\left\{\begin{array}{lllllr}-1 & 0 & 1 & -4 & 4 & 0\end{array}\right\}$
$y_{2}[n]=\left\{\begin{array}{lllllr}0 & 1 & -3 & 5 & -5 & 2\end{array}\right\}$
$y_{3}[n]=\left\{\begin{array}{lllllr}0 & 1 & 0 & 0 & 3 & -2\end{array}\right\}$

Considering the overlapping tails:

$$
\left|\begin{array}{cccccccccccccc}
-1 & 0 & 1 & -4 & 4 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & 1 & -3 & 5 & -5 & 2 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 1 & 0 & 0 & 3 & -2
\end{array}\right|
$$

We get as output:
$y[n]=\left\{\begin{array}{llllllllllllll}-1 & 0 & 1 & -4 & 4 & 1 & -3 & 5 & -5 & 3 & 0 & 0 & 3 & -2\end{array}\right\}$
For the Overlap and Save we need to add two '0's at the beginning to account for the circular convolution effect, the number of zeros added at the beginning of the sequence is the length of the filter minus 1, i.e. 3-1=2

$$
\begin{aligned}
& x_{1}[n]=\left\{\begin{array}{llllll}
0 & 0 & -1 & -1 & 2 & 0
\end{array}\right\} \\
& x_{2}[n]=\left\{\begin{array}{lllllll}
2 & 0 & 0 & 1 & -2 & 1
\end{array}\right\} \\
& x_{3}[n]=\left\{\begin{array}{llllll}
-2 & 1 & 0 & 1 & 1 & -1
\end{array}\right\}
\end{aligned}
$$

Filtering using the circular convolution we get:

$$
\begin{aligned}
& y_{1}[n]=\{4 \\
& y_{2}[n]=\{-3 \\
& y_{3}[n]=\{1
\end{aligned}
$$

In order to remove the spurious components due to the circular convolution we remove the first two samples of every output sequence; the final output sequence will then be:

$$
y[n]=\left\{\begin{array}{lllllllllllll}
-1 & 0 & 1 & -4 & 4 & 1 & -3 & 5 & -5 & 3 & 0 & 0
\end{array}\right\}
$$

That is equal to the previous one apart from the last two samples (the tail of the Overlap and Add) that are not present due to the circular convolution approach.

## Ex. 3

\% Ex. 3 (Pt. 11 - MATLAB code)
\% 1) Generate five cosine tones with the following parameters:
\% Amplitude Frequency [Hz] Phase [deg]
\% x1 1.02200
$\% \times 20.7544045$
\% x3 0.566090
\% x4 0.25880135
\% x5 0.1251100180
\% All five signals have a duration of 1 second and a sampling rate of 8000 Hz

Fs=4000;
t=0:1/Fs:1;

```
X=zeros(5,length(t));
freqs=[220,440,660,880,1100];
phases=[0,45,90,135,180];
ampl=[1,.75,.5,.25,.125];
for i=1:length(freqs)
    X(i,:)=ampl(i)*cos(2*pi*freqs(i)*t+phases(i)/pi);
end
```

\%2) Generate the signal $x 6$ as the sum of the five signal
generated in point 1
x6= sum(X);
\% 3) Compute y6 as x6 filtered with a low-pass filter with
normalized cut-off frequecy of
\% 0.4 and 64 weights
Nf=64;
w_norm=0.4;
b=fir1(Nf-1,w_norm);
y6=filter(b,1,x6);
\% 4) Apply a Hanning window to select the first 512 samples of
the signals $x 6$ and y 6.
win=window(@hanning,512);
x6_win=x6(1:512).*(win.');
y6_win=y6(1:512). *(win.');
\%5) Compute the DFT of the two windowed $x 6$ and $y 6$ using the
MATLAB function fft
X6=fft(x6_win);
Y6=fft(y6_win);
\% 6) Plot the magnitude of the two DFTs computed at step 5,
... with frequency in Hz in the x -axis.
f=linspace(-Fs/2,Fs/2,512);
figure;
subplot(2,1,1);
plot(f, fftshift(abs(X6)));
xlabel('f [Hz]');
ylabel('|X_6(k)|');
subplot $(2,1,2)$;
plot(f, fftshift(abs(Y6)));
xlabel('f [Hz]');
ylabel('|Y_6(k)|');
\% 7) Comment on what you expect from this result.
\% I'm filtering at 0.4 w.r.t. to Fs=4000,
...so omega $=2000 \mathrm{~Hz}$, and cut-off frequency is
... 0.4*2000=800 Hz. For this reason, in Y6 I will
... only see up to x3 sinusoids

