Lezione 10 Filtering

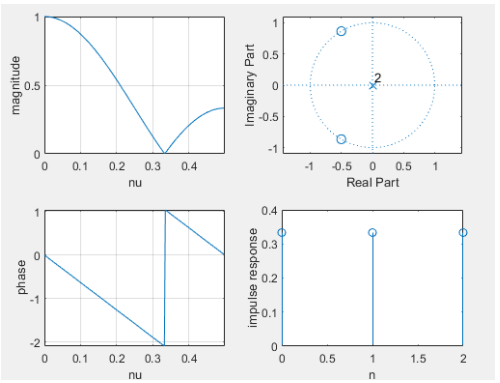
mercoledì 20 ottobre 2021 08.58

Prepare a function for filter characterization

```
function FilterChar(B,A)
[H W]=freqz(B,A,200); %get frequency response (abscissa in rad)
[h t]=impz(B,A);  %get impulse response
figure
subplot(2,2,1)
plot(W/(2*pi),abs(H)/max(abs(H))) %plot magnitude (abscissa in norm freq.)
ylabel('magnitude')
xlabel('nu')
axis([0 0.5 0 1])
grid on
subplot(2,2,3)
plot(W/(2*pi),phase(H)) %plot phase (abscissa in norm freq.)
ylabel('phase')
xlabel('nu')
axis([0 0.5 min(phase(H)) max(phase(H))])
grid on
subplot(2,2,2)
zplane(B,A)      %compute and plot z plane
subplot(2,2,4)
stem(t,h)      %plot impulse response
ylabel('impulse response')
xlabel('n')
```

Test : Moving average filter N=3

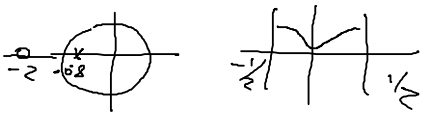
A=[1]
B=[1/3 1/3 1/3]
FilterChar(B,A)

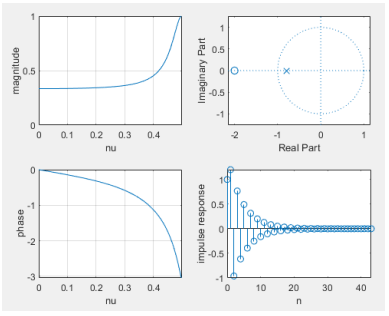


$$H(z) = \frac{1}{3}\left(1 + z^{-1} + z^{-2}\right) = \frac{1}{3}\,\frac{\left(z^2 + z + 1\right)}{z^2}$$

Test on recursive filter:
A=[1 0.8]
B=[1 2]

$$H(z) = \frac{1 + 2z^{-1}}{1 + 0.8z^{-1}} = \frac{z + 2}{z + 0.8}$$

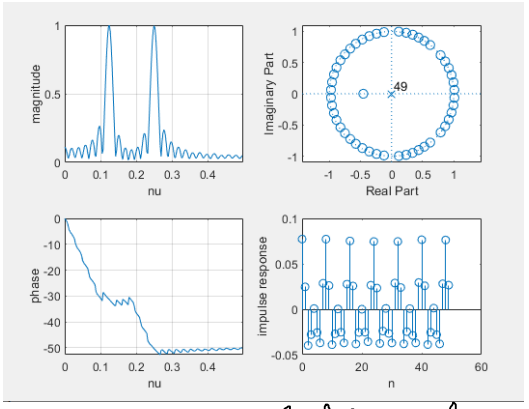




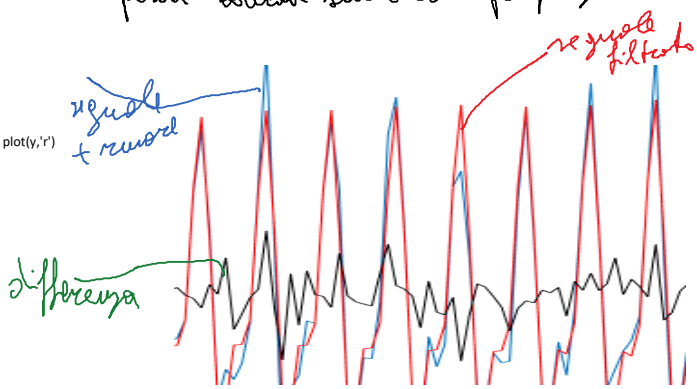
FILTRO DI WIENER (AI MINIMI QUADRATI) USANDO LA PSEUDO-INVERSA

```
clear all
close all
%generate a noisy signal
t=1:1/8000:2;
f1=1000;
f2=2000;
s=0.5*cos(2*pi*f1*t+0.3)+0.3*cos(2*pi*f2*t);
subplot(3,1,1)
plot(t, s)
sound(s,8000)
pause
noise=0.1*randn(1,length(t));
subplot(3,1,2)
plot(t, noise)
sound(noise, 5000)
pause
x=s+noise;
subplot(3,1,3)
plot(t, x)
sound(x,8000)
save s
```

```
close all
clear all
%wiener using the pseudo-inverse
%import the clean signal
load s
%load x
%s=x(4000:8000)';
sound(s,8000)
pause

N=50;
d=s(1, N:length(s)); %desired vector
%contaminate the signal with noise
spn=s+0.1*randn(1,length(s));
sound(spn, 8000)
pause

R=fliplr(spn(1,1:N));
C=spn(1,N:end);
A=toeplitz(C,R);
h=pinv(A)*d'

FilterChar(h',[1])
y=filter(h',[1],spn);
% figure
% plot(spn)
% hold on
% plot(y, 'r')
sound(spn)
pause
sound(y)

figure, plot(spn), hold on , plot(y,'r')
hold on, plot(spn-y,'k')
```



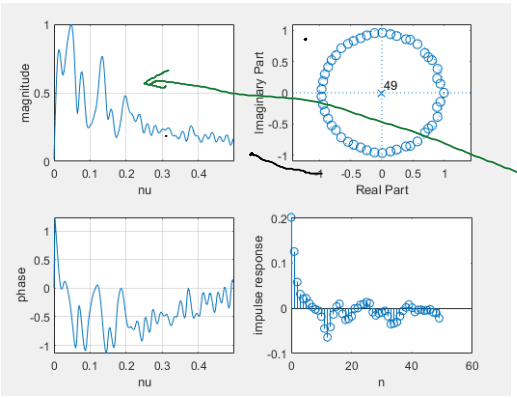
Qui si vede come il filtro seleziona le due sinusoidi diventando un passa-banda sulle due frequenze



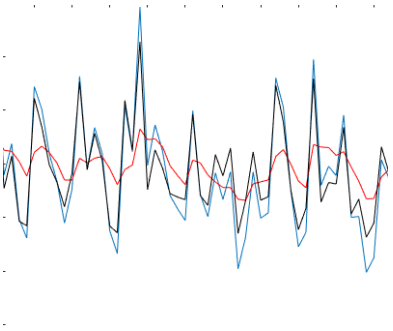
Secondo modo

Segnale vocale in x

```
close all
clear all
%wiener using the pseudo-inverse
%import the clean signal
%load s
load x
s=x(4000:8000)'; %use only a voiced segment
sound(s,8000)
pause

N=50;
d=s(1, N:length(s)); %desired vector
%contaminate the signal with noise
spn=s+0.1*randn(1,length(s));
sound(spn, 8000)
pause

R=fliplr(spn(1,1:N));
C=spn(1,N:end);
A=toeplitz(C,R);
h=pinv(A)*d'

FilterChar(h',[1])
y=filter(h',[1],spn);
% figure
% plot(spn)
% hold on
% plot(y, 'r')
sound(spn)
pause
sound(y)

figure, plot(spn), hold on , plot(y,'r')
hold on, plot(spn-y,'k')
```



il filtro si focalizza sulle "formanti"



Risultato non molto buono soprattutto all'ascolto poiché ci sono effetti percettivi non lineari.