Distortion of filtered signals MATLAB tutorial series (Part 3)

Pouyan Ebrahimbabaie

Laboratory for Signal and Image Exploitation (INTELSIG) Dept. of Electrical Engineering and Computer Science University of Liège Liège, Belgium

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A system has distortionless response if the input signal x[n] and the output signal y[n]have the same shape.

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It means:

$$y[n] = \frac{G}{G}x[n - n_d]$$

 G, n_d : constant

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It means:

$$Y(e^{j\omega}) = Ge^{-j\omega n_d}X(e^{j\omega}),$$

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$$H(e^{j\omega}) = \frac{Y(e^{j\omega})}{X(e^{j\omega})} = Ge^{-j\omega n_d}$$

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It means:

 $\left|H(e^{j\omega})\right|=\mathbf{G},$

 $\angle H(e^{j\omega}) = -n_d\omega.$

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It means:

 $\left|H(e^{j\omega})\right|=G,$

$$\angle H(e^{j\omega}) = -n_d\omega.$$

Notice: phase response passes from the origin !

$$x[n] = \cos(\omega_0 n) - \frac{1}{3}\cos(3\omega_0 n) + \frac{1}{5}\cos(5\omega_0 n),$$

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 $y_i[n] = c_1 \cos(\omega_0 n + \varphi_1) + c_2 \cos(3\omega_0 n + \varphi_2)$

 $+\boldsymbol{c_3}\cos(5\omega_0n+\boldsymbol{\varphi_3}).$

$$x[n] = \cos(\omega_0 n) - \frac{1}{3}\cos(3\omega_0 n) + \frac{1}{5}\cos(5\omega_0 n),$$

 $y_0[n] = 1\cos(\omega_0 n + 0) - 1/3\cos(3\omega_0 n + 0)$

 $+1/5\cos(5\omega_0 n + 0).$

Original signal no change !

$$x[n] = \cos(\omega_0 n) - \frac{1}{3}\cos(3\omega_0 n) + \frac{1}{5}\cos(5\omega_0 n),$$

 $y_1[n] = 1/4\cos(\omega_0 n + 0) - 1/3\cos(3\omega_0 n + 0)$

 $+1/5\cos(5\omega_0 n + 0).$

High pass filter Low frequency attenuated !

$$x[n] = \cos(\omega_0 n) - \frac{1}{3}\cos(3\omega_0 n) + \frac{1}{5}\cos(5\omega_0 n),$$

 $y_2[n] = \cos(\omega_0 n + \mathbf{0}) - \frac{1}{6}\cos(3\omega_0 n + \mathbf{0})$

 $+1/10\cos(5\omega_0 n + 0).$

Low pass filter High frequencies attenuated !

$$x[n] = \cos(\omega_0 n) - \frac{1}{3}\cos(3\omega_0 n) + \frac{1}{5}\cos(5\omega_0 n),$$

 $y_3[n] = \cos(\omega_0 n + \pi/6) - 1/3\cos(3\omega_0 n + \pi/6)$

 $+1/5\cos(5\omega_0 n + \pi/6).$

Constant phase

$$x[n] = \cos(\omega_0 n) - \frac{1}{3}\cos(3\omega_0 n) + \frac{1}{5}\cos(5\omega_0 n),$$

 $y_4[n] = \cos(\omega_0 n - \pi/4) - 1/3\cos(3\omega_0 n - 3\pi/4)$

$$+1/5\cos(5\omega_0n-5\pi/4).$$

Linear phase

$$x[n] = \cos(\omega_0 n) - \frac{1}{3}\cos(3\omega_0 n) + \frac{1}{5}\cos(5\omega_0 n),$$

 $y_5[n] = \cos(\omega_0 n - \pi/3) - 1/3\cos(3\omega_0 n + \pi/4)$

 $+1/5\cos(5\omega_0 n + \pi/7).$

Nonlinear phase





FIR has one main advantage and many disadvantages rather IIR ...

FIR has linear phase response !

FIR filters are the best choice to remove the noises from signal without distortion.

Original signal



Signal plus noise v.s. Original signal



Signal plus noise v.s. Original signal



Noise source is known : 12-18 Hz

Single sided Fourier transform



Noise source is known : 12-18 Hz

Single sided Fourier transform



Noise source is known : 12-18 Hz

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Filtered signal using IIR Butterworth filter



Filtered signal v.s. Original signal



Filtered signal v.s. Original signal



IIR filters have nonlinear phase response => Distortion

Persevering the shape of the signals not important in most of the applications

For example in audio applications, because human hearing system is not sensitive to distortion.

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Image: Second and the second and t	ass2: 19				

Filtered signal using FIR



Filtered signal v.s. Original signal



FIR filters have linear phase response !

Persevering the shape of the signals is important in bio-signals applications

%% Producing the oregingal signal

% Sampling period Fs = 500;% Sampling interval Ts=1/Fs;% Length of the signal N=2000; % Maximum time Tmax=(N-1)*Ts; % Time vector t=0:Ts:Tmax;

% Main frequencies & phase of the oreginal signal

- F1=10; F2=20; phi1=1.4; % Oreginal signal x=cos(2*pi*F1*t)+0.5*cos(2*pi*F2*t+phi1); % Plot range plot range =(N/2-100:N/2+100);
- % Plot signal in the range

figure(1)

plot(t(plot_range),x(plot_range),'LineWidth',2.5);
axis tight

%% Generate noise in a specific frequency band (12-18 Hz)

% Generate white Gaussian noise ns = randn(1, length(x))*3;% Design and load pass band filter: 12 to 18 Hz load PB 12 18; fvtool(PB 12 18) % Construct in-band noise ns filtered=filter(PB 12 18,ns); % Signal + Noise x ns=x+ns filtered;

% Plot oreginal signal and signal plus noise figure(3) plot(t(plot_range),x(plot_range),'LineWidth',2.5); hold on plot(t(plot_range),x_ns(plot_range),'LineWidth',2.5); axis tight

%% single-sided frequency spectrum of the signal plus noise

% Compute fft X=fft(x ns); % Take abs and scale it X2=abs(X/N);% Pick the first half X1=X2(1:N/2+1); % Multiply by 2 (except the DC part), to compenseate % the removed side from the spectrum. X1(2:end-1) = 2*X1(2:end-1);

% Frequency range F = Fs*(0:(N/2))/N; % Plot single-sided spectrum figure(4) plot(F,X1,'LineWidth',2.5) title('Single-Sided Amplitude Spectrum') xlabel('f (Hz)');

%% Remove noise usin band-stop IIR filter

% Design and load IIR band stop filter: 12 to 18 Hz load SB_12_18 fvtool(SB_12_18) % Filter the noise out x_clean_IIR=filter(SB_12_18,x_ns);

% Single sided spectrum of cleaned signal % Compute fft X=fft(x clean IIR); % Take abs and scale it X2=abs(X/N);% Pick the first half X1=X2(1:N/2+1);% Multiply by 2 (except the DC part), to compenseate % the removed side from the spectrum. X1(2:end-1) = 2*X1(2:end-1);

```
% Plot single-sided spectrum
figure(6)
plot(F,X1,'LineWidth',2.5)
title('Single-Sided Amplitude Spectrum')
xlabel('f (Hz)');
figure(7)
plot(t(plot_range),x(plot_range),'LineWidth',2.5);
hold on
plot(t(plot range), x clean IIR(plot range), 'LineWidth',
2.5);
axis tight
```

%% Remove noise usin band-stop FIR filter % Design and load FIR band stop filter: 12 to 18 Hz load SB 12 18 FIR fvtool(SB 12 18 FIR) % Filter the noise out x clean FIR=filter(SB 12 18 FIR, x ns); % Single sided spectrum of cleaned signal % Compute fft X=fft(x clean FIR); % Take abs and scale it X2=abs(X/N);% Pick the first half X1=X2(1:N/2+1);

% Multiply by 2 (except the DC part), to compenseate % the removed side from the spectrum. X1(2:end-1) = 2*X1(2:end-1);% Frequency range $F = Fs^{*}(0:(N/2))/N;$ % Plot single-sided spectrum figure(9) plot(F,X1,'LineWidth',2.5) title('Single-Sided Amplitude Spectrum') xlabel('f (Hz)');

figure(10)
plot(t(plot_range),x(plot_range),'LineWidth',2.5);
hold on
plot(t(plot_range),x_clean_FIR(plot_range),'LineWidth'
,2.5);
axis tight

Useful links

 <u>http://www.montefiore.ulg.ac.be/~ebrahimb</u> <u>abaie/applieddigtial.htm</u>