% Tune-Up Tuesday #7 for October 23, 2018

% Play $x[n]$, which is an 800 Hz tone for 3s at a sampling rate of 8000 Hz:

```
fs = 8000;
Ts = 1/fs;
t = 0 : Ts : 3;
f0 =  800;
x = cos(2*pi*f0*t);
sound(x, fs);       % please use the sound command for tune-up
pause(3);
```

```
% Discrete-time frequency for the cosine is
% w0 = 2*pi*f0/fs = 2*pi*800/8000 = 0.2*pi
```

% (a) Define an impulse response $h[n]$ of an averaging filter of 10 coefficients.

```
h10 = (1/10)*ones(1, 10);
```

% (b) Plot its magnitude/phase response

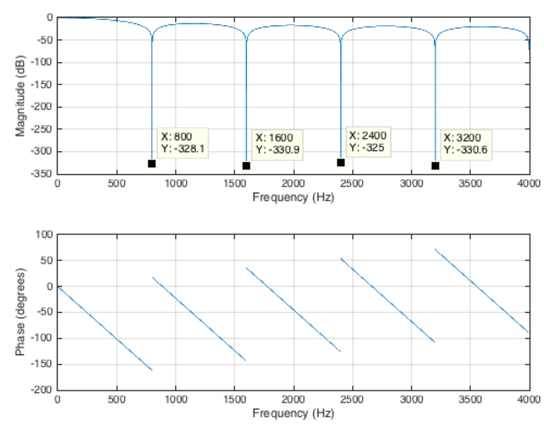
```
freqz(h10);  % not shown
```

% (c) At what frequencies (in Hz) does the magnitude response equal zero?
% *Hint: You can use the data cursor tool in the freqz plot window.*

```
% For 1 Hz accuracy in freqz and
% horizontal axis in Hz, use

freqz(h10, 1, fs, fs);  % plot on right

%  800,  1600,  2400,  3200, 4000 Hz
% -800, -1600, -2400, -3200 Hz
```



% Are the frequencies harmonically related?

```
% Yes, over frequencies captured via sampling at sampling rate fs,
% i.e. -fs/2 to fs/2, integer multiples are 800 Hz have been
% zeroed out except 0 Hz.
```

% Can you give a formula for the frequencies in terms for an N-point averaging filter?

```
%  fs/N,  2*fs/N,  3*fs/N, etc.
% -fs/N, -2*fs/N, -3*fs/N, etc.
```

% (d) Filter $x[n]$ using the averaging filter $h[n]$ and play the result:

```
y10 = filter(h10, 1, x);
sound(y10, fs);
pause(3);
```

```
% Playback is silent because the filter filters out (rejects)
% the frequency of the input sinusoid (800 Hz).  See Epilog.
```

% (e) Filter $x[n]$ using a five-point averaging filter and play the result

```
h5 = (1/5)*ones(1, 5);
y5 = filter(h5, 1, x);
sound(y5, fs);
pause(3);
freqz(h5, 1, fs, fs);
```

```
% From the freqz plot, the filter reduces amplitude of the cosine at
% 800 Hz (0.2pi) by about -3.7 dB.  AdB = 20 log10 A = -3.7 dB, which
% means that A = 10^(-3.7/20) = 0.653.  See Epilog.
```

% (f) Filter *x*[*n*] using a 15-point averaging filter and play the result

```
h15 = (1/15)*ones(1, 15);
y15 = filter(h15, 1, x);
sound(y15, fs);
pause(3);
freqz(h15);
```

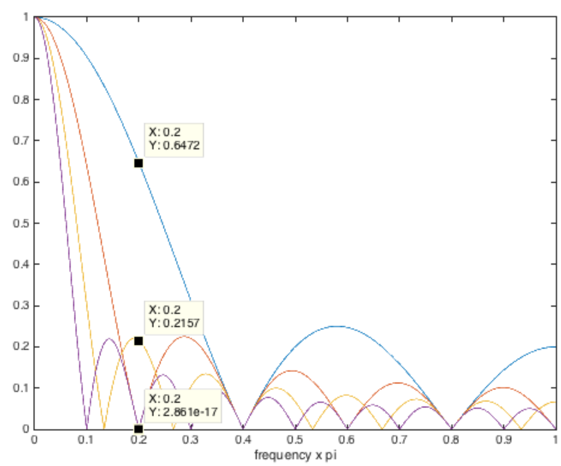
```
% From the freqz plot, the filter reduces amplitude of the cosine at
% 800 Hz (0.2pi) by about -13.37 dB, which is a gain of 0.2145.  See Epilog.
```

% (g) Filter *x*[*n*] using a 20-point averaging filter and play the result

```
h20 = (1/20)*ones(1, 20);
y20 = filter(h20, 1, x);
sound(y20, fs);
pause(3);
freqz(h20);
```

```
% Playback is silent because the averaging filter filters out (rejects)
% the frequency of the input sinusoid (800 Hz).  See Epilog.
```

% ***Epilog***. Here we superimpose the magnitude responses for the four averaging
% filters: 5-point (blue), 10-point (red), 15-point (yellow), and 20-point (purple).
% The data cursor indicates the magnitude response at 0.2*pi (i.e. 800 Hz).
% Lowpass filter: passes low frequencies and attenuates high frequencies.




```
fs = 8000;
for N = [5 10 15 20]
    coeffs = (1/N)*ones(1, N);
    [H, W] = freqz(coeffs, 1, fs);
    plot( W/pi, abs(H) );
    hold on;
end
xlabel('frequency x pi');
```

% *N*-point averaging filter: (a) extent in positive frequencies that have magnitude
% response in linear units close to 1 is proportional to 2*pi/*N*, and (b) zeros out
% discrete-time frequencies that are multiples of 2*pi/N but not multiples of 2*pi.