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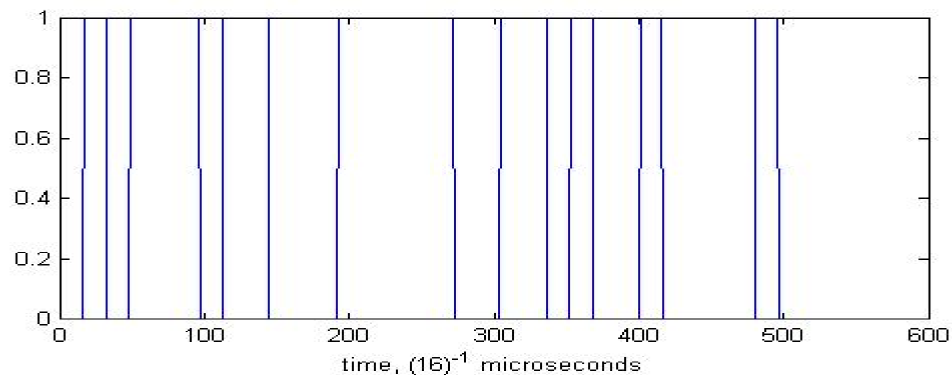
CES 400
Fall 2005

Project 3 – Noise reduction using a filter

a. Plot the signal.

```
a = [0 1 0 1 1 1 0 1 1 0 0 0 1 1 1 1 1 0 0 1 1 0 1 0 0 1 0 0 0 0 1 0 ]
for i = 1:32; for j = 1:16; S((i - 1)*16 + j) = a(i); end; end
plot(S) ; pbaspect([2,1,1]) ; xlabel('time, (16)^-^1 microseconds')
```

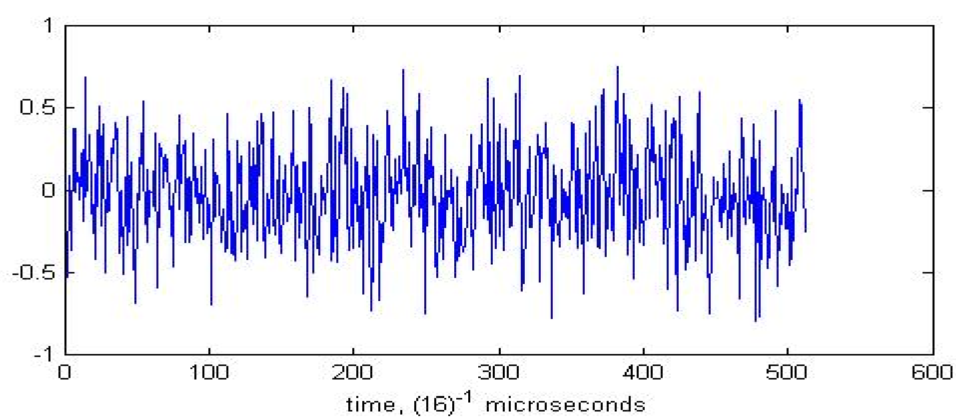


b. Signal average power is 0.5.

```
avP = sum(S)/512
```

c. Plot the noise.

```
Y = randn(1, 512)
Y = Y * sqrt(0.1)
plot(Y) ; pbaspect([2,1,1]) ; xlabel('time, (16)^-^1 microseconds')
```

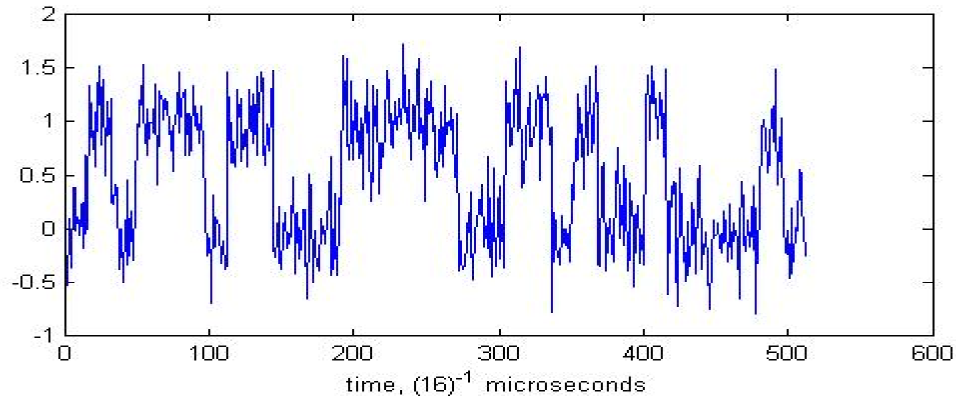


d. Noise average power = 0.09.

```
avPY = sum(Y.*Y)/512
```

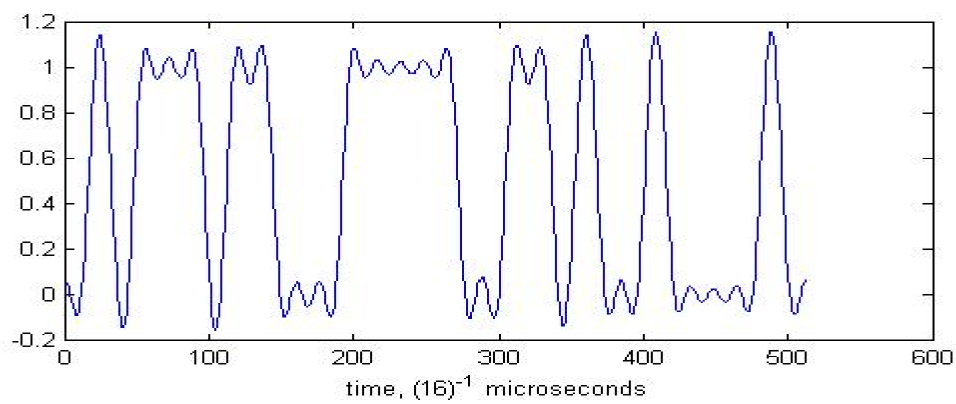
e. Plot signal + noise. SNR = 10*log (0.5/0.9) = 7.5 dB

```
plot(S+Y) ; pbaspect([2,1,1]) ; xlabel('time, (16)^-^1 microseconds')
```



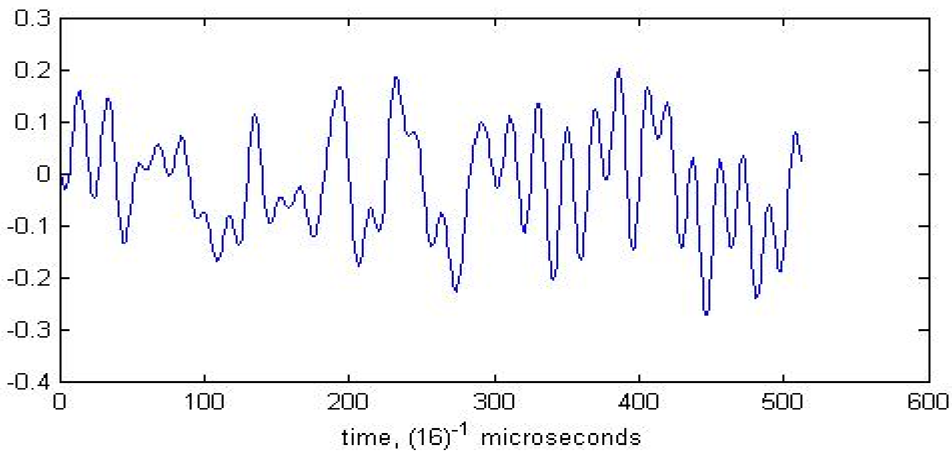
f. Plot the filtered signal. Since the period of the sampling signal is 32 x 10^{-6} s, then the frequency spacing of the points in Fourier domain is 1 / (32 x 10^{-6} s) = (1/32) x 10^{6} s^{-1}. This shows up as $e^{-jk(2\pi/T)t}$ in the Fourier series analysis equation. Signal power is 0.4760.

```
filter_S = (1/512)*fft(S)
for k = 34:480; filter_S(k) = 0; end
S1 = real(512 * ifft(filter_S))
plot(S1) ; pbaspect([2,1,1]) ; xlabel('time, (16)^-^1 microseconds')
avPY1 = sum(S1.*S1)/512
```



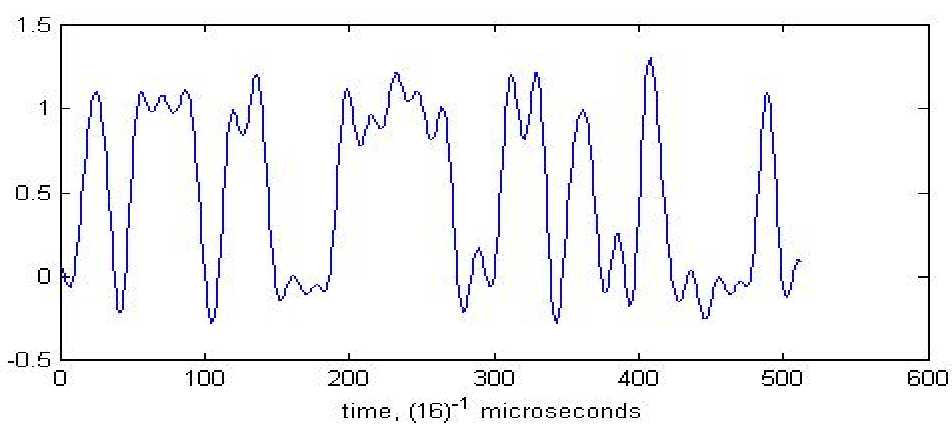
g. Plot the filtered noise. Noise power is 0.0107.

```
filter_Y = (1/512)*fft(Y)
for k = 34:480; filter_Y(k) = 0; end
Y1 = real(512 * ifft(filter_Y))
plot(Y1) ; pbaspect([2,1,1]) ; xlabel('time, (16)^-^1 microseconds')
avPY1 = sum(Y1.*Y1)/512
```



h. Add signal & noise together. Filter the combined signal. Plot the filtered (signal + noise).

```
SY = S1 + Y1
filter_SY = (1/512)*fft(SY)
for k = 34:480; filter_SY(k) = 0; end
filter_SY
SY1 = real(512 * ifft(filter_SY))
plot(Y1) ; pbaspect([2,1,1]) ; xlabel('time, (16)^-^1 microseconds')
```



i. SNR of the filtered (signal + noise) = 10*log(0.4760 / 0.0107) = 16.5 dB.

j. Conclusions: In the filtered (signal + noise), all the higher-frequency components that sharpen the vertical rise (because of speed) and smooth out the horizontal sections (by cancellation) have been filtered out. Also, the filtered noise is superimposed on the filter-distorted signal. So it's not a faithful replica of the input signal. But the signal-to-noise ratio has improved. In some applications this could be OK. For example if the signal is being used to switch a device between two states, and is in an environment where it picks up noise, then the filtered (signal + noise) would be usable, whereas the unfiltered (signal + noise) would not be usable.