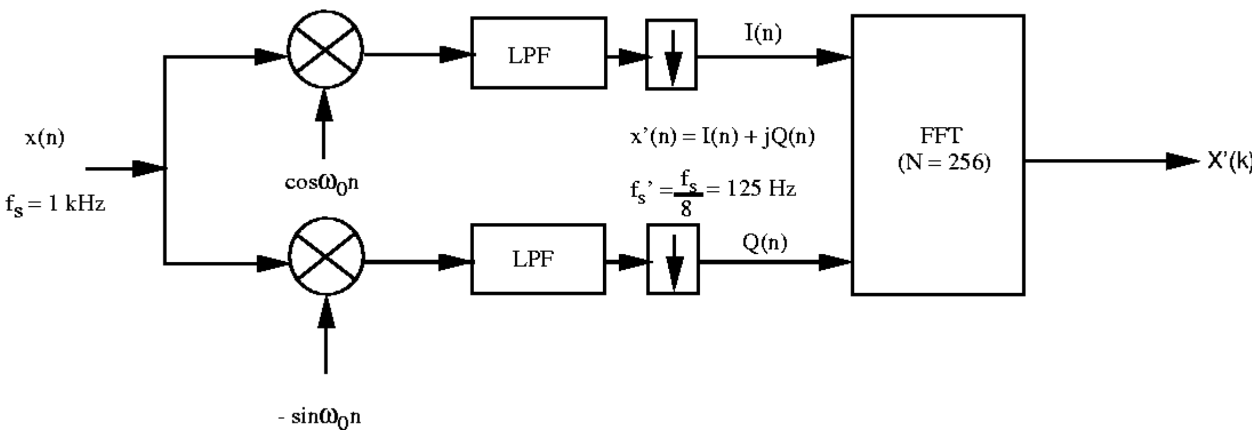
Mid-Term Project

Note: You should treat this project as a take-home exam. Thus, you should neither give nor receive assistance on completing the project. See the "Academic Integrity" document on TritonEd (Misc_Handouts/Overview folder). Select one of the following four project suggestions to work. Include in an appendix your Matlab code.

I. Complex Representation of Bandpass Signals

A. Complex Basebanding

As an alternative to the bandpass sampling architecture described in Sect. 11.4.3 (Oppenheim, Schafer, and Buck, 1998) or Sect. 12.4.3 (Oppenheim and Schafer, 2010), implement the following complex basebanding and FFT architecture in software:



Pass a data file consisting of three sinusoids through the complex basebander and observe the spectral characteristics of the signal at various stages through the structure. **Note**: See additional implementation details and specific questions to answer. This project is not available to students who previously have taken SIO 207A.

II. Homomorphic Signal Processing

A. Automatic Gain Control

Consider the following received signal modeling a fading channel situation: r(n) = a(n)s(n). Demonstrate the ability of homomorphic processing to compress and expand the envelope distortion on s(n) due to a(n). References available on TritonEd : (1) T.G. Stockham, "The application of generalized linearity to automatic gain control," IEEE Trans. Audio and Electroacoustics, AU-16: 267-270 (1968) (2) A. Oppenheim et. al., "Nonlinear filtering of multiplied and convolved signals," Proc. IEEE 56: 1264-1291 (1968). **Note:** See additional implementation details and specific questions to answer.

III. Power Spectral Estimation

A. Quantization Noise

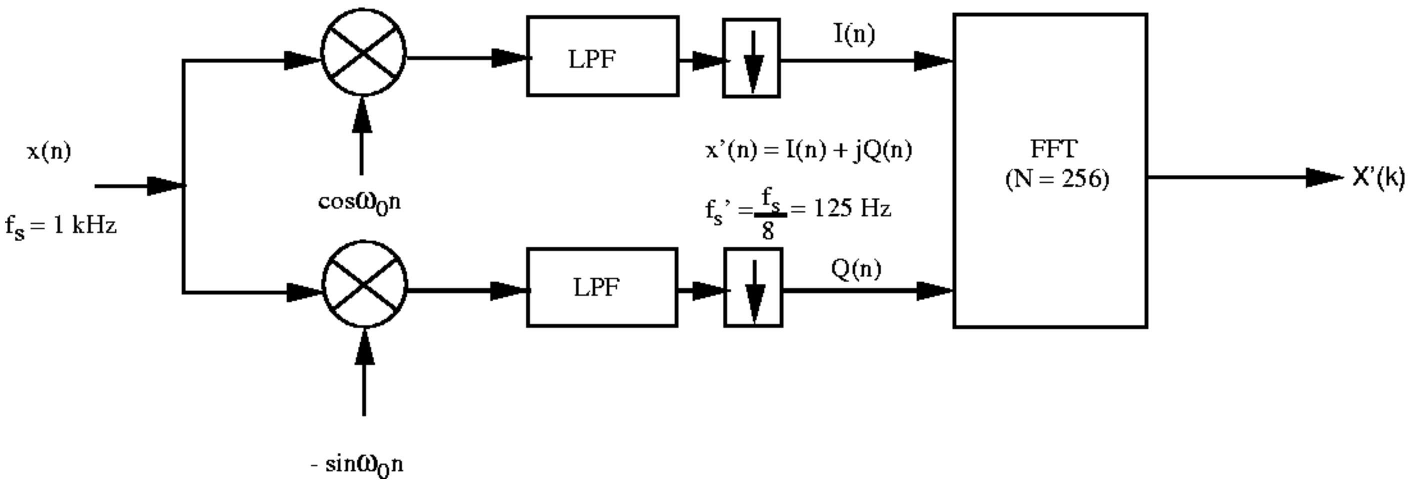
Investigate the correlation and spectral properties of quantization noise for various numbers of fractional bits. Specifically, let $x(n) = \sin\left(\frac{2\pi}{N}\right)n$, N = 256, and investigate quantization to 1, 3, and 7 fractional bits. Plot the N = 256 point time series x(n), Q[x(n)], and e(n) = Q[x(n)] - x(n). Plot the |FFT| (dB) of x(n) and Q[x(n)]. Estimate and plot the histogram, autocorrelation function, and power spectrum (dB) of e(n). Also investigate the cross-correlation between e(n) and x(n). See Chapters 3.7.3 and 11.7.2 (Oppenheim and Schafer, 1989), Chapters 4.8.3 and 10.7.2 (Oppenheim, Schafer, and Buck, 1998), or Chapters 4.8.3 and 10.6.2 (Oppenheim and Schafer, 2010).

B. Ambient Ocean Noise and Channel Impulse Response Estimation

Shallow water, vertical line array (VLA) acoustic data was collected north of Elba Island, Italy, in July 2004. The data available on TritonEd consists of ambient noise followed by the multipath arrival of a LFM chirp transmitted by a moving source 4.2 km away. First, estimate the time-evolving power spectrum of the observed time series from the deepest and most shallow hydrophones (Els #1 and #32). Second, estimate the channel impulse response from the source to each of the array elements. **Note:** See additional implementation details and specific questions to answer. This project is not available to students who previously worked on this data as a project for ECE 254.

Complex Basebanding

The following complex basebanding and FFT architecture is to be implemented in software:



LPF specifications: 64-coefficient, linear phase, FIR
passband cutoff frequency = 40 Hz (analog)
stopband cutoff frequency = 85 Hz (analog)
passband/stopband weighting ratio = 50

I. Data Set

A. Generate a 4096-point data file containing samples of the following signal:

$$x(t) = \sum_{l=1}^{3} s_l(t) \text{ where: } s_l(t) = A_l \cos\left(2\pi f_l\, t + \phi_l\right)$$

l	A_l	f_l	ϕ_l
1	100	160	0
2	10	237	0
3	1	240	0

Note: fs = 1 kHz (sampling rate)

B. Plot x(n) (n = 0, ..., 255) (amplitude). Take a N = 256 FFT of that portion of the data set using a KB window (α = 2.5 or β = 7.85). Plot | X(k)| (dB) and identify the locations of $\pm f_1$, $\pm f_2$, and $\pm f_3$. What is the bin width (analog) of the FFT? In what bins do f_1, f_2, and f_3 reside (bins numbered –N/2, …, 0, …, N/2-1)? Speculate on whether or not a rectangularly windowed FFT would have indicated distinct spectral peaks corresponding f_2 and f_3.

II. Decimation Filter Design

A. Using an equiripple FIR filter design algorithm, design the decimation filter.

B. Plot h(n) (amplitude). Take a N = 1024 FFT of h(n) (rectangular window). Plot |H(k)| (dB). Note that the sidelobe level should be –40 dB. Plot an expanded version of |H(k)| (dB) in the region $0 \le |f| \le 40$ Hz (analog) to illustrate the passband ripples.

III. Complex Basebanding and Desampling

A. Consider a center frequency $f_0 = 250$ Hz (analog). Implement the complex multiplication $e^{-j\omega_0 n} x(n)$. Take a N = 256 FFT of the new complex sequence for n = 0,..., 255 using a KB window ($\alpha = 2.5$ or $\beta = 7.85$). Plot | FFT| (dB) and identify the locations of $\pm f_1$, $\pm f_2$, and $\pm f_3$. Note: $\omega_0 = 2\pi(f_0/f_s)$

B. Pass the new 4096-point complex sequence through the LPF (i.e. implement the filtering operation in the time domain). Take a N = 256 FFT of the filtered complex sequence for n = 256,..., 511 using a KB window ($\alpha = 2.5$ or $\beta = 7.85$). Plot | FFT| (dB) and identify the locations of $\pm f_1$, $\pm f_2$, and $\pm f_3$.

C. Desample the complex filtered sequence by a factor of 8 (i.e. $f_s' = \frac{f_s}{8} = 125\,Hz$) yielding the 512-point sequence x'(n).

IV. High Resolution Spectral Analysis

A. Take a N = 256 FFT of x'(n) for n = 256, ..., 511 using a KB window ($\alpha = 2.5$ or $\beta = 7.85$). Plot $|X'(k)|$ (dB) and identify the spectral components present (i.e. $+f_1$, $+f_2$, $+f_3$, $-f_1$, $-f_2$, and $-f_3$). What is the bin width (analog) of the FFT?

B. Make recommendations regarding a second iteration in the design of the LPF. All specifications are to remain as originally given with the exception of the passband/stopband weighting ratio. Support your recommendations with plots as generated in II and IVA. Discuss both stopband attenuation and passband ripples.

Notes

(1) Work through the entire problem on paper *first* including hand-drawn sketches showing the impact of the filter sidelobe level and explain the aliasing mapping. Include as an appendix.

(2) All | FFT| plots must include both negative (to the left) and positive (to the right) frequencies. Rather than use the frequency index k as the horizontal axis variable, use f reflected back in terms of analog frequency (Hz) by multiplying normalized frequency f (-0.5 to 0.5 cycles/sample) by f_s or f_s’ as appropriate. Indicate explicitly the frequency of each component on the spectral plots (i.e. $+f_1$, $+f_2$, $+f_3$, $-f_1$, $-f_2$, and $-f_3$).

Automatic Gain Control

Consider the following received signal modeling a fading channel situation:

$r(n) = a(n)s(n) + n(n)$

where: $s(n)$ is the desired component

$a(n)$ is a multiplicative amplitude distortion

$n(n)$ is an additive noise component.

Assume prefiltering has been done such that $n(n) \approx 0$. Thus, $r(n) \approx a(n)s(n)$.

Let: $a(n) = 1 + A \cos\left(\frac{2\pi}{N}\right) k_d n$

$s(n) = \cos\left(\frac{2\pi}{N}\right) k_s n$

$A = \frac{1}{2}$, $N = 256$, $k_d = 4$, $k_s = 32$

Plot the following:

A. Time series

1. $a(n)$, $s(n)$, and $r(n)$.
2. $|a(n)|$, $|s(n)|$, and $|r(n)|$.
3. $\ln |a(n)|$, $\ln |s(n)|$, and $\ln |r(n)|$.
4. $\ln |r'(n)| = K \ln |a(n)| + \ln |s(n)|$, $K<1$ and $K>1$.
5. $r'(n) = a^K(n)s(n)$, $K<1$ and $K>1$.
6. $\ln |r''(n)| = h(n) * \ln |r(n)|$, envelope compressed and expanded.
7. $r''(n)$.

B. Spectra (dB) of all time series A(1) - A(7)

Comment on your results. Include a discussion on how you designed and implemented $H(f)$. Plot $|H(f)|$ (use a linear magnitude scale not dB).

Notes

(1) Consider carefully an appropriate way to handle cases when $s(n) \simeq 0$ or $r(n) \approx 0$.
(2) A(6) is most easily done via multiplication in the frequency domain.
(3) Must have $H(0) = 1$, otherwise DC term in $\ln|s(n)|$ will be altered.
(4) Use $K = 0.1$ and $K = 10$.

References

[1] T.G. Stockham, "The Application of Generalized Linearity to Automatic Gain Control," IEEE Trans. Audio and Electroacoustics, AU - 16, pp. 267 - 270 (1968).

[2] A. Oppenheim, et al, "Nonlinear Filtering of Multiplied and Convolved Signals," Proc. IEEE 56, pp. 1264 - 1291 (1968).

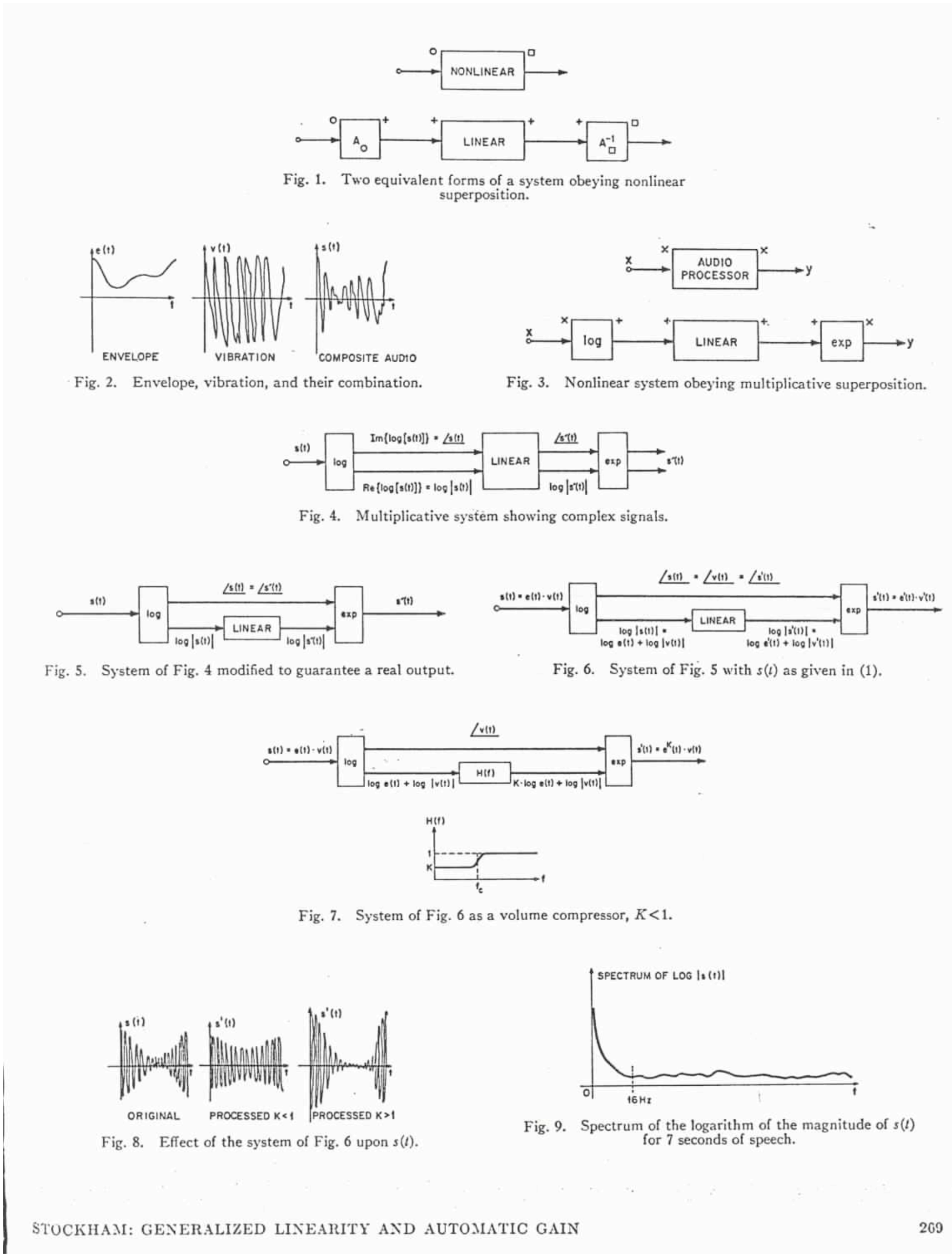


Fig. 1. Two equivalent forms of a system obeying nonlinear superposition.

Fig. 2. Envelope, vibration, and their combination.

Fig. 3. Nonlinear system obeying multiplicative superposition.

Fig. 4. Multiplicative system showing complex signals.

Fig. 5. System of Fig. 4 modified to guarantee a real output.

Fig. 6. System of Fig. 5 with $s(t)$ as given in (1).

Fig. 7. System of Fig. 6 as a volume compressor, $K<1$.

Fig. 8. Effect of the system of Fig. 6 upon $s(t)$.

Fig. 9. Spectrum of the logarithm of the magnitude of $s(t)$ for 7 seconds of speech.

STOCKHAM: GENERALIZED LINEARITY AND AUTOMATIC GAIN 269

Ambient Ocean Noise and Channel Impulse Response Estimation

Shallow water, vertical line array (VLA) acoustic data was collected north of Elba Island, Italy, in July 2004 [1]. The data available consists of ambient noise followed by the multipath arrival of a single LFM chirp transmitted by a moving source 4.2 km away.

The data consists of a 500 ms time series observed from 32 hydrophones spaced 2 m apart and sampled at fs = 12 kHz (see figure below). The signal conditioning for each array element includes a high pass filter to attenuate low frequency shipping noise and a low pass (anti-aliasing) filter prior to A/D conversion. The source transducer was at a range of 4.2 km, depth of 70 m, and towed at 4 knots. The transmission was a 2-4 kHz LFM chirp of duration 100 ms. The chirp arrival at the array begins approximately 200 ms into the time series with the beginning and end of the observation consisting of ambient noise.

Provide the following analyses of this data:

A. Time-Evolving Power Spectral Analysis

For both the deepest and most shallow hydrophones (Els #1 and #32), plot the time-evolving power spectrum (dB) (e.g. “spectrogram” in Matlab) as a color or gray scale plot (e.g. “imagesc” and “colorbar” in Matlab) and discuss the major features of the results. In addition, plot the averaged power spectrum (dB) (e.g. “pwelch” in Matlab) before and during the chirp arrival.

B. Channel Impulse Response Estimation

Assuming a channel impulse response of the medium $h_i(t)$ from the source to the ith element of the array, express analytically the output of the filter matched to the source waveform s(t) with the ith array element time series $r_i(t)$ as input in terms of the autocorrelation of the LFM chirp and $h_i(t)$.

Plot the time series of the LFM chirp, a spectrogram (dB) showing its time-evolving frequency content, and its spectrum (dB). With the LFM chirp s(t) as the input, plot the time series at the output of the filter matched to s(t) (optionally, the time series envelope computed via a Hilbert transform).

Using the known LFM chirp waveform, match filter (pulse compress) the multipath arrival structure observed on Els #1 and #32. Display the matched filter input and output time series (optionally, the matched filter output time series envelope computed via a Hilbert transform).

Similarly, match filter (pulse compress) the multipath arrival structure observed on all array elements to estimate the channel impulse response from the source to each element of the VLA. Display the matched filter output (or envelope) in a vertical waterfall display (see figure below) or as a color or gray scale plot.

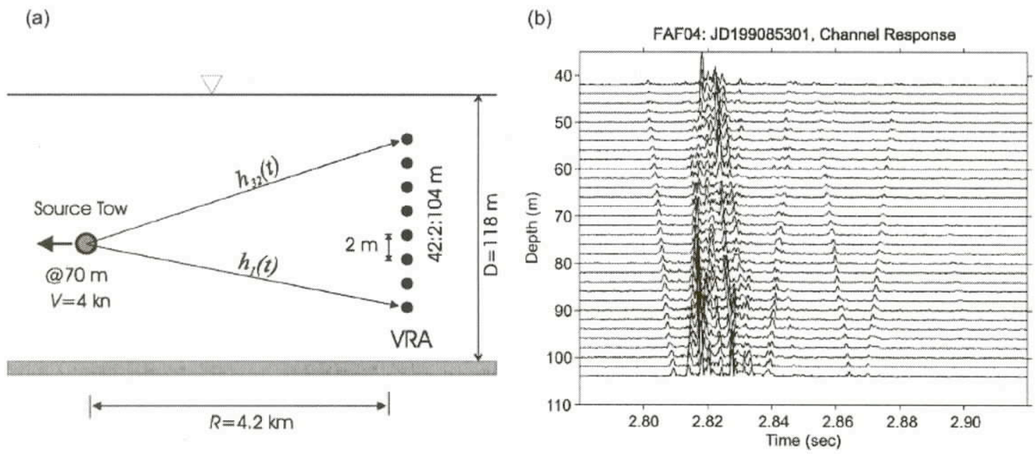


FIG. 8. (Color online) (a) Schematic of passive time reversal communications with a moving source at about 4 knots. (b) The channel responses (envelope) received by the vertical receiver array from a probe source at 70 m depth and 4.2 km range.

Reference

[1] H. Song et. al., "Spatial diversity in passive time reversal communications," J. Acoust. Soc. Am. 120(4): 2067-2076 (2006).