ES 442 Homework #6 – Solutions  
(Spring 2020 – March 11, 2020 )

Write all answers on homework pages – show your work.  
100 points total possible.

Question 1  Shannon-Weaver Model  (5 points)

According to the Shannon-Weaver model what elements or components 
(\textit{i.e.}, represented in a block diagram) must a communication system have? 

\textbf{Answer:} They are information source, transmitter, channel, receiver and 
destination (or recipient). Sometimes a noise source is included, but not 
necessary to receive full credit in this examination.  
See EE 442 Lecture 1, slide 14 for Shannon-Weaver model; Agbo & 
Sadiku Section 1.3 on pages 3-6.

![Block diagram of a communication system](https://example.com/block_diagram.png)

Question 2  Analog & Digital Signal Properties  (6 points)

List the principle properties for both physically realizable analog and digital 
signals.

\textbf{Analog:}  \begin{itemize}
\item (1) Continuous valued
\item (2) Time-varying
\item (3) Limited to a finite range of values (limited in magnitude) 
\textit{Also accepted the answer: causal}
\end{itemize}

\textbf{Digital} \begin{itemize}
\item (1) Discrete valued
\item (2) Time-varying
\item (3) Limited to a finite number of values 
\textit{Also accepted the answer: causal}
\end{itemize}

See EE 442 Lecture 1, slide 34 and EE 442 Lecture 2, slide 28.
Question 3 Adding Phasors  (10 points)

You are given two sinusoidal signals: $\varphi_1(t) = 2 \cdot \cos(\omega_1 t + \pi)$ and $\varphi_2(t) = \sin(\omega_2 t)$. Show how you would add these signals using phasors. But do not do the mathematics to find a value – show the basic principle you use in combining the two signals on the polar graph. You will want to assume a value for time $t$ to do the drawing.

Does linearity hold when combining signals?

Answer: Yes, linearity holds for phasors.  
See EE 442 Lecture 2, slide 19 for phaser definition.
\[ |V| = \sqrt{(2)^2 + (1)^2} = \sqrt{5} = 2.236 \]

\[
\tan \theta = \frac{y}{x} = \frac{1}{-2} \implies \theta = \tan^{-1}\left(\frac{-1}{2}\right) = 153.43^\circ
\]

We assumed \( t = 0 \) for this plot.

**Question 4 Using Single Tone for Analysis**  (5 points)

In EE442 we have used tone modulation to discuss the operation of communication components and systems. Of course, a tone modulation message signal carries no information. So why is tone modulation useful for analyzing communication components and systems?

**Answer:** Although a simple tone carries no information, actual communication signals are made up of combinations of sinusoids (we know this from Fourier analysis). It is far simpler to analyze a single sinusoidal signal and it provides insight into how a component or system operates.
For real signals we use the linearity property of the Fourier analysis by adding all of the sinusoids together forming the communication signal.

**Question 5  Group Delay Interpretation**  (5 points)

What is the physical meaning of the group delay of a two-port network (such as a transmission line of a filter)?

**Answer:** Group delay is how long it takes a signal to traverse a network or transmission line; it is the **transit time** (i.e., transit time of the energy flow of the signal). Added note: Group delay is defined as the rate of change of transmission phase angle with respect to frequency. See EE 442 Lecture 3, slides 31 to 34.

**Question 6  Channel Distortion**  (5 points)

To avoid signal distortion over a channel or transmission medium, what properties are required of the network or channel?

**Answer:** There are two properties required: (1) **Constant amplitude** and (2) **linear phase**. See EE 442 Lecture 3, slide 10.

**Question 7  Signal Power in a Receiver**  (10 points)

The receiver shown below has the following features: (1) An antenna for receiving the input RF signal, (2) a preselect filter with an attenuation loss of 5 dB, (3) a mixer with a 7.5 dB conversion loss (defined as the IF output power to the RF input power), (4) an IF filter with a 3.5 dB nominal loss (as defined as the ratio of the power at point B relative to the power at point A), (5) an IF amplifier with 26 dB gain, and a local oscillator (LO) delivering a power of +12 dBm to the mixer’s LO port.

![Diagram of receiver system](image)
If the power delivered to the input of the detector (at point C) must be at least –20 dBm to reliably detect the message signal, what is the lowest power in dBm that must be received by the antenna for the receiver to operate?

**Answer:** The equation for the losses and gain between the antenna and the input of the detector is

\[
P_{\text{antenna}} - 5 \text{ dB} - 7.5 \text{ dB} - 3.5 \text{ dB} + 26 \text{ dB} = -20 \text{ dBm},
\]

\[
P_{\text{antenna}} = -30 \text{ dBm} \quad \text{(or 1 microwatt received by antenna)}
\]

See Handout #1 on *The Definition of ‘dB’, ‘dBm’ and ‘dBW’.*

**Question 8 Signal Duration vs. Bandwidth** (5 points)

Explain the “signal duration versus bandwidth” tradeoff which arises when comparing the time domain viewpoint and the frequency domain viewpoint associated with a signal.

**Answer:** The bandwidth of a signal is inversely proportional to the width or duration of the signal. The time-scaling property of the Fourier transform says that narrower time signals (related to more rapid time variation) require larger bandwidths to support the faster time variations.

See EE 442 Lecture 4, slide 28. Also, Section 2.8.2 in Agbo and Sadiku on pages 46-48.
Question 9  AM Crystal Radio Operation  (9 points)

You are given a crystal radio schematic (shown below) like the foxhole radio we discussed in class. Here a germanium diode replaced the razor blade/safety pin “point contact” and a high impedance earphone is used. Explain how this radio works to (a) select a radio signal, (b) how the detector works (Hint: Is this an envelope detector or rectifier detector?), and (c) why can’t a person hear the carrier frequency?

Answer:
(a) The coil combines with capacitance capacitance to form a resonator thereby selecting the radio signal picked up by the antenna (the antenna has no selectivity at all). The AM signal drives the detector diode which rectifies the signal and the amplitude of the signal causes the earplug speaker can be heard when inserted into an ear. (b) This is not an envelope detector because no capacitance is used to hold the peak signal values; it is a rectifier detector. (c) The carrier frequency to too high to be heard by a person and furthermore the earphone is not capable of producing mechanical vibrations at the frequency of the carrier. See EE 442 Lecture 6, slides 14 through 17.

Question 10  DSB-SC Modulator  (10 points)

You are given a balanced modulator as shown below. It is used to generate a “double sideband-suppressed carrier “ (DSB-SC) signal.
Assume both AM modulators are identical. The information signal is denoted by $m(t)$ with the upper branch’s input being $m(t)$ and the lower branch’s input being $-m(t)$. The upper branch AM modulator is driven by an oscillator supplying $A \cdot \cos(\omega_c t)$. The lower branch’s AM modulator is driven from the same oscillator, but with an appropriate phase shift $\phi$. Be careful to note that the summing node sums the positive values of both branches (upper and lower).

(a) What must the phase shift $\phi$ be to generate the output DSB-SC signal as indicated in the diagram?

**Answer:** It must be $\pi$ radians or 180 degrees. This allows the two branch signals to constructively add together. 

**Note:** This problem makes use of Lecture 6, slide 62 for DSB-SC generation. Changing the sign on the summing node of the lower branch requires 180 degree phase shift to change recover the negative sign required to produce the DSB-SC signal. Admittedly, this is a hard problem.

(b) If the summing node subtracted the lower branch’s modulated from the upper branch’s modulated signal, then what would the phase shift angle be to generate DSB-SC?
Answer: It would have to be -π/2 radians, or -90 degrees (LO driven by \(\sin(\omega_c t)\)) to produce the DSB-SC output signal.

**Question 11 Advantages of Superheterodyne Receiver  (14 points)**

Explain what the advantages are in using a superheterodyne receiver. List the advantages (at least three) and draw a block diagram of a single-mixer superheterodyne receiver able to tune among multiple channels or stations.

**Answer:** Advantages of superheterodyne receiver are:

1. Superior selectivity because of use of a broadband mixer.
2. Able to receive multiple modulation schemes.
3. Able to receive very high frequencies that are down converted.
4. Much of the signal processing is done at a conveniently chosen IF frequency (which is incoming frequency independent).
5. Generally the gain, selectivity and sensitivity are superior.

A superheterodyne receiver is a type of radio receiver that uses frequency mixing to convert a received signal to a fixed intermediate frequency (IF) which can be more conveniently processed than the original carrier frequency. See EE 442 Lecture 6; slides 83, 85 and 90.

**Block Diagram:**
**Question 12: Switching Modulator**  (16 points)

Explain the basic principle behind the operation of a switching modulator (*i.e.*, how does a switching modulator work?). You can explain by showing a circuit diagram and explain it from the circuit schematic or give a generic explanation.

**Answer:**

A switching modulator works by turning the message signal on and off, alternatively at a set rate. In the square law modulator, a single diode is operated in a non-linear mode, whereas, in the switching modulator, the diode is driven very hard and operates as an ideal switch. Multiple diodes may be used such as the ring diode modulator topology.

The example presented in class was the balanced ring diode mixer (modulator) shown below. In this circuit four diodes are used as switches driven by a high level LO signal (at the carrier frequency).

![Circuit Diagram](image)

The DSB-SC output is a chopped signal with the amplitude set by the message signal $m(t)$ [$x(t)$ in the above diagram] as shown here:
The square wave’s switching rate (labelled LO) alternates the output amplitude at the carrier frequency (even though the carrier power is suppressed, the DSB-SC signal still propagates in a band centered upon the carrier frequency) and its envelope is the message signal. Thus, a receiver must intercept the modulated signal in a band centered upon the carrier frequency. Bandpass filtering smooths the waveform contained within the envelope, thus making it appear sinusoidal inside the envelope. It therefore contains only frequencies comprising the message signal. By this process the carrier and the message signals have be combined to form the modulation required.

See EE 442 Lecture 6, slides 64 to 76. See Section 3.3.1 in Agbo and Sadiku, pages 103 to 108.