ZJU-UIUC INSTITUTE

ECE-210 Analog Signal Processing Spring 2023 **Laboratory Exercise #5**

Student ID	3210111519	**Pre-lab**	**/26**
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Semester/Section		**Total**	**/80**

Lab 5: Sampling, reconstruction, and software radio

Until this point, your study of signals and systems has concerned only the continuous-time case[1], which dominated the early history of signal processing. About 60 years ago, however, the development of the modern computer generated research interest in digital signal processing (DSP), a type of discrete-time signal processing. Although hardware limitations made most real-time DSP impractical at the time, the continuing maturation of the computer has been matched with a continuing expansion of DSP. Much of that expansion has been into areas previously dominated by continuous-time systems: our telephone network, medical imaging, music recordings, wireless communications, and many more.

You do not need to worry whether the time and effort you have invested in studying continuous-time systems will be wasted because of the growth of DSP — digital systems are practically always hybrids of analog and digital subsystems. Furthermore, many DSP systems are linear and time-invariant, meaning that the same analysis techniques apply, although with some modifications. In this lab, you will explore some of the parallels between continuous-time systems and DSP with a "software radio" designed to the same specifications as the receiver circuit you developed on your protoboard.

1 Prelab

Our software radio is typical of many DSP systems in that both the available input and required output are continuous-time signals. The conversion of a continuous-time input signal to a discrete-time signal is called **sampling** (or A/D conversion), and the conversion of a discrete-time signal to a continuous-time output signal is called **reconstruction** (or D/A conversion). As discussed in class, **samples** $f(nT)$ of a **band-limited** analog signal $f(t)$ can be used to reconstruct $f(t)$ exactly when the **sampling interval** T and **signal bandwidth** $\Omega = 2\pi B$ satisfy the **Nyquist criterion** $T < \frac{1}{2B}$.

This is illustrated by the hypothetical system shown in Figure 1, where the analog signal $f(t)$ defined at the output stage of a low-pass filter $H_1(\omega)$ has a bandwidth $\Omega = 2\pi B$ limited by the bandwidth $\Omega_1 = 2\pi B_1$ of the filter. A/D converter extracts the samples $f(nT)$ from $f(t)$ with a sampling interval of T, and D/A conversion of samples $f(nT)$ into an analog signal $y(t)$ can be envisioned as low-pass filtering of a hypothetical signal $f_T(t) = \sum_n f(nT)\delta(t - nT)$ using the filter $H_2(\omega)$. With an appropriate choice of $H_2(\omega)$, the system output $y(t)$ will be identical to $f(t)$ in all its details as long as $T < \frac{1}{2B_1}$. The reason for that can be easily appreciated after comparing the Fourier transforms $F(\omega)$ and $F_T(\omega)$ of signals $f(t)$ and $f_T(t)$ with the help of Figure 2.

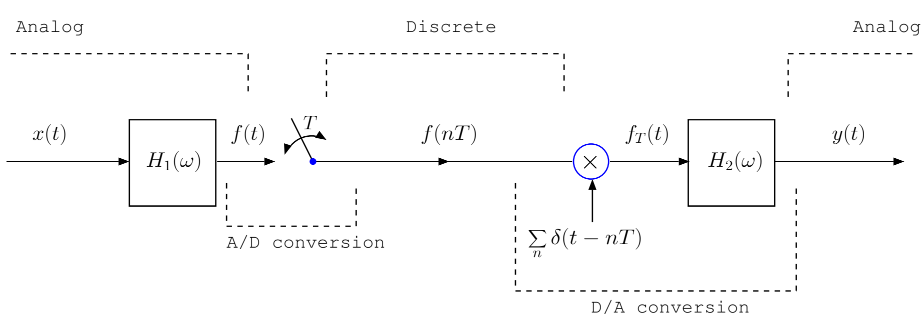


Figure 1: A conceptual system that samples a band-limited continuous-time signal $f(t)$ and reconstructs a continuous-time output $y(t)$. An A/D converter generates the samples $f(nT)$ of its analog input $f(t)$. An ideal D/A converter generates a signal $f_T(t) = \sum_n f(nT)\delta(t - nT)$ from samples $f(nT)$ and low-pass filters $f_T(t)$ with $H_2(\omega)$ to produce an analog $y(t)$. In a practical D/A converter the impulse train $\sum_n \delta(t - nT)$ is replaced by a practical pulse train $\sum_n p(t - nT)$ such that $p(t) * h_2(t)$ is a closed approximation of a delayed $\mathrm{sinc}(\frac{\pi}{T}t)$.

[1]The term "continuous time" is used generically to refer to signals that are functions of a continuous independent variable. Often that variable represents time, but it may instead represent distance, etc. "Discrete time" is used in the same way.

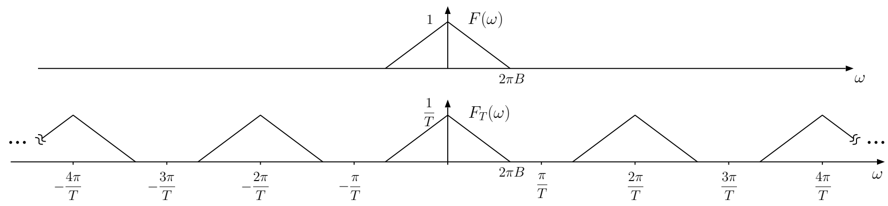


Figure 2: An example comparing the Fourier transforms of signals $f(t)$ and $f_T(t)$ defined in Figure 1. Since for $|\omega| < \frac{\pi}{T}$ the two Fourier transforms have the same shape, low-pass filtering of $f_T(t)$ yields the original analog signal $f(t)$. $F_T(\omega)$ is constructed as a superposition of replicas of $\frac{F(\omega)}{T}$ shifted in ω by all integer multiples of $\frac{2\pi}{T}$ (see item 25 in Table 7.2 in the text).

The following prelab exercises concern the system shown Figure 1. Assume that $T = \frac{1}{44100}$ s (i.e., the sampling frequency is $T^{-1} = 44100$ Hz) and signal $x(t)$ has a Fourier transform $X(\omega)$ shown in Figure 3.

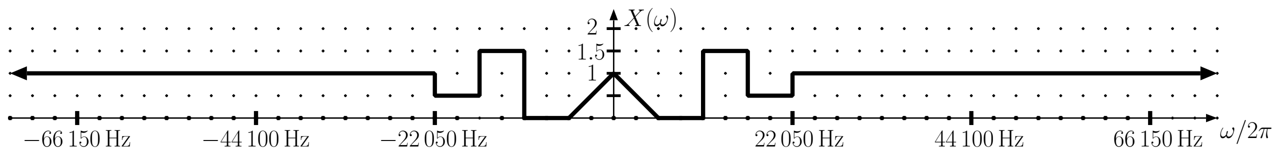


Figure 3: Fourier transform of $x(t)$ for prelab questions.

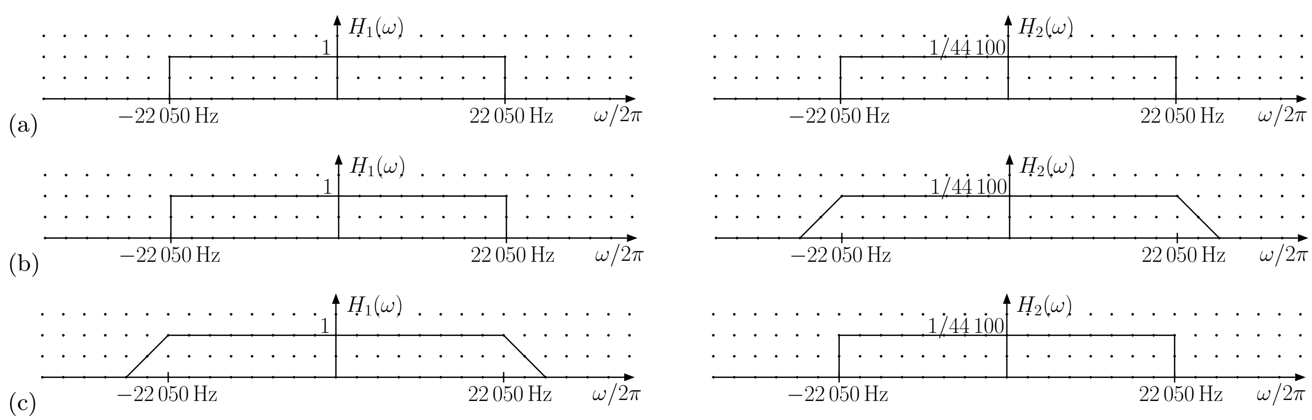
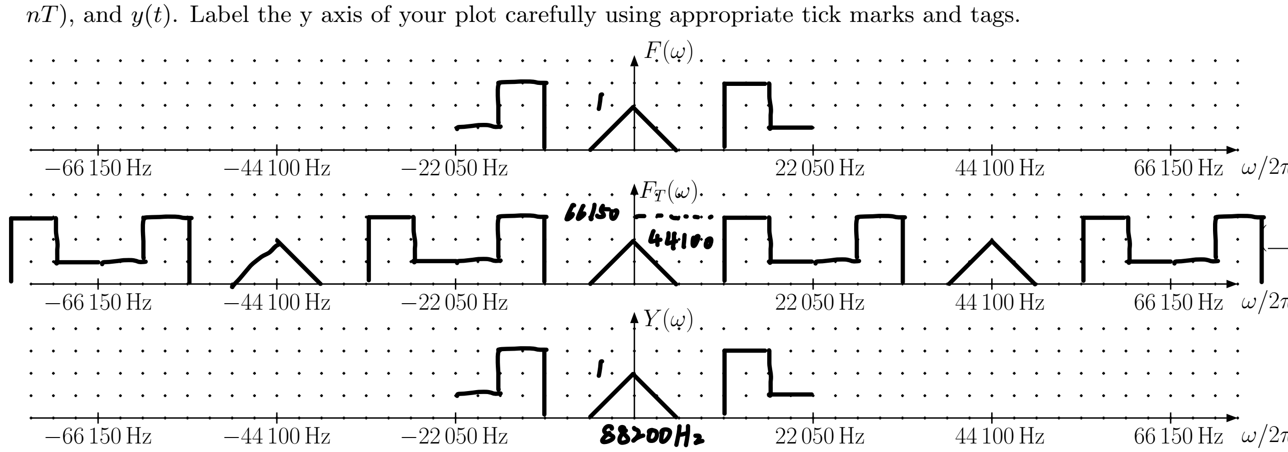


Figure 4: Filter frequency responses for the prelab problems.

1. For frequency responses $H_1(\omega)$ and $H_2(\omega)$ in Figure 4a, sketch the Fourier transform of $f(t)$, $f_T(t) = \sum_n f(nT)\delta(t-nT)$, and $y(t)$. Label the y axis of your plot carefully using appropriate tick marks and tags.



(____/4)

Is $y(t)$ a perfect reconstruction of $x(t)$? Explain your reasoning.

(____/2)

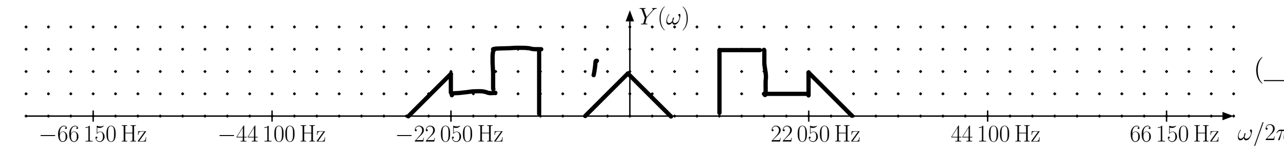
No, signal with $f > 22050$ Hz is lost

Is $y(t)$ a perfect reconstruction of $f(t)$? Explain your reasoning.

(____/2)

Yes, because f(t) is in the range of 22050 Hz

2. Now suppose an ideal $H_1(\omega)$ but a non-ideal $H_2(\omega)$ given by Figure 4b. The signals $f(t)$ and $f_T(t)$ are unchanged, but sketch the Fourier transform of the new $y(t)$.



(____/2)

Is $y(t)$ a perfect reconstruction of $f(t)$? Explain your reasoning.

(____/2)

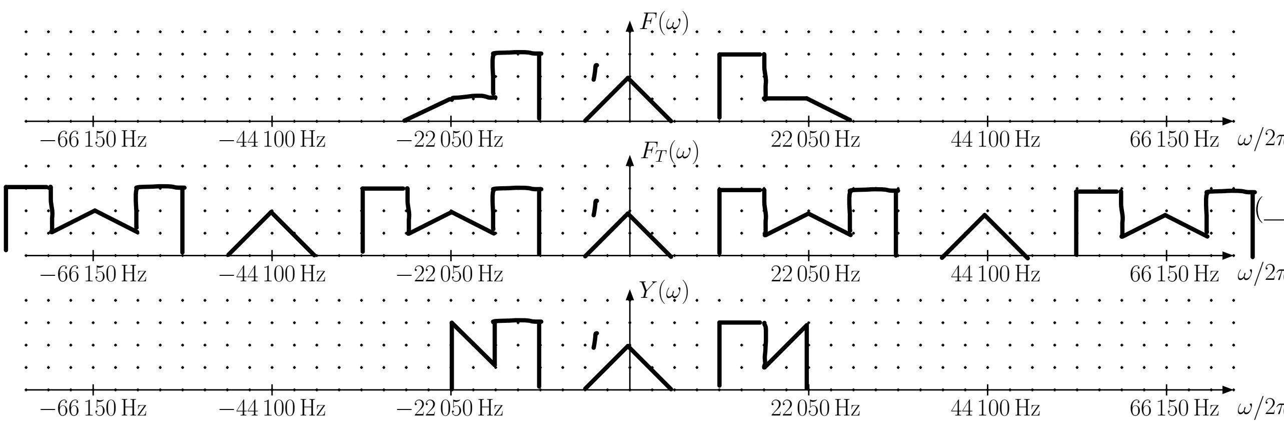
No, their fourier transform is not the same.

Is it still possible to have a perfect reconstruction of $f(t)$ based on the signal $y(t)$? Explain your reasoning.

(____/2)

Yes. use a LP filter to cut off frequency over 22500Hz

3. Now suppose a non-ideal $H_1(\omega)$ and an ideal $H_2(\omega)$ given by Figure 4c. Sketch the Fourier transform of $f(t)$, $f_T(t)$, and $y(t)$.



(____/4)

Is $y(t)$ a perfect reconstruction of $f(t)$? Explain your reasoning.

(____/2)

No, their frequency responce is different.

Is it still possible to have a perfect reconstruction of $f(t)$ based on the signal $y(t)$? Explain your reasoning.

(____/2)

No. because some data is losted in y(t), and it can't recover.

4. Discuss the role of the filter $H_1(\omega)$ in the system examined above. In what way does it impact the system output?

(____/2)

H1(w) is a LP filter, it's to avoid overlap cause by frequency shift.

5. Discuss the role of the filter $H_2(\omega)$ in the system examined above. In what way does it impact the system output?

(____/2)

H2(w) reduce the duplicate created by F1(w)

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2 Laboratory exercise

- Equipment: Function generator, PC with a sound card, and wires.
- Components: antenna, RF amplifier, mixer, stereo jack.
- Software: MATLAB (R2016a) and softRx6.m.

2.1 Sampling and Reconstruction

In this section, you will observe a real system much like the one you studied in the prelab exercises. The lab also will illustrate the phenomenon of aliasing which occurs when the analog input is under-sampled.

1. Connect the function generator's output to the "mic in" jack on the front of the computer at your lab station: Use BNC "Y" cables to bring the signal from lab equipment to the protoboard, then use a stereo jack and a stereo cable to run the signal from the protoboard to the computer. Ask your TA if you need help.

2. After you connect the stereo cable to the computer, right click on the speaker icon at the bottom right cornor of the screen and select Recording devices.

   (a) Under the Recording TAB, you should see one of the icons shows the microphone is plugged in, double click on it.

   (b) Under the Levels TAB, set Microphone to 100.

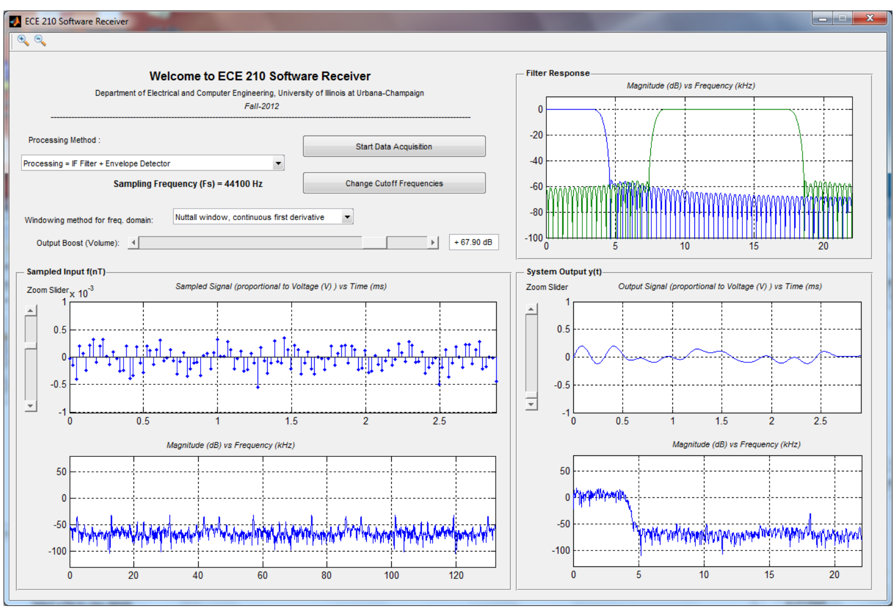


Figure 5: Screenshot of the softRx Graphical User Interface (GUI).

3. Start MATLAB: In Windows, go to Start –> All Programs –> MATLAB, and click MATLAB R2016a (MATLAB versions are continuously updated. If you do not see MATLAB R2016a, start the latest version installed on the computer). Wait for MATLAB to start and at the command prompt. Then go to "Open" button, go to "Computer", double click on Class (V:) folder and open ece210/lab5/softRx6.m file.

4. Start the softRx6.m code by pressing "Run", the green play button. This will launch the graphical user interface (GUI) for the software AM receiver shown in Figure 5.

5. Select "No processing" from the pull-down options menu near the top left corner of the GUI, in which case analog "line in" signal is sampled and reconstructed as an analog signal as depicted in Figure 6 and explained in the caption. Note the resemblance of the diagram to the system studied in the prelab. The analog low-pass filters shown in Figure 6 are part of the sound card and serve the same role as those in the prelab.

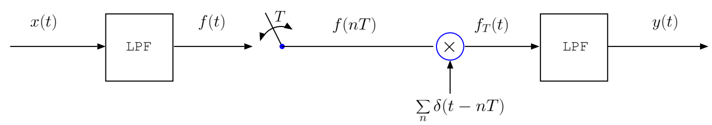


Figure 6: Block diagram of the system implemented by softRx when "No processing" option is selected. Note that no signal processing is applied to samples $f(nT)$ before D/A conversion.

6. Set the function generator, "High Z" mode and a 2 kHz sinusoid with amplitude 800 mV (High Z mode) peak-to-peak. Click the "Start Data Acquisition" button in the GUI to begin sampling and reconstruction.

   (a) Describe what you see in time domain of the sampled input signal $f(nT)$ (left column in the GUI). (____/2)

   There is a sine wave composed by many dots.

   (b) Describe what you see in frequency domain of the output $Y(\omega)$ (right column in the GUI). (____/2)

   There is a peak at 2kHz.

7. Slowly sweep the input frequency up to 20 kHz. Does the output $y(t)$ look like a perfect reconstruction of the input signal $x(t)$? Be aware that $x(t)$ is the signal you will see using an oscilloscope and is NOT the signal seen on the left column of the GUI. If not, then explain the differences. Disregard the amplitude and focus on the signal shape and the frequency content. (____/1)

   Yes

8. Now slowly sweep the input frequency from 20 to 24 kHz passing through 22.05 kHz. Describe what happens to the output signal in the time and frequency domain? (____/2)

   It become weaker, after 22.05kHz the Y(t) is 0

   At 24 kHz, does the output signal look like a perfect reconstruction or do we have an **aliased** component? (____/1)

   We have an aliased component.

   What is the significance of the frequency 22.05 kHz? (____/2)

   It's the threshoud of cutoff frequency.

9. Which component(s) in the system could be improved to reduce the aliasing effect? Note that the answer to this is not the sampling frequency. (____/2)

   Use a lowpass filter like W2

10. Finally, sweep the input frequency from 24 to 30 kHz. What do you observe? Explain what is happening.

(____/2)

Nothing changed about output, it's still 0 .

2.2 Digital Filtering

In this section, you will examine the digital filter option of softRx.

1. Stop the data acquisition and select the processing method to "Processing = IF Bandpass Filter" setting in the GUI. The samples $f(nT)$ will be processed by the digital filter shown in Figure 7 to generate samples $g(nT)$. You will be asked to enter two cutoff frequencies for the filter. Accept the default values (8 - 18 kHz) for this step.

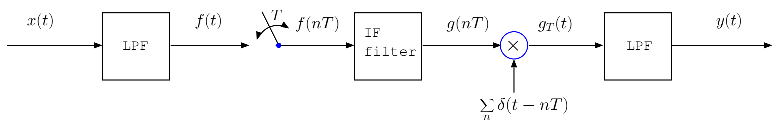


Figure 7: Block diagram for the Software Receiver when "Processing = IF Bandpass Filter" option is selected.

2. Note that the filter frequency-response depicted in the GUI depends on cutoff frequency inputs. Change them, after clicking "Change Cutoff Frequencies," to $f_{cl} = 6\,\text{kHz}$ and $f_{cu} = 9\,\text{kHz}$ and observe the new filter response.

3. Switch the input waveform from the sinusoid to a square wave with $f = 1.5\,\text{kHz}$ and 250 mV peak-to-peak(High Z mode). You should see a sinusoidal wave at the output. Indicate its frequency: $f =$ 7.5 kHz (____/2)

   Explain why do you have that output, when the input is a 1.5 kHz square wave. (*Hint: Having in mind the analysis of the periodic square wave done in lab #3, recall that the input signal should consist on decreasing odd harmonics of the fundamental frequency.*)

   (____/2)

   Explain the output.
   The only harmonic of the input signal is the 5th , 5×1.5=7.5kHz

4. Now return the processing method to "No processing", and change the input signal to a square wave with $f = 11\,\text{kHz}$ and 250 mV peak-to-peak (High Z mode). Describe and explain what you obtain at the output. (*Hint: What harmonic frequencies has the input signal now?.*)

   (____/2)

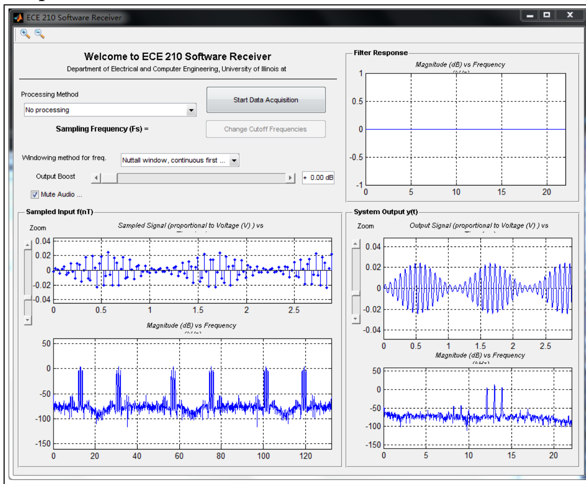
   Explain the output.
   The output is a sine wave with f=11kHz

2.3 Receiving Synthetic AM

Now you will generate a synthetic AM signal and process it with softRx.

1. Set the function generator to create an AM signal:
   - Set the function generator to create a 13 kHz sine wave measuring 250 mV peak-to-peak, no DC offset
   - Press "Shift" then "AM" to enable amplitude modulation
   - Press "Shift" then "<" then ∨ to select the shape of the message signal. You can select a sine, square, triangle, or arbitrary waveform by pressing ">". For this part of the lab, select a sine waveform.
   - Press "Enter" to save the change and turn off the menu
   - Press "Shift" then "Freq" to set the message signal frequency to 880 Hz
   - Press "Shift" then "Level" to set the modulation depth to 80%. This adjusts the DC component added to the message signal before modulation.

2. Select the "No processing" option. Sketch what you see on the GUI output. Disregard the spurious signals at frequencies lower than 5 kHz.



/2)

What does the strong impulse at $f = 13$ kHz represent in this AM signal? (____/2)

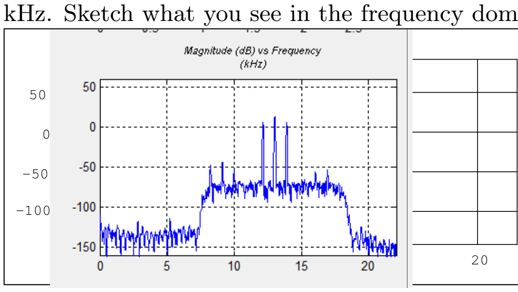
The carrier frequency

At what frequency locations are the other two impulses surrounding the 13 kHz impulse, and what do they represent? (frequencies in kHz with 2 decimals of accuracy) (____/2)

left = 12.11 kHz

right = 13.90 kHz

3. Select the "Processing = IF Bandpass Filter" option. Change the cutoff frequencies back to $f_{cl} = 8$ and $f_{cu} = 18$ kHz. Sketch what you see in the frequency domain of the output.



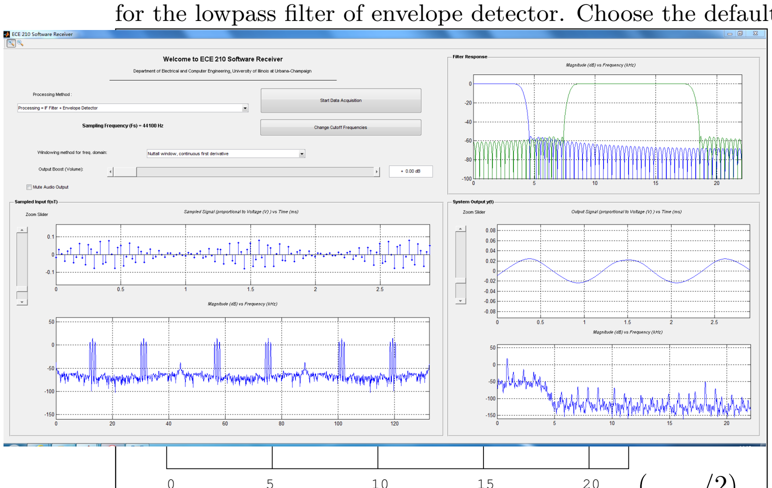
(____/2)

Explain why the noise has been reduced in some regions of the spectrum.

The bandpass filter stop the component with $f < 8$ kHz or $f > 18$ kHz

(____/1)

4. Select the "Processing = IF Filter + Envelope Detector" option. This introduces an envelope detector to the system as shown in Figure 8. You will be asked to enter three frequencies, two for the bandpass filter and one for the lowpass filter of envelope detector. Choose the default values. Sketch what you see on output panel.



(____/2)

In the time domain, what type of signal is that. Describe it. And in the frequency domain what is the dominant signal and how large is it in decibels, compared with the other signals?

it's a sine wave.

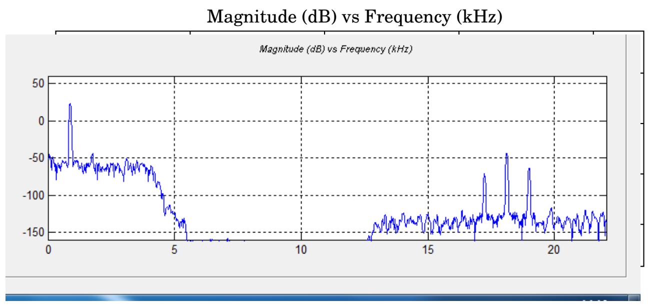
$f < 5$ kHz, Mag ≈ -60 dB

$f \geq 5$ kHz Mag ≈ -150 dB

(____/2)

5. Select the "Processing = IF Filter + Coherent Detector[2]" option. This implements a coherent detector. You will be asked to enter three frequencies, two for the bandpass filter and one for the lowpass filter of envelope detector. Choose the default values. Sketch what you see on frequency domain of the output panel.

[2]A coherent detector works as follows. For a received signal of the form $m(t)\cos(\omega_c t + \theta_o)$, the carrier frequency ω_c and the phase delay θ_o are estimated. Then a cosine signal with the same carrier frequency and phase delay is generated and mixed with the received signal, yielding $m(t)\cos(\omega_c t + \theta_o)\cos(\omega_c t + \theta_o) = m(t)\cos^2(\omega t + \theta_o) = \frac{1}{2}m(t)(1 + \cos(2\omega_c t))$. Finally a lowpass filter is applied to filter out the $2\omega_c$ modulated signal in order to obtain the message $m(t)$.



(____/2)

Describe the differences in frequency domain between the output of the "coherent detector" and the "envelope detector".

The component between 5-12.5 kHz is reduced by coherent dector.

(____/2)

6. Overall, what does the envelope detector or the coherent detector accomplish?

(____/2)

It gets the message signal

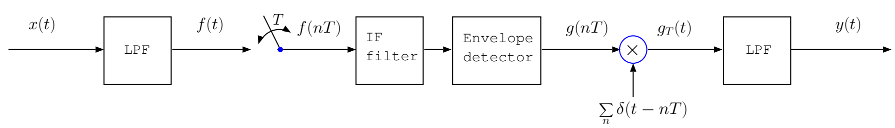


Figure 8: Block diagram for the Software Receiver when "Output = Envelope Detected Input" is selected.

2.4 Receiving Broadcast AM

In this last section, you will mix a broadcast AM signal to an IF of 13 kHz, as in Lab 4. You will then process the IF with softRx and eventually listen to its detected version.

1. Connect the antenna, RF amplifier, and frequency mixer modules provided (as in Lab 4). Use the DC source to power the modules by connecting +15 V to blue terminals, -15 V to purple terminals, and the bench ground ("Com") to black terminals. Also connect the Com to a power outlet ground using a banana-to-banana cable.

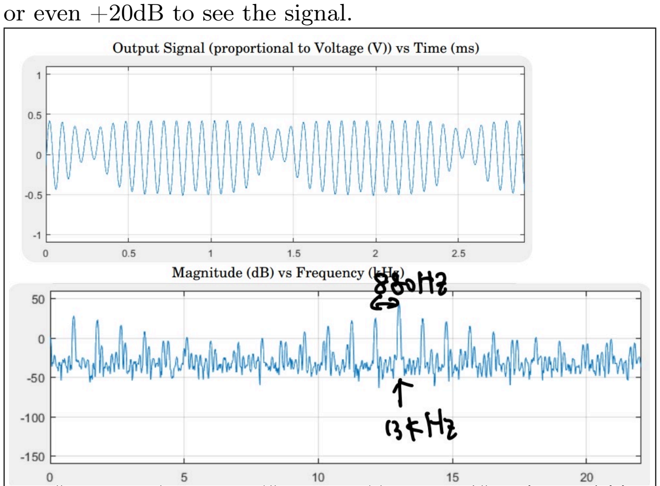
2. Use the function generator as your local oscillator (L.O.). Generate a sinusoid with 100 mV peak-to-peak(High Z mode) amplitude (do not forget to turn off the AM feature of the function generator). **What frequencies could be used as L.O.** to tune to AM 1400 kHz or AM 580 kHz radio station via 13 kHz IF?

(____/2)

For AM: 1400kHz, LO=1387kHz. For AM: 580kHz, LO=567kHz

Select one of those frequencies and connect the function generator to LO input of the frequency mixer.

3. Connect the output of the frequency mixer to "mic in" on the sound card. Select "No processing" option in the GUI. Sketch what you see on the output panel. You might need to set the system Microphone Boost to +10dB or even +20dB to see the signal.



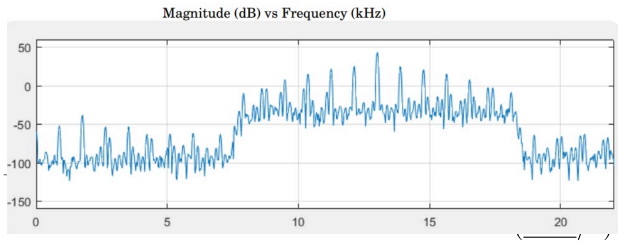
You should identify an impulse in the frequency domain. What is the frequency of that impulse?

main peak: 13 kHz

distance: 880Hz

(____/2)

4. Select "Processing = IF Bandpass Filter" option. Set the cutoff frequencies of the IF filter to $f_{cl} = 8$ and $f_{cu} = 18$ kHz. Sketch the frequency domain of the output.

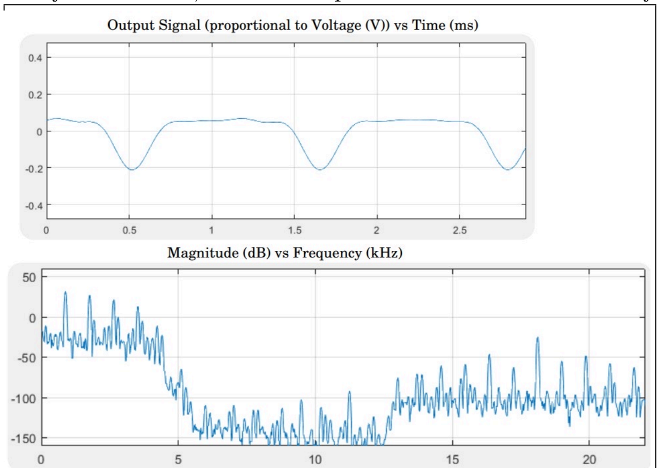


Approximately what is the bandwidth of the output signal? (Hint: This is a bandpass signal)

10 kHz

(____/1)

5. Select the "Detector" of your preference and sketch output. Set the cutoff frequencies of the IF filter to $f_{cl} = 8$ and $f_{cu} = 18$ kHz, and the lowpass filter of the detector to $f_c = 4$ kHz.



In the frequency domain what is approximately the bandwidth of the output signal? (Hint: It is a lowpass signal)

4kHz

(____/2)

How do the signals in each of these settings resemble those you observed in the continuous-time case during Lab 4? Explain briefly the differences.

(____/2)

In Lab 4, the signal without frequency filter only has 3 peaks. But this signal has many peaks.

6. Connect a pair of loudspeakers to "speaker out" or use the internal loudspeakers to listen to AM 1400 or WILL 580. Your software radio should be working now. Note: You might need to move the antenna or change the way you are holding the antenna in order to increase signal quality. Touching the antenna increases the noise level, so in order to increase the signal level use the "Output Boost (Volume)" in the softRx GUI.

The End

Congratulations on completing the ECE 210 lab! Over these five labs you have learned and applied the most important principles of continuous-time signals and systems and explored their parallels in discrete-time signals and systems. Advanced coursework in ECE will require you to apply these principles again and again. You are well prepared!

DO NOT FORGET TO RETURN YOUR PROTOBOARD AND COMPONENTS TO YOUR TA!