



**King Fahd University of Petroleum and Minerals**

**Department of Electrical Engineering**

**EE-370**

**Communication Engineering I  
Laboratory Manual**

**By**

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# PREFACE

Welcome to the EE370: Communication Engineering-I Lab. As course & Lab instructors, we do hope that you will enjoy working out the experiments. The idea of these experiments is that you will do a combination of MATLAB simulation experiment followed by a hardware circuit experiment to really grasp and fully understand the main concepts of communication engineering. We tried our best to relate what you are learning in class with what you are taking in the lab. The following outlines the general policies for the lab:

## Grading Policy:

- 8% for Lab Reports
  - 2% for Pre-labs
  - 4% for Lab performance
  - 6% for Lab Final
- which will add-up to 20% for the Lab grade.

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## Lab Report:

- You must submit the report at the beginning of the new following lab.
- The report must have a cover page.
- It must have the following:
  1. Lab objective.
  2. Clear theoretical concepts & equations used.
  3. Description of the lab results or data collected and plotted.
  4. Conclusions.
  5. Appendices showing your source code and results of the simulations or plotted data.

**Note:** You can follow the sample report included below.

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## Important Notes:

- 1) Always bring an empty floppy disk with you.
  - 2) Each unexcused absence will result in a grade of ZERO for that experiment.
  - 3) As per university rules, **three** unexcused absences will cost the student a DN grade.
  - 4) No make-ups are allowed.
  - 5) Any kind of cheating in is forbidden and will be treated according to the university rules. Just do your work independently.
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Lab Report Sample

King Fahd University of Petroleum & Minerals  
Electrical Engineering Department

EE370  
Communication Engineering I

Lab Report for Experiment No#

*Place the title of the experiment over here*

Prepared for

*(Place the name of your lab instructor)*

By

*(Place your)* Name  
ID No.  
Section No.

*(Place the date of the lab experiment)*

- Objective:

*Place the objective of the experiment here*

- Equations, general concepts and theoretical background:

*Place the equations if they exist & your general fundamentals of the experiment here*

- Observations & discussion:

*Write your comments independently regarding the simulations you've done or the results you've obtained through the whole experiment in details. They should be very clear, well written & logically convincing the reader of the report. You have to refer to the simulations or plots accordingly*

- Conclusion:

*Write a brief conclusion stating the main lessons that you've learnt from running the experiment.  
DON'T REPEAT THE OBJECTIVE HERE OR DESCRIBE THE EXPERIMENT AGAIN!*

- Appendix I: *(put Matlab Code here, or other relevant material)*

- Appendix II: *(put Matlab simulation figures, or other lab plots and results)*

# LIST OF EXPERIMENTS

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# Fourier Series & Fourier Transforms

## MATLAB Simulation

### Objectives

Fourier analysis plays an important role in communication theory. The main objectives of this experiment are:

- 1) To gain a good understanding and practice with Fourier series and Fourier Transform techniques, and their applications in communication theory.
- 2) Learn how to implement Fourier analysis techniques using MATLAB.

### Pre-Lab Work

You are expected to do the following tasks in preparation for this lab:

- MATLAB is a user-friendly, widely used software for numerical computations (as you learned in EE207). You should have a quick review of the basic commands and syntax for this software. The following exercises will also help in this regard.

Note: it is important to remember that Matlab is vector-oriented. That is, you are mainly dealing with vectors (or matrices).

- 1) Consider the following code:  **$Y=3+5j$** 
  - a. How do you get MATLAB to compute the magnitude of the complex number  $Y$ ?
  - b. How do you get MATLAB to compute the phase of the complex number  $Y$ ?
- 2) Vector manipulations are very easy to do in MATLAB. Consider the following:

```
xx=[ones(1,4), [2:2:11], zeros(1,3)]  
xx(3:7)  
length(xx)  
xx(2:2:length(xx))
```

Explain the result obtained from the last three lines of this code. Now, the vector **xx** contains 12 elements. Observe the result of the following assignment:

```
xx(3,7)=pi*(1:5)
```

---

Now, write a statement that will replace the odd-indexed elements of **xx** with the constant  $-77$  (i.e., **xx(1)**, **xx(3)**, etc). Use vector indexing and vector replacement.

3) Consider the following file, named example.m:

```
f=200;
tt=[0:1/(20*f):1];
z=exp(j*2*pi*f*tt);
subplot(211)
plot(real(z))
title('REAL PART OF z')
subplot(212)
plot(imag(z))
title('IMAGINARY OF z')
```

- How do you execute the file from the MATLAB prompt?
- Suppose the file name was "example.cat". Would it run? How should you change it to make it work in MATLAB?
- Assuming that the M-file runs, what do you expect the plots to look like? If you're not sure, type in the code and run it.

## Introduction

Recall from what you learned in EE207 that the input-output relationship of a linear time-invariant (LTI) system is given by the convolution of the input signal with the impulse response of the LTI system. Recall also that computing the impulse response of LTI systems when the input is an exponential function is particularly easy. Therefore, it is natural in linear system analysis to look for methods of expanding signals as the sum of complex exponentials. Fourier series and Fourier transforms are mathematical techniques that do exactly that!, i.e., they are used for expanding signals in terms of complex exponentials.

### Fourier Series:

A Fourier series is the orthogonal expansion of periodic signals with period  $T_o$  when the signal set  $\{e^{j2\pi nt/T_o}\}_{n=-\infty}^{\infty}$  is employed as the basis for the expansion. With this basis, any given periodic signal  $x(t)$  with period  $T_o$  can be expressed as:

$$x(t) = \sum_{n=-\infty}^{\infty} x_n e^{j2\pi nt/T_o}$$

where the  $x_n$ 's are called the Fourier series coefficients of the signal  $x(t)$ . These coefficients are given by:

$$x_n = \frac{1}{T_o} \int_0^{T_o} x(t) e^{-j2\pi nt/T_o} dt$$

This type of Fourier series is called the exponential Fourier series. The frequency  $f_o = 1/T_o$  is called the fundamental frequency of the periodic signal. The  $n^{\text{th}}$  harmonic is given by the frequency  $f_n = nf_o$ .

If  $x(t)$  is a real-valued periodic signal, then the conjugate symmetry property is satisfied. This basically states that  $x_{-n} = x_n^*$ , where  $*$  denotes the complex conjugate. That is, one can compute the negative coefficients by only taking the complex conjugate of the positive coefficients. Based on this result, it is obvious to see that:

$$\begin{aligned} |x_n| &= |x_{-n}| \\ \angle x_n &= -\angle x_{-n} \end{aligned}$$

### Fourier Transforms:

The Fourier transform is an extension of the Fourier series to arbitrary signals. As you have seen in class, the Fourier Transform of a signal  $x(t)$ , denoted by  $X(f)$ , is defined by:

$$X(f) = \int_{-\infty}^{\infty} x(t)e^{-2\pi ft} dt$$

On the other hand, the inverse Fourier Transform is given by:

$$x(t) = \int_{-\infty}^{\infty} X(f)e^{2\pi ft} df$$

If  $x(t)$  is a real signal, then  $X(f)$  satisfies the following conjugate symmetry property:

$$X(-f) = X^*(f)$$

In other words, the magnitude spectrum is even while the phase spectrum is odd. There are many properties satisfied by the Fourier Transform. These include Linearity, Duality, Scaling, Time Shift, Modulation, Differentiation, Integration, Convolution, and Parseval's relation.

## Lab Work

### Part A: Fourier Series

- 1) In MATLAB, go to the command window and type **Fourier\_series\_demo.m**. This will bring up a Graphical User Interface (GUI) that can be used to test and demonstrate many concepts and properties of the Fourier series expansion.



- Try different types of functions, starting with the square wave, fully rectified sine, sawtooth, etc. Run different examples while changing the fundamental frequency and number of harmonics in the FS expansion. Report your observations. In particular, explain why the Fourier series for the square and sawtooth waves require many more harmonics than the rectified sine waves in order to get a close match between the FS and the original function?
  - Consider the plots for amplitude and phase spectra. State what can of symmetry is present in each type of spectrum, and why? The plots also indicate the presence of FS terms with “negative” frequencies! What’s the interpretation of that? Are there really negative frequencies? Explain.
- 2) Now, consider a periodic signal  $x(t)$ . Compute and plot the discrete magnitude and phase spectra of this signal given by  $x(t) = e^{-t/2}$  where  $t \in [0, \pi]$ . For this, you need to use the Fast Fourier Transform (FFT) function in MATLAB (refer to the notes below for more details). For the expansion of the signal  $x(t)$ , the number of harmonics  $N_o$  to be used should be 32, the period  $T_o$  is  $\pi$ , and the step size is  $t_s = T_o / N_o$ . The output should be in two figure windows. The first window should contain  $x(t)$  while the second window should contain both the magnitude and phase spectra versus a vector of harmonics indices (for example,  $n$ ). You also need to include labels and titles in all plots. What can you observe from these plots?

**Notes:** In MATLAB, Fourier series computations are performed numerically using the Discrete Fourier Transform (DFT), which in turn is implemented numerically using an efficient algorithm known as the Fast Fourier Transform (FFT). Refer to the textbook (Sect.2.10 & 3.9) for more theoretical details. You should also type: **help fft** at the MATLAB prompt and browse through the online description of the **fft** function.

Because of the peculiar way MATLAB implements the FFT algorithm, the **fft** MATLAB function will provide you with the positive Fourier coefficients including the coefficient located at 0 Hz. You need to use the even amplitude symmetry and odd phase symmetry properties of the Fourier series for real signals (see the introduction to Fourier series of this experiment) in order to find the coefficients for negative harmonics.

As an illustration, the following code shows how to use **fft** to obtain Fourier expansion coefficients. You can study this code, and further enhance it to complete your work.

```
Xn = fft(x,No)/No;
Xn = [conj(Xn(No:-1:2)), Xn];
Xnmag = abs(Xn);
Xnangle = angle(Xn);
k=-No/2+1:No/2-1
stem(k, Xnmag(No/2+1:length(Xn)-No/2))
stem(k,Xnangle(No/2+1:length(Xn)-No/2))
```

**Useful MATLAB Functions:** exp, fft(x,No), length( ), conj, abs, angle, stem, figure, xlabel, ylabel, title.

## Part B: Fourier Transform

- 3) In the MATLAB command window, type **Fourier\_trans\_demo.m** to launch a GUI that will demonstrate and review the basic properties of the Fourier transform. The basic function used is a rectangular unit pulse.
- First, introduce a certain time delay in the function, and notice what happens to the amplitude spectra. Explain why?
  - Next, introduce different scaling factors and comment on what you are observing.
  - Now, introduce a frequency shift, which means that the unit pulse is multiplied by a given sine or cosine signal with some frequency (later, we will see this is known as Amplitude Modulation). Referring to the basic properties of the FT, explain what you are observing in the plots.
- 4) Now, consider the signals  $x_1(t)$  and  $x_2(t)$  described as follows:

$$x_1(t) = \begin{cases} t + 1, & -1 \leq t \leq 0 \\ 1, & 0 < t \leq 1 \\ 0, & \text{elsewhere} \end{cases}$$

$$x_2(t) = \begin{cases} t, & 0 \leq t \leq 1 \\ 1, & 1 < t \leq 2 \\ 0, & \text{elsewhere} \end{cases}$$

Plot these signals and their relative spectra in MATLAB. What do you conclude from the results you obtained? Are there any differences?

You need to plot both time signals in one figure window. Similarly, you need to plot the magnitude and phase spectra for both signals in one figure window, i.e, overlapping each other. For the phase, display small values by using the **axis** command. You also need to normalize the magnitude and phase values, and you should include the labels, titles, grid, etc. Assume the x-axis to work as a ruler of units. Each unit contains 100 points and let the starting point to be at -5 and the last point to be at 5.

**Notes:** Similar to Fourier series, Fourier transform computations in MATLAB are easily implemented using the **fft** function. The following code illustrates that. Notice in particular the function **fftshift** is very useful for presenting the Fourier spectrum in an understandable format. The internal algorithm used in MATLAB to find the FFT points spreads the signal points in the frequency domain at the edges of the plotting area, and the function **fftshift** centers the frequency plots back around the origin.

```
X = fft(x);
X = fftshift(X);
Xmag = abs(X);
Xmag = Xmag/max(X1mag);%Normalization
Xangle = angle(X);
Xangle = Xangle/max(Xangle);
F = [-length(X)/2:(length(X)/2)-1]*fs/length(X);
plot(F, Xmag), plot(F, Xangle);
```

- 5) Repeat the above for the following signals, and report your observations&conclusions

$$x_1(t) = \begin{cases} 1, & |t| \leq 3 \\ 0, & \text{elsewhere} \end{cases}$$

$$x_2(t) = \begin{cases} 1, & |t| \leq 1 \\ 0, & \text{elsewhere} \end{cases}$$

- 6) In the MATLAB directory you are working in, you will find a MAT-file named **Exp1Part4.mat**. You need to load that file as follows:

```
load Exp1Part4.mat
```

After you successfully loaded the file, go to the command window and type **whos** and press **Enter**. You will notice three stored variables fs (sampling frequency or 1/ts), t (time axis vector) and m (speech signal). These correspond to a portion of speech recording.

The next step is to plot the speech signal versus the time vector t. In the same figure window and a second window panel, display the magnitude spectrum of m (call it M). What is the bandwidth of the signal? What can you notice in terms of the speech signal? In order to play the signal properly, make sure that the speakers are turned on and write the following MATLAB statement:

```
sound(m,fs)
```

---

## The Analog Communication Board (ACB)

### Hardware Experimentation

#### Objectives

The objective of this lab is to give the students a first introduction to the Analog Communication Board (ACB), which will be extensively used in subsequent experiments.

- The Lab will enable the students to identify the different circuits of the ACB, and understand their respective functions.
- The configuration of the different circuit blocks to implement analog communication systems is introduced.
- Some testing and experimentation with the ACB, Function Generator and Oscilloscope equipment is also included.

#### Pre-Lab Work

In preparation for this lab, do the following tasks:

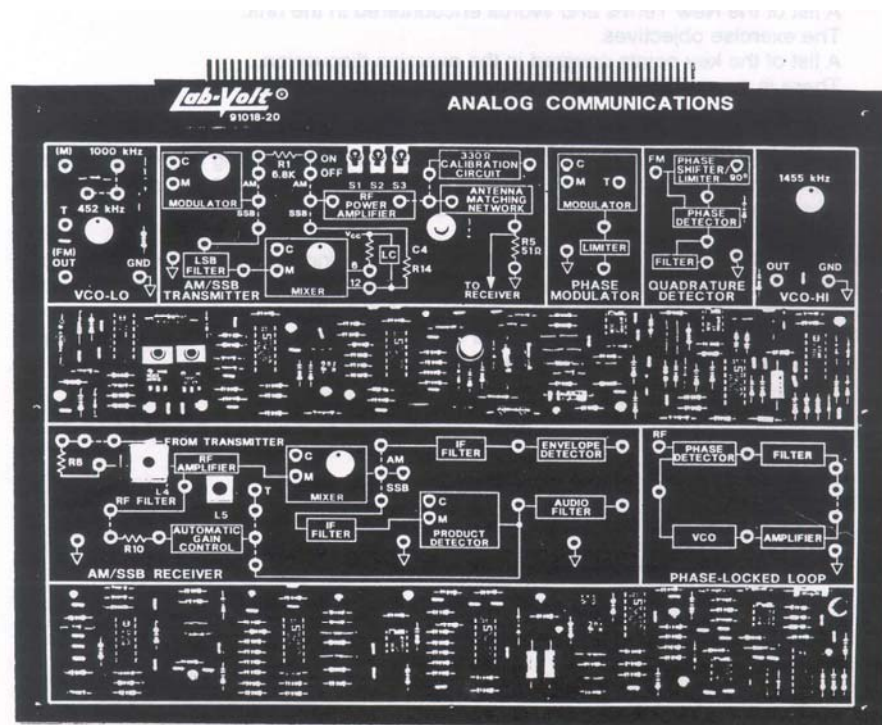
- 1) Consider a linear time-invariant (LTI) system such as a filter, etc. Explain how to characterize the input-output relationship for such a system. Give the results both in time-domain and in frequency-domain (using convolution, frequency response, etc).
- 2) The function generator that you will be using in the lab generates many special functions such as sine waves, rectangular pulses, triangular signals, etc. How do you classify these signals: as “energy” signals or “power” signals? Explain why.
- 3) Assume now for simplicity that these signals are truly periodic, with a given period  $T_0$ . Give the Fourier series for each type of the basic waveforms: sinusoidal, rectangular and triangular.

## Overview

The ACB is made of several building blocks, including the following:

1. Voltage-Controlled Oscillators: VCO-LO (low frequency), and VCO-HI (high frequency)
2. AM/SSB Transmitter
3. AM/SSB Receiver
4. Phase Modulator
5. Phase Locked Loop (PLL)
6. Quadrature Detector

The architecture of the ACB is illustrated in the following block diagram:



Analog Communications circuit board

The basic operation of the different ACB blocks is briefly summarized below. Throughout many of the subsequent experiments, you will become more familiar with their different functions.

### 1. VCO-LO & VCO-HI:

These voltage-controlled oscillators generate sinusoidal signals, with VCO-LO providing frequencies in the range of 452 KHz or 1000 KHz, and VCO-HI a single frequency in the range of 1455 KHz. These signals are typically used in the modulation & demodulation operations. The basic operation of the VCOs is described as follows:

- Choosing either frequency for VCO-LO is done by switching a two-post connector in the corresponding position.
- The potentiometer knob adjusts the oscillator amplitude.
- The negative supply knob on the upper left side of the unit allows the user to do small adjustments of the frequency of VCO-LO, while the positive supply knob adjusts the frequency of VCO-HI.
- VCO-LO can also be used for frequency modulation (FM).

### 2. AM/SSB Transmitter:

This block can be configured to generate an AM signal (including a carrier component), a Double-Sideband Suppressed-Carrier (DSB-SC) signal or a Single Sideband (SSB) signal. The AM/SSB block can be further divided into the following circuits:

#### **Modulator:**

This component is a balanced modulator used for the generation of AM or DSB-SC signals. It basically combines the message signal  $m(t)$  with the carrier signal  $c(t)$  to produce a modulated signal of the form  $[A + m(t)] c(t)$ . (DSB-SC modulation is obtained when  $A = 0$ ).

#### **RF Filter:**

This block filters the signal to the desired transmission band and it also provides a power amplifier to boost the signal power so that it can be radiated over a long distance.

#### **Mixer:**

This is a three-port system that works as a signal multiplier. It multiplies two signals (one is the message signal, and the other is the carrier signal). Notice that this essentially to the modulator functionality. However, a mixer can also be used in other applications (including frequency translation, demodulation, etc).

#### **S1-S2-S3 Switches:**

When these switches are in the ON positions, they function as follows:

S1: produces a DSB signal at the Modulator output (disables the modulator potentiometer).

S2: produces a DSB signal at the Mixer output (disables the mixer potentiometer).

S3: automatically matches the antenna impedance.

**Note:** R5 is a  $51\Omega$  resistor that emulates the antenna impedance.

### 3. AM/SSB Receiver:

This block can be configured to receive and demodulate an AM, DSB-SC or SSB signal. It consists of the following:

#### **R8:**

This is a  $1\text{ M}\Omega$  resistor that emulates transmission losses.

#### **RF Filter:**

This is a tunable band-pass filter (BPF). It should be tuned to pass the desired transmission band and reject other noise and out-of-band interference.

#### **RF Amplifier:**

This amplifier is used to boost the level of the weak received signal so that can be processed and demodulated.

#### **Mixer:**

This is used to translate the signal from its RF band to an intermediate frequency (IF) set at 455 KHz.

#### **IF Filter:**

This is a good quality (high Q) band-pass filter (BPF) that is factory-tuned at a fixed IF frequency (455 KHz). This filter can be tuned to provide the desired selectivity to select the AM band for a given transmitting station.

#### **Envelope Detector:**

This is used for non-coherent envelope detection of AM-modulated signals.

#### **Product Demodulator:**

This is used for coherent demodulation of DSB and SSB signals.

#### **Audio Filter:**

This is a LPF that passes the audio band and rejects higher frequencies.

#### 4. Phase Modulator:

This block is used for Phase Modulation (PM). It consists of a phase modulator and an amplitude limiter, and is used to demonstrate PM operation with a message signal of 5 KHz sinusoidal tone and a carrier frequency of 452 KHz.

#### 5. PLL:

The PLL block is a very useful component in communication systems. It has three main components: a phase comparator, a LPF and VCO. The PLL is used for many applications in communication, including: carrier acquisition and tracking, FM demodulation, etc. A subsequent lab will provide more details on the PLL.

#### 6. Quadrature Detector:

This block is used for FM detection, and consists of three main components: a phase shifter & limiter, a phase detector and a LPF.

### Lab Work

1. Use the oscilloscope to measure the range of amplitudes and frequencies that are available for VCO-LO and VCO-HI.
2. Use the Signal Generator to apply a sinusoidal signal with frequency 1000 KHz to the input of the input of the RF amplifier of the transmitter circuit. Use the appropriate equipment to measure the voltage gain and phase shift of the resulting output signal.

**Note:** You should set the signal amplitude at low level (for example  $V_{pp} = 50\text{mV}$ ), because the RF amplifier gain is very high.

3. In this part, you are expected to measure the amplitude transfer function  $H(f) = V_{out}/V_{in}$  of the IF filter. The procedure is as follows. First apply a sinusoidal signal from the external function generator to the input port of the IF filter. Vary the frequency of this signal in the range of 435KHz to 465KHz, in steps of 5KHz, and measure  $V_{out}$  at each frequency (you can set the input amplitude to a fixed value in order to simplify the computation of  $H(f)$ ). You should also reduce the frequency step size around the frequency of 455 KHz (use, for example, 1 or 2 KHz).
4. From the measured transfer function  $H(f)$ , determine the center frequency of the IF filter and its 3-dB bandwidth. Recall that the 3dB cutoff frequency is defined as the frequency for which the output amplitude gain drops to  $1/\sqrt{2}$  of its maximum value.

**Note:** Recall that the signal power is proportional to the squared amplitude. Hence, a  $1/\sqrt{2}$ -factor in signal amplitude corresponds to a  $1/2$ -factor in signal power, which in terms of dB units corresponds to 3dB. This explains why we use the term 3dB bandwidth in the above part.

5. Now, change the input signal shape to apply a rectangular pulse stream from the external signal generator. Set the frequency of this signal in the range of the central frequency (around 455 KHz) with  $V_{pp} = 1\text{V}$ , and observe the output signal. Record your observations and explain your results. What happens if you change the rectangular signal frequency?



(Hint: refer to the pre-lab exercise related to the Fourier series expansion of the rectangular signal).

## Homework Questions

- Q1.** If a message signal  $\cos(2,000 \pi t)$  and a carrier signal  $\cos(20,000\pi t)$  are fed to a mixer, what is the resulting output signal? Assume that the mixer passes the higher frequency term and suppresses the lower one.
- Q2.** Consider Part 5 above and imagine the case where the input rectangular pulse train had double the central IF frequency (i.e., around  $2 \times 455\text{KHz}$ ). Explain, in a simple and precise way, the kind of output you expect to get from this filter.

# Amplitude Modulation

## MATLAB Simulation

### Objectives

The main objectives of this experiment are:

- 1) To gain a clearer understanding about double-side band suppressed carrier (DSB-SC) and amplitude modulation (AM)
- 2) To learn how to simulate modulation/demodulation systems for DSB-SC and AM using MATLAB for synthetic & real signals (such as speech).

### Pre-Lab Work

- 1) Read the relevant material in your textbook (Chapter 4).
- 2) Using MATLAB, perform the following:
  - a.  $X=[0,1,2,3,4,5]$
  - b.  $Y=[1,2,3,4,5,6]$

Now multiply  $X$  by  $Y$  using two ways. The first one is the usual MATLAB multiplication (star  $X*Y$ ) and the other one is what is called as point-wise array multiplication ( a dot followed by a star  $X.*Y$ ). What is the difference between the two?

- 3) Using MATLAB, generate a vector  $t = [0:0.001:1]$ . Then generate  $m = \cos(2*\pi*t)$  and  $v = \cos(4*\pi*t)$ . Plot  $m$ ,  $v$ , and the product  $x=mv$ . Are you going to use multiplication between matrices or vectors that are representing functions?

### Introduction

Amplitude modulation (AM) is the family of modulation schemes in which the amplitude of a sinusoidal carrier is changed as a function of the modulating message signal. This type of modulation schemes includes many variants, such as double-sideband suppressed carrier (DSB-SC), single-sideband (SSB), conventional AM, and vestigial-sideband (VSB). Refer to

your textbooks for ample details on amplitude modulation techniques. In this lab, we focus in particular on DSB-SC and conventional AM.

DSB-SC AM:

In DSB-SC AM, the amplitude of the modulated signal is proportional to the message signal. The time-domain representation of this scheme is given by:

$$y(t) = A_c x(t) \cos(2\pi f_c t)$$

where  $A_c \cos(2\pi f_c t)$  is the carrier signal with a carrier frequency  $f_c$ , and  $x(t)$  is the message signal. The transmission bandwidth is twice the bandwidth of the message signal.

Conventional AM:

AM is similar to DSB-SC, but it also includes a pure carrier (non-modulated) component in the transmitted signal. The message signal  $x(t)$  is replaced by  $[1 + \mu x_n(t)]$ , where  $x_n(t)$  is the normalized message signal and  $\mu$  is the modulation index. Therefore the AM signal will be:

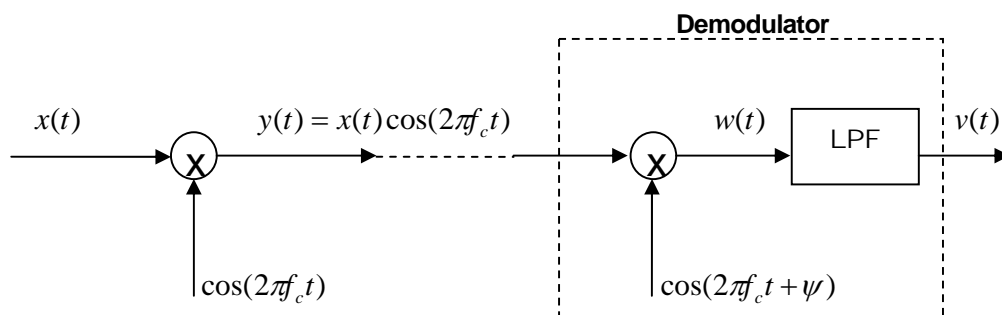
$$y(t) = A_c [1 + \mu x_n(t)] \cos(2\pi f_c t)$$

The existence of the sinusoidal component makes the AM scheme less economical in terms of power utilization as compared to the DSB-SC scheme. However, the demodulation of AM signals is much cheaper than the demodulation process of DSB-SC signals. The conventional AM demodulation process is simply done by employing envelope detectors.

For the bandwidth, the AM signal has the same transmission bandwidth as the DSB-SC transmission bandwidth.

## Lab Work

- 1) Use MATLAB to simulate the following block diagram



Assume  $\psi = 0$  and let  $x(t) = \cos(2\pi 2000t)$ . Use a carrier frequency of  $f_c = 20$  kHz. Plot  $x(t)$ ,  $y(t)$ ,  $w(t)$ , and  $v(t)$ , and their magnitude spectrums each

in a two-panel figure. Define the time vector  $t$  as  $[0:200]*t_s$  where  $t_s$  is the step size given by  $t_s = 1/(10f_c)$ .

At the receiver end, you need to design a Low Pass Filter (LPF). In MATLAB, you can use a type of filters known as Butterworth filters. For example, you can design a given filter with some order  $n$  and cut-off frequency  $f_c$  which is typically normalized in Matlab and given by  $2f_c t_s$  (where  $t_s$  is the sampling period). To obtain the filter coefficients, the statement will be: `[num,den]=butter(n, 2*f_c*t_s)`, where **num** and **den** are the numerator and denominator coefficients of the rational function representing the analog filter. You can use  $n = 5$  for example. Once you obtained these coefficients, you can use the Matlab function **filter** to filter the signal  $w(t)$  using the designed LPF. That is, `v = filter(num,den,w)`.

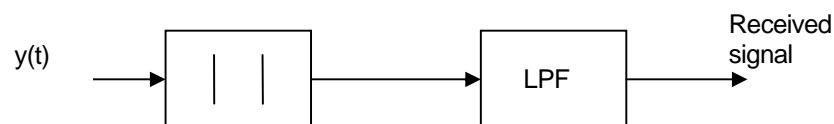
Refer to the additional notes below for further discussion on how to use filters in Matlab.

**Useful MATLAB Functions:** `cos`, `fftshift(fft( ))`, `butter`, `filter`, `abs`, `plot`, `subplot`, `figure`, `xlabel`, `ylabel`, `title`.

- 2) Repeat Part 1 with 2 different values for the receiver phase offset:  $\psi = \pi/2$  &  $\pi$ . What do you notice at the receiver end? Is there any difference between the recovered signal here and the one obtained in Part 1? Why is that? And what is the solution to this problem?

Repeat Part 1 by making  $y(t) = A_c(1 + \mu x_n(t))\cos(2\pi f_c t)$  where  $\mu$  is the modulation index of the AM wave,  $A_c$  is the carrier amplitude (set it equal to 4), and  $x_n(t)$  is the normalized version of  $x(t)$ . Set it to be 0.5 (50% modulation).

- 3) At the demodulator, you can implement the functionality of the simple envelope detector that you studied in class (built with a diode, a capacitor and resistor) by using simple MATLAB code to produce full-wave rectification (absolute value function), followed by low-pass filtering. This is illustrated in the following block diagram. You need to think about setting the appropriate cutoff frequency for the LPF. In addition, you can also add a mechanism to remove the DC component from the signal



- 4) Repeat Part 3 by letting the modulation index equal to 1.2. What will happen to the received signal? Explain.

- 5) Load the file called Exp3Part5.mat. This data file contains:
- A vector called **ms**, which is a speech signal sampled with  $t_s = 1/96E3$  s.
  - A vector called **t** that represents time.

With a carrier of 24kHz, transmit and receive ms using the AM system in part 3 with  $\mu = 0.5$  &  $1.5$ . For both cases show the following:

- In one figure with two panels, the time and frequency domain representations of the modulated waves
- Listen to **ms** by typing (**sound(ms,96E3)**, **pause**, then press **Enter** to continue). Also listen to the received signal **v** by typing **sound(v,96E3)**. Comment on the differences between the two signals.

**Note:** in order to be able to see the spectrum of the signal **ms**, after plotting the magnitude spectrum of **ms**, (denoted by **Ms**) vs. **f**, type:

**axis([-4E3,4E3,0,max(|Ms|)])**

**Useful MATLAB Functions:** load, sound, pause, axis.

### Additional Notes (Filters in MATLAB):

To further understand how to use filters in Matlab, recall from EE207 that the rational transfer function of a filter can be expressed in terms of the Laplace variable  $s$  by:

$$H(s) = \frac{b_m s^m + b_{m-1} s^{m-1} + \dots + b_1 s + b_0}{a_n s^n + a_{n-1} s^{n-1} + \dots + a_1 s + a_0}$$

where the a's and b's define the transfer function coefficients. These coefficients completely characterize the filter response. Matlab returns these two vectors as a result of designing a filter with a certain type (e.g., Butterworth, etc) and cutoff frequency. For example, with  $\text{num}=[b_m, b_{m-1}, \dots, b_0]$  and  $\text{den}=[a_n, a_{n-1}, \dots, a_0]$ , the filter design is done by :  $[\text{num}, \text{den}] = \text{butter}(n, 2 f_c t_s)$ ,

Notice also that the filter order  $n$  is important to specify. For example, it is shown that for these Butterworth-type filters, as the filter order increases the filter response will approach that of an ideal "brick-wall" response.

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# DSB-SC & AM Modulation with coherent & non-coherent Detection

## Hardware Experimentation

### Objectives

The goal of this lab is to allow the students to experiment with various types of amplitude Modulation & demodulation hardware systems. The detailed objectives include the following:

- 1) Implementing DSB-SC with coherent detection.
- 2) Implementing AM with non-coherent envelope detection.
- 3) Experimenting with the super-heterodyne receiver structure for commercial AM radio.

### Pre-Lab Work

Answer the following questions:

1. Explain the difference between coherent and non-coherent detection of AM signals
2. Can DSB-SC be detected non-coherently with an envelope detector? Explain why.
3. Explain the meaning of “super-heterodyne” receiver (refer to your textbook, Ch.4).

### Overview

Refer to your textbook and to the previous experiment for an overview of AM modulation and the related mathematical expressions. Also refer to Experiment 3 for a description of the ACB board that will be used in this lab.

Basically, Amplitude Modulation (whether it is DSB-SC or AM with a carrier component) is the process of varying the carrier signal in accordance with the message signal. The AM receiver can use coherent or non-coherent detection. In the first case, the receiver is able to extract a synchronized local version of the transmitted carrier (by using carrier synchronization techniques such as a Costas loop for example, or through other methods).

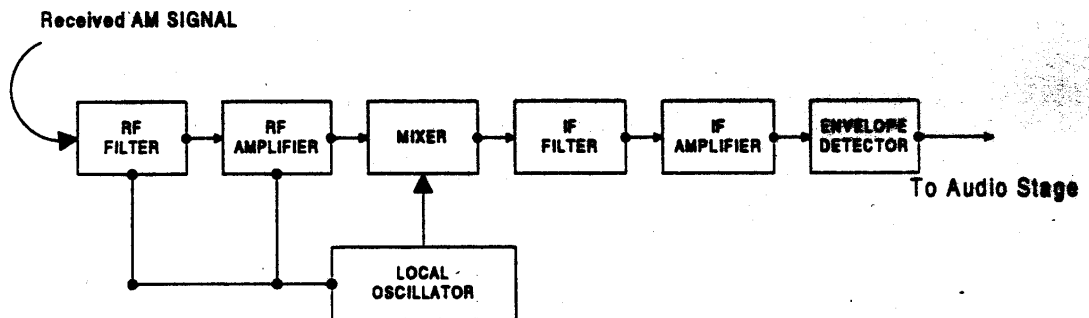
In the case of this experiment, you will be “artificially” testing coherent detection by feeding the same carrier signal used at the transmitter directly to the receiver.

In the second case of non-coherent detection, the receiver doesn't have access to a synchronized local replica of the carrier, so other techniques are necessary. For example, AM (in contrast with DSB-SC) is specifically devised to allow non-coherent detection by transmitting a pure un-modulated carrier component in addition to the one modulated by the message signal. This mechanism allows the receiver to perform simple non-coherent envelope detection (as you studied in class). Notice that it is also possible to detect AM coherently (just like DSB-SC!).

Because of the simple, low-cost receiver architecture for non-coherent AM detection, it is widely used in commercial systems such as radio broadcasting. In this experiment, you will be studying a basic example of a non-coherent AM receiver (known as the super-heterodyne receiver) as illustrated in the block diagram below.

In this radio architecture, the RF stage and local oscillator are simultaneously adjusted to select the desired station, and the RF stage amplifies the AM signal. By adjusting the band-pass RF filter on the ACB board, you can select a frequency range that covers the desired AM signal. The RF amplifier can also be tuned by adjusting the variable inductors to maximize its gain. The following Mixer stage converts the AM frequency to an IF (Intermediate Frequency) of 455 kHz. It is this mixing operation that gives the name of heterodyne receiver to this architecture (see your textbooks for more details). The fixed IF frequency allows the AM receiver to be highly optimized regardless of the selected RF frequency. The IF filter stage is a high-performance ceramic band-pass filter with 20 kHz bandwidth from 445 kHz to 465 kHz. This filter essentially performs the selectivity required to tune to a certain station. The final envelope detector stage demodulates the 455 kHz IF signal to recover the original message signal (which, in the case of voice, can be applied to the audio stage and converted to sound by the Speaker).

**Figure 1: Super-heterodyne AM Receiver**



## Lab Work

### 1) Part 1: DSB-SC Modulation Set-up

- a. Adjust the VCO-LO circuit to produce a 1000 KHz sinusoidal signal, and connect it to Terminal C (carrier) of the Modulator block in the transmitter. This signal will be the carrier signal.
- b. Use the signal generator to obtain a sinusoidal signal of frequency 10 KHz and amplitude 50 mV peak-to-peak. Use this signal as a message signal and connect it to Terminal M of the modulator block.
- c. Set Switch S1 to "ON" (so that a DSB waveform is generated). S2 and S3 should be OFF. Then observe and sketch the modulated signal on the oscilloscope.

### 2) Part 2: DSB-SC Demodulation

- a. Connect the modulated signal directly to the message input M of the Product Detector block (i.e., bypass the RF and IF stages of the receiver), and connect the same carrier to the carrier-input of the Product detector. Make sure the input signal level is 100mV.
- b. Compare the demodulated signal (output of the audio filter) and the message signal. Sketch both waveforms.
- c. Try increasing the frequency of the message signal (up from 10 KHz), and observe at which frequency does the demodulated signal disappear. Explain your observations.

### 3) Part 3: DSB-SC with Triangular Signal

- a. Now, change the message signal to a triangular waveform of frequency 10 KHz. Observe if this signal is successfully recovered at the demodulator. Explain.
- b. Reduce the frequency of the triangular waveform to 2 KHz. Is there any improvement in the demodulated signal? Explain why?

### 4) Part 4: Effect of Frequency Mismatch

- a. Now, you need to explore the effect of frequency mismatch. First, start by disconnecting VCO-LO from the Product Detector, and connect instead a signal from the external function generator (with frequency 1000 KHz). Observe the output of the audio amplifier. Can you recover the message signal? Explain why?
- b. Try varying the frequency of the oscillator finely around 1000 KHz, and observe what happens.

### 5) Part 5: AM Setup

- a. Re-connect VCO-LO (at 1000 KHz) to the carrier terminal C of the Modulator, and adjust its amplitude to 1V peak-to-peak. Then apply a sinusoidal message signal of frequency 10 KHz and 0.1 V peak-to-peak to Terminal M of the modulator.
- b. Turn Switch S1 and Switch S2 OFF, and set Switch S3 to ON.



- c. Adjust the modulator potentiometer knob so that the AM modulation index  $\mu$  is about 50%. Sketch the modulated waveform.
  - d. Try setting the modulation index to 100%, and then over 100%. Sketch the signals. Discuss your observations.
- 6) **Part 6: AM super-heterodyne receiver**
- a. Connect the following in series: Modulator Output-Resistor R1-RF Power Amplifier-Antenna Matching Network with 3 two-post connectors.
  - b. Using a two-post connector, connect the modulator output to the  $1\text{ M}\Omega$  R8 resistor from the Receiver Block (to simulate transmission losses), and then adjust the L4 and L5 inductors to maximize the RF amplifier output signal.
 

**Important Notes:** 1) the L4&L5 inductors are quite fragile and often get damaged (by the students!!!). Care should be exercised while tuning them very slightly with the special tools provided.

2) If these inductors are damaged, or can not be tuned properly, connect the antenna matching network output directly to the Mixer input M of the receiver, and go to step c)
  - c. Connect the output of the 1455 KHz VCO-HI to the carrier input C of the Receiver Mixer, and set the potentiometer knob on the VCO-HI fully clockwise. Also connect the RF received signal to the Receiver Mixer input M.
  - d. While observing the IF filter output, adjust the mixer potentiometer knob and the positive supply knob on the base unit to obtain the maximum signal at the filter output.
  - e. Compare the signal at the output of the envelope detector to the original message signal. Sketch both waveforms.
  - f. Now, adjust the modulator potentiometer to obtain a modulation index of 100%, and repeat the above.

## Homework Questions

- Q1.** In Part 2 above, explain why the received signal disappears at some point when the transmitter message frequency exceeds a certain threshold
- Q2.** In Part 3, explain why there is an improvement when the message frequency is 1 KHz.
- Q3.** In Part 4, analyze mathematically the effect of frequency mismatch between the transmitter and receiver carrier signals, and show that the demodulated baseband signal appears as a modulated signal.

## Frequency Modulation (FM)

### MATLAB Simulation

#### Objectives

The main objectives of this experiment are:

- 1) To gain a good understanding of Frequency Modulation.
- 2) To learn how to implement FM modulation & demodulation in software.
- 3) To demodulate an FM signal using the Differentiation (or Slope) technique.

#### Pre-Lab Work

- 1) Integration can be used to find the area under a given curve defined by some function  $f(u)$ . Our concern in this experiment is to learn how to numerically integrate a function  $f(u)$  up to some point  $t$  (i.e. to get a function of  $t$ , say  $m(t)$ ). To do so consider the following formula ( $i$ : is the index in MATLAB):

$$m = \frac{1}{\text{length}(f)} \sum_{i=1}^{\text{length}(f)} f(i)$$

Generate  $\mathbf{f} = \cos(2*\pi*t)$  for  $\mathbf{t} = 0:0.001:1$ . Find  $m$ ? In order to speed up the time of computation, avoid using **for**-loops. Think of the functions: **sum** or **cumsum**. You also need to plot  $\mathbf{f}$  vs.  $\mathbf{t}$  and  $\mathbf{m}$  vs.  $\mathbf{t}$ .

- 2) How to differentiate a function in MATLAB? First, generate a function  $f$  in MATLAB like the one in Part 1. Use the function **diff(f)** and store it as a vector called **df**. Now plot  $\mathbf{f}$  vs.  $\mathbf{t}$  and  $\mathbf{df}$  vs.  $\mathbf{t}$ . What will happen? Type **whos** in the MATLAB command window to see the dimensions of  $\mathbf{t}$ ,  $\mathbf{f}$ , and  $\mathbf{df}$ , what do you notice? What do you need to be able to plot  $\mathbf{df}$  vs.  $\mathbf{t}$ ? Think about the length of the  $\mathbf{t}$  and  $\mathbf{df}$  vectors. Read carefully how **diff(f)** works by typing **help diff** in the MATLAB command window.

## Overview

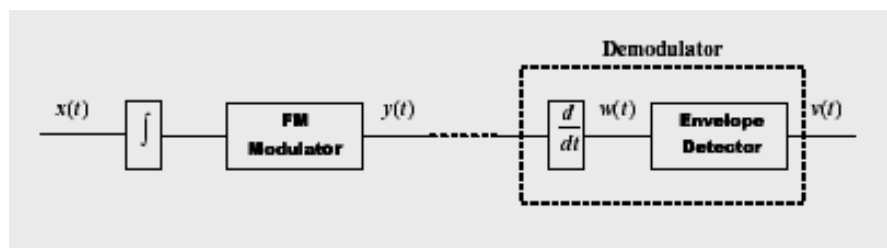
Angle modulation includes both Frequency and Phase modulation schemes (FM and PM), which are characterized by their superior performance (compared to AM) in the presence of noise at the expense of higher bandwidth requirements. As you studied in class, FM and PM are very similar. In fact, an FM signal can be interpreted as PM signal and vice-versa (refer to your textbooks). As such, our focus in this lab will be on FM modulation exclusively.

An FM-modulated signal has its instantaneous frequency that varies linearly with the amplitude the message signal. For example, a message signal  $x(t)$  causes the frequency of the FM signal  $y(t)$  to vary linearly around a central carrier frequency  $f_c$ . The following formula describes this relationship:

$$y(t) = A_c \cos(2\pi f_c t + k_f \int_0^t x(\tau) d\tau)$$

Here,  $k_f$  is known as the sensitivity factor, and represents the frequency deviation rate as a result of message amplitude change. In practice, FM modulation is implemented by controlling the instantaneous frequency of a voltage-controlled oscillator (VCO). The amplitude of the input signal voltage controls the oscillation frequency of the VCO output signal.

FM signals can be demodulated using different techniques (refer to your textbook for details). Our focus in this experiment will be on the Slope Method, which uses a cascaded differentiator with an envelope detector circuit as illustrated in the diagram below. The differentiator basically produces an AM-like signal that is then demodulated by the envelope detector block. Refer to your textbooks for the mathematical details of this method.



## Lab Work

- 1) Implement the FM Modulator function for the message signal  $x(t) = \sin(2\pi t)$ . Use a carrier frequency of  $f_c = 100$  Hz and a frequency sensitivity factor  $k_f = 160\pi$ .
- 2) Plot  $x(t)$  and  $y(t)$ . Also, plot the magnitude spectrum for  $y(t)$ .
- 3) Next, implement the FM demodulator part.
- 4) Plot  $w(t)$ , and  $v(t)$  both in Figure 3. You need to present only one period of the message signal with  $t_s = 1/(10f_c)$ .

**Useful MATLAB Functions:** `fftshift(fft( ))`, `cumsum`, `diff`, `plot`, `subplot`, `figure`, `xlabel`, `ylabel`, `title`.

- 5) Repeat the above for a sinc message signal  $x(t) = \text{sinc}(100\pi t)$  with  $f_c = 250$  Hz and  $k_f = 200\pi$ , and where  $t = -0.1:ts:0.1$ .

What is the bandwidth of  $x(t)$  and  $y(t)$ ?

## Homework Questions

**Q1.** What is the effect of changing the value of the sensitivity factor  $k_f$ ?

**Q2.** List and compare all the other methods you know about for demodulating an FM signal?

---

# Frequency Modulation Using the ACB

## Hardware Experimentation

### Objectives

The main objectives of this experiment are:

- 1) To build on the introduction to FM in the previous lab (based on Matlab Simulation), and to gain a clearer understanding of its concepts using hardware experimentation with the ACB.
- 2) To perform FM bandwidth computations for different message waveforms.
- 3) To experiment with another common demodulation scheme for FM, known as Quadrature Detection.

### Pre-Lab Work

- 1) Consider the case of FM modulation with a sinusoidal message signal (tone modulation). Give a simple validation of Carson's formula for bandwidth estimation based on the use of mathematical derivations involving *Bessel Functions* (refer to your textbook for relevant details).
- 2) As a preparation for the bandwidth computation in this lab, review Fourier series techniques for rectangular and triangular signals. Give the Fourier series for these signals assuming some signal period  $T$ .

### Overview

Refer to your textbook and to the previous lab for a general overview and mathematical expressions of FM-modulated signals. The bulk of this hardware experiment will be based on the ACB board, where we use the VCO-LO for FM signal generation, and the Quadrature Detector for FM signal demodulation.

Part 1 of the work in this lab consists of FM bandwidth evaluation. A useful expression for the computation of FM bandwidth is given by Carson's rule:

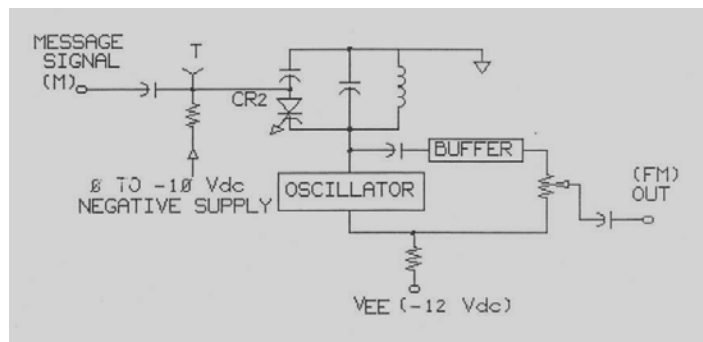
$$B_T = 2(1 + \beta)W$$

where  $W$  is the message signal bandwidth, while the modulation index is given by

$$\beta = \frac{k_f \max |x(t)|}{W}$$

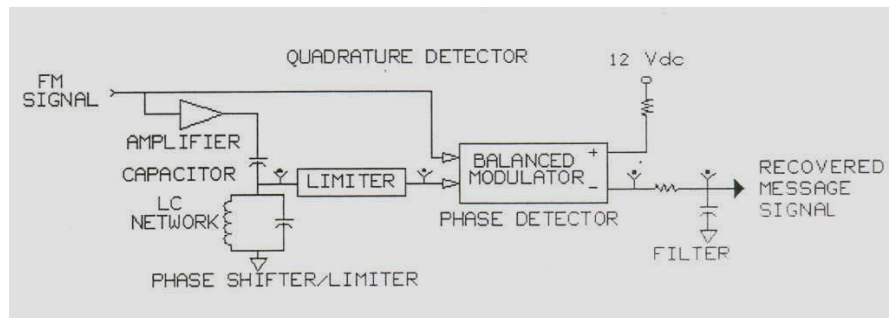
In this experiment, the FM modulator is part of the VCO-LO block. As illustrated below, the circuit essentially uses the input voltage to change the capacitance of a varactor diode that modifies the resonance frequency of an LC circuit, which in turn tunes the oscillation frequency of the VCO. You can browse through the WinFACET lab overview for more details on the circuit diagram and operation details.

Figure 1: VCO Circuit Diagram



On the other hand, the demodulator uses a circuit known as the Quadrature Detector. This is a type of FM discriminators that converts the FM carrier frequency deviation into amplitude and frequency of the original message signal. The quadrature detector circuit block on the ACB includes a Phase Shifter/Limiter, a Phase Detector, and a Filter. Its operation is described as follows. At the detector input, the received FM signal is split into two paths, with one going into the phase shifter/limiter which introduces a phase shift into the signal by  $\sim 90$  degrees. The shifted signal is then clipped by the Limiter, and combined with the original FM signal. The resulting sum is applied to a balanced modulator which will work as a Phase Detector. This phase detector will produce two signals: one signal at double the carrier frequency, and another baseband signal that varies with the phase difference between the two inputs, or equivalently, with the FM signal frequency deviation. The final low-pass Filter block removes the double frequency term and output the reconstructed message signal. Refer to WinFACET software to get more details on the circuit diagram for the Quadrature Detector block.

Figure 2: FM Quadrature Detector



## Lab Work

### 1) FM Modulation:

- For signal generation, you will be using the VCO-LO block. First, use a 2-post connector to connect the 452 kHz terminals. Notice that this produces an un-modulated carrier signal at FM Out (which may not be exactly at 452 kHz).

Note: Use the oscilloscope to tune the carrier frequency around 500 kHz by adjusting the negative supply knob. This will be useful to get a good display later on.

- Select a square waveform from the external function generator, and set it to a 1 kHz frequency and 4V pk-pk amplitude. Apply this signal to terminal M of the VCO-LO block.
- Observe the modulated signal from FM Out on the oscilloscope, and sketch it. Explain intuitively why you are getting two sinusoids on the screen.
- Measure the periods and frequencies of these sinusoids. Then, estimate the FM sensitivity factor  $K_f$  (refer to the previous lab or your textbooks for exact definitions) from your measurements.
- Try changing the amplitude of the input message signal. Does the change in the FM output make sense? Explain.
- Switch now to a sinusoidal (or triangular) input signal, and observe the FM signal on the oscilloscope. Sketch and comment.

### 2) FM Bandwidth:

- In practice, the spectral properties of modulated signals are normally observed on a spectrum analyzer (which you don't have access to at present). In this part, you will resort to theoretical expressions (supported by lab measurements) to estimate FM signal bandwidth.
- For the 1 kHz sinusoidal signal used in Part 1, what is the bandwidth of the message signal, and what is the maximum frequency deviation of the FM modulated signal? You will need to use the  $K_f$  factor measured in Part 1,
- Use Carson's Rule and the above results to estimate the bandwidth of the FM signal modulated by the 1 kHz sinusoidal message signal.

### 3) FM Demodulation:

- In this part, you will be using the Quadrature Detector block on the ACB. This is another type of FM demodulators (in addition to the slope detector, PLL detector, etc, that you studied in class).
- Start WinFACET software, and go to Exercise2: Demodulation. Go through the discussion to understand the main concepts of Quadrature Detection for FM.

- Connect the FM Out signal from VCO-LO to the FM input of the Quadrature Detector block. Go through the different steps of the procedure up to Step 35 (Slide 26). Report your observations and answers to questions.
- In addition, change the signal waveform shape (from sinusoidal to rectangular and triangular) to see its impact. Comment on the results. Change also the message signal amplitude and frequency and study the impact as well.

## Homework Questions

- Q1.** In Part 1 above, specify the expression for the instantaneous frequency of the modulated signal.
- Q2.** Consider two message signals,  $m_1(t) = A \cos(2 \pi f_m t)$  and  $m_2(t) = A \cos(4 \pi f_m t)$ , with the same amplitude but  $m_2$  having double the frequency. Would the two modulated FM signals (by  $m_1$  and  $m_2$ ) have different frequency deviations? What about their bandwidth? Explain.
- Q3.** Re-do the bandwidth computation of Part 2 with the triangular and rectangular signals. Note: you need to use Fourier techniques to estimate the bandwidth of the message signals (you can truncate to a few harmonics as appropriate).



## Experiment 7

# Sampling & Quantization

## MATLAB Simulation

### Objectives

The main objectives of this experiment are:

- 1) To gain a clearer understanding of sampling theory and uniform quantization.
- 2) To learn how to simulate these operations using MATLAB.

### Pre-Lab Work

- Use the MATLAB function **pulstran** to generate and plot three periodic pulse trains with different shapes each with three different pulse durations. Read the help of that function and try to take advantage of the example given is at the end of the help for **pulstran**. However, do not copy it!
- Given a vector  $x=[1.1,2.6,-3.5,-0.001,0.768]$ , apply the MATLAB functions **fix**, **round**, **floor**, & **ceil** on  $x$ . State the differences between the resulting vectors.

### Overview

An analog signal is characterized by the fact that its attributes (like: amplitude, frequency and phase) can take any value over a continuous range. On the other hand, digital signals can take only discrete and finite values. One can convert an analog signal to a digital signal by sampling and quantizing (collectively called analog-to-digital conversion, or ADC).

It is typically more efficient to process the resulting discrete signals by digital signal processors. The processed signals are then converted back into analog signals using a reconstruction or interpolation operation (called digital-to-analog conversion, or DAC).

### Sampling:

To sample a continuous-time signal  $x(t)$  is to represent  $x(t)$  at a discrete number of points,  $t = nT_s$ , where  $T_s$  is the sampling period. The sampling theorem states that a band-limited signal  $x(t)$  with a bandwidth  $W$  can be reconstructed from its sample values  $x(n) = x(nT_s)$  if

the sampling frequency  $f_s = 1/T_s$  is greater than twice the bandwidth  $W$  of  $x(t)$ . Otherwise, aliasing would result in  $x(t)$ . The minimum sampling rate of  $2f_s$  for an analog band-limited signal is called the Nyquist rate.

### Quantization :

In order to process the sampled signal digitally, the sample values have to be quantized to a finite number of levels, and each value can then be represented by a string of bits. For example, if the signal is quantized to  $N$  different levels, then  $\log_2(N)$  bits per sample are required.

Notice that to quantize a sample value is to round it to the nearest point among a finite set of permissible values. Therefore, a distortion will inevitably occur. This is called quantization noise (or error).

Quantization can be classified as uniform and non-uniform. In the case of uniform quantization, the quantization regions are chosen to have equal length. However, in non-uniform quantization, regions of various lengths are allowed. Non-uniform quantization can be implemented through compression-expansion (or companding) of the signal, and this is commonly used (as in telephony) to maintain a uniform signal-to-quantization noise ratio over the full dynamic range of the signal (refer to your textbook for more details).

### Lab Work

- 1) Generate a time vector  $\mathbf{t}$  from -0.5 to 0.5 with a step size of 0.001. Implement the function  $x(t) = e^{-|t|/\tau}$  where  $\tau = 0.1$ . In order to simulate an A/D converter, perform the following tasks:
  - a. Plot  $\mathbf{x}$  and its magnitude spectrum in a two-panel figure window. What is the bandwidth of  $\mathbf{x}$ ?
  - b. Set the sampling frequency to twice the bandwidth of the signal  $\mathbf{x}$  (which is approximately 25Hz). Generate a rectangular pulse train starting at -2 to 2 where the step size is  $1/(\text{sampling frequency})$  and with a duration  $1\text{E-}10$ . In a two-panel figure window, plot the pulse train and its magnitude spectrum. What is the relation between the time and frequency domain representation of the pulse train?
  - c. Sample  $\mathbf{x}$  using the pulse train and plot the resulting sampled version of  $\mathbf{x}$ , (say  $\mathbf{xs}$ ). Also, plot its magnitude spectrum. What can you observe from both plots?

Now, to be able to simulate a D/A converter, perform the following tasks:

- d. Let  $\mathbf{h}$  be a vector of forty ones (40). Plot its magnitude spectrum. In order to correctly plot  $H(\mathbf{f})$ , use the `fft` function with a number of FFT points equal to the length of  $\mathbf{x}$ .

e. Use the function **filter** to form the reconstructed signal (say **xd**) from **xs** using the filter **h**, i.e, **xd=filter(h,1,xs)**. In a 4-panel figure window, show the plots of **x**, **xd**, and their magnitude spectrums. Comment on your findings.

**Useful MATLAB functions:** pulstran

- 2) Repeat 1 using a sampling frequency equal to 1.5 times the bandwidth of  $x(t)$ .
- 3) Repeat 1 using a sampling frequency equal to 10 times the bandwidth of  $x(t)$ .
- 4) Generate a sine signal (call it **x**) with frequency equal to  $1/2\pi$  over a time interval of three periods. The step size should be equal to 1E-4. The quantized signal will be equal to  $x_q = \Delta x \text{round}(x/\Delta)$ , where  $\Delta$  is the quantization step size. You can compute it using the formula  $\Delta = 2 \max(|x|)/L$  where  $L = 2^b$  and  $b$  is the number of quantization bits. Use 2 bits to quantize **x** and show the following:
  - a. In one figure plot **x** vs. time and the quantized signal vs. time.
  - b. Compute the Signal-to-Quantization Noise Ratio (SQNR) defined by the formula:  $SQNR = 10 \log_{10}(\sum x^2 / \sum (x - x_q)^2)$  in dB.
- 5) Repeat Part 4 using 3, 4 and 5 bits per sample.
- 6) Plot the SQNR vs. the number of bits per sample for 2, 3, 4, and 5 bits. What is your conclusion?

**Additional Questions:**

- Q1.** Can you process real-world analog signal by digital computers? List possible applications where sampling and quantization are necessary.
- Q2.** From the above results, what can you conclude about the improvement in SQNR with every additional quantization bit?

---

# Pulse Amplitude Modulation

## Hardware Experimentation

### Objectives

The main objectives of this lab are:

- 1) To gain a practical understanding of the concepts of analog signal sampling, their usefulness and limitations.
- 2) To learn about Pulse Amplitude Modulation (PAM) as a transmission scheme.
- 3) To experiment with the realization of PAM modulation & demodulation using the DCB.

### Pre-Lab Work

- 1) Explain the main result of the Sampling Theorem, and give a simple graphical proof for it (there is no need for full analytical details).
- 2) Discuss why is the sampling of analog signals quite useful in many applications?

### Overview

An analog message signal, representing voice for example, has continuous amplitude and frequency values that vary with time. Analog communication systems transmit the complete analog waveform. But instead of transmitting the analog signal, it is possible to transmit discrete pulses (or samples) that represent some parameter of the message signal's waveform at regular intervals in time.

In the case of pulse amplitude modulation (PAM), the amplitude of the analog waveform is sampled at discrete (i.e., discontinuous) time instants, and transmitted as a sequence of pulses. Notice, however, that this sampling produces amplitudes that can still take *any value*, hence the scheme is not fully digital (unlike PCM which will be studied in a subsequent lab, and which uses quantization to represent the discrete samples by a finite number of bits).

As long as the sampling of the analog signal is taken with a sufficiently high frequency (higher than the minimum Nyquist rate of twice the signal largest frequency), it can be shown that there is *no loss* in information as a result of taking discrete samples. In fact, the PAM receiver

simply passes the received samples through a low-pass interpolating filter to fully recover the analog waveform. Notice also that the results of the sampling theorem, which are derived for the case of ideal sampling with Dirac delta functions, are equally applicable for practical sampling with pulses that have a finite duration (refer to your textbook for more details).

Different types of sampling can be distinguished. With “natural” PAM signals, the amplitude of each pulse follows the amplitude of the message signal for the duration of the pulse. Another type of sampling results in “flat-top” PAM signals where a sample&hold circuit holds the sampled amplitude at a constant level between the sampling times, thus resulting in a stair-case PAM signal shape. This is sometimes used at the receiver end prior to low-pass filtering since it helps increase the amplitude of the reconstructed signal.

There are a number of advantages for transmitting PAM pulses rather than complete analog signals. For example, if the duration of the PAM pulse is small, the energy required to transmit the pulses is much less than the energy required to transmit the full analog signal. For a sampling pulse train with a duty cycle fraction  $PW/T$  (i.e., the sampling pulse duration is  $PW$ , and its period is  $T$ ), it can be shown that the power of the sampled PAM signal  $P_p$  is only a fraction of the total analog signal power  $P_s$ , given by:  $P_p = (PW/T) \times P_s$ .

Another advantage of PAM has to do with the ability to multiplex (or combine) several different signals and transmit them on the same communication channel. Although this is also possible with analog communication (using for example frequency division multiplexing-FDM), it is much simpler and more economical to implement with digital or discrete systems like PAM using what's known as time division multiplexing (TDM). Since PAM sends amplitude pulses of a given signal at discrete periodic time slots (or intervals), it is then possible to assign the remaining time slots for other signals, thereby maximizing the utilization of the channel. There is obviously a need for maintaining adequate synchronization at the transmitter and receiver levels to distinguish the different signals. This type of TDM transmission is very efficient and widely used in practical communication systems such as telephone networks, particularly in combination with Pulse Code Modulation-PCM (which is similar to PAM, but uses additional amplitude quantization as will be seen in a following lab).

## Lab Work

### 2) PAM Signal Generation

- Use a two-post connector to apply the M2 message signal to the SAMPLER input. Measure the amplitude and frequency of this signal on the oscilloscope.
- Check the sampling pulse frequency on port SP of the DCB using the oscilloscope (it should be 8KHz). Is this sampling rate sufficient for the given input signal?
- Apply the input analog signal and resulting PAM signal from the SAMPLER output to channels 1 & 2 of the oscilloscope. Observe and sketch both signals. Is the reconstruction quality good?
- What type of sampling is being used: natural sampling, or sample & hold?

### 3) Power Calculation

- Use the oscilloscope to measure the pulse width  $PW$  of the PAM signal, and the period of the sampling pulse train  $T_p$ .

- Calculate the pulse duty cycle  $PW/T_p$ .
- Calculate the power of the analog sinusoidal signal and the discrete PAM signal across a hypothetical  $1\text{ k}\Omega$  resistor. Comment on the efficiency of the PAM scheme.

4) **PAM Signal Demodulation**

- First, you need to measure the receiver filter characteristics. Use the external signal generator to apply sinusoidal signals with  $2\text{V}$  pk-pk and increasing frequency to obtain a rough sketch of the filter frequency response.

**Hint:** The filter 3dB cutoff frequency should be around  $2.6\text{ kHz}$ .

- Disconnect the M2 input, and use the external function generator to apply a new sinusoidal input signal to the SAMPLER. Set the frequency to  $2\text{ kHz}$  and the amplitude to  $4\text{V}$  pk-pk.
- Now, use a 2-post connector to apply the sampled PAM signal at the output of the SAMPLER to the FILTER input. Observe the reconstructed signal at the FILTER output, and sketch it. Comment on the quality of the reconstructed signal.
- Let the input signal frequency now be changed in the range  $3$  to  $3.5\text{ kHz}$ . Is the Nyquist criterion still met (given that SP is  $8\text{kHz}$ )? Observe the demodulated signal at the FILTER output. Explain why it is still strongly attenuated (although the sampling frequency is sufficient!)

5) **Effect of Aliasing**

- When the sampling rate is  $8\text{kHz}$ , what is the theoretical input frequency at which aliasing starts to appear?
- Select an input message signal frequency of  $5\text{kHz}$ . Apply the original analog input and PAM demodulated signals on both channels of the oscilloscope. Sketch the resulting demodulated waveforms and comment.
- Sketch a frequency-domain description of the problem that you just observed in time-domain on the oscilloscope. Notice that it possible to observe things in frequency-domain too (using spectrum analyzers, which are currently not available in sufficient numbers for the EE370 lab requirements).

## Additional Questions

- Q1.** Explain why it is sufficient to use a low-pass filter at the receiver to recover the original transmitted analog signal? Hint: you can assume that sampling is ideal (i.e., with delta pulses), then use frequency-domain diagrams to show the effect of sampling on the signal spectrum.
- Q2.** Consider the sampling of a signal which has some maximum frequency  $f_m$ , but that may be corrupted by high-frequency noise. What should you do first? Explain why.

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# Pulse Code Modulation and Time Division Multiplexing

## Hardware Experimentation

### Objectives

PCM and time-division multiplexing are widely used in modern telecommunication networks. The main objectives of this experiment are to use to DCB board in order to:

1. Gain a good understanding and hands-on experience with Pulse Code Modulation
2. Learn about non-uniform quantization and its applications (e.g., in telephony)
3. Understand and experiment with the concept of Time Division Multiplexing.

### Pre-Lab Work

1. Read the relevant material in your textbook.
2. Describe the use of PCM and TDM in modern telephony: give the basic architecture of modern telephony networks from the user phone to the central office switch, discuss T1/E1 multiplexing, etc.
3. Discuss and compare the advantages & disadvantages of TDM vs. FDM (frequency-division multiplexing).

### Overview

Pulse-code Modulation (PCM), like PAM, is a digital communication technique that sends samples of the analog signal taken at a sufficiently high rate (higher than the Nyquist rate). In addition, PCM differs than PAM in that it quantizes the samples by constraining them to only take a limited number of values, and then converts each value into a binary string of bits that are transmitted on the communication line. Typically, in digital telephony where PCM is widely used, the sampling rate is 8 kHz (higher than twice the voice band), and the quantization uses 256 levels (i.e., each sample is mapped into an 8-bit PCM code).

In practice, PCM is typically combined with Time Division Multiplexing (TDM), which is the process of combining many PCM signals representing different messages and transmitting them over the same channel on a time-sharing basis. Each PCM signal is assigned a time period called a slot on the transmission line, and slots are arranged in groups called frames. The main advantages of PCM transmission are: lower cost, ease of multiplexing and switching, and better noise immunity. Its main disadvantage is the stringent timing and synchronization requirements. Nowadays, PCM-TDM systems form the backbone for all digital telephony networks worldwide (refer to your textbooks for more details).

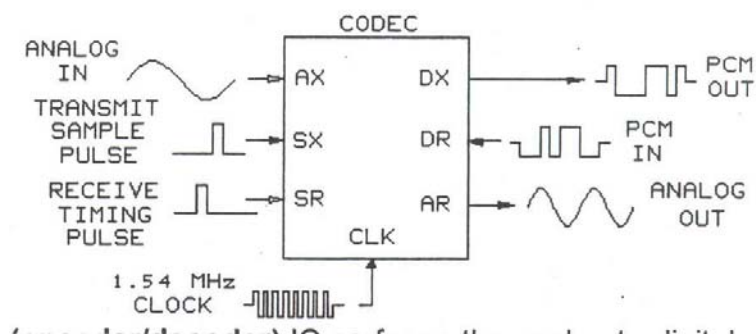
In this experiment, you will be using the PCM block in the DCB board, which is centered on a pair of voiceband CODECs (a codec is a Coder-Decoder) for PCM transmission and reception. These CODECs are IC (integrated circuit) chips that incorporate the major PCM-TDM functionalities. The basic operation of the CODEC is described below.

First, the analog signal at the AX input port is converted into a digital PCM signal at the DX output port. There is a timing signal SX giving an 8 kHz transmit pulse that occurs one clock cycle before the assigned time slot. Each pulse enables the codec to sample/encode the message signal to be transmitted in the following time slot.

At the receiver codec, the PCM signal at the DR port is recovered into an analog signal at the AR port. Similar to the transmitter codec, the receiver has an 8 kHz clock signal SR that enables the operation of the codec. In order for the receiver to decode a transmitted PCM code in a given time slot, the SX and SR pulses must be synchronized.

The transmission rate between codecs, i.e., the PCM line rate, is governed by a 1.54 MHz clock (typically referred to as T1 rate). Notice that this is much faster than the bit stream generated from a single user (which is  $8\text{bits} \times 8\text{ kHz}$ , or 64 kbps). This is because the PCM line actually carries a number of PCM signals (a T1 line corresponds to 24 PCM signals) in a TDM fashion, as explained above, and therefore needs to run at a much higher speed than a single 64 kbps PCM stream.

The actual architecture of the codecs used in the DCB board comprises many circuit blocks. On the transmitter (encoder) side, there is a voice-band anti-aliasing filter (limited to [0.2 kHz, 3.5 kHz]), followed by an 8 kHz sample & hold circuit, an ADC (analog-to-digital converter) that implements 8-bit quantization and compressor ( $\mu$ -law or A-law), and finally a parallel-to-serial converter and output register. On the receiver (decoder) path, there is an input register with serial to parallel conversion, a sample & hold circuit, an expander (to remove the transmitter compression), and a receiver interpolation filter.





### 1) Part 1: PCM Signal Modulation & Demodulation

- Use the external function generator to apply a sinusoidal signal with frequency 1 kHz and amplitude 2 V peak-to-peak to Input AX of CODEC 1.
- Connect CODEC 1 to CODEC 2 with a two-post connector. CODEC 1 is the transmitter, and CODEC 2 is the receiver.
- Use the oscilloscope to simultaneously observe the AX signal of CODEC 1 and the AR signal of CODEC 2. Is the demodulation working properly?
- Observe the signal DX of CODEC 1 (or similarly, DR of CODEC 2) on the oscilloscope. What does this signal correspond to?
- If you keep increasing the amplitude of the input AX signal, what type of distortion do you get? What is the reason for that?
- Now, try to change the input frequency for AX. If the frequency is outside the range [0.2kHz, 3.5kHz], what happens to the received AR signal? What is the reason for that?

### 2) Part 2: PCM Timing Signals

- Disconnect the external function generator, and use a 2-post connector to apply the signal M1 to Input AX of CODEC 1. Record its frequency and amplitude.
- Observe the input sampling signal SX of CODEC 1 on the oscilloscope. Measure its period and frequency. What is the role of this SX signal?
- Check the SR signal of CODEC 2 on the oscilloscope. Is it in synch with the SX signal? What is the reason for that (i.e., will PCM decoding work if not)?
- Observe the relative timings of the signal SX and DX of CODEC 1 (same for RX and DX of CODEC 2). Apply these signals to channels 1&2 of the oscilloscope, and record your observations. When does DX start relative to SX? Explain why?
- Now, observe the characteristics of the DX signal (or RX) on the oscilloscope (which corresponds to PCM codes). How many bits per PCM code are there? What is the duration of one PCM code? Try displaying a few different code words on the oscilloscope. Are they having the same string of bits? Why?
- Using the oscilloscope, try to measure the PCM stream bit rate. Compare this with the theoretical bit rate you expect to see.

### 3) Part 3: Companding and Non-Uniform Quantization

- Launch the WinFACET software, and go through the material covering the overview and description of  $\mu$ -law and A-law companding.
- Explain the meaning of non-uniform quantization. What is its main advantage?

- What is the difference between  $\mu$ -law and A-law companding?

#### **4) Part 4: Time-division Multiplexing (optional part)**

- Launch WinFACET and go to Exercise 2 (Time Division Multiplexing).
- Go over the overview, then start the procedure, and execute all steps up to Step 26, and report your observations and answers to questions.

### Additional Questions

**Q1.** How is PCM different than PAM?

**Q2.** Does a digital PCM stream of bits representing analog voice consume more or less bandwidth than the original analog signal?

**Q3.** What is the main advantage of PCM-TDM?

**Q4.** What is the difference between simplex and full-duplex communication links?

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# Communication Channel Effects

## Hardware Experimentation

### Objectives

The main objectives of this experiment are:

1. To obtain a better understanding of the impact of channel bandwidth limitation on the performance of communication systems.
2. To learn how to measure channel bandwidth.
3. To experiment with channel distortion impact on PAM and PCM transmission using the DCB platform.

### Pre-Lab Work

You should read the relevant material in your textbook, and answer the following questions:

1. By referring back to basic Fourier theory results, explain why the time duration of a pulse signal and its bandwidth are inversely proportional, i.e., if time duration is large, bandwidth is small, and vice-versa.
2. What is Inter-symbol Interference (ISI)? Is it caused by channel bandwidth limitation, external noise interference, or other factors? Explain.

### Overview

A typical communication system consists of 3 main parts: the transmitter, the receiver, and the communication channel. This channel is a physical propagation medium used to carry signals between the transmitter and the receiver. For example, a propagation channel can be a twisted copper pair, an optical fiber, a coaxial cable, or a wireless RF link.

The propagation channel has many characteristics that can introduce distortion and cause a severe limitation to the proper reception of the transmitted signal. Some of these limitations include:

- Bandwidth limitation: where, for instance, a channel with poor high frequency characteristics (i.e., similar to a low-pass filter) will affect a digital pulse rise and fall times. Thus, limited bandwidth increases the time a pulse needs to attain the full amplitude required by the receiver. Also, if the channel has poor low-frequency characteristics, it will affect the top and bottom of the transmitted pulses, which no longer maintain a stable voltage, and become more difficult to detect.
- Additive noise: coming in the form of undesirable, random-like interference that corrupts the transmitted signal, and affects the ability of the receiver to detect it properly.
- Timing variations: also known as timing jitter, which is due to random variations in the channel time delay. As a result, the received signal is delayed by a varying amount of time from one signal period to the next, and become therefore difficult to track properly.
- Channel resonance: if the channel contains reactive elements, then oscillation can arise and cause amplitude ringing effects, with overshoots and undershoots than can lead to signal detection errors at the receiver.

In this experiment, we focus on bandwidth-related aspects of digital communication channels. For a digital transmitted pulse to be detected at the receiver, the significant frequency components of the pulse must pass undistorted through the channel. But since practical channels have limited bandwidth, only the frequency components inside the channel passband will be received properly. The elimination of some frequency components will change the shape of the received pulse, and may cause loss of information at the receiver.

To get a basic idea of how to assess channel bandwidth, consider for example the case where the channel can be approximated by a simple low-pass filter of 1<sup>st</sup> order (e.g., an R-C circuit). It is then found that the 3-dB pass-band of this channel (which can taken as a measure of bandwidth) is given by  $BW = 1/(2 \pi RC)$  or  $1/(2 \pi \tau)$ , where  $\tau$  is the time constant of the circuit. Recall also from electrical circuit theory that one RC time constant is required for the output voltage to reach 63% of the input voltage (when the input is a stair-case pulse).

The concept of *rise time* can be linked to signal bandwidth as well. Rise time, denoted by  $T_r$ , is the time required for a pulse to rise from 10 to 90% of its final value. It can be shown that bandwidth is given by:  $BW = 1/(\pi T_r)$ . Notice that for a pulse to reach its full amplitude at the receiver, the pulse duration (or period) must be greater than the rise time imposed by the channel. This clearly shows that the pulse rise time (or, equivalently the channel bandwidth) sets a practical limit on the minimum signal pulse duration and, therefore, the maximum permissible symbol transmission rate.

In this experiment, you will be using the DCB board with the PAM, PCM and Channel Simulator blocks. The Channel Simulator is simply a configurable filter that emulates the response of a real communication channel. Additionally, the DCB channel block is also capable of including additive noise, but this is outside our scope at this time.

## Lab Work

### 1) Part 1: PAM Transmission through a band-limited channel

- Launch WinFACET software and select the “Channel Effects” experiment under Digital Communications 1.
- Go through the overview section, then start Exercise 1: Channel Bandwidth. Notice that this lab does not cover Exercise 2 (with noise effects).

**Important Note:** Do NOT follow the procedures for using the PAM-TDM block (bottom left of the DCB board) to connect this experiment setup. Instead, use the PAM block (top left of the DCB). Make sure you observe the output PAM signal from the channel (and not from the RX filter).

- Apply the sinusoidal input signal M2 from the PAM block to the Sampler input and follow the rest of the procedure steps up to Step 19 (on Slide 12). Notice also that you will be using the receive filter of the PAM block (and not the Filter 1 in the PAM-TDM block).
- Follow all steps, and use the software control for toggling the CM switch to change the channel simulator bandwidth. Record all your observations, and report your answers to the questions.
- As an additional question, when the channel bandwidth is not limited, does the channel still have some attenuation in its pass-band (i.e., within 80 kHz)? Give measurements and explain whether this is considered as distortion or not?

### 2) Part 2: PCM Transmission through a band-limited channel

- Disconnect the previous setup, and proceed from Step 21 (slide 14) in the Exercise 1 procedure using the PCM block of the DCB board.
- Now, use CODEC 1 as a transmitter and CODEC 2 as a receiver, with the channel simulator in between. Connect the M1 signal to Input AX of CODEC 1, the PCM output DX to the channel IN, and the Channel OUT to DR of CODEC 2. The analog recovered signal is obtained at AR of CODEC 2.
- Follow all steps, and use the software control for toggling the CM switch to change the channel simulator bandwidth. Write your observations and answers to the questions.
- Finally, go through the five review questions after finishing the experimental procedure, and report your answers.

## Additional Questions

- Q1.** In the lab procedure you went through, can you reliably transmit the PCM signal through a channel with 80 kHz bandwidth? Why? What should be the required channel bandwidth (approximately) to allow adequate reception?
- Q2.** Can you suggest a typical method for controlling the bandwidth of a transmitted signal? Explain how. Hint: think of the Fourier transform of various pulse shapes!

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## Delta Modulation (DM)

### MATLAB Simulation & Hardware Experimentation

#### Objectives

The main objectives of this experiment are:

1. To obtain a better understanding of Delta Modulation (DM) theory.
2. To learn how to simulate this digital transmission scheme using MATLAB.
3. To experiment with DM hardware implementation using the DCB platform.

#### Pre-Lab Work

You should read the relevant material in your textbook, and answer the following questions:

1. What is Delta Modulation comparable to (AM, FM, PCM, DPCM)? Explain.
2. Explain the meaning of “Delta” in the definition of Delta Modulation.
3. What are the advantages of Delta Modulation?

#### Overview

Like PCM and DPCM, Delta Modulation (DM) is a form of digital signal encoding. DM can be considered as a 1-bit differential PCM (DPCM) system. It converts each message signal amplitude change to a 1-bit sample: Logic 1 (or +E) for an increase, or Logic 0 (or -E) for a decrease. Therefore, the transmitted DM signal is a continuous series of bits indicating an increase or decrease in the message signal's amplitude.

In a typical realization of DM, the transmitter circuit contains a digital sampler which generates the DM logic pulses. The digital sampler is a compare/hold circuit that compares the message signal's amplitude with a reconstructed amplitude from a feedback integrator. Ideally, the feedback amplitude should be a close approximation of what the message-signal amplitude was at the previous clock cycle.

The DM receiver contains another integrator (similar to the transmitter's feedback integrator) and a low-pass filter which smoothes the distorted recovered signal out of the integrator.

DM transmission introduces some noise (or distortion) into the signal. There are different sources of distortion, including quantization noise and slope overload (refer to your textbook for a more in-depth description of these problems). DM relies on a very high sampling rate (typically much more than the minimum Nyquist rate of twice the signal bandwidth) to minimize the effects of distortion and to be able to produce an accurate re-construction of the signal.

In the hardware part of this lab, you will be using the DCB which contains a DELTA circuit block. In this block you will identify a comparator and a D-type flip flop which constitute the digital sampler in the DM transmitter. The comparator output feeds the D flip-flop input which latches (i.e., holds) the input value at its Q output (giving the DM transmitted bit sequence) until the next rising edge of the clock signal. The clock frequency is adjustable by control switches SW3&4, and can take any value among 8, 16, 32 and 64 kHz.

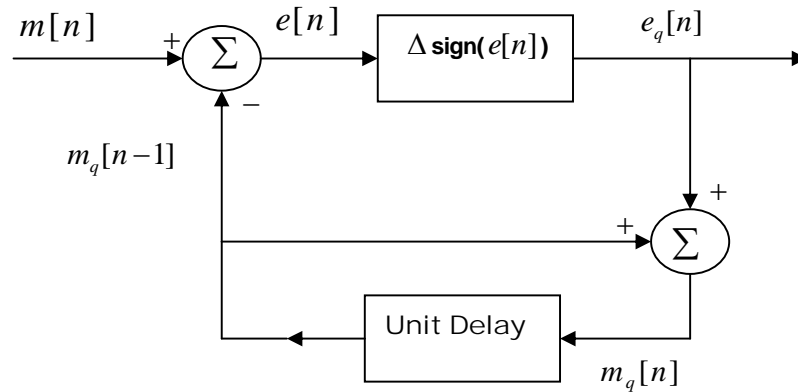
The transmitter feedback integrator is implemented by the INTEGRATE 1 circuit. The REF voltage determines how closely the feedback signal resembles the message signal. This REF value can be adjusted by the positive supply voltage potentiometer on the base unit. An RC time constant sets the positive and negative slope of the integrator output voltage. Each logic 1 DM pulse cause the integrator output voltage to increase by a fixed amount, and each logic 0 pulses causes it to decrease by a similar amount. Notice also the definition of the Baud rate for the transmitter. This simply means the number of signal changes per second. In the case of DM, this Baud rate is simply the bit rate out of the transmitter (true for binary modulation schemes only).

The DM receiver circuit is quite simple. It consists of a receive integrator and a low-pass filter. On the DCB, the receive integrator is the INTEGRATE2 circuit, and the LPF is the FILTER 1 circuit in the PAM-TDM circuit block. This filter recovers an approximate of the original message signal by smoothing out the high-frequency distortion in the integrated signal out of the INTEGRATE 2 circuit. Notice also that clock and synchronization circuits are not required in the demodulation process, which is another advantage of delta modulation.

## Lab Work

### Part I - MATLAB Simulation:

Consider the signal  $m(t) = 2e^{-t/4} \cos(2\pi f_m t)$  where  $t$  is between 0 and 20 seconds with a step size equal to 1/the sampling frequency. The sampling frequency should be large as compared to the signal bandwidth. For example, let it equal to 40 times the  $f_m$  where  $f_m = 0.4$  Hz. In order to specify the quantization step size, use the formula:  $\Delta = \max \left| \frac{dm(t)}{dt} \right| / 5$ . Now, quantize  $m(t)$  based on the following DM block diagram:



You need first to write down the equations of  $e_q[n]$  and  $m_q[n]$ . Then use a **for**-loop to create  $m_q[n]$ . That is, implement the formula:

$$m_q[n] = \sum_{i=1}^{\text{length}(m[n])} e_q[i]$$

**Note:** before you start the **for**-loop, you need to initialize the value of  $m_q[1]$  to be zero.

**I-1)** In figure 1, **plot**  $m[n]$  vs. t, then **hold** that figure and use **stairs** function to plot  $m_q[n]$  vs. t also. Finally, use **hold** again to disable that figure.

**I-2)** Generate figure 2 with two panels where the first panel plot  $e_q[n]$  vs. t using **stem** and in the second panel, **plot**  $(m[n] - m_q[n])$  vs. t. What are your observations? Indicate on the figures by using **gtext** command the types of noise you encountered in the quantized signal.

**I-3)** The above system described so far is the DM Transmitter. Using your knowledge about DM theory, propose a receiver block diagram, and specify its building blocks and how to simulate them in MATLAB.

**Useful MATLAB Functions:** sign, for, end, stairs, stem, plot, gtext, xlabel, ylabel, title.

## Part II - Hardware Experimentation:

### II.1) DM Transmission & Reception:

1. On the DELTA and PAM-TDM blocks, use three 2-post connectors to connect the M1 signal to the Digital Sampler input, the output of the Sampler to the input of INTEGRATE 2, and the output of INTEGRATE 2 to the input of FILTER 1 (in PAM-TDM). Verify the frequency & amplitude of the signal M1. Also check the frequency of the clock signal.



2. On the oscilloscope, observe and comment on the analog input M1 and the output signal of FILTER 1. **Note:** you should adjust the reference voltage for INTEGRATE 1 so that the output signal from FILTER 1 is maximized and not distorted. This is done by using the positive supply knob on the base unit.
3. Observe the output of the Digital Sampler on the oscilloscope. What does it correspond to? Measure the DM baud rate (or bit rate). Why is it equal to the clock frequency?
4. Use the oscilloscope to simultaneously display the input message signal and the fed-back signal from INTEGRATE 1. Sketch both signals, and explain why INTEGRATE 1 output closely tracks the input analog signal.

## II.2) DM Performance Factors:

1. Observe the output of INTEGRATE 1 (or INTEGRATE 2) on the oscilloscope. Is it similar to the “accumulator” integrator used in the Matlab software part of the lab? Explain why they produce “saw-tooth” signals, and not stair-case signals. Hint: think about the typical response of 1<sup>st</sup> order, RC & Op-Amp integrator.
2. Delta Modulation introduces **quantization noise**. Explain the impact of the integrator’s RC time constant on this noise. Hint: think of the relation between the time constant RC, the integrator slope, and the amount of voltage increase or decrease in a given clock period (i.e., the quantization step size).

**Slope overload noise** occurs when the input signal varies too fast for the DM bits to track closely. To study this, disconnect M1 and use the external function generator to apply a sinusoidal input signal of variable frequency. Study the impact of increasing the input signal frequency. To do so, observe the DM bits at the output of the sampler, the signal from INTEGRATE 1 (compared to the analog input) and the receiver output from Filter 1 as well. **Note:** you should also keep in mind that Filter 1 is a low-pass filter with 3dB cutoff frequency around 2.6 kHz.

## Homework Questions

- Q1.** Explain the role of the receive filter (FILTER 1) in the operation of the DM receiver?
- Q2.** Discuss how the quantization step size affects quantization noise and slope overload noise and why?

## **Appendix**

### **(LABORATORY REGULATIONS AND SAFETY RULES)**

The following Regulations and Safety Rules must be observed in all concerned laboratory locations.

1. It is the duty of all concerned parties who use any electrical laboratory to take all reasonable steps to safeguard the HEALTH and SAFETY of themselves and all other users and visitors.
2. Be sure that all equipment is properly working before using them for laboratory exercises. Any defective equipment must be reported immediately to the Lab. Instructors or Lab. Technical Staff.
3. Students are allowed to use only the equipment provided in the experiment manual or equipment used for senior project laboratory.
4. Power supply terminals connected to any circuit are only energized with the presence of the Instructor or Lab. Staff.
5. Students should keep a safe distance from the circuit breakers, electric circuits or any moving parts during the experiment.
6. Avoid any part of your body to be connected to the energized circuit and ground.
7. Switch off the equipment and disconnect the power supplies from the circuit before leaving the laboratory.
8. Observe cleanliness and proper laboratory house keeping of the equipment and other related accessories.
9. Wear proper clothes and safety gloves or goggles required in working areas that involves fabrications of printed circuit boards, chemicals process control system, antenna communication equipment and laser facility laboratories.
10. Double check your circuit connections specifically in handling electrical power machines, AC motors and generators before switching "ON" the power supply.
11. Make sure that the last connection to be made in your circuit is the power supply and first thing to be disconnected is also the power supply.
12. Equipment should not be removed, transferred to any location without permission from the laboratory staff.
13. Software installation in any computer laboratory is not allowed without the permission from the Laboratory Staff.
14. Computer games are strictly prohibited in the computer laboratory.
15. Students are not allowed to use any equipment without proper orientation and actual hands on equipment operation.
16. Smoking and drinking in the laboratory are not permitted.

All these rules and regulations are necessary precaution in Electrical Laboratory to safeguard the students, laboratory staff, the equipment and other laboratory users.