UNIVERSITÉ Concordia UNIVERSITY

Department of Electrical and Computer Engineering

TELECOMMUNICATIONS LAB

INTRODUCTION

Summer, 2014

IN CASE OF EMERGENCY REMAIN CALM

AND FOLLOW THESE INSTRUCTIONS

Fire/Evacuation

Fire:
- If you see smoke or fire activate the nearest fire alarm.

Evacuation:
- Stay calm; do not rush or panic
- Safely stop your work,
- Gather your personal belongings; coat, purse, etc...
- Close and lock your door and windows.
- Use stairs only; do not use elevators or escalators,
- Once outside, move away from the building.
- Do not re-enter the building until instructed to do so by Security.

Suspicious Person/Package

Suspicious Person:
- Do not physically confront the person,
- Do not let anyone into a locked building/office,
- Call Security @ 514-848-(3717),
- Provide as much information as possible about the person and his or her direction of travel.

Suspicious Package:
- Do not touch or disturb object,
- Call Security @ 514-848-(3717),
- Notify your Supervisor.

Medical Emergencies

- In the event of a serious or life threatening injury or illness;
- From a safe location; call Security immediately at 514-848-(3717),
- Ensure your personal security before attempting first-aid,
- Provide the victim appropriate first-aid & comforting,
- Do not give the victim anything to drink or eat.

**If the injury is the result of a fall or significant trauma: Do not move the victim unless absolutely necessary.*

Shelter In Place

Communication:
- Shelter-in-Place will be announced by intercom P.A. voice communication, text messaging,
- Fire alarms will not be sounded.

Procedures:
- Lock classroom, office and lab doors if possible, remain quiet and do not enter the hallway,
- Should the fire alarm sound, DO NOT evacuate the building unless:
  1. *You have first hand knowledge that there is a fire in the building,*
  2. *You are in imminent danger, or*
  3. *You have been advised by Security or Police to evacuate the building.*
- Crouch down in the areas that are out of sight from doors and windows,
- Anyone in the hallways are to seek shelter in the nearest classroom,
- Anyone outdoors on campus should immediately take cover,
- If safe you can call 514-848-(8800) for more information on the situation.

Hazardous Materials

- If an emergency develops or if anyone is in danger, call 514-848-(3717),
- Move away from the site of the hazard to a safe location,
- Follow the instructions of Emergency Personnel,
- Alert others to stay clear of the area,
- Notify Emergency Personnel if you have been exposed to the hazard or have information about the release.

Power Failure

- Remain calm and move cautiously to a lighted area,
- Do not evacuate unless asked to by Emergency Personnel,
- Do not use candles!
- For localized outages, contact Security at 514-848-(3717).

UNIVERSITÉ Concordia UNIVERSITY

TABLE OF CONTENTS

Laboratory rules iv

Organization of the manual.. iv

Execution of the experiments... v

The lab report.. vi

Grading scheme vi

Submission of the report vii

Samples of cover page and results viii

Expectations of Originality…ENCS-SAS form…………………………………xi

LABORATORY RULES

Considering the large number of students attending the labs and in order for the lab to operate properly, the students are asked to abide by the following rules:

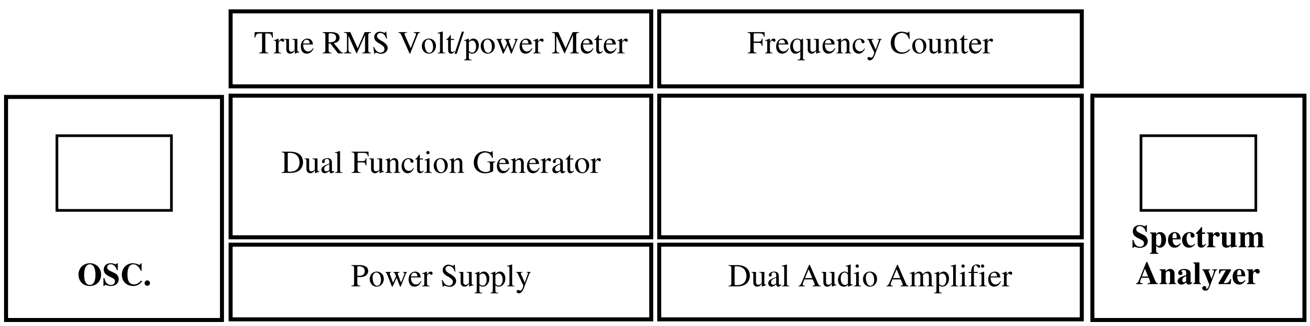
1. No eating or drinking is permitted in the laboratory.
2. CELL PHONES MUST BE SWITCHED OFF DURING THE SESSIONS. STUDENTS ARE NOT PERMITTED TO LEAVE THE LAB TO HOLD CELL PHONE CONVERSATIONS
3. Students should bring their own laboratory instructions.
4. No equipment is allowed to be exchanged from one bench to another.
5. All damaged or missing equipment and cables must be reported immediately to the demonstrator.
6. All data must be recorded neatly in the laboratory on a clean piece of paper and must be signed by the demonstrator.
7. Two students per laboratory station.
8. Answers to pre-lab questions are to be submitted to the TA at the beginning of the lab session.
9. After your laboratory session is completed all components, connecting jumpers, and cables must be returned to their respective places.
10. No student is allowed to move or stack the modules. Your laboratory demonstrator will provide you with all necessary modules.

LAB MANUAL ORGANIZATION

This manual contains five experiments; each experiment consists of the following sections:

- I. Objectives
- II. Introduction
- III. Pre-lab questions
- IV. Experimental procedure

The first part gives the objectives of the experiment. The second part provides a brief introduction to the experiment and relevant theory and formulas used in the experiment. The third part gives a list of questions, which should be answered by the student before coming to the lab. The fourth part describes the experimental procedure to be followed and the communication system block diagrams. All modules in the block diagram are prepared and plugged into the suitable slots of the experiment setup. The students will use the suitable cables (such as BNC or D connectors) to connect these modules. The basic setup on each bench in the communication lab consists of main power supply and dual audio amplifier, spectrum analyzer, dual function generator, true rms voltmeter/power meter, frequency counter, oscilloscope and enclosure/supply regulator.



Basic setup for communication systems

EXECUTION OF THE EXPERIMENTS

Each experiment must be studied in advance and the theory must be reviewed before conducting the experiment. Carefully follow the procedure and make the necessary connection as shown in each block diagram. If you are in doubt about the use of a particular module, consult the manual of such module before connecting it to other modules. All manuals are placed on a table in the lab and students are not allowed to remove or take these manuals out of the lab room.

The results and the collected data must be tabulated and written. Before leaving the lab have your data record checked by the lab demonstrator. The results should be presented in tables or graphs with clear headings and labels as shown in the samples.

<u>Each student must submit an individual report.</u>

Different marks will be assigned to group members according to their participation during the performance of the experiments. It is important to show the demonstrator that you are working with the group and understand the operation of all modules. The benches must be left clean at the end of the experiment.

THE LAB REPORT

Each lab report should be divided into five parts as follows:

Cover page: with group names and Ids (see sample)

Objectives: they have to be stated clearly and can be copied from the lab manual.

Introduction: it should be brief, written clearly in your own terms, and explain the relevant theory or formulas used in the experiment.

Experimental Results: should be broken down into sections as in the lab manual. See the sample of numerical and graphical presentations.

Questions and Discussion: answer all the questions (if any) posed in the lab manual or by the lab demonstrator. Discussion of any problems encountered during the experiment and any important observations made during the report write-up.

Conclusions: Express what was learned from the experiment and make any comments about the results.

GRADING SCHEME

Each lab report (including the pre-lab) will be marked out of fifteen. Late lab reports will be marked out of ten and no lab will be accepted after the last day of classes. The grading scheme is as follows:

Objectives and brief introduction	1/15
Results presentation and discussion	6/15
Conclusions	1/15
Participation**	3/15
Overall presentation and appearance	2/15
Pre-lab and preparation**	2/15

**It is important that the student prepares for each experiment by reading the instructions and the theory before conducting the experiment. Recording the data only is not considered participation, participation means that the student works on all modules (changing the settings, make the necessary connections, and interpret the results). The preparation and participation of each member in the group will be evaluated during each laboratory session.

SUBMISSION OF THE REPORT

Laboratory reports for each experiment are to be submitted at the beginning of the next lab with the "Expectations of Originality" form. The report will be returned on the day that the next experiment is to be performed.

ELEC 363

FUNDAMENTALS OF TELECOMMUNICATION SYSTEMS

EXPERIMENT #1

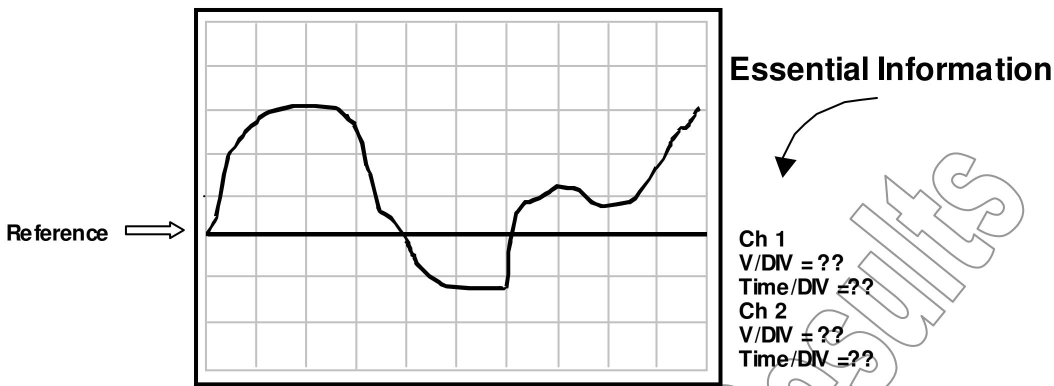
Signal Representation in Time & Frequency Domain

Name:	**ID:**
Partner's Name:	**ID:**
Partner's Name:	**ID:**
Section:	**###**

Lab Instructor:
Date Performed:
Date Submitted:

Presentation of the graphical Records

Oscilloscope



Explain and Comments on the above results

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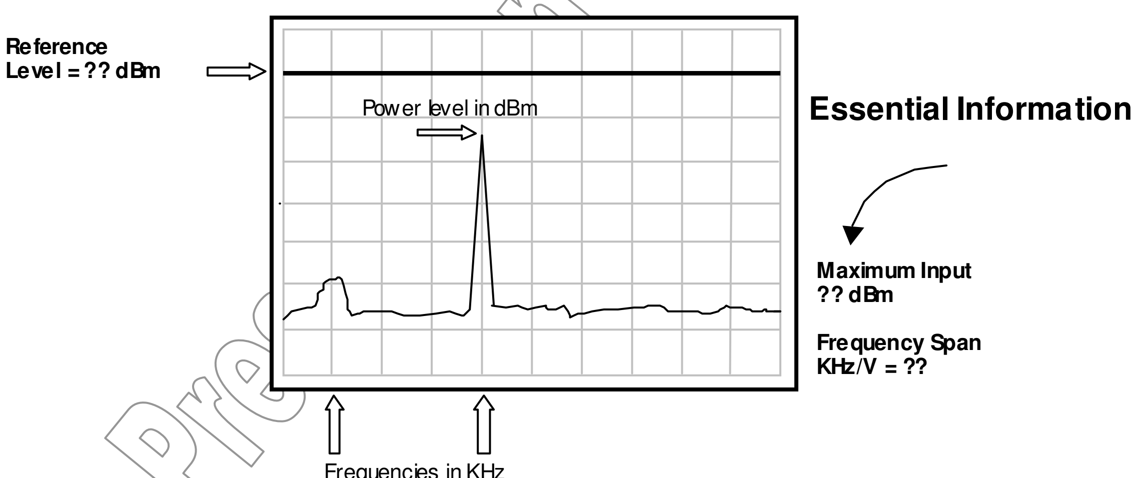
Spectrum Analyzer



Explain and Comments on the above results

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Presentation of the numerical Records

***Fixed variables*: X1**= xx , **X2**= xx

Sample of the Theoretical Calculations:

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Explain and Comments on the above results

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PLEASE NOTE THAT:

- The presentation of any step in the experimental procedure must be complete and followed by the necessary theoretical calculations and the comments.
- The comment statement must indicate the correctness and the validation of the results through a meaningful discussion.
- You must write the following statement on each lab report that you submit: "I certify that this submission is my original work and meets the Faculty's Expectations of Originality"

Include your signature, I.D. #, and the date. With the above statem

Faculty of Engineering and Computer Science
Expectations of Originality

This form sets out the requirements for originality for work submitted by students in the Faculty of Engineering and Computer Science. Submissions such as assignments, lab reports, project reports, computer programs and take-home exams must conform to the requirements stated on this form and to the Academic Code of Conduct. The course outline may stipulate additional requirements for the course.

1. Your submissions must be your own original work. Group submissions must be the original work of the students in the group.
2. Direct quotations must not exceed 5% of the content of a report, must be enclosed in quotation marks, and must be attributed to the source by a numerical reference citation[1]. Note that engineering reports rarely contain direct quotations.
3. Material paraphrased or taken from a source must be attributed to the source by a numerical reference citation.
4. Text that is inserted from a web site must be enclosed in quotation marks and attributed to the web site by numerical reference citation.
5. Drawings, diagrams, photos, maps or other visual material taken from a source must be attributed to that source by a numerical reference citation.
6. No part of any assignment, lab report or project report submitted for this course can be submitted for any other course.
7. In preparing your submissions, the work of other past or present students cannot be consulted, used, copied, paraphrased or relied upon in any manner whatsoever.
8. Your submissions must consist entirely of your own or your group's ideas, observations, calculations, information and conclusions, except for statements attributed to sources by numerical citation.
9. Your submissions cannot be edited or revised by any other student.
10. For lab reports, the data must be obtained from your own or your lab group's experimental work.
11. For software, the code must be composed by you or by the group submitting the work, except for code that is attributed to its sources by numerical reference.

You must write one of the following statements on each piece of work that you submit:
For individual work: **"I certify that this submission is my original work and meets the Faculty's Expectations of Originality",** with your signature, I.D. #, and the date.
For group work: **"We certify that this submission is the original work of members of the group and meets the Faculty's Expectations of Originality",** with the signatures and I.D. #s of all the team members and the date.

A signed copy of this form must be submitted to the instructor at the beginning of the semester in each course.

I certify that I have read the requirements set out on this form, and that I am aware of these requirements. I certify that all the work I will submit for this course will comply with these requirements and with additional requirements stated in the course outline.

Course Number:	______________________	Instructor:	______________________
Name:	______________________	I.D. #	______________________
Signature:	______________________	Date:	______________________

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[1] Rules for reference citation can be found in "Form and Style" by Patrich MacDonagh and Jack Bordan, fourth edition, May, 2000, available at http://www.encs.concordia.ca/scs/Forms/Form&Style.pdf.
Approved by the ENCS Faculty Council February 10, 2012

LIST OF LABORATORY EXPERIMENTS
FOR
FUNDAMENTALS OF TELECOMMUNICATION SYSTEMS

ELEC 363, FALL 2013

EXP. 1: SIGNAL REPRESENTATIONS IN THE TIME AND FREQUENCY DOMAINS.................. 1

EXP. 2: GENERATION AND RECEPTION OF AM SIGNALS 6

EXP. 3: GENERATION AND RECEPTION OF FM SIGNALS.. 13

EXP. 4: PULSE MODULATION AND SAMPLING: PAM, PWM AND PPM.............................. 18

EXP. 5: WAVEFORM CODING TECHNIQUES: PCM 24

REFERENCES.. 31

APPENDIX.. 32

EXPERIMENT #1
Signal Representation in the Time and Frequency Domains

OBJECTIVES

Gain an understanding of signal representation in time and frequency domains. Become familiar with the operation and use of spectrum analyzers.

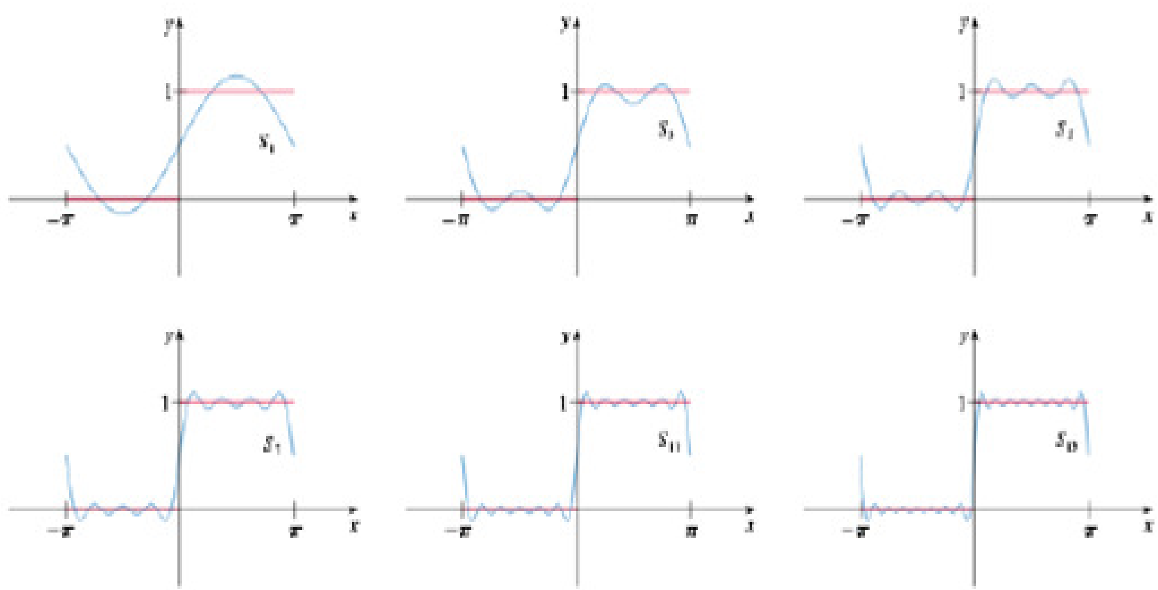
INTRODUCTION

Signal analysis can be conducted in time or frequency domains. Time domain analysis is used when the waveform, amplitude, period and phase of a signal are our main concern. In the laboratory, time domain observation and measurements of a signal are done using a menu driven digital oscilloscope. Frequency domain analysis helps us to allocate the right bandwidth to a specific signal and also study the mutual effect of nearby signals in frequency domain.
The basic idea of frequency domain analysis can be summarized as follows:
The signal (must be a time dependent function) is broken down into sine waves with different frequencies and amplitudes (Fourier Transform) and then displayed as vertical lines on the frequency-axis (horizontal-axis).

Fourier Transform:

$$f(t) = \sum_{n=1}^{\infty} b_n \sin\left(\frac{n\pi t}{L}\right)$$

What it means in time-domain:



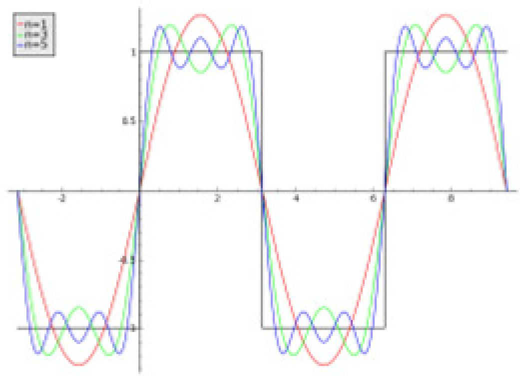


Figure1-1: Fourier series of the Square Wave

If a perfect sinusoidal signal (perfect being a distortion free, pure sine wave) with a frequency F_0 is injected into the input of the spectrum analyzer, one vertical line would be seen at the precise frequency F_0 of the signal, and the image will be displayed at $-F_0$ at the left hand side of the center frequency.

Fourier transform of Cosine:

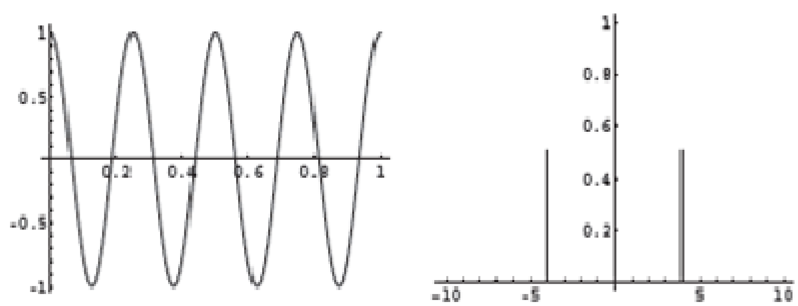


Figure 1-2: Cosine in time and frequency domain

What does spectrum analyzer show?

The height of each line represents the relative strength of that frequency component. Frequency domain observations and measurements are done in the lab using a spectrum analyzer. Spectrum analyzers aid in observing and measuring the power spectra of signals. Power spectral density or PSD describes how the power of the signal is distributed with frequency.

If the signal being analyzed can be considered a stationary process, the spectrum analyzer shows an estimate of its power spectral density.

Autocorrelation:

Autocorrelation refers to the correlation of a time series with its own past and future values.

$$R_g(\tau) = \lim_{T \to \infty} \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} g(t) g(t - \tau) dt$$

Power spectra is the Fourier transform of the autocorrelation function:

$$S_g(f) = \int_{-\infty}^{\infty} R_g(\tau) e^{-i2\pi f\tau} d\tau$$

Using the Agilent N9320B

The main parameters to set are:

- Frequency (the frequency represented at the center of the screen, Center Frequency, CF)
- Span (left side of the screen = CF-Span/2, right side of the screen = CF + Span/2)
- Amplitude (power level at the top of the screen)

The BW (bandwidth) of the N9320B is 9 KHz to 3GHz. Below 9 KHz it may be necessary to manually set the Resolution BW. The soft menu to set resolution BW is accessed by pressing the "BW/ Avg" button and can be reduced by turning the dial.

It is important to keep the frequency range of test equipment in mind when making measurements. We sometimes use the N9320B below its range, but the frequency readings are still accurate. Only amplitude measurements below 9 KHz are not accurate.

0 Hz reference

A 0 Hz reference is a signal generated by the analyzer to indicate 0 Hz. You can see it if you set the center frequency to 0 Hz, It is generated automatically without applying any input signal to the spectrum analyzer and is always visible if 0 Hz is displayed on screen.

FREQUENCY SPAN

The horizontal divisions on the screen correspond to the frequency axis and the vertical division correspond to the power axis. Similar to the time base definition on the oscilloscope (for example 1ms/div) the frequency span of spectrum analyzer defines the width of each horizontal division in Hz/V (or simply Hz/div). For example a frequency span of 100 KHz means that the frequency changes by (100 kHz/10) 10 KHz per division.

The following example will give you an idea about using the center frequency display selection of the frequency range and frequency span. Assume the input signal is centered at 10MHz with a variation of ± 20 KHz and you want examine the spectrum of this signal. The first step is to set the center freq to 10MHz, then set frequency span to 40 KHz or more. This will display 40 KHz across the screen (10 divisions) which is 4 KHz/div.

You may want to become more familiar with the analyzer by changing frequency and span settings and observing the position of the 0Hz reference on the display.

For further information about the function and other settings please refer to the spectrum analyzer section in the appendix.

PRELIMINARY REPORT

1. Define the peak-to-peak, RMS amplitudes, and average power of a sinusoidal wave and a 50% duty cycle square wave.

2. Explain the relations between the unit dBm, dBW, Volts, Ohms and Amps.

3. Find the amplitude and power spectra of a sinusoidal wave and a 50% duty cycle square pulse of amplitude (A) and frequency (F_0). Explain what the fundamental and harmonic components are.

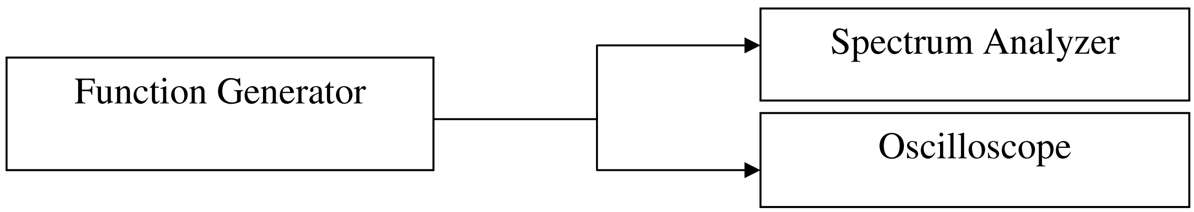
4. Explain in brief the operation and function of an oscilloscope. What is the purpose or function of the trigger section?

PROCEDURE

EQUIPMENT:

- Function Generator
- Oscilloscope
- Spectrum Analyzer
- Cables

1. Connect the modules as shown in Figure 4.



2. Obtain a 1Vp-p, 90 KHz sinusoidal signal from the 50 Ohms output of the signal generator. Use a BNC-T connector. Make sure to record the exact values used. Observe and record the output of the oscilloscope and spectrum analyzer, taking readings as accurately as possible.

3. Disconnect the spectrum analyzer while observing the signal on the scope. What happens and why? Record the result.

4. Repeat step 2 with a triangular wave.

5. Repeat step 2 with a 50% duty cycle square pulse. What is the amplitude of the 50% duty cycle square pulse? Why?

6. Repeat step 2 with a 20% duty cycle square pulse.

7. Compare experimental and theoretical results.

8. Assume that if you monitor two sinusoidal signals, one with a frequency of 5 kHz on channel 1 and 5.5 kHz on channel 2 of the oscilloscope respectively, what do you see if the oscilloscope is triggered on channel 1?

EXPERIMENT #2
GENERATION AND RECEPTION OF AM SIGNALS

OBJECTIVE

To understand the generation of AM signals using different modulation processes (DSB-LC, DSB-SC). The characteristics of these processes, in time and frequency domains, will also be studied. Examine the determination of the modulation index, sideband power and transmission efficiency for AM signals. Obtain an understanding of the demodulation process and receiver structures.

INTRODUCTION

If the amplitude of the modulating signal is greater than the amplitude of the carrier, distortion will occur. Distortion caused by over modulation also produces adjacent channel interference. For undistorted AM, the modulating signal voltage must be less than the carrier voltage, this ratio between the peak values of the message signal and the unmodulated carrier is known as the *modulation index m* (also called the modulating factor or coefficient, or the degree of modulation). Multiplying the modulation index by 100 gives the *percentage of modulation.*

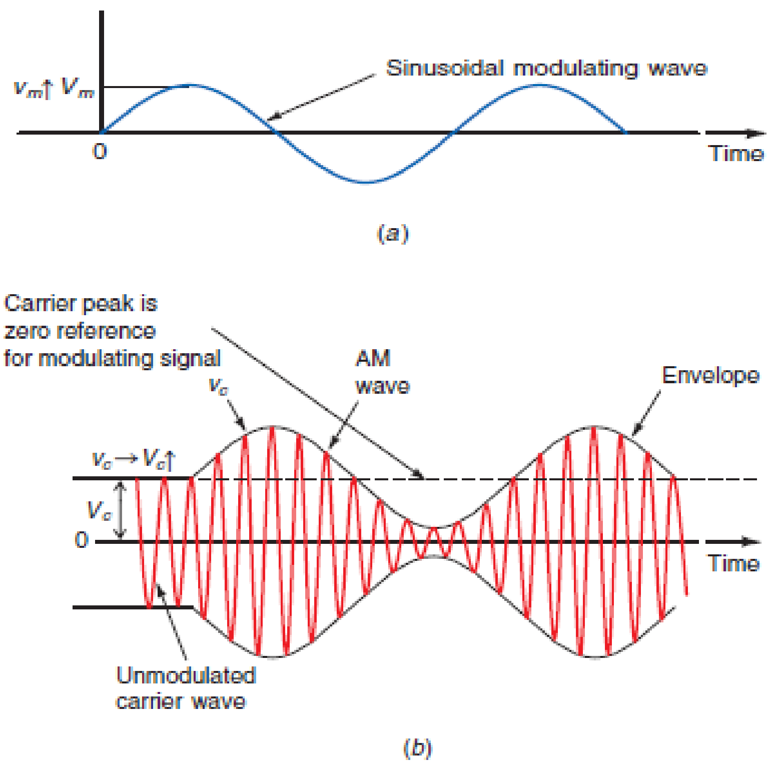
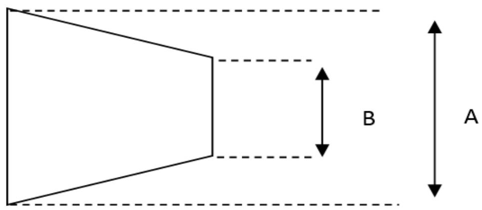


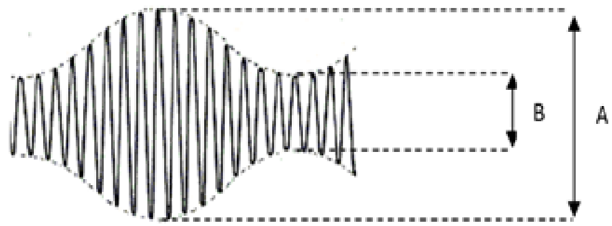
Figure 2-1: Amplitude modulation. (*a*) The modulating or information signal. (*b*) The modulated carrier.

Modulation index is measured using a single-tone sine wave as the message signalin both time and frequency domain. In time domain the oscilloscope can be set in either time-base operation mode or in X-Y mode and m is determined using the following formula:

$$m = \frac{A - B}{A + B} * 100\%$$



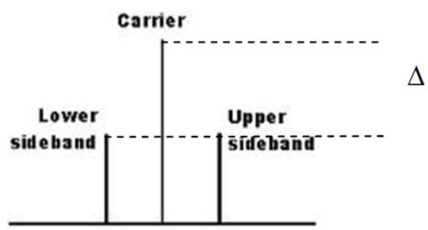
X-Y mode



Time-base mode

The measurement of the AM modulation index in frequency domain is done using spectrum analyzer. The difference Δ(delta in dB) between carrier power and the power in one sideband corresponds to the modulation index. Use the following formula to determine the percentage of modulation index.

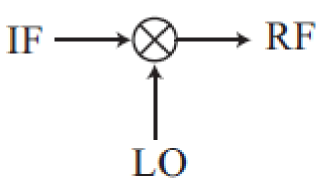
$$m = \frac{2}{10^{\frac{\Delta}{20}}}$$



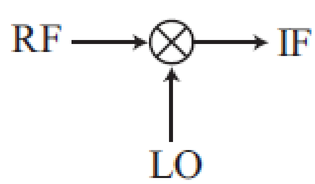
Mixer:

As shown below, an ideal mixer translates the modulation around one carrier to another. In a receiver, this is usually from a higher RF frequency to a lower IF frequency. In a transmitter, it's the inverse.

We see that the modulation is indeed translated to two new frequencies, LO + RF and LO − RF. We usually select either the upper or lower "sideband" by filtering the output of the mixer.



Up-conversion



Down-conversion

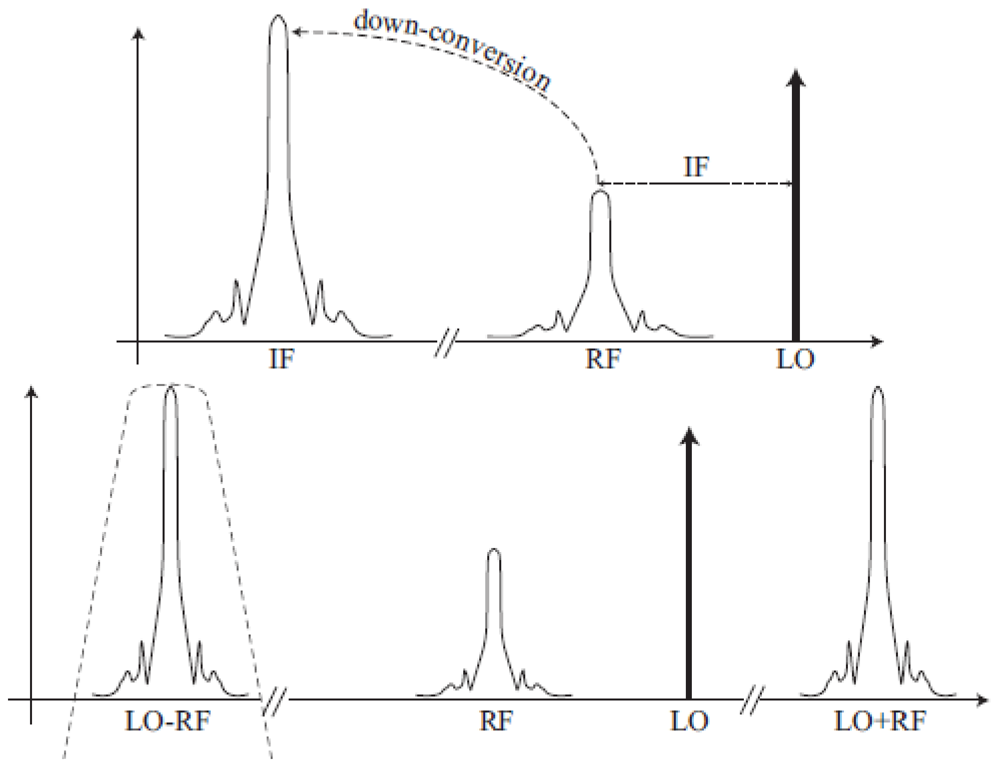


Figure2-2: Mixer and filter

Automatic Gain Control (AGC):

To prevent overloading of the IF amplifiers when a strong signal is received some form of gain control is necessary. By passing the received signal through a long time constant LPF, an average of the received signal power can be formed. This control voltage is applied to the base of the amplifier transistors in the IF stages and possibly the RF stage thereby changing their gain to suit the received signal power.

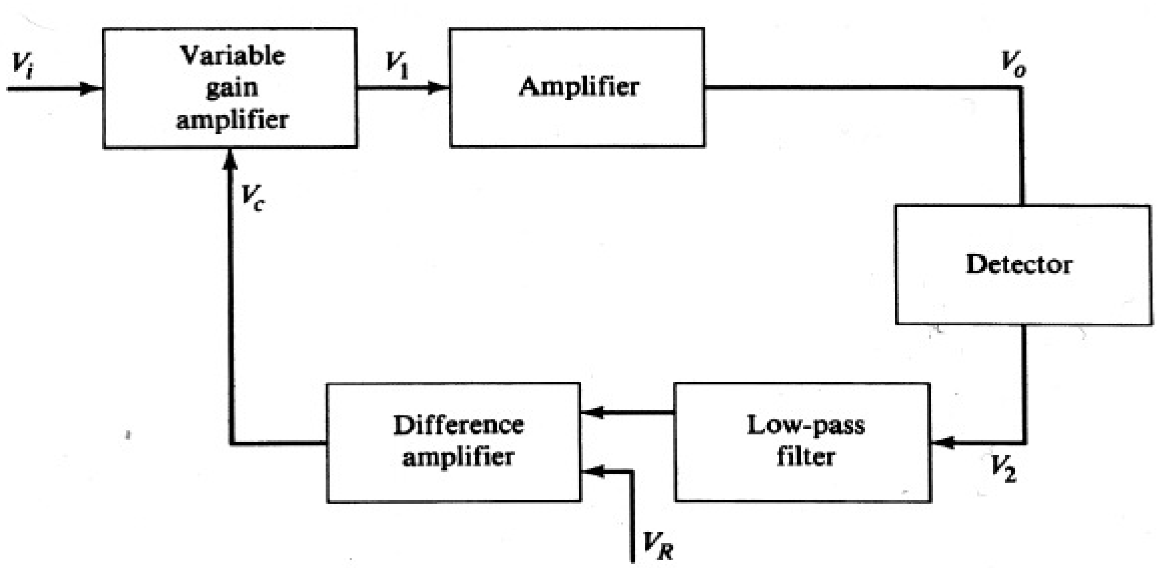


Figure2-3: AGC block diagram

Envelope Detector (Rectifier):

The output voltage of the envelope detector is proportional to the magnitude of the instantaneous RF input voltage. Assuming that sufficient low pass filtering is applied at its output to eliminate RF ripple, this detector produces a voltage proportional to the envelope amplitude of the RF signal.

3. PRELIMINARY REPORT

Consider a sinusoidal carrier c(t) defined by:

$$c(t) = A_c \cos(2\pi f_c t)$$

Where A_c and f_c are the amplitude and frequency of the carrier respectively. Let a(t) denote the baseband modulating signal, which contains the information to be transmitted. As an illustrative example, choose a(t) to be a single-tone signal.

$$a(t) = A_m \cos(2\pi f_m t + \varphi)$$

Where φ is an arbitrary phase.

1. In one paragraph describe with equations the modulated DSB-SC and DSB-LC signals in time and frequency domains. Use f_c =1 MHz and f_m=10 KHz.
2. Define and derive the modulation index (m) for the time domain representation of a DSB-LC signal.
3. Evaluate and compare carrier power with sideband power and transmission efficiency of a DSB-LC signal in terms of m. Give the numerical results for m=0.25 and 1.
4. Normally, the power spectrum is shown on the screen of a spectrum analyzer in dB. Derive the modulation index (m) from the difference (in dB) between carrier and sideband power.
5. Give some reasons why IF should not be zero?
6. What is the limitation on transmission efficiency of DSB-LC?

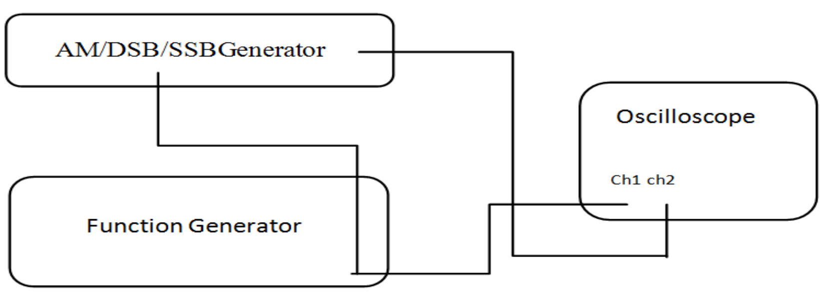
4. PROCEDURE

EQUIPMENT:

- Function Generator
- AM/DSB/SSB Generator #9410
- AM/DSB Receiver #9411
- Spectrum Analyzer
- Oscilloscope

a) AM/DSB GENERATION

1. On the AM/DSB/SSB Generator, turn the carrier level and the RF gain-control tuning knobs to MAX. Adjust the RF tuning knob to generate a carrier frequency of 1100 KHz at the AM/DSB RF output. Use a frequency counter to make this measurement.
2. Obtain a 250 mV, 10 KHz sinusoidal signal from a signal generator. It will be used as the modulating signal. Use the BW limit on the scope to hide the noise (why do we have noise?) so the signal is easy to read.
3. Use a T connector to connect the signal generator output to the audio input of the AM/DSB/SSB generator and to Channel #1 of the oscilloscope. Connect the AM DSB output to Channel #2. Describe and sketch the waveforms.



4. Compute the percentage modulation of the output waveform (using the oscilloscope and the spectrum analyzer).
   What is the maximum BW occupied?

5. Use a 10 KHz sinusoidal modulating signal and generate DSB-LC waveforms with 20%, 50%, 100%, and >100% modulation. Sketch the measured waveforms in both the time and frequency domains. Measure Δ , the difference in power levels (dB) between the carrier and the sidebands and calculate the percentage of modulation in this way.

M	A, B	M (TIME DOMAIN)	Δ	M (FREQ DOMAIN)
20%				
50%				
100%				
>100%				

6. Set the carrier level to MIN. Sketch the measured results in both time and frequency domains. Is the modulated signal a DSB-SC or DSB-LC signal?

b) AM/DSB RECEPTION

1. **<u>Mixer:</u>** Obtain a DSB-LC signal from the AM/DSB/SSB generator with a carrier frequency of 1 MHz +/-500Hz, 50% modulation index and a modulating signal of 3KHz. Adjust the RF gain-control 10% turn clockwise from Min. Apply this signal to the 50 ohm RF input of the AM/DSB Receiver (model 9441). Adjust the RF tuning for an OSC output frequency of 1455 KHz +/-500Hz. Turn the AGC off.
   a. Monitor the signal at the Mixer output of the AM/DSB receiver, using both the oscilloscope and the spectrum analyzer. Describe the obtained waveform and spectrum. Record the relative levels and frequencies.
   b. Set the spectrum analyzer to view 0 to 3MHz. Vary the generator RF gain from min to max and observe the spectrum. What happens and why?
   c. Explain the experimental results.

2. **AGC:**
    a. Remove the modulating signal from the Audio input of the AM/DSB/SSB generator.
    b. Set the RX Oscillator to get an IF of 455kHz +/-500Hz. Adjust the RF gain-control to mid level (half way). Adjust the RF tuning to obtain 455 KHz +/-500Hz at the IF test point.
    c. Reapply the modulating signal to the Audio input of the generator 9410. Vary the RF gain of the generator from MIN to MAX in steps while observing the signal at the IF output of the receiver on the oscilloscope and spectrum analyzer *(do not connect the scope and analyzer at the same time to avid loading the signal)*. Show images of the IF output signal with AGC "on" and "off" for several RF gain settings (low, medium, high) that illustrate the functioning of AGC. Show using pictures and explain the results.

3. **Detector:**
    a. Set the generators to get a carrier frequency of 1455kHz; modulating frequency of 1kHz and a modulation index of 50%.
    b. Select the ENV (ENVELOPE) detector on the receiver.
    c. Maintain an IF frequency of 455kHz.
    d. Adjust the AM generator's RF gain to get a distortion free signal at the receiver's audio output (with AGC on).
    e. Vary the modulating signal from 1 kHz to 15 kHz in 2kHz steps. Plot the amplitude of the demodulated signal versus frequency in the table below. Explain the observed results.

FREQUENCY(KHZ)	**AUDIO OUTPUT (V**p-p)
1	
3	
5	
7	
9	
11	
13	
15	

EXPERIMENT 3

GENERATION AND RECEPTION OF FM SIGNALS

I. OBJECTIVES

The student will have an understanding of the generation of FM signals and their characteristics after completing the experiment. The student will also develop an understanding of the reception of FM signals and the structures of various FM demodulators.

II. INTRODUCTION

Two methods are used to alter frequency of the carrier signal according to the modulating signal. The direct method uses a voltage-controlled oscillator VCO with operating range (or the linear region) that cover the maximum and the minimum swing of the modulating signal. One of the main problems of the direct FM generation is instability of the carrier frequency. The Indirect method of FM generation eliminates this problem. The Indirect method of generating the Narrow Band FM signal (NBFM) is known as Armstrong method (see the front panel of 9414 module). The following definitions are important to know:

- The sensitivity (k_f) of the FM modulator: When the carrier frequency oscillates between a minimum & maximum, these frequency limits are determined by the sensitivity (k_f) of the FM modulator.

- Frequency deviation Δf is the frequency difference between the un-modulated carrier and the maximum (or the minimum) frequency of the modulated carrier.

- The frequency deviation of the modulated FM is $\Delta f = k_f\ A_m$ where A_m is the modulating signal amplitude.

In frequency modulation (FM), the instantaneous frequency, $\omega_i\ (t)$, is varied linearly with the modulating signal, $a\ (t)$:

$$\omega_i(t) = 2\pi f_c + 2\pi k_f a(t)$$

Equivalently,

$$f_i(t) = f_c + k_f a(t)$$

Where

$$\frac{d\phi(t)}{dt} = 2\pi k_f a(t)$$

And k_f represents the frequency sensitivity expressed in Hz/volt, and is defined as above.

Let $a(t)$ be a single-tone sinusoidal signal, expressed by:

$$a(t) = A_m \cos(2\pi f_m t)$$

The instantaneous linear frequency, $f_i\ (t)$, of the FM signal becomes:

$$f_i(t) = f_c + k_f A_m \cos(2\pi f_m t)$$

This indicates that $f_i\ (t)$ is varied from (f_c - Δf) to (f_i + Δf). $\Delta f = k_f A_m$, and is called the frequency deviation. The angle, $\phi(t)$, of the modulated signal x(t) becomes:

$$\phi(t) = \frac{\Delta f}{f_m}\sin(2\pi f_m t)$$

Where

$$\beta = \frac{\Delta f}{f_m}$$

Is called the modulation index of the FM signal. The FM signal can be represented as:

$$x_{FM}(t) = A\cos[2\pi f_c t + \beta\sin(2\pi f_m t)]$$

The modulation index is a function of the modulating signal frequency and amplitude as given in this formula $\beta = \frac{k_f\ A_m}{f_m}$. To set the modulation index to a specific value, simply change the modulating signal amplitude. Since β can take any positive value starting from 0, two formulas will be used in this lab to determine the modulation as follows:

If $\beta < 0.5$ the number of the most significant sideband pairs in FM spectrum will be 2 (according to the Bessel coefficients Jo=0.94, J1=0.24 and J2=0.03), notice that the power contained in the 2nd component is very low. The spectrum in this case is similar to the one obtained in the previous lab when analyzing the DSB-LC signal and the formula for modulation index will be $\beta = \frac{2}{10^{\frac{\Delta}{20}}}$. The bandwidth occupied by the FM signal in this case will be approximated by the value $BW = 2f_m$.

If $\beta > 0.5$ the number of the most significant sideband pairs in FM spectrum start to grow (see the table below) and the formula for the modulation index will be $\beta = \frac{\Delta f}{f_m}$. In the lab Δf can be measured using DEVIATION kHz display on the front panel of the FM/PM RECEIVER. The bandwidth occupied by the FM signal in this case will be approximated by the value $BW = 2\Delta f$.

III. PRELIMINARY REPORT

1. Expand the expression of $x_m(t)$ as a sum of two product terms:

$$x_{FM}(t) = A\cos[\beta\sin(2\pi f_m t)]\cos(2\pi f_c t) - A\sin[\beta\sin(2\pi f_m t)]\sin(2\pi f_c t)$$

Find the condition on β where $x_{FM}(t)$ can be approximated as an AM signal and explain the meaning of a NBFM signal and show that NBFM is a linear modulation. Draw a block diagram of a NBFM modulator.

2. Describe the indirect method of generating FM signals and the Armstrong FM Modulator. Explain the operation of the indirect FM/PM Generator shown in Figure 4.1. For an audio input having a frequency of 1 kHz, describe the expected signals at the outputs of the elements in this block diagram, assuming that FM generation is chosen with an initial value of beta = 0.2

3. Describe the direct method of generating FM.

4. Describe the indirect method of demodulating FM signals using a phase' locked loop (PLL). Explain the operation of a PLL, its lock range, and its capture range.

IV. PROCEDURE

Equipment:

- Power Supply/Dual Audio Amplifier #9401
- Indirect FM/PM Generator #9414
- FM/PM Receiver #9415-10
- AC Voltmeter #9404
- Oscilloscope
- Spectrum Analyzer

The modulated FM signals (direct FM) and the effect of the frequency deviation can be observed in time domain using the dual function generator and oscilloscope as follows:

Obtain a Sine wave with 1kHz and 8Vp-p from channel A, and square wave with 100Hz and 8Vp-p from channel B of the dual function generator. Connect channel B output to the frequency modulation input located on the front panel of the Dual function generator and use a T-connector to display the signal on channel 1 of the oscilloscope. Connect channel A of the dual function generator to channel 2 of the oscilloscope and set the frequency deviation to 50% of its range. Explain the displayed waveforms and measure the frequency deviation Δf . Set the frequency deviation to maximum and measure Δf again. Record you observations on the oscilloscope for Sine and Triangle waveforms.

<u>Tech note</u>: *By default, the scope is triggered by the modulating signal on Ch1*

a) INDIRECT FM GENERATION

1. Obtain a sinusoidal modulating signal with a frequency (f_m) of 1 kHz, and an amplitude (A_m) of $50\,mV_{p-p}$ *(set the scope BW limit to 20MHz)* from the function generator and apply it to the *Audio Input* of the Indirect FM/PM Generator (model 9414). Select FM mode and set the RF GAIN to 50% clockwise. Observe the signals at *Mixer Output* (point *2*), *Amplifier Output* (point *3*), *NBFM* (point *4*), *WBFM* (point 5) using the oscilloscope. Also look at the spectrums of points 4 and 5. Record the measured results and explain.
<u>Tech note</u>: trigger the scope with the modulating signal when looking at point 2.

2. Monitor the NBFM signal (point 4) using the spectrum analyzer. Vary the modulating frequency fm and the amplitude A_m. Observe the variations of the spectrum and comment on the differences compared to an AM signal.

3. Set the modulating frequency to 3kHz, amplitude to 0 (small as possible). Monitor the NBFM signal (point 4) using a spectrum analyzer. Slowly increase the amplitude of the modulating signal until the power difference (D in dB) between the spectral lines representing the carrier and the first sideband pair is 20 dB and all other sideband pairs are zero. Calculate the modulation index β as

$$\beta = 2(10^{-D/20})$$

Explain the meaning of the above formula. Now apply the NBFM signal to one of the *PM/NBFM RF Inputs* of the FM/PM Receiver (model 9415-10). Set the *Deviation* push button to NBFM. Record the frequency deviation delta f indicated by the display. Calculate the modulation index β, and compare it to the previously obtained value of β. Calculate the bandwidth.

b) INDIRECT FM RECEPTION

1. Obtain a sinusoidal modulating signal with a frequency of 2kHz and amplitude of 30 mV RMS from the signal generator. Apply this signal to the *Audio* Input of the Indirect FM/PM Generator Model 9414. Adjust the Indirect FM generator NBFM "RF GAIN" to get a $300mV_{rms}$ (no load) RF Signal. Connect the *NBFM* Output (point 4 of Generator 9414) to one of the *PM/NBFM RF* Inputs (points 4 or 5 of the FM/PM Receiver 9415-10). Monitor the NBFM deviation. Adjust the amplitude of the modulating signal to obtain a deviation reading of 4kHz <u>*with the lowest amplitude possible*</u> (do not forget to set the deviation button to NBFM). Observe the *NBFM Audio* output (point 9 on the FM/PM Receiver) signal on the oscilloscope.

2. Vary the modulating frequency from 0kHz to 20kHz in 2 kHz steps and note the distortion effect of the *Audio* output signal due to the effect of the discriminator response. Explain the obtained results.

3. Set the modulating frequency to 1 kHz, RF Gain of the WBFM signal to 50% and connect the 101.7MHz *WBFM RF Output* (point 5 of the Generator 9414) to one of the *WBFM RF Inputs* (point 1 or 2 of Receiver 9415-10).
   - Observe the *Baseband Output* (point 11 of Receiver 9415-10) on the oscilloscope. Set the deviation button to WBFM.
   - Adjust the *RF Tuning to 101.7MHz: M*aximum clean signal level on the scope and proper center frequency (i.e. GREEN tuning LED should be on). Alternately vary the modulating amplitude and RF tuning to get maximum deviation (typically 140kHz, at least 120kHz) while maintaining a clean baseband signal on the scope. If needed, increase the RF gain (while maintaining a distortion free baseband signal) to increase the deviation.
   - Reduce the modulating amplitude to get deviations from maximum down to 10 kHz in 10 kHz steps (by adjusting the input amplitude) and record the peak-to-peak amplitude (Vp-p) of the baseband output signal (use the AC volt meter to do this, remembering that the readout gives RMS not peak-to-peak so convert).
   - Plot the deviation vs. $V_{p\text{-}p}$ curve. Explain the obtained results.

4. Compare all experimental results with the theoretical results

EXPERIMENT **4**

PULSE MODULATION AND SAMPLING: PAM, PWM AND PPM

I. OBJECTIVES

To understand the fundamentals of sampling and pulse modulation with pulse amplitude modulation (PAM), pulse-width modulation (PWM), and pulse-position modulation (PPM) techniques.

II. INTRODUCTION

Pulse modulation is based on the idea of having an analog signal sampled at regular intervals, then using these samples to vary the parameter of a pulse waveform. For example, in Pulse **Amplitude** Modulation (PAM), the amplitude of the pulse waveform is varied in proportion to the amplitude of the sampled signal. Similarly Pulse **Width** Modulation (PWM) and Pulse **Position** Modulation (PPM), the pulse width or position on the time axis is varied in proportion to the amplitude of the sampled signal.

The minimum sampling rate T_S for the message signal is called Nyquist rate. The sampling frequency $f_S = 1/T_S$ must be greater than twice the highest frequency of the signal being sampled (message signal, f_S). For example, the sampling frequency of a 4kHz voice signal is at least 8kHz. When the sampling rate is less than the Nyquist rate, a phenomenon called aliasing occurs which produces distortion in the recovered signal. To reduce the aliasing to acceptable level, the sampled signal must be band-limited. This can be accomplished by passing the message signal through a low-pass filter to attenuate the higher frequency components of the analog signal before sampling. This process is known as pre-filtering. As a result, the message signal becomes a band-limited signal prior to sampling.

During this experiment, students will enhance their understanding of different types of pulse modulation techniques through time domain and frequency domain observation and measurements. The main tool in this case is the dual channel oscilloscope. Measurements and observations are taken at designated points on the modulator and demodulator modules. These "test points" (TP1…TP7) are shown on the detailed block diagram of each module (please see the appendix). Access to these points is done via a 9-pin connector (8 test points and pin #9 is common). The "signal interrupt/select" module allows the student to observe two test points and/or interrupt the 8 data bit (1 byte code word) signal by disconnecting or dropping any bit in the code word as described below.

SIGNAL INTERRUPTER/SELECTOR

Data (Parallel output) or Test point (Test Bus) selection for each "Selector output" is made by pushing a button. An LED will indicate the position of the selector. There is a single input to the module labeled as **Inputs,** and the outputs can be taken after the interruptor (8-bits) and/or from selectors (selector1 and selector2). In other words the module will allow you to plug in 8-bits of data using the 9 pin D-connector and provide multiple outputs known as parallel (From interrupter) and sequential (From selector1 or selector2) outputs.

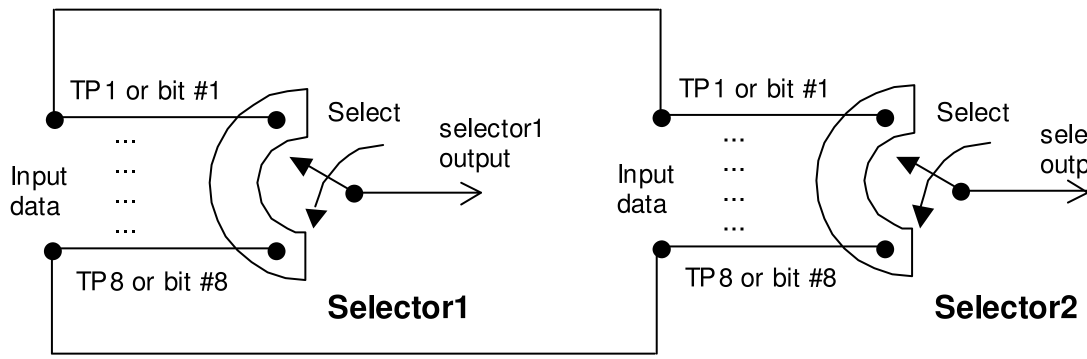


Figure (a) Block diagram of selector1 and selector2 internal connections and operation

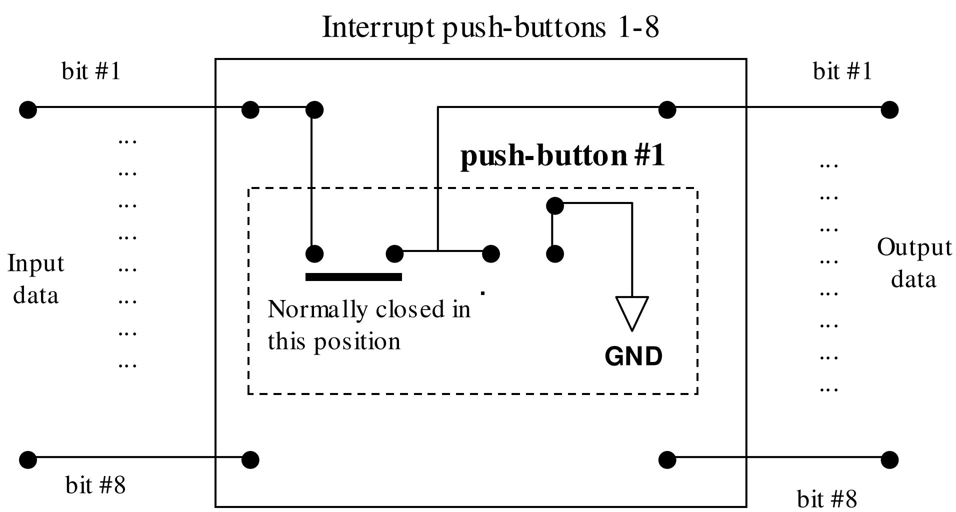


Figure (b) connection details of the push-button #1 in normal position (not depressed)

Figure (b) above shows the connection details of a single push-button in the interrupter section of the module. Depressing push-button #1 will interrupt the connection for bit #1 and replaces the data bit by ground at the output. This means that if the data on connection #1 were logic 0 before interruption, you wouldn't observe any difference after this action.

III. PRELIMINARY REPORT

1) With the help of graphs, show the difference between the natural and flat-top sampling PAM signal.
2) If the message signal contains frequency components ranging from 0 Hz to $f_{\max}$ Hz and the sampling signal rate is f_S Hz, draw the PAM signal spectrum up to 3 replicas. Explain the effect of increasing $f_{\max}$ on the frequency range of each replica.
3) What is a band-limited signal? Why is the pre-filtering necessary?

VI. PROCEDURE

Equipment Required

Dual Function Generator
Sync. Audio Generator
Signal Int/Selector
Noise Measurement Filters
PAM/ASK Generator
PAM/ASK Receiver
PWM/PPM Generator
PWM/PPM Receiver
Low Pass Audio Filter
Spectrum Analyzer

1- PAM

In this part of the experiment you will observe the effects of aliasing in both time and frequency domains. The message signal will be varied between 1kHz and 6kHz while maintaining the sampling frequency at 8kHz. Graphically record all observations in time and frequency domains.

Figures 1 and 2 show the block diagrams of the experiment set up for observations in the time domain and frequency domains respectively. A DC voltage from the DC Voltmeter/Source is applied to the Frequency Modulation Input on the Dual Function Generator to fine-tune the frequency of Ch. A. This allows a stable display of PAM signals on the oscilloscope.

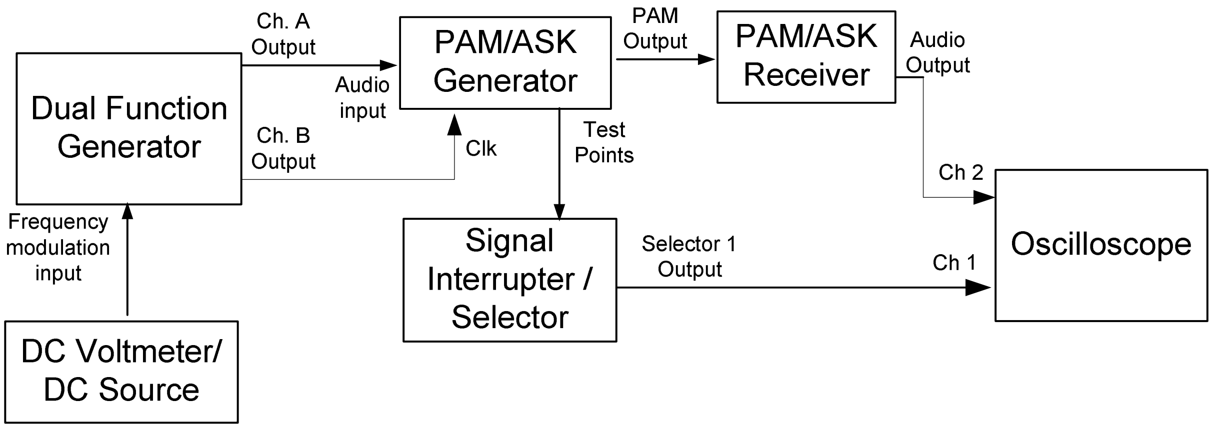


Figure 1 Time Domain Observation

Set the Dual Function Generator:
Output A: 1kHz +/- 0.5Hz, Sine Wave, 2Vp-p
Output B: 8kHz +/- 0.5Hz, pulse with 50% duty cycle, MAX output level, 0dB attenuator.
Deviation: 50%

1. Observe the message signal (TP1) and the PAM signal (TP7) for both NATural and FLAT-top modes. Record your observation (graphically) and explain the obtained results.
Note that each time a PAM signal is observed on the oscilloscope you need to adjust the DC Voltmeter/DC Source output to stabilize the display of the PAM signal.
2. Vary the duty cycle of the sampling signal from 90% to 10%. Describe the effect of changing the duty cycle on the reconstructed signal. Record and explain the obtained results.
3. Set the duty cycle of the sampling signal to 50%. Change the frequency of the message signal to 2.0kHz, 4.0kHz and 6.0kHz. Does the PAM signal seem to resemble the message signal? Compare the sampling rate to the Nyquist rate in each case. Compare the frequency of the message and reconstructed signals in each case. Explain the obtained results and comment.
4. Set up the system in figure 2. Observe PAM spectrums for message signals of 2kHz, 4kHz and 6kHz.

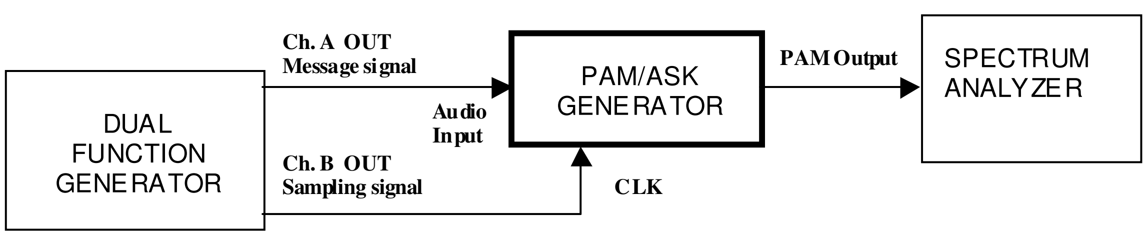


Figure 2 Frequency Domain observation of PAM signal

Set the sampling signal duty cycle to 50%.
Adjust the spectrum analyzer center freq and span to display the message signal and replicas on the spectrum analyzer. Set the Spectrum analyzer "Res BW" to 30Hz

5. What happens to the spectrum as the message signal frequency is increased? Record and describe all the observations.

2- PWM/PPM

Figures 3 and 4 show the block diagram of the setup for PWM and PPM respectively. The ramp signal (sawtooth wave) is provided by channel A of the Dual Function Generator. The Synchronous Audio Generator and the Low Pass Audio Filter provide the Sine wave message signal. These signals are applied to the inputs of the PWM/PPM Generator. The PWM/PPM Receiver is used to reconstruct the message signal. In order to transmit DC levels, the output of the PWM/PPM Receiver must be DC-coupled.

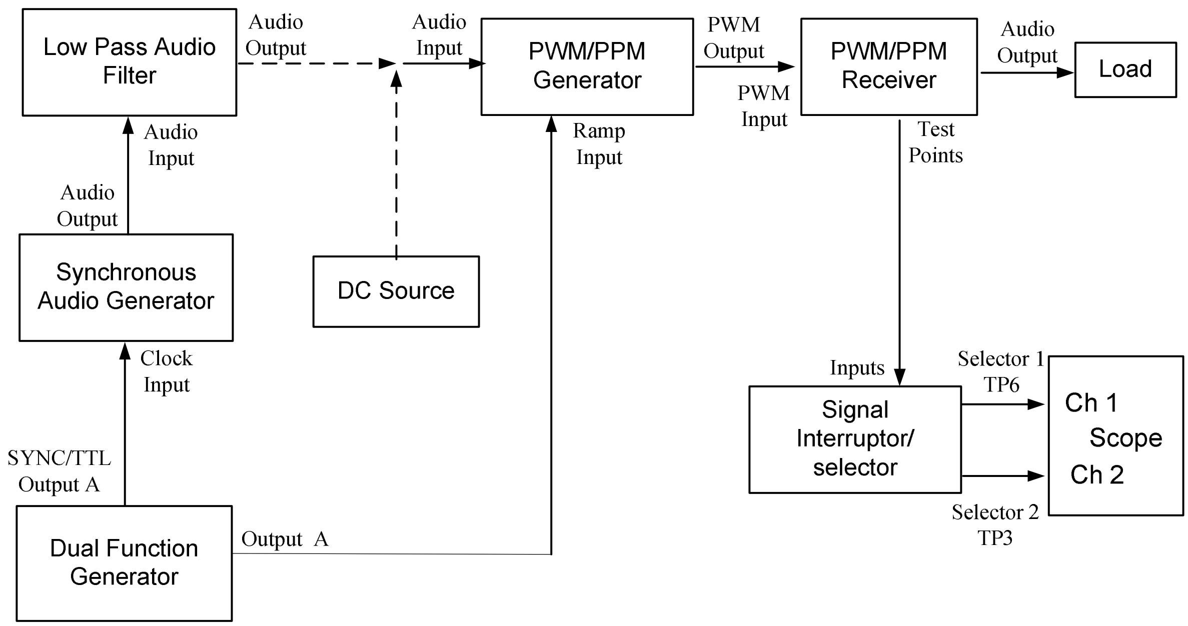


Figure 3 **PWM System**

Set the equipment as follows:

- Dual Function Generator: 8kHz, sawtooth wave, 2.7Vp-p (not including the voltage spikes at the end of the sawtooth).
- Synchronous Audio Generator: N=10, GAIN control is set to CAL.
- Low pass Audio Filter: **-** 4^{th} Order - CUTOFF FREQUENCY control fully counterclockwise (to minimum) and then turn it one half turn clockwise. - Pull the GAIN control out to the VAR position, and adjust the GAIN until the amplitude of the sine wave message signal is 1Vp-p. (If the output is too high, reduce the cutoff frequency.)

- Use the PWM system shown in figure 3 and connect audio output of the Low Pass filter to the Audio input of the PWM/PPM.
- Observe the message signal (TP6 on Ch 1 of the oscilloscope) and the PWM signal (TP3, Ch2 on the oscilloscope). Sketch and explain the obtained results.
- Slowly increase the GAIN on the Low Pass Audio Filter. What happens to the PWM signal? Sketch and explain the obtained results.
- Disconnect the Low Pass audio filter and connect the DC Source to Audio Input of the PWM/PPM Generator.
- Set the DC Message Signal Amplitude to 0V, -1.0V and 1.0V. Sketch and explain the obtained results.

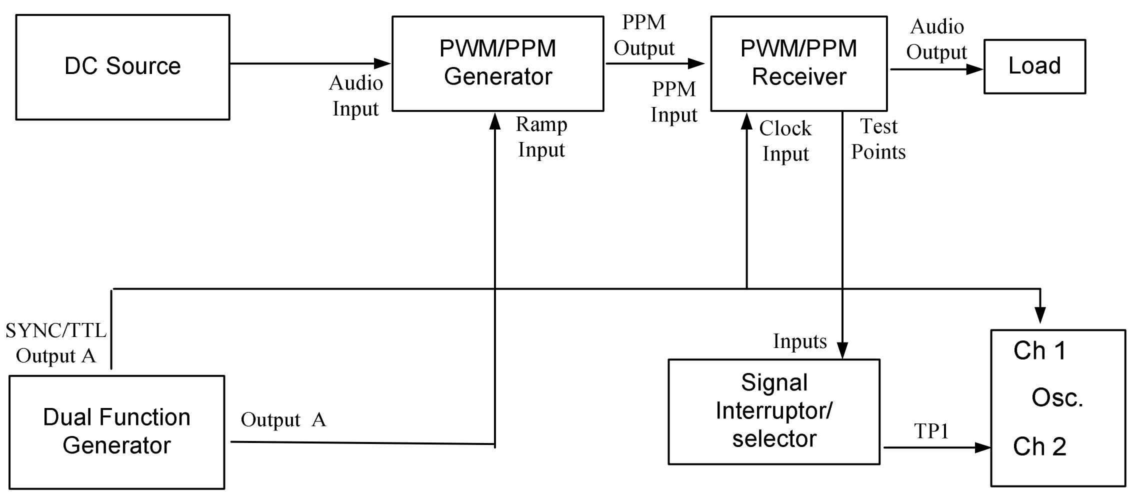


Figure 4 **PPM System**

- Use the PPM system shown in figure 4. PPM generator – pulse gen - "course" set to min, "fine" set to max.
- Explain why the clock is connected to Ch 1 of the oscilloscope instead of Ch 2.
- Set the DC Message Signal Amplitude to 0V, -0.75V and +0.75V. Sketch and explain the obtained results.

Experiment 5

WAVEFORM CODING TECHNIQUES: PCM

I. OBJECTIVES

To understand the fundamentals of quantization processes and waveform coding techniques.
To become familiar with Pulse-Code Modulation (PCM).

II. INTRODUCTION

The conversion of an analog signal (signal that is characterized as a continuous time function) into digital form is accomplished by sampling the signal at Nyquist rate, then assign a binary code to each samples. The process is called A/D or *QUANTIZING*. In this process, the maximum swing of the analog signal is divided into regions (or voltage levels) with equal width (uniform quantization) or non-equal width (non-uniform quantization), and a binary code is assigned to each region (voltage level). The number of bits used in the binary code is related to the number of regions or signal levels as follows:

$$number\ of\ levels \quad L = 2^{n-1}$$

Where n is the number of bits in the binary code word. The resolution of an A/D converter is a measure of the nominal analog change required for a 1 bit change in the binary code word. For example given an analog signal with input range -1 to +1 volts, and an 8 bit A/D converter, the resolution can be determined as follows:

$$A/D\ resolution = \frac{input\ range}{2^8} = \frac{+1-(-1)}{2^8} = \frac{2.00}{256} = 7.813mV$$

The value obtained (7.813mV) is referred to the region width. Notice that all function values within a particular region are coded into the same binary number. In practice the resolution is expressed as a number of bits or the length of the code word, for example 4bits A/D, 8bits A/D, 16bits A/D, and so on.

The A/D and D/A converters are considered basic parts of PCM systems. In the PCM the sample value is coded into a binary number that represents all values over the dynamic range of the analog signal. The resolution is therefore dependent on the dynamic range of the analog signal.

The following formula will be used to determine the signal-to-quantization noise (SQNR):

$$SQNR = \frac{Signal\ Power}{Quantaization\ noise\ power} = \frac{P_S}{P_{QN}}$$

In decibels

$$SQNR = 10\ \log(P_S / P_{QN}) = 20\ \log(V_S / V_{QN})$$

During this experiment, students will enhance their understanding of quantization process and waveform coding techniques through time domain and frequency domain observations and measurements. The main tool in this case is the dual channel oscilloscope, which exist on the left hand side of the basic set up. The measurements and observations are taken at designated points on the Encoder and Decoder modules. These points are known as "test points" (TP1…TP7) and shown on the detailed block diagram of each module (please see the appendix). These points are accessed through a 9-pin connector (8 test points and pine #9 is common). In order to observe these points simultaneously we need 8-channel oscilloscope, which is not available in the lab. However, the signal interrupt/select module can be used to solve this problem. Incorporating this module into the communication system set up will allow the student to observe two test points and/or interrupt the 8 bits digital signal by disconnecting or dropping any bit in the code word as described below.

SIGNAL INTERRUPTER/SELECTOR

The main idea of the signal selector is self explained in the figure (a) below. The selection is done through pushing a button and a led will indicate the position of the selector. There is a single input to the module labeled as Inputs, and the outputs can be taken after the interrupter (8-bits +common) and/or from selectors (selector1 and selector2). In other words the module will allow you to plug in 8-bits data + common using the D-connector (9 pine connector) and provide multiple outputs known as parallel (from interrupter) and sequential (From selector1 or selector2) outputs.

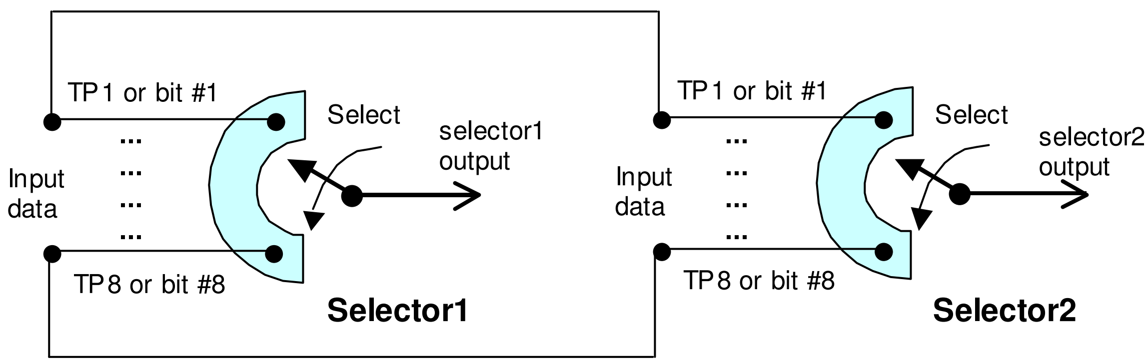


Figure (a) Block diagram of selecter1 and selecter2 internal connections and operation

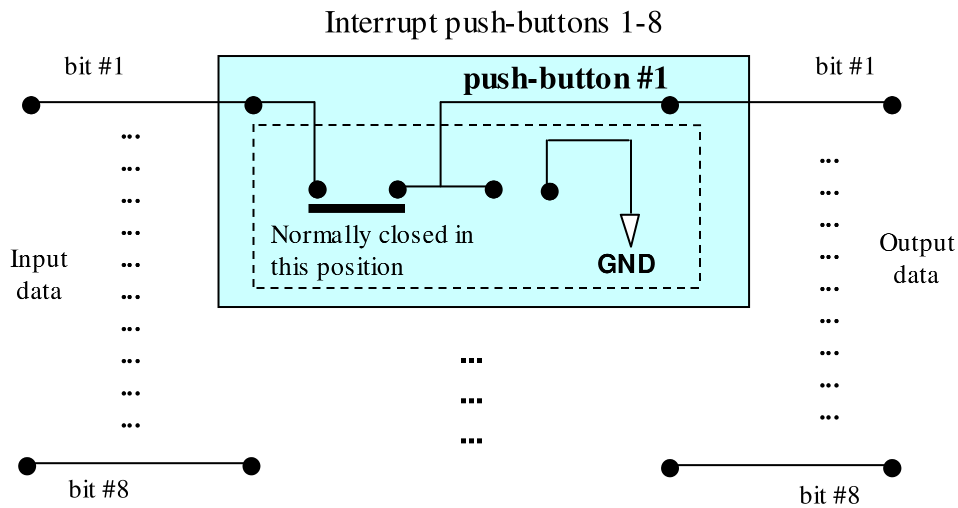


Figure (b) connection details of the push-button #1 in normal position (not depressed)

Figure (b) above shows the connection details of single push-button in the interrupter section of the module. Notice that depressing the push-button #1 will interrupt the connection for bit #1 and replaces the data bit by ground connection at the output. This means that if the data on connection #1 were logic 0 before interruption, you wouldn't observe any difference after this action.

III. PRELIMINARY REPORT

1) Explain the A/D and D/A conversions, PCM Encoding/Decoding and Companding processes.
2) What is the quantization? Derive the signal-to-quantization-noise-ratio (SQNR) for linear quantization. What are the effects of companding on the signal-to-quantization-noise-ratio?
3) Explain the concept of the PCM encoder (9444) and PCM decoder (9445). Please refer to the appendix.

IV. PROCEDURE

Equipment Required

Dual Function Generator	Sync. Audio Generator	Low Pass Audio Filter.
Signal Int/Selector	Noise Measurement Filter	Clock Generator.
PCM Encoder	PCM Decoder.	

1. Figure 4.1 shows the block diagram of the experimental setup used to observe the PCM encoding/decoding and the effects of companding on the reconstruction of a received signal. The 8.0 kHz CLK signal (square wave) is obtained from channel A of the Dual Function Generator (Attenuator: 0dB, Output Level: 1/4 turn CW).

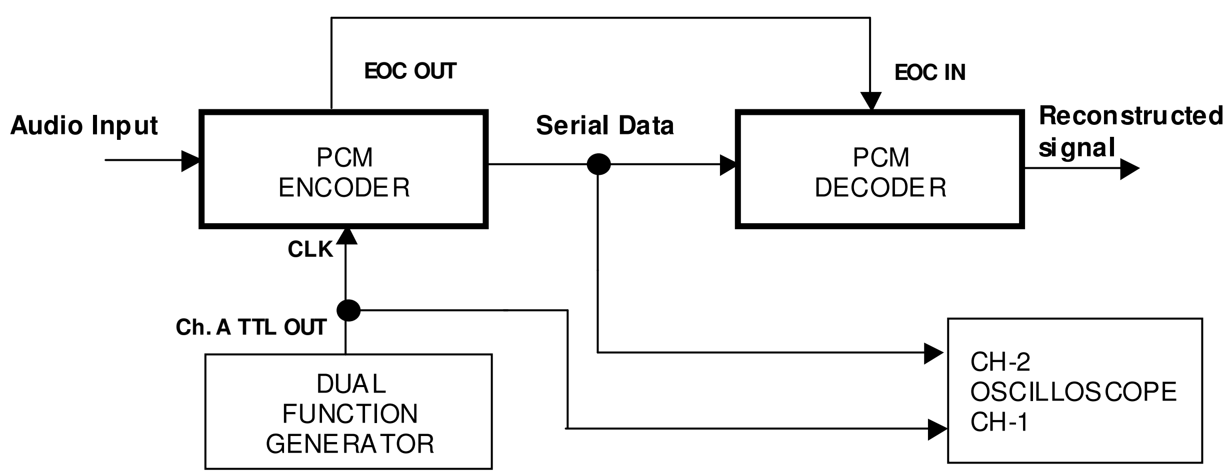


Figure 4.1

2. Apply a DC voltage (0.25V, 0.5V, and 0.75V) to the Audio Input of the PCM Encoder. Record the companding binary values at the serial output (of the PCM Encoder) for the following selections of Compression law: DIR, μ_2 and A_1. Explain the obtained results.

3. Apply a sine wave of 300 Hz, 0.75V p-p (from channel B of Dual Function Generator) to the Audio Input of the PCM encoder. Set the INPUT CODE of the PCM Decoder to “OFFSET”. Observe the original and reconstructed signals on the oscilloscope for DIR, μ_2, A_1. Explain the obtained results.

4. Apply a saw tooth wave of 100 Hz, 0.75V p-p to the Audio Input of the PCM Encoder. Set the INPUT CODE of the PCM Decoder to “SIGN” (signed binary). Observe the original and reconstructed signals on the oscilloscope for the following selections of compression law of the PCM Encoder: DIR, μ_2 and A_1. Explain the obtained results.

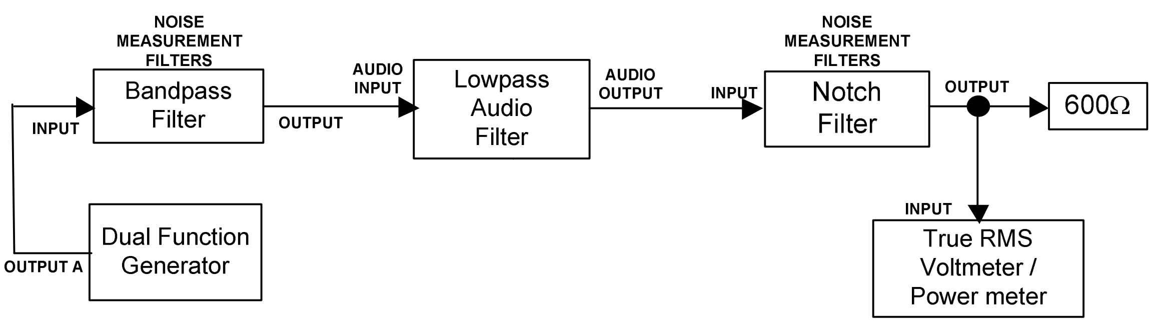
5.1 Measuring V_R and adjusting for filter optimum frequency:

The notch and bandpass filter sections of the Noise Measurement Filters have been calibrated to give the best performance as a pair, near the “Center Frequency” (100, 300 1k) Hz. For 300Hz, the typical range is between 290 and 310Hz. For this experiment the “300Hz” from the signal generator must be adjusted to the optimal frequency of the filters. *Each setup has a different optimum frequency.*

Connect the equipment as in the following diagram and set the following:

- Noise Measurement Filters: 300 Hz center frequency.
- Low pass Audio Filter: 4th order, 3.4 kHz cutoff frequency, “Gain” to Cal.
- Channel A of the Dual Function Generator for a 300Hz sine wave (Attenuator: 0dB, Output level: 1/4 turn CW).
- Select the 10mV range on the true RMS voltmeter to measure the output voltage of the Notch Filter. You can also monitor the signal on the scope to find the minimum.
- Slowly adjust the frequency of the audio input signal around 300 Hz until the reading on the true RMS voltmeter (or scope) is as small as possible.

Record the voltage read on the true RMS voltmeter, it is the residual noise voltage, V_R, of the signal components not removed by the filters. (V_R should be < 5mV)



Block Diagram of system used to measure the residual noise voltage (V_R)

5.2 Figure 4.2 shows the block diagram of the experimental setup used to measure the signal-to-quantization-noise-ratio (SQNR). The true RMS voltmeter / Power Meter, is used to measure the voltage of the reconstructed voltage signal and quantization noise, V_{SN} point 1. The NOTCH FILTER of the Noise Measurement Filters will remove the message signal leaving the quantization noise. The voltage measured at the output of the Notch Filter, V_{NT} (point 2) represents the total noise voltage. Therefore we can derive the following equations:

Signal Voltage: $V_s = [V_{SN}^2 - V_{NT}^2]^{1/2}$

Quantization noise voltage: $V_Q = [V_{NT}^2 - V_R^2]^{1/2}$

Signal to quantization noise ratio: $SQNR = 20\log(V_s/V_Q)$

<u>Note:</u> *<u>The optimum filter frequency is critical to obtaining good test results. If the frequency is accidentally changed, redo the frequency setting portion of step 5.1. The 1Hz accuracy of the generator counter is not enough to obtain the optimal frequency</u>.*

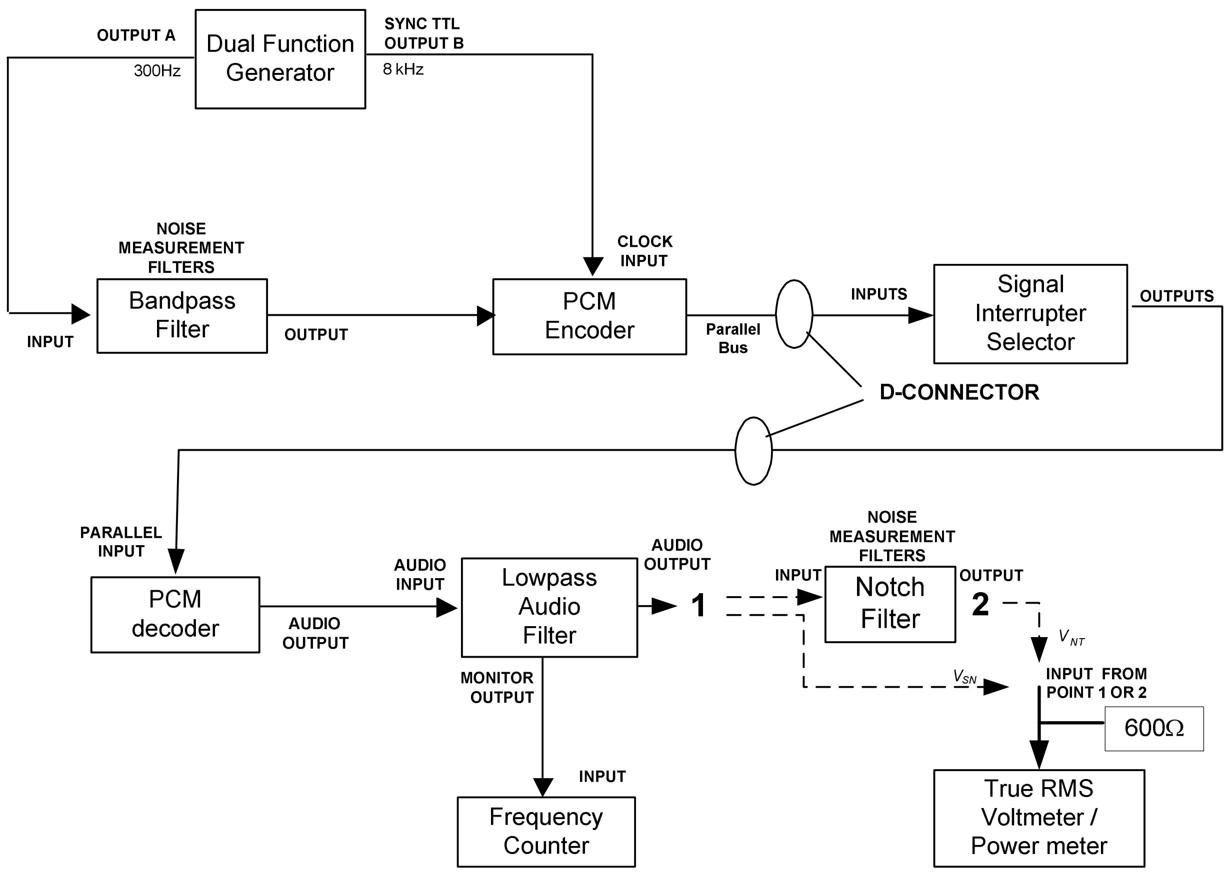


Figure 4.2 **Block Diagram of the setup to Measure SQNR**

Figure 4.2 setup:

- Set the compression law of the PCM encoder to DIR
- Set the PCM Decoder ‘s Input Code to OFFSET, Gain to Cal
- Set the low pass filter to 3.4kHz, 4^{th} order, Gain to Cal.
- Using the Power meter with the 600Ω termination, adjust the function generator output A level to get 0.650 Vrms at point 1 (Low Pass Audio filter output, Fig 2.2).

6. Use the Signal Interrupter/Selector module to remove codeword bits to simulate systems with 7, 6, 5, 4, 3 and 2 bits. Start with the least significant bit and work towards the most significant. For each case measure V_{SN} and V_{NT}, and compute SQNR using the value of V_R measured earlier. Compare this value to the theoretical SQNR.

7. Set the Compression Law (of the PCM Encoder) and INPUT CODE (of PCM Decoder) to A_l, and 8 bits/code. Vary the FunctionGenerator output to get the following voltages at point 1: 20, 50, 100, 200, 500, 650 mV, Measure and record V_{SN}, V_{NT}, V_R, and compute SQNR. Explain the obtained results.

References

[1] S. Haykin Digital Communications John Wiley & Sons inc., 1988, Chapters 6 and 7

[2] Staff of Lab Volt Inc., Vol 3: Modems and Data Transmission, Units land 2

[3] F.G. Stremler, Introduction to Communications Systems, 3rd edition, Addison-Wesley, 1990.

[4] Indirect FM/PM Generator Model 9414. Instruction Manual, Lab Volt Ltd., March 1988, Sections 3 and 4, pp 3-8.

[5] Direct FM Multiplex Generator Model 9413. Instruction Manual, Lab Volt Ltd., March 1988, Sections 3 and 4, pp. 8-9.

[6] FM/PM Receiver Model 9415-10. Instruction Manual 1988, Sections 3 and 4, pp. 3-10.

[7] Staff of Lab Volt Ltd., Analog Communications. Volume 3:FM/PM, 2nd Edition, Jan. 1989; Units 2 to 5: pp. 2-1 to 5-21; Unit 6: pp 6-1 to 6-15; Unit 7: pp., 7-1 to 7-31.

[8] AM/DSB/SSB Generator 9410 Instruction Manual, Lab Volt Ltd., June 1987, Section 4, pp. 3-10.

[9] Analog Communications, volume 2: AM/DSB/SSB, Lab Volt, Jan 1988, Unit 2: pp. 2-1 to 15, Unit 4: pp. 4-1 to 12, Unit 5: pp. 5-1 to 12.

[10] AM/DSB Receiver Model 9411 Instruction Manual, July 1987, Sections 3 and 4, pp. 2-8.

[11] SSB Receiver Model 9412 Instruction Manual, March 1988, Sections 3 and 4, pp. 3-7.

[12] Analog Communications Volume 2: AM/DSB/SSB, Lab Volt, Jan 1988, Unit 3: pp. 3-1 to 26, Unit 4: pp. 4-13 to 21, Unit 5: pp. 5-13 to 23.

APPENDIX

Equipment description, specifications and operating instructions:

- Spectrum Analyzer………………………………………………… 1
- Low pass audio filter……………………………………….…...18
- Noise measurement filter……………………………….……….21
- Clock Generator……………………………………………........26
- Signal Interrupter / Selector…………………………………......31
- PRBS (Pseudo-Random Binary Sequence) Generator……….....35
- BER (Bit Error Rate) Indicator……………………………….…42
- AM/DSB/SSB Generator…………………………………………..46
- AM/DSB Receiver……………………………………………........50
- Indirect FM/PM generator……………………………………....54
- FM/PM Receiver……………………………………………….......57
- PCM Encoder……………………………………………….........61
- PCM Decoder………………………………………………….........65
- PAM/ASK Generator…………………………………………..........69
- PAM/ASK Receiver…………………………………………………..73
- Tektronix TDA3000 oscilloscope…………………………….…75