2020-21

Communications Systems Fundamentals Laboratory

EE 333

Eng. Mashhour AL Tarayrah

Electrical Engineering Department

Major Topics covered and schedule in weeks:

Topic	Week #	Courses Covered
Introduction and Lab safety.	1	Introduction
Lab Familiarization and Introduction	2	EE 332
Fourier Series Using MATLAB Simulation	3	EE 332
Fourier Transform Using MATLAB Simulation	4	EE 332
AM Modulation	5	EE 332
DSB-SC Modulation and Demodulation Experiment	6	EE 332
AM demodulation	7	EE 332
SSBSC modulation and demodulation	8	EE 332
FM Modulation Experiment	9	EE 332
FM Demodulation Experiment	10	EE 332
Pulse Amplitude Modulation Experiment	11	EE 332
Quantization and coding Experiment (Sampling and reconstruction)	12	EE 332
Pulse Code Modulation Experiment	13	EE 332
Pulse Code demodulation Modulation Experiment	14	EE 332
Final Exam	15	

Specific Outcomes of Instruction (Lab Learning Outcomes):

1. An ability to run the communication lab measurement equipments **(6)**
2. An ability to Validate the theory of FS for periodic signals through MATLAB **(1,3,5)**
3. An ability to Use MATLAB to find the Fourier transform of non-periodic signals **(1,3,5)**
4. An ability to Simulate and perform DSB-SC and SSB modulation and demodulation **(1,3,5,6)**
5. An ability to Implement both AM modulator and demodulator **(1,3,5,6)**
6. An ability to Conduct the FM modulator and demodulator circuitry **(1,3,5,6))**
7. An ability to Perform Pulse Amplitude Modulation (PAM) experiment **(1,3,5,6)**
8. An ability to Investigate the sampling, quantization and coding processes of analogue signals **(1,3,5,6)**
9. An ability to Conduct Pulse Code Modulation (PCM) experiment **(1,3,5,6)**

Student Outcomes (SO) Addressed by the Lab:

z	Outcome Description General Engineering Student Outcomes	Contribution
1.	an ability to identify, formulate, and solve complex engineering problems by applying principles of engineering, science, and mathematics	M
2.	an ability to apply engineering design to produce solutions that meet specified needs with consideration of public health, safety, and welfare, as well as global, cultural, social, environmental, and economic factors	
3.	an ability to communicate effectively with a range of audiences	L
4.	an ability to recognize ethical and professional responsibilities in engineering situations and make informed judgments, which must consider the impact of engineering solutions in global, economic, environmental, and societal contexts	
5.	an ability to function effectively on a team whose members together provide leadership, create a collaborative and inclusive environment, establish goals, plan tasks, and meet objectives	L
6.	an ability to develop and conduct appropriate experimentation, analyze and interpret data, and use engineering judgment to draw conclusions	H
7.	an ability to acquire and apply new knowledge as needed, using appropriate learning strategies	

Goal

The goal of this laboratory is to study communication systems through experimentation. Upon completion of this course, students should be able to use standard laboratory equipment to analyze the behavior of basic communication systems and to design and construct simple communication experiments.

The purpose of the experiments described here is to acquaint the student with:

- Analog communication systems.
- Digital communication systems.
- Instruments & procedures for communication test & measurement.

The aim is to teach a practical skill that the student can use in his or her own experimental projects in digital and analog communications.

At the end of this course, the student should be able to:

- Design and build simple communication circuits of his or her own design.
- Use communication test & measurement instruments such as oscilloscopes, CASSY, etc.

General Guidelines

Attendance Policy:

Students are required to attend all lab sessions. A student who misses a lab will receive a grade of zero for the lab and any associated reports. No make-up labs will be given.

Homework Policy:

Every lab requires preparation prior to performing the experiments. Most of the labs require MATLAB simulations, and some of the labs require calculations of design parameters before beginning the experiment. Students are required to perform this preparatory work prior to coming to the lab. In order to fulfill the written communication component of the course, students are required to turn in pre-lab reports written in a proposal format prior to performing the laboratory work.

Pre-lab reports:

Each pre-lab report is due at the beginning of the lab period. The required report format is found in the lab manual. Each student is required to keep a copy of all pre-labs submitted. The pre-lab report should be used as the basis of the lab report, which will be written sometime after completion of the lab. In certain cases, an instructor may require the student to make corrections to simulations in the pre-lab reports, which will be due the following lab meeting. A 10-point deduction will be taken on every resubmission.

Lab reports:

Lab report must be completed independently. You can share only the collected data sets with your lab partner. Coping any part of the report from others is strictly prohibited and is against the college integrity policy. Lab reports are always due the next session after the lab is completed. Late report will be subjected to a penalty of 10% per day. Late reports will be accepted up to 3 days after the due date. No late reports are accepted after that. A student who misses a lab report will receive a grade of zero.

Safety

Safety is always an important topic whenever laboratory work is being considered, and it is certainly true in the case of EE 333 lab. Safety is important.

The experiments in the laboratory use low voltages and low currents. However, the lab equipment is powered by the 220V, 60Hz, line voltage. Be careful with the line voltages. Do not touch exposed prongs on the equipment plugs when connecting the equipment to the lines.

Take care when using power supplies, which may be low voltage but can supply currents in the ampere range. Shorting such a supply can lead to a serious burn as high currents arc and can ignite flammable material. This is precisely why a car battery needs to be treated with respect. The hundreds of amps a battery can supply are sufficient to cause serious burns.

The equipment is heavy enough to be generally stable on the bench. Be sure to keep the equipment away from the edges of the benches to avoid having a piece of equipment fall off the bench. Besides endangering people who might be struck, falling equipment endangers everyone in vicinity by stressing the power cords, possibly causing a line short or live fault on the equipment, not to mention damage to the expensive lab equipment. In general electronic equipment does not survive harsh treatment.

General Report Guidelines

A technical report is expected to contain the following items or subsections:

- Title page
- Introduction
- Theoretical discussion or background
- Experimental procedure and methodology and experimental results
- Discussion of results
- Conclusion or summary
- Acknowledgements
- Appendices
- References

You may find some reports have one or more of these sections removed.

Title page: This page is the first page of the report and should act as the cover page. The title will often be the same as the title given in the lab manual. The title is centered on the top half of the page and is written in bold type, all capitals. Centered on the bottom half of the page is your name, the name of your lab partner(s), the course name, the course title, the instructor or the name of the institution for whom the report is being prepared.

Introduction: The introduction should explain the background of the work. It should put the experimental work into perspective and should lead the reader into the subject matter. It should have at least one sentence explaining why the work was undertaken. It should end with one or two sentences describing the general experimental approach and results.

Theoretical discussion or background: This section is used to develop the theoretical aspects of the experiment. Any relevant theory from class or from the lab manual can be used.

Experimental Procedure and Methodology and Experimental Results: This section is for explanation of apparatus, circuit configuration, and procedures used in this experiment. The title will probably change from experiment-to-experiment. You can put drawings and circuit diagrams into this section.

Discussion of experimental results: This section is probably the most important part of the report. The data should be presented in a reduced form. Usually figures are the easiest form to present data, but tables or lists can be used, where appropriate. Do not assume that the reader knows what you are talking about. Be descriptive. Include sample calculations along with your calculated data. Do not present data or graphs without explanation. Be sure to compare your measured results with those that you expected. Here is where theory meets reality. If the results agree with the theory show how. If the results do not agree with the predicted values, try to explain why you think they are different or where any errors in data-taking could have occurred. Explain any anomalous data.

Conclusion-Summary: This section should provide closure to your report. Conclusions should be based on the information described in the report. The conclusions may not exactly match the lab's objectives, but make sure your conclusions are supported by your data. Any advantages and/or limitations of the information presented here should be included. You may want to include any personal observation that may not be reflected in the data, e.g., the problems encountered while using a particular instrument or when performing a particular step in the experiment.

Acknowledgements: This section is used to acknowledge any technical or financial aid that was received in support of this work. You should state who your lab partner was in this section.

Appendices: This section should contain any miscellaneous calculations, any mathematical derivations or proofs, and any computer programs or SPICE simulations.

References: This section contains all bibliographical work cited in the report. Usually you will reference your textbook and lab manual here. The format of these is

1. Author, *Title of reference,* page numbers, who published, where published, when published.
1. B. Grob, *Basic Electronics,* pg. 43 to 435, McGraw Hill, New York, NY, 1943.

Week 1: EXPERIMENT 1
INTRODUCTION TO THE COMMUNICATION LAB

OBJECTIVES:

The goals of the communication laboratory are:

1. To allow you to perform experiments that demonstrates the theory of signals and communication systems that discussed in the lecture course.
2. To introduce you to some electronic equipments that make up the communication system.
3. To familiarize you with proper laboratory procedure; this include precise record keeping, logical troubleshooting, and learning the capabilities as well as the limitations of your measuring equipments.

Test Standard :IEEE 802

GENERAL LABORATORY PROCEDURE:

The most important rule to follow in any laboratory is: *Think before you do anything*. If you follow this one rule you will avoid injury to yourself, damage to the system you are testing, damage to your measurement equipments, and you will not waste time going down dead-ends streets.

Safety: In general you will not be using voltage level high enough to cause injury; nevertheless, you should always pay attention to what you are doing.

Circuit damage: your voltage levels can cause damage to the circuit under test if you are not carful. Make sure your circuit diagram is correct. If you need to make changes to the circuit, disconnect (turn off) the power supply and the input signal.
Measurement equipment: each lab station has the following permanent equipment that you will use for most of the labs.

- CASSY: Sensor-CASSY is an interface device for recording measurement data, along with CASSY software, the time domain and frequency domain measurements are applicable.
- Oscilloscope: is a typical electronic test instrument that allows the observation of constantly varying signals, usually as a two-dimensional graph of one or more electrical potential differences. Using the vertical or 'Y' axis, the signal is plotted as a function of time, (horizontal or 'x' axis). Although an oscilloscope displays voltage on its vertical axis, any other quantity that can be converted to a voltage can be displayed as well. In most instances, oscilloscopes show events that repeat with either no change or change slowly.
- DMM: stands for digital multi-meter, a typical multi-meter include features such as the ability to measure voltage, current and resistance.

PROCEDURE:

PART A – Experiment setup + pulse train:

Connect the experiment as shown in figure 1.

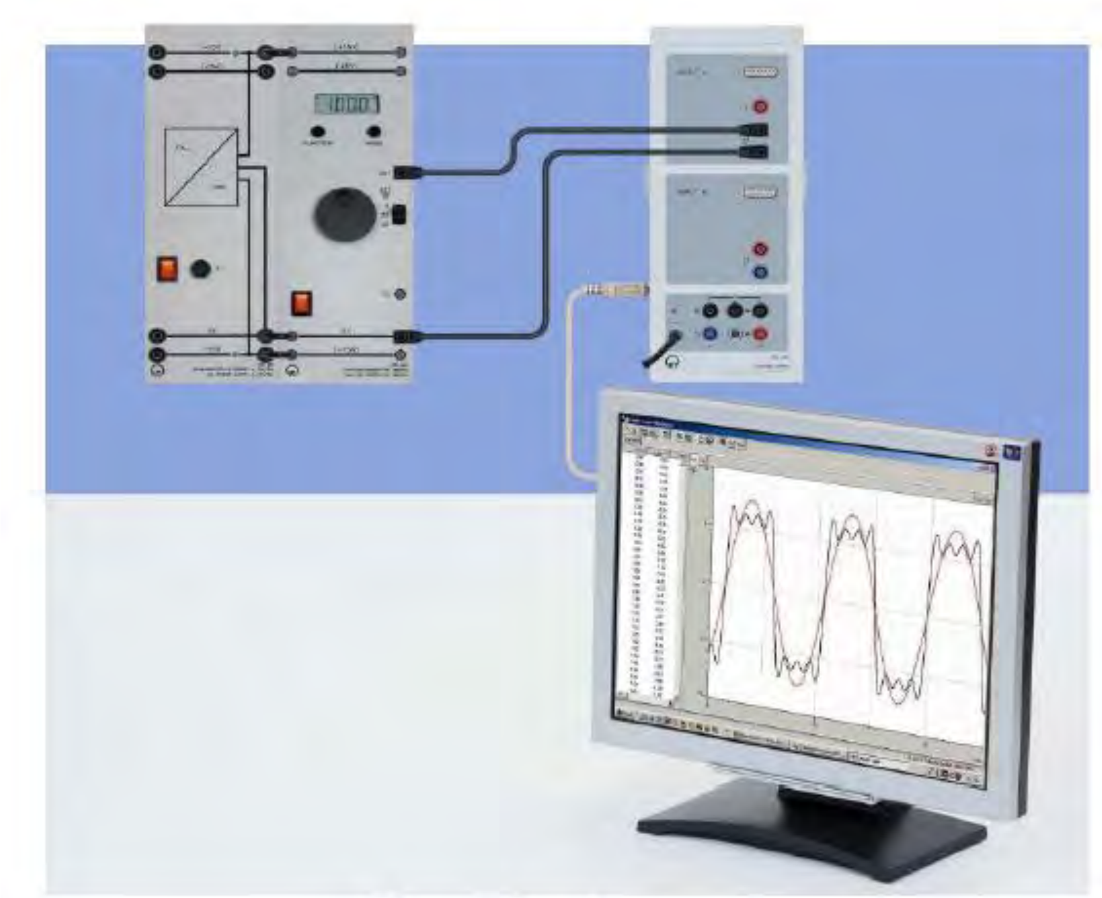


Figure 1: Experiment setup.

1. Select a pulse train at the function generator with f_p = 1 kHz, pulse amplitude A_P = 5 V (10 V_{pp}) and duty cycle t_1/T_P = 1/10.
2. Set the measurement parameters so that you can see a clear time and frequency response of the input signal (Clock generator).
3. Display the time characteristic of the pulse train.
4. Display the spectrum content of the input signal.
5. Where are in general the zero crossings in the envelope of the pulse spectrum?
6. How many spectral lines *l* arise between two zero crossings of the envelope (sync function)?
7. Repeat the measurement of the spectra and time characteristics for the same pulse frequency $f_P = 1$ kHz and pulse amplitude A_P for different duty cycles $\tau_2/T_P = 2/10$, $\tau_3/T_P = 3/10$, $\tau_5/T_P = 6/10$ and $\tau_6/T_P = 9/10$. Proceed as described above.
8. Why do pulse trains require large transmission bandwidths?

9. What is the structure of the spectrum of a pulse train?
10. What kind of characteristic curve is the envelope curve of the pulse spectrum?
11. Find the complex Fourier coefficients for triangular signal?

PART B – Sinusoidal signal:

1. Select a sinusoidal input signal at the function generator with f_p = 1 kHz, pulse amplitude A_P = 10 V (10 V_{pp}).
2. Repeat the steps from 1 to 4 in part A.
3. Change the input signal frequency and record your results.

PART C – Triangular signal:

1. Select a triangular input signal at the function generator with f_p = 0.5 kHz, pulse amplitude A_P = 10 V (10 V_{pp}).
2. Repeat the steps from 1 to 4 in part A.
3. Change the input signal frequency and record your results.

Week 2: Experiment 2-Part A - An introduction to the NI ELVIS II test equipment

Test Standard :IEEE 802

Preliminary discussion

The digital multimeter and oscilloscope are probably the two most used pieces of test equipment in the electronics industry. The bulk of measurements needed to test and/or repair electronics systems can be performed with just these two devices.

At the same time, there would be very few electronics laboratories or workshops that don't also have a DC Power Supply and Function Generator. As well as generating DC test voltages, the power supply can be used to power the equipment under test. The function generator is used to provide a variety of AC test signals.

Importantly, NI ELVIS II has these four essential pieces of laboratory equipment in one unit (and others). However, instead of each having its own digital readout or display (like the equipment pictured), NI ELVIS II sends the information via USB to a personal computer where the measurements are displayed on one screen.

On the computer, the NI ELVIS II devices are called "virtual instruments". However, don't let the term mislead you. The digital multimeter and scope are real measuring devices, not software simulations. Similarly, the DC power supply and function generator output real voltages.

The experiments in this manual make use of all four NI ELVIS II devices and others so it's important that you're familiar with their operation.

The experiment

This experiment introduces you to the NI ELVIS II digital multimeter, variable DC power supplies (there are two of them), oscilloscope and function generator. Importantly, the oscilloscope can be a tricky device to use if you don't do so often. So, this experiment also gives you a procedure that'll set it up ready to display a stable 2kHz 4Vp-p signal every time. Importantly, it's recommended that you use this procedure as a starting point for the other experiments in this manual.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads

Some things you need to know for the experiment

This box contains definitions for some electrical terms used in this experiment. Although you've probably seen them before, it's worth taking a minute to read them to check your understanding.

The amplitude of a signal is its physical size and is measured in volts (V). It is usually measured either from the middle of the waveform to the top (called the peak voltage) or from the bottom to the top (called the peak-to-peak voltage).

The period of a signal is the time taken to complete one cycle and is measured in seconds (s). When the period is small, it is expressed in milli seconds (ms) and even micro seconds (J.s).

The frequency of a signal is the number of cycles every second and is measured in hertz (Hz). When there are many cycles per second, the frequency is expressed in kilo hertz (kHz) and even mega hertz (MHz).

A sinewave is a repetitive signal with the shape shown in Figure 1.

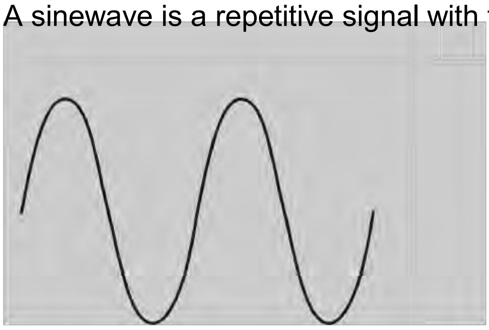


Figure 1

A squarewave is a repetitive signal with the shape shown in Figure 2.

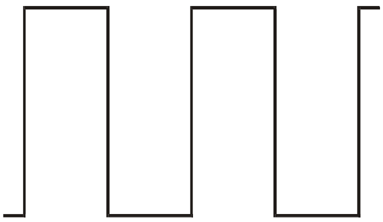


Figure 2

Procedure

Part A - Getting started

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to Manual.

4. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

5. Turn on the PC and let it boot-up.

6. Launch the NI ELVISmx software per the instructor's directions.

   Note: If the NI ELVISmx software has launched successfully, the window called "ELVIS - Instrument Launcher" will be visible (see Figure 3).

Figure 3

Part B - The NI ELVIS II Digital Multimeter

The NI ELVIS II Digital Multimeter (DMM) is an instrument that can measure the following electrical properties: DC & AC voltages, DC & AC currents, resistance, capacitance and inductance. Its operation is briefly introduced next.

7. Use the mouse to click on the "DMM" button on the NI ELVISmx Instrument Launcher.

   Note: If the digital multimeter virtual instrument has launched successfully, the instrument's window will be visible (see Figure 4).

Figure 4

The digital multimeter's measurement options are selected using the Measurement and Settings controls on the virtual instrument (near the mouse-pointer in Figure 4).

8. Move the mouse-pointer over these controls but don't click on any of them yet.

   Note: As you do this, you'll notice that a pop-up appears to tell you by name what measurement mode the controls activate.

9. Click back and forth between one of the Voltage controls (marked V) and one of the Current controls (marked A).

   Note 1: As you do, notice that the buttons on the virtual instrument are animated. The selected control fades as though it has been physically pressed in.

   Note 2: Notice also that the Banana Jack Connections window updates to tell you which of the DMM's banana jacks to use on the left side of the NI ELIVS II for that particular measurement.

Importantly, simply launching the DMM virtual instrument doesn't activate the instrument's hardware. This must be done every time the DMM virtual instrument is launched using its Run control (the button with the green arrow).

10. Click on the DMM's Run control.

11. Click on each of the Measurement and Settings controls in turn while watching the DMM's readout.

    Note: As you do, notice that the readout updates to tell you the unit of measurement (eg V for volts, A for amps, etc). The readout also indicates the relative size of the measurement (for example, m for milli, M for mega, etc). See the instructor for more information if you're not familiar with the metric system of multiples and sub-multiples.

    Question 1
    Given you've not been asked to connect the digital multimeter's inputs to anything yet, why does the DMM read very small values of voltage and current instead of zero?

    ____

    ____

If you examine the DMM virtual instrument closely you'll notice that there are other settings on the DMM virtual instrument that can be adjusted including the Mode, Null Offset and Acquisition Mode. These controls default to appropriate settings for regular use so we'll not discuss them further here. Where adjustment of these controls is necessary, they'll be explained at the appropriate place in the experiments.

Part C - The NI ELVIS II Variable Power Supplies

The NI ELVIS II Variable Power Supplies (VPS) is an instrument that can simultaneously output two DC voltages (one positive and one negative) to terminals on the Emona DATEx. Its operation is briefly discussed next.

12. Use the mouse to click on the "VPS" button on the NI ELVISmx Instrument Launcher.

    Note 1: Don't close the NI ELVISmx DMM virtual instrument because you'll be using it to verify the operation of the Variable Power Supplies.

    Note 2: If the Variable Power Supplies virtual instrument has launched successfully, the instrument's window will be visible (see Figure 5).

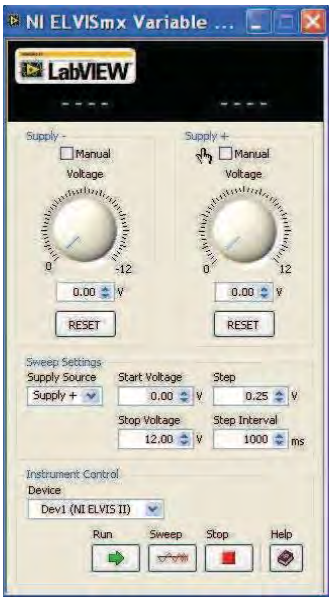


Figure 5

13. Put the NI ELVIS II Variable Power Supplies into Manual mode by checking the boxes next to the word Manual on both the positive and negative sides of the virtual instrument.

    Tip: One of the boxes is near the mouse-pointer in Figure 5.

    Note: Once you've performed this step, you'll notice that the virtual controls fade. This tells you that the virtual power supplies' outputs are controlled manually using the controls on the top right of the NI ELVIS II (directly below the USB Ready & Active indicators).

14. Click on the  DMM's control to put the unit into DC voltage measuring mode.

15. Set the two Variable Power Supplies Voltage controls to about half of their travel.

16. Connect the set-up shown in Figure 6 below.

Tip: Use the 4mm banana plug to 2mm banana plug patch lead.

Note: As you perform this step, you should see some activity on the DMM virtual instrument and the measurement on its readout change to about 6V.

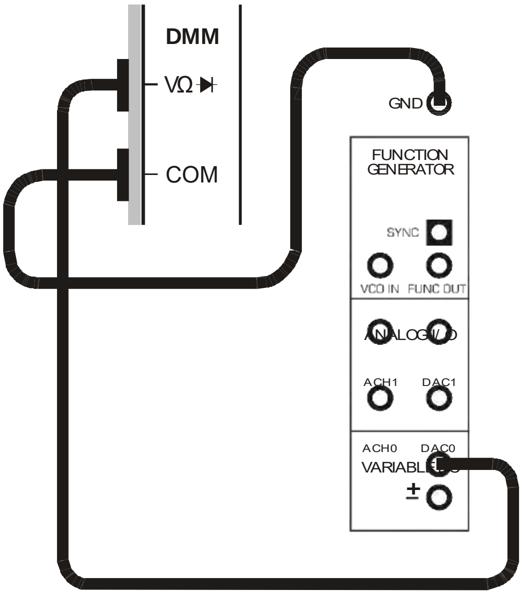


Figure 6

17. Use the Voltage control to determine the Variable Power Supplies' minimum and maximum positive output voltages. Record these in Table 1 below.

18. Connect the DMM to the Variable Power Supplies' negative output and repeat Step 18.

Table 1	Minimum output voltage	Minimum output voltage
Positive (+) output		
Negative (-) output		

While the DMM can be used for measuring the Variable Power Supplies' outputs, the instrument can monitor its own outputs freeing the digital multimeter for other uses. The next steps demonstrate this.

19. Check the box at the top of the NI ELVIS II Variable Power Supplies' virtual instrument shown in Figure 7 below.

Figure 7

20. Vary the Variable Power Supplies' negative Voltage control and compare the values on the displays of the two virtual instruments - they should be the same.

Part D - The NI ELVIS II Oscilloscope

The NI ELVIS II Oscilloscope (or just "scope") is a fully functional dual channel oscilloscope that allows engineers and technicians to measure AC waveforms and view their shape. Its operation is briefly discussed next.

21. Close the virtual instruments for the digital multimeter and Variable Power Supplies.

22. Use the mouse to click on the "Scope" button on the NI ELVISmx Instrument Launcher.

    Note: If the scope virtual instrument has launched successfully, the instrument's window will be visible (see Figure 8).

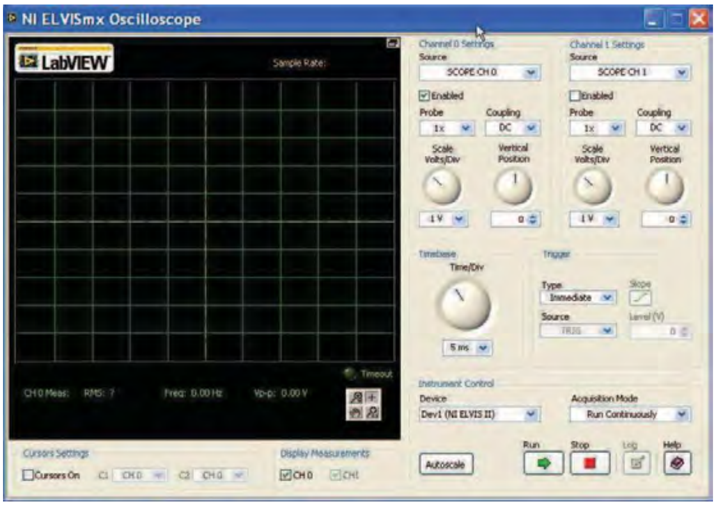


Figure8

The NI ELVIS II Oscilloscope is operated using the controls on its virtual instrument. Although operating the NI ELVIS II Oscilloscope is much easier than operating other types of scopes, it can still be a little tricky to use when you're new to this piece of test equipment. The procedure on the next page is one that you can use to set it up ready to reliably view waveforms and take measurements when undertaking DATEx experiments

Procedure for setting up the NI ELVIS II Oscilloscope

23. Follow the procedure below. Call the instructor for assistance if you can't find a particular control.
Note: Much of this procedure simply involves checking that control settings are in the default positions used by the NI ELVIS II Oscilloscope at the time of writing this manual.

General
i) Check that the Cursors On box is doesn't have a tick in it.

Vertical
i) Check that the Channel 0 Source control is set to SCOPE CH 0 and the Channel 1 Source control is set to SCOPE CH 1.
ii) Check that the Probe control for both channels is set to 1x.

iii) Set the Coupling control for both channels to AC.

iv) Check that the Scale Volts/Div control for both channels is set to 1V/div.

v) Check that the Vertical Position control for both channels is in the middle of their travel.

Timebase
i) Set the Time/Div control to the 500J.s/div position.

Trigger
i) Set the Type control to Edge.

ii) Set the Source control to CH 0 Source.

iii) Check that the Level control is set to 0.

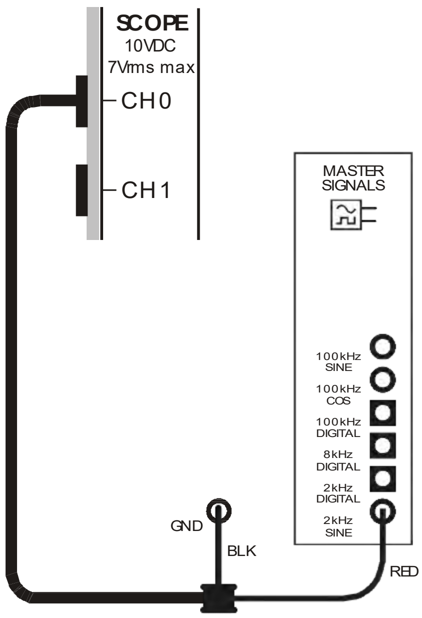
iii) Check that the Slope control is set to the ⎯╱⎺ position.

iv) Activate the scope's hardware by clicking on its Run control

The next part of this experiment lets you familiarise yourself with NI ELVIS II Oscilloscope by observing and measuring a DATEx signal.

24. Connect the set-up shown in Figure 9 below.

Note: Notice that the connection to the Master Signals' 2kHz SINE output must be made with the red banana plug. The black banana plug should be connected to any one of the ground (GND) sockets on the Emona DATEx.



When measuring the amplitude of an AC waveform using a scope, it's common to measure its peak-to-peak voltage. That is, the difference between its lowest point and its highest point. This is shown in Figure 10. Importantly, knowing the waveform's peak-to-peak voltage allows us to calculate its RMS voltage where required.

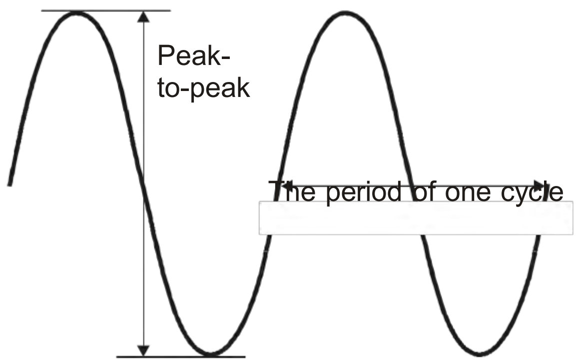


Figure 10

The other dimension of an AC waveform that's important to measure is its period. The period is the time it takes to complete one cycle and this is also shown in Figure 10. While knowing the waveform's period may be useful in its own right, it also allows us to calculate the signal's frequency using the equation:

$$f = \frac{1}{\text{Period}}$$

Measuring the amplitude of signals and determining their frequency using conventional scopes is a little more involved that using a digital multimeter. As such, it can be easy for the novice to make mistakes. Helpfully, the NI ELVIS II Oscilloscope includes meters that measure voltage and frequency for you and readout the information on the display. The location of this information on the virtual instrument is below the graticule as shown in Figure 11 below.

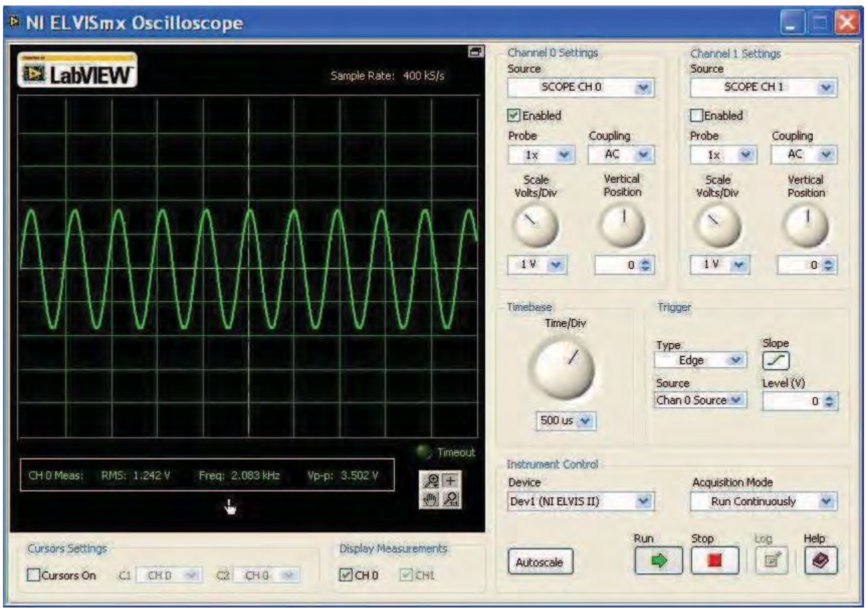


Figure 11

1. Record the scope's measured values for voltage (RMS and peak-to-peak) and frequency in Table 2 below.

2. Use the signal's frequency to work backwards to calculate and record its period.

   Tip: You'll have to transpose the equation on the previous page to make period (P) the subject.

Table 2

RMS voltage	
Frequency	
Pk-Pk voltage	
Period	

Part E - The NI ELVIS II Function Generator

The NI ELVIS II Function Generator (FGEN) is an instrument that can output AC signals of various shapes and at various frequencies to terminals on the Emona DATEx. Its operation is briefly discussed next.

25. Use the mouse to click on the "FGEN" button on the NI ELVISmx Instrument Launcher.

    Note 1: Don't close the NI ELVISmx Scope virtual instrument because you'll be using it to verify the operation of the function generator.

    Note 2: If the function generator virtual instrument has launched successfully, the instrument's window will be visible (see Figure 12).

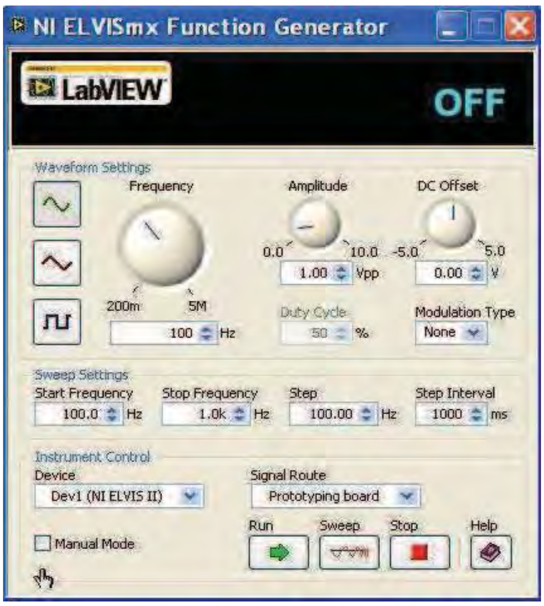


Figure 12

26. Check the box that puts the function generator into Manual Mode.

Note: Once you've performed this step, you'll notice that the virtual controls fade. This tells you that the function generator's output is now a sinewave whose amplitude and frequency are controlled manually by the Frequency and Amplitude controls on the right side of the NI ELVIS II.

27. Set the function generator's Amplitude control on the right side of the NI ELVIS II to about half its travel.

28. To observe the function generator's output, connect the set-up shown in Figure 13 below.

Note: Again, the connection to the function generator's output must be made using the lead's red banana plug.

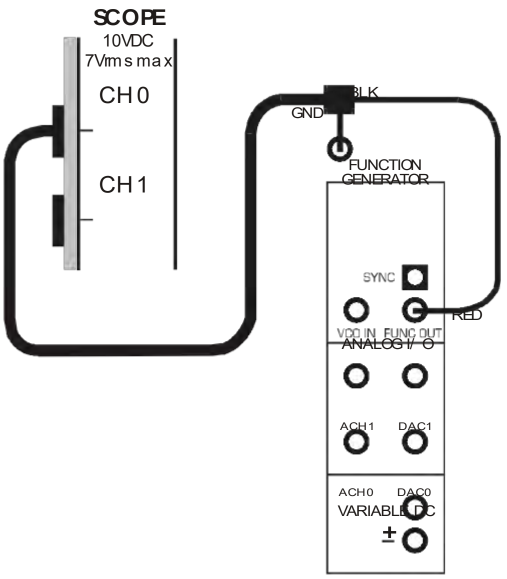


Figure 13

29. Adjust the scope's Timebase control for 1ms/div.

Note: Once this step has been performed, you should see one complete cycle of a sinewave.

30. Vary the function generator's Amplitude control left and right and observe the effect on the function generator's output.

31. Determine the function generator's manually adjustable minimum and maximum output voltages and record your measurements in Table 3 below.

Table 3

Minimum output V	
Maximum output V	

32. Vary the function generator's Frequency control on the right side of the NI ELVIS II and observe the effect on the function generator's output.

Note 1: If this control doesn't seem to have an effect, keeping turning it.

Note 2: This control can vary the frequency of the function generator's output between 0.2Hz and 5MHz. However, it would take a lot of turns of the manual Frequency control to sweep between them. Experiment 3 introduces you to the controls on the function generator's virtual instrument which are more convenient to use.

Experiment 2-Part B -Using the Emona DATEx to model equations

Preliminary discussion

This may surprise you, but mathematics is an important part of electronics and this is especially true for communications and telecommunications. As you'll learn, the output of all communications systems can be described mathematically with an equation.

Although the math that you'll need for this manual is relatively light, there is some. Helpfully, the Emona DATEx can model communications equations to bring them to life.

The experiment

This experiment will introduce you to modelling equations by using the Emona DATEx to implement two relatively simple equations.

Something you need to know for the experiment

This box contains the definition for an electrical term used in this experiment. Although you've probably seen it before, it's worth taking a minute to read it to check your understanding.

When two signals are 180° out of phase, they're out of step by half a cycle. This is shown in Figure 1 below. As you can see, the two signals are always travelling in opposite directions. That is, as one goes up, the other goes down (and vice versa).

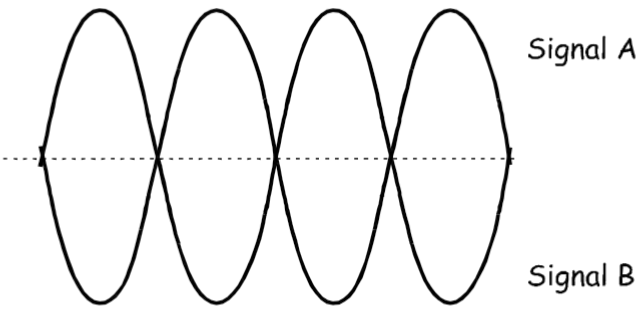


Figure 14

Procedure

In this part of the experiment, you're going to use the Adder module to add two electrical signals together. Mathematically, you'll be implementing the equation:

Adder module output = Signal A + Signal B

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELVISmx software.

8. Launch and run the NI ELVIS II Oscilloscope virtual instrument (VI).

9. Set up the scope per the procedure in Experiment 1 (page 1-12) ensuring that the Trigger Source control is set to CH 0.

10. Launch the DATEx soft front-panel (SFP).

11. Check you now have soft control over the DATEx by activating the PCM Encoder module's soft PDM/TDM control on the DATEx SFP.

    Note: If you're set-up is working correctly, the PCM Decoder module's LED on the DATEx board should turn on and off.

12. Locate the Adder module on the DATEx SFP and set its soft G and g controls to about the middle of their travel.

13. Connect the set-up shown in Figure 2 below.

    Note: Although not shown, insert the black plugs of the oscilloscope leads into a ground (GND) socket.

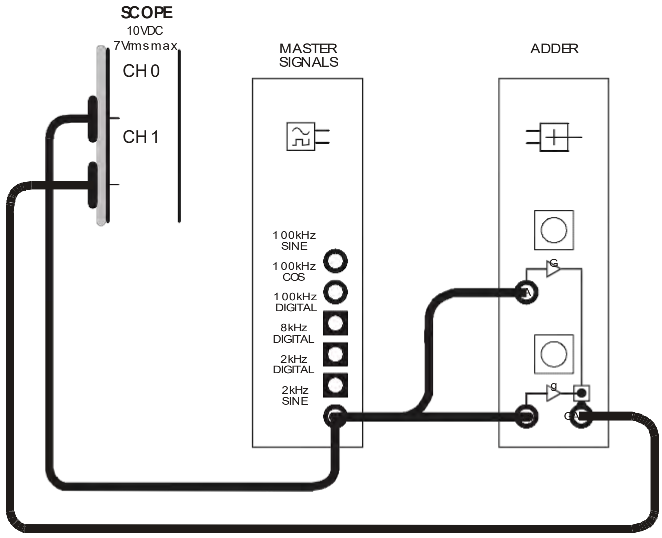


Figure 15

This set-up can be represented by the block diagram in Figure 3 below.

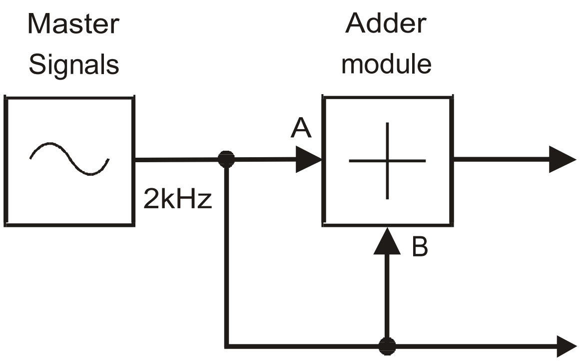


Figure 16

14. Adjust the scope's Timebase control to view two or so cycles of the Master Signals module's 2kHz SINE output.

15. Measure the amplitude (peak-to-peak) of the Master Signals module's 2kHz SINE output. Record your measurement in Table 4 on the next page.

16. Disconnect the lead to the Adder module's B input.

17. Activate the scope's Channel 1 input by checking the Channel 1 Enabled box to observe the Adder module's output as well as its input.

18. Adjust the Adder module's soft G control until its output voltage is the same size as its input voltage (measured in Step 15).

    Note 1: This makes the gain for the Adder module's A input -1.

    Note 2: Remember that you can use the keyboard's TAB and arrow keys for fine adjustment of the DATEx SFP's controls.

19. Adjust the scope's Timebase control to view two or so cycles of the Master Signals module's 2kHz SINE output.

20. Measure the amplitude (peak-to-peak) of the Master Signals module's 2kHz SINE output. Record your measurement in Table 4 on the next page.

21. Reconnect the lead to the Adder module's B input.

22. Disconnect the lead to the Adder module's A input.

23. Adjust the Adder module's soft g control until its output voltage is the same size as its input voltage (measured in Step 15).

    Note: This makes the gain for the Adder module's B input -1 and means that the Adder module's two inputs should have the same gain.

24. Reconnect the lead to the Adder module's A input.

The set-up shown in Figures 3 and 4 is now ready to implement the equation:

Adder module output = Signal A + Signal B

Notice though that the Adder module's two inputs are the same signal: a 4Vp-p 2kHz sinewave. So, for these inputs the equation becomes:

Adder module output = 4Vp-p (2kHz sine) + 4Vp-p (2kHz sine)

When the equation is solved, we get:

Adder module output = 8Vp-p (2kHz sine)

Let's see if this is what happens in practice.

25. Measure and record the amplitude of the Adder module's output.

Table 4

Input voltage	Output voltage

Question 2
Is the Adder module's measured output voltage exactly 8Vp-p as theoretically predicted?

Question 3
What are two reasons for this?

In the next part of the experiment, you're going to add two electrical signals together but one of them will be phase shifted. Mathematically, you'll be implementing the equation:

Adder module output = Signal A + Signal B (with phase shift)

26. Locate the Phase Shifter module on the DATEx SFP and set its soft Phase Change control to the 0° position.
27. Set the Phase Shifter module's soft Phase Adjust control about the middle of its travel.
28. Connect the set-up shown in Figure 4 below.

Note: Insert the black plugs of the oscilloscope leads into a ground (GND) socket.

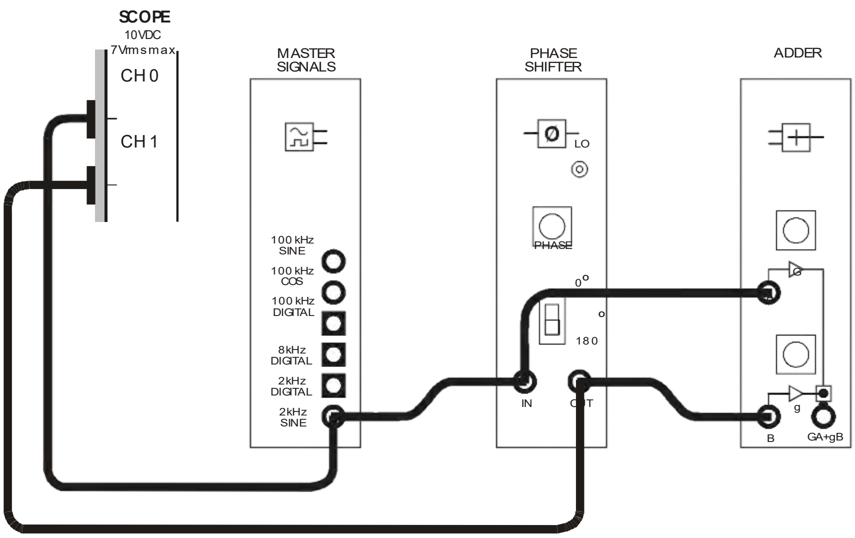


Figure 17

This set-up can be represented by the block diagram in Figure 5 on the next page.

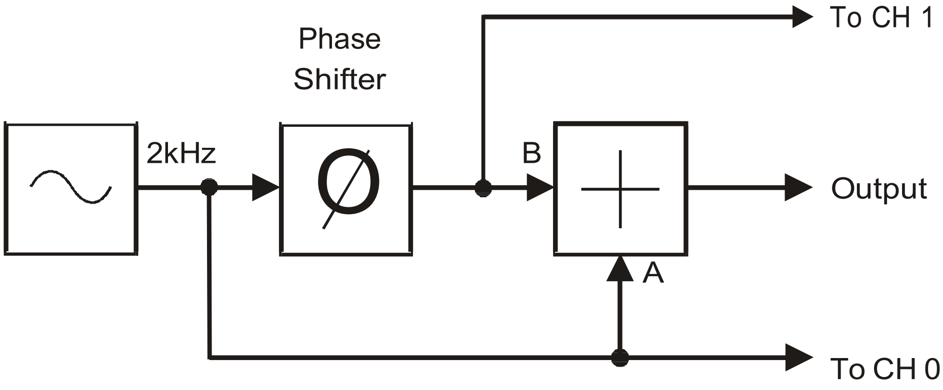


Figure 18

The set-up shown in Figures 4 and 5 is now ready to implement the equation:

Adder module output = Signal A + Signal B (with phase shift)

The Adder module's two inputs are still the same signal: a 4Vp-p 2kHz sinewave. So, with values the equation is:

Adder module output = 4Vp-p (2kHz sine) + 4Vp-p (2kHz sine with phase shift)

As the two signals have the same amplitude and frequency, if the phase shift is exactly 180° then their voltages at any point in the waveform is always exactly opposite. That is, when one sinewave is +1V, the other is -1V. When one is +3.75V, the other is -3.75V and so on. This means that, when the equation above is solved, we get:

Adder module output = 0Vp-p

Let's see if this is what happens in practice.

29. Adjust the Phase Shifter module's soft Phase Adjust control until its input and output signals look like they're about 180° out of phase with each other.

30. Disconnect the scope's Channel 1 lead from the Phase Shifter module's output and connect it to the Adder module's output.

31. Adjust Channel 1's Scale control to resize the signal on the display.

32. Measure the amplitude of the Adder module's output. Record your measurement in Table 5 below.

Table 5

Output voltage

Question 4

What are two reasons for the output not being 0V as theoretically predicted?

The following procedure can be used to adjust the Adder and Phase Shifter modules so that the set-up has a null output. That is, an output that is close to zero volts.

33. Use the keyboard's TAB and arrow keys to vary the Phase Shifter module's soft Phase Adjust control left and right a little and observe the effect on the Adder module's output.

34. Use the keyboard to make the necessary fine adjustments to the Phase Shifter module's soft Phase Adjust control to obtain the smallest output voltage from the Adder module.

Question 5

What can be said about the phase shift between the signals on the Adder module's two inputs now?

____

35. Use the keyboard to vary the Adder module's soft g control left and right a little and observe the effect on the Adder module's output.

36. Use the keyboard to make the necessary fine adjustments to the Adder module's soft g control to obtain the smallest output voltage.

Question 6

____

You'll probably find that you'll not be able to null the Adder module's output completely. Unfortunately, real systems are never perfect and so they don't behave exactly according to theory. As such, it's important for you to learn to recognise these limitations, understand their origins and quantify them where necessary.

Week 3: EXPERIMENT 3 FOURIER SERIES With 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 their applications in communication theory.
2. Learn how to implement Fourier series technique using MATLAB.

Test Standard :IEEE 802

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 Engr. 101). 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) 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')
```

a) How do you execute the file from the MATLAB prompt?
b) Suppose the file name was "example.cat". Would it run? How should you change it to make it work in MATLAB?
c) 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 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_0 when the signal set $\{e^{j2\pi nt/T_0}\}_{n=-\infty}^{\infty}$is employed as the basis for the expansion. With this basis, any given periodic signal *x*(*t*) with period T_0 can be expressed as:

$$x(t) = \sum_{n=-\infty}^{\infty} x_n \, e^{j2\pi nt/T_0}$$

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

$$x_n = \frac{1}{T_0}\int_0^{T_0} x(t) e^{-j2\pi nt/T_0} dt$$

This type of Fourier series is called the complex exponential Fourier series. The frequency $f_0 = 1/T_0$ is called the fundamental frequency of the periodic signal. The n_{th}harmonic is given by the frequency $f_n = nf_0$.

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:

$$|x_n| = |x_{-n}|$$
$$\angle x_n = -\angle x_-$$

PROCEDURE:

PART A – Using MATLAB:

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_0 to be used should be 32, the period T_0 is π, and the step size is $t_s = T_0 / N_0$. 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=-N0/2+1:N0/2-1
stem(k, Xnmag(No/2+1:length(Xn)-No/2))
stem(k,Xnangle(No/2+1:length(Xn)-No/2))
```

- plot the pulse train, and obtain the Fourier expansion coefficient.

Week 4:EXPERIMENT 4

FOURIER TRANSFORM Using MATLAB Simulation

OBJECTIVES:

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

3. To gain a good understanding and practice with Fourier Transform, and their applications in communication theory.
4. Learn how to implement Fourier Transform technique using MATLAB.

Test Standard :IEEE 802

FOURIER TRANSFORM:

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:

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

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

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

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 Parserval's relation.

USING MATLAB:

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.

1. First, introduce a certain time delay in the function, and notice what happens to the amplitude spectra. Explain why?
2. Next, introduce different scaling factors and comment on what you are observing.

3. 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.

Now, consider the $x_1(t)$ and $x_2(t)$ signals and described as follows:

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

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

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);
```

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

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

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

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)
```

Week 5: Experiment 5 - Amplitude modulation

Test Standard :IEEE 802

Introduction:

In an amplitude modulation (AM) communications system, speech and music are converted into an electrical signal using a device such as a microphone. This electrical signal is called the message or baseband signal. The message signal is then used to electrically vary the amplitude of a pure sinewave called the carrier. The carrier usually has a frequency that is much higher than the message's frequency.

Figure 1 below shows a simple message signal and an unmodulated carrier. It also shows the result of amplitude modulating the carrier with the message. Notice that the modulated carrier's amplitude varies above and below its unmodulated amplitude.

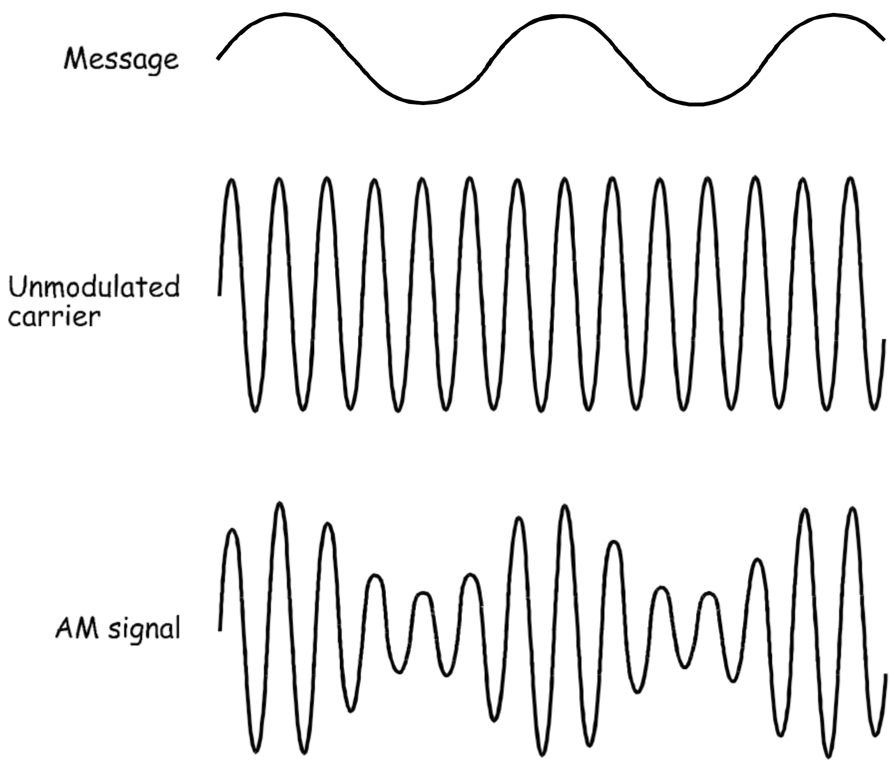


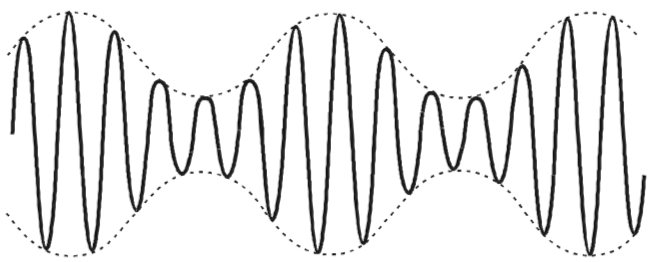
Figure 1

Experiment 5 - Observations of AM and DSBSC signals in the frequency domain

Preliminary discussion

Experiments 5 and 6 use the Emona DATEx to demonstrate the differences you would see on a scope between the output signals of an AM and DSBSC modulator. To refresh your memory, Figure 1 below shows the AM and DSBSC signals that would be produced by identical inputs (for example, a 1kHz sinewave for the message and a 100kHz sinewave for the carrier).

AM signal



DSBSC signal

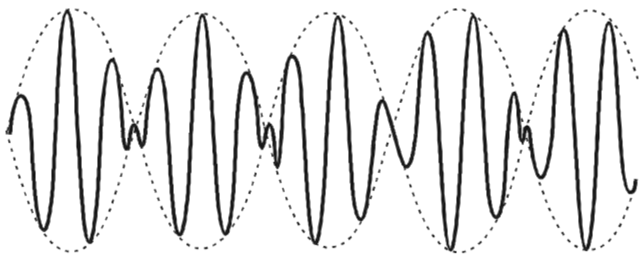


Figure 1

The two signals look different because they contain different sinewaves. That is, they have a different spectral composition. The reason for this is explained by the mathematical models of AM and DSBSC. Side-by-side, it's easy to see that the equations are a little different.

AM = (DC + message) x the carrier

DSBSC = the message x the carrier

And, when the equations are solved for the inputs specified above, we find that the AM and DSBSC signals consist of the following:

AM	DSBSC	Description
100kHz	-	A sinewave at the carrier frequency
101kHz	101kHz	A sinewave with a frequency equal to the sum of the carrier and message frequencies (the upper sideband or USB)
99kHz	99kHz	A sinewave with a frequency equal to the difference between the carrier and message frequencies (the lower sideband or LSB)

As you can see, AM signals include the carrier signal whereas DSBSC signals don't.

When you think about it, a scope's display is actually a graph of time (on the X-axis) versus voltage (on the Y-axis). Importantly, graphs plotted this way are said to be drawn in the time domain.

Another way of representing signals like AM and DSBSC signals involves drawing all the sinewaves that they contain on a graph that has frequency for the X-axis instead of time. In other words, they're drawn in the frequency domain. When the AM and DSBSC signals in Figure 1 are drawn this way, we get the graphs in Figure 2 below.

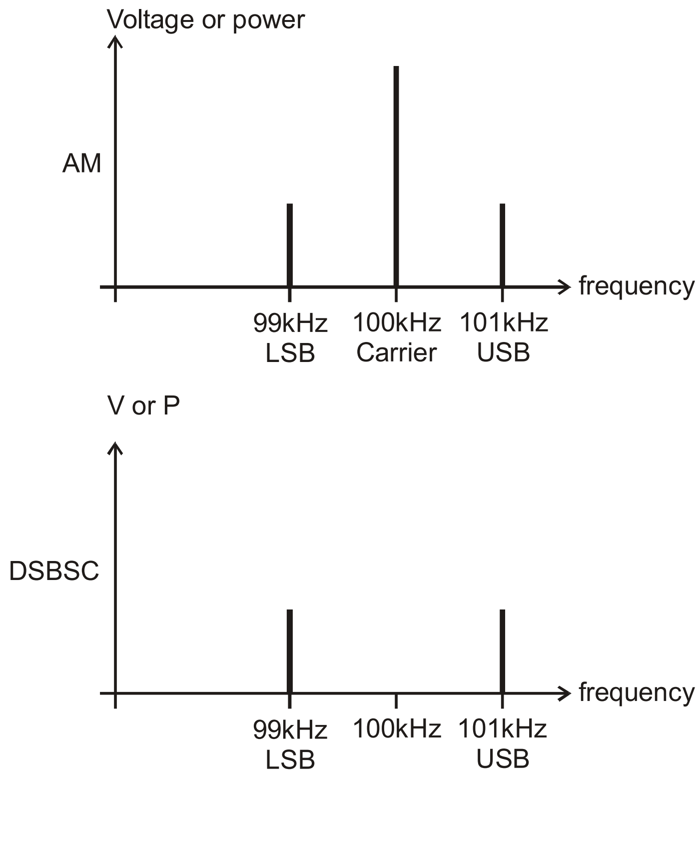


Figure 2

Frequency domain representations of complex signals are very useful for thinking about their spectral composition. They give you a tool for visualising the sinewaves that the signal is made up of. They also help you to see how much of the frequency spectrum the signal occupies. This is the signal's bandwidth and is a critical issue in communications and telecommunications.

The bandwidth of AM and DSBSC signals can be calculated in one of two ways. The frequency domain graphs in Figure 2 shows that the signals occupy a portion of the spectrum from the lower sideband up to the upper sideband. That being the case, the bandwidth can be found using the equation:

$$BW = USB - LSB$$

Using this equation we find that the bandwidth of the AM and DSBSC signals in Figure 2 are 2kHz. In situations where the sidebands are made up of more than one sinewave, you must solve the equation using the highest frequency in the USB and the lowest frequency in the LSB.

Now, compare the bandwidth of the signals in Figure 2 (2kHz) with the original signals used to produce them (that is, a 1kHz message and a 100kHz carrier). Notice that their bandwidths are twice the frequency of their message. This gives us the second equation for calculating bandwidth:

$$BW = 2 \times f_m \qquad \text{where } f_m = \text{the message frequency}$$

In situations where the message is made up of more than one sinewave, you must solve the equation using the highest frequency in the message.

The experiment

For this experiment you'll use the Emona DATEx to generate a real AM and DSBSC signal then analyse the spectral elements of the two signals using the NI ELVIS Dynamic Signal Analyzer.

It should take you about 50 minutes to complete this experiment.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads

Procedure

Part A - Setting up the AM modulator

To experiment with AM spectrum analysis, you need an AM signal. The first part of the experiment gets you to set one up.

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELVISmx software.

8. Launch the NI ELVIS II Variable Power Supplies VI and click on its Run control to activate the hardware.

9. Adjust the Variable Power Supplies negative output Voltage control for an output of about -6V (the exact value is not critical).

10. Launch the NI ELVIS II DMM VI and click on its Run control to activate the hardware.

11. Set up the DMM VI for measuring DC voltages.

12. Launch the DATEx soft front-panel (SFP).

13. Check you have soft control over the DATEx by activating the PCM Encoder module's soft PDM/TDM control on the DATEx SFP.

    Note: If you're set-up is working correctly, the PCM Decoder module's LED on the DATEx board should turn on and off.

14. Locate the Adder module on the DATEx SFP and turn its soft G and g controls fully anti-clockwise.

15. Connect the set-up shown in Figure 3 below.

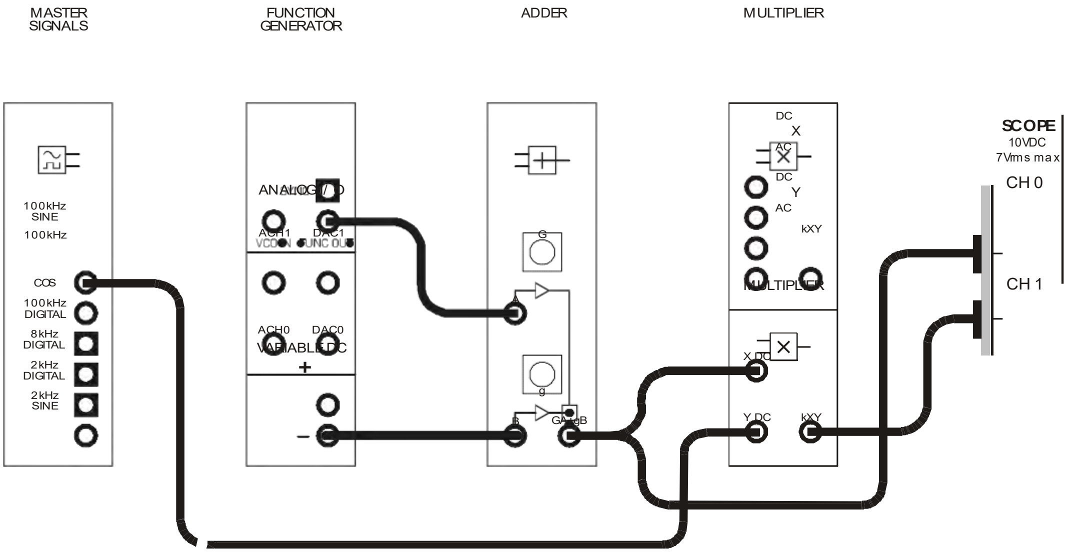


Figure 3

16. Connect the Adder module's output to the DMM and adjust the module's soft g control to obtain a 1V DC output.

Note 1: You must also connect the DMM's COM input to a ground terminal on the Emona DATEx.

Note 2: Remember also that you can use the keyboards tab and arrow keys for fine adjustment of DATEx soft controls.

17. Disconnect the DMM and close its VI.

18. Launch the NI ELVIS II Function Generator VI and click on its Run control to activate the hardware.

19. Adjust the function generator using its soft controls for an output with the following specifications:

- Waveshape: Sine
- Frequency: 10kHz exactly
- Amplitude: 4Vpp
- DC Offset: 0V

20. You'll be using the function generator VI again later but minimise its window for now.

21. Launch the NI ELVIS II Oscilloscope VI and click on its Run control to activate the hardware.

22. Set up the scope per the procedure in Experiment 1 (page 1-12) with the following changes:

- Channel 0 Coupling control to the DC position instead of AC
- Channel 0 Scale control to the 500mV/div position instead of 1V/div
- Timebase control to the 50.us/div position instead of 500.us/div
- Trigger Level control to the 1V position instead of 0V

23. Adjust the Adder module's soft G control to obtain a 1Vp-p sinewave.

24. Activate the scope's Channel 1 input (by checking the Channel 1 Enabled box) to view both the message and the modulated carrier.

    Self check: If the scope's Scale control for Channel 1 is set to the 1V/div position, the scope should now display an AM signal with envelopes that are the same shape and size as the message. If not, close all windows, check your wiring then repeat the process starting from Step 7.

The set-up can be represented by the block diagram in Figure 4 below. It implements the equation: AM = (1VDC + 1Vp-p 10kHz sine) x 4Vp-p 100kHz sine.

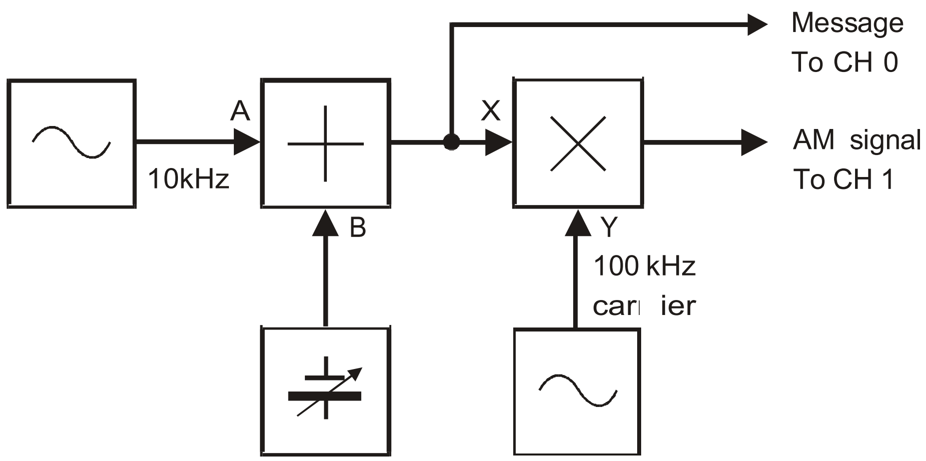


Figure 4

Question 1

For the given inputs to the Multiplier module, what are the frequencies of the three sinewaves on its output?

Question 2

Use this information to calculate the AM signal's bandwidth. Tip: **If** you're not sure how to do this, read the preliminary discussion.

Ask the instructor to check your work before continuing.

Part B - Setting up the NI ELVIS II Dynamic Signal Analyzer (DSA)

25. Suspend the scope's operation by clicking on its Stop control once.

    Note: The scope's display should freeze and its hardware has been deactivated. This is a necessary step as the scope and signal analyzer share hardware resources and so they cannot be operated simultaneously.

26. Minimise the scope's VI.

27. Launch the NI ELVIS II Dynamic Signal Analyzer VI.

    Note: If the Dynamic Signal Analyzer VI has launched successfully, the instrument's window will be visible (see Figure 5).

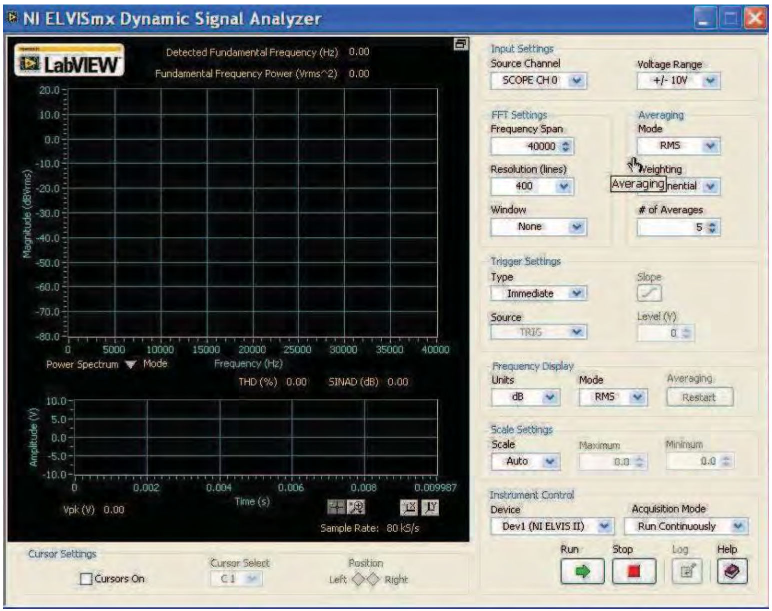


Figure 5

28. Adjust the signal analyzer's controls as follows:

Input Settings

- Source Channel to SCOPE CH 1
- Voltage Range to ±10V

FFT Settings

- Frequency Span to 150,000
- Resolution to 400
- Window to 7 Term B-Harris

Averaging

- Mode to RMS
- Weighting to Exponential
- # of Averages to 3

Trigger Settings

- Type to Edge

Frequency Display

- Units to dB
- Mode to RMS
- Scale to Auto

Cursor settings

- Cursors On box unchecked (for now)

29. Click on the signal analyzer's Run control to activate its hardware.

Note: If the Signal Analyzer VI has been set up correctly, its display should look like Figure 6 below.

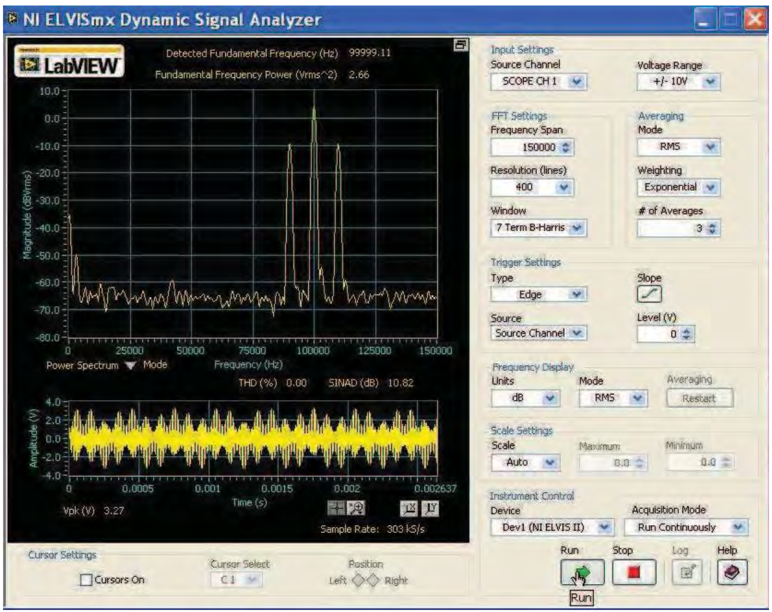


Figure 6

The Signal Analyzer's display needs a little explaining here. There are actually two displays, a large one on top and a much smaller one underneath. The smaller one is a time domain representation of the input (in other words, the display is a scope). Notice that it's showing the AM signal that you set up earlier and saw in Step 24.

The larger of the two displays is the frequency domain representation of the input. Notice that it looks fairly similar to the frequency domain graph for an AM signal in Figure 2 (in the preliminary discussion). The Signal Analyzer's display doesn't have single sharp lines for each of the sinewaves present in the signal because the practical implementation of FFT is not as precise as the theoretical expectation.

Part C - Spectrum analysis of an AM signal

The next part of this experiment let's you analyze the frequency domain representation of the AM signal to see if its frequency components match the values that you mathematically predicted for Questions 1 and 2.

30. Activate the signal analyzer's cursors by checking (that is, ticking) Cursors On box.

    Note: When you do, green horizontal and vertical lines should appear on the signal analyzer's frequency domain display.

The NI ELVIS II Dynamic Signal Analyzer has two cursors C1 and C2 that default to the left most side of the display when the signal analyzer's VI is launched. They're repositioned by "grabbing" their vertical lines with the mouse and moving the mouse left or right.

31. Use the mouse to grab and move the vertical line of cursor C1.

    Note: As you do, notice that cursor C1 moves along the signal analyzer's trace and that the vertical and horizontal lines move so that they always intersect at C1.

32. Repeat Step 31 for cursor C2.

    Note: Fine control over the cursors' position is obtained by using the cursor's Position control in the Cursor Settings area (below the display).

The NI ELVIS II Dynamic Signal Analyzer includes a tool that measures the difference in magnitude and frequency between the two cursors. This information is displayed in green between the upper and lower parts of the display.

33. Move the cursors while watching the measurement readout to observe the effect.

34. Position the cursors so that they're on top of each other and note the measurement.

    Note: When you do, the measurement of difference in magnitude and frequency should both be zero.

Usefully, when one of the cursors is moved to the extreme left of the display, its position on the X-axis is zero. This means that the cursor is sitting on 0Hz. It also means that the measurement readout gives an absolute value of frequency for the other cursor. This makes sense when you think about it because the readout gives the difference in frequency between the two cursors but one of them is zero.

35. Move C1 to the extreme left of the display.

36. Align C2 with the highest point in the AM signal's lower sideband.

    Note: This is the sinewave just to the left of the largest sinewave in the display.

37. Measure the sinewave's frequency and record this in Table 1 on the next page.

38. Align C2 with the highest point in the AM signal's carrier and repeat Step 37.

    Note: This is the largest sinewave in the display.

39. Align C2 with the highest point in the AM signal's upper sideband and repeat Step 37.

    Note: This is the sinewave just to the right of the carrier.

40. Align C1 with the highest point in the AM signal's lower sideband and measure the AM signal's bandwidth.

Table 1

LSB frequency	
Carrier frequency	
USB frequency	
Bandwidth	

Question 3
How do the measured values in Table 1 compare with your theoretically predicted values (see Questions 1 and 2)? Explain any differences.

Ask the instructor to check your work before continuing.

As an aside, at this point it looks as though the sidebands are nearly as large as the carrier. However, this is misleading because the vertical axis is logarithmic (that is, it's non-linear). The sidebands are actually much smaller than the carrier. This can be proven as follows:

41. Set the Signal Analyzer's Units control to Linear instead of dB.

    Note: This sets the vertical axis to a simple linear voltage measurement instead of decibels.

42. Note the relative sizes of the sinewaves in the signal.

43. Return the Signal Analyzer's Units control to dB.

44. Maximise the function generator's VI and increase its output frequency to 20kHz.

45. Use the signal analyzer's cursors to find the AM signal's new bandwidth. Record this in Table 2 below.

    Note: It'll take up to thirty seconds for the display to be fully up to date with the change because it's an average of three sweeps.

46. Increase the function generator's output frequency to 30kHz.

47. Find and record the AM signal's new bandwidth.

Table 2

Bandwidth for fm = 20kHz	
Bandwidth for fm = 30kHz	

Question 4
What's the relationship between the message signal's frequency and the AM signal's bandwidth?

Ask the instructor to check your work before continuing.

48. Return the function generator's output frequency to 10kHz.

49. Wait until the signal analyzer's frequency domain display has fully updated then disconnect the banana plug to the Multiplier module's X input.

50. Wait until the display has fully updated then investigate the frequency of the most significant sinewave on the Multiplier module's output.

Question 5
What is this signal?

Question 6
What's missing and why?

51. Reconnect the banana plug to the Multiplier module's X input.

52. Disconnect the banana plug to the Multiplier module's Y input.

53. Wait until the display has fully updated then investigate the frequency of the most significant sinewave on the Multiplier module's output.

Question 7
What is this signal?

Question 8
Why are the sidebands missing when there's a message?

Ask the instructor to check your work before continuing.

Part D - Setting up the DSBSC modulator

To experiment with DSBSC spectrum analysis, you need a DSBSC signal. This part of the experiment gets you to set one up.

54. Suspend the signal analyzer's operation by clicking on its Stop control once.

55. Maximise the function generator's VI and check that its output frequency has been returned to 10kHz.

56. Set the function generator's output to 1Vpp.

57. Disassemble the current set-up.

58. Connect the set-up shown in Figure 7 below.

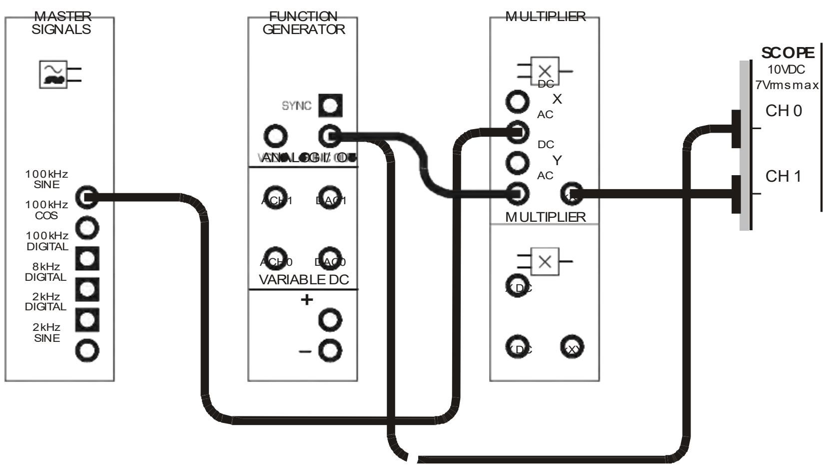


Figure 7

This set-up can be represented by the block diagram in Figure 8 on the next page. It implements the equation: DSBSC = 1Vp-p 10kHz sine x 4Vp-p 100kHz sine.

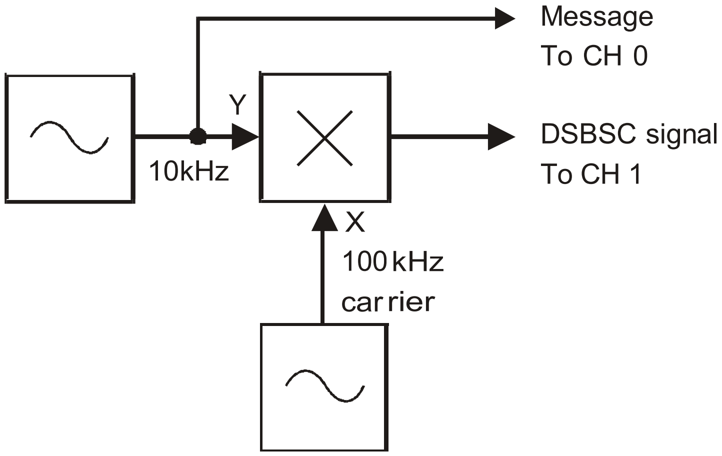


Figure 8

59. Restart the scope and make the following checks and adjustments:

- Check that the Timebase control is set to 50.us/div
- Return the Trigger Level control to 0V

Self check: The scope should now display a DSBSC signal with alternating halves of the envelope forming the same shape as the message and is about the same size.

Question 9
For the given inputs to the Multiplier module, what are the frequencies of the two sinewaves on its output?

Question 10
Use this information to calculate the DSBSC signal's bandwidth.

Ask the instructor to check your work before continuing.

Part E - Spectrum analysis of a DSBSC signal

60. Close the scope's VI.

61. Restart the NI ELVIS II Dynamic Signal Analyzer VI.

    Note: Once the display has had time to update, you should be able to clearly see the DSBSC signal's two sidebands.

You'll also see that the signal has a carrier. However, despite appearances, this signal is very small relative to the sidebands (remember, the scale for the Y-axis is decibels which is a logarithmic unit of measurement). Design limitations in implementing DSBSC mean that there will always be a small carrier component in the DSBSC signal. That's why the second "s" in DSBSC is for "suppressed".

62. Move C1 to the extreme left of the display.

63. Align C2 with the DSBSC signal's lower sideband.

64. Measure the sinewave's frequency and record this in Table 3 below.

65. Align C2 with the DSBSC signal's upper sideband and repeat Step 64.

66. Use the signal analyzer's two cursors to determine and record the DSBSC signal's bandwidth.

Table 3

LSB frequency	
USB frequency	
Bandwidth	

Question 11
How do the measured values in Table 3 compare with your theoretically predicted values (see Questions 9 and 10)?

Question 12
Compare the DSBSC signal's bandwidth with the bandwidth for the AM signal with a 10kHz message (in Table 1). What can you say about the bandwidth requirements of AM and DSBSC signals?

Ask the instructor to check your work before continuing.

67. Find the DSBSC signal's bandwidth for two other message frequencies (say 20kHz and 30kHz).

Question 13
What's the relationship between the message signal's frequency and the DSBSC signal's bandwidth?

Ask the instructor to check your work before finishing

Figure 2 below shows the AM signal at the bottom of Figure 1 but with a dotted line added to track the modulated carrier's positive peaks and negative peaks. These dotted lines are known in the industry as the signal's envelopes. If you look at the envelopes closely you'll notice that the upper envelope is the same shape as the message. The lower envelope is also the same shape but upside-down (inverted).

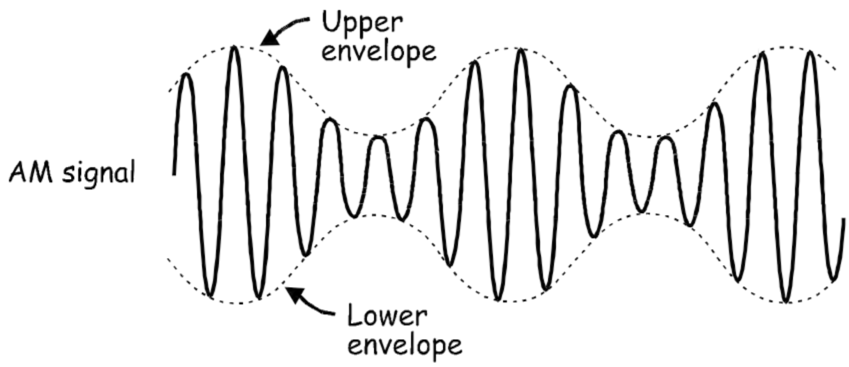


Figure 2

In telecommunications theory, the mathematical model that defines the AM signal is:

AM = (DC + message) x the carrier

When the message is a simple sinewave (like in Figure 1) the equation's solution (which necessarily involves some trigonometry that is not shown here) tells us that the AM signal consists of three sinewaves:

- One at the carrier frequency
- One with a frequency equal to the sum of the carrier and message frequencies
- One with a frequency equal to the difference between the carrier and message frequencies

In other words, for every sinewave in the message, the AM signal includes a pair of sinewaves - one above and one below the carrier's frequency. Complex message signals such as speech and music are made up of thousands sinewaves and so the AM signal includes thousands of pairs of sinewaves straddling carrier. These two groups of sinewaves are called the sidebands and so AM is known as double-sideband, full carrier (DSBFC).

Importantly, it's clear from this discussion that the AM signal doesn't consist of any signals at the message frequency. This is despite the fact that the AM signal's envelopes are the same shape as the message.

Experiment Objectives:

For this experiment you'll use the Emona DATEx to generate a real AM signal by implementing its mathematical model. This means that you'll add a DC component to a pure sinewave to create a message signal then multiply it with another sinewave at a higher frequency (the carrier). You'll examine the AM signal using the scope and compare it to the original message. You'll do the same with speech for the message instead of a simple sinewave.

Following this, you'll vary the message signal's amplitude and observe how it affects the modulated carrier. You'll also observe the effects of modulating the carrier too much. Finally, you'll measure the AM signal's depth of modulation using a scope.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads

Procedure

Part A - Generating an AM signal using a simple message

1. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

2. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

3. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

4. Turn on the PC and let it boot-up.

5. Launch the NI ELVISmx software.

6. Launch the NI ELVIS II Variable Power Supplies VI and click on its Run control to activate the hardware.

7. Adjust the Variable Power Supplies negative output Voltage control for an output of about -6V (the exact value is not critical).

8. You will not need to adjust the Variable Power Supplies VI again so minimize it (but do not close it).

9. Connect the set-up shown in Figure 3 below.

   Note: The NI ELVIS II DMM's inputs are on the left side of the unit.

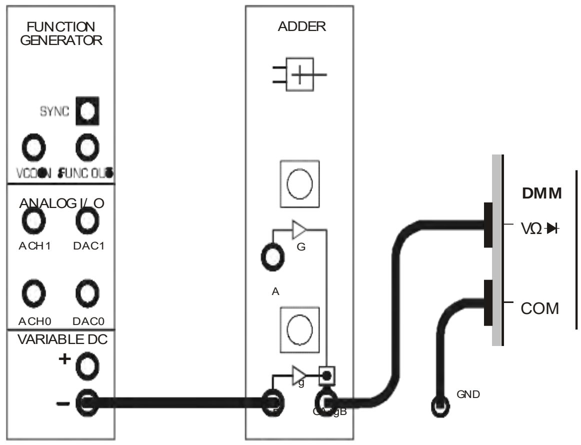


Figure 3

10. Launch the NI ELVIS II DMM VI and press its Run button to activate its hardware.

11. Set up the DMM for measuring DC voltages.

12. Launch the DATEx soft front-panel (SFP).

13. Check you have soft control over the DATEx by activating the PCM Encoder module's soft PDM/TDM control on the DATEx SFP.

    Note: If you're set-up is working correctly, the PCM Decoder module's LED on the DATEx board should turn on and off.

14. Locate the Adder module on the DATEx SFP and turn its soft G control fully anti-clockwise.

15. Adjust the Adder module's soft g control to obtain a 1V DC output (as measured by the DMM).

16. Close the DMM VI - you'll not need it again (unless you accidentally change the Adder module's soft g control).

17. Connect the set-up shown in Figure 4 below.

Note 1: The NI ELVIS II scope inputs are on the left side of the unit.

Note 2: Insert the black plugs of the oscilloscope lead into a ground (GND) socket.

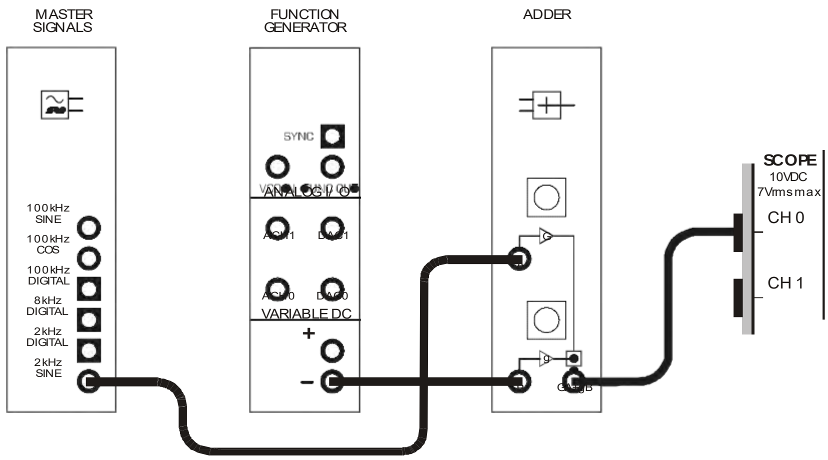


Figure 4

This set-up can be represented by the block diagram in Figure 5 below. It implements the highlighted part of the equation: AM = (DC + message) x the carrier.

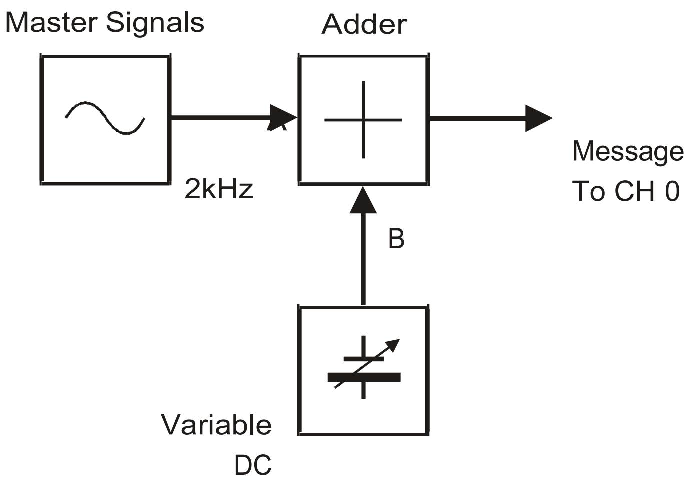


Figure 5

18. Launch and run the NI ELVIS II Oscilloscope VI.

19. Set up the scope per the procedure in Experiment 2 with the following changes:

- Channel 0 Coupling control to the DC position instead of AC
- Channel 0 Scale control to the 500mV/div position instead of 1V/div
- Trigger Level control to the 1V position instead of 0V

Now, the scope's display will not have a trace and there should be a message at the top stating, "***Waiting for Trigger***". This situation is normal and will be corrected when you perform the next step.

20. While watching the Adder module's output on the scope, turn its soft G control clockwise to obtain a 1Vp-p sinewave.

    Tip: Remember that you can use the keyboard's TAB and arrow keys for fine adjustment of the DATEx SFP's controls.

The Adder module's output can now be described mathematically as:

AM = (1VDC + 1Vp-p 2kHz sine) x the carrier

Question 1
In what way is the Adder module's output now different to the signal out of the Master Signals module's 2kHz SINE output?

---

21. Modify the set-up as shown in Figure 6 below.

Before you do.
The set-up in Figure 6 builds on Figure 4 so don't pull it apart. Existing wiring is shown as dotted lines to highlight the patch leads that you need to add.

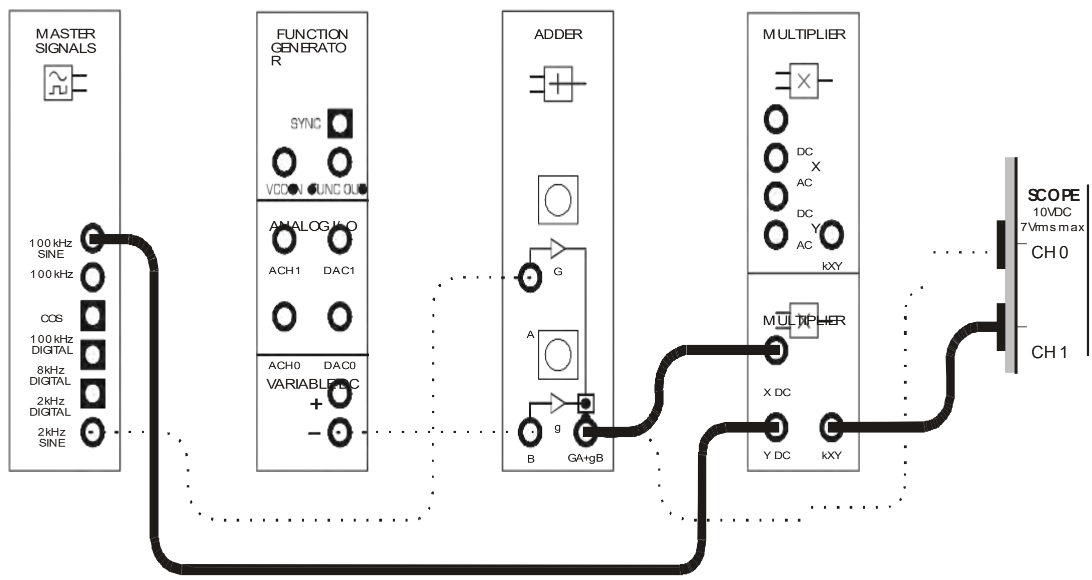


Figure 6

This set-up can be represented by the block diagram in Figure 7 below. The additions that you've made to the original set-up implement the highlighted part of the equation:
AM = (DC + message) x the carrier.

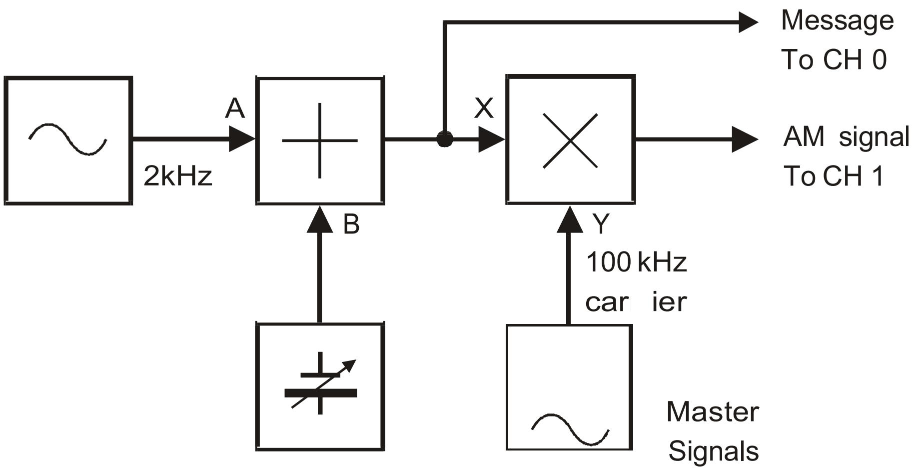


Figure 7

With values, the equation on the previous page becomes:

AM = (1VDC + 1Vp-p 2kHz sine) x 4Vp-p 100kHz sine.

22. Adjust the scope's Timebase control to view only two or so cycles of the message signal.

23. Activate the scope's Channel 1 input (by checking the Channel 1 Enabled box) to view the Multiplier module's output as well as the message signal.

24. Draw the two waveforms to scale on the graph provided below.

    Tip: Draw the message signal in the upper half of the graph and the AM signal in the lower half.

25. Use the scope's Channel 0 Position control to overlay the message with the AM signal's upper envelope then lower envelope to compare them.

Tip: If you haven't do so already, set the Channel 1 Scale control to 500mV/div.

Question 2
What feature of the Multiplier module's output suggests that it's an AM signal? Tip: If you're not sure about the answer to the questions, see the preliminary discussion.

Question 3
The AM signal is a complex waveform consisting of more than one signal. Is one of the signals a 2kHz sinewave? Explain your answer.

Question 4
For the given inputs to the Multiplier module, how many sinewaves does the AM signal consist of, and what are their frequencies?

Part B – Generating an AM signal using speech

This experiment has generated an AM signal using a sinewave for the message. However, the message in commercial communications systems is much more likely to be speech and music. The next part of the experiment lets you see what an AM signal looks like when modulated by speech.

26. Disconnect the plug on the Master Signals module's 2kHz SINE output that connects to the Adder module's A input.

27. Connect it to the Speech module's output as shown in Figure 8 below.

    Remember: Dotted lines show leads already in place.

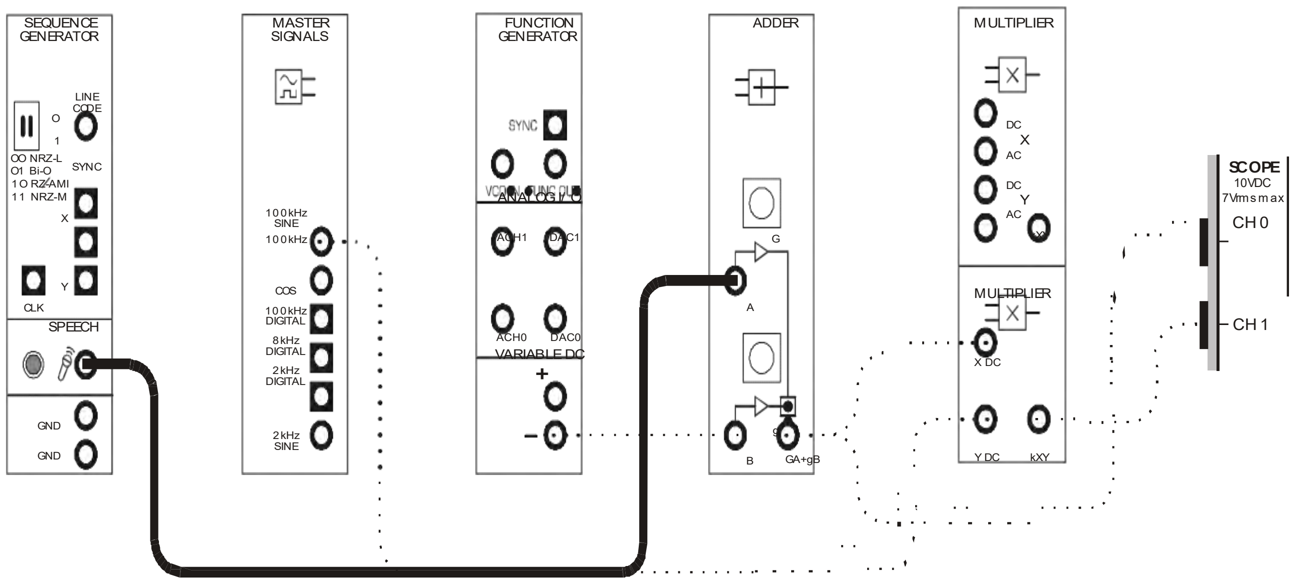


Figure 8

28. Set the scope's Timebase control to the 1ms/div position.

29. Hum and talk into the microphone while watching the scope's display.

Question 5

Why is there still a signal out of the Multiplier module even when you're not humming (or talking, etc)?

___

Part C - Investigating depth of modulation

It's possible to modulate the carrier by different amounts. This part of the experiment let's you investigate this.

30. Return the scope's Timebase control to the 100JJs/div position.

31. Disconnect the plug to the Speech module's output and reconnect it to the Master Signals module's 2kHz SINE output.

    Note: The scope's display should now look like your drawings on the graph paper on page 5-10.

32. Vary the message signal's amplitude a little by turning Adder module's soft G control left and right and notice the effect on the AM signal.

    Question 6
    What is the relationship between the message's amplitude and the amount of the carrier's modulation?

    ____

    ____

You probably noticed that the size of the message signal and the modulation of the carrier are proportional. That is, as the message's amplitude goes up, the amount of the carrier's modulation goes up.

The extent that a message modulates a carrier is known in the industry as the modulation index (m). Modulation index is an important characteristic of an AM signal for several reasons including calculating the distribution of the signal's power between the carrier and sidebands.

Figure 9 below shows two key dimensions of an amplitude modulated carrier. These two dimensions allow a carrier's modulation index to be calculated.

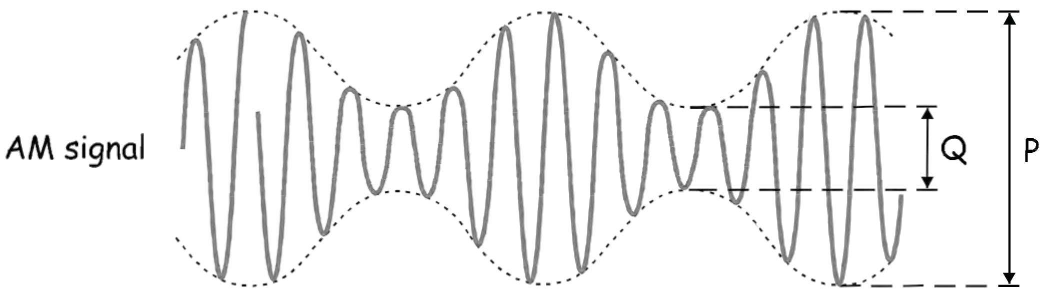


Figure 9

The next part of the experiment lets you practise measuring these dimensions to calculate a carrier's modulation index.

33. Adjust the Adder module's soft G control to return the message signal's amplitude to 1Vp-p.

34. Measure and record the AM signal's P dimension. Record your measurement in Table 1 below.

35. Measure and record the AM signal's Q dimension.

36. Calculate and record the AM signal's depth of modulation using the equation below.

$$m = \frac{P - Q}{P + Q}$$

Table 1

P dimension	Q dimension	m

A problem that is important to avoid in AM transmission is over-modulation. When the carrier is over-modulated, it can upset the receiver's operation. The next part of the experiment gives you a chance to observe the effect of over-modulation.

37. Increase the message signal's amplitude to maximum by turning the Adder module's soft G control to about half its travel then fully clockwise and notice the effect on the AM signal.

38. Set the scope's Channel 0 Scale control to 1V/div and the Channel 1 Scale control to 500mV/div.

39. Use the scope's Channel 0 Position control to overlay the message with the AM signal's envelopes and compare them.

Question 7
What is the problem with the AM signal when it is over-modulated?

____

____

Question 8
What do you think is a carrier's maximum modulation index without over-modulation?

A- A minus number

B- 0

C- 1

D- Greater than 1

40. Draw the two waveforms to scale in the space provided below.

Week 6: Experiment 6 - DSBSC modulation

Test Standard :IEEE 802

Preliminary discussion

DSBSC is a modulation system similar but different to AM (which was explored in Experiment 5).

Like AM, DSBSC uses a microphone or some other transducer to convert speech and music to an electrical signal called the message or baseband signal. The message signal is then used to electrically vary the amplitude of a pure sinewave called the carrier. And like AM, the carrier usually has a frequency that is much higher than the message's frequency.

Figure 1 below shows a simple message signal and an unmodulated carrier. It also shows the result of modulating the carrier with the message using DSBSC.

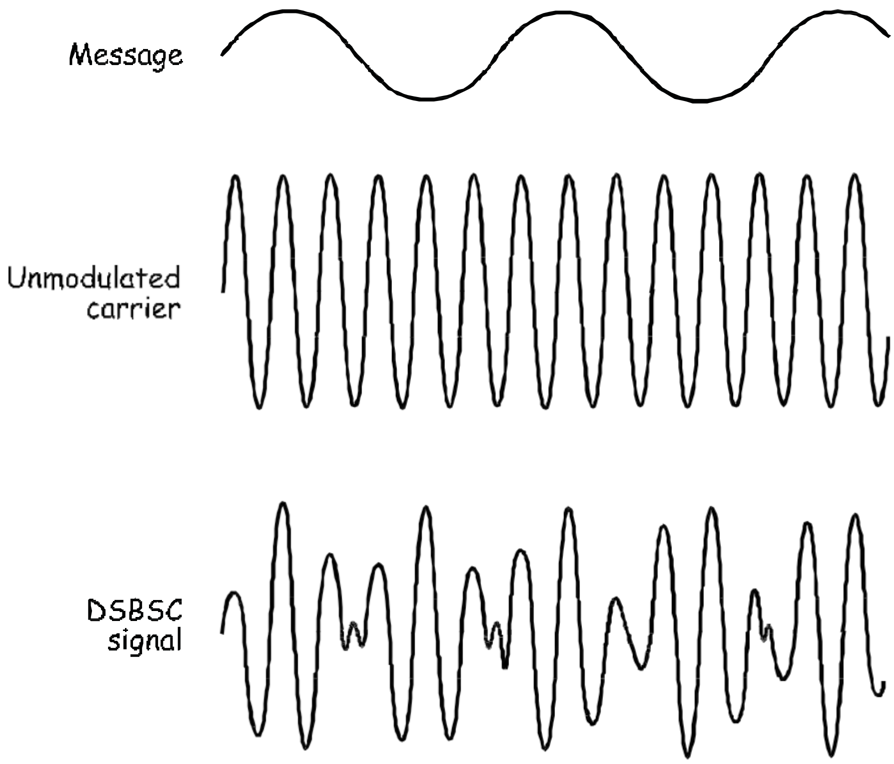


Figure 1

So far, there doesn't appear to be much difference between AM and DSBSC. However, consider Figure 2 below. It is the DSBSC signal at the bottom of Figure 1 but with dotted lines added to track the signal's envelopes (that is, its positive peaks and negative peaks). If you look at the envelopes closely you'll notice that they're not the same shape as the message as is the case with AM (see Experiment 5 page 5-3 for an example).

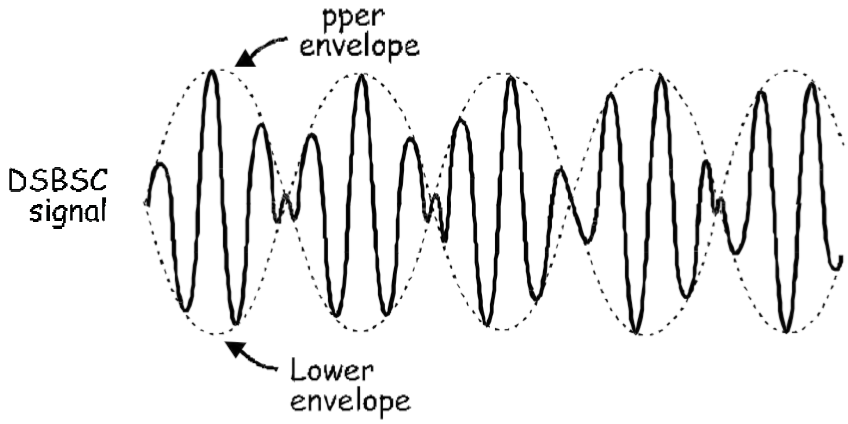


Figure 2

Instead, alternating halves of the envelopes form the same shape as the message as shown in Figure 3 below.

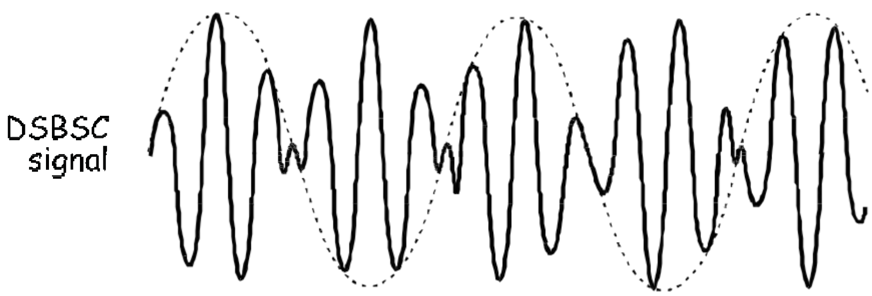


Figure 3

Another way that DSBSC is different to AM can be understood by considering the mathematical model that defines the DSBSC signal:

DSBSC = the message x the carrier

Do you see the difference between the equations for AM and DSBSC? If not, look at the AM equation in Experiment 5 (page 5-3).

When the message is a simple sinewave (like in Figure 1) the equation's solution (which necessarily involves some trigonometry) tells us that the DSBSC signal consists of two sinewaves:

- One with a frequency equal to the sum of the carrier and message frequencies
- One with a frequency equal to the difference between the carrier and message frequencies

Importantly, the DSBSC signal doesn't contain a sinewave at the carrier frequency. This is an important difference between DSBSC and AM.

That said, as the solution to the equation shows, DSBSC is the same as AM in that a pair of sinewaves is generated for every sinewave in the message. And, like AM, one is higher than the unmodulated carrier's frequency and the other is lower. As message signals such as speech and music are made up of thousands of sinewaves, thousands of pairs of sinewaves are generated in the DSBSC signal that sit on either side of the carrier frequency. These two groups are called the sidebands.

So, the presence of both sidebands but the absence of the carrier gives us the name of this modulation method - double-sideband, suppressed carrier (DSBSC).

The carrier in AM makes up at least 66o of the signal's power but it doesn't contain any part of the original message and is only needed for tuning. So by not sending the carrier, DSBSC offers a substantial power saving over AM and is its main advantage.

Experiment Objectives:

For this experiment you'll use the Emona DATEx to generate a real DSBSC signal by implementing its mathematical model. This means that you'll take a pure sinewave (the message) that contains absolutely no DC and multiply it with another sinewave at a higher frequency (the carrier). You'll examine the DSBSC signal using the scope and compare it to the original message. You'll do the same with speech for the message instead of a simple sinewave.

Following this, you'll vary the message signal's amplitude and observe how it affects the carrier's depth of modulation. You'll also observe the effects of modulating the carrier too much.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads

Procedure

Part A - Generating a DSBSC signal using a simple message

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELVISmx software.

8. Launch and run the NI ELVIS II Oscilloscope virtual instrument (VI).

9. Set up the scope per the procedure in Experiment 1 (page 1-12) ensuring that the Trigger Source control is set to CH 0.

10. Launch the DATEx soft front-panel (SFP).

11. Check you now have soft control over the DATEx by activating the PCM Encoder module's soft PDM/TDM control on the DATEx SFP.

    Note: If you're set-up is working correctly, the PCM Decoder module's LED on the DATEx board should turn on and off.

12. Connect the set-up shown in Figure 4 below.

Note: Insert the black plugs of the oscilloscope leads into a ground (GND) socket.

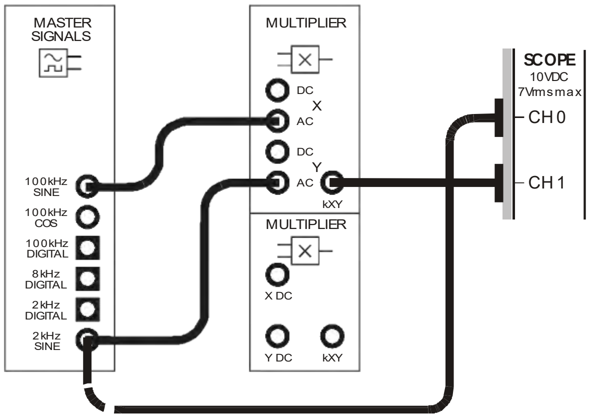


Figure 4

This set-up can be represented by the block diagram in Figure 5 below. It implements the entire equation: DSBSC = the message x the carrier.

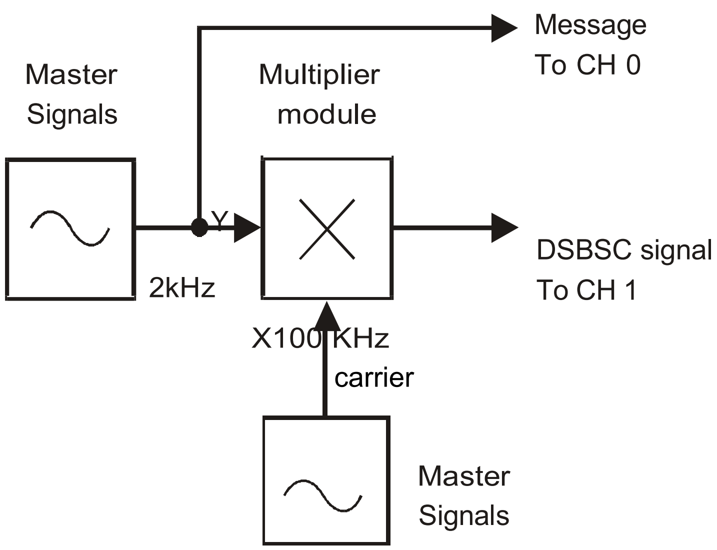


Figure 5

With values, the equation on the previous page becomes:

DSBSC = 4Vp-p 2kHz sine x 4Vp-p 100kHz sine.

13. Adjust the scope's Timebase control to view two or so cycles of the Master Signals module's 2kHz SINE output.

14. Activate the scope's Channel 1 input (by checking the Channel 1 Enabled box) to view the DSBSC signal out of the Multiplier module as well as the message signal.

15. Set the scope's Channel 0 Scale control to the 500mV/div position and the Channel 1 Scale control to the 1V/div position (if it's not already).

16. Draw the two waveforms to scale in the space provided below.

    Tip: Draw the message signal in the upper half of the graph and the DSBSC signal in the lower half.

17. If they're not already, overlay the message with the DSBSC signal's envelopes to compare them using the scope's Channel 0 Position control.

Question 1
What feature of the Multiplier module's output suggests that it's a DSBSC signal? Tip: If you're not sure about the answer to the questions, see the preliminary discussion.

Question 2
The DSBSC signal is a complex waveform consisting of more than one signal. Is one of the signals a 2kHz sinewave? Explain your answer.

Question 3
For the given inputs to the Multiplier module, how many sinewaves does the DSBSC signal consist of, and what are their frequencies?

Question 4
Why does this make DSBSC signals better for transmission than AM signals?

Part B – Generating a DSBSC signal using speech

This experiment has generated a DSBSC signal using a sinewave for the message. However, the message in commercial communications systems is much more likely to be speech and music. The next part of the experiment lets you see what a DSBSC signal looks like when modulated by speech.

18. Disconnect the plugs to the Master Signals module's 2kHz SINE output.

19. Connect them to the Speech module's output as shown in Figure 6 below.

    Remember: Dotted lines show leads already in place.

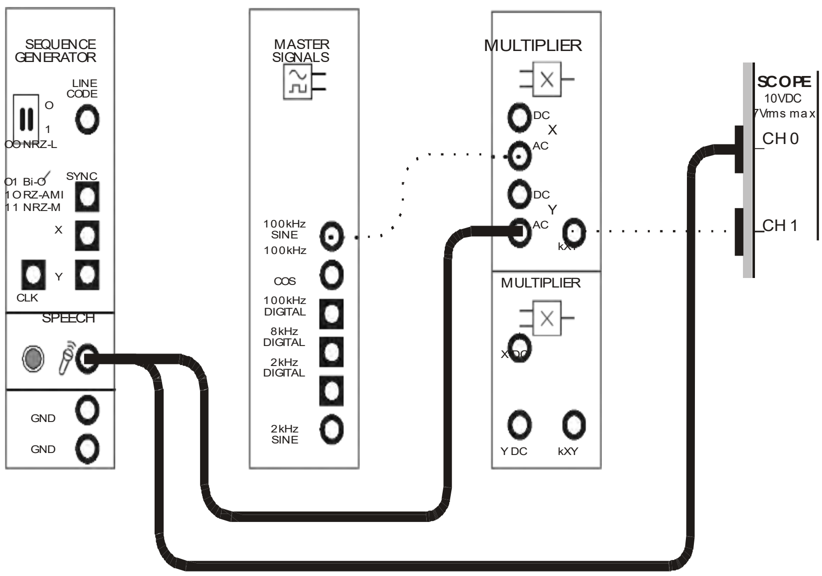


Figure 6

20. Set the scope's Timebase control to the 1ms/div position.

21. Hum and talk into the microphone while watching the scope's display.

Question 5

Why isn't there any signal out of the Multiplier module when you're not humming or talking?

Part C - Investigating depth of modulation

It's possible to modulate the carrier by different amounts. This part of the experiment let's you investigate this.

22. Return the scope's Timebase control to the 100JJs/div position.

23. Locate the Amplifier module on the DATEx SFP and set its soft Gain control to about a quarter of its travel (the control's line should be pointing to where the number nine is on a clock's face).

24. Modify the set-up as shown in Figure 7 below.

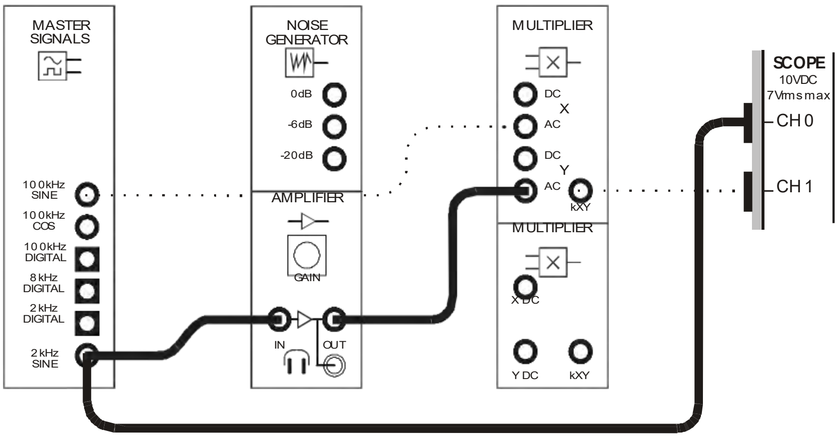


Figure 7

The set-up in Figure 7 can be represented by the block diagram in Figure 8 below. The Amplifier allows the message signal's amplitude to be adjustable.

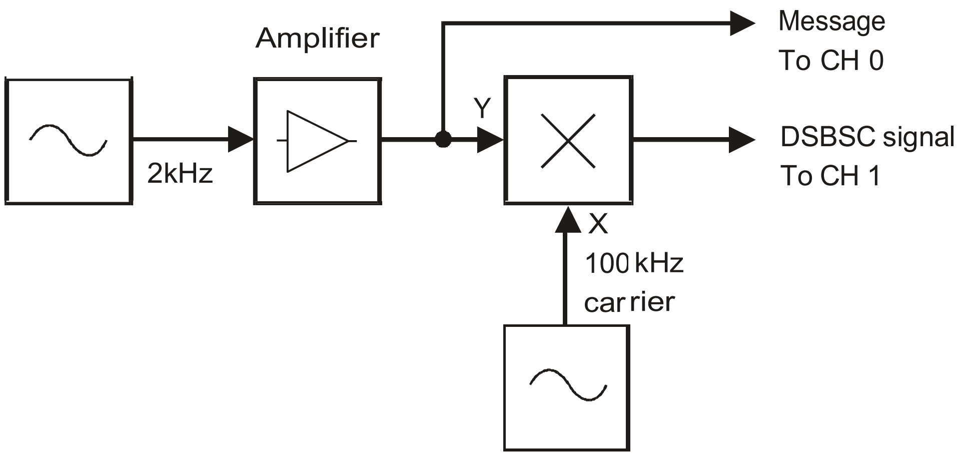


Figure 8

Note: At this stage, the Multiplier module's output should be the normal DSBSC signal that you sketched earlier.

Recall from Experiment 5 that an AM signal has two dimensions that can be measured and used to calculated modulation index (m). The dimensions are denoted P and Q. If you've forgotten which one is which, take a minute to read over the notes at the top of page 5-14 before going on to the next step.

25. Vary the message signal's amplitude a little by turning the Amplifier module's soft Gain control left and right a little. Notice the effect that this has on the DSBSC signal's P and Q dimensions.

Question 6

Based on your observations in Step 25, when the message's amplitude is varied

a) neither dimensions P or Q are affected.

b) only dimension Q is affected.

c) only dimension P is affected.

d) both dimensions P and Q are affected.

On the face of it, determining the depth of modulation of a DSBSC signal is a problem. The modulation index is always the same number regardless of the message signal's amplitude. This is because the DSBSC signals Q dimension is always zero.

However, this isn't the problem that it seems. One of the main reasons for calculating an AM signal's modulation index is so that the distribution of power between the signal's carrier and its sidebands can be calculated. However, DSBSC signals don't have a carrier (remember, it's suppressed). This means that all of the DSBSC signal's power is distributed between its sidebands evenly. So there's no need to calculate a DSBSC signal's modulation index.

The fact that you can't calculate a DSBSC signal's modulation index might imply that you can make either the message or the carrier as large as you like without worrying about over-modulation. This isn't true. Making either of these two signals too large can still overload the modulator resulting in a type of distortion that you've seen before. The next part of the experiment lets you observe what happens when you overload a DSBSC modulator.

26. Set the Amplifier module's soft Gain control to about half its travel and notice the effect on the DSBSC signal.

    Note 1: Resize the display as necessary using the scope's Channel 1 Scale control.

    Note 2: If doing this has no effect, turn up the gain control a little more.

27. Draw the new DSBSC signal to scale in the space provided below.

Question 7
What is the name of this type of distortion?

Week 7: Experiment 7 - AM demodulation

Test Standard :IEEE 802

Preliminary discussion

If you've completed Experiment 5 then you've seen what happens when a 2kHz sinewave is used to amplitude modulate a carrier to produce an AM signal. Importantly, you would have seen a key characteristic of an AM signal - its envelopes are the same shape as the message (though the lower envelope is inverted).

Recovering the original message from a modulated carrier is called demodulation and this is the main purpose of communications and telecommunications receivers. The circuit that is widely used to demodulate AM signals is called an envelope detector. The block diagram of an envelope detector is shown in Figure 1 below.

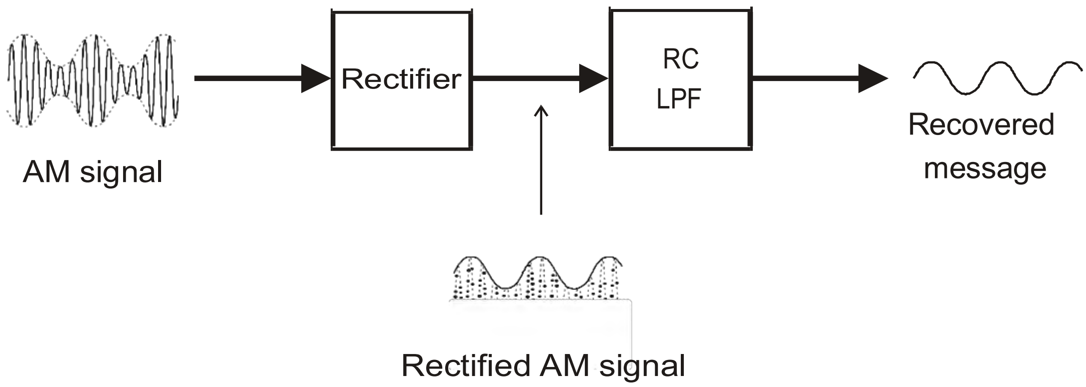


Figure 1

As you can see, the rectifier stage chops the AM signal in half letting only one of its envelopes through (the upper envelope in this case but the lower envelope is just as good). This signal is fed to an RC LPF which tracks the peaks of its input. When the input to the RC LPF is a rectified AM signal, it tracks the signal's envelope. Importantly, as the envelope is the same shape as the message, the RC LPF's output voltage is also the same shape as the message and so the AM signal is demodulated.

A limitation of envelope detector shown in Figure 1 is that it cannot accurately recover the message from over-modulated AM signals. To explain, recall that when an AM carrier is over-modulated the signal's envelope is no-longer the same shape as the original message. Instead, the envelope is distorted and so, by definition, this means that the envelope detector must produce a distorted version of the message.

The experiment

For this experiment you'll use the Emona DATEx to generate an AM signal by implementing its mathematical model. Then you'll set-up an envelope detector using the Rectifier and RC LPF on the trainer's Utilities module.

Once done, you'll connect the AM signal to the envelope detector's input and compare the demodulated output to the original message and the AM signal's envelope. You'll also observe the effect that an over-modulated AM signal has on the envelope detector's output.

Finally, if time permits, you'll demodulate the AM signal by implementing by multiplying it with a local carrier instead of using an envelope detector.

It should take you about 50 minutes to complete Parts A to D of this experiment and another 20 minutes to complete Part E.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads
- One set of headphones (stereo)

Procedure

Part A - Setting up the AM modulator

To experiment with AM demodulation you'll need an AM signal. The first part of the experiment gets you to set one up.

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELVISmx software.

8. Launch and run the NI ELVIS II Variable Power Supplies VI.

9. Adjust the Variable Power Supplies negative output Voltage control for an output of about -6V (the exact value is not critical).

10. Launch and run the NI ELVIS II DMM VI.

11. Set up the DMM VI for measuring DC voltages.

12. Launch the DATEx soft front-panel (SFP) and check that you have soft control over the DATEx board.

13. Locate the Adder module on the DATEx SFP and turn its soft G and g controls fully anti-clockwise.

14. Connect the set-up shown in Figure 2 below.

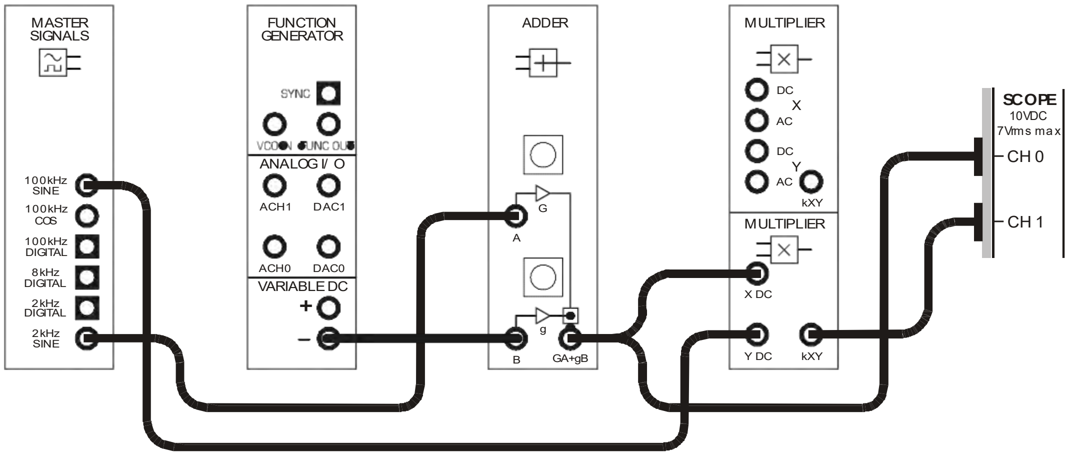


Figure 2

15. Connect the Adder module's output to the DMM and adjust the module's soft g control to obtain a 1V DC output.

Note 1: Remember that you must also connect the DMM's COM input to a ground terminal on the Emona DATEx.

Note 2: Remember also that you can use the keyboards tab and arrow keys for fine adjustment of DATEx soft controls.

16. Disconnect the DMM and close its VI.

17. Launch and run the NI ELVIS II Oscilloscope VI.

18. Set up the scope per the procedure in Experiment 1 with the following changes:

- Channel 0 Coupling control to the DC position instead of AC
- Channel 0 Scale control to the 500mV/div position instead of 1V/div
- Trigger Level control to the 1V position instead of 0V

19. Adjust the scope's Timebase control to view only two or so cycles of the message signal.

20. Adjust the Adder module's soft G control to obtain a 1Vp-p sinewave.

21. Activate the scope's Channel 1 input to view both the message and the modulated carrier.

Self check: If the scope's Scale control for Channel 1 is set to the 1V/div position, the scope should now display an AM signal with envelopes that are the same shape and size as the message. If not, close all windows, check your wiring then repeat the process starting from Step 7.

The set-up in Figure 2 on the previous page can be represented by the block diagram in Figure 3 below. It generates a 100kHz carrier that is amplitude modulated by a 2kHz sinewave message.

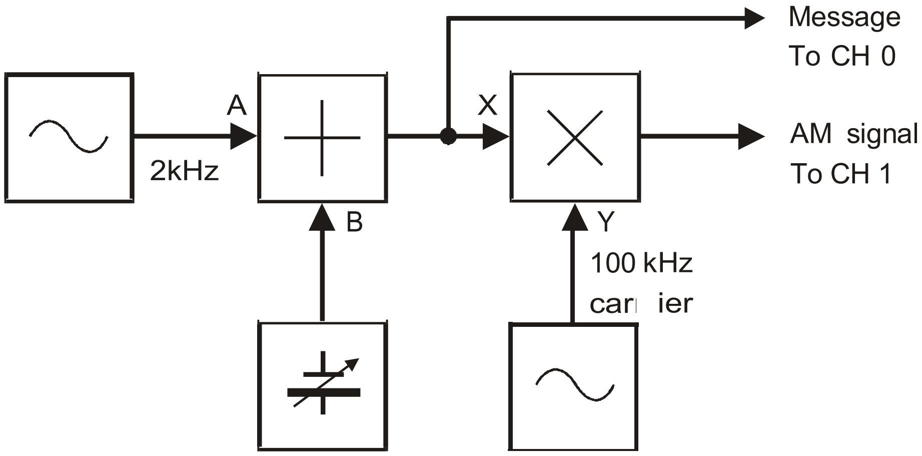


Figure 3

Ask the instructor to check
your work before continuing.

Part B - Recovering the message using an envelope detector

22. Modify the set-up as shown in Figure 4 below.

Remember: Dotted lines show leads already in place.

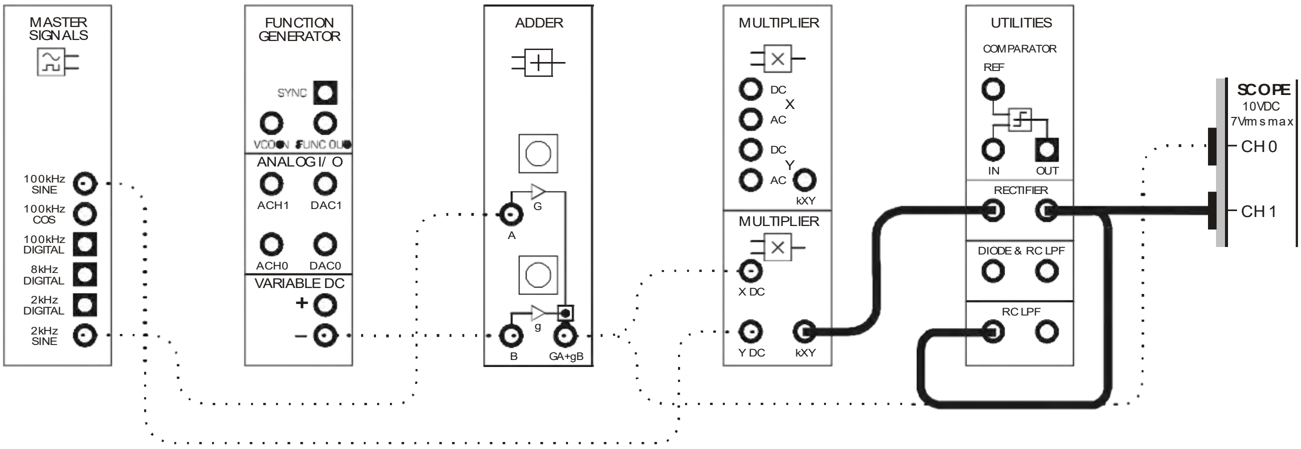


Figure 4

The additions to the set-up can be represented by the block diagram in Figure 5 below. As you can see, it's the envelope detector explained in the preliminary discussion.

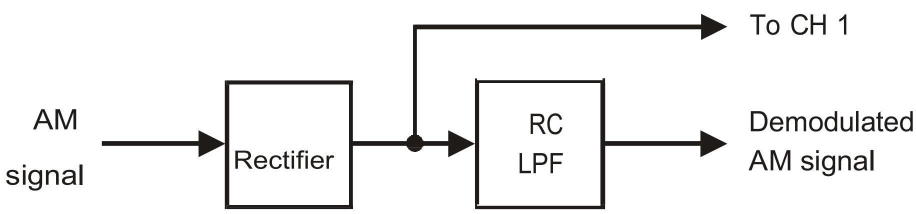


Figure 5

23. Adjust the scope's Scale and Timebase controls to appropriate settings for the signals.

24. Draw the two waveforms to scale in the space provided below leaving room to draw a third waveform.

    Tip: Draw the message signal in the upper third of the graph and the rectified AM signal in the middle third.

25. Disconnect the scope's Channel 1 input from the Rectifier's output and connect it to the RC LPF's output instead.

26. Draw the demodulated AM signal to scale in the space that you left on the graph paper.

Question 1
What is the relationship between the original message signal and the recovered message?

Ask the instructor to check your work before continuing.

Part C - Investigating the message's amplitude on the recovered message

27. Vary the message signal's amplitude up and down a little (by turning the Adder module's soft G control left and right a little) while watching the demodulated signal.

Question 2
What is the relationship between the amplitude of the two message signals?

---

28. Slowly increase the message signal's amplitude to maximum while watching the demodulated signal.

Question 3
What do you think causes the heavy distortion of the demodulated signal? Tip: If you're not sure, connect the scope's Channel 0 input to the AM modulator's output.

Question 4
Why does over-modulation cause the distortion?

Ask the instructor to check your work before continuing.

Part D - Transmitting and recovering speech using AM

This experiment has set up an AM communication system to "transmit" a message that is a 2kHz sinewave. The next part of the experiment lets you use the set-up to modulate, transmit, demodulate and listen to speech.

29. If you moved the scope's Channel 0 input to help you answer Question 4, reconnect it to the Adder module's output.

30. Return the message signal's amplitude to 1Vpp (by adjusting the Adder module's soft G control).

31. Modify the set-up as shown in Figure 6 below.

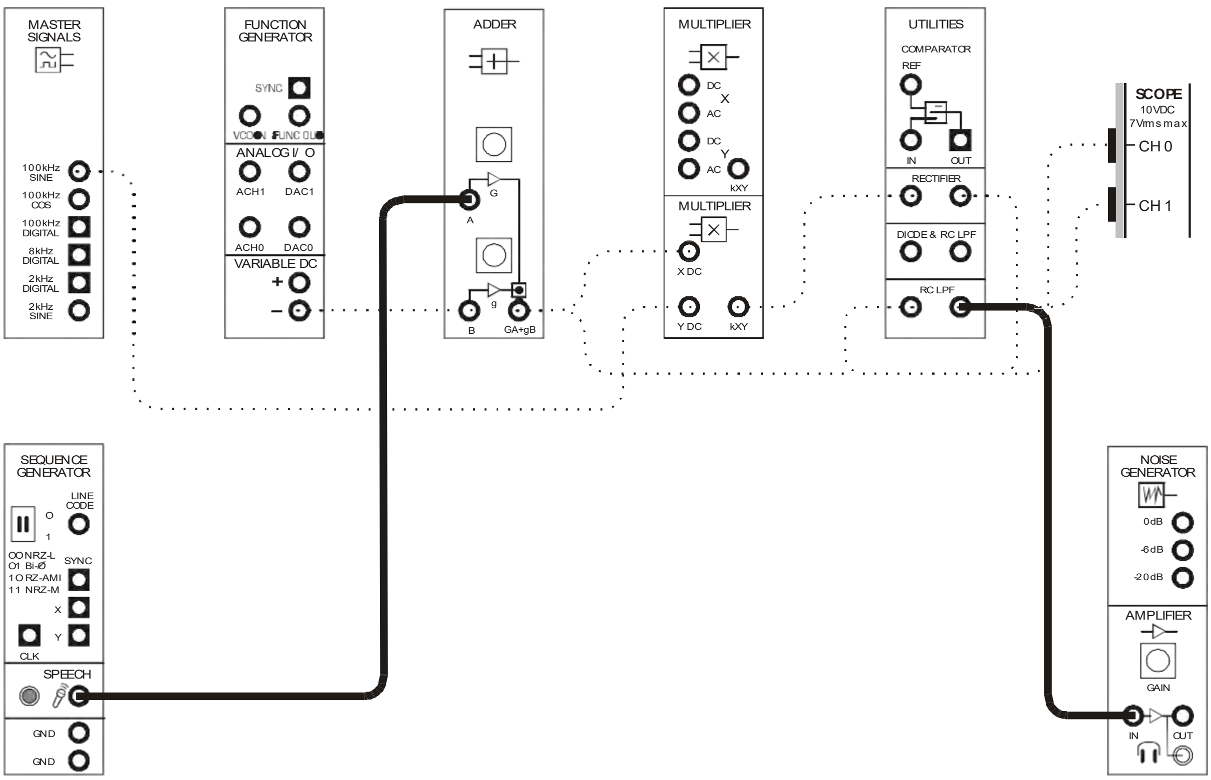


Figure 6

32. Set the scope's Timebase control to the 2ms/div position.

33. Turn the Amplifier module's soft Gain control fully anti-clockwise.

34. Without wearing the headphones, plug them into the Amplifier module's headphone socket.

35. Put the headphones on.

36. As you perform the next step, set the Amplifier module's soft Gain control to a comfortable sound level.

37. Hum and talk into the microphone while watching the scope's display and listening on the headphones.

Ask the instructor to check
your work before continuing.

Part E - The mathematics of AM demodulation

AM demodulation can be understood mathematically because it is uses multiplication to reproduce the original message. To explain, recall that when two pure sinewaves are multiplied together (a mathematical process that necessarily involves some trigonometry that is not shown here) the result gives two completely new sinewaves:

- One with a frequency equal to the sum of the two signals' frequencies
- One with a frequency equal to the difference between the two signals' frequencies

The envelope detector works because the rectifier is a device that multiplies all signals on its one input with each other. Ordinarily, this is a nuisance but not for applications like AM demodulation. Recall that an AM signal consists of a carrier, the carrier plus the message and the carrier minus the message. So, when an AM signal is connected to a rectifier's input, mathematically the rectifier's cross multiplication of all of its sinewaves looks like:

Rectifier's output = carrier x (carrier + message) x (carrier - message)

If the message signal used to generate the AM signal is a simple sinewave then, when the equation above is solved, the rectifier outputs six sinewaves at the following frequencies:

- Carrier + (carrier + message)
- Carrier + (carrier - message)
- (carrier + message) + (carrier - message)
- Carrier - (carrier + message) which simplifies to just the message
- Carrier - (carrier - message) which also simplifies to just the message
- (carrier + message) - (carrier - message)

To make this a little more meaningful, let's do an example with numbers. The AM modulator that you set up at the beginning of this experiment uses a 100kHz carrier and a 2kHz message (with a DC component). So, the resulting AM signal consists of three sinewaves: one at 100kHz, another at 102kHz and a third at 98kHz. Table 1 below shows what happens when these sinewaves are cross-multiplied by the rectifier.

Table 1	100kHzx102kHz	100kHzx98kHz	98kHzx102kHz
Sum	202kHz	198kHz	200kHz
Difference	2kHz	2kHz	4kHz

Notice that two of the sinewaves are at the message frequency. In other words, the message has been recovered! And, as the two messages are in phase, they simply add together to make a single bigger message.

Importantly, we don't want the other non-message sinewaves so, to reject them but keep the message, the rectifier's output is sent to a low-pass filter. Ideally, the filter's output will only consist of the message signal. The chances of this can be improved by making the carrier's frequency much higher than the highest frequency in the message. This in turn makes the frequency of the "summed" signals much higher and easier for the low-pass filter to reject.

[As an aside, the 4kHz sinewave that was generated would pass through the low-pass filter as well and be present on its output along with the 2kHz signal. This is inconvenient as it is a signal that was not present in the original message. Luckily, as the signal was generated by multiplying the sidebands, its amplitude is much lower than the recovered message and can be ignored.]

An almost identical mathematical process can be modelled using the Emona DATEx module's Multiplier module. However, instead of multiplying the AM signal's sinewaves with each other (the Multiplier module doesn't do this), they're multiplied with a locally generated 100kHz sinewave. The next part of this experiment lets you demodulate an AM signal this way.

38. Return the scope's Timebase control to its earlier setting (probably 200J.s/div).

39. Disconnect the envelope detector and modify the set-up to return it to just an AM modulator with a 2kHz sinewave for the message as shown in Figure 7 below.

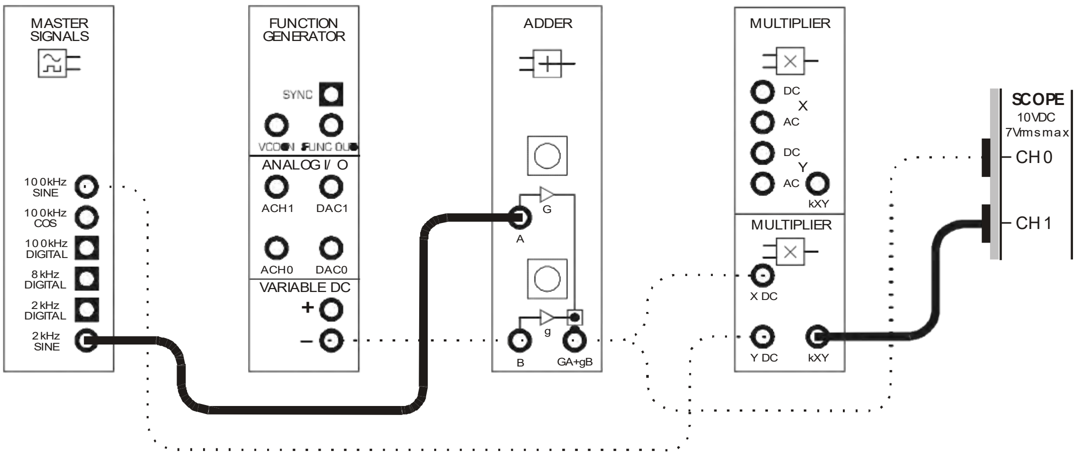


Figure 7

40. Modify the set-up as shown in Figure 8 below.

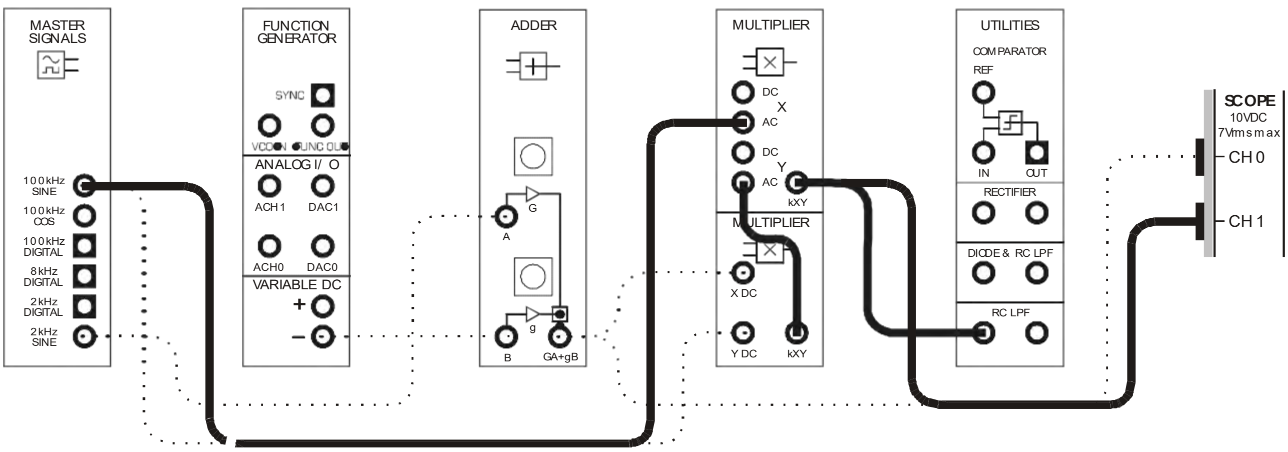


Figure 8

The additions to the set-up in Figure 8 on the previous page can be represented by the block diagram in Figure 9 below. The Multiplier module models the mathematical basis of AM demodulation and the RC Low-pass filter on the Utilities module picks out the message while rejecting the other sinewaves generated.

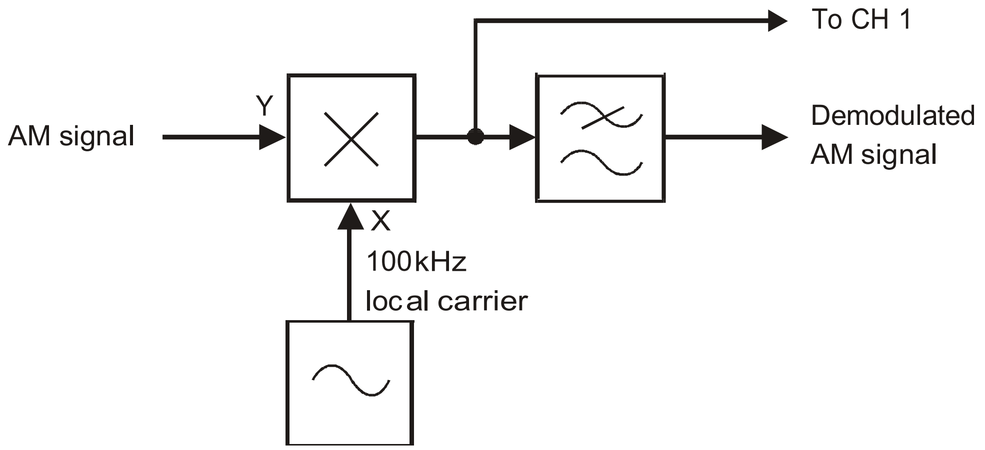


Figure 9

41. Compare the Multiplier module's output with the Rectifier's output that you drew earlier (see page 8-8).

Question 5
Given the AM signal (which consists of 100kHz, 102kHz and 98kHz sinewaves) is being multiplied by a 100kHz sinewave:

A) How many sinewaves are present in the Multiplier module's output?

B) What are their frequencies?

____

____

42. Disconnect the scope's Channel 1 input from the Multiplier module's output and connect it to the RC LPF's output instead.

43. Compare the RC LPF's output with the message and the output RC LPF's that you drew earlier (see page 8-8).

Ask the instructor to check
your work before continuing.

A common misconception about AM is that, once the signal is over-modulated, it's impossible to recover the message. However, when the AM signal is generated using an ideal or near-ideal modulator (like Figure 3) this is only true for the envelope detector.

The AM demodulation method being implemented in this part of the experiment (called product detection - though it is more accurate to call it product demodulation) doesn't suffer from this problem as it's not designed to recover the message by tracking one of the AM signal's envelopes. The final part of this experiment demonstrates this.

44. Connect the scope's Channel 0 to the AM modulator's output.

45. Set the scope's Trigger Level to 0V.

    Note: The scope will lose triggering but the display will be adequate for the next steps.

46. Slowly increase the message signal's amplitude to produce a near 100o modulated AM signal by adjusting the Adder module's soft G control.

    Note: Resize the AM and demodulated message signals on the screen as necessary.

47. Slowly increase the message signal's amplitude to produce an AM signal that is modulated by more than 100o while paying close attention to the demodulated message signal.

As an aside, the commercial implementation of AM modulation commonly involves a Class C amplifier for efficiency (that is, to minimise power losses). When a Class C amplifier is operated at depths of modulation above 100o the circuit's operation no-longer corresponds with the model of an AM modulator in Figure 3. Importantly, in addition to producing an envelope that is not the same as the original message, the over-modulated Class C circuit produces extra frequency components in the spectrum. This means that neither the envelope detector nor the product demodulator can reproduce the message without distortion.

Experiment 9 - DSBSC demodulation

Preliminary discussion

Experiment 8 shows how the envelope detector can be used to recover the original message from an AM signal (that is, demodulate it). Unfortunately, the envelope detector cannot be used to demodulate a DSBSC signal.

To understand why, recall that the envelope detector outputs a signal that is a copy of its input's envelope. This works well for demodulating AM because the signal's envelopes are the same shape as the message that produced it in the first place (that is, as long as it's not over-modulated). However, recall that a DSBSC signal's envelopes are not the same shape as the message.

Instead, DSBSC signals are demodulated using a circuit called a product detector (though product demodulator is a more appropriate name) and its basic block diagram is shown in Figure 1 below. Other names for this type of demodulation include a synchronous detector and switching detector.

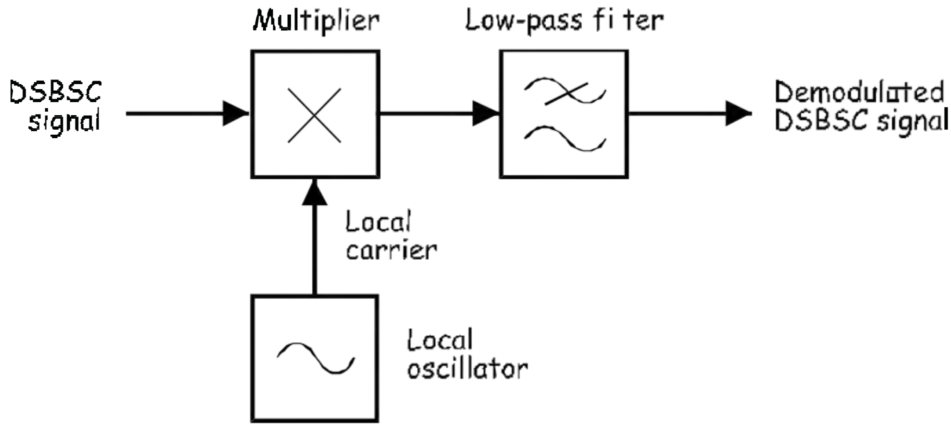


Figure 1

As its name implies, the product detector uses multiplication and so mathematics are necessary to explain its operation. The incoming DSBSC signal is multiplied by a pure sinewave that must be the same frequency as the DSBSC signal's suppressed carrier. This sinewave is generated by the receiver and is known as the local carrier.

To see why this process recovers the message, let's describe product detection mathematically:

DSBSC demodulator's output = the DSBSC signal x the local carrier

Importantly, recall that DSBSC generation involves the multiplication of the message with the carrier which produces sum and difference frequencies (the preliminary discussion in Experiment 6 summarises DSBSC generation). That being the case, this information can be substituted for the DSBSC signal and the equation rewritten as:

DSBSC demodulator's output = [(carrier + message) + (carrier - message)] x carrier

When the equation is solved, we get four sinewaves with the following frequencies:

- Carrier + (carrier + message)
- Carrier + (carrier - message)
- Carrier - (carrier + message) which simplifies to just the message
- Carrier - (carrier - message) which also simplifies to just the message

(If you're not sure why these sinewaves are produced, it's important to remember that whenever two pure sinewaves are multiplied together, two completely new sinewaves are generated. One has a frequency equal to the sum of the original sinewaves' frequencies and the other has a frequency equal to their difference.)

Importantly, notice that two of the products are sinewaves at the message frequency. In other words, the message has been recovered. As the two message signals are in phase, they simply add together to make one larger message.

Notice also that two of the products are non-message sinewaves. These sinewaves are unwanted and so a low-pass filter is used to reject them while keeping the message.

The experiment

For this experiment you'll use the Emona DATEx to generate a DSBSC signal by implementing its mathematical model. Then you'll set-up a product detector by implementing its mathematical model also.

Once done, you'll connect the DSBSC signal to the product detector's input and compare the demodulated output to the original message and the DSBSC signal's envelopes. You'll also observe the effect that a distorted DSBSC signal due to overloading has on the product detector's output.

Finally, if time permits, you'll investigate the effect on the product detector's performance of an unsynchronised local carrier.

It should take you about 1 hour to complete the whole experiment.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads
- One set of headphones (stereo)

Procedure

Part A - Setting up the DSBSC modulator

To experiment with DSBSC demodulation you need a DSBSC signal. The first part of the experiment gets you to set one up.

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELVISmx software.

8. Launch and run the NI ELVIS II Oscilloscope VI.

9. Set up the scope per the procedure in Experiment 1 ensuring that the Trigger Source control is set to CH 0.

10. Connect the set-up shown in Figure 2 below.

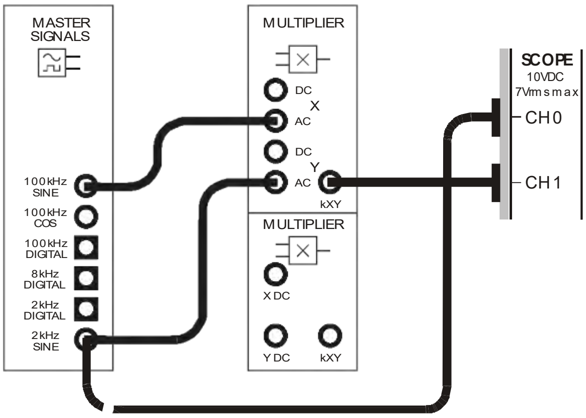


Figure 2

This set-up can be represented by the block diagram in Figure 3 below. It generates a 100kHz carrier that is DSBSC modulated by a 2kHz sinewave message.

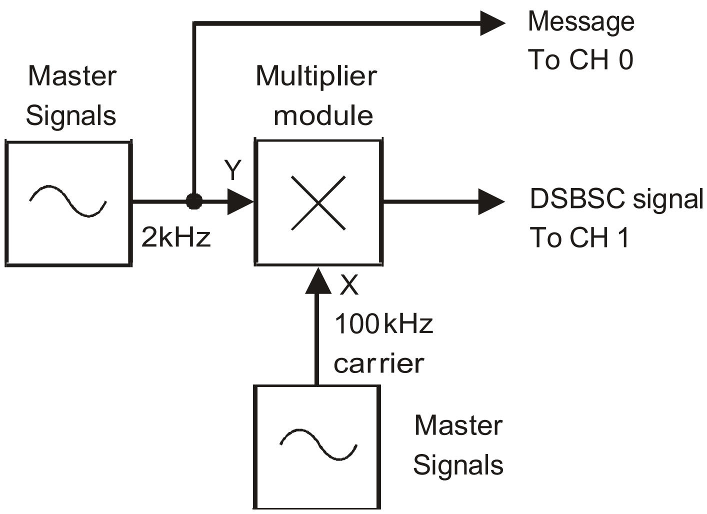


Figure 3

11. Adjust the scope's Timebase control to view two or so cycles of the Master Signals module's 2kHz SINE output.

12. Activate the scope's Channel 1 input to view the DSBSC signal out of the Multiplier module as well as the message signal.

    Note: If the Multiplier module's output is not a DSBSC signal, check your wiring.

13. Set the scope's Channel 0 Scale control to the 1V/div position and the Channel 1 Scale control to the 2V/div position.

14. Draw the two waveforms to scale in the space provided on the next page leaving room to draw a third waveform.

    Tip: Draw the message signal in the upper third of the graph and the DSBSC signal in the middle third.

Ask the instructor to check
your work before continuing.

Part B - Recovering the message using a product detector

15. Launch the DATEx soft front-panel (SFP) and check that you have soft control over the DATEx board.

16. Locate the Tuneable Low-pass Filter module on the DATEx SFP and set its soft Gain control to about the middle of its travel.

17. Turn the Tuneable Low-pass Filter module's soft Cut-off Frequency Adjust control fully clockwise.

18. Modify the set-up as shown in Figure 4 below.

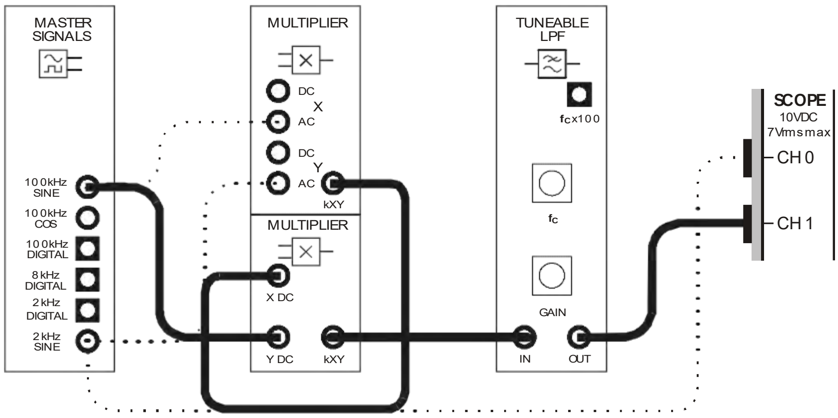


Figure 4

The additions to the set-up can be represented by the block diagram in Figure 5 on the next page. The Multiplier and Tuneable Low-pass Filter modules are used to implement a product detector which demodulates the original message from the DSBSC signal.

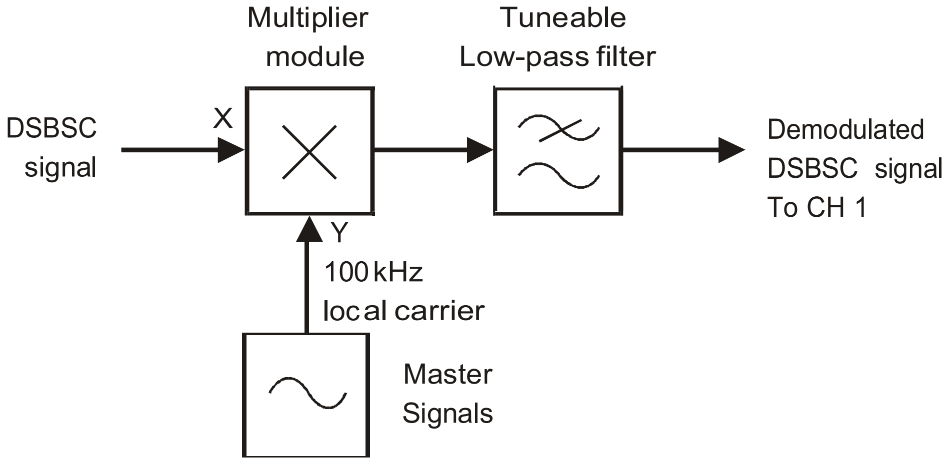


Figure 5

The entire set-up is represented by the block diagram in Figure 6 below. It highlights the fact that the modulator's carrier is "stolen" to provide the product detector's local carrier. This means that the two carriers are synchronised which is a necessary condition for DSBSC communications.

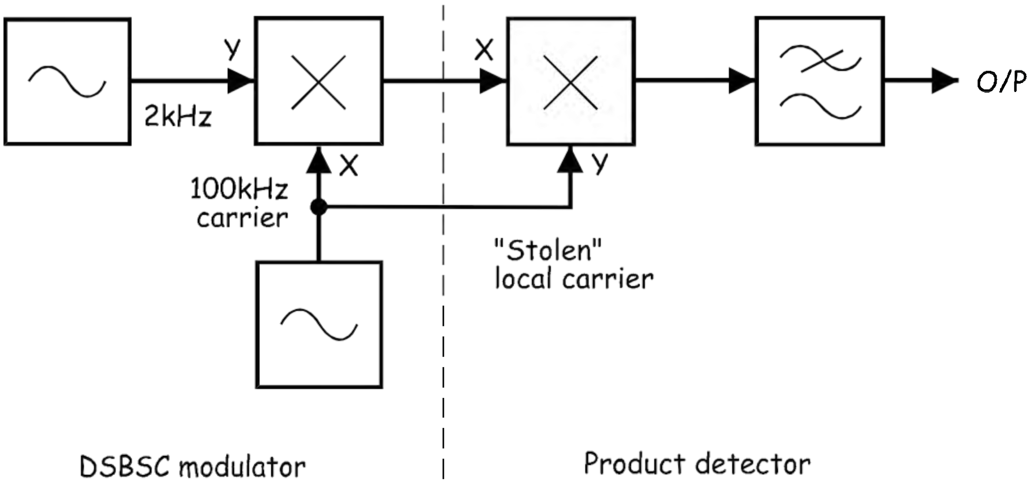


Figure 6

19. Draw the demodulated DSBSC signal to scale in the space that you left on the graph paper.

Question 1

Why must a product detector be used to recover the message instead of an envelope detector? Tip: If you're not sure, refer to the preliminary discussion.

Ask the instructor to check your work before continuing.

Part C - Investigating the message's amplitude on the recovered message

20. Locate the Amplifier module on the DATEx SFP and turn its soft Gain control to about a quarter of its travel.

21. Disconnect the plugs to the Master Signals module's 2kHz SINE output.

22. Use the Amplifier module to modify the set-up as shown in Figure 7 below.

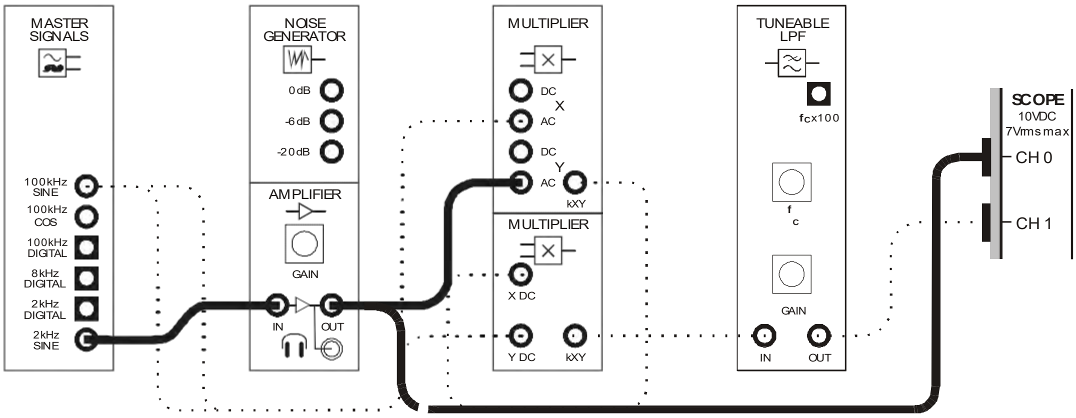


Figure 7

The addition to the set-up can be represented by the block diagram in Figure 8 below. The amplifier's variable gain allows the message's amplitude to be adjustable.

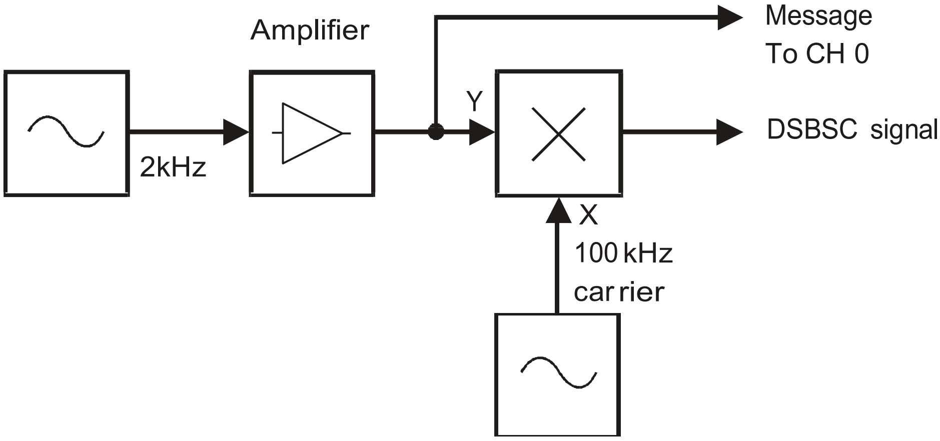


Figure 8

23. Vary the message signal's amplitude up and down a little (by turning the Amplifier module's soft Gain control left and right a little) while watching the demodulated signal.

Remember: You can use the keyboard's TAB and arrow keys for fine adjustments of DATEx controls.

Question 2
What is the relationship between the amplitude of the two message signals?

24. Slowly increase the message signal's amplitude to maximum until the demodulated signal begins to distort.

Question 3
What do you think causes the distortion of the demodulated signal? Tip: If you're not sure, connect the scope's Channel 0 input to the DSBSC modulator's output and set its Trigger Source control to the CH 1 position.

Ask the instructor to check
your work before continuing.

Part D - Transmitting and recovering speech using DSBSC

This experiment has set up a DSBSC communication system to "transmit" a 2kHz sinewave. The next part of the experiment lets you use it to modulate, transmit, demodulate and listen to speech.

25. If you moved the scope's Channel 0 input and adjusted its Trigger Source control to help answer Question 3, return them to their previous positions.

26. Disconnect the leads to the Amplifier module and modify the set-up as shown in Figure 9 below.

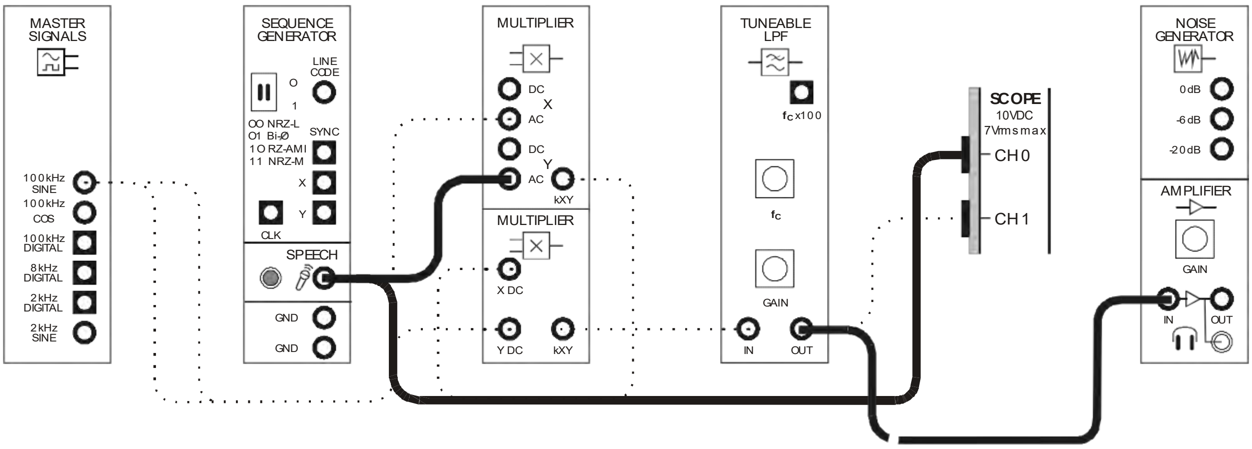


Figure 9

27. Set the scope's Timebase control to the 2ms/div position.

28. Turn the Amplifier module's soft Gain control fully anti-clockwise.

29. Without wearing the headphones, plug them into the Amplifier module's headphone socket.

30. Put the headphones on.

31. As you perform the next step, set the Amplifier module's soft Gain control to a comfortable sound level.

32. Hum and talk into the microphone while watching the scope's display and listening on the headphones.

Ask the instructor to check
your work before continuing.

Part E - Carrier synchronisation

Crucial to the correct operation of a DSBSC communications system is the synchronisation between the modulator's carrier signal and the product detector's local carrier. Any phase or frequency difference between the two signals adversely affects the system's performance.

The effect of phase errors

Recall that the product detector generates two copies of the message. Recall also that they're in phase with each other and so they simply add together to form one bigger message. However, if there's a phase error between the carriers, the product detector's two messages have a phase error also. One of them has the sum of the phase errors and the other the difference. In other words, the two messages are out of phase with each other.

If the carriers' phase error is small (say about 10°) the two messages still add together to form one bigger signal but not as big as when the carriers are in phase. As the carriers' phase error increases, the recovered message gets smaller. Once the phase error exceeds 45° the two messages begin to subtract from each other. When the carriers' phase error is 90° the two messages end up 180° out of phase and completely cancel each other out.

The next part of the experiment lets you observe the effects of carrier phase error.

33. Turn the Amplifier module's soft Gain control fully anti-clockwise again.

34. Return the scope's Timebase control to about the 100JJs/div position.

35. Locate the Phase Shifter module on the DATEx SFP and set its soft Phase Change control to the 180° position.

36. Set the Phase Shifter module's soft Phase Adjust control to about the middle of its travel.

37. Disconnect the leads to the Speech output and modify the set-up as shown in Figure 10 below.

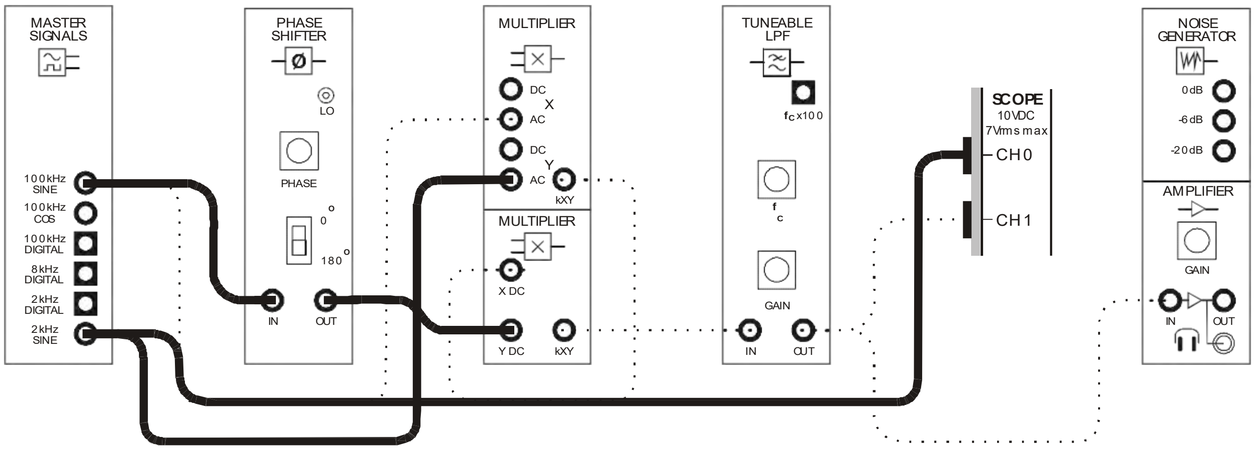


Figure 10

The set-up in Figure 10 on the previous page can be represented by the block diagram in Figure 11 below. The Phase Shifter module allows a phase error between the DSBSC modulator's carrier and the product detector's local carrier to be introduced.

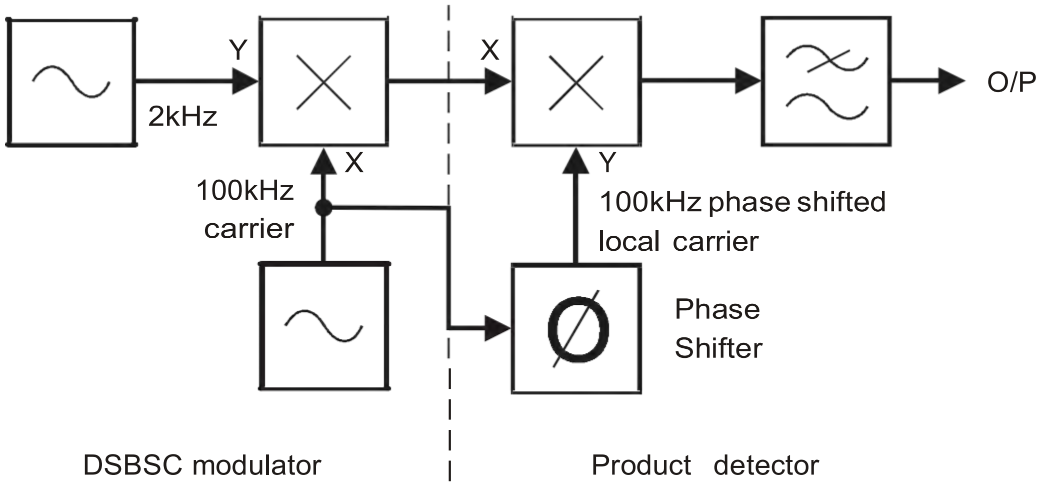


Figure 11

38. Slowly increase the Amplifier's module's gain until you can comfortably hear the demodulated 2kHz tone.

39. Vary the Phase Shifter module's soft Phase Adjust control left and right while watching and listening to the effect on the recovered message.

40. Use the keyboard's TAB and left arrow keys to turn the Phase Shifter module's soft Phase Adjust control anti-clockwise until the recovered message is smallest.

Question 4
Given the size of the recovered message's amplitude, what is the likely phase error between the two carriers? Tip: If you're not sure about the answer to this question (and the next one), reread the notes on page 9-13.

---

41. Verify your answer to Question 4 by connecting the scope's Channel 0 input to the Master Signals module's 100kHz SINE output, its Channel 1 input to the Phase Shifter module's output and setting its Timebase control to the 5JJs/div setting.

42. Use the keyboard's TAB and left arrow keys to adjust the Phase Shifter module's soft Phase Adjust control until the two signals are in phase.

Question 5
Given the two carriers are in phase, what should the amplitude of the recovered message be?

---

43. Verify your answer to Question 5 by reconnecting the scope's Channel 0 input to the Master Signals module's 2kHz SINE output, reconnecting its Channel 1 input to the Tuneable Low-pass Filter module's output and setting its Timebase control back to the 100JJs/div setting.

Ask the instructor to check
your work before continuing.

The effect of frequency errors

When there's a frequency error between the DSBSC signal's carrier and the product detector's local carrier, there is a corresponding frequency error in the two products that usually coincide. One is at the message frequency minus the error and the other is at the error frequency plus the error.

If the error is small (say 0.1Hz) the two signals will alternately reinforce and cancel each other which can render the message periodically inaudible but otherwise intelligible. If the frequency error is larger (say 5Hz) the message is reasonably intelligible but fidelity is poor. When frequency errors are large, intelligibility is seriously affected.

The next part of the experiment lets you observe the effects of carrier frequency error.

44. Launch and run the NI ELVIS II Function Generator VI.

45. Adjust the function generator's soft controls for an output with the following specifications:

- Waveshape: Sine
- Frequency: 100kHz exactly
- Amplitude: 4Vpp
- DC Offset: 0V

46. Disconnect the leads to the Phase Shifter module and modify the set-up as shown in Figure 12 below.

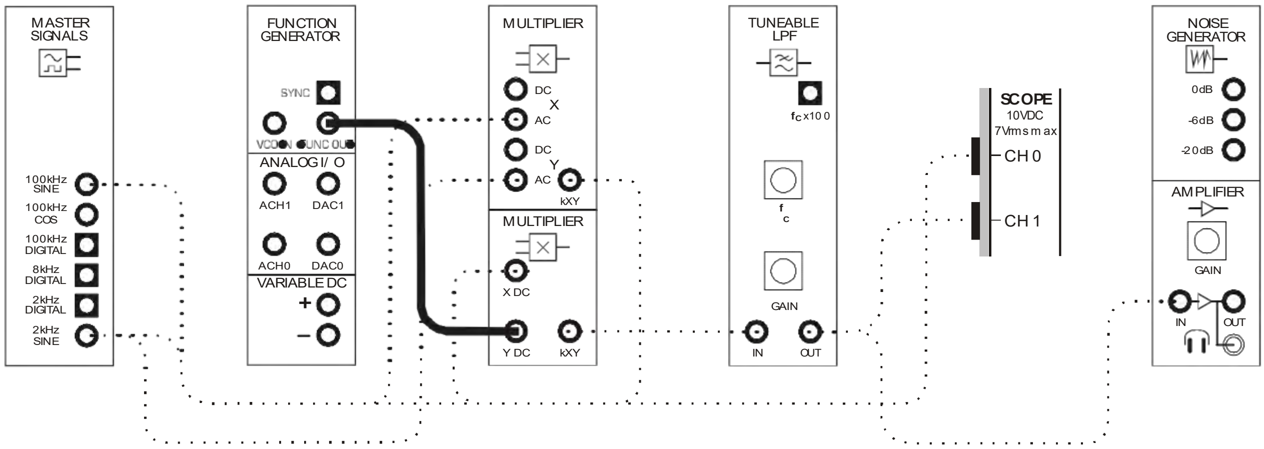


Figure 12

The entire set-up can be represented by the block diagram in Figure 13 below. The function generator allows the local oscillator to be completely frequency (and phase) independent of the DSBSC modulator.

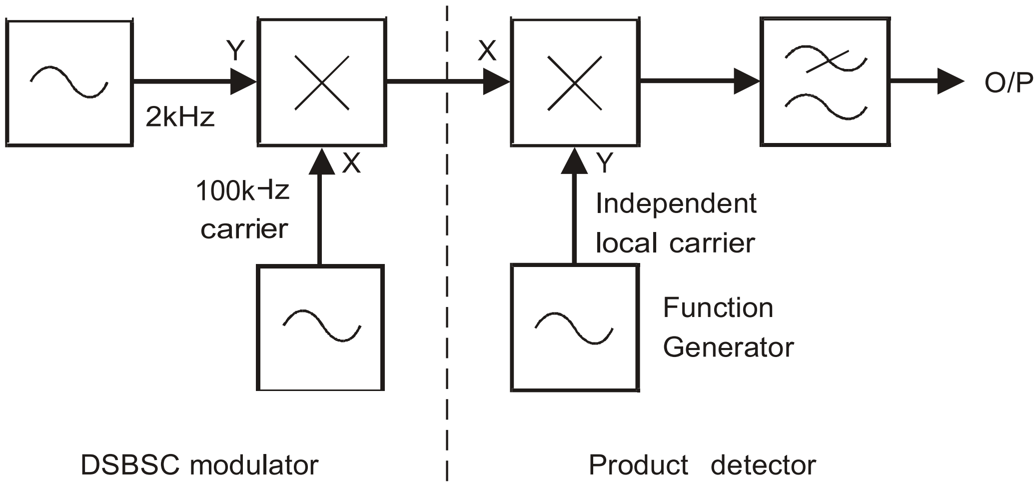


Figure 13

47. If you're not doing so already, listen to the recovered message using the headphones.

48. Compare the scope's frequency measurements for the original message and the recovered message.

    Note 1: You should find that they're very close in frequency.

    Note 2: You'll notice that the volume of the recovered messages varies. This is due to the phase error between the two carriers and should be ignored for the following steps.

49. Reduce the function generator's output frequency to 99.8kHz.

50. Give the function generator a moment to achieve the correct frequency and note the change in the tone of recovered message.

    Tip: If you can't remember what 2kHz sounds like, set function generator's output to 100kHz for a moment then return it to 99.8kHz.

51. Experiment with other local carrier frequencies around 100kHz and listen to the effect on the recovered message.

52. Return the function generator's output to 100kHz.

53. Disconnect the plugs to the Master Signals module's 2kHz SINE output and connect them to the Speech module's output.

54. Hum and talk into the microphone to check that the whole set-up is still working correctly.

55. Vary the function generator's frequency again and listen to the effect of an unsynchronised local carrier on speech.

Ask the instructor to check
your work before finishing.

week 8: Experiment 8 - SSBSC modulation and demodulation

Test Standard :IEEE 802

Preliminary discussion

Comparing the two communications systems considered earlier in this manual, DSBSC offers considerable power savings over AM (at least 66o) because a carrier is not transmitted. However, both systems generate and transmit sum and difference frequencies (the upper and lower sidebands) and so they have the same bandwidth for the same message signal.

As its name implies, the Single Sideband Suppressed Carrier (SSBSC or just SSB) system transmits only one sideband. In other words, SSB transmits either the sum or the difference frequencies but not both. Importantly, it doesn't matter which sideband is used because they both contain all of the information in the original message.

In transmitting only one sideband, SSB requires only half the bandwidth of DSBSC and AM which is a significant advantage.

Figure 1 below shows a simple message signal and an unmodulated carrier. It also shows the result of modulating the carrier with the message using SSBSC. If you look closely, you'll notice that the modulated carrier is not the same frequency as either the message or the carrier.

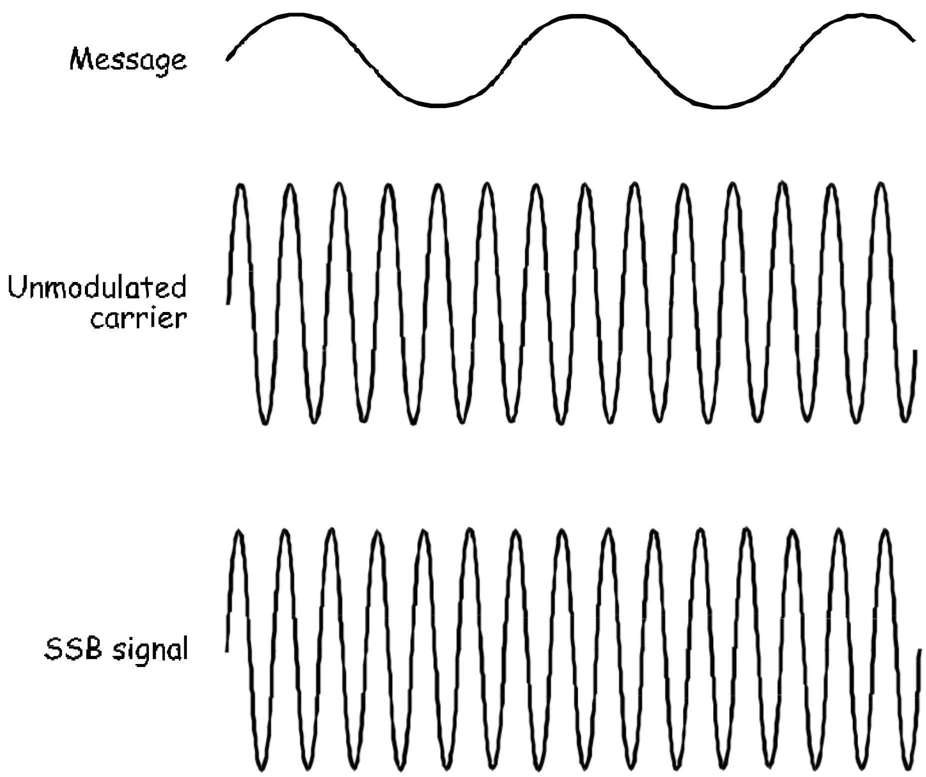


Figure 1

A common method of generating SSB simply involves generating a DSBSC signal then using a filter to pick out and transmit only one of the sidebands. This is known as the filter method. However, the two sidebands in a DSBSC signal are close together in frequency and so specialised filters must be used. This means that the filters for non-mainstream communications systems can be expensive.

Another way of generating SSB that is becoming increasingly popular is called the phasing method. This uses a technique called phase discrimination to cancel out one of the sidebands at the generation stage (instead of filtering it out afterwards).

In telecommunications theory, the mathematical model that defines this process is:

SSB = (message x carrier) + (message with 90° of phase shift x carrier with 90° of phase shift)

If you look closely at the equation you'll notice that it's the sum of two multiplications. When the message is a simple sinewave the solution of the two multiplications tells us that four sinewaves are generated. Depending on whether the message's phase shift is +90° or -90° their frequencies and phase differences are:

These.	Or these.
• Carrier + message	• Carrier + message
• Carrier - message	• Carrier - message
• Carrier + message	• Carrier + message (180° phase shifted)
• Carrier - message (180° phase shifted)	• Carrier - message

Regardless of whether the message's phase shift is +90° or -90°, when the four sinewaves are added together, two of them are in phase and add together to produce one sinewave (either carrier + message or carrier - message) and two of the sinewaves are phase inverted and completely cancel. In other words, the process produces only a sum or difference signal (that is, just one sideband).

The block diagram that implements the phasing type of SSB modulator is shown in Figure 2 below.

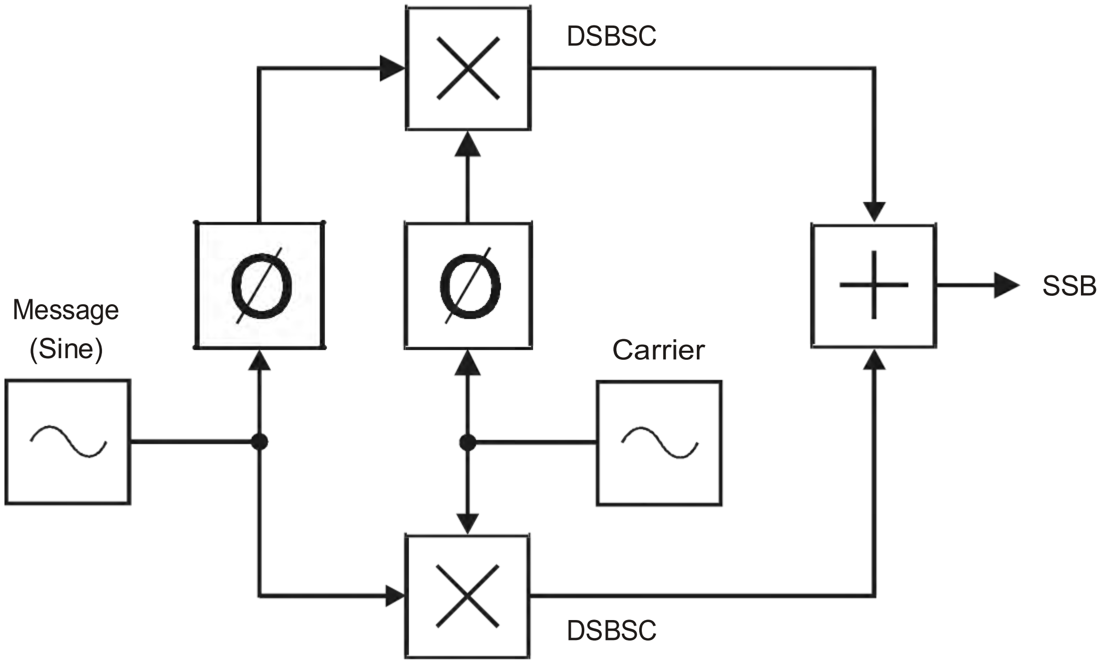


Figure 2

As SSB signals don't contain a carrier, they must be demodulated using product detection in the same way as DSBSC signals (the product detector's operation is summarised in the preliminary discussion of Experiment 9).

The experiment

For this experiment you'll use the Emona DATEx to generate an SSB signal by implementing the mathematical model for the phasing method. You'll then use a product detector (with a stolen carrier) to reproduce the message.

Importantly, you'll only do so for a sinewave message (that is, you'll not SSB modulate then demodulate speech). There's a practical reason for this. The phase shift introduced by the DATEx Phase Shifter module is frequency dependent (that is, for any given setting, the phase shift is different at different frequencies). A wideband phase shifting circuit is necessary to provide 90° of phase shift for all of the sinewaves in a complex message like speech.

It should take you about 40 minutes to complete this experiment.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads

Procedure

Part A - Generating an SSB signal using a simple message

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELVISmx software.

8. Launch and run the NI ELVIS II Function Generator VI.

9. Adjust the function generator using its soft controls for an output with the following specifications:

   - Waveshape: Sine
   - Frequency: 10kHz exactly
   - Amplitude: 4Vpp
   - DC Offset: 0V

10. Minimise the function generator's VI.

11. Connect the set-up shown in Figure 3 below.

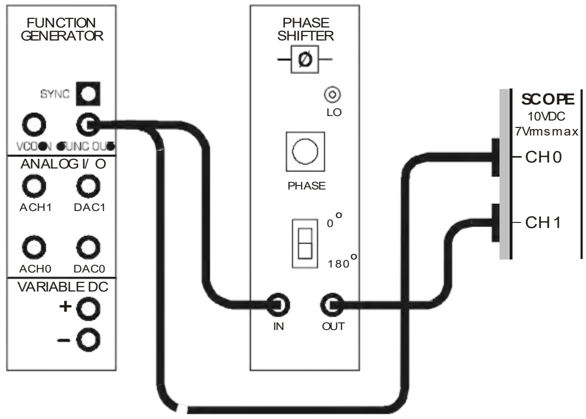


Figure 3

This set-up can be represented by the block diagram in Figure 4 below. It is used to set up two message signals that are out of phase with each other.

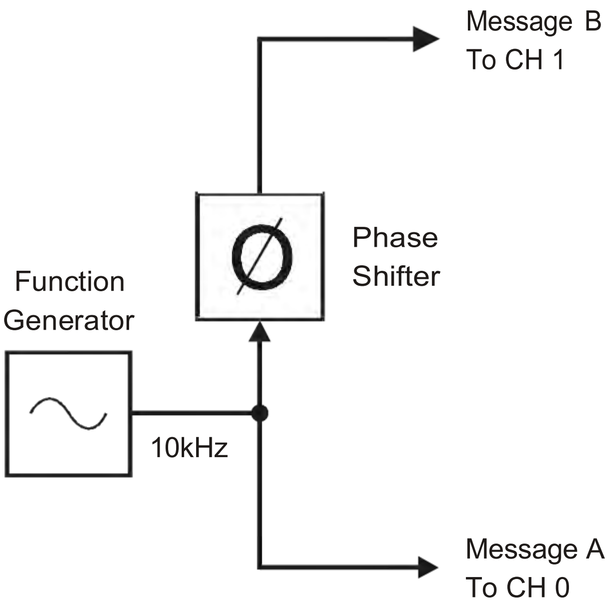


Figure 4

12. Launch the DATEx soft front-panel (SFP) and check that you have soft control over the DATEx board.

13. Locate the Phase Shifter module on the DATEx SFP and set its soft Phase Change control to the 180° position.

14. Set the Phase Shifter module's soft Phase Adjust control to about the middle of its travel.

15. Launch and run the NI ELVIS II Oscilloscope VI.

16. Set up the scope per the procedure in Experiment 1 ensuring that the Trigger Source control is set to CH 0.

17. Adjust the scope's Timebase control to view two or so cycles of the function generator's output.

18. Activate the scope's Channel 1.

19. Check that the two message signals are out of phase with each other.

    Note: At this stage, it doesn't matter what the phase difference is.

20. Modify the set-up as shown in Figure 5 below.

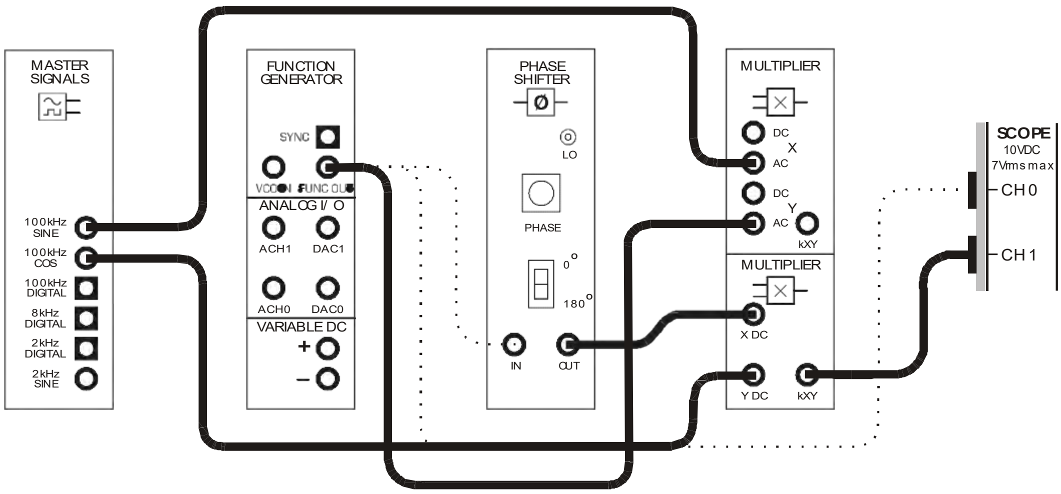


Figure 5

This set-up can be represented by the block diagram in Figure 6 on the next page. It is used to multiply the two message signals with two 100kHz sinewaves (the carriers) that are exactly 90° out of phase with each other.

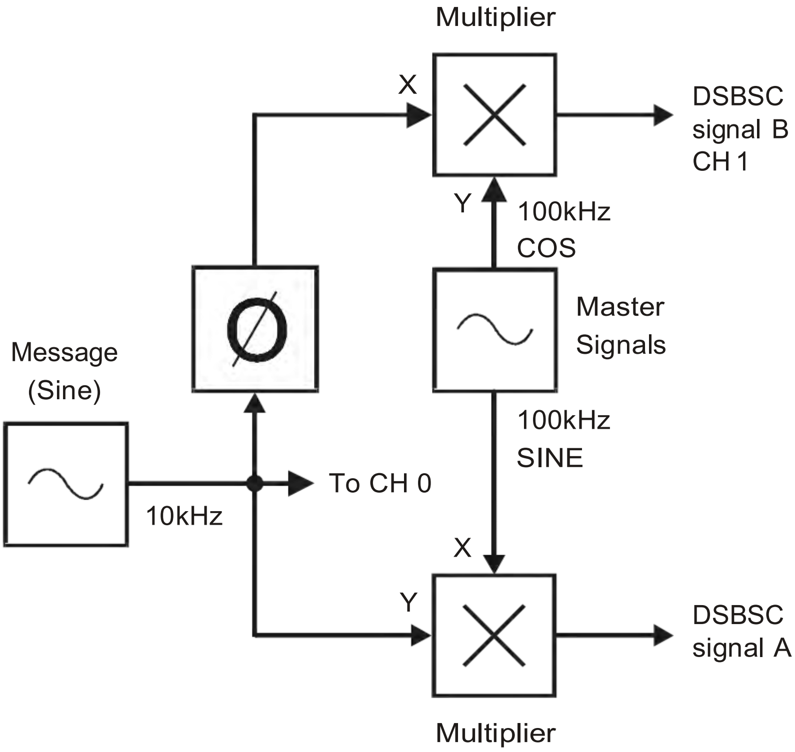


Figure 6

21. Use the scope to check that the lower Multiplier module's output is a DSBSC signal.

    Tip: Temporarily set the scope's Channel 1 Scale control to the 2V/div position to do this.

22. Disconnect the scope's Channel 1 input from the lower Multiplier module's output and connect it to the upper Multiplier module's output.

23. Check that the upper Multiplier module's output is a DSBSC signal as well.

24. Locate the Adder module on the DATEx SFP and set its soft G and g controls to about the middle of their travel.

25. Modify the set-up as shown in Figure 7 below.

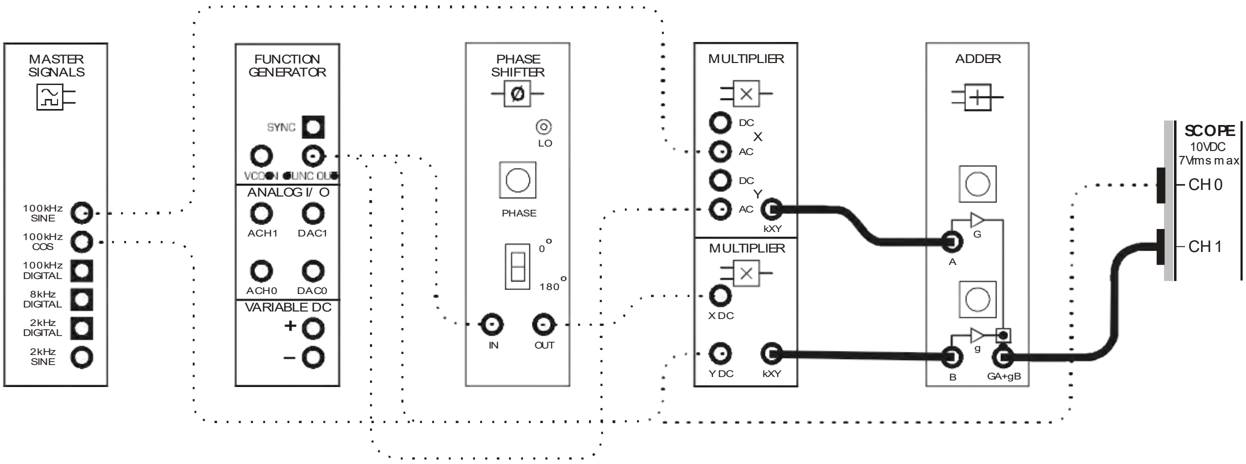


Figure 7

This set-up can be represented by the block diagram in Figure 8 below. The Adder module is used to add the two DSBSC signals together. The phase relationships between the sinewaves in the DSBSC signals means that two of them (one in each sideband) reinforce each other and the other two cancel each other out.

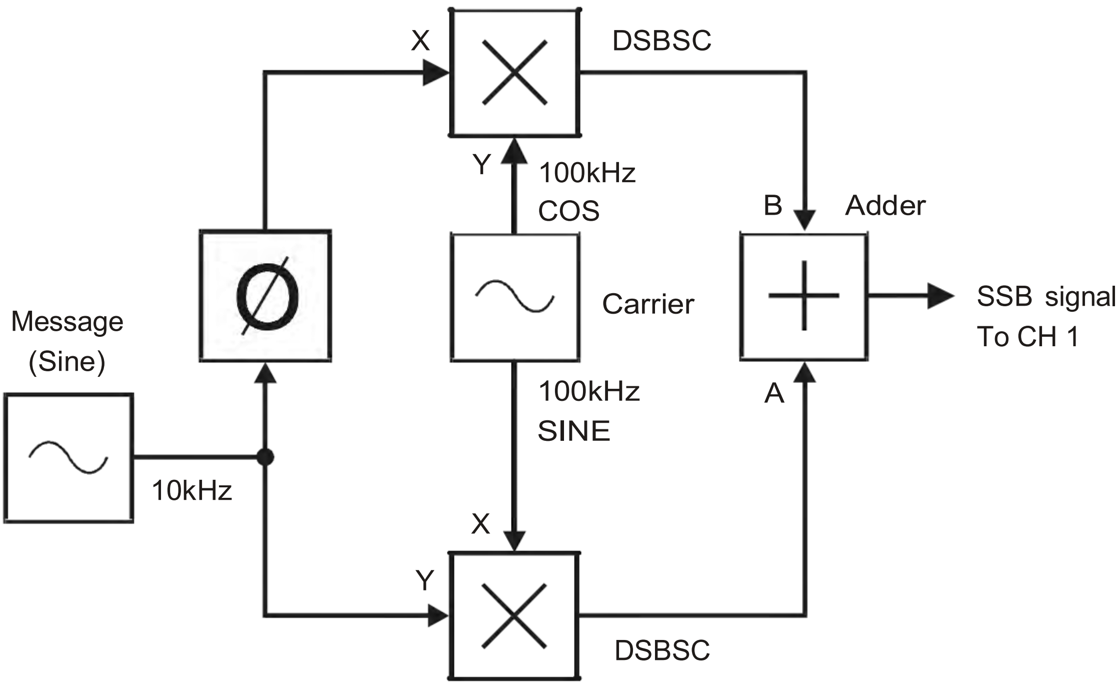


Figure 8

Question 1

The signal out of the Adder module is highly unlikely to be an SSB signal at this stage. What are two reasons for this? Tip: If you're not sure, one of them can be worked out by reading the preliminary discussion.

Ask the instructor to check your work before continuing.

The next part of the experiment gets you to make the fine adjustments necessary to turn the set-up into a true SSB modulator.

26. Deactivate the scope's Channel 0 input.

27. Disconnect the patch lead to the Adder module's B input.

    Note: This removes the signal on the Adder module's B input from the set-up's output.

28. Adjust the Adder module's soft G control to obtain a 4Vp-p output.

    Tip: Remember that you can use the keyboard's TAB and arrow keys for fine adjustment of the DATEx SFP's controls.

29. Reconnect the Adder module's B input and disconnect the patch lead to its A input.

    Note: This removes the signal on the Adder module's A input from the set-up's output.

30. Adjust the Adder module's soft g control to obtain a 4Vp-p output.

31. Reconnect the patch lead to the Adder module's A input.

The gains of the Adder module's two inputs are now nearly the same. Next, the correct phase difference between the messages must be achieved.

32. Slowly vary the Phase Shifter module's soft Phase Adjust control left and right and observe the effect on the envelopes of the set-up's output.

    Note: For most of the soft Phase Adjust control's travel, you'll get an output that looks like a DSBSC signal. However, if you adjust the control carefully, you'll find that you're able to flatten-out the output signal's envelope.

33. Set the scope's Channel 1 Scale control to the 500mV/div position.

34. Adjust the Phase Shifter module's soft Phase Adjust control to make the envelopes as "flat" as possible.

The phase difference between the two messages is now nearly 90°.

35. Tweak the Adder module's soft G control to see if you can make the output's envelopes flatter.

36. Tweak the Phase Shifter module's soft Phase Adjust control to see if you can make the output's envelopes flatter still.

Once the envelopes are as flat as you can get, the gains of the Adder module's two inputs are very close to each other and the phase difference between the two messages are very close to 90°. That being the case, the signal out of the Adder module is now SSBSC.

Question 2
How many sinewaves does this SSB signal consist of? Tip: If you're not sure, see the preliminary discussion.

Question 3
For the given inputs to the SSB modulator, what two frequencies can this signal be?

Ask the instructor to check
your work before continuing.

Part B – Spectrum analysis of an SSB signal

The next part of this experiment let's you analyse the frequency domain representation of the SSB signal to see if its spectral composition matches your answers to Questions 2 and 3.

37. Suspend the scope's operation by clicking on its Stop control once.

    Note: The scope's display should freeze and its hardware has been deactivated. This is a necessary step as the scope and signal analyzer share hardware resources and so they cannot be operated simultaneously.

38. Launch and run the NI ELVIS II Dynamic Signal Analyzer VI.

39. Set up the signal analyzer per the procedure in Experiment 7 (page 7-10).

40. Activate the signal analyzer's cursors by checking the Cursors On box.

41. Align C1 with the most significant sinewave in the signal's spectrum and determine its frequency.

    Question 4
    Based on your measurement for the step above, which sideband does your SSB modulator generate?

    ____

42. Align C1 with some of the other significant sinewaves close to this sideband and note their frequencies.

    Note: You should find that there's a sinewave at the carrier frequency and another at the frequency for the other sideband. Importantly, despite appearances, these signals are very small relative to the significant sideband (the scale used for the Y-axis is decibels which is not a linear unit of measurement).

    Question 5
    Give two reasons for the presence of a small amount of the other sideband.

    ____

    ____

43. Tweak the Phase Shifter module's soft Phase Adjust control and note the effect on the size of the carrier and other sideband.

Note: Give the signal analyzer's display time to update after each adjustment.

Question 6
Why doesn't varying the Phase Shift module's Phase Adjust control affect the size of the carrier in the SSBSC signal?

____

____

44. Adjust the two controls to obtain the smallest size for the insignificant sideband.

45. Close the Signal Analyzer's VI.

46. Restart the scope's VI by clicking on its Run control once.

47. Note whether there is any improvement in the SSB signal's envelope (that is, note whether the envelope is any flatter).

Ask the instructor to check
your work before continuing.

Part C - Using the product detector to recover the message

48. Reactivate the scope's Channel 0 input and return the Channel 1 Scale control to the 1V/div position.

49. Locate the Tuneable Low-pass Filter module on the DATEx SFP and set its soft Gain control to about the middle of its travel.

50. Turn the Tuneable Low-pass Filter module's soft Cut-off Frequency Adjust control fully clockwise.

51. Modify the set-up as shown in Figure 9 below.

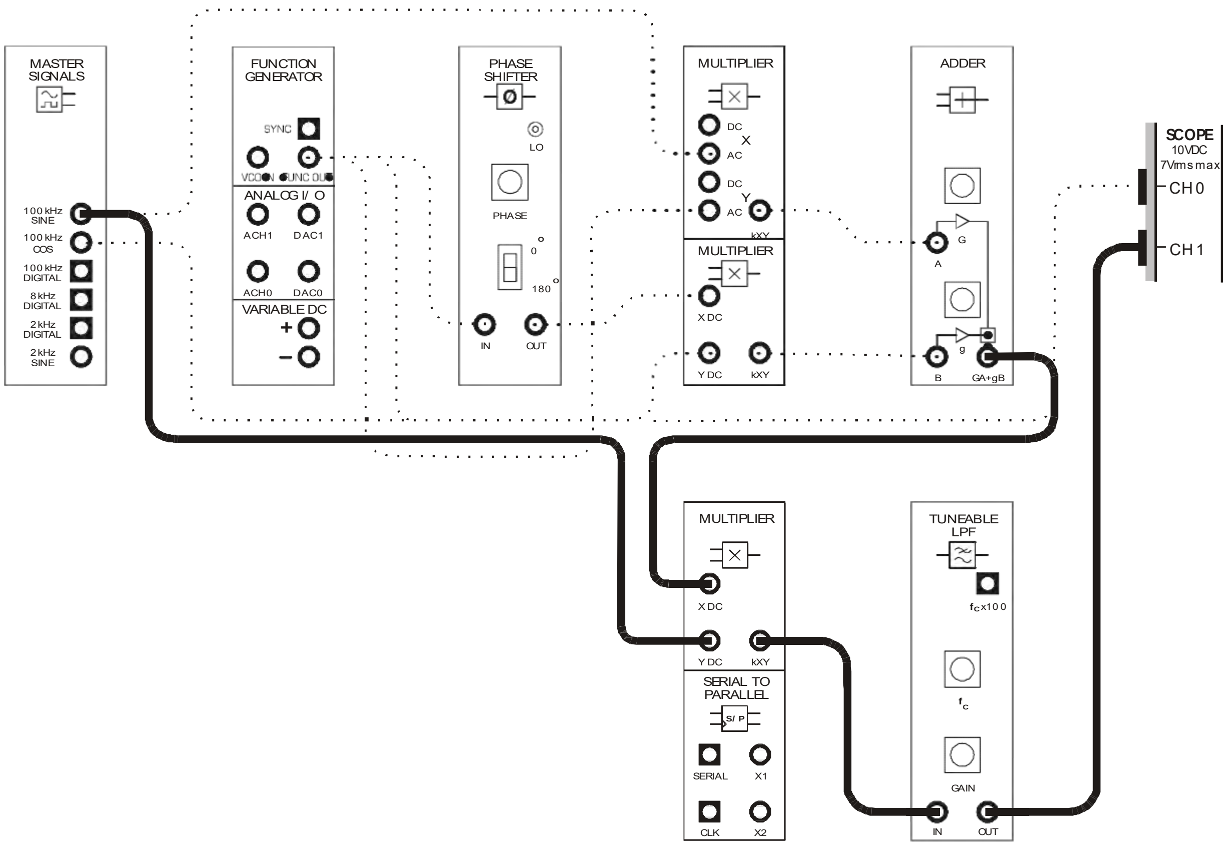


Figure 9

The additions to the set-up shown in Figure 9 on the previous page can be represented by the block diagram in Figure 10 below. The Multiplier and Tuneable Low-pass Filter modules are used to implement a product detector which demodulates the original message from the SSB signal.

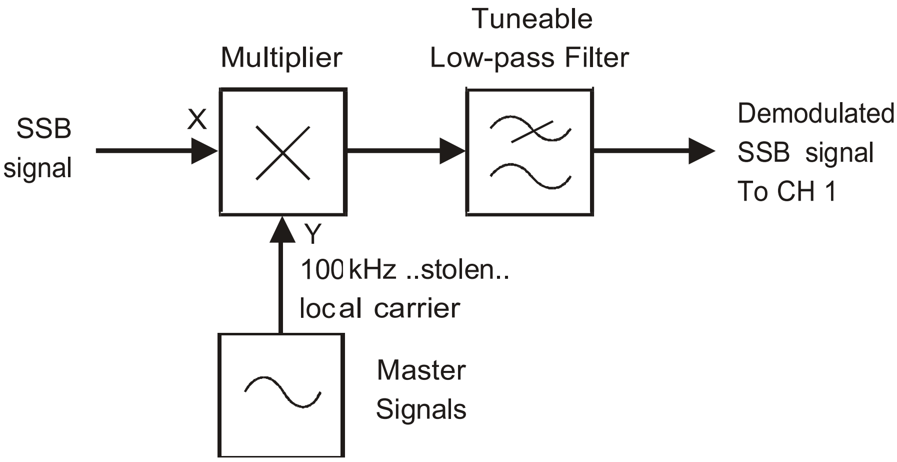


Figure 10

52. Use the scope to compare the original message with the recovered message.

Question 7
What is the relationship between the original message and the recovered message?

Ask the instructor to check
your work before finishing

Week 9: Experiment 9 - Frequency modulation

Test Standard :IEEE 185-1974

Preliminary discussion

A disadvantage of the AM, DSBSC and SSB communication systems is that they are susceptible to picking up electrical noise in the transmission medium (the channel). This is because noise changes the amplitude of the transmitted signal and the demodulators of these systems are designed to respond to amplitude variations.

As its name implies, frequency modulation (FM) uses a message's amplitude to vary the frequency of a carrier instead of its amplitude. This means that the FM demodulator is designed to look for changes in frequency instead. As such, it is less affected by amplitude variations and so FM is less susceptible to noise. This makes FM a better communications system in this regard.

There are several methods of generating FM signals but they all basically involve an oscillator with an electrically adjustable frequency. The oscillator uses an input voltage to affect the frequency of its output. Typically, when the input is OV, the oscillator outputs a signal at its rest frequency (also commonly called the free-running or centre frequency). If the applied voltage varies above or below OV, the oscillator's output frequency deviates above and below the rest frequency. Moreover, the amount of deviation is affected by the amplitude of the input voltage. That is, the bigger the input voltage, the greater the deviation.

Figure 1 below shows a bipolar squarewave message signal and an unmodulated carrier. It also shows the result of frequency modulating the carrier with the message.

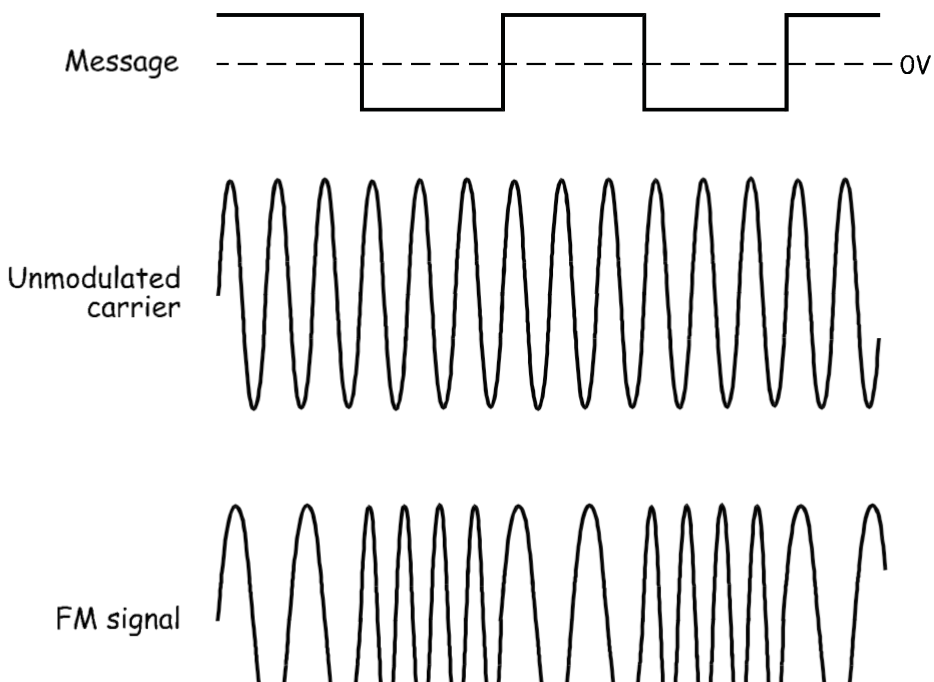


Figure 1

There are a few things to notice about the FM signal. First, its envelopes are flat - recall that FM doesn't vary the carrier's amplitude. Second, its period (and hence its frequency) changes when the amplitude of the message changes. Third, as the message alternates above and below OV, the signal's frequency goes above and below the carrier's frequency. (Note: It's equally possible to design an FM modulator to cause the frequency to change in the opposite direction to the change in the message's polarity.)

Before discussing FM any further, an important point must be made here. A squarewave message has been used in this discussion to help you visualise how an FM carrier responds to its message. In so doing, Figure 1 suggests that the resulting FM signal consists of only two sinewaves (one at a frequency above the carrier and one below). However, this isn't the case. For reasons best left to your instructor to explain, the spectral composition of the FM signal in Figure 1 is much more complex than implied.

This highlights one of the important differences between FM and the modulation schemes discussed earlier. The mathematical model of an FM signal predicts that even for a simple sinusoidal message, the result is a signal that potentially contains many sinewaves. In contrast, for the same sinusoidal message, an AM signal would consist of three sinewaves, a DSBSC signal would consist of two and an SSBSC signal would consist of only one. This doesn't automatically mean that the bandwidth of FM signals is wider than AM, DSBSC and SSBSC signals (for the same message signal). However, in the practical implementation of FM communications, it usually is.

There's another important difference between FM and the modulation schemes discussed earlier. The power in AM, DSBSC and SSBSC signals varies depending on their modulation index. This occurs because the carrier's RMS voltage is fixed but the RMS sideband voltages are proportional to the signals' modulation index. This is not true of FM. The RMS voltage of the carrier and sidebands varies up and down as the modulation index changes such that the square of their voltages always equal the square of the unmodulated carrier's RMS voltage. That being the case, the power in FM signals is constant.

Finally, when reading about the operation of an FM modulator you may have recognised that there is a module on the Emona DATEx that operates in the same way - the VCO output of the function generator. In fact a voltage-controlled oscillator is sometimes used for FM modulation (though there are other methods with advantages over the VCO).

The experiment

For this experiment you'll generate a real FM signal using the VCO module on the Emona DATEx. First you'll set up the VCO module to output an unmodulated carrier at a known frequency. Then you'll observe the effect of frequency modulating its output with a squarewave then speech. You'll then use the NI ELVIS II Dynamic Signal Analyzer to observe the spectral composition of an FM signal in the frequency domain and examine the distribution of power between its carrier and sidebands for different levels of modulation.

It should take you about 4O minutes to complete this experiment.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads

Procedure

Part A - Frequency modulating a squarewave

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.
2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.
3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.
4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.
5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.
6. Turn on the PC and let it boot-up.
7. Launch the NI ELVISmx software.
8. Launch and run the NI ELVIS II Function Generator VI.
9. Adjust the function generator using its soft controls for an output with the following specifications:

   - Waveshape: Sine
   - Frequency: 2OkHz
   - Amplitude: 4Vpp
   - DC Offset: OV
   - Modulation type: FM

1O. Connect the set-up shown in Figure 2 below.

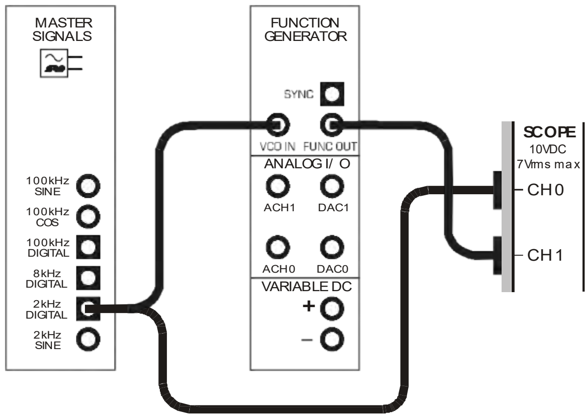


Figure 2

This set-up can be represented by the block diagram in Figure 3 below. The Master Signals module is used to provide a 2kHz squarewave message signal and the VCO is the FM modulator with a 2OkHz carrier.

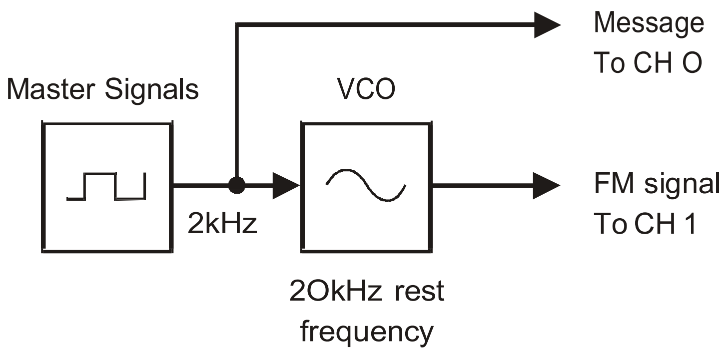


Figure 3

11. Launch and run the NI ELVIS II Oscilloscope VI.

12. Set up the scope per the procedure in Experiment 1 with the following changes:

- Timebase control to the 1OO.us/div position instead of 5OO.us/div
- Trigger Level to the 2.5V position instead of OV

13. Activate the scope's Channel 1 input to view the FM signal on the VCO's output as well as the message signal.

Question 1
Why does the frequency of the carrier change?

____

____

Ask the instructor to check
your work before continuing.

Part B - Generating an FM signal using speech

So far, this experiment has generated an FM signal using a squarewave for the message. However, the message in commercial communications systems is much more likely to be speech and music. The next part of the experiment lets you see what an FM signal looks like when modulated by speech.

14. Return the scope's Trigger Level control to OV.

15. Disconnect the plugs to the Master Signals module's 2kHz SINE output.

16. Connect them to the Speech module's output as shown in Figure 4 below.

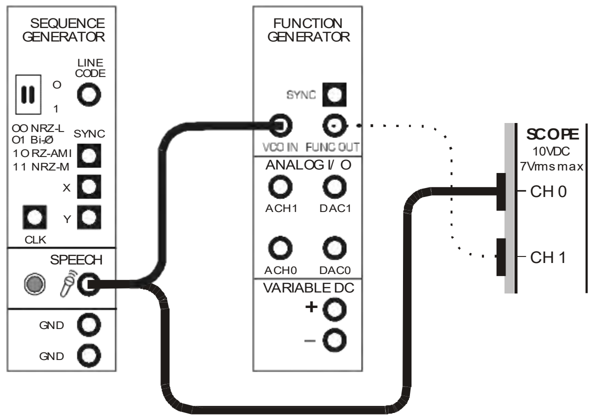


Figure 4

17. Set the scope's Timebase control to the 2OO.us/div position.

18. Hum and talk into the microphone while watching the scope's display.

Ask the instructor to check your work before continuing.

Part C - Power in an FM signal
As mentioned earlier, the power in an FM signal is constant regardless of its level of modulation. This part of the experiment lets you see this for yourself.

19. Launch the DATEx soft front-panel (SFP) and check that you have soft control over the DATEx board.

2O. Locate the Amplifier module on the DATEx SFP and turn soft Gain control fully anti-clockwise.

21. Completely dismantle the previous set-up.

22. Connect the set-up shown in Figure 5 below.

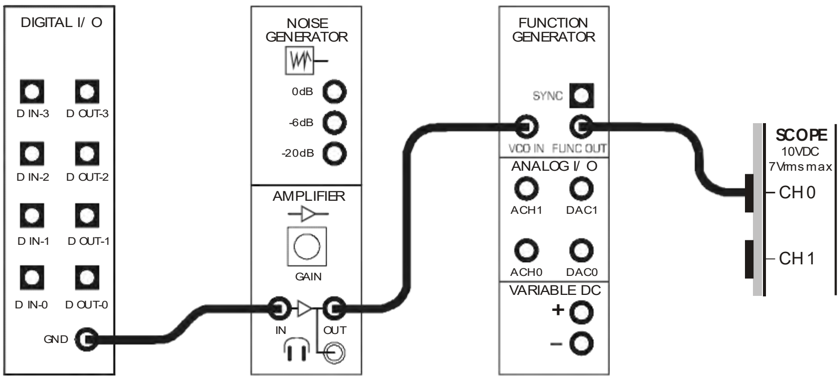


Figure 5

This set-up can be represented by the block diagram in Figure 6 on the next page. With the VCO's input connected to ground, its output is a single sinewave at 2OkHz.

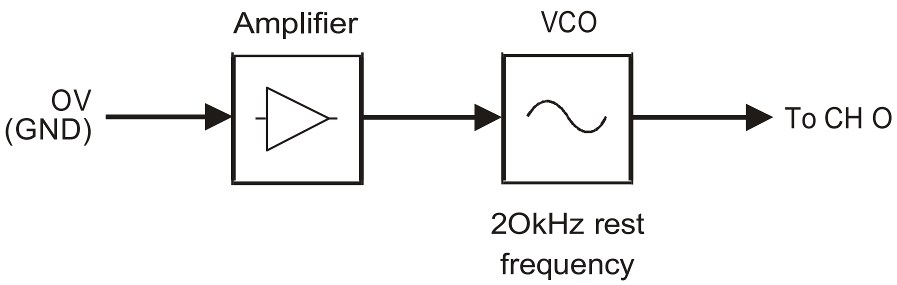


Figure 6

23. Use the scope to check that the VCO's output is a 2OkHz sinewave.

24. Close the scope's VI.

25. Launch and run the NI ELVIS II Dynamic Signal Analyzer VI.

26. Adjust the signal analyzer's controls as follows:

Input Settings

- Source Channel to SCOPE CH O
- Voltage Range to ±1OV

FFT Settings

- Frequency Span to 6O,OOO
- Resolution to 4OO
- Window to 7 Term B-Harris

Averaging

- Mode to RMS
- Weighting to Exponential
- # of Averages to 3

Trigger Settings

- Type to Edge

Frequency Display

- Units to Linear
- Mode to RMS
- Scale to Auto

Cursor settings

- Cursors On box unchecked (for now)

27. Once done, one significant sinewave should be displayed.

Note: It's important at this point to double-check that the signal analyzer's Frequency Display Units option is set to Linear.

28. Activate the signal analyzer's cursors by checking the Cursors On box.

29. Use the scope's C1 cursor to measure the frequency of the sinewave and verify that it's the VCO's rest frequency (that is, 2OkHz).

Note: Recall that cursor measurements of frequency and voltage are difference values between C1 and C2. However, when cursor C2 is moved to the extreme left side of the display, its frequency and voltage is zero so the cursor measurements become absolute values for C1.

3O. To the left of the cursor's frequency measurement readout is the measurement of the signal's RMS voltage. Record this in Table 1 below.

31. Square and record this voltage.

Table 1

Unmodulated Carrier V_{RMS}	Unmodulated Carrier V_{RMS}^2

Why square the signal's RMS voltage? To answer this question, remember that we're investigating the power in an FM signal but signal analyzers (and most other test equipment) can't

measure power. However, one of the power equations ($P = \frac{V_{RMS}^2}{R}$) tells us that power and the square of a signal's RMS voltage (that is, V_{RMS}^2) are proportional values. That being the case, we can investigate power in an FM signal indirectly by investigating the square of the signal's RMS voltage because whatever is true of one must also be true of the other (regardless of R).

32. Disconnect the plug to the GND socket and modify the set-up as shown in Figure 7 below.

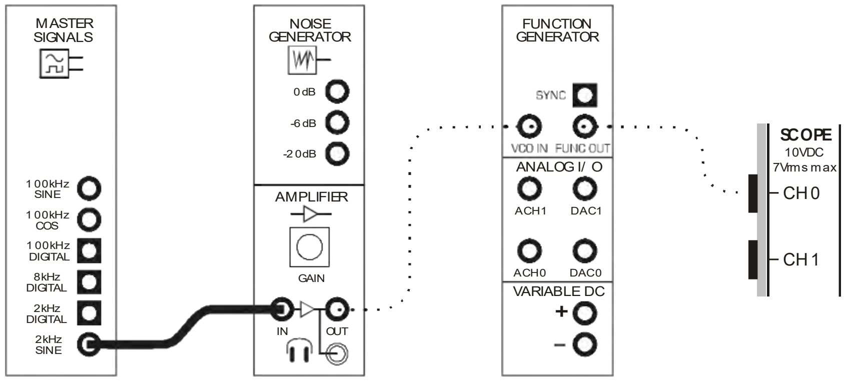


Figure 7

This set-up can be represented by the block diagram in Figure 8 below. Importantly, as the Amplifier module's minimum gain isn't zero, the carrier will now be frequency modulated by a low level message signal. This means that the signal analyzer's display will show about four sidebands. As these sidebands are small relative to the carrier, they can be better observed by temporarily setting the analyzer's Units option to dB instead of Linear.

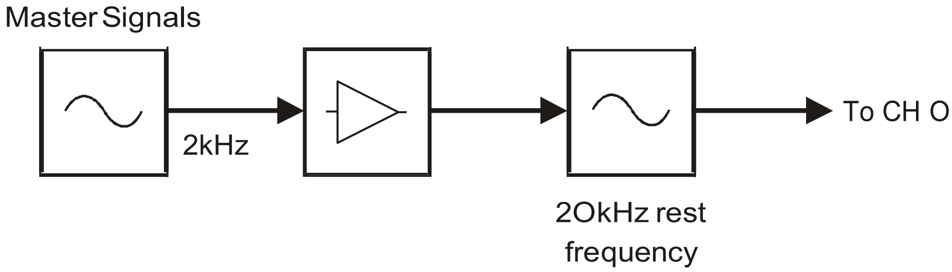


Figure 8

33. If you haven't already done so, return the analyzer's Units option to Linear.

34. Use the Amplifier module's soft Gain control to adjust the modulation of the FM signal slightly until only five sinewaves are clearly visible in the signal's spectrum.

35. Use the cursor to measure the RMS voltage of these sinewaves and record them in Table 2 on the next page.

36. Square and record the voltages.

37. Add and record the squared voltages.

Table 2

Sinewave	V_{RMS}	V^2_{RMS}
1		
2		
3		
4		
5		
	Total	

38. Use the Amplifier module's soft Gain control to increase the modulation of the FM signal until the carrier drops to zero for the first time.

39. Repeat Steps 35 and 37 for the six most significant sinewaves in the signal recording your measurements in Table 3 below.

Table 3

Sinewave	V_{RMS}	V^2_{RMS}
1		
2		
3		
4		
5		
6		
	Total	

Question 2
How do the totals in Tables 2 and 3 compare with each other and the value in Table 1?

____

____

Question 3
What do these measurements help to prove? Explain your answer.

____

____

____

Ask the instructor to check
your work before continuing.

Part D - Bandwidth of an FM signal

The spectral composition of an FM signal can be complex and consist of many sidebands. Usually, many of them are relatively small in size and so an engineering decision must be made about how many of them to include as part of the signal's bandwidth. There are several standards in this regard and a common one involves including all sidebands that are equal to or greater than 1o of the unmodulated carrier's power (or V^2_{RMS}). This part of the experiment lets you use this criterion to measure FM signal bandwidth.

4O. Use the signal analyzer's C1 cursor to identify the lowest frequency sinewave in the FM signal with a V^2_{RMS} equal to or greater than 1o of the value in Table 1.

Note: You have to do this by measuring the RMS voltage of the smallest sinewaves and square the value until you find the first one with a V^2_{RMS} equal to or greater than 1o of the value in Table 1.

41. Use the signal analyzer's C2 cursor to identify the highest frequency sinewave in the FM signal with a voltage equal to or greater than 1o of the value in Table 1.

42. The signal analyzer's df (Hz) reading is a measurement of the difference in frequency between its cursors. Following Steps 4O and 41, this reading is the FM signal's bandwidth. Record this value in Table 4 below.

Table 4

FM signal's bandwidth

Question 4

Calculate the bandwidth of a 2OkHz carrier amplitude modulated by a 2kHz sinewave.

____

____

Question 5

How does the FM signal's bandwidth compare to an AM signal's bandwidth for the same inputs?

____

____

Ask the instructor to check
your work before continuing.

43. Increase the Amplifier module's gain until the cursor on its soft Gain control points to the 9 o'clock position.

44. Repeat steps 4O to 42 recording your measurement in Table 5 below.

Table 5

FM signal's bandwidth

Question 6
What is the relationship between the message signal's amplitude and the FM signal's bandwidth?

Ask the instructor to check
your work before finishing

Week 10:Experiment 10 - FM demodulation

Test Standard IEEE 185-1974

Preliminary discussion

There are as many methods of demodulating an FM signal as there are of generating one. Examples include: the slope detector, the Foster-Seeley discriminator, the ratio detector, the phase-locked loop (PLL), the quadrature FM demodulator and the zero-crossing detector. It's possible to implement several of these methods using the Emona DATEx but, for an introduction to the principles of FM demodulation, the zero-crossing detector is used here.

The zero-crossing detector

The zero-crossing detector is a simple yet effective means of recovering the message from FM signals. Its block diagram is shown in Figure 1 below.

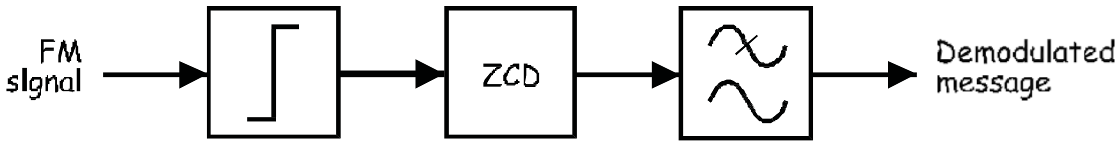


Figure 1

The received FM signal is first passed through a comparator to clip it heavily, effectively converting it to a squarewave. This allows the signal to be used as a trigger signal for the zero-crossing detector circuit (ZCD).

The ZCD generates a pulse of fixed duration every time the squared-up FM signal crosses zero volts (either on the positive or the negative transition but not both). Given the squared-up FM signal is continuously crossing zero, the ZCD effectively converts the squarewave to a rectangular wave with a fixed mark time.

When the FM signal's frequency changes (in response to the message), so does the rectangular wave's frequency. Importantly though, as the rectangular wave's mark is fixed, changing its frequency is achieved by changing the duration of the space and hence the signal's mark/space ratio (or duty cycle). This is shown in Figure 2 on the next page using an FM signal that only switches between two frequencies (because it has been generated by a squarewave for the message).

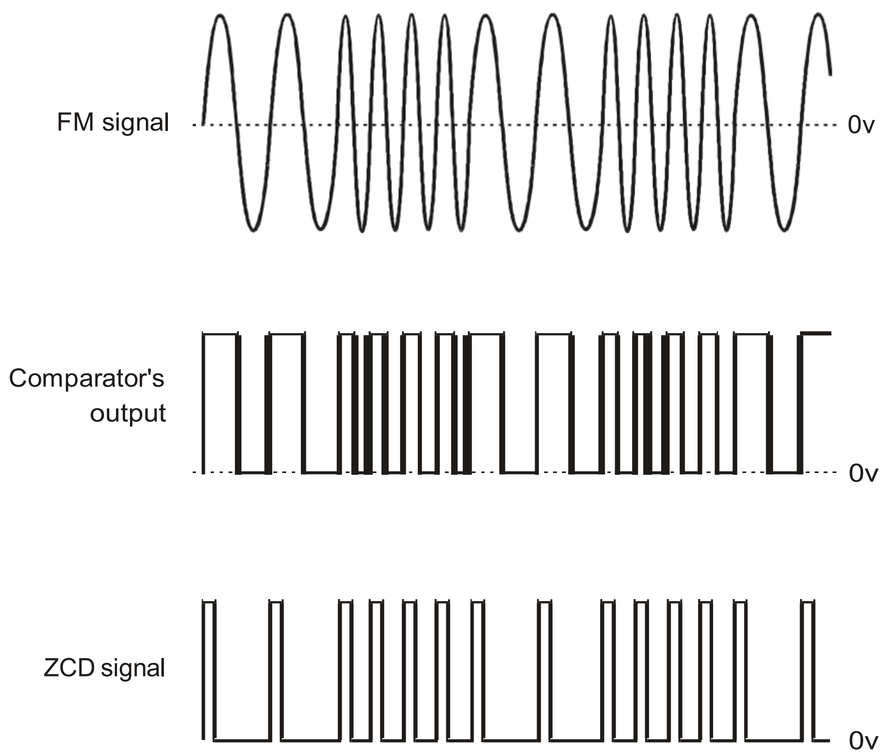


Figure 2

Recall from the theory of complex waveforms, pulse trains are actually made up of sinewaves and, in the case of Figure 2 above, a DC voltage. The size of the DC voltage is affected by the pulse train's duty cycle. The greater its duty cycle, the greater the DC voltage.

That being the case, when the FM signal in Figure 2 above switches between the two frequencies, the DC voltage that makes up the rectangular wave out of the ZCD changes between two values. In others words, the DC component of the rectangular wave is a copy of the squarewave that produced the FM signal in the first place. Recovering this copy is a relatively simple matter of picking out the changing DC voltage using a low-pass filter.

Importantly, this demodulation technique works equally well when the message is a sinewave or speech.

The experiment

For this experiment you'll use the Emona DATEx to generate an FM signal using a vCO. Then you'll set-up a zero-crossing detector and verify its operation for variations in the message's amplitude.

It should take you about 50 minutes to complete this experiment.

Equipment

- Personal computer with appropriate software installed
- NI ELvIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads
- One set of headphones (stereo)

Procedure

Part A - Setting up the FM modulator

To experiment with FM demodulation you need an FM signal. The first part of the experiment gets you to set one up. To make viewing the signals around the demodulator possible, we'll start with a DC voltage for the message.

1. Ensure that the NI ELvIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELvIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELvIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELvIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELvISmx software.

8. Launch and run the NI ELvIS II Function Generator vI.

9. Adjust the function generator for an output with the following specifications:

   - Waveshape: Sine
   - Frequency: 15kHz
   - Amplitude: 4vpp
   - DC Offset: 0v
   - Modulation type: FM

10. Check that the function generator's Signal Route option (near the bottom right-hand corner of the vI is set to Prototyping Board.

    Important note: There is the potential for a hardware conflict when using the scope's external triggering input (TRIG) at the same time as the function generator. This is because the connector for the TRIG input doubles as an output for the function generator when its Signal Route option is set to FGEN BNC. To avoid hardware conflicts always set the Signal Route to Prototyping Board.

11. Launch and run the NI ELvIS II variable Power Supplies vI.

12. Turn the variable Power Supplies negative output soft voltage control fully anti-clockwise to output 0v.

13. Connect the set-up shown in Figure 3 below.

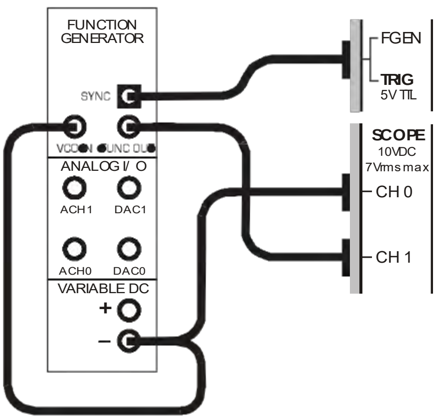


Figure 3

This set-up can be represented by the block diagram in Figure 4 on the next page. The negative output of the variable DC Power Supplies is being used to provide a simple DC message and the vCO implements the FM modulator with a carrier frequency of 15kHz.

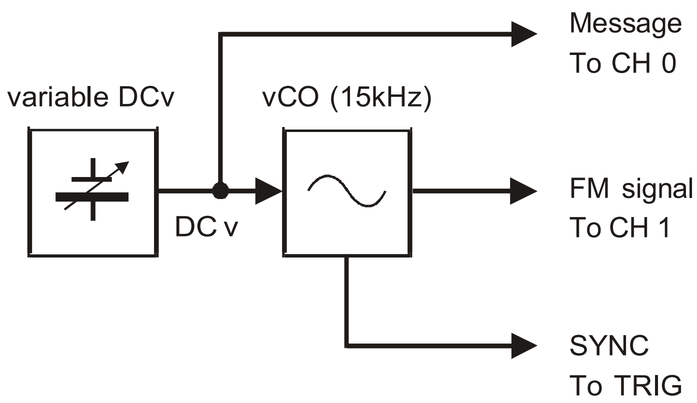


Figure 4

14. Launch and run the NI ELvIS II Oscilloscope vI.

15. Set up the scope per the procedure in Experiment 1 with the following changes:

- Scale control for Channel 0 to 2v/div instead of 1v/div
- Trigger Type control to Digital instead of Edge
- Coupling controls for both channels to DC instead of AC

16. Activate the scope's Channel 1 input to view the FM signal on the vCO's output as well as the DC message signal.

17. Set the scope's Timebase control to view two or so cycles of the vCO output.

18. vary the variable Power Supplies negative output soft voltage control and check that the vCO's output frequency changes accordingly.

Ask the instructor to check your work before continuing.

Part B - Setting up the zero-crossing detector

19. Launch the DATEx soft front-panel (SFP) and check that you have soft control over the DATEx board.

20. Locate the Twin Pulse Generator module on the DATEx SFP and turn its soft Width control fully anti-clockwise.

21. Set the Twin Pulse Generator module's soft Delay control fully anti-clockwise.

22. Locate the Tuneable Low-pass Filter module on the DATEx SFP and turn its soft Gain control fully clockwise.

23. Turn the Tuneable Low-pass Filter module's soft Cut-off Frequency Adjust control fully clockwise.

24. Modify the set-up as shown in Figure 5 below.

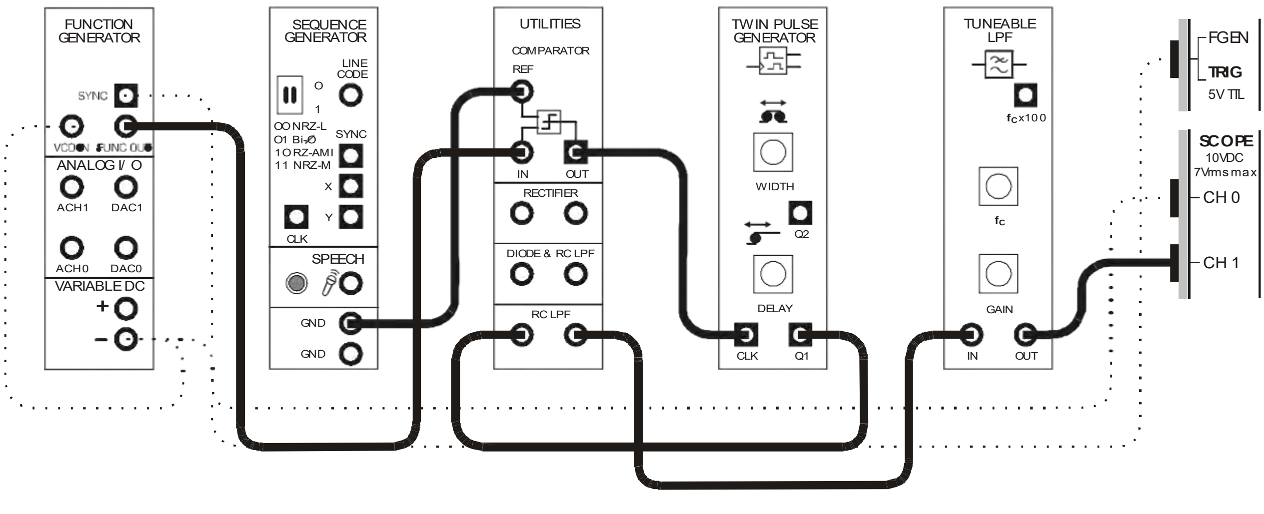


Figure 5

The additions to the set-up can be represented by the block diagram in Figure 6 on the next page. The comparator on the Utilities module is used to clip the FM signal, effectively turning it into a squarewave. The positive edge-triggered Twin Pulse Generator module is used to implement the zero-crossing detector. To complete the FM demodulator, the RC Low-pass Filter module and Tuneable Low-pass Filter module combination is used to pick-out the changing DC component of the Twin Pulse Generator module's output.

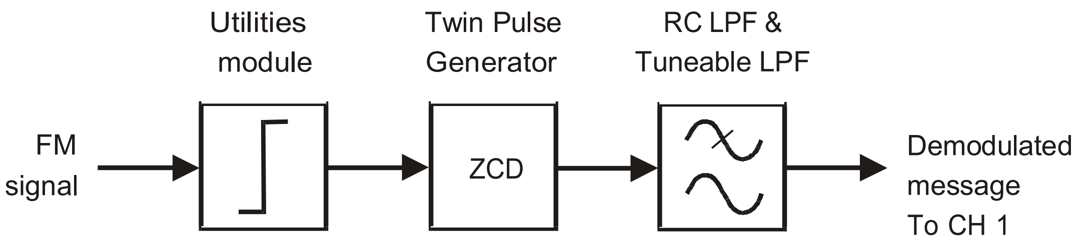


Figure 6

Note: The RC LPF and Tuneable LPF modules are used in combination because the Tuneable LPF is a clocked "switched capacitor" filter which uses internal sampling technology. These types of filters are subject to aliasing (a concept that is covered in later experiments) which can cause problems in sampled systems. To avoid such problems, we use the RC LPF as an anti-aliasing pre-filter for the Tuneable LPF. If time permits, explore this part of the experiment without the RC LPF.

The entire set-up can be represented by the block diagram in Figure 7 below.

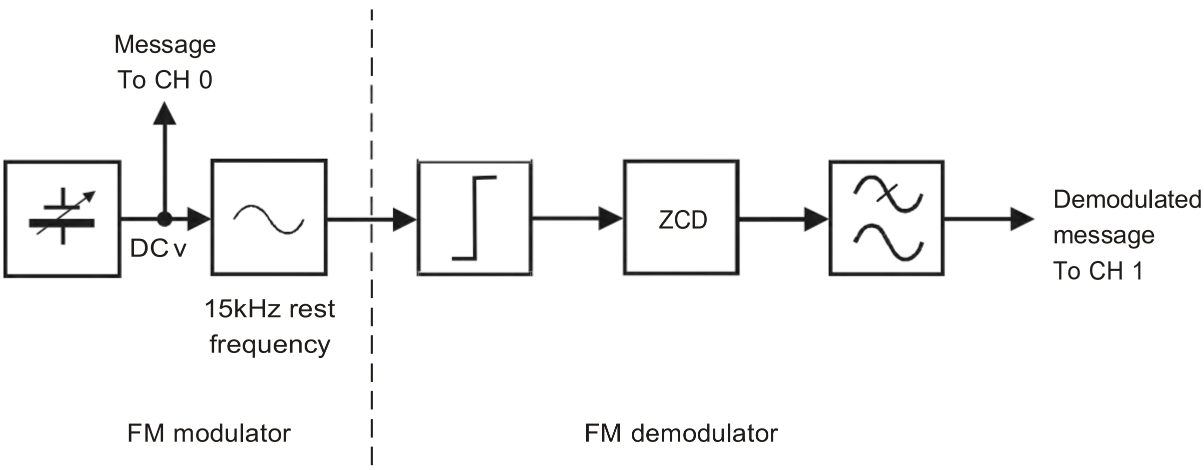


Figure 7

You will now tune the Tuneable Low-pass Filter to isolate the DC component of the ZCD for the voltage range of the input signal we plan to use.

25. Set the variable Power Supplies negative output to -2v.

26. Set the scope's Channel 1 Scale control to the 100mv/div position.

    Note: You should now see a 430mvp-p sinewave with a DC offset of approximately 230mv. This is the filtered pulse train out of the ZCD.

27. Slowly turn the Tuneable Low-pass Filter module's soft Cut-off Frequency Adjust control anti-clockwise until the sine wave becomes a DC voltage. Stop the moment this happens.

    Note 1: You have now eliminated all non-DC components from the signal. That said, the filter will still allow the message signal to pass.

    Note 2: Don't change the Tuneable Low-pass Filter module's soft Cut-off Frequency Adjust control setting for the rest of the experiment unless instructed to.

28. vary the variable Power Supplies negative output between 0v and a maximum of -2v.

    Note 1: As you do, you should notice that the DC voltage out of the Tuneable Low-pass Filter module varies as well.

    Note 2: If this doesn't happen, check that the scope's Channel 1 Coupling control is set to the DC position.

Ask the instructor to check
your work before continuing.

Part C - Investigating the operation of the zero-crossing detector

The next part of the experiment lets you verify the operation of the zero-crossing detector.

29. Rearrange the scope's connections to the set-up as shown in Figure 8 below.

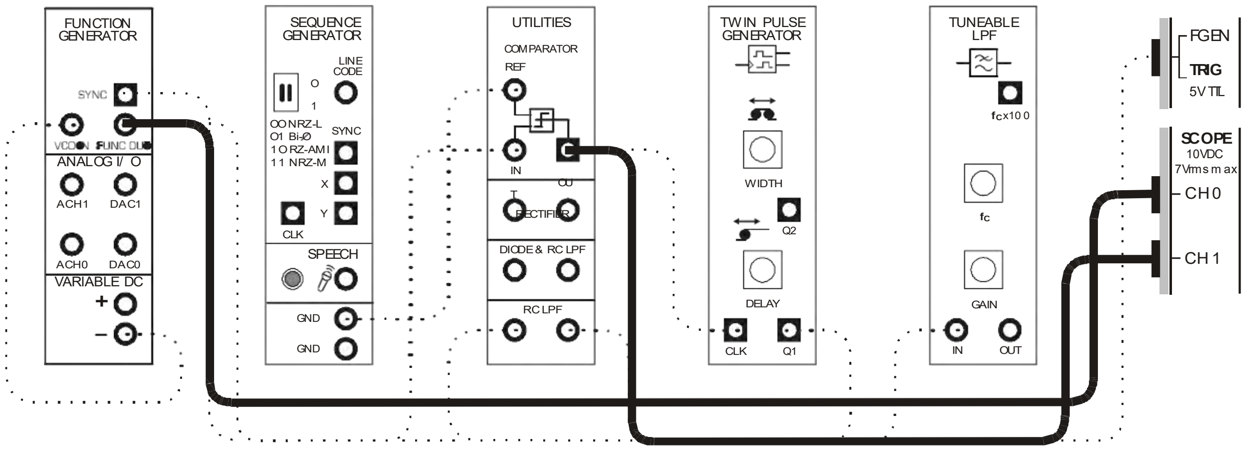


Figure 8

The new scope connections can be shown using the block diagram in Figure 9 below.

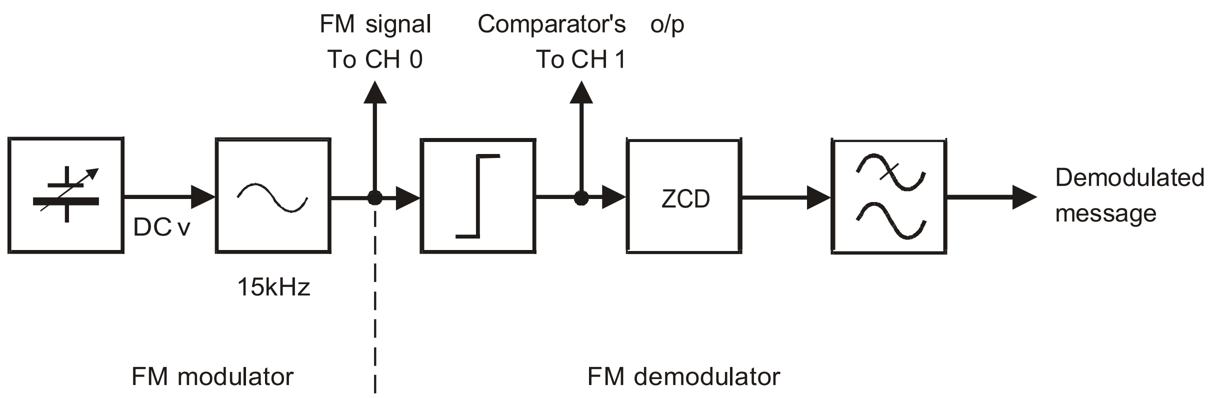


Figure 9

30. vary the variable Power Supplies negative output in small steps using the up and down arrow buttons on the vI.

Note: This will cause small but noticeable changes in the FM signal's frequency.

31. As you vary the FM signal's frequency, pay close attention to the mark-space ratio (that is, the duty cycle) of the Comparator's output.

Tip: You may find it helpful to adjust the scope's vertical Position controls to separate the signals on the display.

Question 1
Does the mark-space ratio of the signal on the Comparator's output change?

Question 2
What does this tell us about the DC component of the comparator's output?

Ask the instructor to check
your work before continuing.

32. Rearrange the scope's connections to the set-up as shown in Figure 10 below.

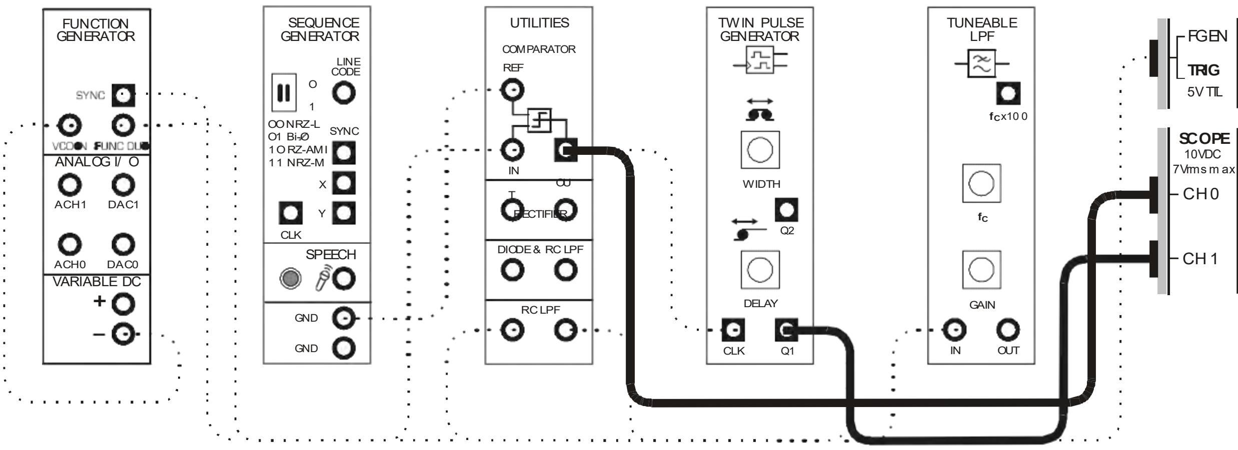


Figure 10

The new scope connections can be shown using the block diagram in Figure 11 below.

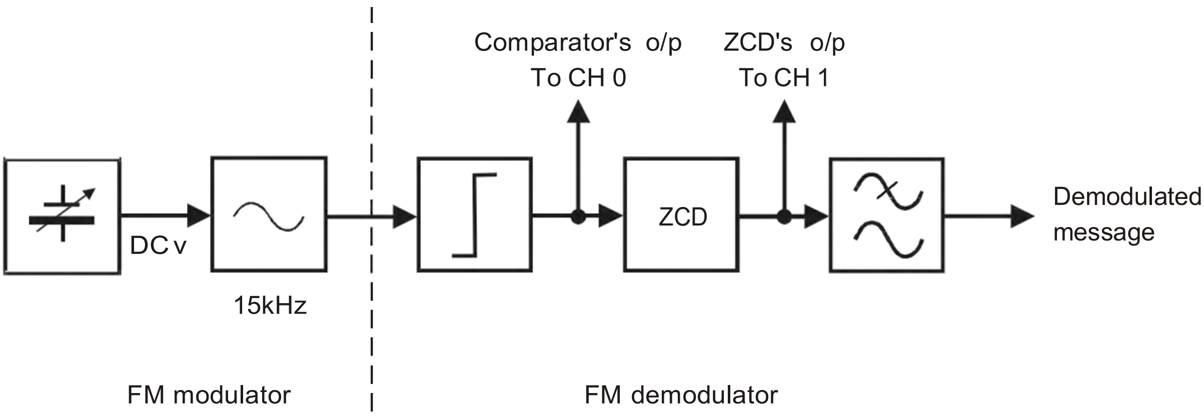


Figure 11

33. vary the variable Power Supplies negative output in small steps again to model an FM signal's changing frequency.

34. As you perform the step above, note how the frequency of the two signals changes.

35. Turn on the scope's cursors.

36. Use the scope's cursors to measure the width of the ZCD output's mark and space for different DC input voltages.

    Note: The time difference between the two cursors is displayed directly above the Channel 0 & 1 measurements and is denoted as dT.

    Tip: You may find it helpful to turn the scope's Channel 0 off as you do this and set its Timebase control to 10J,s/div when measuring the mark's width.

Question 3
As the FM signal changes frequency so does the ZCD's output. What aspect of the ZCD's output signal changes to achieve this?

D Neither the signal's mark nor space

D Only the signal's mark

D Only the signal's space

D Both the signal's mark and space

Question 4
What does this tell us about the DC component of the comparator's output?

____

____

Ask the instructor to check
your work before continuing.

The next part of the experiment lets you verify your answer to the previous question.

37. If you deactivated the scope's Channel 0 then reactivate it and return its Timebase control to 50 μ,s/div.

38. Rearrange the scope's connections to the set-up as shown in Figure 12 below.

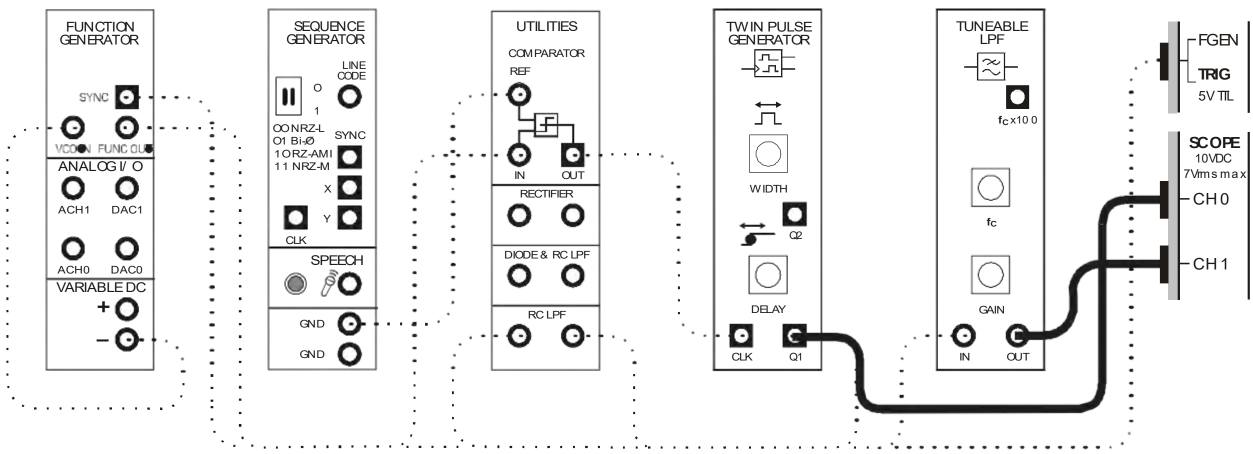


Figure 12

The new scope connections can be shown using the block diagram in Figure 13 below.

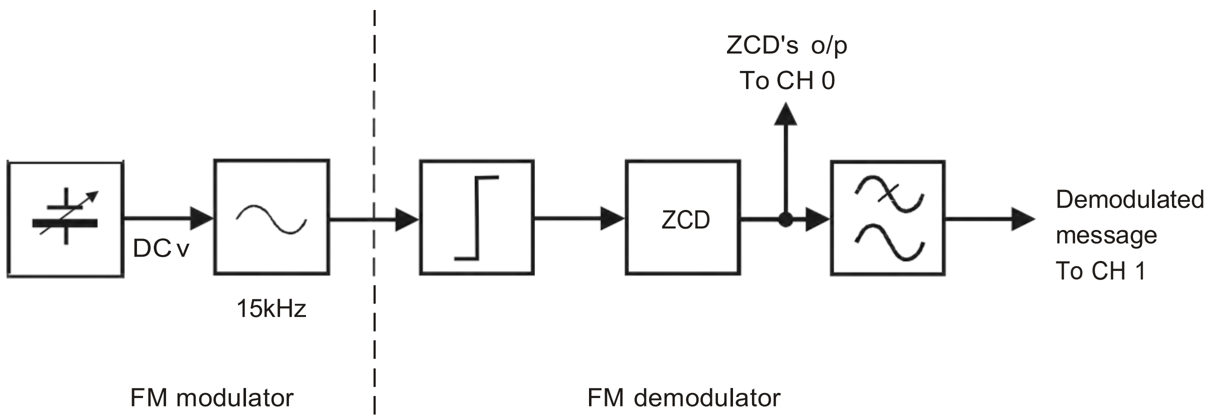


Figure 13

39. If you've adjusted the scope's Channel 1 vertical Position control, re-zero it.

40. vary the variable Power Supplies negative output in small steps again to model an FM signal's changing frequency.

41. As you perform the step above, compare the outputs from the Twin Pulse Generator module (the ZCD) and the Tuneable Low-pass Filter module.

Question 5
Why does the Tuneable Low-pass Filter module's DC output go up as the mark-space ratio of the ZCD's output goes up?

Question 6
If the original message is a sinewave instead of a variable DC voltage, what would you expect to see out of the Tuneable Low-pass Filter module?

Ask the instructor to check your work before continuing.

Part D - Transmitting and recovering a sinewave using FM

This experiment has set up an FM communication system to "transmit" a message that is a DC voltage. The next part of the experiment lets you use the set-up to modulate, transmit and demodulate a test signal (a sinewave).

42. If it's not already, turn the Tuneable Low-pass Filter module's soft Gain control fully clockwise.

43. Modify the set-up as shown in Figure 14 below.

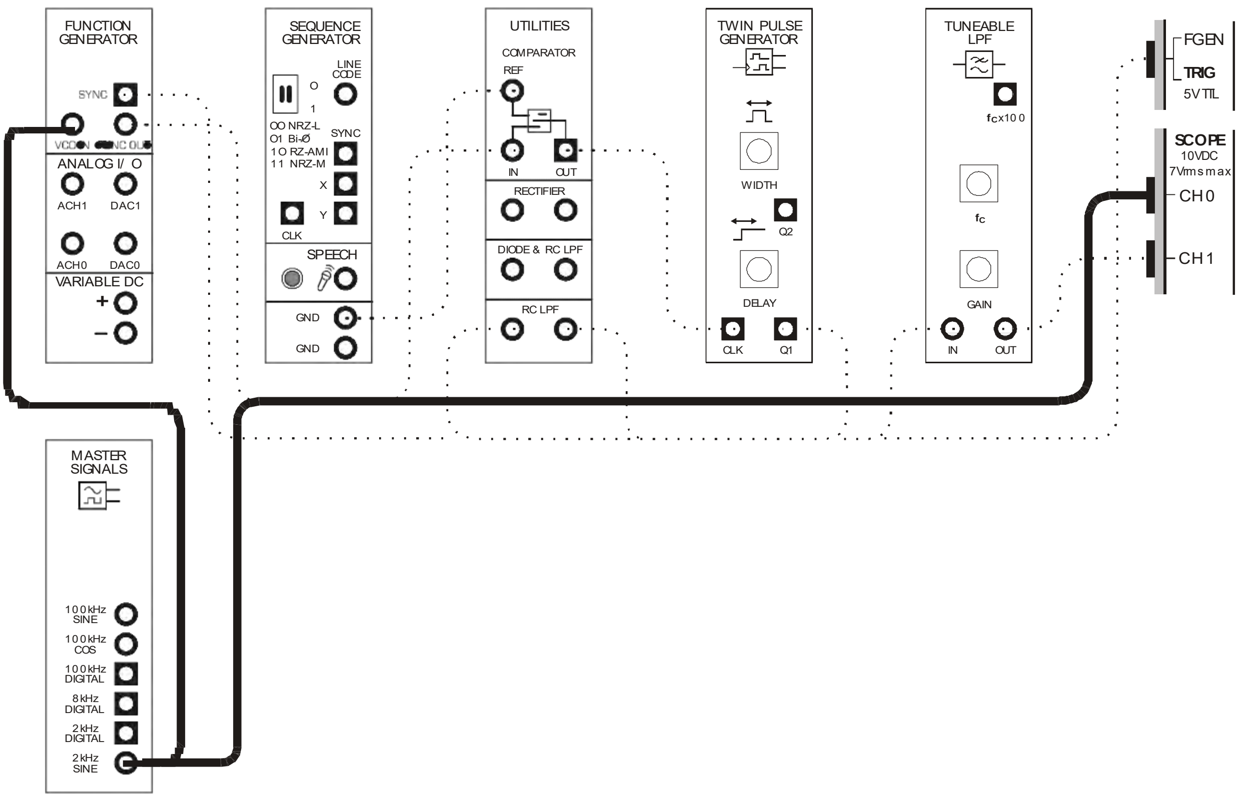


Figure 14

This modification to the FM modulator can be shown using the block diagram in Figure 15 on the next page. Notice that the message is now provided by the Master Signals module's 2kHz SINE output.

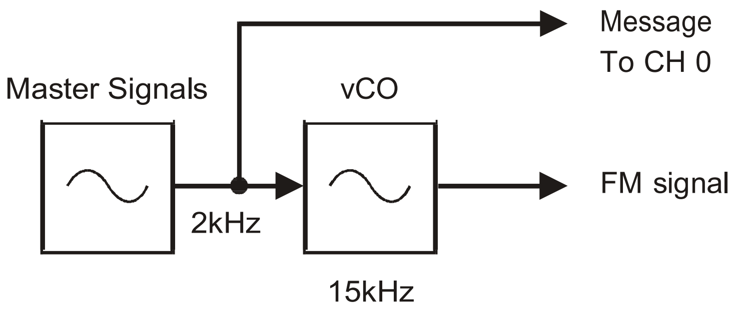


Figure 15

44. Make the following adjustments to the scope's controls:

- Scale control for Channel 0 to 2v/div and to 100mv/div for Channel 1
- Input Coupling control for both channels to AC
- Trigger Type to Edge
- Trigger Source to CH 0
- Timebase control to 200J,s/div

45. Use the TAB and arrow keys to turn the Tuneable Low-pass Filter module's soft Cut-off Frequency Adjust control slightly anti-clockwise to fine tune the filter's cut-off frequency.

Note: You will already be viewing the demodulated 2kHz message sinewave with an amplitude of approximately 250mvp-p.

Question 7
What does the FM modulator's output signal tell you about the ZCD signal's duty cycle?

___

___

Ask the instructor to check your work before continuing.

Part E - Transmitting and recovering speech using FM

The next part of the experiment lets you use the set-up to modulate, transmit and demodulate speech.

46. Disconnect the plugs to the Master Signals module's 2kHz SINE output.

47. Modify the set-up as shown in Figure 16 below.

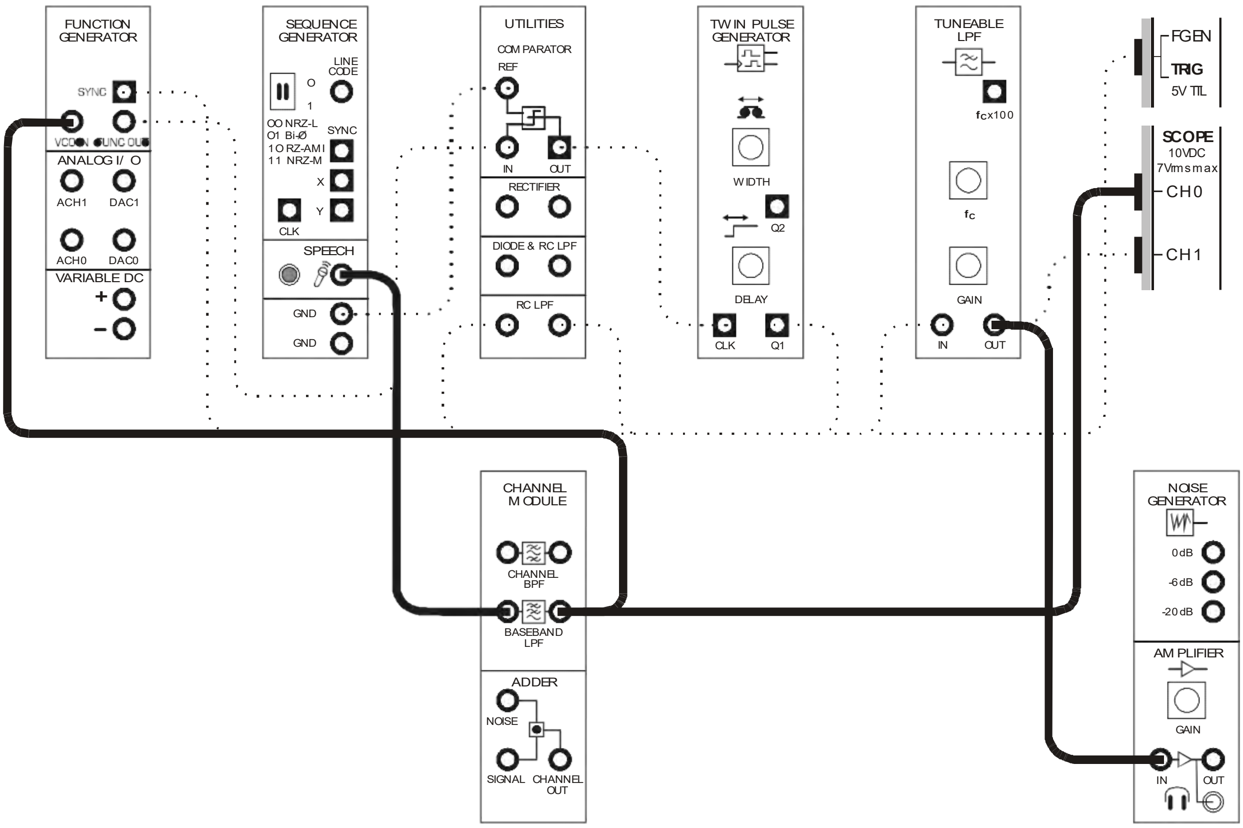


Figure 16

48. Set the scope's Timebase control to the 2ms/div position.

49. Locate the Amplifier module on the DATEx SFP and turn its soft Gain control fully anti-clockwise.

50. Without wearing the headphones, plug them into the Amplifier module's headphone socket.

51. Put the headphones on.

52. As you perform the next step, set the Amplifier module's soft Gain control to a comfortable sound level.

53. Hum and talk into the microphone while watching the scope's display and listening on the headphones.

60. Once you have completed viewing the signal with the scope, stop the scope and start the NI ELvIS II Dynamic Signal Analyzer to view the spectrum of the frequency modulated speech. Set the DSA frequency setting to 40kHz. Try whistling into the microphone. This will help you to see the difference between single tones and speech during modulation.

Ask the instructor to check your work before finishing.

Week 11: EXPERIMENT 11

Design of A PULSE AMPLITUDE MODULATION (PAM)

OBJECTIVES:

The purpose of this experiment is to design and conduct pulse amplitude modulation experiment to achieve the following:

1. The PAM signals in the time domain.
2. The spectrum of PAM signals.
3. The influence of the duty cycle on PAM signals.

Test Standard :IEEE 802

INTRODUCTION:

In pulse amplitude modulation, the amplitude of individual pulses in the pulse train is varied from its default value in accordance with the instantaneous amplitude of the modulating signal at sampling intervals. The width and position of the pulses is kept constant. As shown in Figure 1.

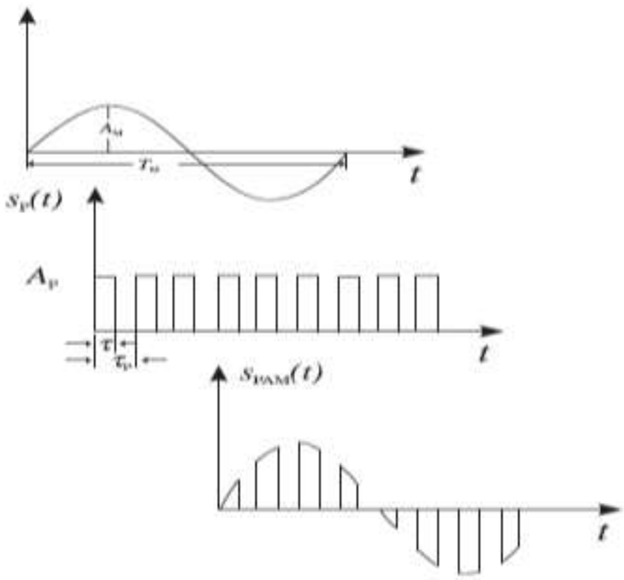


Figure 1: Pulse Amplitude Modulation.

Prior to sampling, a low-pass pre-alias filter is used to attenuate those higher-frequency components of the message signal that are not essential to the information being conveyed by the signal.

The message signal is sampled with a train of narrow rectangular pulses (see Figure 1). To closely approximate the instantaneous sampling process. The sampling rate must be greater than twice the highest frequency component f_M of the message signal in accordance with the sampling theorem (Nyquist rate).

EQUIPMENTS DESCRIPTIONS:

The two main parts in this experiment are the PAM Modulator and PAM Demodulator. Figure 2 shows the PAM modulator (to the right) and PAM demodulator (to the left).

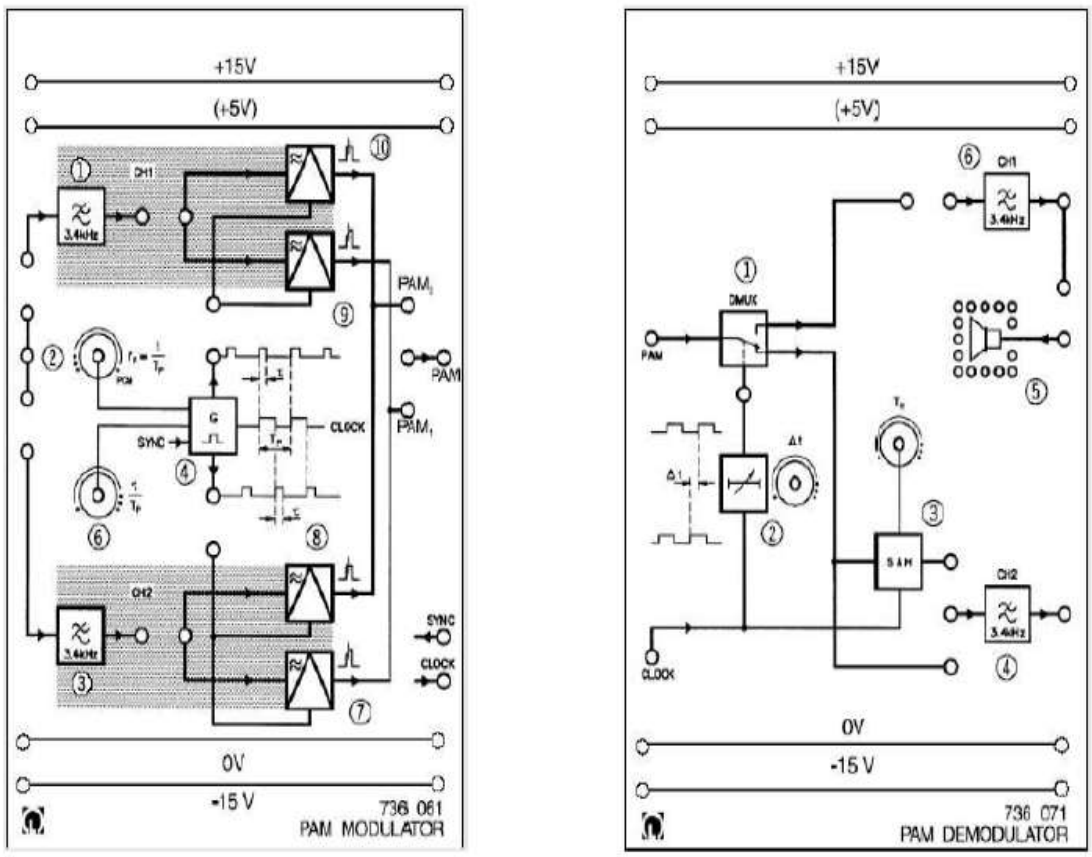


Figure 2: PAM modulator, and PAM demodulator.

PAM Modulator

1- Input filter channel 1 (low-pass filter with 3.4 kHz limit).
2- Socket field for the connection of the function generator.
3- Input filter channel 2 (low-pass filter with 3.4 kHz limit).
4- Clock generator $1\ KHz < f_p < 15\ KHz$.
5- Control knop for f_p.
6- Control for duty cycle.
7- Modulator for PAM 1 channel 2.
8- Modulator for PAM 2 channel 2.
9- Modulator for PAM 1 channel 1.

10- Modulator for PAM 2 channel 1.

PAM Demodulator

1- Demultiplexer.
2- Variable skew Δt (simulation of channel cross-talk).
3- Sample and hold circuit with adjustable hold time T_H.
4- Demodulator low-pass channel 2.
5- Loudspeaker with integrated push-pull stag.
6- Demodulator low-pass channel 1.

PROCEDURE:

PART A - Adjusting the sampling frequency

Connect the experiment as shown in figure 3.

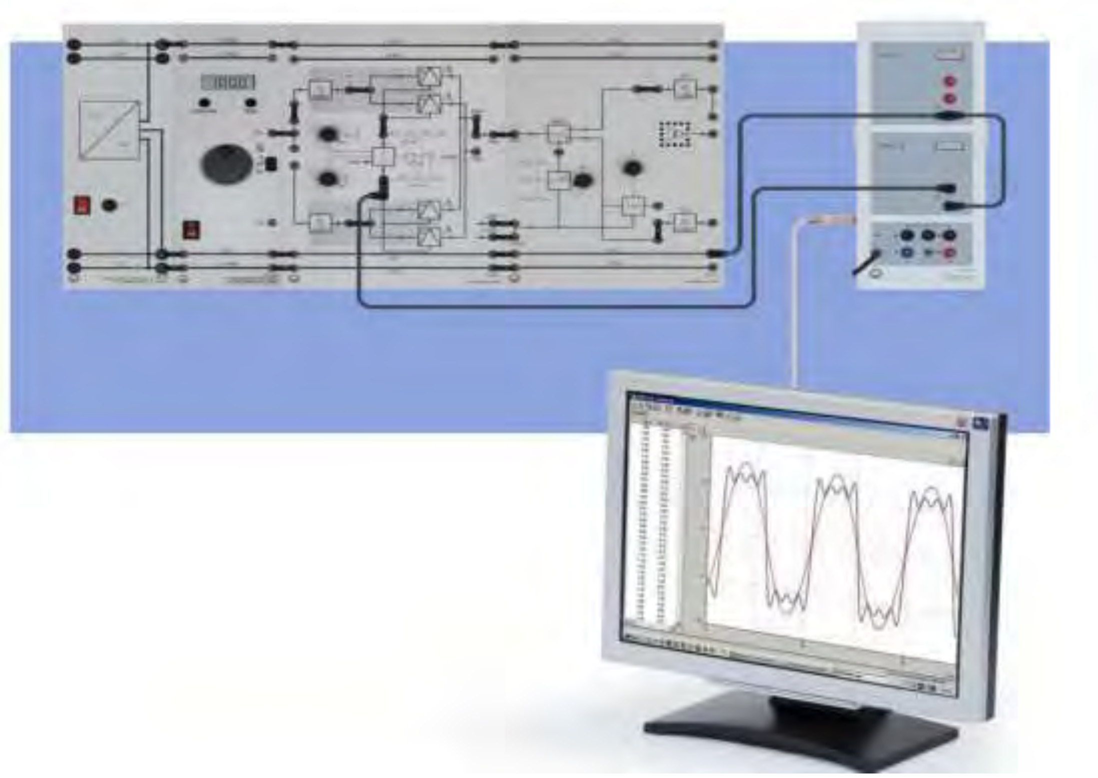


Figure 3: Experiment setup.

- The sampling frequency f_p is set using the FFT analyzer. For that purpose set the PAM modulator:
  Control for duty cycle $\tau/T_P \rightarrow$ PCM.
  Control for sampling frequency $f_p \rightarrow \cdots$ (max)
  CASSY UB1 $\rightarrow$ Clock generator G.

- Set the measurement parameters so that you can see a clear time and frequency response of the input signal (Clock generator).
- Start the measurement.
- Now slowly adjust the pulse frequency f_p, until the spectral line of the fundamental mode appears at f_0 = 5000 Hz (3 f_0 = 15 kHz, etc). Don't change the sampling frequency f_p anymore.

PART B- Time characteristic of the PAM

- Measure the input and output of channel filter CH1.
    - Function generator: sine, 500 Hz, A= 10 V_{pp}.
    - CASSY UA1 → input of channel filter CH1.
    - CASSY UB1 → output of channel filter CH1.
    - Set the measurement parameters so that you can see a clear time and frequency response of the signal.
    - Start the measurement.
- Display the time characteristic of the PAM
    - Function generator: sine, 500 Hz, A= 10 V_{pp}.
    - CASSY UA1 → input PAM modulator channel CH1.
    - CASSY UB1 → output PAM_1.
    - Set the measurement parameters so that you can see a clear time and frequency response of the signal.
    - Start the measurement.
- Measure the modulating signal $S_M(t)$ and the demodulated signal $S_D(t)$ as a function of duty cycle.
    - Adjust the controller for the sampling frequency f_p → ⋯ (max)
    - Adjusting the duty cycle:
      CASSY UB1 → clock generator *G*.
      Load the CASSY lab example DutyCycle.labx.
      Start the measurement.
    - Slowly readjust the duty cycle τ/T_P, until the display of the CASSY instrument shows τ/T_P = 50%. Eventually correct the display, for that make a right click into the instrument Duty Cycle and match the factor 1.1 to your special situation. For the maximum position (PCM) is true τ/T_P = 50%.
    - CASSY UA1 → input of channel filter CH1 at PAM modulator.
    - CASSY UB1 → output of channel filter CH1 at PAM demodulator.
    - Set the measurement parameters so that you can see a clear time and frequency response of the signal.
    - Start the measurement.
    - Repeat the measurement for τ/T_P = 30% and τ/T_P = 10%.

- Sketch your results.

PART C- Spectrum of PAM

- The PAM1 spectrum as a function of the frequency of the modulating signal.
  - All the measurements are made for f_p = 5 kHz. Follow the hints Adjusting the sampling frequency.
  - Function generator: sine, 500 Hz, A= 10 V_{pp}.
  - CASSY UA1 → output PAM1 at the PAM modulator.
  - CASSY UB1 → output of the clock generator.
  - Set the measurement parameters so that you can see a clear time and frequency response of the signal.
  - Start the measurement.
  - Repeat the measurement for f_M = 1 kHz and f_M = 2 kHz.
  - Sketch your results.
- The PAM1 spectrum as a function of the duty cycle.
  - Function generator: sine, 1000 Hz, A= 10 V_{pp}.
  - CASSY UA1 → output PAM1 at the PAM modulator.
  - CASSY UB1 → clock generator.
  - Set the duty cycle for τ/T_P = 30%.
  - Set the measurement parameters so that you can see a clear time and frequency response of the signal.
  - Start the measurement.
  - Sketch your results. Mark in the spectrum the position of the suppressed carrier lines. Compare the PAM spectra with the pulse spectra.

Week 12: Experiment I2 - Sampling and reconstruction

Test Standard :IEEE 802

Preliminary discussion

So far, the experiments in this manual have concentrated on communications systems that transmit analog signals. However, digital transmission is fast replacing analog in commercial communications applications. There are several reasons for this including the ability of digital signals and systems to resist interference caused by electrical noise.

Many digital transmission systems have been devised and several are considered in later experiments. Whichever one is used, where the information to be transmitted (called the message) is an analog signal (like speech and music), it must be converted to digital first. This involves sampling which requires that the analog signal's voltage be measured at regular intervals.

Figure Ia below shows a pure sinewave for the message. Beneath the message is the digital sampling signal used to tell the sampling circuit when to measure the message. Beneath that is the result of "naturally" sampling the message at the rate set by the sampling signal. This type of sampling is "natural" because, during the time that the analog signal is measured, any change in its voltage is measured too. For some digital systems, a changing sample is unacceptable. Figure Ib shows an alternative system where the sample's size is fixed at the instant that the signal measured. This is known as a sample-and-hold scheme (and is also referred to as pulse amplitude modulation).

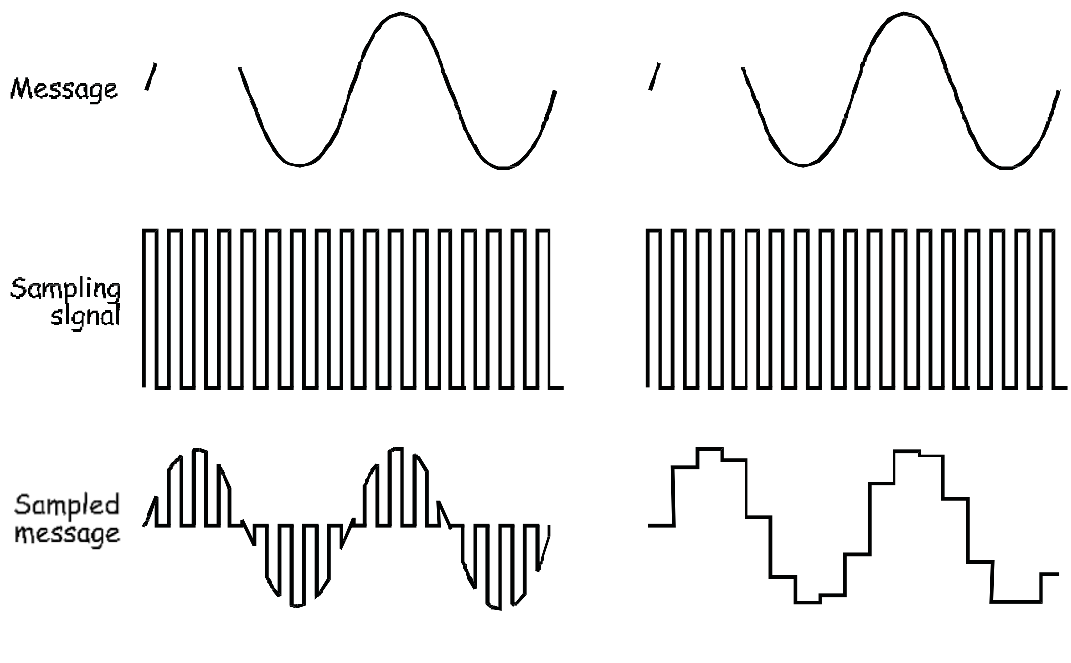


Figure Ia Figure Ib

Regardless of the sampling method used, by definition it captures only pieces of the message. So, how can the sampled signal be used to recover the whole message? This question can be answered by considering the mathematical model that defines the sampled signal:

Sampled message = the sampling signal x the message

As you can see, sampling is actually the multiplication of the message with the sampling signal. And, as the sampling signal is a digital signal which is actually made up of a DC voltage and many sinewaves (the fundamental and its harmonics) the equation can be rewritten as:

Sampled message = (DC + fundamental + harmonics) x message

When the message is a simple sinewave (like in Figure I) the equation's solution (which necessarily involves some trigonometry that is not shown here) tells us that the sampled signal consists of:

- A sinewave at the same frequency as the message
- A pair of sinewaves that are the sum and difference of the fundamental and message frequencies
- Many other pairs of sinewaves that are the sum and difference of the sampling signals' harmonics and the message

This ends up being a lot of sinewaves but one of them has the same frequency as the message. So, to recover the message, all that need be done is to pass the sampled signal through a low-pass filter. As its name implies, this type of filter lets lower frequency signals through but rejects higher frequency signals.

That said, for this to work correctly, there's a small catch which is discussed in Part E of the experiment.

The experiment

For this experiment you'll use the Emona DATEx to sample a message using natural sampling then a sample-and-hold scheme. You'll then examine the sampled message in the frequency domain using the NI ELVIS II Dynamic Signal Analyzer. Finally, you'll reconstruct the message from the sampled signal and examine the effect of a problem called aliasing.

It should take you about 50 minutes to complete this experiment.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads

Part A - Sampling a simple message

The Emona DATEx has a Dual Analog Switch module that has been designed for sampling. This part of the experiment lets you use the module to sample a simple message using two techniques.

Procedure

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELVISmx software.

Ask the instructor to check your work before continuing.

8. Connect the set-up shown in Figure 2 below.

Note: Insert the black plugs of the oscilloscope leads into a ground (GND) socket.

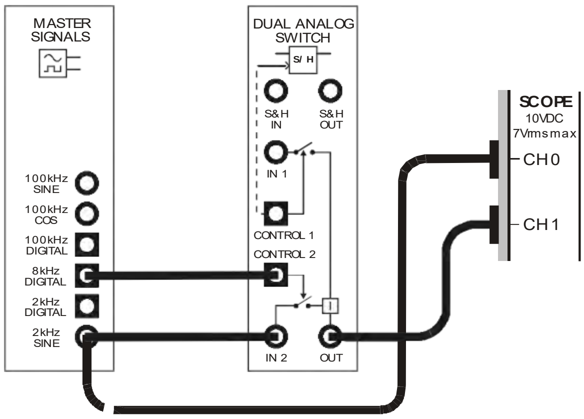


Figure 2

This set-up can be represented by the block diagram in Figure 3 below. It uses an electronically controlled switch to connect the message signal (the 2kHz SINE output from the Master Signals module) to the output. The switch is opened and closed by the 8kHz DIGITAL output of the Master Signals module.

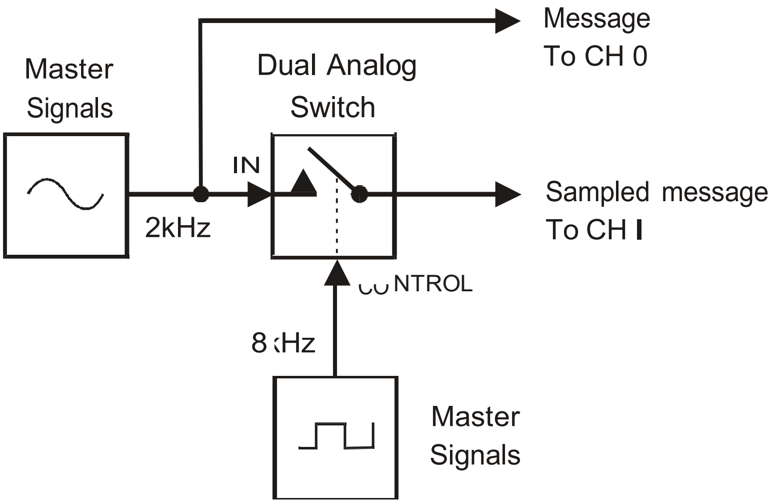


Figure 3

9. Launch and run the NI ELVIS II Oscilloscope VI.

I0. Set up the scope per the procedure in Experiment I (page I-I2) with the following change:

- Timebase control to the I00.us/div position instead of 500.us/div

II. Adjust the scope's Timebase control to view two or so cycles of the Master Signals module's 2kHz SINE output.

I2. Activate the scope's Channel I input by (by checking the Channel I Enabled box) to observe the sampled message out of the Dual Analog Switch module as well as the message.

Tip: To see the two waveforms clearly, you may need to adjust the scope so that the two signals are not overlayed.

I3. Draw the two waveforms to scale in the space provided on the next page leaving room to draw a third waveform.

Tip: Draw the message signal in the upper third of the graph and the sampled signal in the middle third.

Question I
What type of sampling is this an example of?

D Natural
D Sample-and-hold

Question 2
What two features of the sampled signal confirm this?

Ask the instructor to check
your work before continuing.

I4. Modify the set-up as shown in Figure 4 below.

Before you do.
The set-up in Figure 4 below builds on the set-up that you've already wired so don't pull it apart. To highlight the changes that we want you to make, we've shown your existing wiring as dotted lines.

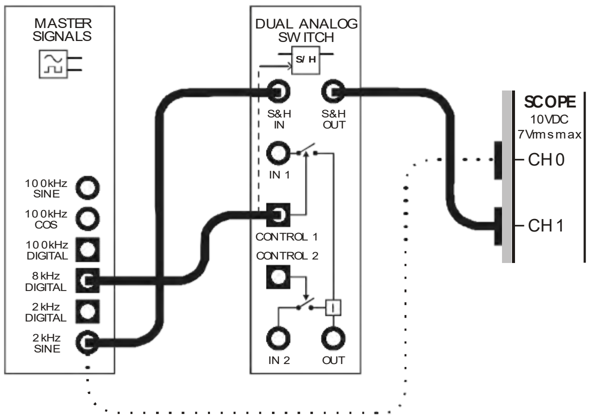


Figure 4

This set-up can be represented by the block diagram in Figure 5 on the next page. The electronically controlled switch in the original set-up has been substituted for a sample-and-hold circuit. However, the message and sampling signals remain the same (that is, a 2kHz sinewave and an 8kHz pulse train).

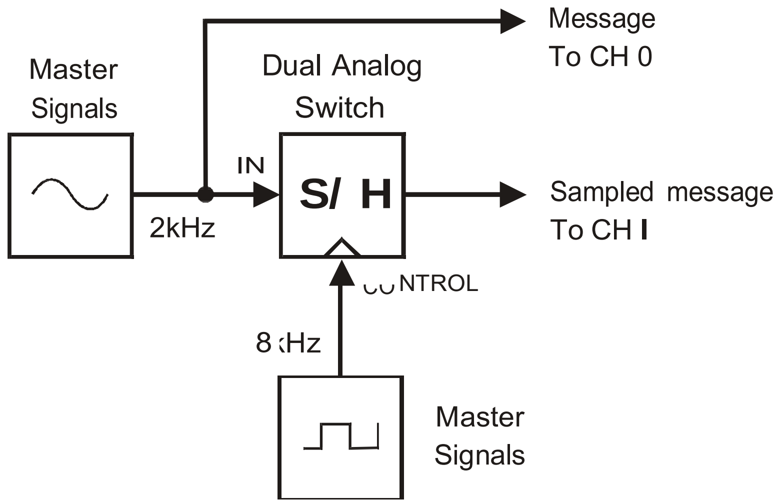


Figure 5

I5. Draw the new sampled message to scale in the space that you left on the graph paper.

Question 3
What two features of the sampled signal confirm that the set-up models the sample- and-hold scheme?

____

____

Ask the instructor to check
your work before continuing.

Part B - Sampling speech

This experiment has sampled a 2kHz sinewave. However, the message in commercial digital communications systems is much more likely to be speech and music. The next part of the experiment lets you see what a sampled speech signal looks like.

I6. Disconnect the plugs to the Master Signals module's 2kHz SINE output.

I7. Connect them to the Speech module's output as shown in Figure 6 below.

Remember: Dotted lines show leads already in place.

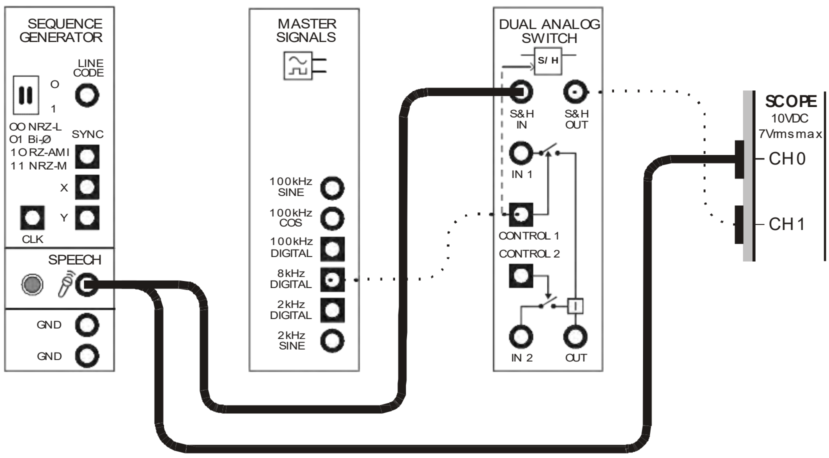


Figure 6

I8. Set the scope's Timebase control to the 500.us/div position.

I9. Hum and talk into the microphone while watching the scope's display.

Ask the instructor to check your work before continuing.

Part C - Observations and measurements of the sampled message in the frequency domain

Recall that the sampled message is made up of many sinewaves. Importantly, for every sinewave in the original message, there's a sinewave in the sampled message at the same frequency. This can be proven using the NI ELVIS II Dynamic Signal Analyzer. This device performs a mathematical analysis called Fast Fourier Transform (FFT) that allows the individual sinewaves that make up a complex waveform to be shown separately on a frequency-domain graph. The next part of the experiment lets you observe the sampled message in the frequency domain.

20. Return the scope's Timebase control to the 100.us/div position.

21. Disconnect the plugs to the Speech module's output and reconnect them to the Master Signals module's 2kHz SINE output.

    Note: The scope should now display the waveform that you drew for Step 15.

22. Suspend the scope VI's operation by clicking on its Stop control once.

    Note: The scope's display should freeze and its hardware has been deactivated. This is a necessary step as the scope and signal analyzer share hardware resources and so they cannot be operated simultaneously.

23. Launch the NI ELVIS II Dynamic Signal Analyzer VI.

    Note: If the signal analyzer's VI has launched successfully, the instrument's window will be visible (see Figure 7).

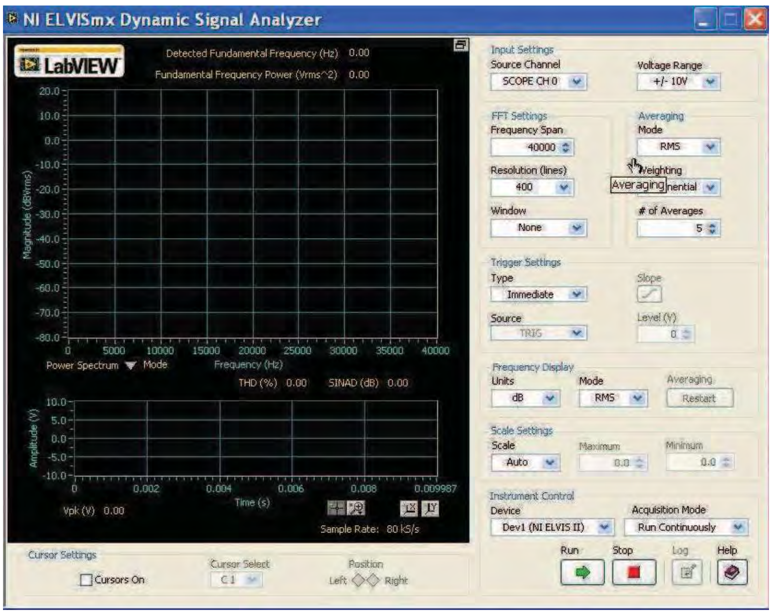


Figure 7

24. Adjust the Signal Analyzer's controls as follows:

Input Settings

- Source Channel to SCOPE CH I
- Voltage Range to ±l0V

FFT Settings

- Frequency Span to 40,000
- Resolution to 400
- Window to 7 Term B-Harris

Averaging

- Mode to RMS
- Weighting to Exponential
- # of Averages to 3

Trigger Settings

- Type to Edge

Frequency Display

- Units to dB (for now)
- Mode to RMS
- Scale to Auto
- Cursors On box unchecked (for now)

25. Click on the signal analyzer's Run control.

Note: If the Signal Analyzer VI has been set up correctly, its display should look like Figure 8 below.

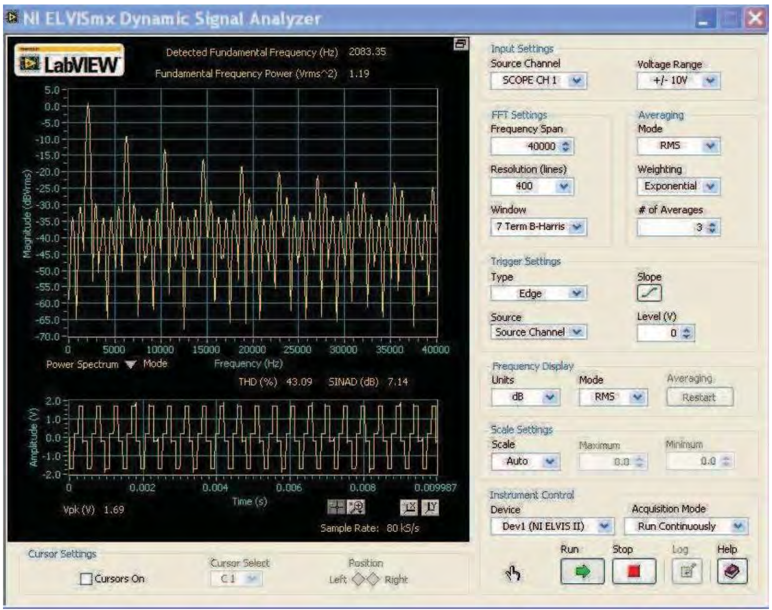


Figure 8

If you've not attempted Experiment 7, the signal analyzer's display may need a little explaining here. There are actually two displays, a large one on top and a much smaller one underneath. The smaller one is a time domain representation of the input (in other words, the display is a scope).

The larger of the two displays is the frequency domain representation of the complex waveform on its input (the sampled message). The humps represent the sinewaves and, as you can see, the sampled message consists of many of them. As an aside, these humps should just be simple straight lines, however, the practical implementation of FFT is not as precise as the theoretical expectation.

If you have done Experiment 7, go directly to Step 33 on the next page.

26. Activate the signal analyzer's cursors by checking (that is, ticking) Cursors On box.

    Note: When you do, green horizontal and vertical lines should appear on the signal analyzer's frequency domain display.

The NI ELVIS II Dynamic Signal Analyzer has two cursors Cl and C2 that default to the left most side of the display when the signal analyzer's VI is launched. They're repositioned by "grabbing" their vertical lines with the mouse and moving the mouse left or right.

27. Use the mouse to grab and move the vertical line of cursor Cl.

    Note: As you do, notice that cursor Cl moves along the signal analyzer's trace and that the vertical and horizontal lines move so that they always intersect at Cl.

28. Repeat Step 27 for cursor C2.

    Note: Fine control over the cursors' position is obtained by using the cursor's Position control in the Cursor Settings area (below the display).

The NI ELVIS II Dynamic Signal Analyzer includes a tool that measures the difference in magnitude and frequency between the two cursors. This information is displayed in green between the upper and lower parts of the display.

29. Move the cursors while watching the measurement readout to observe the effect.

30. Position the cursors so that they're on top of each other and note the measurement.

    Note: When you do, the measurement of difference in magnitude and frequency should both be zero.

Usefully, when one of the cursors is moved to the extreme left of the display, its position on the X-axis is zero. This means that the cursor is sitting on 0Hz. It also means that the measurement readout gives an absolute value of frequency for the other cursor. This makes sense when you think about it because the readout gives the difference in frequency between the two cursors but one of them is zero.

31. Move C2 to the extreme left of the display.

32. Align C1 with the highest point of any one of the humps.

    Note: The readout will now be showing you the frequency of the sinewave that the hump represents.

Recall that the message signal being sampled is a 2kHz sinewave. This means that there should also be a 2kHz sinewave in the sampled message.

33. Use the signal analyzer's C1 cursor to locate sinewave in the sampled message that has the same the frequency as the original message.

Ask the instructor to check your work before continuing.

As discussed earlier, the frequency of all of the sinewaves in the sampled message can be mathematically predicted. Recall that digital signals like the sampling circuit's clock signal are made up out of a DC voltage and many sinewaves (the fundamental and harmonics). As this is a sample-and-hold sampling scheme, the digital signal functions as a series of pulses rather than a squarewave. This means that the sampled signal's spectral composition consists of a DC voltage, a fundamental and both even and odd whole number multiples of the fundamental. For example, the 8kHz sampling rate of your set-up consists of a DC voltage, an 8kHz sinewave (fs), a 16kHz sinewave (2fs), a 24kHz sinewave (3fs) and so on.

The multiplication of the sampling signal's DC component with the sinewave message gives a sinewave at the same frequency as the message and you have just located this in the sampled signal's spectrum.

The multiplication of the sampling signal's fundamental with the sinewave message gives a pair of sinewaves equal to the fundamental frequency plus and minus the message frequency. That is, it gives a 6kHz sinewave (8kHz - 2kHz) and a I0kHz sinewave (8kHz + 2kHz).

In addition to this, the multiplication of the sampling signal's harmonics with the sinewave message gives pairs of sinewaves equal to the harmonics' frequency plus and minus the message frequency. That is, the signal also consists of sinewaves at the following frequencies: I4kHz (I6kHz - 2kHz), I8kHz (I6kHz + 2kHz), 22kHz (24kHz - 2kHz), 26kHz (24kHz + 2kHz) and so on.

All of these sum and difference sinewaves in the sampled signal are appropriately known as aliases.

34. Use the signal analyzer's CI cursor to locate and measure the exact frequency of the sampled signal's first six aliases. Record your measurements in Table I below.

    Tip: Their frequencies will be close to those listed above.

Table I

Alias I		Alias 4	
Alias 2		Alias 5	
Alias 3		Alias 6	

Ask the instructor to check your work before continuing.

Why aren't the alias frequencies exactly as predicted?
You will have notice that the measured frequencies of your aliases don't exactly match the theoretically predicted values. This is not a flaw in the theory. To explain, the Emona DATEx has been designed so that the signals out of the Master Signals module are synchronised. This is a necessary condition for the implementation of many of the modulation schemes in this manual. To achieve this synchronisation, the 8kHz and 2kHz signals are derived from a I00kHz master crystal oscillator. As a consequence, their frequencies are actually 8.3kHz and 2.08kHz.

Part D - Reconstructing a sampled message

Now that you have proven that the sampled message consists of a sinewave at the original message frequency, it's easy to understand how a low-pass filter can be used to "reconstruct" the original message. The LPF can pick-out the sinewave at the original message frequency and reject the other higher frequency sinewaves. The next part of the experiment lets you do this.

35. Suspend the Signal Analyzer VI's operation by clicking on its Stop control once.

    Note: The analyzer's display should freeze.

36. Restart the scope's VI by clicking its Run control once.

37. Launch the DATEx soft front-panel (SFP).

38. Check you now have soft control over the DATEx by activating the PCM Encoder module's soft PDM/TDM control on the DATEx SFP.

    Note: If you're set-up is working correctly, the PCM Decoder module's LED on the DATEx board should turn on and off.

39. Locate the Tuneable Low-pass Filter module on the DATEx SFP and set its soft Gain control to about the middle of its travel.

40. Turn the Tuneable Low-pass Filter module's soft Cut-off Frequency Adjust control fully anti-clockwise.

4l. Modify the set-up as shown in Figure 9 below.

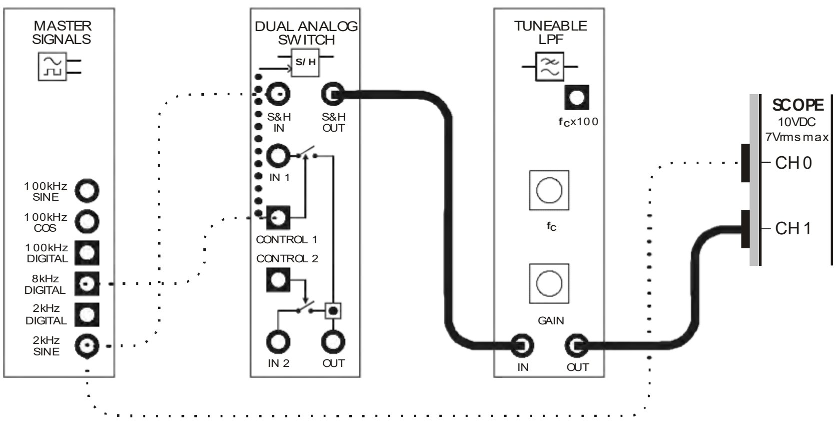


Figure 9

The set-up in Figure 9 can be represented by the block diagram in Figure 10 below. The Tuneable Low-pass Filter module is used to recover the message. The filter is said to be "tuneable" because the point at which frequencies are rejected (called the cut-off frequency) is adjustable.

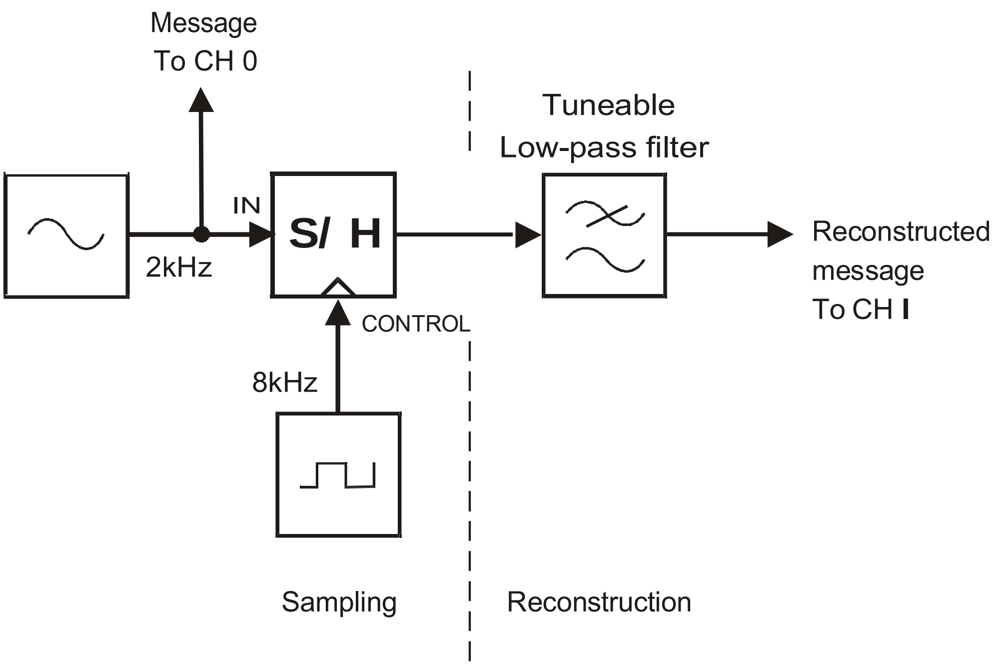


Figure 10

At this point there should be nothing out of the Tuneable Low-pass Filter module. This is because it has been set to reject almost all frequencies, even the message. However, the cut-off frequency can be increased by turning the module's Cut-off Frequency Adjust control clockwise.

42. Slowly turn the Tuneable Low-pass Filter module's soft Cut-off Frequency control clockwise and stop when the message signal has been reconstructed and is roughly in phase with the original message.

Ask the instructor to check your work before continuing.

Part E - Aliasing

At present, the filter is only letting the message signal through to the output. It is comfortably rejecting all of the other sinewaves that make up the sampled message (the aliases). This is only possible because the frequency of these other sinewaves is high enough. Recall from your earlier measurements that the lowest frequency alias is 6kHz.

Recall also that the frequency of the aliases is set by the sampling signal's frequency (for a given message). So, suppose the frequency of the sampling signal is lowered. A copy of the message would still be produced because that's a function of the sampling signal's DC component. However, the frequency of the aliases would all go down. Importantly, if the sampling signal's frequency is low enough, one or more of the aliases pass through the filter along with the message. Obviously, this would distort the reconstructed message which is a problem known as aliasing.

To avoid aliasing, the sampling signal's theoretical minimum frequency is twice the message frequency (or twice the highest frequency in the message if it contains more than one sinewave and is a baseband signal). This figure is known as the Nyquist Sample Rate and helps to ensure that the frequency of the non-message sinewaves in the sampled signal is higher than the message's frequency. That said, filters aren't perfect. Their rejection of frequencies beyond the cut-off is gradual rather than instantaneous. So in practice the sampling signal's frequency needs to be a little higher than the Nyquist Sample Rate.

The next part of the experiment lets you vary the sampling signal's frequency to observe aliasing.

43. Launch and run the NI ELVIS II Function Generator VI.

44. Adjust the function generator for an 8kHz output.

    Note: It's not necessary to adjust any other controls as the function generator's SYNC output will be used and this is a digital signal.

45. Modify the set-up as shown in Figure 11 below.

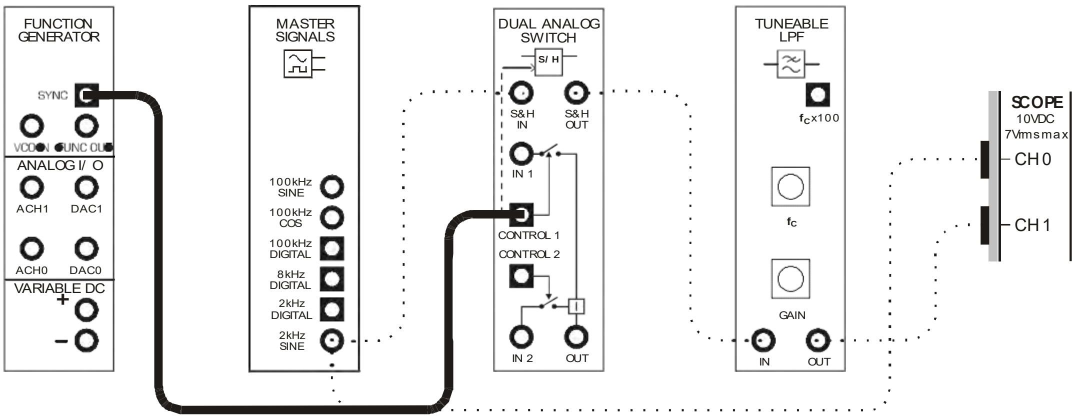


Figure 11

This set-up can be represented by the block diagram in Figure 12 below. Notice that the sampling signal is now provided by the function generator which has an adjustable frequency.

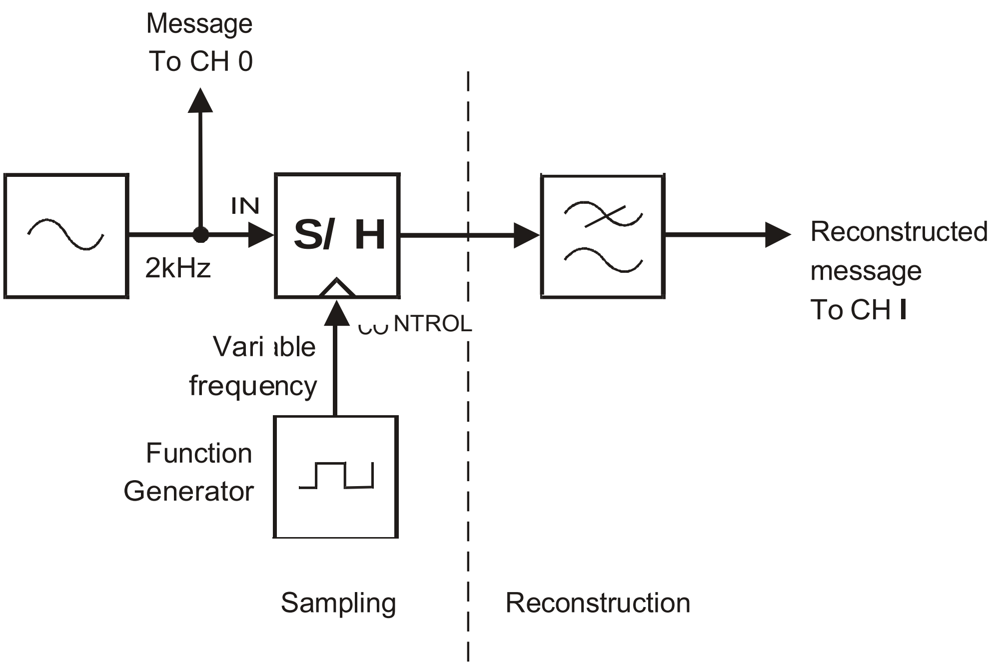


Figure 12

At this point, the sampling of the message and its reconstruction should be working as before.

46. Set the scope's Timebase control to the 500.us/div position.

47. Reduce the frequency of the frequency generator's output by l000Hz and observe the effect this has (if any) on the reconstructed message signal.

    Note: Give the function generator time to output the new frequency before you change it again.

48. Disconnect the scope's Channel I input from the Tuneable Low-pass Filter module's output and connect it to the Dual Analog Switch module's S&H output.

49. Suspend the scope VI's operation.

50. Restart the signal analyzer's VI.

Question 4
What has happened to the sampled signal's aliases?

____

5l. Suspend the signal analyzer VI's operation.

52. Restart the scope's VI.

53. Return the scope's Channel I input to the Tuneable Low-pass Filter module's output.

54. Repeat Steps 47 to 53 until the function generator's output frequency is 3000Hz.

Question 5
What's the name of the distortion that appears when the sampling frequency is low enough?

____

Question 6
What happens to the sampled signal's lowest frequency alias when the sampling rate is 4kHz?

____

____

Ask the instructor to check
your work before continuing.

55. If you've not done so already, repeat Steps 51 to 53.

56. Increase the frequency of the frequency generator's output in 200Hz steps and stop the when the recovered message is a stable, clean copy of the original.

57. Record this frequency in Table 2 below.

Table 2	Frequency
Minimum sampling frequency (without aliasing)	

Question 7
Given the message is a 2kHz sinewave, what's the theoretical minimum frequency for the sampling signal? Tip: If you're not sure, see the notes on page 13-18.

Question 8
Why is the actual minimum sampling frequency to obtain a reconstructed message without aliasing distortion higher than the theoretical minimum that you calculated for Question 5?

Ask the instructor to check
your work before finishing.

Week 13 Experiment 13 - PCM encoding

Test Standard :IEEE 802

Preliminary discussion

As you know, digital transmission systems are steadily replacing analog systems in commercial communications applications. This is especially true in telecommunications. That being the case, an understanding of digital transmission systems is crucial for technical people in the communications and telecommunications industries. The remaining experiments in this book use the Emona DATEx to introduce you to several of these systems starting with pulse code modulation (PCM).

PCM is a system for converting analog message signals to a serial stream of 0s and 1s. The conversion process is called encoding. At its simplest, encoding involves:

- Sampling the analog signal's voltage at regular intervals using a sample-and-hold scheme (demonstrated in Experiment 13).
- Comparing each sample to a set of reference voltages called quantisation levels.
- Deciding which quantisation level the sampled voltage is closest to.
- Generating the binary number for that quantisation level.
- Outputting the binary number one bit at a time (that is, in serial form).
- Taking the next sample and repeating the process.

An issue that is crucial to the performance of the PCM system is the encoder's clock frequency. The clock tells the PCM encoder when to sample and, as the previous experiment shows, this must be at least twice the message frequency to avoid aliasing (or, if the message contains more than one sinewave, at least twice its highest frequency).

Another important PCM performance issue relates to the difference between the sample voltage and the quantisation levels that it is compared to. To explain, most sampled voltages will not be the same as any of the quantisation levels. As mentioned above, the PCM Encoder assigns to the sample the quantisation level that is closest to it. However, in the process, the original sample's value is lost and the difference is known as quantisation error. Importantly, the error is reproduced when the PCM data is decoded by the receiver because there is no way for the receiver to know what the original sample voltage was. The size of the error is affected by the number of quantisation levels. The more quantisation levels there are (for a given range of sample voltages) the closer they are together. This means that the difference between the quantisation levels and the samples is smaller and so the error is lower.

A little information about the PCM Encoder module on the Emona DATEx

The PCM Encoder module uses a PCM encoding and decoding chip (called a codec) to convert analog voltages between -2V and +2V to an 8-bit binary number. With eight bits, it's possible to produce 256 different numbers between 00000000 and 11111111 inclusive. This in turn means that there are 256 quantisation levels (one for each number).

Each binary number is transmitted in serial form in frames. The number's most significant bit (called bit-7) is sent first, bit-6 is sent next and so on to the least significant bit (bit-0). The PCM Encoder module also outputs a separate Frame Synchronisation signal (FS) that goes high at the same time that bit-0 is outputted. The FS signal has been included to help with PCM decoding (discussed in the preliminary discussion of Experiment 15) but it can also be used to help "trigger" a scope when looking at the signals that the PCM Encoder module generates.

Figure 1 below shows an example of three frames of a PCM Encoder module's output data (each bit is shown as both a 0 and a 1 because it could be either) together with its clock input and its FS output.

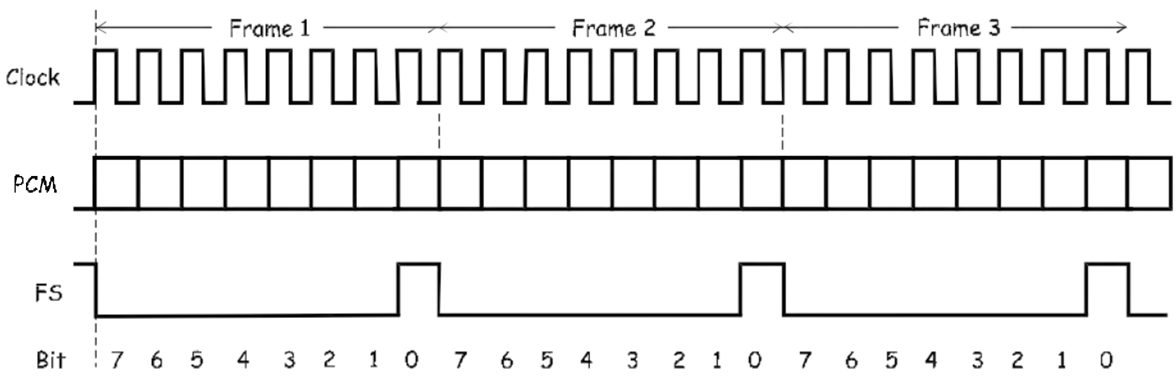


Figure 1

The experiment

For this experiment you'll use the PCM Encoder module on the Emona DATEx to convert the following to PCM: a fixed DC voltage, a variable DC voltage and a continuously changing signal. In the process, you'll verify the operation of PCM encoding and investigate quantisation error a little.

It should take you about 1 hour to complete this experiment.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads

Procedure

Part A - An introduction to PCM encoding using a static DC voltage

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELVISmx software.

8. Launch and run the NI ELVIS II Function Generator VI.

9. Adjust the function generator for a 10kHz output.

   Note: It's not necessary to adjust any other controls as the function generator's SYNC output will be used and this is a digital signal.

10. Connect the set-up shown in Figure 2 below.

Note: Insert the black plugs of the oscilloscope leads into a ground (GND) socket.

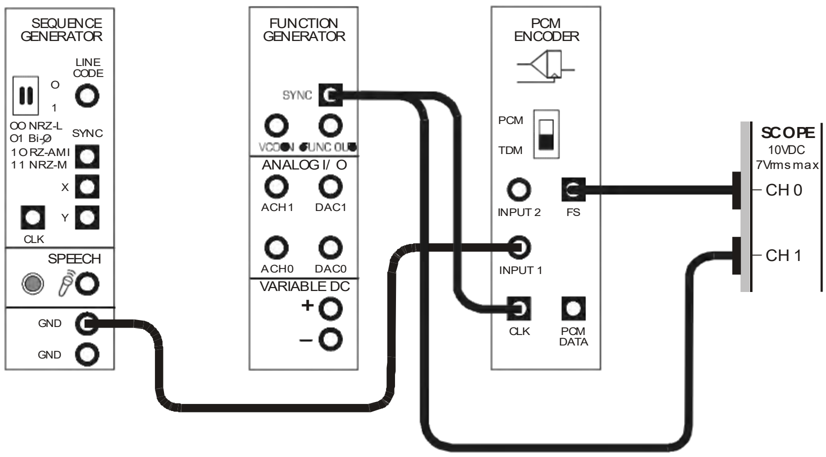


Figure 2

This set-up can be represented by the block diagram in Figure 3 below. The PCM Encoder module is clocked by the function generator output. Its analog input is connected to 0V DC.

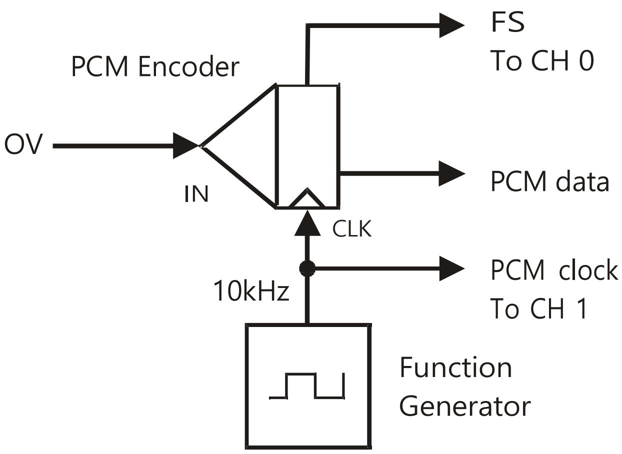


Figure 3

11. Launch the DATEx soft front-panel (SFP).

12. Check you now have soft control over the DATEx by activating the PCM Encoder module's soft PDM/TDM control on the DATEx SFP.

    Note: If you're set-up is working correctly, the PCM Decoder module's LED on the DATEx board should turn on and off.

13. Locate the PCM Encoder module on the Emona DATEx SFP and set its soft Mode switch to the PCM position.

14. Launch and run the NI ELVIS II Oscilloscope VI.

15. Set up the scope per the procedure in Experiment 1 (page 1-12) with the following changes:

    - Scale control for both channels to 2V/div instead of 1V/div
    - Coupling control for both channels to DC instead of AC
    - Trigger Level control to 2V instead of 0V
    - Timebase control to 200JJs/div instead of 500JJs/div

16. Set the scope's Slope control to the  position.

Setting the Slope control to the "-" position makes the scope start its sweep across the screen when the FS signal goes from high to low instead of low to high. You can really notice the difference between the two settings if you flip the scope's Slope control back and forth. If you do this, make sure that the Slope control finishes on the "-" position.

17. Set the scope's Timebase control to the 100JJs/div position.

    Note 1: The FS signal's pulse should be one division wide as shown in Figure 4. If it's not, adjust the function generator's output frequency until it is.

    Note 2: Setting the function generator this way makes each bit in the serial data stream one division wide on the graticule's horizontal axis.

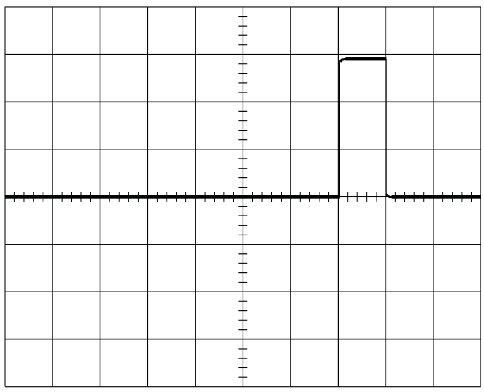


Figure 4

18. Activate the scope's Channel 1 input (by checking the Channel 1 Enabled box) to observe the PCM Encoder module's CLK input as well as its FS output.

    Tip: To see the two waveforms clearly, you may need to adjust the scope so that the two signals are not overlayed.

19. Draw the two waveforms to scale in the space provided below leaving enough room for a third digital signal.

    Tip: Draw the clock signal in the upper third of the graph paper and the FS signal in the middle third.

Ask the instructor to check
your work before continuing.

20. Connect the scope's Channel 1 input to the PCM Encoder module's output as shown in Figure 5 below.

Remember: Dotted lines show leads already in place.

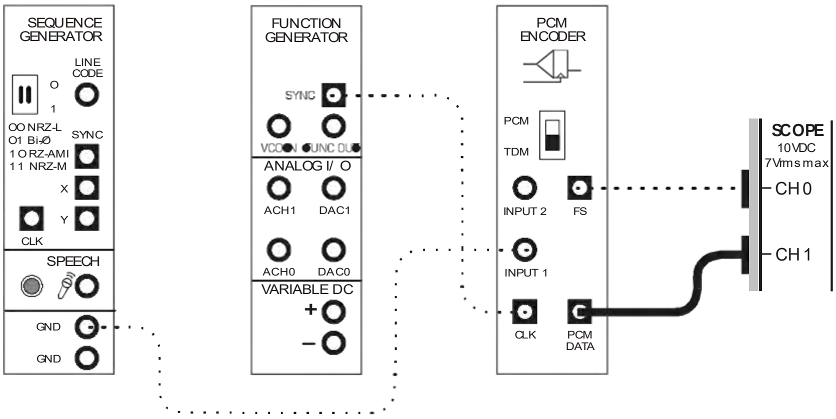


Figure 5

This set-up can be represented by the block diagram in Figure 6 below. Channel 1 should now display 10 bits of the PCM Encoder module's data output. Reading from the left of the display, the first 8 bits belong to one frame and the last two bits belong to the next frame.

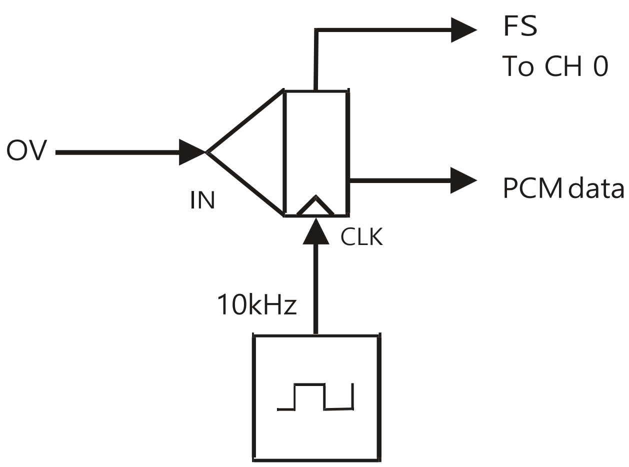


Figure 6

21. Draw this waveform to scale in the space that you left on the graph paper.

Question 1
Indicate on your drawing the start and end of the frame. Tip: If you're not sure where these points are, see the preliminary discussion.

Question 2
Indicate on your drawing the start and end of each bit.

Question 3
Indicate on your drawing which bit is bit-0 and which is bit-7.

Question 4
What is the binary number that the PCM Encoder module is outputting?

Question 5
Why does the PCM Encoder module output this code for 0V DC and not 0000000?

Ask the instructor to check your work before continuing.

Part B - PCM encoding of a variable DC voltage

So far, you have used the PCM Encoder module to convert a fixed DC voltage (0V) to PCM. The next part of the experiment lets you see what happens when you vary the DC voltage.

22. Launch and run the NI ELVIS II Variable Power Supplies VI.

23. Set the Variable Power Supplies two outputs to 0V.

24. Unplug the patch lead connected to the ground socket.

25. Modify the set-up as shown in Figure 7 below.

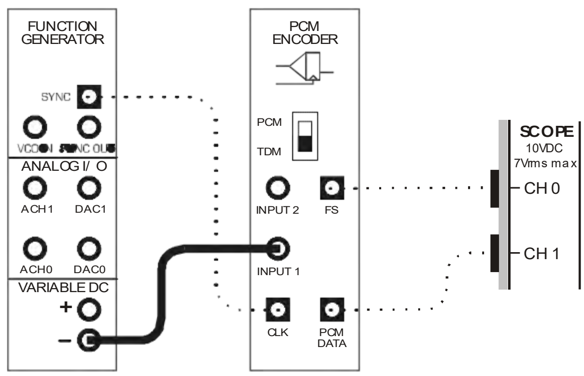


Figure 7

This set-up can be represented by the block diagram in Figure 8 on the next page. The NI ELVIS II Variable Power Supplies is used to let you vary the DC voltage on the PCM Encoder module's input. The scope's external trigger input is used to obtain a stable display.

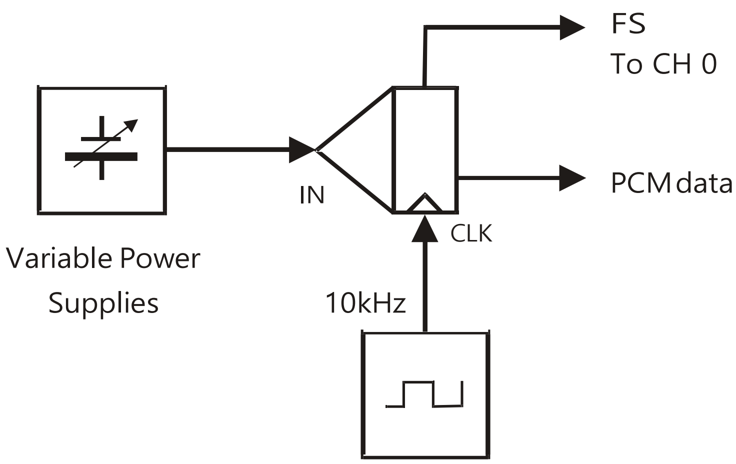


Figure 8

26. Determine the code on the PCM Encoder module's output.

Tip: Remember, the first eight horizontal divisions of the scope's graticule correspond with one frame of the PCM Encoder module's output.

Note: You should find that the PCM Encoder module's output is a binary number that is reasonably close to the code you determined earlier when the module's input was connected directly to ground.

Code: ______________________________

Ask the instructor to check your work before continuing.

27. Increase the Variable Power Supplies' negative output voltage in -0.1V increments and note what happens to the binary number on the PCM Encoder module's output.

Tip: This is easiest to do by simply typing the required voltage in the field under the negative output's Voltage control. When you do, don't forget to put a minus sign in front of the voltage you enter.

Question 6
What happens to the binary number as the input voltage increases in the negative direction?

---

28. Determine the first instance of the changing negative voltage that produces the number 00000000 on the PCM Encoder module's output.

29. Record this voltage in Table 1 below.

Table 1

PCM Encoder's output code	PCM Encoder's input voltage
00000000	

Ask the instructor to check your work before continuing.

30. Modify the set-up as shown in Figure 9 below.

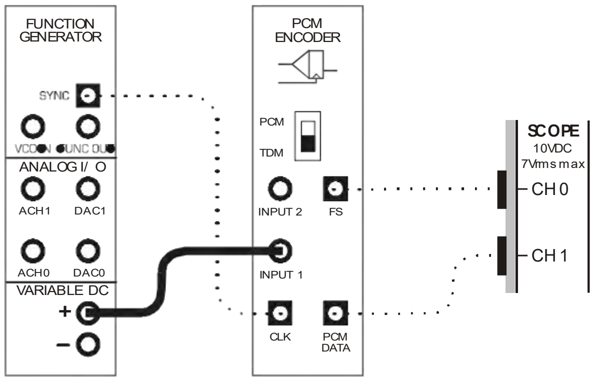


Figure 9

This set-up can be represented by the block diagram in Figure 10 below.

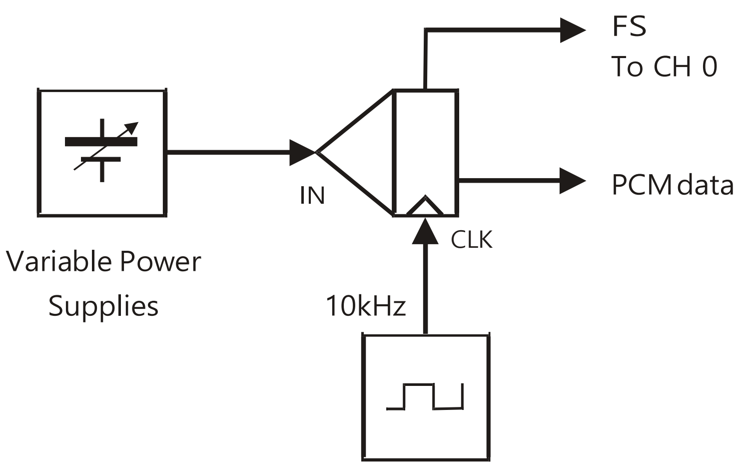


Figure 10

31. Increase the Variable Power Supplies' positive output voltage in +0.1V increments and note what happens to the binary number on the PCM Encoder module's output.

Question 7
What happens to the binary number as the input voltage increases in the positive direction?

32. Determine the lowest positive voltage that produces the number 11111111 on the PCM Encoder module's output.

33. Record this voltage in Table 2 below.

Table 2

PCM Encoder's output code	PCM Encoder's input voltage
11111111	

Question 8
Based on the information in Tables 1 & 2, what is the maximum allowable peak-to-peak voltage for an AC signal on the PCM Encoder module's INPUT?

Question 9
Calculate the difference between the PCM Encoder module's quantisation levels by subtracting the values in Tables 1 & 2 and dividing the number by 256 (the number of codes).

Ask the instructor to check
your work before continuing.

Part C - PCM encoding of continuously changing voltages

Now let's see what happens when the PCM encoder is used to convert continuously changing signals like a sinewave.

34. Close the Variable Power Supplies VI.

35. Disconnect the plugs to the Variable Power Supplies positive output.

36. Modify the set-up as shown in Figure 11 below.

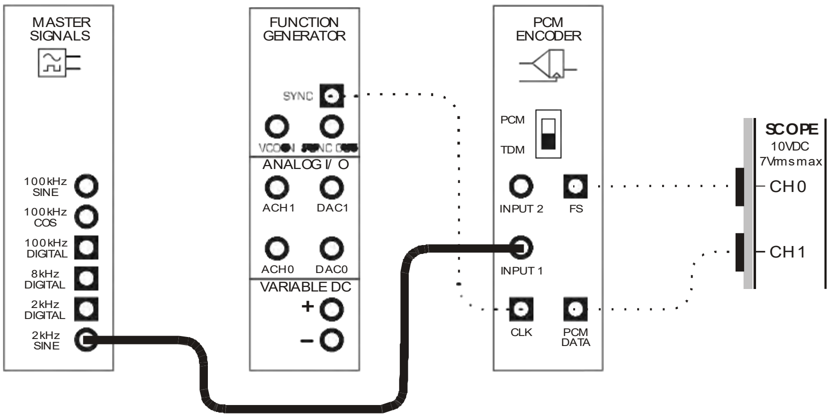


Figure 11

37. Set the function generator's output frequency to 50kHz.

38. Watch the PCM Encoder module's output on the scope's display.

Question 10

Why does the code on PCM Encoder module's output change continuously?

Ask the instructor to check your work before finishing.

Week 14: Experiment 14 - PCM decoding

Test Standard :IEEE 802

Preliminary discussion

The previous experiment introduced you to the basics of pulse code modulation (PCM) which you'll recall is a system for converting message signals to a continuous serial stream of binary numbers (encoding). Recovering the message from the serial stream of binary numbers is called decoding.

At its simplest, decoding involves:

- Identifying each new frame in the data stream.
- Extracting the binary numbers from each frame.
- Generating a voltage that is proportional to the binary number.
- Holding the voltage on the output until the next frame has been decoded (forming a pulse amplitude modulation (PAM) version of the original message signal).
- Reconstructing the message by passing the PAM signal through a low-pass filter.

The PCM decoder's clock frequency is crucial to the correct operation of simple decoding systems. If it's not the same frequency as the encoder's clock, some of the transmitted bits are read twice while others are completely missed. This results in some of the transmitted numbers being incorrectly interpreted, which in turn causes the PCM decoder to output an incorrect voltage. The error is audible if it occurs often enough. Some decoders manage this issue by being able to "self-clock".

There is another issue crucial to PCM decoding. The decoder must be able to detect the beginning of each frame. If this isn't done correctly, every number is incorrectly interpreted. The synchronising of the frames can be managed in one of two ways. The PCM encoder can generate a special frame synchronisation signal that can be used by the decoder though this has the disadvantage of needing an additional signal to be sent. Alternatively, a frame synchronisation code can be embedded in the serial data stream that is used by the decoder to work out when the frame starts.

A little information about the DATEx PCM Decoder module

Like the PCM Encoder module on the Emona DATEx, the PCM Decoder module works with 8-bit binary numbers. For 00000000 the PCM Decoder module outputs -2V and for 11111111 it outputs +2V. For numbers in between, the output is a proportional voltage between ±2V. For example, the number 10000000 is half way between 00000000 and 11111111 and so for this input the module outputs 0V (which is half way between +2V and -2V).

The PCM Decoder module is not self-clocking and so it needs a digital signal on the CLK input to operate. Importantly, for the PCM Decoder module to correctly decode the PCM data generated by the PCM Encoder module, it must have the same clock signal. In other words, the decoder's clock must be "stolen" from the encoder.

Similarly, the PCM Decoder module cannot self-detect the beginning of each new frame and so it must have a frame synchronisation signal on its FS input to do this.

The experiment

For this experiment you'll use the Emona DATEx to convert a sinewave and speech to a PCM data stream then convert it to a PAM signal using the PCM Decoder module. For this to work correctly, the decoder's clock and frame synchronisation signal are simply "stolen" the PCM Encoder module. You'll then recover the message using the Tuneable Low-pass filter module.

It should take you about 45 minutes to complete this experiment.

Equipment

- Personal computer with appropriate software installed
- NI ELVIS II plus USB cable and power pack
- Emona DATEx experimental add-in module
- Two BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads
- One set of headphones (stereo)

Procedure

Part A - Setting up the PCM encoder

To experiment with PCM decoding you need PCM data. The first part of the experiment gets you to set up a PCM encoder.

1. Ensure that the NI ELVIS II power switch at the back of the unit is off.

2. Carefully plug the Emona DATEx experimental add-in module into the NI ELVIS II.

3. Set the Control Mode switch on the DATEx module (top right corner) to PC Control.

4. Connect the NI ELVIS II to the PC using the USB cable.

   Note: This may already have been done for you.

5. Turn on the NI ELVIS II power switch at the rear of the unit then turn on its Prototyping Board Power switch at the top right corner near the power indicator.

6. Turn on the PC and let it boot-up.

7. Launch the NI ELVISmx software.

8. Launch and run the NI ELVIS II Variable Power Supplies VI.

9. Set the Variable Power Supplies' positive output to 0V.

10. Launch the DATEx soft front-panel (SFP) and check that you have soft control over the DATEx board.

11. Set the PCM Encoder module's soft Mode switch to the PCM position.

12. Connect the set-up shown in Figure 1 below.

Note: Insert the black plugs of the oscilloscope leads into a ground (GND) socket.

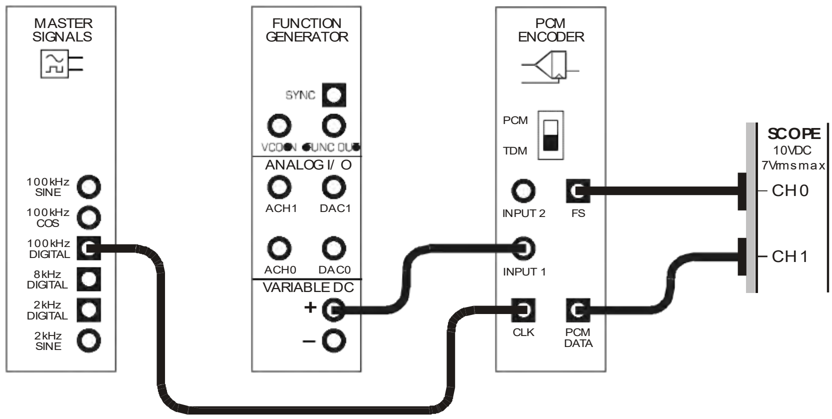


Figure 1

This set-up can be represented by the block diagram in Figure 2 below. The PCM Encoder module is clocked by the Master Signals module's 100kHz DIGITAL output. Its analog input is the Variable Power Supplies' positive output.

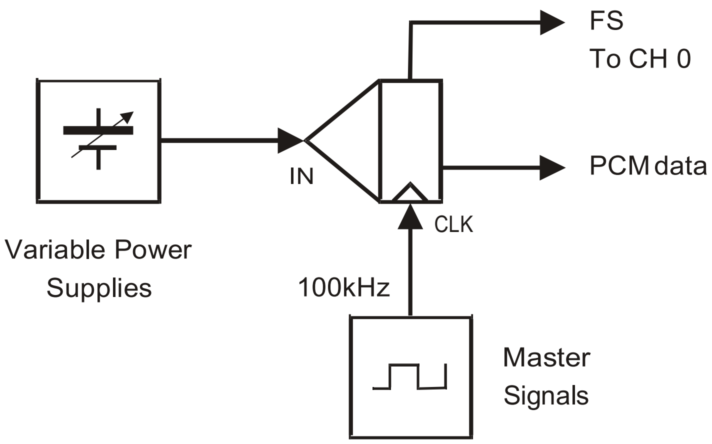


Figure 2

13. Launch and run the NI ELVIS II Oscilloscope VI.

14. Set up the scope per the procedure in Experiment 1 (page 1-12) with the following changes:

    - Scale control for both channels to 2VIdiv instead of 1VIdiv
    - Coupling control for both channels to DC instead of AC
    - Trigger Level control to 2V instead of 0V
    - Timebase control to 10psIdiv instead of 500psIdiv

15. Set the scope's Slope control to the "-" position.

16. Activate the scope's Channel 1 input (by checking the Channel 1 Enabled box) to observe the PCM Encoder module's PCM DATA output as well as its FS output.

17. Vary the Variable Power Supplies positive output Voltage control left and right (but don't exceed 2.5V).

    Note: If your set-up is working correctly, this last step should cause the number on PCM Encoder module's PCM DATA output to go down and up. If it does, carry on to the next step. If not, check your wiring or ask the instructor for help.

18. Close the Variable Power Supplies VI.

19. Launch and run the NI ELVIS II Function Generator VI.

20. Adjust the function generator using its soft controls for an output with the following specifications:

    - Waveshape: Sine
    - Frequency: 500Hz
    - Amplitude: 4Vpp
    - DC Offset: 0V

21. Disconnect the plug to the Variable Power Supplies' positive output.

22. Modify the set-up as shown in Figure 3 below.

Remember: Dotted lines show leads already in place.

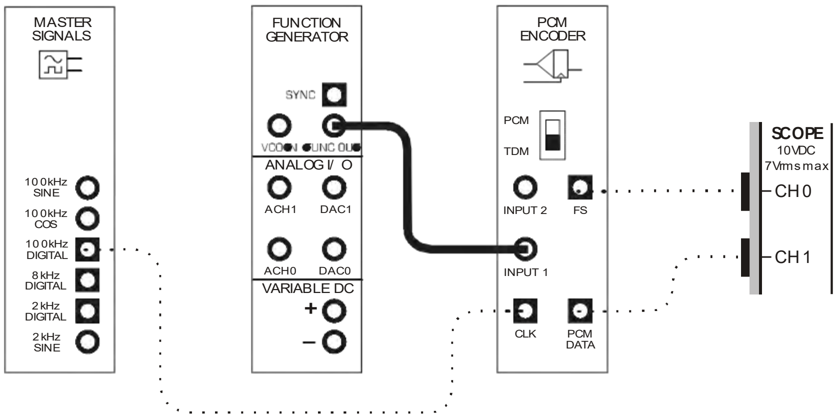


Figure 3

This set-up can be represented by the block diagram in Figure 4 below. Notice that the PCM Encoder module's input is now the function generator's output.

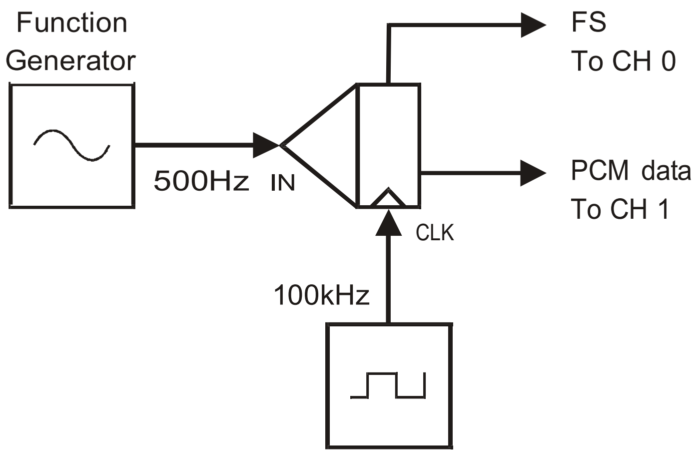


Figure 4

As the PCM Encoder module's input is a sinewave, the module's input voltage is continuously changing. This means that you should notice the PCM DATA output changing continuously also.

Ask the instructor to check
your work before continuing.

Part B - Decoding the PCM data

23. Deactivate the scope's Channel 1 input.

24. Modify the set-up as shown in Figure 5 below.

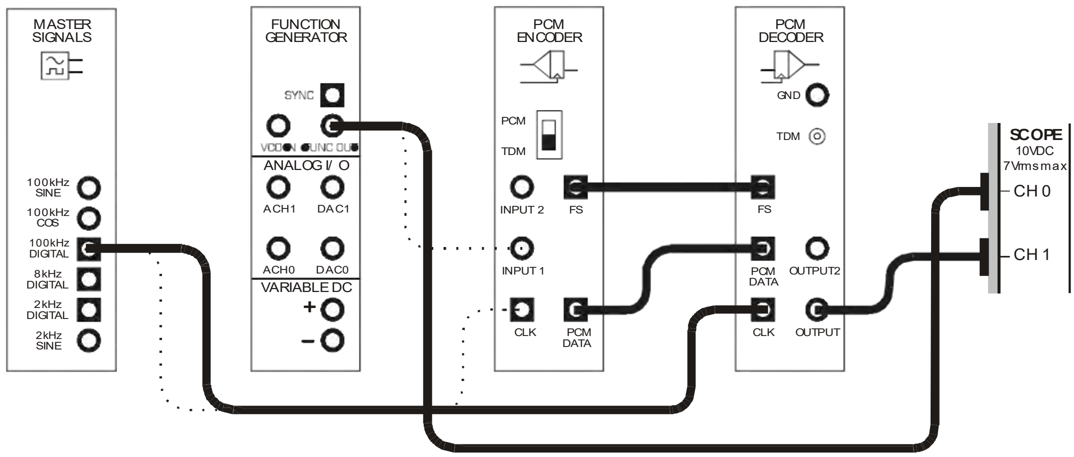


Figure 5

The entire set-up can be represented by the block diagram in Figure 6 on the next page. Notice that the decoder's clock and frame synchronisation information are "stolen" from the encoder.

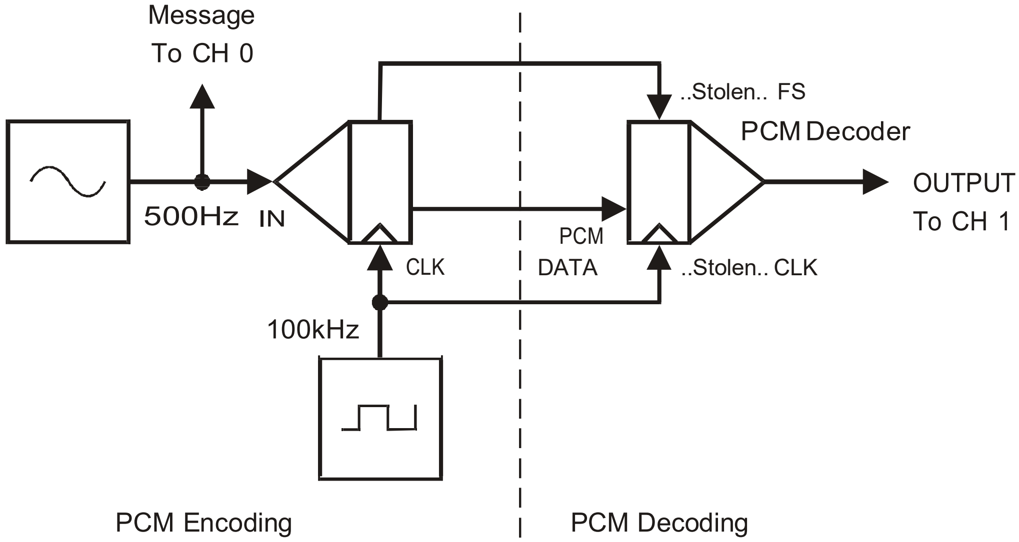


Figure 6

25. Adjust the scope as follows:

- Scale control for both channels to 1VIdiv
- Coupling control for both channels to AC
- Trigger Level control to 0V
- Slope control back to the "+" position
- Timebase control to 500psIdiv

26. Activate the scope's Channel 1 input to observe the PCM Decoder module's output as well as the message signal.

Question 1
What does the PCM Decoder's "stepped" output tell you about the type of signal that it is? Tip: If you're not sure, see the preliminary discussion for this experiment or for Experiment 13.

Ask the instructor to check your work before continuing.

The PCM Decoder module's output signal looks very similar to the message. However, they're not the same. Remember that a "sampled" message contains many sinewaves in addition to the message. The next part of this experiment lets you verify this using the NI ELVIS II Dynamic Signal Analyzer.

27. Suspend the scope's VI.

28. Launch and run the NI ELVIS II Dynamic Signal Analyzer VI.

29. Adjust the signal analyzer's controls as follows:

Input Settings

- Source Channel to SCOPE CH 1
- Voltage Range to ±10V

FFT Settings

- Frequency Span to 40,000
- Resolution to 400
- Window to 7 Term B-Harris

Averaging

- Mode to RMS
- Weighting to Exponential
- # of Averages to 3

Trigger Settings

- Type to Edge

Frequency Display

- Units to dB
- Mode to RMS
- Scale to Auto
- Cursors On box unchecked (for now)

30. Activate the signal analyzer's cursors (by checking Cursors On box).

31. Use the signal analyzer's C1 cursor to examine the frequency of the significant sinewaves that make up the sampled message.

32. Use the C1 cursor to locate the sinewave in the sampled message that has the same the frequency as the original message.

Ask the instructor to check
your work before continuing.

You have probably just noticed that some of the extra sinewaves in the sampled message are at audible frequencies (that is, between about 20Hz and 20kHz, and in particular those about 12.5kHz). This means that, although the message and sampled messages are similar in shape, you may be able to hear a difference between them. Keep in mind that many of the components below the -40dB line are not significant. Signals that are visually below this level (but bigger numerically) are less than 1o of the level of any signals at 0dB.

Where do these "extra" components come from?
As the data clock rate is 100kHz, the "frame rate" is 1l8th of that because there are 8 bits per frame. Therefore, the PCM system is sampling the message at 12.5kHz or every 80ps.

Importantly, this means that the PCM module's output changes voltage in steps every 80ps. The spectral composition of these steps creates extra "images" of the message at 12.5kHz, 2 x 12.5kHz (25kHz), 3 x 12.5kHz (37.5kHz) and so on. It is these "images" that you can see in the spectrum.

33. Add the Amplifier module to the set-up as shown in Figure 7 below leaving the scope's connections as they are.

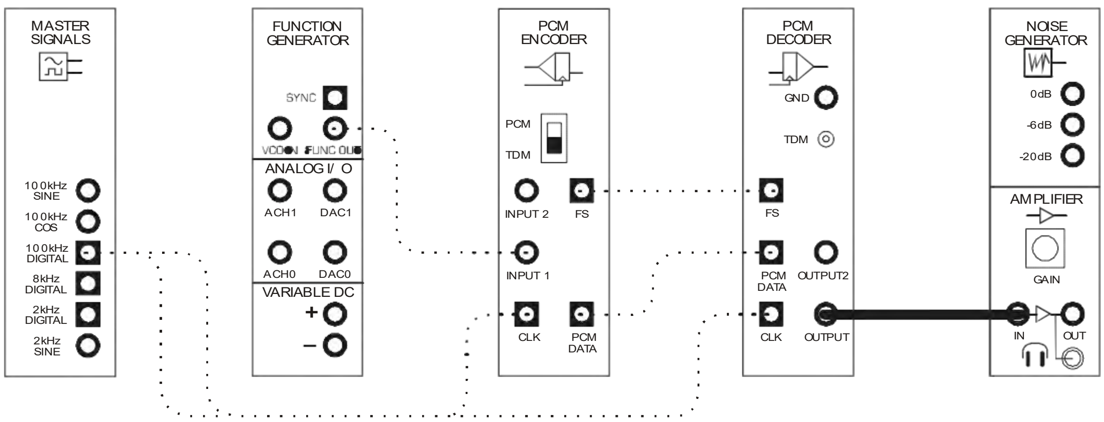


Figure 7

34. Locate the Amplifier module on the DATEx SFP and turn its soft Gain control fully anti-clockwise.

35. Without wearing the headphones, plug them into the Amplifier module's headphone socket.

36. Put the headphones on.

37. Turn the Amplifier module's soft Gain control clockwise until you can comfortably hear the PCM Decoder module's output.

38. Listen to how the sampled message sounds and commit it to memory.

39. Disconnect the Amplifier module's lead where it plugs to the PCM Decoder module's output.

40. Modify the set-up as shown in Figure 8 below, again leaving the scope's connections as they are.

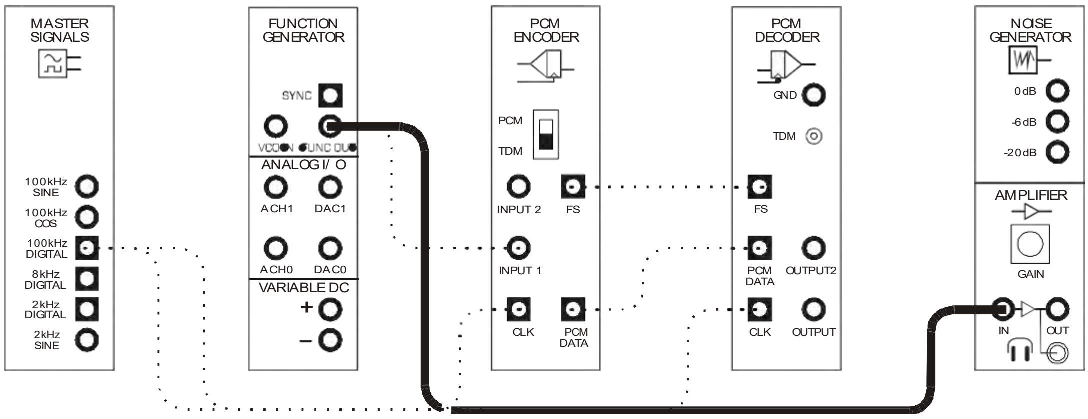


Figure 8

41. Compare the sound of the two signals. You should notice that they're similar but clearly different.

Question 2
What must be done to the PCM Decoder module's output to reconstruct the message properly?

Ask the instructor to check your work before continuing.

Part C - Encoding and decoding speech

So far, this experiment has encoded and decoded a sinewave for the message. The next part of the experiment lets you do the same with speech.

42. Close the signal analyzer's VI and restart the scope's VI.

43. Adjust the scope so that you can observe two or so cycles of the original and sampled messages again.

    Tip: Don't forget to check that the scope's Trigger Source control is set to the CH 0 position.

44. Completely remove the Amplifier module from the set-up while leaving the rest of the leads in place.

45. Disconnect the plugs to the function generator's output.

46. Modify the set-up as shown in Figure 9 below.

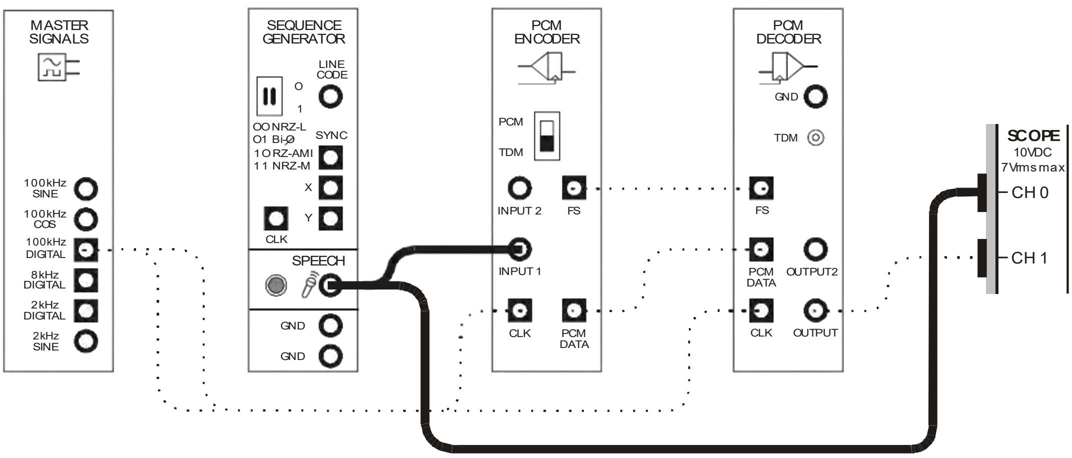


Figure 9

47. Hum and talk into the microphone while watching the scope's display.

Ask the instructor to check
your work before continuing.

Part D - Recovering the message

As mentioned earlier, the message can be reconstructed from the PCM Decoder module's output signal using a low-pass filter. This part of the experiment lets you do this.

48. Locate the Tuneable Low-pass Filter module on the DATEx SFP and set its soft Gain control to about the middle of its travel.

49. Turn the Tuneable Low-pass Filter module's soft Cut-off Frequency Adjust control fully anti-clockwise.

50. Disconnect the plugs to the Speech module's output.

51. Modify the set-up as shown in Figure 10 below.

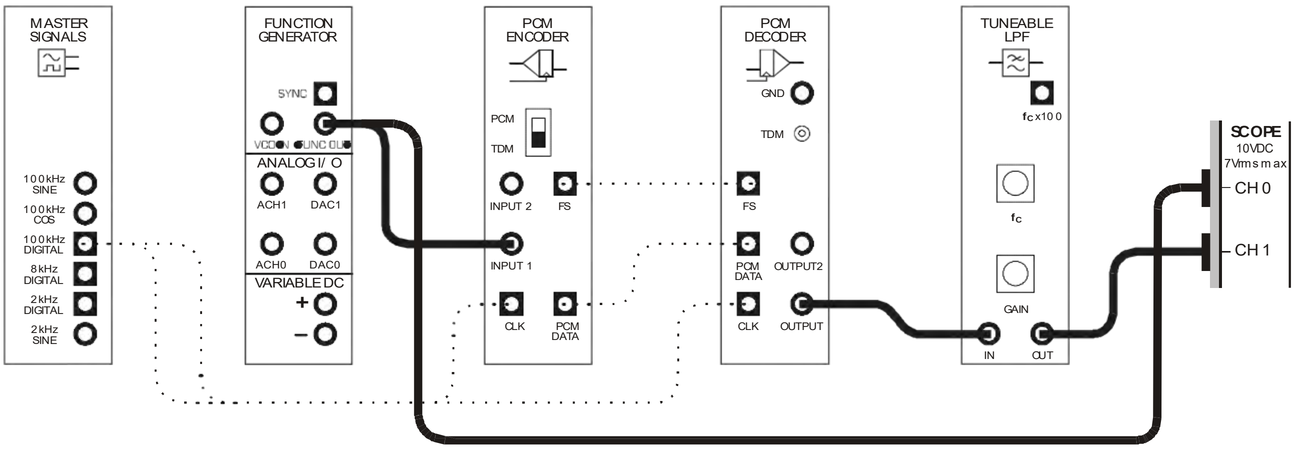


Figure 10

The entire set-up can be represented by the block diagram in Figure 11 on the next page. The Tuneable Low-pass Filter module is used to reconstruct the original message from the PCM Decoder module's PAM output.

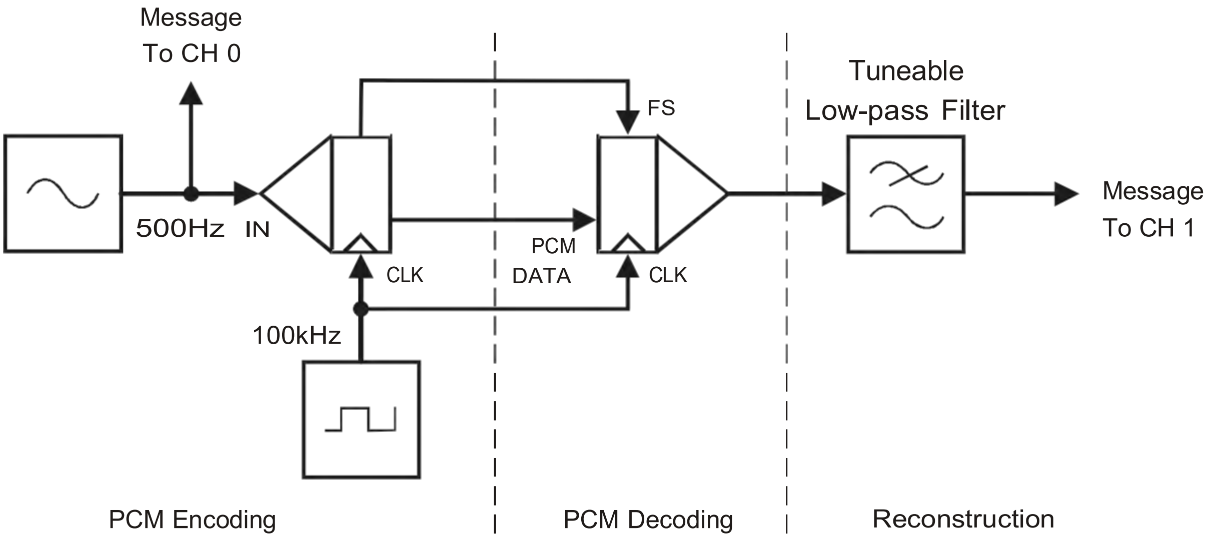


Figure 11

52. Slowly turn the Tuneable Low-pass Filter module's soft Cut-off Frequency control clockwise and stop the moment the message signal has been reconstructed (ignoring phase shift).

The two signals are clearly the same so let's see what your hearing tells you.

53. Add the Amplifier module to the set-up as shown in Figure 12 below leaving the scope's connections as they are.

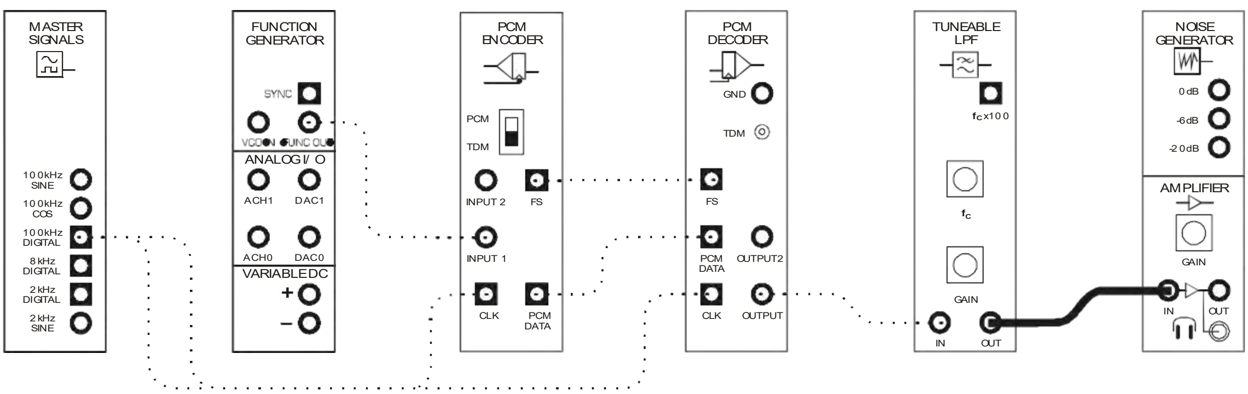


Figure 12

54. Turn the Amplifier module's soft Gain control fully anti-clockwise.

55. Put the headphones on.

56. Turn the Amplifier module's soft Gain control clockwise until you can comfortably hear the Tuneable Low-pass Filter module's output.

57. Commit the recovered message's sound to memory.

58. Disconnect the Amplifier module's lead where it plugs to the PCM Decoder module's output and connect it to the function generator's output (in the same way that you did when wiring the set-up in Figure 8).

59. Compare the sound of the two signals. You should find that they're very similar.

Question 3
Even though the two signals look and sound the same, why isn't the reconstructed message a perfect copy of the original message? Tip: If you're not sure, see the preliminary discussion for Experiment 14.

Ask the instructor to check your work before finishing.