Emona 101 Trainer SAMPLE Lab Manual

Volumes 1 and 2 Experiments in Modern Analog & Digital Telecommunications

Barry Duncan



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Emona-101 Experimenter SAMPLE Lab Manual Volumes 1 and 2 -Experiments in Modern Analog and Digital Telecommunications.

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Introduction

The ETT-101 Experimenter

As its name implies, the Emona Telecoms-Trainer 101 is used to help students learn about communications and telecommunications principles. It lets you bring to life the block diagrams that fill communications textbooks. A "block diagram" is a simplified representation of a more complex circuit. An example is shown in Figure 1 below.

Block diagrams are used to explain the principle of operation of electronic systems (like a radio transmitter for example) without worrying about how the circuit works. Each block represents a part of the circuit that performs a separate task and is named according to what it does. Examples of common blocks in communications equipment include the adder, multiplier, oscillator, and so on.





The Emona Telecoms-Trainer 101, illustrated below, has a collection of blocks (called *modules*) that you can put together to implement dozens of communications and telecommunications block diagrams.



The ETT-101 Lab Manual

The ETT-101 Lab Manual Volume One, is written specifically for students enrolled in a first course in telecommunications at unviersity or a specilaist course at a vocational college. The experiments are intended to support the theoretical concepts introduced in class work. It is assumed that the students only have a basic knowledge of mathematics and a limited background in physics and electricity.

The 20 chapters cover a broad range of telecommunications concepts, from fundamental topics familiar to all students, such as AM and FM broadcasting, through to the latest technologies used in mobile telepohnes and wireless networking. In each experiment, the core technology is revealed to the student, at its most fundamental and simplest level.

In order to make the student's learning experience more memorable, the student is usually able to both view signals on the oscilloscope and then listen to their own voice undergoing the modulation or coding scheme being investigated.

Another important factor which makes the learning experimence more valuable for the student is the ETT-101 board's experimental approach: the student is "allowed to make mistakes". As the student builds the experiment, they need to make constant observations and adjustments. If signals are not as expected then the student needs to make a decision as to whether the correction required is an adjustment or an incorrect interconnection.

Each experiment in the ETT-101 Lab Manual provides a basic introduction to the topic under investigation, followed by a series of carefully graded hands-on activities. At the conclusion of each sub section the student is asked to answer questions to confirm their understanding of the work before proceeding.

Chapters can be covered in any order, however, it is imperative that all students complete the first three before proceeding to the subsequent chapters. Chapter 1 introduces the use of an oscilloscope to view electrical signals. Chapter 2 follows with an introduction to the functions of the ETT-101 and chapter 3 introduces the concept of mathematical modeling using electronic functional blocks.

Since the ETT-101 Experimenter is a true modeling system, the instructor has the freedom to modify existing experiments or even create completely new experiments to convey new concepts to students.

Finally, it should be noted that the ETT-101 Experimenter can implement many more experiments than are documented in this Volume One Lab Manual and further experiments can be found in later manuals.

3 - Using the Telecoms-Trainer 101 to model equations

Name: Class:

Experiment 3 - Using the Telecoms-Trainer 101 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 Telecoms-Trainer 101 can model communications equations to bring them to life.

The experiment

This experiment will introduce you to modelling equations by using the Emona Telecoms-Trainer 101 to implement two relatively simple equations.

It should take you about 45 minutes to complete this experiment.

Equipment

- Emona Telecoms-Trainer 101 (plus power-pack)
- Dual channel 20MHz oscilloscope
- two Emona Telecoms-Trainer 101 oscilloscope leads
- assorted Emona Telecoms-Trainer 101 patch leads

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 1

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

Procedure

- 1. Gather the equipment listed on page 3-2.
- 2. Set up the scope per the instructions in Experiment 1. Ensure that:
 - the *Trigger Source* control is set to the *CH1* (or *INT*) position.
 - the *Mode* control is set to the *CH1* position.
- 3. Locate the Adder module and set its G and g controls to about the middle of their travel.
- 4. 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 2

This set-up can be represented by the block diagram in Figure 3 on the next page.





- 5. Adjust the scope's *Timebase* control to view two or so cycles of the Master Signals module's *2kHz SINE* output.
- 6. Disconnect the lead to the Adder module's B input.
- 7. Measure the amplitude (peak-to-peak) of the Master Signals module's *2kHz SINE* output.

Record your measurement here:

- 8. Set the scope's *Mode* control to the *CH2* position.
- 9. Adjust the Adder module's G control until its output voltage is the same size as its input voltage (measured in Step 7).

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

- 10. Reconnect the lead to the Adder module's B input.
- 11. Disconnect the lead to the Adder module's A input.
- 12. Adjust the Adder module's g control until its output voltage is the same size as its input voltage (measured in Step 7).

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.

13. 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, with values the equation is:

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 we get in practice.

- 14. Set the scope's *Mode* control to the *CH1* position.
- 15. Measure the amplitude of the Master Signals module's *2kHz SINE* output. Record your measurement in Table 1 below.

Note: The voltage may be a little different to that measured in Step 7 due to loading of the Master Signals module's *2kHz SINE* output.

- 16. Set the scope's *Mode* control to the *CH2* position.
- 17. Measure and record the amplitude of the Adder module's output.

Table 1			
Input voltage	Output voltage		

Question 1

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

No.

Question 2 What are two reasons for this?

1) Measurement error; and

2) The gains aren't exactly -1. (Scopes are not accurate measuring instruments.)



Ask the instructor to check your work before continuing.

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)

- 18. Locate the Phase Shifter module and set its *Phase Change* control to the *180°* position.
- 19. Set the Phase Shifter module's *Phase Adjust* control about the middle of its travel.
- 20. Connect the set-up shown in Figure 4 below.

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





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





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 = OVp-p

Let's see if this is what we get in practice.

- 21. Set the scope's *Mode* control to the *DUAL* position to view the Phase Shifter module's output **as well as** the Master Signals module's *2kHz SINE* output.
- 22. Adjust the Phase Shifter module's *Phase Adjust* control until the two signals look like they're 180° out of phase with each other.
- 23. Disconnect the scope's Channel 2 lead from the Phase Shifter module's output and connect it to the Adder module's output.
- 24. Set the scope's *Mode* control to the *CH2* position.
- 25. Measure the amplitude of the Adder module's output. Record your measurement in Table 2 (on the next page).

Tip: You'll probably need to adjust the Channel 2 *Vertical Attenuation* control to obtain an appropriate display (try the *0.1V/div* setting).



Question 3

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

1) The phase difference between the Adder's two inputs is not exactly 180°; and

2) The gains aren't exactly the same.



Ask the instructor to check your work before continuing.

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.

- 26. Vary the Phase Shifter module's *Phase Adjust* control left and right and observe the effect on the Adder module's output.
- 27. Adjust the Phase Shifter module's *Phase Adjust* control to obtain the smallest output voltage.

Question 5

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

The phase shift is much closer to 180° (but it's still not exactly 180°)

- 28. Vary the Adder module's g control left and right and observe the effect on the Adder module's output.
- 29. Adjust the Adder module's g control to obtain the smallest output voltage.

Question 6

What can be said about the gain of the Adder module's two inputs now?

They're much closer to each other (but they're still not exactly the same)



Ask the instructor to check your work before finishing.

11 - Sampling and reconstruction

Name: Class:

Experiment 11 - Sampling and reconstruction

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 1a 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 1b 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 1a

Figure 1b

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 × 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) × message

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 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 lowpass 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 C of the experiment.

The experiment

In this experiment you'll use the Emona Telecoms-Trainer 101 to sample a message using natural sampling and a sample-and-hold scheme. You'll then reconstruct the message from the sampled signal and examine the effect of aliasing.

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

Equipment

- Emona Telecoms-Trainer 101 (plus power-pack)
- Dual channel 20MHz oscilloscope
- two Emona Telecoms-Trainer 101 oscilloscope leads
- assorted Emona Telecoms-Trainer 101 patch leads

Part A - Sampling a simple message

The Emona Telecoms-Trainer 101 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. Gather a set of the equipment listed above.
- 2. 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

The set-up in Figure 2 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.





- 3. Set up the scope per the instructions in Experiment 1. Ensure that:
 - the *Trigger Source* control is set to the *CH1* (or *INT*) position.
 - the *Mode* control is set to the *CH1* position.
- 4. Adjust the scope's *Timebase* control to view two or so cycles of the Master Signals module's *2kHz SINE* output.
- 5. Set the scope's *Mode* control to the *DUAL* position to view the sampled message out of the Dual Analog Switch module as well as the message.
- 6. Set the scope's Vertical Attenuation controls to the 1V/div position.
- 7. 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 1 What type of sampling is this an example of?

- 🗆 Natural 🗸
- □ Sample-and-hold

Question 2

What two features of the sampled signal confirm this?

1) The sample voltages change during sampling.

2) The signal's voltage returns to zero volts between the samples.



Ask the instructor to check your work before continuing.

8. 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

The set-up in Figure 4 can be represented by the block diagram in Figure 5 below. 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

9. 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 sampleand-hold scheme?

1) The sample voltages don't change during sampling.

2) There's no space between the samples.



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.

- 10. Disconnect the plugs to the Master Signals module's 2kHz SINE output.
- 11. Connect them to the Speech module's output as shown in Figure 6 below.

Remember: Dotted lines show leads already in place.





- 12. Set the scope's *Timebase* control to the *2ms/div* position.
- 13. Talk, sing or hum while watching the scope's display.



Ask the instructor to check your work before continuing.

Part C - Reconstructing a sampled message

Recall that the sampled message is made up of many sinewaves. Importantly, for every sinewave in the message, there's a sinewave in the sampled message with the same frequency. So "reconstructing" the original message involves passing the sampled message signal through a low-pass filter. This lets the sinewave (or sinewaves) with the same frequency as the message through while rejecting the other sinewaves. The next part of the experiment lets you do this.

- 14. Return the scope's *Timebase* control to the *0.1ms/div* position.
- 15. Locate the Tuneable Low-pass Filter module and set its *Gain* control to about the middle of its travel.
- 16. Turn the Tuneable Low-pass Filter module's *Cut-off Frequency Adjust* control fully anticlockwise.
- 17. Disconnect the plugs to the Speech module's output.
- 18. 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 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 8

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.

19. Slowly turn the Tuneable Low-pass Filter module's *Cut-off Frequency* control clockwise and stop the moment the message signal has been reconstructed.



Ask the instructor to check your work before continuing.

Part D - Aliasing

At present, the filter is only letting the message signal through to the output. It is comfortably rejecting all of the other sinewaves (called *aliases*) that make up the sampled message. This is only possible because the frequency of these other sinewaves is high enough. But, this isn't an accident. Their frequency is determined by the sampling rate (that is, the sampling signal's frequency).

To explain, recall that the sampled message consists of the following:

- 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

In your set-up, a 2kHz sinewave message is sampled using an 8kHz sampling signal. That being the case, the lowest frequency components of the sampled message are:

- 2kHz
- 6kHz
- 10kHz

Now, suppose the frequency of the sampling signal is lowered. You'd still get the message but the frequency of the aliases would go down as well. So for example, if the sampling signal is 7kHz, the lowest frequency components of the sampled message become 2kHz, 5kHz and 9kHz. Clearly, if the sampling signal's frequency is low enough, one or more of the lower frequency aliases can pass through the filter along with the reconstructed 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*. So for this set-up, the minimum sampling rate is 4kHz and the lowest frequency components would be 2kHz, 2kHz and 6kHz.

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.

- 20. Locate the VCO module and set its *Frequency Adjust* control fully clockwise.
- 21. Set the VCO module's Range control to the LO position.
- 22. Modify the set-up as shown in Figure 9 below.





This set-up can be represented by the block diagram in Figure 10 below. Notice that the sampling signal is now provided by the VCO module which has a manually adjustable frequency.



Figure 10

At this point, the sampling of the message and its reconstruction should be working normally.

23. Slowly reduce the frequency of the VCO module's output (by turning its *Frequency Adjust* control anti-clockwise) while watching the reconstructed message signal.

Question 4

What's the name of the distortion that appears when the VCO module's *Frequency Adjust* control is turned far enough?

Aliasing.

Question 5

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 11-12.

4kHz (2 × 2kHz)

- 24. Turn the VCO module's *Frequency Adjust* control clockwise and stop the moment the reconstructed message is no-longer distorted.
- 25. Connect the scope's Channel 1 input to the VCO module's DIGITAL output.
- 26. Set the scope's *Mode* control to the *CH1* position.
- 27. Adjust the scope's *Timebase* control to view two or so cycles of the VCO module's *DIGITAL* output.
- 28. Measure the signal's period and record this in Table 1 on the next page.

Tip: If you're not sure how to measure the signal's period, see Experiment 1 (page 1-7).

29. Use the period to calculate and record the signal's frequency.

Tip: If you're not sure how to calculate the signal's frequency, see Experiment 1 (page 1-8).

Table 1	Period	Frequency
VCO module's DIGITAL output		

Question 6

Why is the actual minimum sampling frequency higher than the theoretical minimum that you calculated for Question 5?

Because filters aren't perfect - their cut-off is not instantaneous.



Ask the instructor to check your work before finishing.

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Appendix A - Initial set-up of an oscilloscope

Introduction - ETT-101 Lab Manual Volume 2

Introduction to the ETT-101 Experimenter

As its name implies, the Emona Telecoms-Trainer 101 is used to help students learn about communications and telecommunications principles. It lets you bring to life the block diagrams that fill communications textbooks. A "block diagram" is a simplified representation of a more complex circuit. An example is shown in Figure 1 below.

Block diagrams are used to explain the principle of operation of electronic systems (like a radio transmitter for example) without worrying about how the circuit works. Each block represents a part of the circuit that performs a separate task and is named according to what it does. Examples of common blocks in communications equipment include the adder, multiplier, oscillator, and so on.





The Emona Telecoms-Trainer 101, illustrated below, has a collection of blocks (called *modules*) that you can put together to implement dozens of communications and telecommunications block diagrams.



One-to-One Relationship

The figure on the right illustrates the one-to-one relationship between each block of the BLOCK DIAGRAM and the independent functional circuit blocks of the ETT-101 trainer board.

The functional blocks of the ETT-101 board are used and re-used in experiments, just as blocks of the block diagram reappear in many different implementations.



Examples of ETT-101 functional blocks

Guidelines for Using the ETT-101 Lab Manual Volume 2

The ETT-101 Lab Manual Volume 2 covers a broad range of introductory digital and analog telecommunications topics through a series of 22 hands-on laboratory experiments, which build on the theory and topics covered in the Volume 1 Lab Manual. Each experiment is written to support the theoretical concepts introduced in the class work of a first course in modern telecommunications.

Chapters can be covered in any order, however, it is imperative that all students complete the first three chapters of the Volume 1 Lab Manual before proceeding to the Volume 2 experiments.

Each ETT-101 experiment presents an interesting, hands-on learning experience for the student. In each experiment the student is challenged to build, measure and consider: there are no "instant" or "cookbook-style" experiments. The ETT-101 is actually a true engineering modeling system where students see that the block diagrams so common in their textbooks represent real functioning systems.

In order to make the student's learning experience more memorable, the student is usually able to both view signals on the NI ELVIS oscilloscope and then listen to their own voice undergoing the modulation or coding being investigated.

Test Equipment Required

Along with the complete **ETT-101 Kit** - ETT-101 Trainer Board, power pack, patching leads and accessories - an **oscilloscope** is required for each experiment. The minimum requirement is that the oscilloscope has 2 channels and a minimum bandwidth of 1MHz. Therefore, a commonly available 20MHz, 2 channel oscilloscope is ideal. Alternatively, a PC-based virtual oscilloscope with at least 2 channels and a minimum sample rate of 1MS/s per channel - can also be used.

Note that a PC-based virtual oscilloscope will usually also provide spectrum analyzer (FFT) and volt meter displays. These functions are required or recommended in a number of the Lab Manual Volume 2 experiments, as listed below.

TEST INSTRUMENT:

EXPERIMENTS:

OSCILLOSCOPE

Experiments:

Required to complete all experiments in Volume 2.

SPECTRUM ANALYZER (FFT) Experiments:

Usually provided as an additional function of most PC-based virtual instruments. REQUIRED - required to complete each of the following experiments.

- 13 Principles of GFSK
- 14 PN sequence spectra and noise generation
- 15 Line coding and bit clock regeneration
- 18 Observations of AM & DSBSC signals in the freq domain

SPECTRUM ANALYZER (FFT) Experiments:

OPTIONAL - use is defined in the SUPPLEMENTARY TASKS described below.

- 3 PCM and time division multiplexing (TDM) Supplement should be undertaken between Parts B & C
- 5 Phase division modulation and demodulation Supplement should be undertaken between Parts A & B
- 6 Pulse-width modulation & demodulation
 Supplement should be undertaken between Parts C & D
- 7 Message translation & inversion Supplement should be undertaken between Parts B & C
- 12 FSK generation (switching method) & demodulation Supplement should be undertaken between Parts B & C

TRUE RMS VOLTMETER

Experiments:

Usually provided as an additional function of most PC-based virtual instruments. REQUIRED - required to complete each of the following experiments.

- 2 Noise in AM communications
- 9 SNR & eye diagrams
- 10 PCM and SNDR
- 16 Delta modulation & demodulation

STANDARD VOLTMETER

Experiments:

Usually provided as an additional function of most PC-based virtual instruments. REQUIRED - required to complete the following experiment.

6 - Pulse-width modulation & demodulation

Student Academic Level

Experiments in this volume have been prepared for students with only a basic knowledge of mathematics and a limited background in physics and electricity.

Students with a higher level of competence in mathematics will also gain a deeper understanding of telecommunications theory by using the ETT-101 trainer. Due to the engineering "modeling" nature of the ETT-101, they will be able to investigate more complex issues, carry out additional measurements and then contrast their findings to their theoretical understanding and mathematical analysis.

Making Mistakes and Mis-wiring

An important factor which makes the learning experience more valuable for the student is that the student is allowed to make wiring mistakes. The ETT-101 inputs and outputs can be connected in any combination, without causing damage. As the student builds the experiment, they need to make constant observations, adjustments and corrections. If signals are not as expected then the student needs to make a decision as to whether the correction required is an adjustment or an incorrectly placed patching wire.

Flexibility and Modification of the Experiments and Topics

Since the ETT-101 Trainer is a true modeling system, the instructor has the freedom to modify existing experiments or even create completely new experiments to convey new and course specific concepts to students.

8 - Carrier acquisition using the phase-locked loop

Name: Class:

Experiment 8 - Carrier acquisition using the phase-locked loop

Preliminary discussion

Product detection is a long-standing commercial method used for demodulating SSB signals that can also be used to demodulate DSB and AM signals. Recall though that, crucial to the correct operation of the product detector is the synchronisation of the modulator's carrier signal and the product detector's local carrier. A small frequency error between the carriers causes the message signal's amplitude to vary continuously (at the rate of the error). Larger frequency errors can render the recovered message completely unintelligible. Phase errors between the two signals result in a reduced output from the receiver and possibly even no output at all (for a phase error of 90°).

When implementing product detection of SSB, DSB and AM signals for other Emona Telecoms-Trainer 101 experiments, these synchronisation errors are avoided by simply "stealing" the modulator's carrier (with a patch lead) for use as the product detector's local carrier. This is obviously not possible in practice for wireless communications! As such, a method of remotely synchronising the two signals is necessary.

Until relatively recently, the most common method of achieving carrier synchronisation for SSB receivers involved inserting the transmitter's original carrier into the SSB signal after it has been generated. This signal is called a *pilot carrier* and is at a much lower level than the sidebands (typically -20dB). The low level carrier can then be acquired by the receiver and used for the product detector's local carrier.

An obvious method of acquiring the pilot is to use a band-pass filter. However, this is impractical due to the complexity of balancing the contradictory needs of filtering out the nearby sideband (or sidebands if the signal is DSB or AM) while maintaining a wide enough pass-band to cater for any carrier instability (that is, drifts in the transmitter's carrier frequency).

A better solution involves using the phase-locked loop (PLL). The PLL effectively acquires the pilot carrier by generating its own carrier signal that is synchronised to the pilot. The block diagram of a PLL is shown in Figure 1 below - the VCO is the demodulator's local oscillator.



Figure 1

A mathematical analysis of PLL operation is complicated. For our purposes here, it's better to think of its operation this way. Imagine the connection between the low-pass filter and the VCO is broken. If this is the case, the arrangement is basically the product detector shown in Figure 2 below.





Suppose the PLL's input is an unmodulated carrier and the VCO's output is a sinewave with an identical frequency. The math predicts that the multiplier's output is a DC component and a sinewave equal to the sum of the inputs' frequencies. Importantly, the magnitude of the DC component is a function of the phase difference between the two carriers. A phase difference of 0° produces the maximum DC output voltage and a phase difference of $\pm 90^{\circ}$ produces OV DC. Put another way, any phase difference between the multiplier's inputs either side of 90° produces a DC voltage.

Now, suppose that the VCO's output changes to a slightly different frequency. This has two effects on the multiplier's output. First, the frequency of the sinewave on its output changes. Second and more importantly, the resulting continuous change in phase difference between the carriers causes the magnitude of the DC component to change continuously in sympathy (so it's not really DC any more).

The low-pass filter is used to remove the multiplier's sinewave but let the "changing DC" pass. So, when its output is connected to the VCO's input, this changing DC component causes the VCO's output frequency to change as well. Inevitably, at some point the two signals will have the same frequency and have 90° of phase shift. When this happens, the multiplier's DC component is OV and the PLL achieves "lock". Once lock is achieved, changes in the VCO's output frequency automatically produce a DC voltage that corrects it and forces it to return to the frequency of the carrier.

In terms of carrier acquisition, the VCO's output is now a sinewave with an identical frequency to the carrier with a fixed 90° phase shift. Of course, for the VCO's output to be useful as a local carrier, the 90° phase shift between the modulated carrier and the local carrier must be corrected.

The experiment

For this experiment you'll use the Emona Telecoms-Trainer 101 to generate an AM signal. You'll then implement a phase-locked loop (PLL) to generate a local carrier that is synchronised to the AM signal's carrier.

It should take you about 35 minutes to complete this experiment.

Pre-requisites:

Experiment 1 (Vol. 1): Setting up an oscilloscope Experiment 2 (Vol. 1): An introduction to the Telecoms-Trainer 101 Experiment 3 (Vol. 1): Modelling equations Experiment 5 (Vol. 1): DSBSC modulation Experiment 7 (Vol. 1): DSBSC demodulation Experiment 1 (Vol. 2): AM (method 2) and product detection of AM signals (desirable)

Equipment

- Emona Telecoms-Trainer 101 (plus power-pack)
- Dual Channel 20MHz oscilloscope
- Two Emona Telecoms-Trainer 101 oscilloscope leads
- Assorted Emona Telecoms-Trainer 101 patch leads

Procedure

Part A - Generating a 100% modulated signal

To experiment with carrier acquisition, you'll need a modulated carrier signal and either an AM, DSB or SSB signal can be used for this purpose. You'll use an AM signal because it avoids having to re-insert a pilot carrier (which you would have to do for a DSB or SSB signal) and so keeps the wiring relatively simple. Part A of this experiment gets you to set up a 100% modulated AM signal using an 8.33kHz message.

- 1. Gather a set of the equipment listed on the previous page.
- 2. Locate the Tuneable Low-pass Filter module and set its *Gain* control to about the middle of its travel.
- 3. Turn the Tuneable Low-pass Filter module's *Cut-off Frequency Adjust* control fully clockwise.
- 4. Connect the set-up shown in Figure 3 below.





This set-up can be represented by the block diagram in Figure 4 below. It uses the Tuneable Low-pass Filter module to filter the 8.33kHz fundamental from an 8.33kHz digital signal.



Figure 4

5. Set up the scope per the instructions in the Appendix.

Note: Ensure that the scope's *Trigger Source* control is set to the *CH1* (or *INT*) position.

- 6. Adjust the following scope controls as indicated:
 - Timebase control to the 20µs/div position
 - Channel 2 Vertical Attenuation to the 2V/div position
- 7. Verify that the Tuneable Low-pass Filter module's output is an 8.33kHz sinewave.

Tip: Measure the signal's period and use that to determine its frequency.

- 8. Adjust the Tuneable Low-pass Filter module's *Gain* control so that its output is 4Vp-p.
- 9. Modify the set-up as shown in Figure 5 below.





This set-up can be represented by the block diagram in Figure 6 on the next page. It uses the 4Vp-p 8.33kHz sinewave as the message to amplitude modulate a 100kHz carrier. As the 8.33kHz sinewave also consists of a 2V DC offset, the modulation level is 100%.





- 10. Adjust the scope to view the Multiplier module's output as well as the message signal.
- 11. Verify that the Multiplier module's output is an AM signal by checking that its envelopes are a copy of the message.

Note 1: The modulation index of the AM signal is not crucial to this experiment. A 100% modulated AM signal is used purely for simplicity of wiring.

Note 2: If the AM signal appears over-modulated, reduce the Tuneable Low-pass Filter module's cut-off frequency until 100% modulation is achieved.



Part B - Acquiring the carrier using the phase-locked loop

Part B uses a phased-lock loop (PLL) to generate a local carrier signal that is synchronised to the AM signal's carrier. This signal can be used by a product detector to demodulate the AM signal (though that's not possible to do for this experiment because the Tuneable Low-pass Filter module is required but it is used here to create the 8.33kHz message).

- 12. Locate the VCO module and set its *Gain* control to about the middle of its travel.
- 13. Set the VCO module's Frequency Adjust control to about the middle of its travel.
- 14. Set the VCO module's Range control to the HI position.
- 15. Modify the scope's connections as shown in Figure 7 below.



Figure 7

16. Set the scope's *Timebase* control to the *1µs/div* position.

Note: The scope's Channel 2 input will not be a stable display at this stage because the two signals aren't synchronised. You can ignore this for the moment.

- 17. Measure the period of the Master Signals module's *100kHz SINE* output and note the value.
- 18. Set the scope's *Trigger Source* control to the *CH 2* position.

Note: This triggers the scope on the VCO module's output instead of the Master Signals module's *100kHz SINE* output.

19. Adjust the VCO module's *Frequency Adjust* control to sinewave at exactly the same frequency as the Master Signals module's *100kHz SINE* output.

Tip: Set its period to exactly the same value that you measured at Step 17.

20. Switch the scope's *Trigger Source* control between the *CH 1* and *CH 2* positions and compare the two signals.

Question 1

Why can't the scope show the two signals as stable displays at the same time?

Despite the students' best efforts, the two signals are not exactly the same

frequency.



Ask the instructor to check your work before continuing.

- 21. Set the scope's *Trigger Source* control to the *CH 1* position.
- 22. Set the scope's *Timebase* control to the *2µs/div* position.
- 23. Modify the set-up as shown in Figure 8 below.





The additions to this set-up can be represented by the block diagram in Figure 9 on the next page. The arrangement implements a PLL that generates a local carrier synchronised to the AM signal's carrier.





The entire set-up can be represented by the block diagram in Figure 10 below.





24. Compare the two signals.

Note 1: You should now have "lock". That is, the PLL's output is exactly the same frequency as the Master Signals module's *100kHz SINE* output.

Note 2: If "lock" hasn't been achieved, tweak the VCO module's *Frequency Adjust* control left and right.

25. To prove that your PLL has lock, disconnect and reconnect the patch lead to the VCO module's input a couple of times and observe the effect on its output.

Question 2

Once lock is achieved, what's the phase relationship between the AM signal's carrier and the PLL's output?

90°.

Question 3

What problem would this particular amount of phase shift cause if the signal is used for product demodulation of AM and DSBSC?

The recovered message would be completely cancelled (because the two sidebands would

have a total phase error of 180°).

Question 4

How can this problem be resolved?

Phase shift the PLL's output until the message signal has maximum amplitude - at which

point the two carriers are in phase.



Ask the instructor to check your work before finishing.



Emona-101 Experimenter SAMPLE Lab Manual Volumes 1 & 2 -Experiments in Modern Analog and Digital Telecommunications. Author: Barry Duncan

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