ELEN90057 Communication Systems Workshop 3

This workshop is worth 4% of the overall subject assessment and should be completed in pairs. Seeking or providing detailed assistance from/to people other than your workshop partner is collusion - see http://academichonesty.unimelb.edu.au/plagiarism.html

One joint report per pair, containing your worked solutions and experimental results, should be submitted. All reports should come with a Departmental cover-sheet attached with your name, Student ID and signature, and be submitted to your demonstrator's designated pigeon hole in level 4, EE (south stairwell). Please indicate your workshop time and day.

Note that you may be called in at random to verbally explain your results and solutions.

Note that for all TIMS workshops, only authorised results can be used in your report. They must be collected during your designated lab session, signed by your demonstrator and attached with your report as appendix.

It is highly recommended that before presenting results for any particular section that you present a configuration of the circuit *being studied, as well as any pertinent* equipment settings *that are used to obtain your results.*

AIMS:

The main objective of this workshop is to model and observe experimentally some of the properties of both single sideband modulation (SSB) and frequency modulated (FM) signals.

EQUIPMENT:

One Standard Emona TIMS unit with the following additional modules:

- Adder ×2, Audio Oscillator, Multiplier ×2, VCO ×2, Tuneable LPF, Quadrature Utilities, Twin pulse generator, Noise Generator and Utilities.

Assorted cables and leads - standard patch cables and BNC-BNC cables

Oscilloscope and Spectrum Analyser - Tektronix TDS1012B digital scope with FFT function

1. SSB Modulation (8 marks)

The most straightforward way of converting a DSBSC signal into an SSB signal is to use a sideband filter. This is simple in conception, yet requires a far-from-simple filter for its execution. A second method of sideband removal is to use a 90 degree phase-shifter, i.e. a Hilbert transformer. TIMS does not have a true Hilbert transformer module, but does have a QUADRATURE PHASE SPLITTER. This takes an input signal $x(t)$ and yields two output signals that are out phase with each other by 90 degrees, over a wide range of frequencies (however, their phase relationships with the input signal may be frequency-dependent).

$x(t)$
$\hat{x}(t)$

These two output signals are then separately multiplied onto two carriers at frequency f_c that are also out of phase with each other by 90 degrees, i.e. a cosine and a sine wave, to yield two DSBSC-like signals.

must match.

$\frac{e^{j2\pi f_c t} + e^{-j2\pi f_c t}}{2}$

two output signals:

$m(t) = x(t) = \frac{1}{2}\cos(2\pi f_m t) + \cos(2\pi 2 f_m t)$

$\frac{e^{j2\pi f_c t} - e^{-j2\pi f_c t}}{2j}$

$\hat{m}(t) = \frac{1}{2}\sin(2\pi f_m t) + \sin(2\pi 2 f_m t)$

LSSB

two DSBSC like signals: $s(t) = \frac{A_c}{2}[m(t)\cos(2\pi f_c t) \pm \hat{m}(t)\sin(2\pi f_c t)]$

USSB

$$s(t) = \frac{A_c}{4}[m(t)e^{j2\pi f_c t} + m(t)e^{-j2\pi f_c t}] \pm \frac{A_c}{4j}[\hat{m}(t)e^{j2\pi f_c t} - \hat{m}(t)e^{-j2\pi f_c t}]$$

$$s(t) = \frac{A_c}{4}\left(\frac{e^{j2\pi f_m t}+e^{-j2\pi f_m t}}{4} + \frac{e^{j2\pi 2 f_m t}+e^{-j2\pi 2 f_m t}}{2}\right)(e^{j2\pi f_c t}+e^{-j2\pi f_c t}) \quad \checkmark$$

$$\pm \frac{A_c}{4j}\left(\frac{e^{j2\pi f_m t}-e^{-j2\pi f_m t}}{4j} + \frac{e^{j2\pi 2 f_m t}-e^{-j2\pi 2 f_m t}}{2j}\right)(e^{j2\pi f_c t}-e^{-j2\pi f_c t})$$

$$S(f) = \frac{A_c}{4}\Big[\tfrac{1}{4}\delta(f-(f_m+f_c)) + \underline{\tfrac{1}{4}\delta(f-(f_c-f_m))} + \tfrac{1}{2}\delta(f-(2f_m+f_c))$$

$$+ \underline{\tfrac{1}{2}\delta(f-(f_c-2f_m))}$$

$$+ \underline{\tfrac{1}{4}\delta(f-(f_m-f_c))} + \tfrac{1}{4}\delta(f+(f_c+f_m)) + \underline{\tfrac{1}{2}\delta(f-(2f_m-f_c))}$$

$$+ \tfrac{1}{2}\delta(f+(f_c+2f_m))\Big]$$

$$\pm \frac{A_c}{4}\Big[-\tfrac{1}{4}\delta(f-(f_m+f_c)) + \underline{\tfrac{1}{4}\delta(f-(f_c-f_m))} - \tfrac{1}{2}\delta(f-(2f_m+f_c))$$

$$+ \underline{\tfrac{1}{2}\delta(f-(f_c-2f_m))}$$

$$+ \underline{\tfrac{1}{4}\delta(f-(f_m-f_c))} - \tfrac{1}{4}\delta(f+(f_c+f_m)) + \underline{\tfrac{1}{2}\delta(f-(2f_m-f_c))}$$

$$- \tfrac{1}{2}\delta(f+(f_c+2f_m))\Big]$$

if they can cancel out, we only have, (f_c-f_m) (f_c-2f_m), (f_m-f_c) $(2f_m-f_c)$ — 4 spikes. LSSB

2 (a)

if $\oplus$

$v_c(t) = \cos(2\pi f_c t)$
$= \dfrac{e^{j2\pi f_c t}+e^{-j2\pi f_c t}}{2}$.

spikes: $(2f_c-f_m)$ $(2f_c-2f_m)$ (f_m) $(2f_m)$ (f_m-2f_c) $(2f_m-2f_c)$ $(-2f_m)$ $(-f_m)$

8 spikes.

Take the input to the phase splitter to be a two-tone sinusoid signal $x(t)$ with fundamental frequency f_m, where $2f_m \ll f_c$,

V=2 Vpp=2   V=2 Vpp=4.

$A_c = 2.5$ V
$$x(t) = \frac{1}{2}\cos(2\pi \cdot f_m t) + \cos(2\pi \cdot 2 f_m t)$$

Denote the message signal $m(t)$ to be the phase-splitter output that is to be multiplied onto the cosine-carrier,

a) Assuming $m(t)$ is approximately equal to $x(t)$, find time-domain expressions for the two DSBSC-like signals and show that an SSB signal $s(t)$, either upper sideband only or lower sideband only, can be produced by adding these two DSBSC-like signals together. Find the Fourier transform $S(f)$ of the SSB signal.

b) Use appropriate TIMS modules to generate and save/photograph a plot of the time- domain representation of the SSB signal with $f_c = 100$ kHz and $f_m = 2$ kHz. Notice that the amplitude of two DSBS signals must be carefully adjusted so that when added together, the two upper sidebands (say) cancel out. A fine trim of one or other of the ADDER gain controls will probably be necessary.

Demonstrator's Signature

c) Using the FFT functionality of the oscilloscope to observe, and save/photograph a plot of the spectra (frequency domain representation) of the above SSB signal (noting magnitudes and frequencies). Comment on any discrepancies between your experimental results and the analytical result obtained in *a)*.

Demonstrator's Signature

2. SSB Demodulation (9 marks) $\hat{V}_c(t) = \cos(2\pi f_c t)$

At the receiver, an SSB signal can be demodulated with a coherent demodulator similar to the one we have used for DSBSC demodulation in Workshop 2. To down-convert the signal back into baseband, a "stolen carrier" $\tilde{v}_c(t)$ can be simply multiplied with the SSB signal $s(t)$. The resulting signal will then go through a lowpass filter, where all the high frequency terms are rejected.

a) Find time- and frequency-domain expressions for the received signal after the SSB signal has mixed (been multiplied) with the "stolen carrier". In the frequency domain, how many spikes are we expecting now? Observe and save/photograph a plot of the signal spectra after "down-converting" to verify your result.

8 spikes

Demonstrator's Signature

b) What range of values can we choose for the 3-dB bandwidth of the lowpass filter?

$2f_m \sim 2f_c - 2f_m$

4k ~ 196k

can choose 60k LPF.

c) Choose appropriate TIMS modules to build up an SSB demodulator and recover the message signal. Use both channels of the oscilloscope and the FFT functionality of the oscilloscope to save/photograph, and compare the output waveform with the input message signal in both time- and frequency domains. Note down all relevant features and parameters of your plots.

Demonstrator's Signature

3. VCO Characteristics (13 marks)

This rest of this workshop is about frequency modulation (FM). A very simple and direct method of generating an FM signal is to use a voltage controlled oscillator (VCO), with an instantaneous frequency that varies proportionally with the applied voltage. Integrated circuit based VCO's can be made to have a very linear frequency-voltage characteristic, but only over high a certain frequency range. Hence, before generating an FM waveform, it is important to measure and verify the linear region of the VCO to make sure your system is not overloaded, i.e., operating in the non-linear region.

a) In the absence of any input voltage, adjust and verify the output frequency of the VCO, f_0 is at 100 kHz (in High mode). It is also recommended that the GAIN control (sensitivity) is set at approximately centred position. This to make sure the TIMS analogue signal reference level of 4 volt peak-to-peak is well within the VCO's linear region. Once you set the GAIN and f_0 of the VCO, it is strongly recommended that they remain constant throughout the experiment; otherwise a recalibration or recalculation will be required. (Hint: when operating in the nonlinear region, the OVERLOAD LED is lit to indicate that the input voltage has exceeded the oscillator's internal limits.)

Demonstrator's Signature

①

b) With the help of the VARIABLE DC unit, measure and sketch a frequency vs. voltage graph for your VCO. Make sure to take enough data points on both side of the origin so that both linear and nonlinear regions are covered. Determine the linear region of the VCO, and estimate the slope k_f of the linear region. Change the VCO to LO mode and repeat the task.

②

Another way of computing the frequency sensitivity k_f is to use the "Bessel zero" method. In this method, we employ single tone analysis where a pure sinusoid at frequency W and a very small amplitude A_m is applied to the VCO input.

1.9V 2K

$$m(t) = A_m \cos(2\pi W t)$$

The corresponding output, i.e., FM signal, can be expanded into a summation of sinusoids

$$s_{FM}(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta)\cos(2\pi(f_c + nW)t)$$

where the coefficient $J_n(\beta)$ is the n-th order Bessel function of the first kind.

13

$2.4 \quad \beta = \frac{K_f A_m}{W} \rightarrow 2k$

The spectrum of an FM signal hence consists of a centre spike at the carrier-frequency plus an *infinite* number of spikes, where the frequency spacing between two adjacent spikes equals to W. The relative amplitude of these spikes varies in accordance with the modulation index $\beta = k_f A_m / W$ 12.54.

, and hence in turn depends on the amplitude of the message signal.

13.2 kHz.

Demonstrator's Signature

c) Using the attached Bessel chart at the end of the workshop sheet, calculate and sketch the expected spectrum obtained when $\beta = 2$, i.e., determine the Bessel function values of the first 5 spectral components on either side of the carrier.

$\beta = 2 \quad K_f = 13.2 \quad A_m = 1.9$

d) Notice that when Am is gradually increasing, the intensity of the carrier spike, $J_0(\beta)$, gradually decreases and drops to zero when $\beta = \cancel{2.4}$ 2.8 (see Bessel Function graph at the end of the workshop sheet) and certain other higher values of β. Compute k_f using this technique and comment on any discrepancies if your result does not match up with the result obtained in *b)*

4. FM Spectrum (12 marks)

a) Using the 2 kHz single-tone MESSAGE SIGNAL as the message $m(t)$, choose appropriate TIMS modules to implement an FM modulator and save/photograph a plot of the time-domain representation of the output FM signal.

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$\beta = \frac{K_f A_m}{W}$

b) Using the FFT functionality of the oscilloscope to observe, and save/photograph at least 3 different FM spectra with different β values. (The frequency sensitivity k_f, however, must remain constant.) Indicate the FM bandwidth according to Carlson's rule on your plots. Use the "Bessel zero" method outlined in Part 3 to show how the relative amplitudes among spikes can be used to determine β (without any measurement of A_m or W). The β values to be used can be arbitrary. However, some examples that may be of interest are:

varying Am

(i) $\beta = 1.4$ (why?) $A_m = 0.22$.

(ii) $\beta = 2$ (as obtained in Part 3) $A_m = 0.3$

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

1.4, 2,

(3) $A_m = 0.5$, $\beta = 3.3$

For each of these 3 cases, measure A_m and W to confirm your choice of β. Measure relative spectral components of the FM signal, and compare them with the theoretical values.

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5. Demodulating FM by ZX Counting (8 marks)

One way to demodulate FM signals uses the fact that provided the carrier frequency is large, the message resides in the zero crossings of the modulated signal. As shown in Fig. 1, the number of zero-crossings within a small time interval is indeed proportional to the instantaneous frequency of the FM signal, and hence can be related back to the amplitude of the original message signal. As this does not use a local oscillator, this method is thus a non-coherent or asynchronous demodulation method.

In general, it is not easy to detect zeros inside a FM signal. In TIMS, the zero-crossing pulse train is generated indirectly, i.e., in 2 stages. First, we use a COMPARATOR module to boost up the amplitude of the input FM signal by a large factor and then clip the amplitude of the amplified signal with help of the REF signal. When the REF is connected to the GROUND, the COMPARATOR output provides a standard TTL level (0 ~ 5V) waveform corresponding to the positive part of the input FM signal.

CH2

Am=0.3 β=2
for this

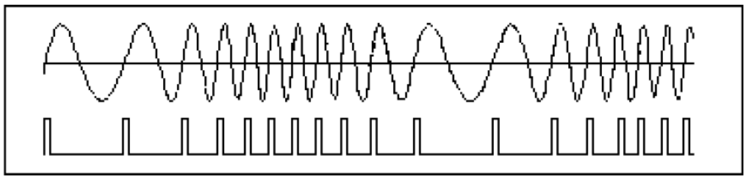


Figure 1: an FM waveform and its corresponding zero-crossing pulse train

In the second stage, the TTL signal is then used to clock a TWIN PULSE GENERATOR module, which produces a train of constant width output pulses (one for each positive going edge of the TTL signal), indicating the zero-crossings in the original FM waveform. This pulse train will then pass through a LPF which essentially serves as an integrator to recover the message.

a) Initially use a 100 kHz sine wave as the input to the zero-crossing detector. Use both channels of the oscilloscope to observe and compare the pulse train at the output of the COMPARATOR and the output from the TWIN PULSE GENERATOR. For demodulating purposes, what is the optimal choice of the pulse width?

b) Replace the 100 kHz sine wave with the FM signal you have generated in part 4 with an
2 arbitrary β of your choice. Choose appropriate TIMS modules to implement a ZX-based FM demodulator to recover the message signal. Use both channels of the oscilloscope and the FFT functionality to save/photograph, and compare the output waveform with the input message signal in both time- and frequency domains. What is your choice of the pulse width? Justify your argument by providing relevant plots and discussions.

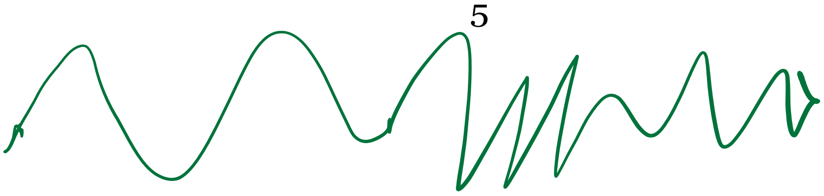
Demonstrator's Signature

CH1 M
CH2 · LPF demod

6. Demodulating FM using PLL (10 marks)

In this section we will use a phase-locked loop (PLL) to demodulate the FM signal. Like a synchronous demodulator for DSBSC or SSB, the incoming signal is multiplied by a local oscillator signal and then passed through a LPF. However, an important difference is that the local oscillator here is voltage-controlled by the demodulated output, which is fed back to it to adjust its instantaneous frequency.

a) Configure the second VCO that will be used as the local carrier in the PLL. Use both



channels of the oscilloscope and the FFT functionality of the oscilloscope to show that both VCO's have nearly matched free run frequencies. The frequency sensitivity k_v, of the second VCO, however, is intentionally set to be different from k_f.

Demonstrator's Signature

b) A FM signal have generated (see part 4 for more details) with an arbitrary β of your choice. In theory, what range of values can we choose for the PPL sensitivity k_v?

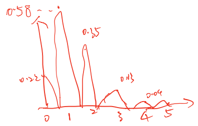
c) Choose appropriate TIMS modules to implement a PLL-based FM demodulator to recover the message signal. (Note that the VCO GAIN was used to control the frequency sensitivity k_v, and hence is different from the loop gain K_a) Use both channels of the oscilloscope and the FFT functionality to save/photograph, and compare the output waveform with the input message signal in both time- and frequency-domains. Note down all relevant features and parameters of your plots.

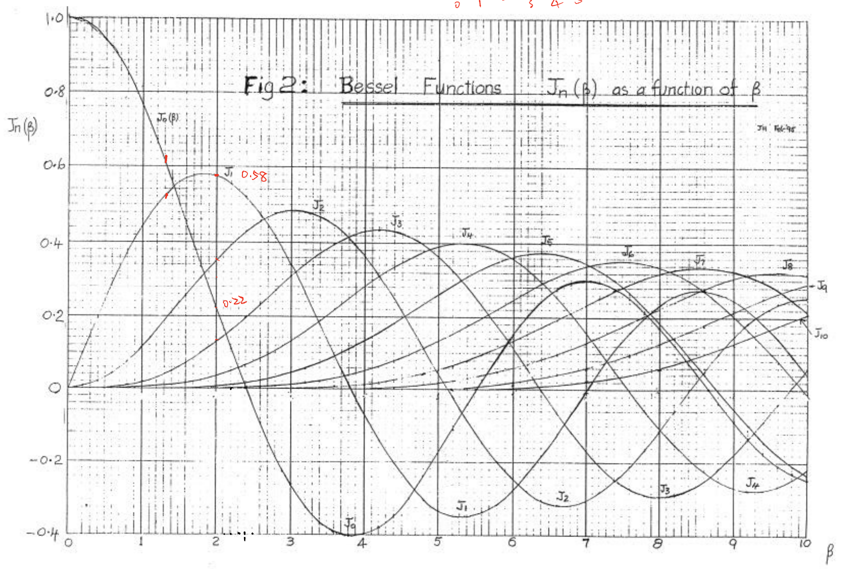
Demonstrator's Signature

cH 1 — demod
CH2 — M

d) Modify the frequency sensitivity k_v of the second VCO in PLL and comment on the effect on the recovered signal in both the time- and frequency-domains. What is the minimum experimental value of k_v for undistorted outputs? What do you think might have determined this minimum?

$J_0 = 0.22$
$J_1 = 0.58$
$J_2 = 0.35$
$J_3 = 0.13$
$J_4 = 0.04$
$J_5 = 0.01$





$J_n(1.0)$	$J_n(2.0)$	$J_n(5.0)$
0.77	0.22	−0.18
0.44	0.58	−0.33
0.11	0.35	0.05
0.02	0.13	0.36
	0.03	0.39
		0.26
		0.13
		0.05

$$W = f_m = \frac{k_f A_m}{\beta}$$