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Linear and Angular Modulator Using an I/Q Mod Topology

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The thesis project was conducted to design and implement I/Q modulator and understand the flexibility of I/Q modulator to perform various kind of modulation with ease. It was also a prime focus of this thesis to identify effect of noise in carrier on the modulated signal and other possible sources of error.

Total of four stages, 90° phase shifter, mixers, gain and summing stages, were designed and their circuit board were constructed. All the components except LT5560 were available in the University laboratory. The modulator simulation was also performed on Multisim software and the results were studied.

After spectrum analysis of LT5560 at lower frequencies below 1 MHz, it was found that mixed signals gets highly attenuated and works pretty well above 5 MHz LO frequencies, but with leakage of LO frequency on output, hence the hardware implementation remained incomplete. Further analysis and fault finding can be done to make the system working and smaller in size.

Keywords

 $\ensuremath{\mathsf{I/Q}}$  modulator, telecommunication, wireless, quadrature modulator, modulation, phase shift



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#### 1 Introduction

Wireless systems have become a revolutionizing development in the electronic systems in today's generation. In my personal opinion, it can be pre-stated that in near future wired communication systems will be confined to a definite area of electronics. Though it is very handy to have such systems, the design process is quite complicated and cumbersome. Among different steps in transmitting data from one place other, there exists a process called modulation where data is manipulated so that it is suitable to transmit to receiver.

Tomasi explains [1, p. 100] that "*The process of impressing low-frequency information signals onto a high-frequency carrier signal is called modulation.*" The low-frequency signals are commonly known as baseband signal or modulating signal, which is produced by the data source, and the high-frequency information carrying signal is called pass-band signal or modulated signal, which is in suitable form to get transmitted [2].

#### 1.1 Advantages of Modulation

Baseband signals from different users can be translated to different frequency bands so that multiple users can be fitted within a band of electromagnetic spectrum. Also when wireless transmissions through long distance is taken in account, low frequency waves are prone to attenuation. As a result, receiving side will not be able to pick the low frequency baseband signals. Besides all, modulation reduces size of antenna, this is because antenna height is inversely proportional to the radiated signal frequency. Higher the frequency, smaller is the size of antenna. [3]

#### 1.2 Linear and Angle Modulation

In general practice sinusoids are used as the carrier signal and which can be represented by following equation 1

$$A\cos(\omega_C t + \phi) \tag{1}$$

In the above equation the parameters which can be changed are amplitude, frequency and phase. If the modulation process impress the data on amplitude then it is called



amplitude modulation. Amplitude modulation is also referred to as Linear Modulation. Linear Modulation has a property, that the input and output relation comply with principle of homogeneity and superposition [3]. And if the data is impressed on either frequency or phase is called angle modulation as suggested by the name itself [1].

Varieties of modulation schemes are available though they are performed by varying the three parameters or combination of these parameters.

## 2 Fundamental Analog and Digital Modulations

Various types of modulation schemes are available that can be implemented according to the requirement of bandwidth, efficiency or cost. In this section AM, ASK and BPSK will be dealt which will be implemented through I/Q modulator.

#### 2.1 Amplitude Modulation

It is relatively low quality and in-expensive type of analog modulation in which the amplitude of the carrier signal is varied proportionally to the instantaneous amplitude of the baseband signal [1].

The simplest way to achieve amplitude modulation is to pass the baseband signal and carrier through non-linear devices called mixers. For a frequency of the carrier signal  $f_c$  and data signal frequency of  $f_m$ , the resulting frequencies are at  $f_c$ - $f_m$  and  $f_c$ + $f_m$ . Any frequency between  $f_c$  and  $f_c$ - $f_m$  are called lower-side band (LSB) and between  $f_c$  and  $f_c$ + $f_m$  are called upper side-band (USB). An intuitive understanding on this concept in frequency domain is shown in Figure 1.



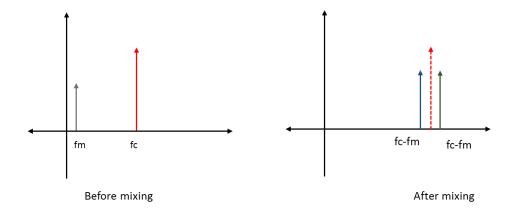


Figure 1 : illustration of modulation at frequency domain

Considering the carrier and data are cosine waves with previously mentioned frequencies, amplitude modulation can be expressed as equation 2.

$$\cos(2\pi f_c) \cdot \cos(2\pi f_m) = \frac{1}{2}\cos[2\pi (f_c + f_m)] + \frac{1}{2}\cos[2\pi (f_c - f_m)]$$
(2)

It can be seen that the message frequency has been translated to two different greater frequencies  $f_c+f_m$  and  $f_c-f_m$ . Also the signals get attenuated by half [4].

The use of quadrature modulator (QM) has made the hardware implementation easier for different types of modulation. Various types of Amplitude modulation can be performed in QM. One common use of QM in Amplitude modulation is to remove one of the sidebands from the modulated wave and eliminate the half attenuation seen in equation 3. To achieve such modulation, the input of QM is driven by message signals that are in quadrature in a way to obtain require side band. [4]. For instance,

$$[\cos(2\pi f_c) \cdot \cos(2\pi f_m)] + [\sin(2\pi f_c) \cdot \sin(2\pi f_m)]$$
  
= $[\frac{1}{2}\cos[2\pi(f_c + f_m)] + \frac{1}{2}\cos[2\pi(f_c - f_m)]]$   
+ $[\frac{1}{2}\cos[2\pi(f_c - f_m)] - \frac{1}{2}\cos[2\pi(f_c + f_m)]]$   
= $\cos[2\pi(f_c - f_m)]$  (3)

In equation (3),  $\cos (2\pi f_c)$  and  $\sin (2\pi f_c)$  are quadrature carriers while  $\cos (2\pi f_m)$  and  $\sin (2\pi f_m)$  are quadrature data signal.



#### 2.2 Amplitude Shift Keying (ASK)

An analogous modulation technique for Amplitude modulation in digital domain is ASK. In ASK, carrier is amplitude modulated with respect to the amplitude of modulating binary signal. For logic 1 and 0, carrier amplitude is shifted between different levels. A common class of Amplitude Shift Keying is On-Off Keying (OOK), where carrier signal is present at logic 1 while absent at logic 0. Amplitude modulated signals can be represented by equation (4). [1]

$$v_{am^{(t)}} = v_m(t)[A\cos(\omega_C t)]$$

Where,

 $v_{am^{(t)}} = digital \ amplitude - modulated \ wave$   $A = unmodulated \ Carrier \ amplitude \ (volts)$   $v_m(t) = modulating \ binary \ signal \ (volts)$  $\omega_c = carrier \ radian \ frequency \ (radians \ per \ second)$ 

Modulating binary signal ( $v_m(t)$ ) have +1V representing logic 1 and 0V representing logic 0.So, for logic 1, above equation becomes  $v_{am^{(t)}} = A \cos(\omega_C t)$  and for logic 0  $v_{am^{(t)}} = 0$ . This clearly explains that for logic 1 there is presence of the carrier signal while for logic 0 the carrier signal is absent, which suggest the name of the modulation as On-Off keying which is represented by Figure 2.

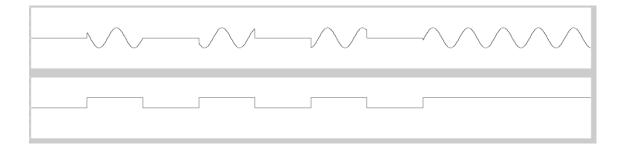


Figure 2 : BPSK constellation diagram



(4)

#### 2.3 Binary Phase Shift Keying (BPSK)

In BPSK logic 1 is represented by a value of carrier's phase and the second phase represents the logic 0, while the amplitude of the carrier remains unchanged. Generally the phases are 180° apart [3]. This can be illustrated by constellation diagram Figure 3 below

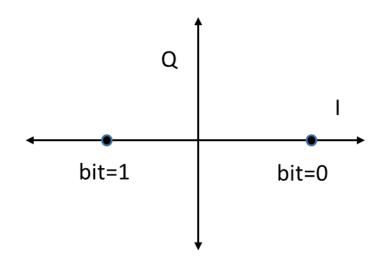


Figure 3 : BPSK constellation diagram

If we have sinusoid of amplitude A, the power is

$$P_s = \frac{1}{2}A^2 \tag{5}$$

Then,

$$A = \sqrt{2P_s} \tag{6}$$

So the two states of BPSK signal are: At 0° phase:

$$W_{BPSK}(t) = \sqrt{2p_s} \cos(\omega_C t) \tag{7}$$

At 180° phase:

$$V_{BPSK}(t) = \sqrt{2p_s}\cos(\omega_C t + \pi)$$



$$V_{BPSK}(t) = -\sqrt{2p_s}\cos(\omega_C t)$$
(8)

### 3 Signal Adder

Summing amplifier can be constructed from inverting amplifier to give negative of sum of voltages of the input signals as shown in Figure 4. [5]

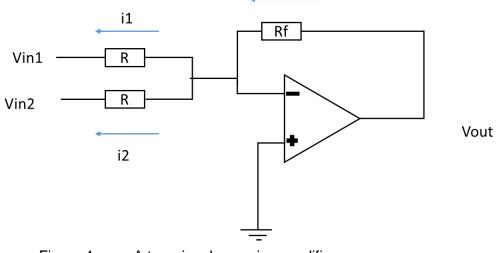


Figure 4 : A two signal summing amplifier

Since the non-inverting node is grounded, the voltage at the inverting input is 0. Taking to consideration, the general characteristics of an Op-Amp, that the input impedance is so high that no current flows into the negative terminal [5], we can deduce following equation at inverting node using KCL law:

$$I = i_1 + i_2 \tag{9}$$

We can solve equation (9) using fundamental circuit equations to estimate the output of summing amplifier.

$$-\frac{v_{out}}{R_f} = \frac{v_{in1}}{R} + \frac{v_{in2}}{R}$$
$$v_{out} = -\frac{R_f}{R}(v_{in1} + v_{in2})$$
(10)



From Equation (10), it can be concluded that input signals are added algebraically and gain of the output signal can be adjusted by use of suitable  $R_f$  and R values. Also the negative sign indicates the output is inverted.

#### 3.1 Multisim Simulation of Signal Adder

A Multisim simulation of summing amplifier was performed using LM324N to have earlier analysis of the signal behaviour. Two sinusoids of same phase (0°) and amplitude (500 mVp) were passed through the input and the output was compared with one of the input. The setup is shown in Figure 5.

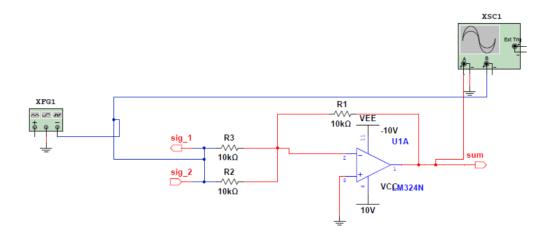
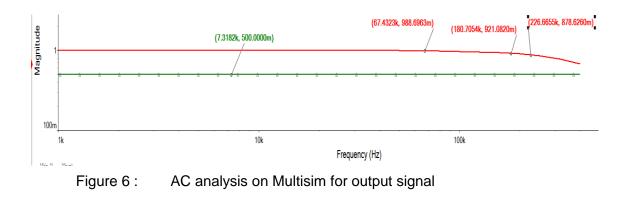


Figure 5 : Multisim setup for signal adder

Magnitude analysis was conducted using built in feature of Multisim, whose result is presented in Figure 6. The green line is the reference input signal, while red line represents the output signal of the summing amplifier.





Theoretically, the sum of the signals must be 1Vp for 500 mVp input, whatever the frequency be, but observing the amplitude response at different frequencies, results tend to approach theoretical estimation at lower frequencies. Frequencies above 200 KHz seems to undergo large attenuation, indicating LM324N not a good choice for higher frequencies.

Summing amplifier was also implemented in hardware. The PADS layout and Schematic for it is attached at Appendix 3.

#### 4 Signal Mixer

Mixers are non-linear devices which performs multiplication of two frequencies that results in amplitude multiplication and shift of frequency. Usually, one of the input signals is low frequency signal and the other one is very high frequency signal, generally from Local oscillator (LO). Frequency shifting can be realized if the LO signal is constant in amplitude and frequency and the other input is quite low compared to LO so that only LO has ability to affect the trans-conductance of the mixer. [6] The multiplication of the signals in mixers can be modelled as a switch, driven by very high frequency, which is in series with the input signal. [6] Figure 7 shows an ideal mixer diagram.

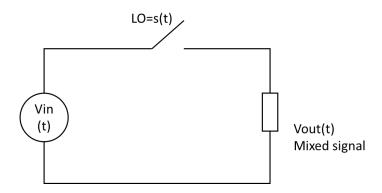


Figure 7 : Ideal mixer model

The switch is generally a nonlinear device like diode or transistor driven by LO. The spectrum of an ideal mixer presented in [6, p. 316] is re-printed in Figure 8



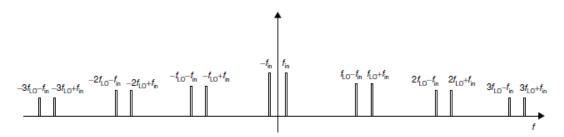


Figure 8 : Spectrum of an ideal mixer

As observed in Figure 8, of an input frequency and LO frequency, an ideal mixer outputs two frequencies at region equal to  $n^*LO \pm f_{in}$ , where n is the harmonics number of LO. Desired frequency thus can be obtained by use of filters.

In real world mixer, the effect of input signal to the trans-conductance property of the mixer cannot be neglected even for small input frequency, resulting in all possible harmonic components, thus in real world output frequencies are  $n^*LO \pm m^*f_{in}$ , where n and m are integers. [7]

A quick note on ports name of a mixer seems beneficial at this instance of this report. Generally mixers are used to translate lower frequencies to upper frequencies (up conversion) and vice-versa (down-conversion). If it is used for up conversion, then the input signal is normally named IF (intermediate frequency) signal and the output is named RF (Radio Frequency) signal. And for the down conversion, opposite is the case. [8] The modulation process is up-conversion process and here onwards naming convention of up-converting mixer shall be used.

#### 4.1 Gilbert Cell Mixer

Gilbert Cell Mixer is an active mixer. Rather than traditional way of implementing diodes, Gilbert Cell utilizes the nonlinear property of transistor to perform mixing action. An advantage of active mixer over passive mixer is that it amplifies the output signal. [8] A Gilbert Cell is shown in Figure 9



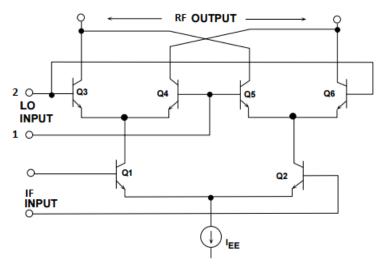


Figure 9 : Gilbert Cell reprinted form [8]

An intuitive understanding on working principle of Gilbert cell presented in Figure 9 can be done by driving LO inputs one by one following certain conditions. If LO-1 input connected to source that drives transistors Q4 and Q5, and LO-2 does not enable Q3 and Q6, Q1 and gets connected to corresponding two RF outputs. And if LO-2 enables transistors Q4 and Q5 then we get same RF outputs but ports are interchanged with respect to previous former case. [8]

#### 4.2 LT5560 Mixer IC

LT5560 is a Low Power double-balanced Gilbert Cell mixer manufactured by Linear Technology. Linear Technology claims that it can perform up or down conversion with 2.4dB typical gain from frequency range of 10 KHz to 4 GHz while the LO source input of -2 dBm can be drive the mixer. [9] After some research at different electronic component suppliers in Helsinki, this IC was found most suitable and chose for the thesis project.

Even though manufacturer claims the IC to work on wide range of frequency applications, there are no any details provided in datasheet for up-conversion of frequencies with LO range below 1 MHz. A dilemma occurred on how to bias and make the IC work on frequency of 200 KHz. In page 25, figure number 22, the datasheet includes schematic of mixer down converting application to 450 KHz form 200.45 MHz LO input and 200 MHz RF input. So modifying the schematic to work under the desired frequency seemed



proper solution at the time and a board was built based on the schematic shown in Figure 10.

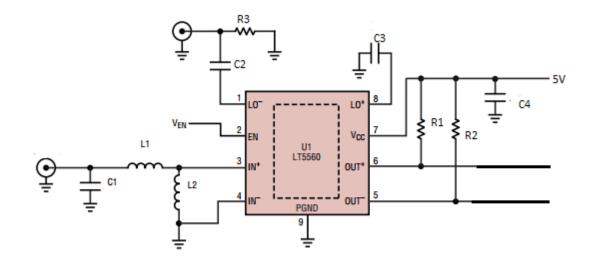


Figure 10 : LT5560 Bias Circuit remodified version of figure 22 of [9]

Resistors R1 and R2 are pull-up resistors connecting the differential output of the IC to the 5V source. R3 is the DC current path to ground from LO<sup>-</sup> port. Capacitor C1 and inductor L1 forms the High-pass filter for input signal while inductor L2 is used as RF choke and DC current path to ground. [9]

At the position of components except R1 and R2 (200 ohm resistors), connectors were placed so that it would facilitate experimenting the board with different through hole components. A reference table for the connectors to the components in the schematic is presented in Table 1. The layout and schematic of the developed board can be found in Appendix 4.

Because of the datasheet not mentioning about impedance of the input port for lower frequencies below 40 MHz, L2 was chosen 40 mH RFC coil and signal input was given through IN<sup>+</sup> directly. C2 and C3 were chosen 10 pF and C4 1nF. Mixer showed good mixing response for LO frequencies above 5 MHz. Spectrum analysis of the mixer at 20 MHz carrier and 500 KHz IF is presented at page 2 of Appendix 4.



Components in Bias Circuit	Corresponding connectors in PCB-layout
C1	J3
L1	J4
L2	J1
C2	J7
C3	J6
C4	J2
OUT <sup>+</sup> /OUT <sup>-</sup>	J10

#### Table 1: Components tally table for LT5560 board

The spectrum of output signal had highest power at 19.950 MHz, 200 MHz and 20.050 MHz. Clearly, the 19.950 MHz and 20.050 MHz signals are the mixed signals. 200 MHz signal is the carrier signal which in ideal case should not be present on the output. No device can be ideal, but since the magnitude of the carrier signal is bigger than the mixed signals, one can conclude that there is leakage of carrier signal to the output port.

Other smaller peaks on the graph are harmonic signals. The harmonic signals on the graph are at frequencies 19.800 MHz, 19.850 MHz, 19.900 MHz, 20.100 MHz, 20.150 MHz, and 20.200 MHz from left to right. As discussed in section Signal Mixer, a real mixer would produce results at all the harmonics of the input signal, in this particular case harmonics are at 200 MHz  $\pm$  m\*500KHz, where m =2,3 and 4.



#### 5 Phase Shifter

If time shift is introduced in a signal in time domain, phase shift is introduced in frequency domain. [10] Let a time domain signal x (t-t<sub>o</sub>) shifted by t<sub>o</sub> be represented in frequency domain as  $X(\omega)$ .

$$F(x(t-t_o)) = \int_{-\infty}^{+\infty} x(t-t_0) \cdot e^{-j\omega t} dt$$
(11)

Let's suppose  $t-t_o = a$ , then  $t = a + t_o$ , differentiating by a, we get dt = da, so the equation (11) becomes

$$F(x(a)) = \int_{-\infty}^{+\infty} x(a) \cdot e^{-j\omega(a+t_0)} da$$
(12)

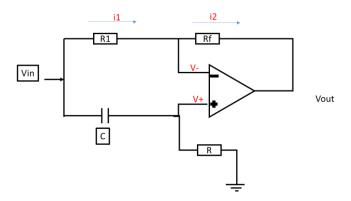
$$F(x(a)) = \int_{-\infty}^{+\infty} x(a) \cdot e^{-j\omega a} \, da \cdot e^{-j\omega t_0} \tag{13}$$

$$F(x(a)) = e^{-j\omega t_0} X(\omega)$$
(14)

Equation (14) suggests that for time shift of  $t_o$  the phase shifts by  $e^{-j\omega to}$ . [10]

#### 5.1 All-Pass Filter

All-pass filter have the property of keeping the amplitude constant but change the phase of a signal. [11] A first order all-pass filter is shown in Figure 11.





#### Figure 11 : First-order all-pass filter [11]

Considering again the general assumptions for the op-amp, V+ = V-. Using KCL law at V-, following equation is obtained

$$i1 = i2$$
 (15)

$$\frac{Vin - V}{R1} = \frac{V - Vout}{Rf}$$
(16)

Putting same value for R1 and Rf,

$$Vout = 2 * V_- Vin \tag{17}$$

V- can be calculated using the voltage divider law at V+ port as follows

$$V_{-} = \frac{R}{\frac{1}{SC} + R} * Vin$$
(18)

$$V_{-} = \frac{SCR}{1 + SCR} * Vin \tag{19}$$

From equation (17) and (19),

$$Vout = 2 * \left(\frac{SCR}{1 + SCR} * Vin\right) - Vin$$
(20)

$$\frac{\text{Vout}}{\text{Vin}} = \frac{2\text{SCR} - 1 - \text{SCR}}{j\omega RC + 1}$$
(21)

$$\frac{\text{Vout}}{\text{Vin}} = \frac{\text{SCR} - 1}{\text{SCR} + 1}$$
(22)

Dividing numerator and denominator on left hand side by CR

$$\frac{\text{Vout}}{\text{Vin}} = \frac{S - \frac{1}{CR}}{S + \frac{1}{CR}}$$
(23)

Since 1/RC sets a frequency, so it is meaningful to represent 1/RC as  $\omega_{o.}$ 

$$H(S) = \frac{S - \omega_o}{S + \omega_o}$$
(24)

Substituting S by  $j\omega$ ,



$$H(j\omega) = \frac{j\omega - \omega_o}{j\omega + \omega_o}$$
(25)

$$H(j\omega) = -\frac{1 - j\frac{\omega}{\omega_o}}{1 + j\frac{\omega}{\omega_o}}$$
(26)

Equation (26) is the transfer function of all-pass filter in Figure 11.

When  $\omega=0$ , i.e. when direct current is passed through the circuit

$$|H(0)| = \left| -\frac{1 - \frac{0}{\omega_o}}{1 + \frac{0}{\omega_o}} \right| = 1$$
(27)

And when,  $\omega = \infty$ ,

$$|H(\infty)| = \left| -\frac{-\frac{\omega}{\omega_o}}{\frac{\omega}{\omega_o}} \right| = 1$$
(28)

Equations (27) and (28) concludes that and ideal all-pass active filter does not alter the magnitude of signal.

Rewriting equation (24) to determine the phase response

$$\phi(j\omega) = \tan^{-1}\left(\frac{\omega}{-\omega_o}\right) - \tan^{-1}\left(\frac{\omega}{\omega_o}\right)$$
(29)

$$\phi(S) = \pi - \tan^{-1}\left(\frac{\omega}{\omega_o}\right) - \tan^{-1}\left(\frac{\omega}{\omega_o}\right) \tag{30}$$

$$\phi(S) = \pi - 2tan^{-1} \left(\frac{\omega}{\omega_o}\right) \tag{31}$$

When  $\omega=0$ ,  $\theta=\pi$  and when  $w=\infty$ ,  $\theta=\pi - 2^*\pi/2 = 0$ . Which concludes that phase changes from 180° to 0° when frequency from low to high are swept through the all-pass filter. [11]

A particular case of interest is when  $\omega = \omega_o$ ,

$$\phi(\omega_o) = \pi - 2tan^{-1} \left(\frac{\omega_0}{\omega_o}\right) \tag{32}$$



$$\phi(\omega_o) = \pi - 2 * \frac{\pi}{4} \tag{33}$$

$$\phi(\omega_o) = \frac{\pi}{2} \tag{34}$$

Equation (34) implies that when R and C are tuned to frequency  $\omega_{o}$ , also known as corner frequency of all pass filter, the phase of the output frequency is shifted by 90° with respect to the corner frequency. [11]

#### 5.2 Phase Shifter Implementation

In the design of I/Q modulator, the signal from Local oscillator (LO) is fed into two mixers, but the mixers receives LO signal in quadrature. In real world implementation, quadrature shifters implemented are wide-band in nature. The all-pass filter presented on section 5.1 can shift only one frequency to 90°. One of the solutions to obtain wideband quadrature shift is to implement all-pass network in parallel.

The explanation of the working of all-pass network is beyond the scope of the report and shall not be discussed more on this but a phase shift networks has been developed using a freeware program called **QuadNet** available at the developer's site <u>www.TonneSoftware.com</u>. The software calculates the values for capacitors and resistors required for generation of quadrature signals from given input. Furthermore, choice to estimate the frequency band, number of all-pass section required and flexibility to keep value of one of the components constant makes it easy to generate the required network in few clicks. For the thesis purpose, all six all-pass section was specified for the frequency band of 100 KHz to 250 KHz and capacitor value was specified to 10 pf, rest of the calculation was done by the software generating network shown in Figure 12



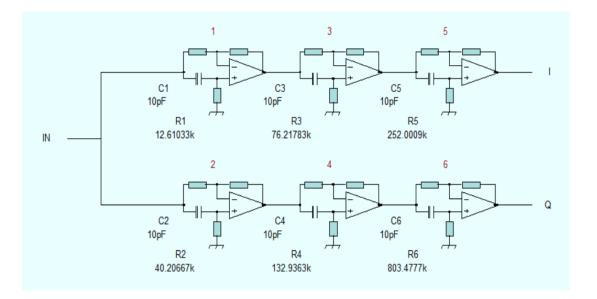


Figure 12 : All-pass network for 100 KHz – 250 KHz built on QuadNet

Network presented in Figure 12 was simulated in Multisim software. All the component values were used as calculated by software. Unlabelled resistors were chosen to be 10 K $\Omega$  and all the op-amp used were LM324N available in Multisim component library. A 1 volt peak to peak sinusoid was passed through input to obtain output at the ports I and Q. AC analyses was performed by sweeping frequency from 10 KHz to 1 GHz which produced the result shown in Figure 13. The red trace represents the phase response of the In-phase LO signal while green trace represents the quadrature LO signal.

The simulation result is summarized in Table 2. Magnitude and phase data are in respective columns as the name suggests. Phase of quadrature signal (Q) with respect to In-phase (I) signal is calculated. And also magnitude percentage with respect to input LO signal is also calculated for both the signals.

	In-phase signal			Quadrature signal			
frequency (KHz)	magnitude(mV)	phase(degrees)	mag(%)	magnitude (mV)	phase(degrees)	phase difference	mag(%)
100	931.7406	-30.3616	93.17406	927.1901	-120.2223	89.8607	92.71901
200	786.5251	-136.4703	78.65251	783.5985	133.7453	-90.2156	78.35985
250	702.1046	-174.4973	70.21046	699.7318	95.7478	-90.2451	69.97318



It can be concluded from the simulation data that the phase difference of the two outputs for the give frequency range remains approximately near 90°, with less than 1% error, But the magnitude attenuation gets bigger with rise in frequency.

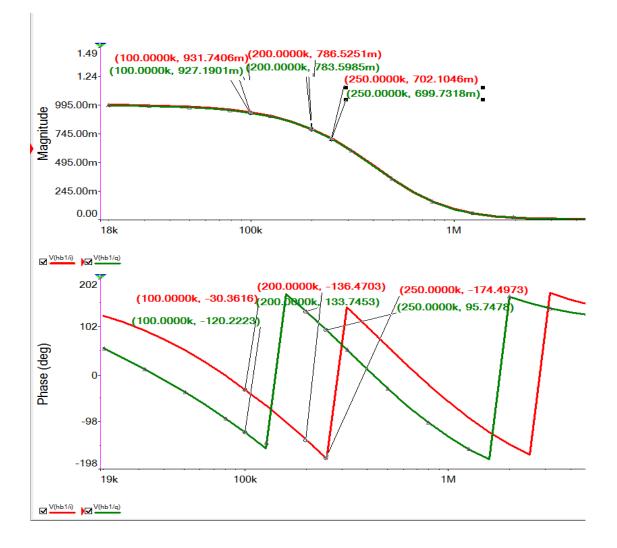


Figure 13 : AC-analyses 100-250 KHz all-pass parallel network

The same network was implemented in hardware and tested at different frequencies, graphs shown in appendix 1. The hardware performs quadrature shift on very narrow region from 190 KHz to 210 KHz with approximately  $\pm 5^{\circ}$  continuous fluctuations. One main reason to this result could be the tolerance value of the components used. As the capacitors and resistors used had 5% tolerance, and to achieve precise resistance value, number of resistors were combined together, resulting in huge addition of tolerance in total.



#### 6 I/Q Modulator

Signals that is described by polar magnitude (A) and phase angle ( $\phi$ ) can also be represented by Cartesian co-ordinates. Cartesian coordinates forms the basis of quadrature modulation. X axis and y axis are designated as In-phase (I) and quadrature (Q) respectively in quadrature modulation. In I axis the unit basis vector has length of one and angle zero, while Q axis basis vector has equal magnitude as I axis but the angle is 90°. [12] This can be represented as:

$$B_I = 1 \cdot e^0 \tag{35}$$

$$B_Q = 1 \cdot e^{j\pi/2} \tag{36}$$

Transforming (35) and (36) to time domain signals

$$B_I(t) = 1 \cdot \cos(\omega_C t + 0) = \cos(\omega_C t) \tag{37}$$

$$B_0(t) = 1 \cdot \cos(\omega_C t + \pi / 2) = \sin(\omega_C t) \tag{38}$$

Scaling the basis vector and adding them can be done to produce any signal with coordinates (I, Q).

$$S = I \cdot B_I + Q \cdot B_Q \tag{39}$$

I and Q are practically time varying, so Equation (39) can be written in time domain as

$$s(t) = I(t) \cdot \cos(\omega_C t) + Q(t) \cdot \sin(\omega_C t)$$
(40)

Equation (40) is the general quadrature modulator (QM) equation. [12] Figure 14 is the model of ideal I/Q modulator showing the implementation of the general QM equation.



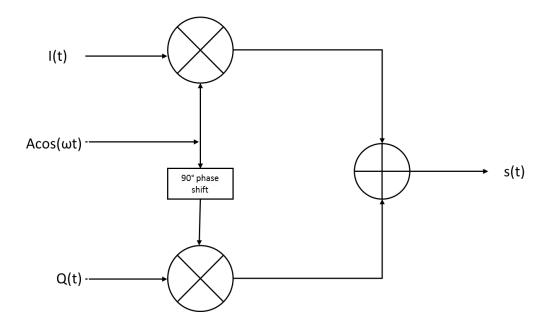


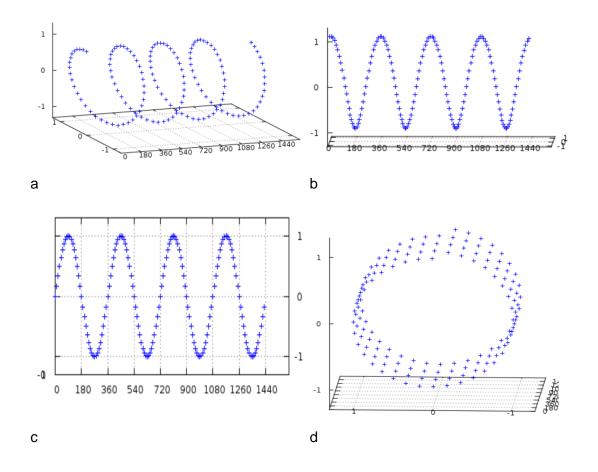
Figure 14 : fundamental block of Quadrature modulator

#### 6.1 I/Q Data

While looking at a simple sinusoidal wave one cannot determine if the frequency is positive or negative because both produce same curve, i.e.  $\cos(x) = \cos(-x)$ . Since mixing frequency f1 with f2 will produce f1+f2 and f1-f2 or f2+f1 and f2-f1, it is hard to tell the outcome. Also it's hard to determine the power of the signal everywhere within the signal. [13] These dilemma can be solved by I/Q data.

Let's examine Figure 15. If the signal is observed in 3 dimensions as a helix I and Q data can be easily studied. If the helix is seen from the front end then it gives us in-phase data (I data) whereas if the same helix is seen from top view Q data can be obtained. The difference is clearly observed that the Q data signal is 90° out of phase and starts at 0. And when the same helix is viewed down the time axis one can quickly figure out that it is turning counter clockwise indicating a positive frequency. [13]





a) Helix view of a signal b) front view of the helix c) top Figure 15 : view of the helix d) view from the time axis (Reprinted from [13])

It can be summarized that expressing a signal in I/Q data form gives more control on signal manipulation. The concept of I/Q data are widely use in DSP (Digital Signal Processing) domain.



#### 6.2 Sources of Errors in Quadrature Modulator

No system is perfect and quadrature modulator is no exception. Errors are introduced from the input stage till the final signal addition stage. This is why Figure 14 is an ideal representation of quadrature modulator and our goal is to get as close as possible to it. [12]

Let's examine the errors at different points in the ideal quadrature in Figure 16 below:

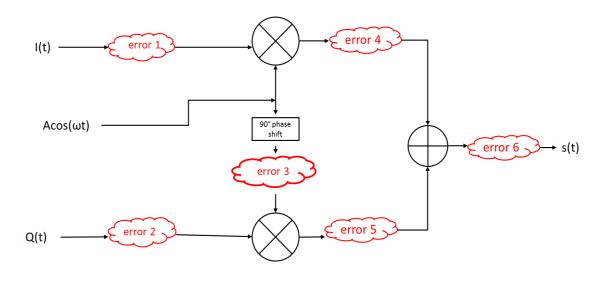


Figure 16 : Introducing errors in ideal quadrature modulator

When a signal is passed through a system, if non-linear distortions are not considered, then a signal can have gain and offset errors. [12] Erro1 and error 2 in Figure 16 can thus be written as follows:

$$I(t) = A_I I(t) + offset_I$$
(41)

$$Q(t) = A_Q \cdot Q(t) + offset_Q$$
(42)

Where,  $A_I$  and off<sub>I</sub> Amplitude error in the I channel respectively whereas  $A_Q$  and off<sub>Q</sub> are errors in Q channel. If we take a look at the basis signals (quadrature signals generally obtained through oscillator), cosine function is passed through 90° phase shifter circuit. If the phase of cosine function is taken to be 0, the phase of output sine function is 90° along with some quantity of phase error introduced by the phase shifter. And we cannot eliminate the Amplitude and offset error in both the quadrature signals. [12] So the basis signals can be re-estimated as



$$\cos(\omega_C t) = A_C \cos(\omega_C t) + offset_C$$
(43)

$$\sin(\omega_C t) = A_S \sin(\omega_C t + \varepsilon_S) + offset_S$$
(44)

Where,  $A_c$ ,  $A_s$ , offset<sub>c</sub>, offset<sub>s</sub> are respective amplitude and offset errors and  $\epsilon_s$  Is the error introduced due to shift.

With these new errors taken in consideration, we can approximate the signals with error 4, error 5 and finally the output signal with error 6.

Using relation (43) and (41) signal at error4 point would be then

$$I(t).\cos(\omega_C t) = (A_I.I(t) + offset_I).(A_C\cos(\omega_C t) + offset_C)$$
(45)

And from relation (44) and (42), signal at error5 would be

$$Q(t).\sin(\omega_C t) = (A_Q, Q(t) + offset_Q).(A_S\sin(\omega_C t + \varepsilon_S) + offset_S)$$
(46)

Combining relation (45), (46) and (40), a better approximation of the signal s (t) could be achieved.

$$s(t) = (A_I \cdot I(t) + offset_I) \cdot (A_C \cos(\omega_C t) + offset_C) + (A_Q \cdot Q(t) + offset_Q) \cdot (A_S \sin(\omega_C t + \varepsilon_S) + offset_S)$$
(47)

Considering ideal condition, no gain errors, offset errors and phase errors,  $A_I=A_Q=A_C=A_S=1$ , offset\_I=offset\_Q=offset\_S=0 and  $\epsilon_s$ , equation (47) reduces to following

$$s(t) = (1 * I(t) + 0). (1 * \cos(\omega_c t) + 0) + (1 * Q(t) + 0). (1 * \sin(\omega_c t + 0) + 0)$$
(48)

$$s(t) = I(t) \cdot \cos(\omega_C t) + Q(t) \cdot \sin(\omega_C t)$$
(49)

The equation (49) is the ideal I/Q modulator equation.



### 7 Implementation

#### 7.1 Simulation in Multisim

An I/Q modulator was constructed in NI Multisim 13.0 and simulated to study its performance. Different stages were individually constructed and combined so that real implementation problems would be easily studied beforehand. The simulation block diagram for this purpose is shown in Figure 17. Description of the blocks are presented in following paragraphs.

#### 90PhaseShifter:

This block is the phase shifter described in section 0. The LO signal from function generator XFG1 is feed into the input and the resulting in-phase and quadrature LO signal flows through nets LO\_in and LO\_90.

A1 and A2:

Both A1 and A2 are the multiplier blocks added from Multisim library. They are the ideal multiplier, and their realization in the real world is impracticable. But for the study purpose, since the library does not contain any mixer IC, it was one of the solutions for easy implementation. Multipliers were here used as mixers. The gain was set to unity for both the mixers.

#### Gain:

GAIN block consist of two independent inverting gain stages built from LM324N op-Amp. Initially, the gain is set to unity but can be changed to different levels adjusting variable resistors. The main purpose of this block is to compensate the gain irregularities in the output signals from the mixers A1 and A2 as it is essential that the amplitude and frequency be similar for both the output for correct operation of I/Q modulator.

Signal Adder:



Signal adder block employs the same circuit discussed in section 3. Modulated signal is obtained through output port of signal adder.

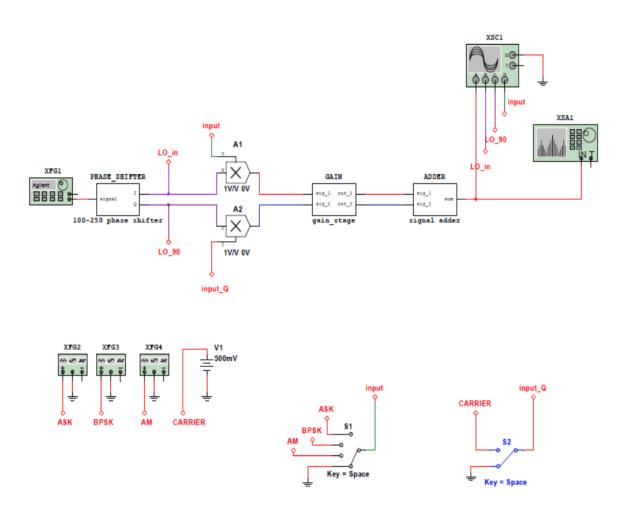


Figure 17 : Multisim I/Q block diagram

Remaining symbols and components are summarized in Table 3. The magnitude of the AC signals are in Vp units meaning peak-peak voltage while the DC volts are denoted by V.



## Table 3: List of components and their functions

Name	Compo- nents	Function	Frequency	magnitude
XFG1	Function generator	Provides LO signal into the phase shifter	200 KHz Sine wave	500 mVp
XFG2	Function generator	Provides Input signal for ASK to the mixer A1	20 KHz square wave 50% duty cy- cle	500 mV with off- set of 500 mV
XFG3	Function generator	Provides Input signal for BPSK to the mixer A1	20 KHz Square wave 50% duty Cy- cle	1 Vp
XFG4	Function generator	Provides Input signal for AM to the mixer A1	20 KHz Sine wave	500 mVp
XSC1	4-channel oscilloscope	Plots the graph of mod- ulated signal, input-I sig- nal, and in-phase and quadrature signals of LO	-	-
XSA1	Spectrum an- alyser	Plots power spectrum of modulated signal	-	-
S1	1 switch Selects the type of mod- ulation to be perfomed		-	-
S2	switch	Controls carrier input	-	-

Carrier signal input is set to 500 mV through a DC source. Switch S2 can be set to ground or the carrier input 500 mV as required.



#### 7.2 Results

Different modulations were performed using the I/Q modulator constructed in Multisim. All the waveforms from the simulation are attached to Appendix 2. For the sake of easiness the waveforms are formatted with different colours.

Coloured Signals	Implemented signal
Green	Baseband input signal
Dark Purple	LO-inphase signal
Light Purple	LO-quadrature signal
Red	Modulated signal

Table 4: Colour representation of simulated signals

Table 4 summarizes the colour codes of the simulated signals. The data readings are as well labelled on the waveform itself where-ever possible.

It is evident from the results that modulated signals form particular wave pattern. ASK without carrier has carrier wave whenever the square base band signal is high and, carrier is not present when the square signal is low. In BPSK, the change in phase of the carrier was observed where the notch was formed at the intersection of input signal and modulated signal.

AM modulation with carrier and no-carrier had similar looking time domain waveforms but looking at the frequency domain, three peaks were observed in former case, where one peak was at 180 KHz, second at 200 KHz and third at 220 KHz, indicating two translated frequencies and one carrier wave from quadrature input while the latter case only had two peaks, at similar region except 200 KHz frequency content missing.

Noise is inevitable in real world implementation as discussed earlier in sources of Errors section. A similar simulation setup was done in LabVIEW and carrier with uniform white noise was introduce to ideal mixer. The resulted BPSK, ASK and AM modulated signals are presented in Appendix 2.



#### 8 Conclusion

An effort to design an I/Q modulator was done. Since this project was confined to certain frequency range from 200 to 250 KHz, the mixer could not fulfil the desired application area, as suggested from the result of the spectrum analysis as the modulated signals were highly attenuated. Furthermore, leakage of the carrier signal from LO port into the output port of LT5560 added hindrance to the mixer's implementation, hardware realization could not be completed. But different stages of I/Q modulators were built and tested. An approach to real implementation was done using Multisim simulation and different modulation were performed through the simulated I/Q modulator and waveforms were studied.

A project might not come to success but one can always learn from failure and try to avoid the errors in next attempt. An important lesson learned was that, an I/Q modulator require very precise component selection and capability to design layouts that minimizes errors as much as possible. The phase shifter would have a better response if 1% toler-ance resistors and capacitors were used. Another important point that could be learnt is that errors are inevitable. Amplitude errors and offset errors can be minimized to almost negligible once they appear in system, but phase are difficult to tackle with and requires effective techniques to develop phase shifter with minimum error, thus providing good performance capability to I/Q modulator.

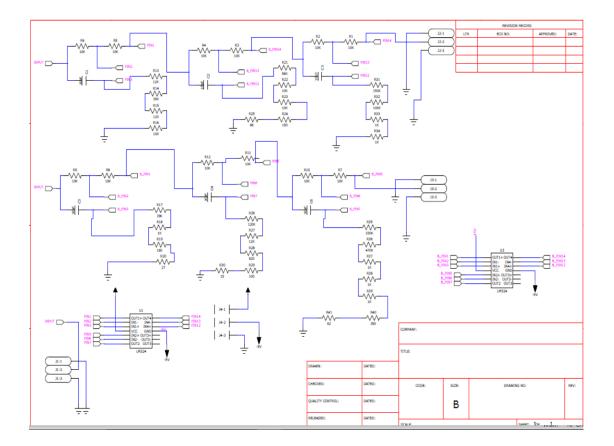


#### 9 References

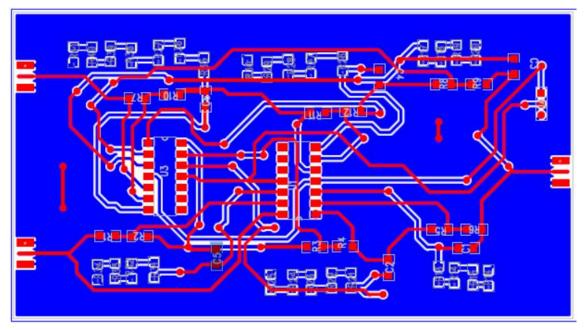
- W. Tomasi, ELECTRONIC COMMUNICATION SYSTEMS FUNDAMENTALS THROUGH ADVANCED, New Jersey: Prentice-Hall, Inc., 2001.
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## All Pass filter Network

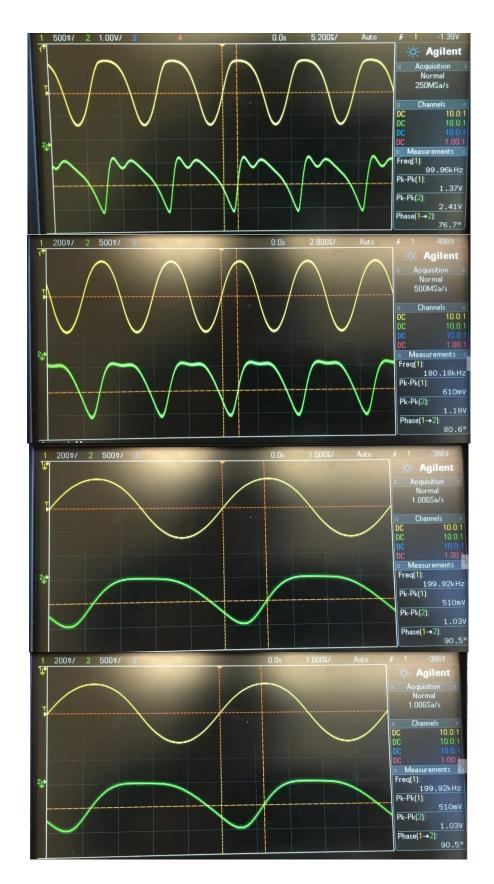


### All Pass Filter Schematic



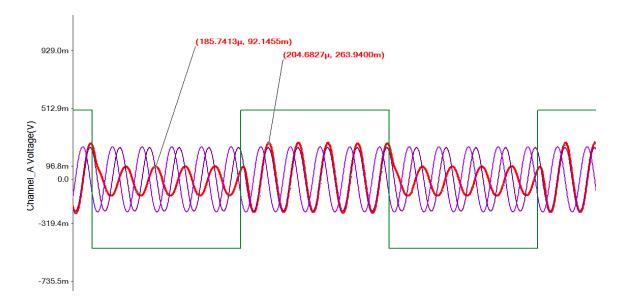
All-Pass Filter layout





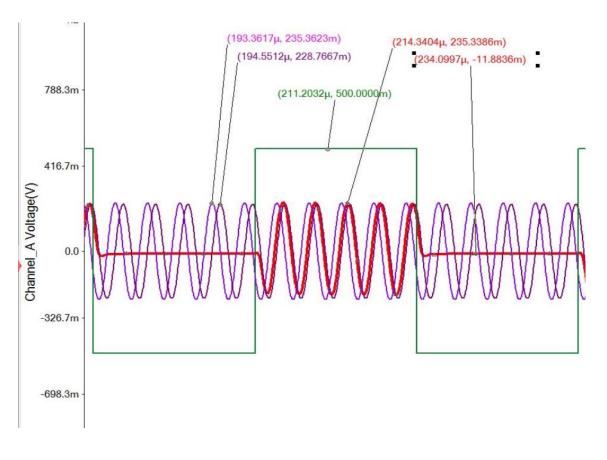
#### Phase shifter I and Q Output measurements at Different Frequencies





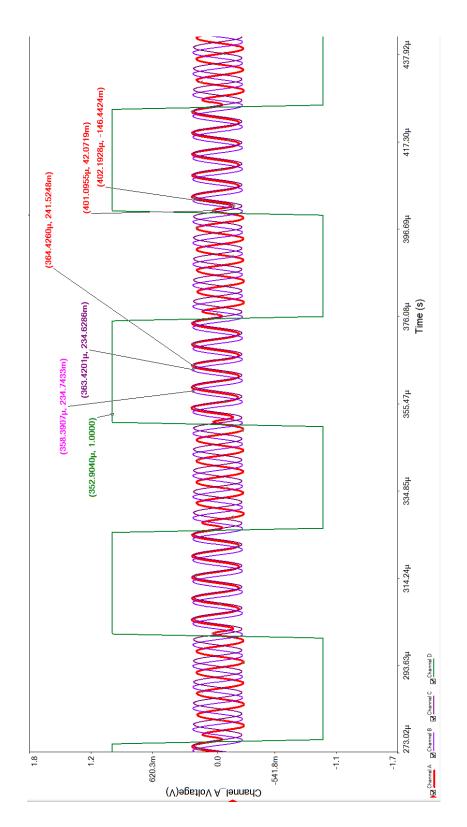
## Simulation Result of I/Q Modulator

ASK with carrier



ASK without carrier



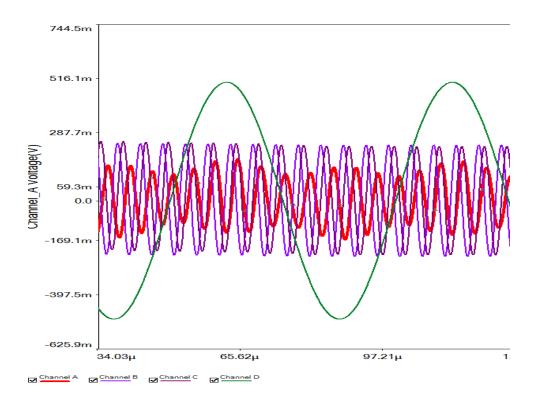


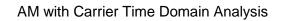
BPSK

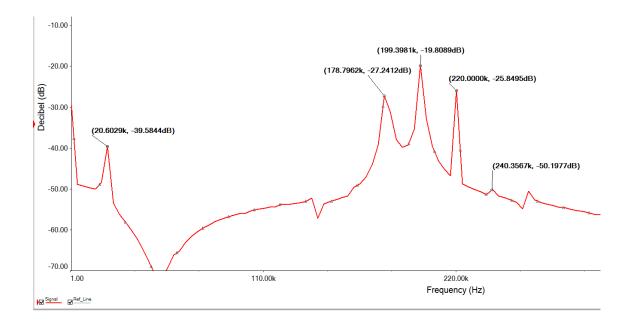


Appendix 2 2(5)

## Appendix 2 3(5)



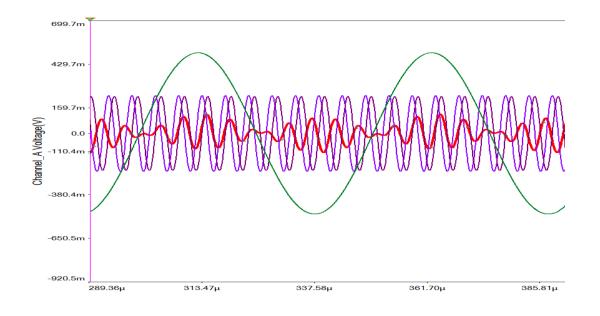




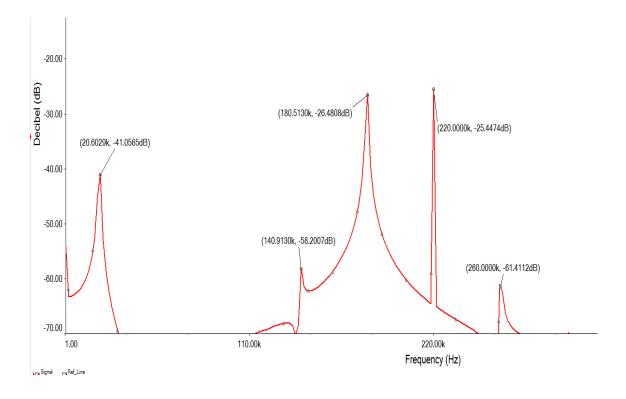
AM with Carrier Spectral Analysis



Appendix 2 4(5)



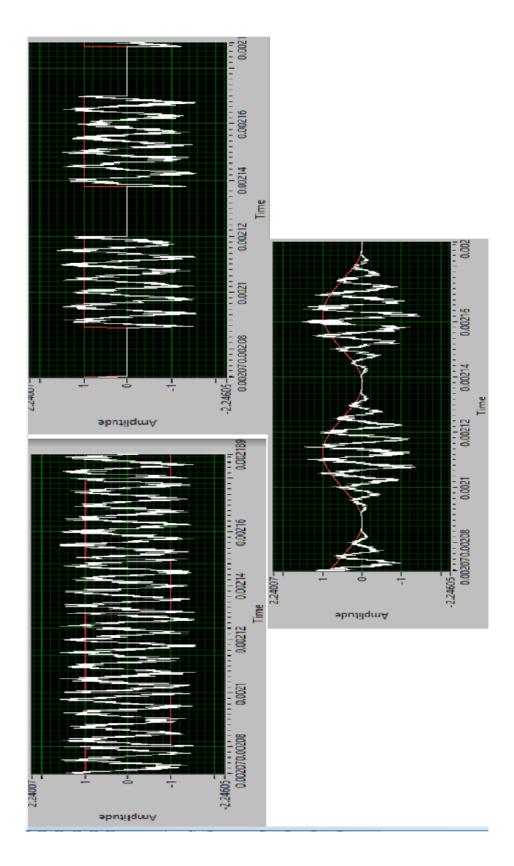
## AM with no carrier Time Domain Analysis



AM with no carrier Spectral Analysis



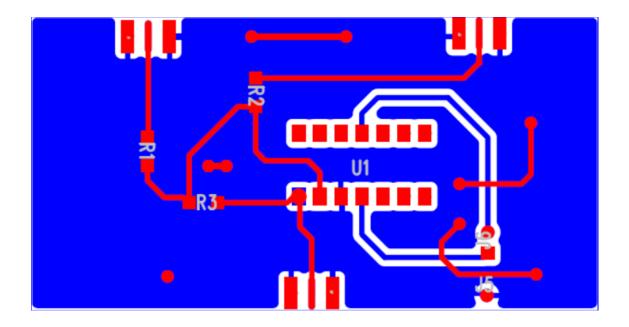
## Appendix 2 5(5)



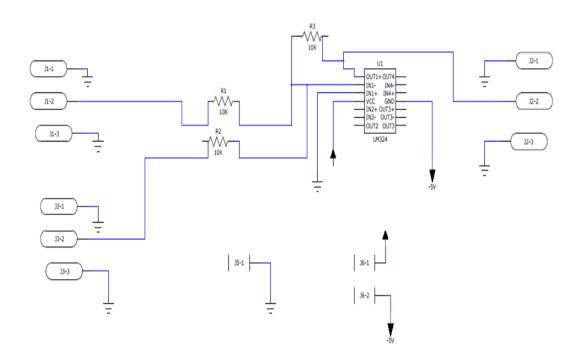
BPSK, ASK and AM with white uniform noise in carrier



## **Summing Amplifier Physical Implementation**



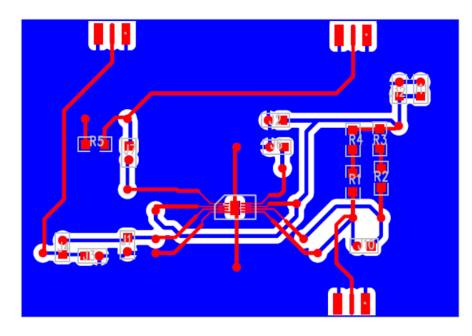
## Summing Amplifier PCB Layout



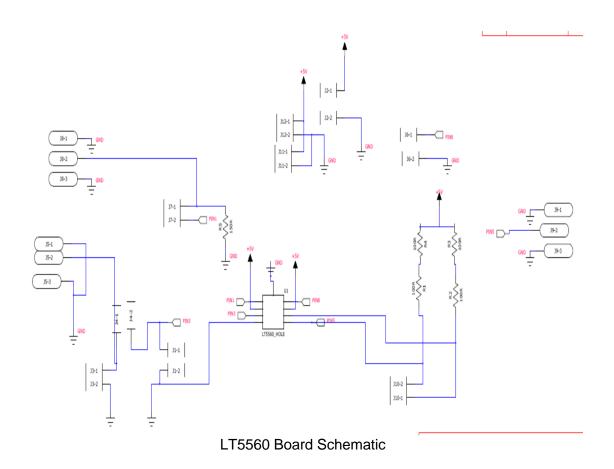
Summing Amplifier Schematic



## LT5560 Board implementation

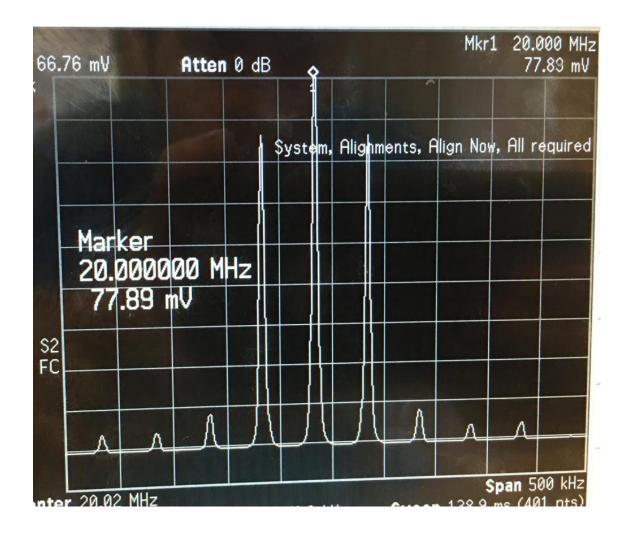


LT5560 Board Layout





Appendix 4 2(2)



### LT5560 spectrum analysis at LO = 20 MHz, IF = 500 KHz

