Introduction to FM Detection: A Necessary Revisit

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Abstract
This paper reports techniques of FM demodulation in time domain. The results have shown obvious advantage of staggered tuned FM demodulator circuit over single tuned circuit. New PLL system to overcome false locking has been proposed.

Keywords
Stagger Tuned Circuit; False-Locking; Compound Loop PLL, FM-AM Conversion; Balanced Slope Detector

I. Introduction

It is to be remembered that FM modulation satisfies main features of transmission, namely, multiplexing, power requirement, antenna size and portability. FM demodulation is inverse process of reverting back the baseband signal [1-3].

The very simplest form of FM demodulation is known as slope detection or demodulation. The origin of the term dates back to period (1896 – 2016). It consists of a tuned circuit that is tuned to a frequency slightly offset from the carrier of the signal. It uses the attenuation slope of the tuned circuit. It converts the received FM signal to AM signal. Then the baseband signal can be recovered using either a square law detector or an envelope detector. Theoretically speaking the received FM signal can be converted to an AM signal by differentiating the received signal. The distortions of the single tuned slope detector can be largely removed by balanced slope detector, reported by Seely in 1936. But, when the carrier frequency is in the HF range the realization of differentiator is not possible.

At HF range the idea of differentiation is the demodulation of FM can be accomplished by a slope detector, which, in its simplest form, can be modeled as a differentiator followed by an envelope detector. A second method that can be used to demodulate an FM is by means of a phase-lock loop (PLL). It is, in fact, the preferred method nowadays because of the availability of inexpensive PLL integrated circuits.

The demodulation of FM signal is done in the frequency domain in almost all the text book but it can also be done in the time domain which is not written. Moreover, the demodulation of subcarrier multiplexed can also be done with the slope detector. Further instead of single tuned or doubled tuned circuit staggered tuned circuit can also be used.

But PLL demodulators at high frequency range cause serious problem in due to transmission delay in the loop. However, it can be considerably mitigated using compound loop that will also be discussed in this tutorial. It is to be noted that attenuation slope of staggered tuned circuit can also be used for FM to AM conversion.

A conventional PLL demodulator at HF can cause serious problem during acquisition because of transmission delay in the loop. The delay in the loop arises not only because of the filter networks, but also due to the finite response time of the Voltage Controlled Oscillator (VCO) and the phase detector. This additional phase shift will have a significant effect on the pull-in characteristics and the tracking behavior of a simple phase lock loop (PLL). The phase detector output in the PLL shows zero voltage although the loop has not achieved locking and this mystifying behavior is known as 'False Locking', shown in Fig. 2. However, this spurious locking can be eliminated using a compound loop PLL shown in Fig.

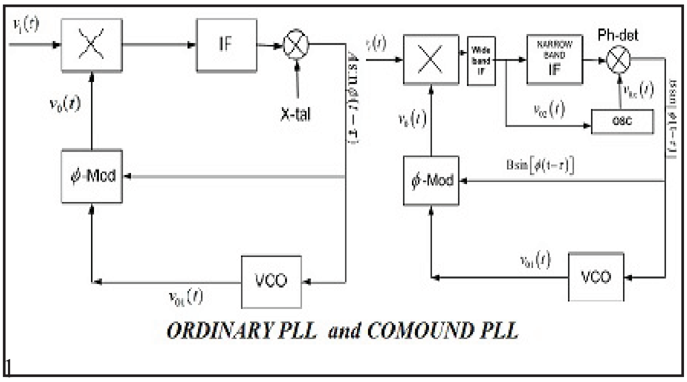


Fig. 1: Block Diagram of Ordinary and Compound PLL

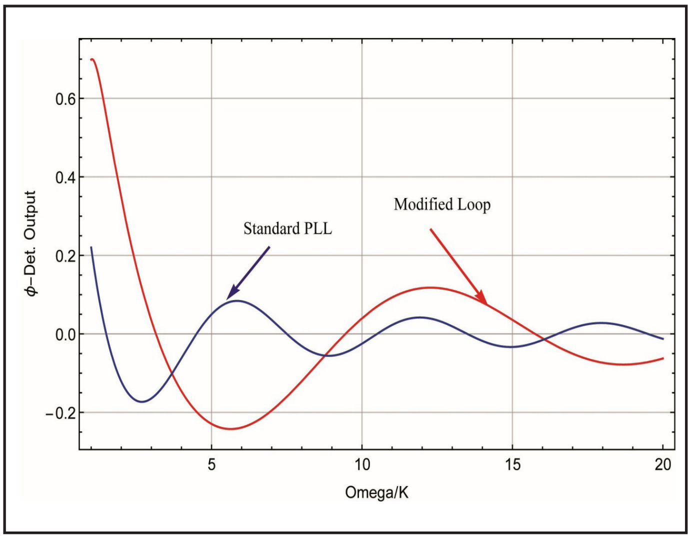


Fig. 2: Phase Detector Output for Ordinary and Compound PLL

The authors feel that that a tutorial session on FM demodulation needs to be organized.

II. Detection of Frequency Modulated Signal

In an FM wave the instantaneous frequency is proportional to the modulating signal. Therefore, the output of an FM detector will have to be proportional to the instantaneous frequency of the modulated wave. A simplest method of recovering the modulating signal from an FM signal is first to convert the FM wave into a corresponding AM signal. Once the conversion is done one can use conventional AM signal. An ordinary

 ISSN : 2230-7109 (Online) | ISSN : 2230-9543 (Print)

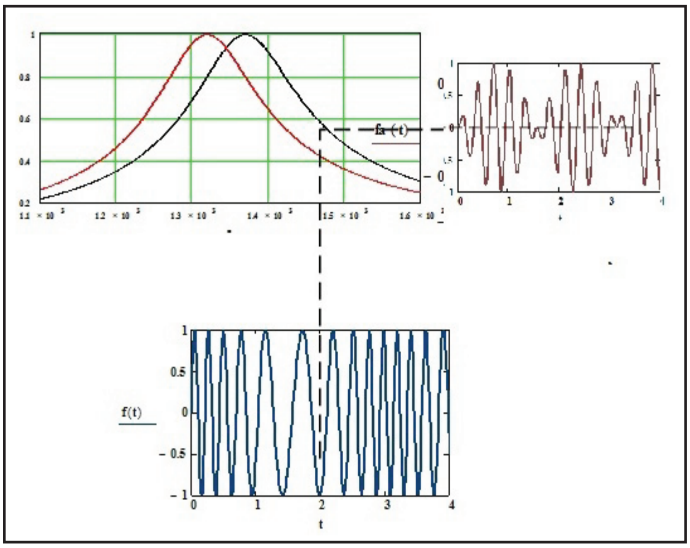


Fig. 3: FM-AM Conversion Employing Single Tuned Amplifier

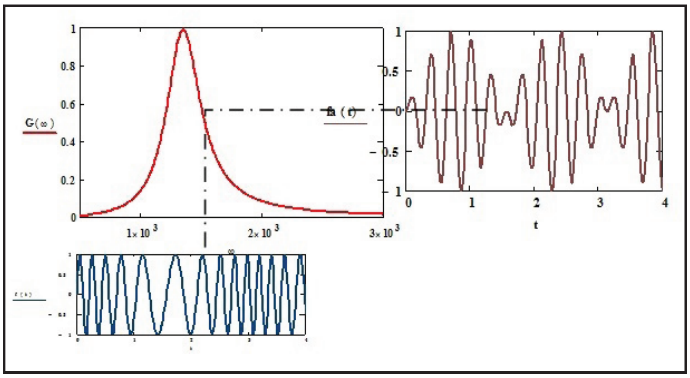
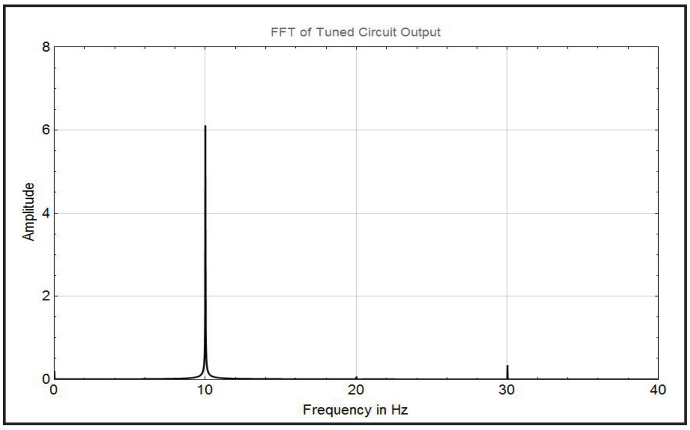
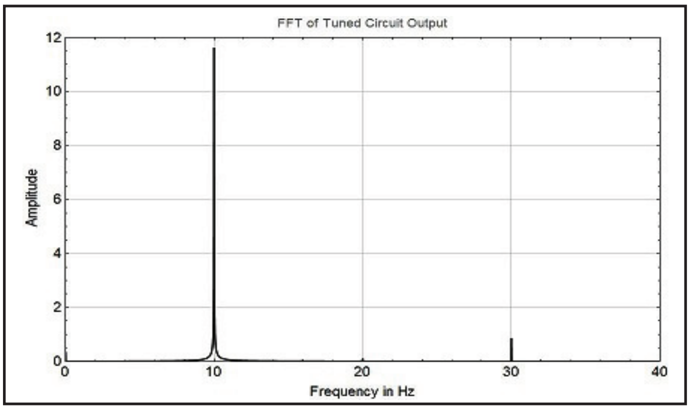


Fig. 4: FM-AM Conversion Using Stagger Tuned Amplifier



(a)



(b)

Fig. 5: Frequency Response of (a) Single Tuned Circuit and (b) Stagger Tuned Circuit

method of converting an FM signal to an AM signal will be to exploit the frequency response characteristic of a single tuned circuit. Thus when the frequency modulated wave is applied at the input of a single tuned circuit, the centre frequency of which is little away from the carrier frequency, the output of the amplifier will be amplitude modulated. FM-AM conversion using single tuned amplifier and stagger tuned amplifier is shown in Fig. 3 and Fig. 4 respectively.

Recalling the gain function of a single tuned amplifier as given below

$$G(j\omega)=\frac{G_0}{1+jQ\left(\frac{\omega}{\omega_0}-\frac{\omega_0}{\omega}\right)} \quad (1)$$

And also the instantaneous frequency of the FM wave as

$$\omega(t)=\omega_c+\Delta_f\cos\omega_m t \quad (2)$$

The output of the single tuned amplifier is then given by

$$v_0=A_cG(j\omega) \quad (3)$$

where A_c is the strength of the carrier and Q is the quality factor of the tuned circuit. Referring to Fig. 1 one finds that when the instantaneous frequency of the modulated wave is changing sinusoidally, the output amplitude is also almost changing sinusoidally. Thus the frequency variation of incoming signal is changed into amplitude variation. Now one can combine (1), (2) and (3) if one remembers that ω of (1) has to be replaced by $\omega(t)$ of (2). Thus

$$v_0=A_cG_0\frac{1}{1+jQ\left[\frac{\omega_c+\Delta_f\cos\omega_m t}{\omega_0}-\frac{\omega_0}{\omega_c+\Delta_f\cos\omega_m t}\right]} \quad (4)$$

Note that ω_0 is the resonant frequency of the tuned circuit.

Since $\left(\frac{Q}{\omega_0}\right)$ is small, one can write (4) as

$$|v_0|\cong\left[1-\frac{2Q^2}{\omega_0^2}\times\{\Omega+\Delta_f\cos\omega_m t\}^2\right]A_cG_0$$
$$=A_cG_0\left[\begin{array}{l}1-\frac{2Q^2\Omega^2}{\omega_0^2}-\frac{4Q^2}{\omega_0^2}\Omega\times\Delta_f\cos(\omega_m t)-\frac{Q^2\Delta_f^2}{\omega_0^2}\\ -\frac{Q^2\Delta_f^2}{\omega_0^2}\cos(2\omega_m t)\end{array}\right] \quad (5)$$

Therefore, the net output of the tuned amplitude is given by

$$v_0=A_cG_0K\left[1-\frac{4Q^2\Omega}{\omega_0^2K}\Delta_f\cos\omega_m t-\frac{Q^2\Delta_f^2}{\omega_0^2K}\cos 2\omega_m t\right] \quad (6)$$
$$\times\cos[\omega_c t+m\sin\omega_m t+\varphi(\omega)]$$

where $\varphi(\omega)$ is the phase shift of the FM wave in going from the input to the output and is given by

$$\phi(\omega)=\tan^{-1}Q\left[\frac{\omega(t)}{\omega_0}-\frac{\omega_0}{\omega(t)}\right] \quad (7)$$

For Ω negative that when $\omega_c<\omega_0$

$$|v_{01}|=A_cG_0\left[\begin{array}{l}1-\frac{2Q^2\Omega^2}{\omega_0^2}+\frac{4Q^2}{\omega_0^2}\Omega\\ \times\Delta_f\cos(\omega_m t)-\frac{Q^2\Delta_f^2}{\omega_0^2}\\ -\frac{Q^2\Delta_f^2}{\omega_0^2}\cos(2\omega_m t)\end{array}\right] \quad (8)$$

$$|v_{01}| - |v_0| = A_c G_0 \left[\frac{8Q^2}{\omega_0^2} \Omega \times \Delta_f \cos(\omega_m t) \right] \quad (9)$$

Thus the second harmonic distortion is eliminated (balanced slope detector)

Referring to (6) one finds that the output of the tuned amplifier is a composite modulated wave in which both the instantaneous frequency and the amplitude are modulated. The amplitude is seen to consist of the fundamental, i.e., the modulated component, and second harmonics component. The amplitude modulation indices of the fundamental and the second harmonic component are respectively given by

$$m_1 = \frac{4 \times Q^2 \times \Omega \times \Delta_f}{\omega_0^2 - 2Q^2\Omega^2 - Q^2\Delta_f^2} \quad (10)$$

$$m_2 = \frac{Q^2 \Delta_f^2}{\omega_0^2 - 2Q^2\Omega^2 - Q^2\Delta_f^2} \quad (11)$$

Therefore the percentage second harmonic distortion is given by

$$D_2\% = \frac{m_2}{m_1} \times 100 = 25 \frac{\Delta_f}{\Omega}\% \quad (12)$$

From (10), (11) and (12) one finds that the percentage second harmonic distortion can be reduced by decreasing the value of the frequency deviation of the FM signal and increasing the detuning of the tuned amplifier from the centre frequency of the amplifier. Increasing the value of the detuning, decreases the modulation component. From the expression of the function for the modulation index of the fundamental component, it is found that m_1 has a maximum value for a value of Ω lying between zero and infinity. The maximum value of m_1 will occur at

$$\Omega_1 = \sqrt{\frac{Q^2\Delta_f^2 - \mathrm{w}_0^2}{2Q^2}} = \frac{\sqrt{\Delta_f^2 - BW^2}}{\sqrt{2}} \quad (13)$$

Therefore, one gets

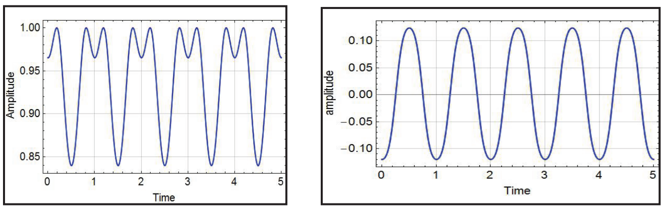
$$(m_1)_{\max} = \frac{\Delta_f}{\Omega_1} \quad (14)$$

Thus in this it is advisable to use a small deviation FM wave.

A typical circuit for the demodulation of an FM wave utilizing this principle is shown in Fig. The output circuit is simply an envelope detector which responds to amplitude modulated wave only. Therefore the frequency modulated part of the wave will not be demodulated.

From the above analysis it is found that when the amplifier is tuned to the centre frequency of the tuned circuit, the output will be distorted and the frequency of the demodulated signal will consist of the even harmonics of the original modulating signal. The waveforms for the single tuned circuit is shown in Fig. 6.

Thus in this it is advisable to use a small deviation FM wave



(a) (b)

Fig. 6: Demodulated Signals for one side of carrier frequency (a) and the resultant waveform considering both sides of carrier frequency

III. Differential Equation for FM Detection: Single Tuned Circuit

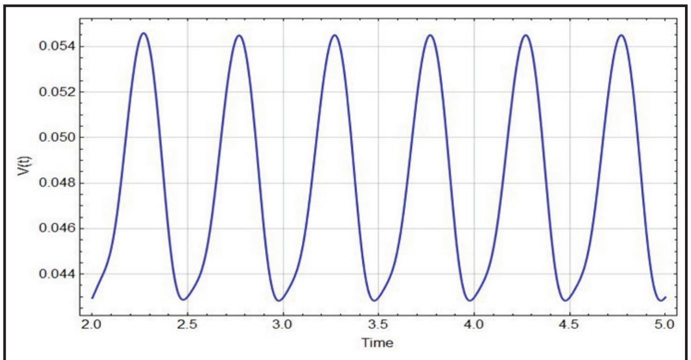
The coupled differential equations governing FM detection can be obtained as

$$\frac{d}{dt}V(t) = \left(\frac{\omega_0}{2Q}\right) \times \left[-V(t) + E\cos(\psi(t) - \theta(t))\right] \quad (15)$$

$$\frac{d\theta(t)}{dt} = (\omega_0 - \omega_c) + \left(\frac{\omega_0}{2Q} \times \frac{E}{V(t)}\right)\sin\{\psi(t) - \theta(t)\} \quad (16)$$

where $E = \frac{I_f}{G}$. Here I_f is the amplitude of the FM current and G is the conductance of the tuned circuit such that $i_{fm}(t) = I_f \exp[j\omega_c t + j\psi(t)]$ and the voltage across the tuned circuit is $v(t) = V(t)\exp[j\omega_c t + j\theta(t)]$

The amplitude and phase response is sown in Fig. 7.



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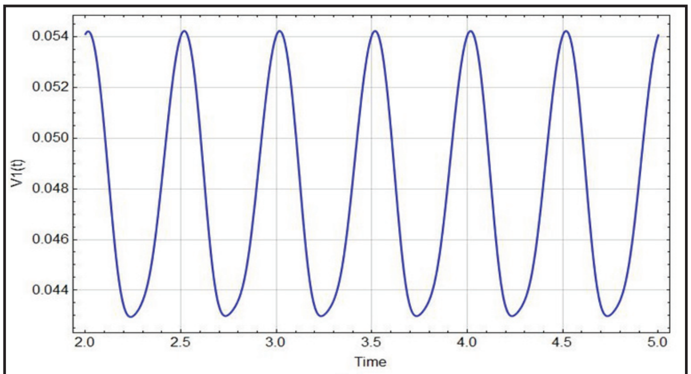

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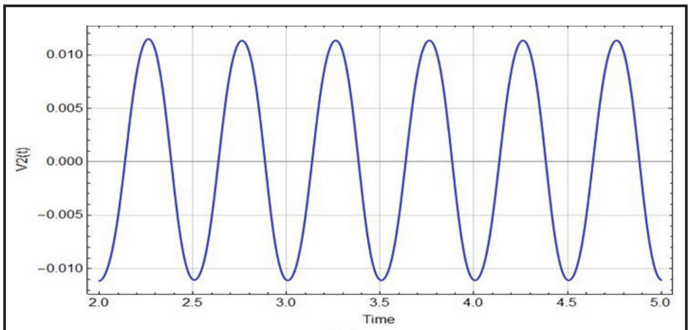

(c)

Fig. 7. (a) Amplitude of the FM demodulated signal considering one side of the carrier frequency; (b) Amplitude of the FM demodulated signal considering the other side of the carrier frequency; (c) resultant amplitude

The single tuned discriminator has a small useful linear region of the frequency response characteristic of the tuned amplifier. As a result, the frequency deviation of the FM signals needs to be small for small distortion at the output of the detector. A circuit that increases the linear region of the operation utilizes two tuned circuits and is, therefore, called a double tuned discriminator. The centre frequency of the tuned amplifier 1 is detuned with respect to the carrier frequency by a certain amount, say $\Delta\omega$. Similarly the centre frequency of the tuned amplifier 2 is also detuned by the same amount. Note that the detuning of the two circuits lie on the opposite sides of the carrier frequency. Let us assume that the two tuned amplifiers are identical in all respects. The net output of the circuit is the difference between the voltages V_{01} and V_{02}. From the fact that the linear regions of the two tuned circuits are simultaneously used it is evident that the net linear region of operation of the circuit is increased.

IV. Sub-Carrier Modulation Demodulation

A significant disadvantage of the double tuned discriminator is that the two tuned amplifiers have to be tuned to frequencies which are identically away from the resonant frequency. Considerable reduction in distortion is observed if one considers a staggered tuned circuit. The above concept can be extended for a staggered tuned circuit also, where the DC gain and the linear zone for FM-AM conversion is further increased by the staggered tuned circuit. The subcarrier signal is represented by

$\psi(t) = m \times \sin\left[\omega_c t + \sum_i \psi_i(t)\right]$, '$\psi_i(t)$' are the subcarrier signals.
Excellent demodulation capabilities are observed.
Let us consider input FM two-tone signal to staggered tuned resonant circuit as

$$\psi(t) = m_i \sin(\omega_s t + m_{i1} \sin(\omega_m t))$$

Now the transfer functions can be written as

$$TFH1(t) = \frac{1}{\sqrt{1 + \left[Q\left(\frac{\omega_{cH1} + d\psi(t)/dt}{\omega_{01}} - \frac{\omega_{01}}{\omega_{cH1} + d\psi(t)/dt}\right)\right]^2}} \quad (17)$$

$$TFH2(t) = \frac{1}{\sqrt{1 + \left[Q\left(\frac{\omega_{cH2} + d\psi(t)/dt}{\omega_{02}} - \frac{\omega_{02}}{\omega_{cH2} + d\psi(t)/dt}\right)\right]^2}} \quad (18)$$

$$TFH(t) = TFH1(t) \times TFH2(t) \quad (19)$$

$$TFL1(t) = \frac{1}{\sqrt{1 + \left[Q\left(\frac{\omega_{cL1} + d\psi(t)/dt}{\omega_{01}} - \frac{\omega_{01}}{\omega_{cL1} + d\psi(t)/dt}\right)\right]^2}} \quad (20)$$

$$TFL2(t) = \frac{1}{\sqrt{1 + \left[Q\left(\frac{\omega_{cL2} + d\psi(t)/dt}{\omega_{02}} - \frac{\omega_{02}}{\omega_{cL2} + d\psi(t)/dt}\right)\right]^2}} \quad (21)$$

$$TFL(t) = TFL1(t) \times TFL2(t) \quad (22)$$

$F\ (t) = TFH(t) - TFL(t)$. The responses are shown in Fig. 8 and the demodulated output in Fig. 9.

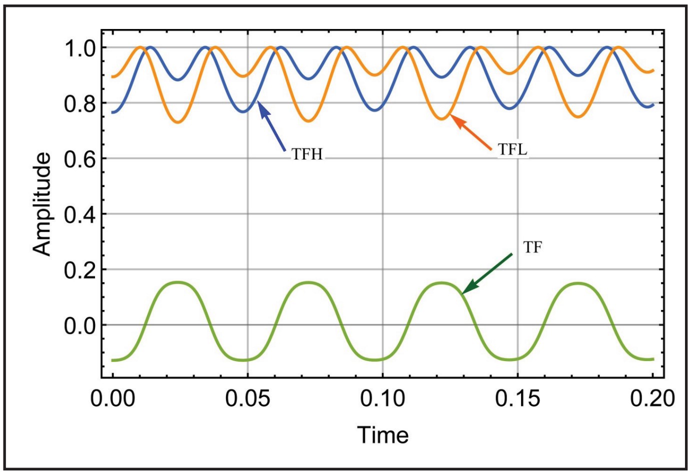

Fig. 8: Response from the Double Tuned Circuit

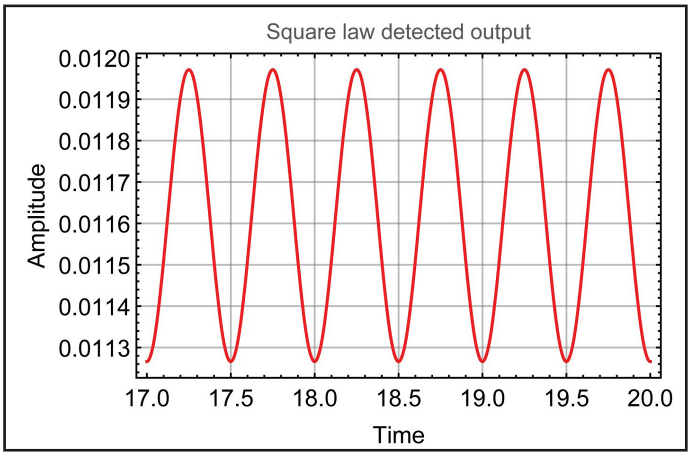

Fig. 9: Square Law Detected Output From the Double Tuned Circuit

V. Conclusion

It is concluded that staggered tuned demodulator circuit has an advantage over single tuned demodulator circuit. Stagger tuned demodulator circuit also increases the useful linear range. Moreover, spurious locking can be eliminated using a compound loop PLL.

VI. Acknowledgment

The authors are thankful to Mr B Guha Mallick, the Chairman of the Supreme Knowledge Group of Institutions for providing facility to work.

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