ADVANCED RESEARCH & INNOVATIONS
International Journal of Advanced Research & Innovations
Volume 15, Issue-2
September, 2025, P75
ISSN: 2319-9253

**International Conference on Emerging Trends in Engineering, Technology & Management (ICETM-2025)**
Conducted by *Viswam Engineering College (UGC—Autonomous Institution)* held on 11th & 12th, April- 2025

# ANALOG FILTERS AT THE RF FRONT OF THE WIRELESS RECEIVERS DISTORT THE SIGNALS

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**ABSTRACT:** This study investigates how analog filters affect signal quality in wireless receivers, highlighting the challenges associated with evaluating their effects and the need for accurate models. The proposed idea implies a Wiener model that combines linear and non-linear elements to rectify the distortions. Improving the accuracy of the models can help decrease demodulation errors, leading to more reliable wireless communication systems. An investigation into analog filters conducted by the study may serve as the basis for a more thorough comprehension of circuit components and signal integrity within wireless systems.

***Keywords:*** *RFFE Circuit, Nonliner Distortion, Analog Filter etc.*

## 1. INTRODUCTION

The arrival of wireless technology has had a substantial effect on modern society. Radio frequency modulation technology functions over a spectrum spanning 20 kHz to 300 GHz to facilitate the establishment of swift and reliable communication systems. Several plans come with different speed options, yet all necessitate robust signals. Weak signal strength leads to incorrect demodulation, ultimately resulting in a decline in the communication quality. The RF front-end circuit is a critical component in wireless systems and is positioned between the antenna and the base-band circuitry. The transmitters' RFFE ensures the delivery of strong, high-quality signals, meeting both range and regulatory requirements. The receiver's RFFE handles the signal processing tasks, including amplification, filtering, and signal recovery to improve the quality and reduce errors. Typically, RFFE circuits consist of power amplifiers, filters, oscillators, and mixers.

Previous research has provided approximate SER expressions for the modulation schemes, assuming ideal signals and Gaussian noise models. In reality, radio signals are susceptible to degradation and distortion as they travel through wireless channels, which is often caused by multipath fading, Doppler effects, and interference. RFFE circuits in receivers are designed to counteract these effects.

Research has pointed out the impact of non-linear behavior and unwanted signals in electrical components such as power amplifiers, mixers, and oscillators on the performance of the communication system. The functions of these components have been explored, and the influence of RFFE circuits on signals has been investigated as well. Real-time scenarios may benefit from adaptive filtering methods that can rectify signal distortion in real-time. A mathematical investigation into the relationship between RFFE circuits and communication performance, particularly in terms of demodulation error probability, has not been fully conducted. Developing a mathematical model of RFFE circuits is crucial for precisely calculating the SER of the modulation schemes.

Despite the limitations of previous research, it is crucial to investigate how analog filters affect system performance under conditions of both linear and non-linear distortions. This study offers a comprehensive

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assessment of the signal distortions caused by analog filters and proposes using a Wiener model to establish a detailed mathematical framework. The primary goal of this model is to offer a deeper understanding of how the RFFE circuit impacts demodulation errors in wireless receivers through analytical comprehension.

## 2. LINEAR ANALOG FILTER MODEL

This section covers the analog filters within a wireless RFFE circuit, specifically highlighting the functions of the BPFs and LPFs in a direct conversion receiver. The discussion then shifts to the impulse response, frequency response, and system function of the LTI filters in relation to the wireless signals, after which the design process and the determination of the transfer function H(s) are explored.

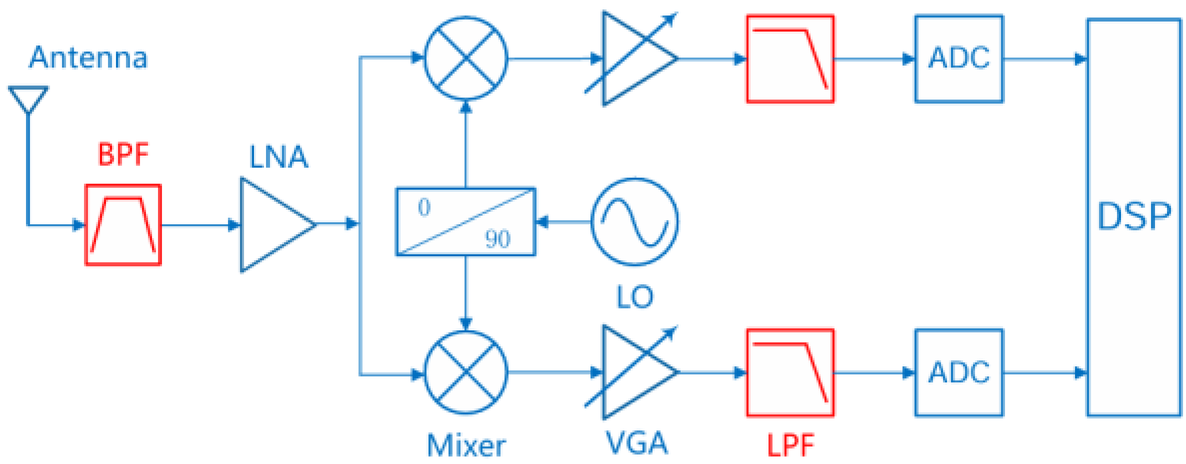


**Fig. 1.** A diagram showing the architecture of the direct conversion.

Fig. 1. The direct conversion architecture directly converts high-frequency radio signals into the base-band with minimal intermediate processing, which is also known as the zero-IF architecture. The process of receiving a signal starts with capturing the signal from the antenna, which is then filtered using band-pass filtering to select specific frequencies and subsequently amplified by a low-noise amplifier (LNA). The Mixing Stage, employing a Local Oscillator, converts the RF signal down to produce I and Q components. The subsequent low-pass filtering process eliminates high frequencies, and the resulting signal is then converted into digital data using an analog-to-digital converter (ADC). This streamlined design offers a cost-effective and energy-efficient approach to signal processing, guaranteeing signal quality throughout the receiver system.

## 3. ANALOG FILTERS RESULT IN SIGNAL DISTORTION.

This section discusses signal distortions produced by analog filters, including amplitude and phase distortions arising from non-ideal frequency responses, non-linear distortions resulting from high-level inputs or parasitic components, and effects linked to noise, such as thermal noise, flicker noise, electromagnetic interference, and parasitic effects.

**1. Significant changes in signal strength with amplitude distortion.**

Parts of a signal are modified at unequal levels, leading to amplitude distortion, which impacts the volume balance, similar to varying the levels of different instruments in music. Signal distortion can alter its original amplitude, changing its shape, a phenomenon similar to when colors in an image are not accurately

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RESEARCH & INNOVATIONS
International Journal of Advanced Research & Innovations
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rendered. The inclusion of spices in the cooking process can have a considerable impact on the overall taste. In the field of signal processing, over-amplification or excessive attenuation leads to distortion of audio signals. Reception errors in wireless systems can stem from amplitude distortion, ultimately impacting the interpretation of data. Ensuring issue resolution necessitates a grasp of, a measurement of, and compensation for distortion through the implementation of enhanced filter designs, thus ensuring signal integrity. Improving these distortions is crucial for bettering wireless communication quality. The summary offers a clear and straightforward overview of amplitude distortion, detailing its effects and providing techniques to mitigate it.

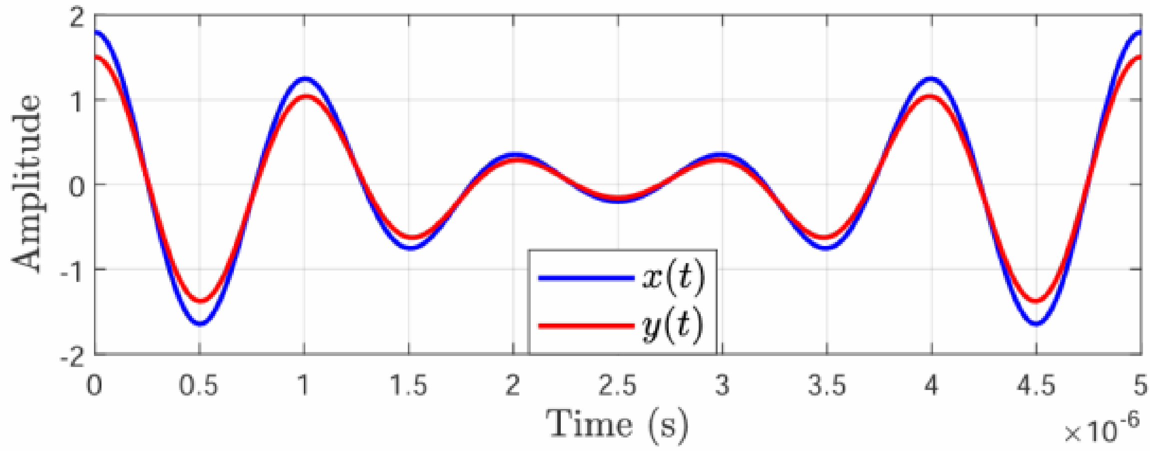


**Fig.A**. Analysis of the comparative effects of amplitude distortion on x(t) and y(t) is conducted.

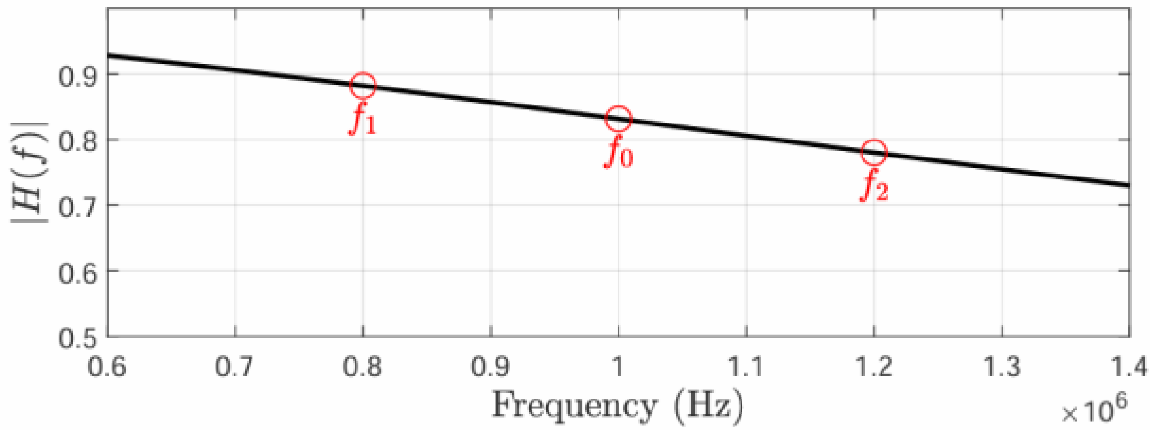


**Fig. B.** The analysis of the filter's magnitude response is performed using H(s).

**Fig.2.** Amplitude Modulation

**Fig.2a.** Fig.2. The focus of the comparison between x(t) and y(t) is on amplitude distortion, with y(t) deviating from x(t) due to the filter's effects. The uneven amplification or damping of the frequency components in a signal leads to amplitude distortion, resulting in an altered output waveform, y(t). Accurate amplitude profiling is crucial for high-precision signal demodulation because any inconsistencies can lead to decoding errors. This distortion can be likened to fine-tuning the volume of specific instruments within a Symphony recording. Identifying these disparities enables the refinement of filter designs through predictions and solutions to amplitude response discrepancies, ultimately improving signal clarity.

Fig. 2b.The filter's magnitude response, as indicated by its transfer function H(s), shows how a signal's different frequencies are either amplified or reduced. H(s) is a key element in comprehending the filter's

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behavior within the frequency spectrum. Converting H(s) to |H(f)| provides insight into the filter's effect on different frequencies, with s replaced by j2πf. The resulting |H(f)| response shows the gain or loss at each frequency after filtering. A good response maintains a consistent volume level, akin to a professional sound engineer, and even minor fluctuations can significantly affect the overall outcome. In real-world scenarios, the response fine-tunes each frequency much like one would add individual spices to a dish. In radio communication, getting the amplitude right is essential; unbalanced responses can damage the integrity of the data signal. Engineers strive to attain flat frequency responses to guarantee equal treatment of all frequencies, thus preserving the signal integrity. The magnitude response helps the designer design efficient filters, thus optimizing the performance in applications such as wireless communication.

**2. Phase distortion alters the waveforms for unique sound timbres.**

Disruptions in timing within a system lead to Phase Distortion when signal components are altered at different intervals, affecting signal clarity and overall reliability. The use of different frequency filters can lead to signal delays, causing corrupted signals and communication failures in systems that depend on the phase information. The concept can be more clearly grasped by making analogies to orchestras and video streaming services. Phase distortion issues occur due to non-linear filter responses and real-world constraints. Linear phase filters and digital signal processing methods were used for the mitigation purposes. Ensuring signal quality in various applications hinges on understanding and addressing phase distortion issues.

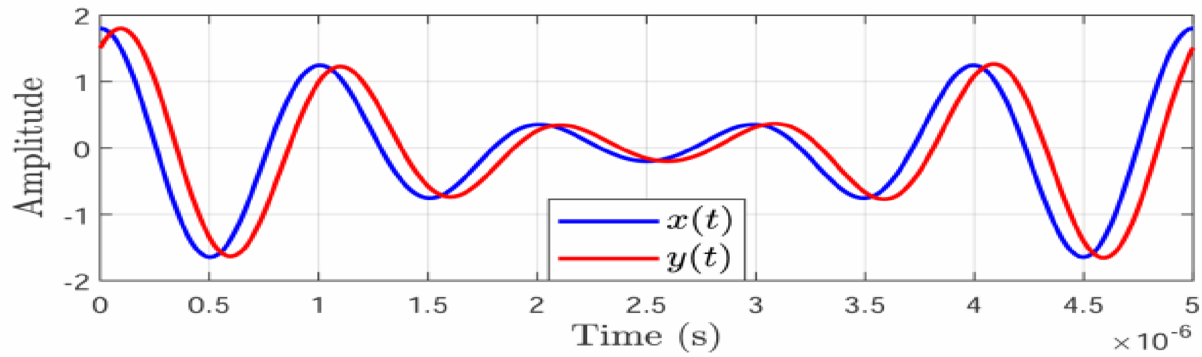


**Fig. a.. Compiling a comparison of x(t) and y(t) solely for phase distortion.**

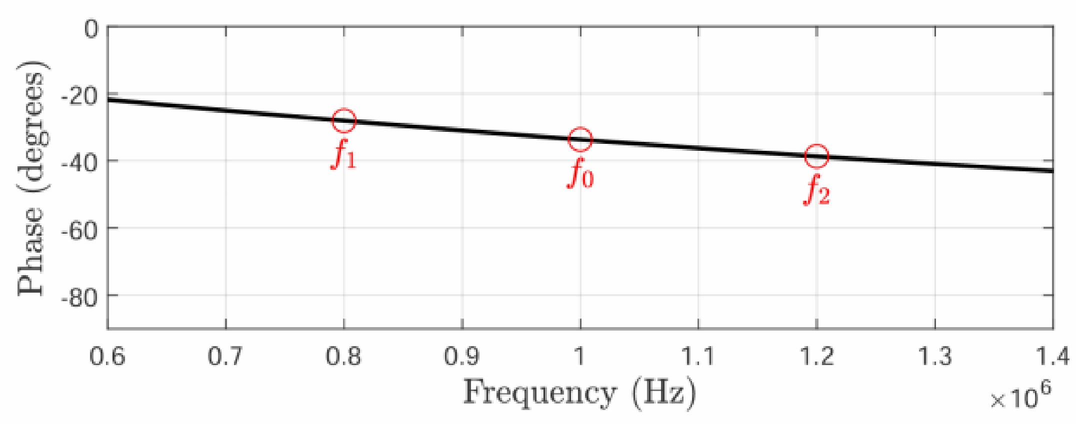


**Fig b. The phase response of the filter corresponds to H(s).**

**Fig.3. Phase Distortion**

Fig.3a. In the context of phase distortion, the function x(t) denotes the original input signal, whereas the function y(t) denotes the output signal following an analog filter. Disruptions to the aligned timing of the frequency components are caused by uneven timing changes resulting from the filter. In an ideal situation,

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the filter's phase response should demonstrate linearity, leading to a uniform delay across all frequencies in order to maintain the signal's original shape. The non-linear response caused by the phase distortion modifies the signal's structure, leading to disparate delays for distinct frequencies. The disparity leads to varying peaks and dips in the output waveform, ultimately affecting the signal clarity. The comparison could be likened to a song that is out of sync or an audio and video feed that is delayed. The accuracy of signal decoding in communication systems depends significantly on the precise synchronization of the frequency components. Maintaining phase integrity is essential for obtaining an accurate signal representation when comparing the original, undistorted input signal x(t) against the linear phase characteristics with the distorted output signal y(t).

Fig.3b. The filter transfer function, denoted by H(s), illustrates the manner in which it alters the signal components with respect to both the amplitude and the phase. The phase response determines the time delay experienced by each frequency, which can impact the signal timing and lead to distortion. Varying time delays across different frequencies can arise from non-linear phase responses, potentially leading to waveform distortions. Successful signal recovery in communication systems relies on understanding the phase response of a filter.

**c. Signal alteration with non-linear distortion.**

Distortion in circuit outputs occurs due to uneven signal processing in analog filter circuits, leading to the presence of extra components. Inter-modulation arises from exceeding the component limitations, material imperfections, and unforeseen effects that cause the issue. When a distortion exceeds its capacity, it starts to behave in a way similar to clay that has lost its original form, resulting in signal degradation and communication system failures. For wireless communication to be stable, engineers need to meticulously plan and build circuits to reduce signal degradation.

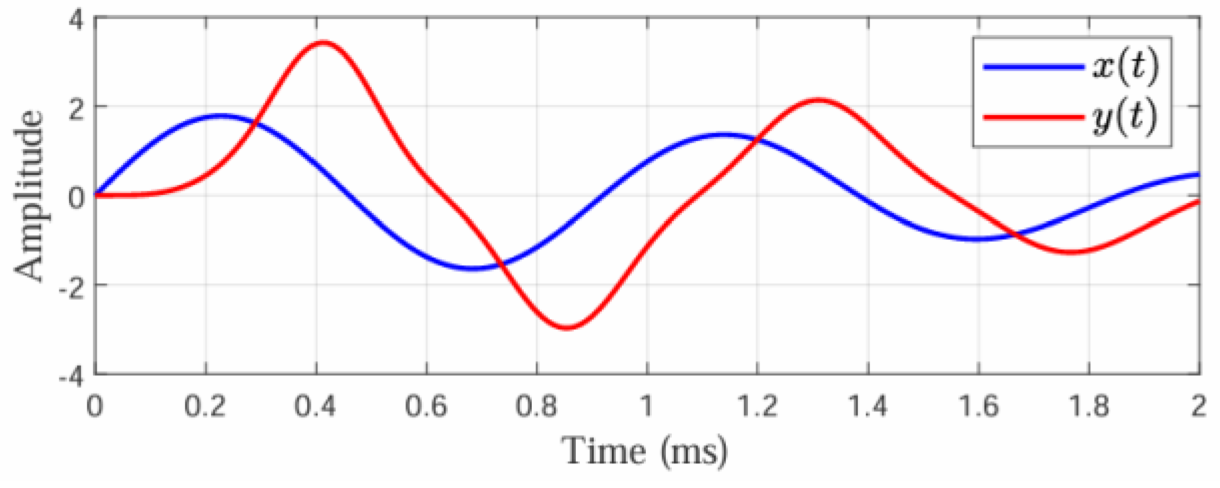


**Fig.a. X(t) and y(t) are compared in the time domain.**

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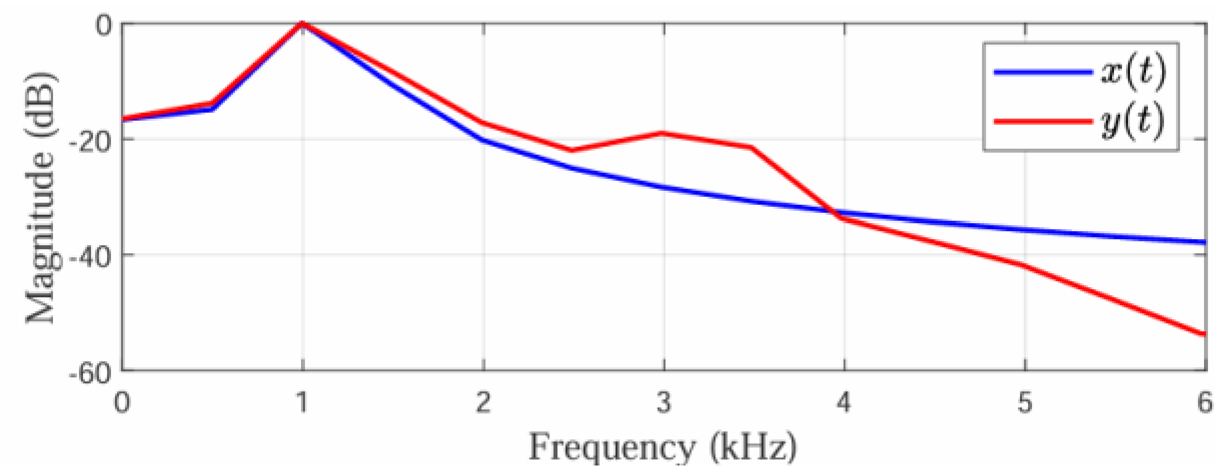


**Fig.b. Comparing the time-dependent functions x(t) and y(t) in the frequency domain.**

**Fig.4. Non-linear Distortion**

Fig. 4a. The input signal x(t) is then passed through an analog filter, producing the output signal y(t). Understanding the basis of this comparison is essential for understanding the signal changes. The signal x(t) originates at the source and carries vital modulated information, specifically designed with unique characteristics for maximum transmission efficiency. In contrast, y(t) is the resulting signal following post-filter processing, which can involve modifications such as noise reduction and boosting. The filter's impulse response changes the output signal y(t) from the original input signal x(t), causing distortions like amplitude and phase fluctuations. These distortions can modify the signal's appearance, possibly having non-linear effects that lead to extra components of y(t) and a more intricate waveform. The presence of noise significantly affects y(t), degrading its signal quality compared to the noise-free signal x(t). Examples of these relationships can be observed in comparable machines, like photocopiers. A comparison of x(t) and y(t) in the time domain graphically demonstrates the influence of the filters on the signal quality within the wireless communication systems.

Fig. 4b. The frequency domain comparison of the input and output signals is demonstrated by x(t) and y(t) being transformed into their respective frequency domains, where an analog filter's effects on the frequency components become apparent. The frequency domain representations of x(t) and y(t) are shown as X(f) and Y(f) spectra, highlighting the energy distribution and filter-induced alterations affecting the amplitude and phase. Non-linear behavior can inject new frequency components, with noise sources incorporating random fluctuations. Comparing the frequency domain representations X(f) and Y(f) facilitates the evaluation of the amplitude and phase variations that are crucial for signal integrity in wireless communication systems.

**Table 1.** Analog filters are susceptible to several forms of noise.

| Noise Type | Origin | Model | Measurement Method |
|---|---|---|---|
| Thermal Noise | Random electron motion | $V_{\text{rms}} = \sqrt{4kTRB}$ | Spectrum analyzer |
| Flicker Noise | Material imperfections | $S(f) \propto 1/f^{\alpha}$ | Low-frequency spectrum analysis |
| Electromagnetic Interference | External Electromagnetic fields | No universal model | Spectrum analyzer, shielding methods |
| Parasitic Effects | Unintended inductance/capacitance | $Z(f) = \frac{1}{j2\pi fC} + j2\pi fL$ | Impedance analyzer |

Signals are disrupted by various types of noise in analog filters, including thermal (Johnson-Nyquist) and flicker (1/f) noise, as well as electromagnetic interference (EMI) and parasitic effects, which ultimately impair communication clarity. Thermal noise originates from the movement of electrons in the resistive components, resulting in a persistent background noise similar to a constant hiss caused by the heating of the

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components. Flicker noise is more pronounced at lower frequencies and originates from material flaws that produce a fading humming sound at higher pitches. Electromagnetic interference (EMI) is caused by external electromagnetic fields, which can be likened to several conversations overlapping in a crowded space. Unintended properties in the design or construction of components can have a significant impact on the circuit behavior, similar to unexpected additives in a recipe that alter its outcome. It is essential for engineers to comprehend different noise types in order to develop more effective filters, thereby improving the signal quality.

## 4. RESULT AND SIMULATION

A 2nd-order Butterworth band-pass filter was simulated to test its performance in predicting the output, with the filter operating at a frequency of 1 GHz, having a bandwidth of 100 MHz, and being of order 2. A Band-Pass Filter was designed and implemented using MATLAB software, and its performance matched the exact specifications of the intended ideal model. The input signal is a 64-QAM type transmitted over a 1-GHz carrier frequency. The non-linear distortion was introduced using a quadratic term. The parameters for the Wiener model were determined using the polit method. The model produces an exact forecast of the filter's output, with a mean square error of 0.002455. The model's reliability is preserved even when the coefficients vary due to the trigonometric functions' similarities. The Wiener model, when combined with a filter, enhances signal processing by addressing both linear and non-linear distortions, ultimately benefiting wireless communication systems.

**Fig. 5. A comparison is drawn between x(t) and y(t).**

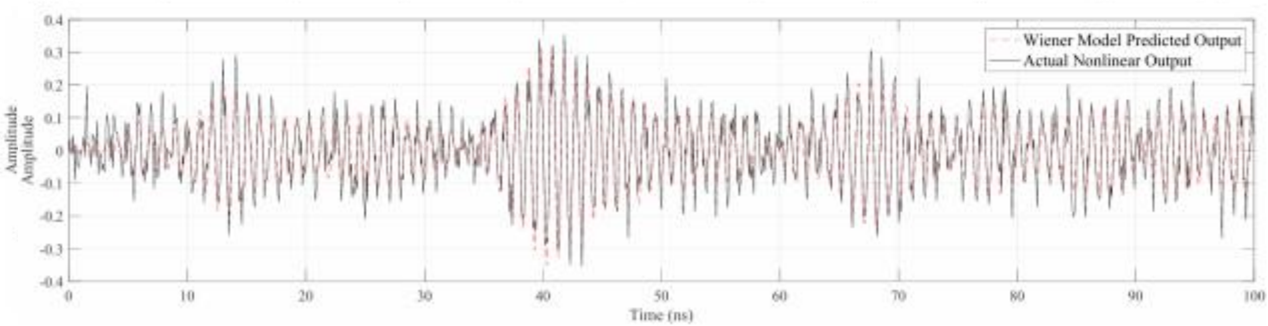


**Fig.6. Comparing the actual and predicted values over time.**

## 5. CONCLUSION

This paper concentrates on the challenges associated with modeling in RFFE circuits, emphasizing the impact of analog filters on the received signals. Filters result in both linear and non-linear distortions, which the Wiener model endeavors to consider in order to make accurate predictions. Analog filter distortions negatively affect the demodulation quality, yet the Wiener model offers effective compensation capabilities. Experimental simulations have verified the accuracy of the model, which is crucial for minimizing demodulation errors and improving wireless systems via a more in-depth comprehension of the circuitry and innovative design guidelines.

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