DESIGN, IMPLEMENTATION AND COMPARISON OF DEMODULATION METHODS IN AM AND FM

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Abstract

Modulation and demodulation hold dominant positions in communication. Communication quality heavily relies on the performance of the detector. A simple and efficient detector can improve the communication quality and reduce the cost.

This thesis reveals the pros and cons of five demodulation methods for Amplitude Modulated (AM) signal and four demodulation methods for Frequency Modulated (FM) signal. Two experimental systems are designed and implemented to finish this task.

This thesis provides the researchers an easier reference of demodulation methods with tables listing their pros and cons.

Index Terms— Demodulation, Envelope, Coherent, Square-law, FM to AM, Zero-crossing, Quadrature.

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Chapter 1

Introduction

Carrier is modulated during amplitude modulation and frequency modulation. After modulation, the modulated wave is sent through medium. The message signal is restored during demodulation at the receiving end.

The output of an ideal Amplitude Modulated (AM) signal detector is proportional to the amplitude of the envelope of the AM signal. In AM signal demodulation, five methods are commonly used:

- Envelope demodulation using filter
- Envelope demodulation with Hilbert transform
- Coherent demodulation
- Square-law demodulation
- Quadrature demodulation

An ideal Frequency Modulated (FM) signal discriminator's output is proportional to the instantaneous frequency of the FM signal. In FM signal demodulation, the basic techniques are:

- FM to AM conversion
- Zero-crossing demodulation
- Quadrature demodulation

1.1 Aim

The aim of this thesis is to compare different demodulation methods of AM and FM respectively. The performances of demodulation methods are compared through simulations.

1.2 Motivation

Many demodulation methods are found dispersed in literature. It is inconvenient to consult huge amount of reference materials, which motivated the authors to analyse and compare these methods in order to reveal their pros and cons in one composition.

Two experimental systems are built to compare the performances of AM signal demodulation methods and FM signal demodulation methods. The design charts are in Chapter 3.

1.3 Division of work

Literature consulting and simulation design are by Sen Lin and Gaojun Chen.

MATLAB simulation and thesis writing are by Gaojun Chen and Sen Lin.

1.4 Outline

- Chapter 2 describes principles of the demodulation methods used in the thesis.
- Chapter 3 covers a detailed description of the simulation.
- Chapter 4 analyses results.
- Chapter 5 draws the conclusion.

Chapter 2

Background

This chapter describes the demodulation methods of AM signal and FM signal.

2.1 AM Signal Demodulation

In AM system, five signals are involved.

1. Message signal $m(t)$, which is also known as modulating signal or modulating wave. The message signal usually contains audio signal [1].

2. Carrier $x_c(t)$

$$x_c(t) = A_c \cdot cos(2\pi \cdot f_c \cdot t) \tag{2.1}$$

where A_c is the carrier amplitude and f_c is the carrier frequency.

3. Local carrier $x_{lc}(t)$, which is generated by the demodulator

$$x_{lc}(t) = cos(2\pi \cdot f_c \cdot t) \tag{2.2}$$

3. Double Side Band Transmitted Carrier (DSB TC) AM signal $s_{tc}(t)$

$$s_{tc}(t) = (m(t) + A_c) \cdot cos(2\pi \cdot f_c \cdot t) \tag{2.3}$$

where $m(t)$ is the message signal.

4. Double Side Band Suppressed Carrier (DSB SC) AM signal $s_{sc}(t)$

$$s_{sc}(t) = m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t) \tag{2.4}$$

5. Single Side Band (SSB) AM signal. There are two kinds of SSB AM signal: upper side band transmitted AM signal $s_u(t)$ and lower side band transmitted AM signal $s_l(t)$:

$$s_u(t) = \frac{1}{2}m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t) + \frac{1}{2}\hat{m}(t) \cdot A_c \cdot sin(2\pi \cdot f_c \cdot t) \tag{2.5}$$

$$s_l(t) = \frac{1}{2}m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t) - \frac{1}{2}\hat{m}(t) \cdot A_c \cdot sin(2\pi \cdot f_c \cdot t) \quad (2.6)$$

where $\hat{m}(t)$ is the Hilbert transform of the message signal $m(t)$.

2.1.1 Envelope Demodulation

Envelope demodulation is the method to recover a message signal by finding the envelope of the DSB TC AM signal.

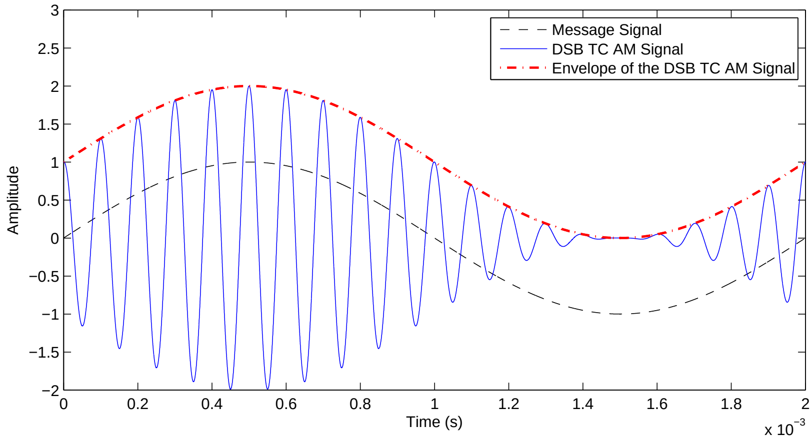

Figure 2.1: Envelope of double side band transmitted carrier AM signal

Figure 2.1 illustrates the relationships among the message signal, the DSB TC AM signal and the envelope of the DSB TC AM signal. A 10000 Hz sinusoidal wave is used as carrier and the message signal is a 500 Hz sinusoidal wave. The carrier is modulated by the message signal (black dash curve) into the DSB TC AM signal (blue). It shows that the envelope (in red) encloses the DSB TC AM signal. Thus, the message signal is extracted from the envelope of the DSB TC AM signal.

Two processes are used in finding the envelope of the DSB TC AM signal: envelope demodulation using filter and envelope demodulation with Hilbert transform.

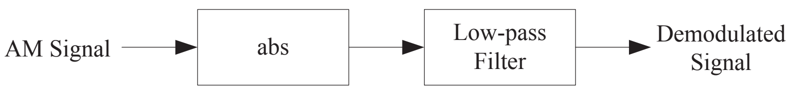

Figure 2.2: Block diagram of envelope demodulation using filter

Figure 2.2 shows the block diagram of the envelope demodulation using filter. The absolute value of the input signal is calculated and then fed to the low-pass filter. The output of the low-pass filter is the demodulated signal.

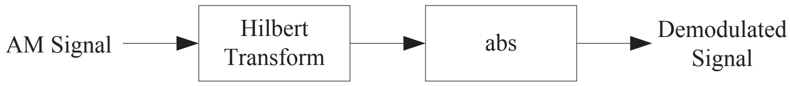

Figure 2.3: Block diagram of envelope demodulation with Hilbert transform

In Figure 2.3, the output of the Hilbert transform module is a complex-valued signal, the real part of the complex-valued signal is the input signal and the imaginary part of the complex-valued signal is the Hilbert transform of the input signal. After the DSB TC AM signal passing through the Hilbert transform module, the absolute value of the complex-valued signal is taken as the demodulated signal.

For any real-valued bandpass signal $g\left(t\right)$, it holds

$$g_{+}\left(t\right) = g\left(t\right) + j\hat{g}\left(t\right) \tag{2.7}$$

where $g_{+}\left(t\right)$ is the pre-envelope of $g\left(t\right)$, $\hat{g}\left(t\right)$ is the Hilbert transform of $g\left(t\right)$ [2, 3].

Set

$$g\left(t\right) = a\left(t\right) cos\left(2\pi f_c t\right) \tag{2.8}$$

When the carrier frequency of $g\left(t\right)$ is larger than the half bandwidth of $g\left(t\right)$, the Hilbert transform of $g\left(t\right)$ is

$$\hat{g}\left(t\right) = a\left(t\right) sin\left(2\pi f_c t\right) \tag{2.9}$$

[4]

When $g\left(t\right)$ is the DSB TC AM signal, the Hilbert transform of the DSB TC AM signal is

$$\hat{g}\left(t\right) = \left(m\left(t\right) + A_c\right) \cdot sin\left(2\pi \cdot f_c \cdot t\right) \tag{2.10}$$

the absolute value of the pre-envelope $g_{+}\left(t\right)$ is

$$\begin{aligned}
\sqrt{g_{+}\left(t\right)} &= \left[g^2\left(t\right) + \hat{g}^2\left(t\right)\right]^{\frac{1}{2}} \\
&= \Bigg\{ \left(m\left(t\right) + A_c\right)^2 \cdot cos^2\left(2\pi \cdot f_c \cdot t\right) \\
&\quad + \left(m\left(t\right) + A_c\right)^2 \cdot sin^2\left(2\pi \cdot f_c \cdot t\right) \Bigg\}^{\frac{1}{2}}
\end{aligned}$$

$$
\begin{aligned}
&= \left\{ \left(m\left(t\right) + A_c \right)^2 \right. \\
&\quad \left. \cdot \left[cos^2\left(2\pi \cdot f_c \cdot t\right) + sin^2\left(2\pi \cdot f_c \cdot t\right) \right] \right\}^{\frac{1}{2}} \\
&= \left\{ \left(m\left(t\right) + A_c \right)^2 \cdot 1 \right\}^{\frac{1}{2}} \\
&= m\left(t\right) + A_c
\end{aligned}
\tag{2.11}
$$

thus the DSB TC AM signal is demodulated.

2.1.2 Coherent Demodulation

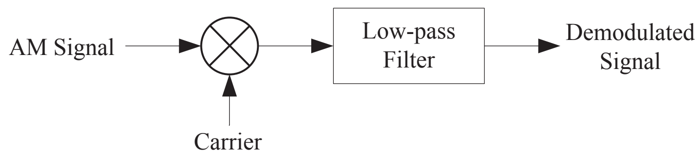

Figure 2.4: Block diagram of coherent demodulation

Figure 2.4 illustrates the block diagram of the coherent demodulation. The input signal is multiplied with the local carrier. After the product passing through the low-pass filter, the demodulated signal is obtained [1].

When the input signal is DSB TC AM signal, the input and carrier are fed to the mixer. Thus the output of the mixer is

$$
\begin{aligned}
& s_{tc}\left(t\right) \cdot x_{lc}\left(t\right) \\
&= \left[\left(m\left(t\right) + A_c \right) \cdot cos\left(2\pi \cdot f_c \cdot t\right) \right] \cdot cos\left(2\pi \cdot f_c \cdot t\right) \\
&= m\left(t\right) \cdot cos\left(2\pi \cdot f_c \cdot t\right) \cdot cos\left(2\pi \cdot f_c \cdot t\right) \\
&\quad + A_c \cdot cos\left(2\pi \cdot f_c \cdot t\right) \cdot cos\left(2\pi \cdot f_c \cdot t\right) \\
&= m\left(t\right) \cdot cos^2\left(2\pi \cdot f_c \cdot t\right) + A_c \cdot cos^2\left(2\pi \cdot f_c \cdot t\right) \\
&= \left[m\left(t\right) + A_c \right] \cdot cos^2\left(2\pi \cdot f_c \cdot t\right) \\
&= \frac{1}{2} \cdot \left[m\left(t\right) + A_c \right] \\
&\quad \cdot \left[cos\left(2\pi \cdot f_c \cdot t + 2\pi \cdot f_c \cdot t\right) + cos\left(2\pi \cdot f_c \cdot t - 2\pi \cdot f_c \cdot t\right) \right] \\
&= \frac{1}{2} \cdot \left[m\left(t\right) + A_c \right] \cdot \left[cos\left(4\pi \cdot f_c \cdot t\right) + 1 \right] \\
&= \frac{1}{2} m\left(t\right) + \frac{1}{2} A_c + \frac{1}{2} \left[m\left(t\right) + A_c \right] cos\left(4\pi \cdot f_c \cdot t\right)
\end{aligned}
\tag{2.12}
$$

In (2.12), the high frequency component is $\frac{1}{2}[m(t) + A_c] \, cos(4\pi \cdot f_c \cdot t)$, which can be eliminated by the low-pass filter. The DC component is $\frac{1}{2}A_c$. And the demodulated signal is $\frac{1}{2}m(t)$, which is proportional to the message signal $m(t)$.

When the input is DSB SC AM signal, the output of the mixer is

$$\begin{aligned}
& s_{sc}(t) \cdot x_{lc}(t) \\
= & [m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t)] \cdot cos(2\pi \cdot f_c \cdot t) \\
= & m(t) \cdot A_c \cdot cos^2(2\pi \cdot f_c \cdot t) \\
= & \frac{1}{2} m(t) \cdot A_c \cdot [cos(4\pi \cdot f_c \cdot t) + 1] \\
= & \frac{1}{2} m(t) \cdot A_c + \frac{1}{2} m(t) \cdot A_c \cdot cos(4\pi \cdot f_c \cdot t) \qquad (2.13)
\end{aligned}$$

In (2.13), the high frequency component is $\frac{1}{2}m(t) \cdot A_c \cdot cos(4\pi \cdot f_c \cdot t)$, which can be rejected with the low-pass filter. The demodulated signal is $\frac{1}{2}m(t) \cdot A_c$, which is proportional to the message signal $m(t)$.

When the input is SSB AM signal, the output of the mixer is

$$\begin{aligned}
& \left[\frac{1}{2} m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t) \pm \frac{1}{2} \hat{m}(t) \cdot A_c \cdot sin(2\pi \cdot f_c \cdot t)\right] \cdot x_{lc}(t) \\
& \left[\frac{1}{2} m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t) \pm \frac{1}{2} \hat{m}(t) \cdot A_c \cdot sin(2\pi \cdot f_c \cdot t)\right] \cdot cos(2\pi \cdot f_c \cdot t) \\
= & \frac{1}{2} m(t) \cdot A_c \cdot cos^2(2\pi \cdot f_c \cdot t) \pm \frac{1}{2} \hat{m}(t) \cdot A_c \cdot sin(2\pi \cdot f_c \cdot t) \, cos(2\pi \cdot f_c \cdot t) \\
= & \frac{1}{4} m(t) \cdot A_c \cdot (cos(4\pi \cdot f_c \cdot t) + 1) \pm \frac{1}{4} \hat{m}(t) \cdot A_c \cdot sin(4\pi \cdot f_c \cdot t) \\
= & \frac{1}{4} m(t) \cdot A_c + \frac{1}{4} m(t) \cdot A_c \cdot cos(4\pi \cdot f_c \cdot t) \\
& \pm \frac{1}{4} \hat{m}(t) \cdot A_c \cdot sin(4\pi \cdot f_c \cdot t) \qquad (2.14)
\end{aligned}$$

In (2.14), the desired demodulated signal is $\frac{1}{4}m(t) \cdot A_c$, which is proportional to the message signal $m(t)$. The high frequency component is $\frac{1}{4}m(t) \cdot A_c \cdot cos(4\pi \cdot f_c \cdot t) \pm \frac{1}{4}\hat{m}(t) \cdot A_c \cdot sin(4\pi \cdot f_c \cdot t)$, which can be removed by the low-pass filter.

If there exists phase differences between the input signal and the local carrier, the phase difference should be taken into consideration. Assume the phase difference is φ, the new local carrier $x_{lcp}(t)$ is defined as:

$$x_{lcp}(t) = cos(2\pi \cdot f_c \cdot t + \varphi) \qquad (2.15)$$

When the input is DSB TC AM signal, the output of the mixer is

$$
\begin{aligned}
& s_{tc}(t) \cdot x_{lcp}(t) \\
= \ & [(m(t) + A_c) \cdot cos(2\pi \cdot f_c \cdot t)] \cdot cos(2\pi \cdot f_c \cdot t + \varphi) \\
= \ & (m(t) + A_c) \cdot [cos(2\pi \cdot f_c \cdot t) \, cos(2\pi \cdot f_c \cdot t + \varphi)] \\
= \ & (m(t) + A_c) \cdot \frac{1}{2} \cdot \Big\{ cos(2\pi \cdot f_c \cdot t + 2\pi \cdot f_c \cdot t + \varphi) \\
& + cos(2\pi \cdot f_c \cdot t - 2\pi \cdot f_c \cdot t - \varphi) \Big\} \\
= \ & (m(t) + A_c) \cdot \frac{1}{2} \cdot [cos(4\pi \cdot f_c \cdot t + \varphi) + cos(\varphi)] \\
= \ & \frac{1}{2} \cdot (m(t) + A_c) \cdot cos(4\pi \cdot f_c \cdot t + \varphi) \\
& + \frac{1}{2} \cdot (m(t) + A_c) \cdot cos(\varphi) \qquad (2.16)
\end{aligned}
$$

In (2.16), the high frequency component is $\frac{1}{2} \cdot (m(t) + A_c) \cdot cos(4\pi \cdot f_c \cdot t + \varphi)$, which can be rejected by the low-pass filter. The desired demodulated signal $\frac{1}{2} \cdot (m(t) + A_c)$ is scaled by an cosine term $cos(\varphi)$, which is determined by the phase difference φ. When the phase difference φ is zero, the demodulated signal has the maximum value, when the phase difference φ is ± 90 degree, the demodulated signal is zero.

When the input is DSB SC AM signal, the output of the mixer is

$$
\begin{aligned}
& s_{sc}(t) \cdot x_{lcp}(t) \\
= \ & [m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t)] \, cos(2\pi \cdot f_c \cdot t + \varphi) \\
= \ & m(t) \cdot A_c \cdot [cos(2\pi \cdot f_c \cdot t) \cdot cos(2\pi \cdot f_c \cdot t + \varphi)] \\
= \ & m(t) \cdot A_c \cdot \frac{1}{2} \cdot [cos(4\pi \cdot f_c \cdot t + \varphi) + cos(\varphi)] \\
= \ & \frac{1}{2} \cdot m(t) \cdot A_c \cdot cos(4\pi \cdot f_c \cdot t + \varphi) + \frac{1}{2} \cdot m(t) \cdot A_c \cdot cos(\varphi) \qquad (2.17)
\end{aligned}
$$

In (2.17), the high frequency component $\frac{1}{2} \cdot m(t) \cdot A_c \cdot cos(4\pi \cdot f_c \cdot t + \varphi)$ can be removed by the low-pass filter. The desired demodulated signal $\frac{1}{2} \cdot m(t) \cdot A_c$ is also scaled by the cosine term $cos(\varphi)$.

When the input is SSB AM signal, the output of the mixer is

$$
\begin{aligned}
& \left[\frac{1}{2} m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t) \pm \frac{1}{2} \hat{m}(t) \cdot A_c \cdot sin(2\pi \cdot f_c \cdot t)\right] \\
& \cdot cos(2\pi \cdot f_c \cdot t + \varphi) \\
= \ & \frac{1}{2} m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t) \cdot cos(2\pi \cdot f_c \cdot t + \varphi)
\end{aligned}
$$

$$
\begin{aligned}
& \pm\frac{1}{2}\hat{m}(t)\cdot A_c\cdot sin(2\pi\cdot f_c\cdot t)\cdot cos(2\pi\cdot f_c\cdot t+\varphi) \\
= & \frac{1}{2}m(t)\cdot A_c\cdot\frac{1}{2}\cdot[cos(4\pi\cdot f_c\cdot t+\varphi)+cos(\varphi)] \\
& \pm\frac{1}{2}\hat{m}(t)\cdot A_c\cdot\frac{1}{2}\cdot[sin(4\pi\cdot f_c\cdot t+\varphi)-sin(\varphi)] \\
= & \frac{1}{4}m(t)\cdot A_c\cdot cos(4\pi\cdot f_c\cdot t+\varphi)+\frac{1}{4}m(t)\cdot A_c\cdot cos(\varphi) \\
& \pm\left[\frac{1}{4}\hat{m}(t)\cdot A_c\cdot sin(4\pi\cdot f_c\cdot t+\varphi)-\frac{1}{4}\hat{m}(t)\cdot A_c\cdot sin(\varphi)\right]
\end{aligned}
\tag{2.18}
$$

In (2.18), the high frequency component $\frac{1}{4}m(t)\cdot A_c\cdot cos(4\pi\cdot f_c\cdot t+\varphi)$ and $\frac{1}{4}\hat{m}(t)\cdot A_c\cdot sin(4\pi\cdot f_c\cdot t+\varphi)$ are eliminated after low-pass filtering. The filter output is $\frac{1}{4}m(t)\cdot A_c\cdot cos(\varphi)\mp\frac{1}{4}\hat{m}(t)\cdot A_c\cdot sin(\varphi)$, which is distorted.

By using the coherent demodulation to demodulate DSB TC AM signal and DSB SC AM signal, the existing phase difference between the input signal and the local carrier leads to the demodulated signal scaled by a cosine term. And for demodulating SSB AM signal, the existing phase difference between the input signal and the local carrier leads to the demodulated signal distorted.

2.1.3 Square-Law Demodulation

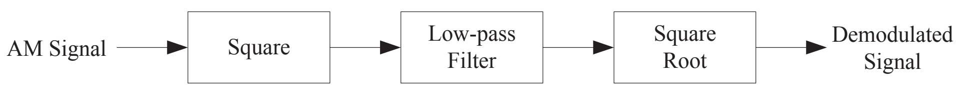

Figure 2.5: Block diagram of square-law demodulation

Figure 2.5 shows the block diagram of the square-law demodulation. In this method, the input DSB TC AM signal is squared before low-passing filtering. After removing the high frequency component, the square root of the filter output is taken. The output of the square root module is the demodulated signal. The output of the square module is:

$$
\begin{aligned}
& (s_{tc}(t)\cdot x_c(t))^2 \\
= & [(m(t)+A_c)\cdot cos(2\pi\cdot f_c\cdot t)]^2 \\
= & (m(t)+A_c)^2\cdot cos^2(2\pi\cdot f_c\cdot t) \\
= & \frac{1}{2}(m(t)+A_c)^2\cdot[cos(4\pi\cdot f_c\cdot t)+1] \\
= & \frac{1}{2}(m(t)+A_c)^2+\frac{1}{2}(m(t)+A_c)^2\cdot cos(4\pi\cdot f_c\cdot t)
\end{aligned}
\tag{2.19}
$$

In the (2.19), the high frequency component is $\frac{1}{2}(m(t)+A_c)^2\cdot cos(4\pi\cdot f_c\cdot t)$, which can be removed by the low-pass filter. After low-pass filtering, only the low frequency component $\frac{1}{2}\left(m\left(t\right)+A_c\right)^2$ is preserved. The square root of the low frequency component is

$$\sqrt{\frac{1}{2}\left(m\left(t\right)+A_c\right)^2}=\frac{\sqrt{2}}{2}\left(m\left(t\right)+A_c\right) \tag{2.20}$$

The result is proportional to the message signal.

2.1.4 Quadrature Demodulation in AM System

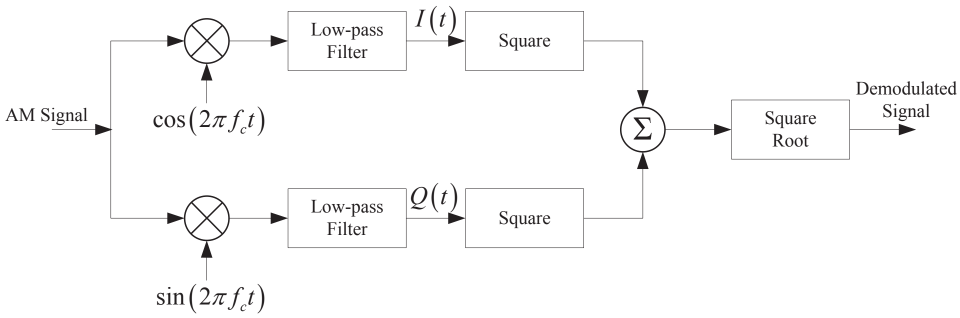

Figure 2.6: Block diagram of quadrature demodulation for AM signal

Figure 2.6 is the block diagram of the quadrature demodulation. In this method, the input DSB TC AM signal is multiplied with $cos\left(2\pi\cdot f_c\cdot t\right)$ and $sin\left(2\pi\cdot f_c\cdot t\right)$ respectively. After low-pass filtering, the input signal is divided into two components, in-phase component $I\left(t\right)$ and quadrature component $Q\left(t\right)$.

$I\left(t\right)$ and $Q\left(t\right)$ contain the amplitude information and the phase information of the input signal [1, 5, 6].

The message signal can be recovered from the norm of $I\left(t\right)$ and $Q\left(t\right)$.

When there is no phase difference between the input signal and the local carrier, the analysis is the same as in the section 2.1.2.

While taking the phase difference φ into consideration, two local carriers become $cos\left(2\pi\cdot f_c\cdot t+\varphi\right)$ and $sin\left(2\pi\cdot f_c\cdot t+\varphi\right)$.

When the input is DSB TC AM signal, the output of the upper mixer is

$$\begin{aligned}
& s_{tc}\left(t\right)\cdot cos\left(2\pi\cdot f_c\cdot t+\varphi\right)\\
= & \left[\left(m\left(t\right)+A_c\right)\cdot cos\left(2\pi\cdot f_c\cdot t\right)\right]\cdot cos\left(2\pi\cdot f_c\cdot t+\varphi\right)\\
= & \left(m\left(t\right)+A_c\right)\cdot\left[cos\left(2\pi\cdot f_c\cdot t\right)cos\left(2\pi\cdot f_c\cdot t+\varphi\right)\right]\\
= & \left(m\left(t\right)+A_c\right)\cdot\frac{1}{2}\cdot\left\{cos\left(2\pi\cdot f_c\cdot t+2\pi\cdot f_c\cdot t+\varphi\right)\right.\\
& \left.+cos\left(2\pi\cdot f_c\cdot t-2\pi\cdot f_c\cdot t-\varphi\right)\right\}
\end{aligned}$$

$$
\begin{aligned}
&= (m(t)+A_c)\cdot\frac{1}{2}\cdot[cos(4\pi\cdot f_c\cdot t+\varphi)+cos(\varphi)] \\
&= \frac{1}{2}\cdot(m(t)+A_c)\cdot cos(4\pi\cdot f_c\cdot t+\varphi)+\frac{1}{2}\cdot(m(t)+A_c)\cdot cos(\varphi)
\end{aligned}
$$

And the output of the lower mixer is

$$
\begin{aligned}
& s_{tc}(t)\cdot sin(2\pi\cdot f_c\cdot t+\varphi) \\
&= [(m(t)+A_c)\cdot cos(2\pi\cdot f_c\cdot t)]\cdot sin(2\pi\cdot f_c\cdot t+\varphi) \\
&= (m(t)+A_c)\cdot[cos(2\pi\cdot f_c\cdot t)\cdot sin(2\pi\cdot f_c\cdot t+\varphi)] \\
&= (m(t)+A_c)\cdot\frac{1}{2}\cdot\Big\{sin(2\pi\cdot f_c\cdot t+\varphi+2\pi\cdot f_c\cdot t) \\
&\quad +sin(2\pi\cdot f_c\cdot t+\varphi-2\pi\cdot f_c\cdot t)\Big\} \\
&= (m(t)+A_c)\cdot\frac{1}{2}\cdot[sin(4\pi\cdot f_c\cdot t+\varphi)+sin(\varphi)] \\
&= \frac{1}{2}\cdot(m(t)+A_c)\cdot sin(4\pi\cdot f_c\cdot t+\varphi)+\frac{1}{2}\cdot(m(t)+A_c)\cdot sin(\varphi)
\end{aligned}
$$

After filtering, the high frequency component $\frac{1}{2}\cdot(m(t)+A_c)\cdot cos(4\pi\cdot f_c\cdot t+\varphi)$ and $\frac{1}{2}\cdot(m(t)+A_c)\cdot sin(4\pi\cdot f_c\cdot t+\varphi)$ are removed, and $I(t)$ and $Q(t)$ are obtained:

$$
\begin{aligned}
I(t) &= \frac{1}{2}\cdot(m(t)+A_c)\cdot cos(\varphi) \\
Q(t) &= \frac{1}{2}\cdot(m(t)+A_c)\cdot sin(\varphi)
\end{aligned}
$$

The norm of $I(t)$ and $Q(t)$ is

$$
\begin{aligned}
& \left[I^2(t)+Q^2(t)\right]^{\frac{1}{2}} \\
&= \left[\frac{1}{4}\cdot(m(t)+A_c)^2\cdot cos^2(\varphi)+\frac{1}{4}\cdot(m(t)+A_c)^2\cdot sin^2(\varphi)\right]^{\frac{1}{2}} \\
&= \left[\frac{1}{4}\cdot(m(t)+A_c)^2\cdot\left(cos^2(\varphi)+sin^2(\varphi)\right)\right]^{\frac{1}{2}} \\
&= \left[\frac{1}{4}\cdot(m(t)+A_c)^2\cdot 1\right]^{\frac{1}{2}} \\
&= \frac{1}{2}\cdot(m(t)+A_c) \qquad (2.21)
\end{aligned}
$$

For (2.21), the demodulated signal is $\frac{1}{2}\cdot(m(t)+A_c)$, which is proportional to the message signal $m(t)$. And the phase difference φ is eliminated.

While taking the phase difference φ into consideration, and the input is DSB SC AM signal, the output of the upper mixer is

$$\begin{aligned}
& s_{sc}(t) \cdot cos(2\pi \cdot f_c \cdot t + \varphi) \\
= & [m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t)] \cdot cos(2\pi \cdot f_c \cdot t + \varphi) \\
= & m(t) \cdot A_c \cdot [cos(2\pi \cdot f_c \cdot t) \, cos(2\pi \cdot f_c \cdot t + \varphi)] \\
= & m(t) \cdot A_c \cdot \frac{1}{2} \cdot \Big\{ cos(2\pi \cdot f_c \cdot t + 2\pi \cdot f_c \cdot t + \varphi) \\
& + cos(2\pi \cdot f_c \cdot t - 2\pi \cdot f_c \cdot t - \varphi) \Big\} \\
= & m(t) \cdot A_c \cdot \frac{1}{2} \cdot [cos(4\pi \cdot f_c \cdot t + \varphi) + cos(\varphi)] \\
= & \frac{1}{2} \cdot m(t) \cdot A_c \cdot cos(4\pi \cdot f_c \cdot t + \varphi) + \frac{1}{2} \cdot m(t) \cdot A_c \cdot cos(\varphi)
\end{aligned}$$

And the output of the lower mixer is

$$\begin{aligned}
& s_{sc}(t) \cdot sin(2\pi \cdot f_c \cdot t + \varphi) \\
= & [m(t) \cdot A_c \cdot cos(2\pi \cdot f_c \cdot t)] \cdot sin(2\pi \cdot f_c \cdot t + \varphi) \\
= & m(t) \cdot A_c \cdot [cos(2\pi \cdot f_c \cdot t) \cdot sin(2\pi \cdot f_c \cdot t + \varphi)] \\
= & m(t) \cdot A_c \cdot \frac{1}{2} \cdot \Big\{ sin(2\pi \cdot f_c \cdot t + \varphi + 2\pi \cdot f_c \cdot t) \\
& + sin(2\pi \cdot f_c \cdot t + \varphi - 2\pi \cdot f_c \cdot t) \Big\} \\
= & m(t) \cdot A_c \cdot \frac{1}{2} \cdot [sin(4\pi \cdot f_c \cdot t + \varphi) + sin(\varphi)] \\
= & \frac{1}{2} \cdot m(t) \cdot A_c \cdot sin(4\pi \cdot f_c \cdot t + \varphi) + \frac{1}{2} \cdot m(t) \cdot A_c \cdot sin(\varphi)
\end{aligned}$$

After filtering, the high frequency component $\frac{1}{2} \cdot m(t) \cdot A_c \cdot cos(4\pi \cdot f_c \cdot t + \varphi)$ and $\frac{1}{2} \cdot m(t) \cdot A_c \cdot sin(4\pi \cdot f_c \cdot t + \varphi)$ are removed, and $I(t)$ and $Q(t)$ are obtained:

$$\begin{aligned}
I(t) &= \frac{1}{2} \cdot m(t) \cdot A_c \cdot cos(\varphi) \\
Q(t) &= \frac{1}{2} \cdot m(t) \cdot A_c \cdot sin(\varphi)
\end{aligned}$$

The norm of $I(t)$ and $Q(t)$ is

$$
\begin{aligned}
& \left[I^2(t)+Q^2(t)\right]^{\frac{1}{2}} \\
= & \left[\frac{1}{4}\cdot(m(t)\cdot A_c)^2\cdot cos^2(\varphi)+\frac{1}{4}\cdot(m(t)\cdot A_c)^2\cdot sin^2(\varphi)\right]^{\frac{1}{2}} \\
= & \left[\frac{1}{4}\cdot(m(t)\cdot A_c)^2\cdot\left(cos^2(\varphi)+sin^2(\varphi)\right)\right]^{\frac{1}{2}} \\
= & \left[\frac{1}{4}\cdot(m(t)\cdot A_c)^2\cdot 1\right]^{\frac{1}{2}} \\
= & \frac{1}{2}\cdot m(t)\cdot A_c
\end{aligned}
\tag{2.22}
$$

For (2.22), the demodulated signal is $\frac{1}{2}\cdot m(t)\cdot A_c$, which is proportional to the message signal $m(t)$. And the phase difference φ is also eliminated.

By using quadrature demodulation to demodulate DSB TC AM signal and DSB SC AM signal, the scaled amplitude caused by the phase difference in the coherent demodulation can be compensated.

2.2 FM Signal Demodulation

In FM system, the process of demodulation is also known as conversion or detection.

2.2.1 FM to AM Conversion

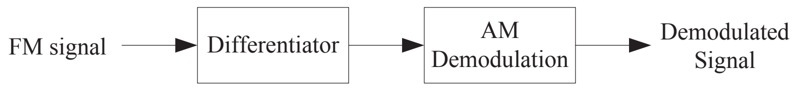

Figure 2.7: Block diagram of FM to AM conversion

FM to AM Conversion, which is also known as slope detection, is shown in Figure 2.7. In this method, the input FM signal is converted to an AM signal by the differentiator [6]. Then an AM demodulation method is used to demodulate the converted signal. Envelope demodulation methods are commonly used in the AM signal demodulation [1, 7, 8]. Thus two FM to AM conversion methods are FM to AM conversion with Hilbert transform and FM to AM conversion using filter.

2.2.2 Zero-Crossing Demodulation

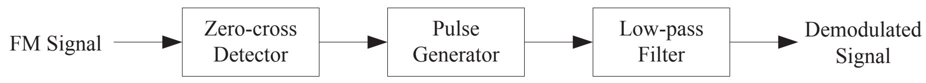

Figure 2.8: Block diagram of zero-crossing demodulation

Figure 2.8 shows the block diagram of the zero-crossing demodulation. The zero-cross detector is used to find positive zero-cross points. When the amplitude of the input signal is changed from negative to positive, an impulse is generated at the zero-cross point. Then the pulse generator converts the impulse chain into a pulse chain [9]. In the pulse chain, the width and amplitude of each pulse are τ and A respectively.

Assume the instantaneous frequency of the FM signal is

$$f = f_c + \Delta f \cdot m\,(t) \tag{2.23}$$

where f_c is the carrier frequency, Δf is the maximum frequency deviation and $m(t)$ is the massage signal.

And

$$T = \frac{1}{f} \tag{2.24}$$

Thus, the output of the low-pass filter is

$$\begin{aligned} A \cdot \frac{\tau}{T} &= A\tau f \\ &= A\tau \cdot [f_c + \Delta f \cdot m\,(t)] \\ &= A\tau f_c + A\tau\Delta f \cdot m\,(t) \end{aligned}$$

where the DC component is $A\tau f_c$, the demodulated signal is $A\tau\Delta f \cdot m\,(t)$.

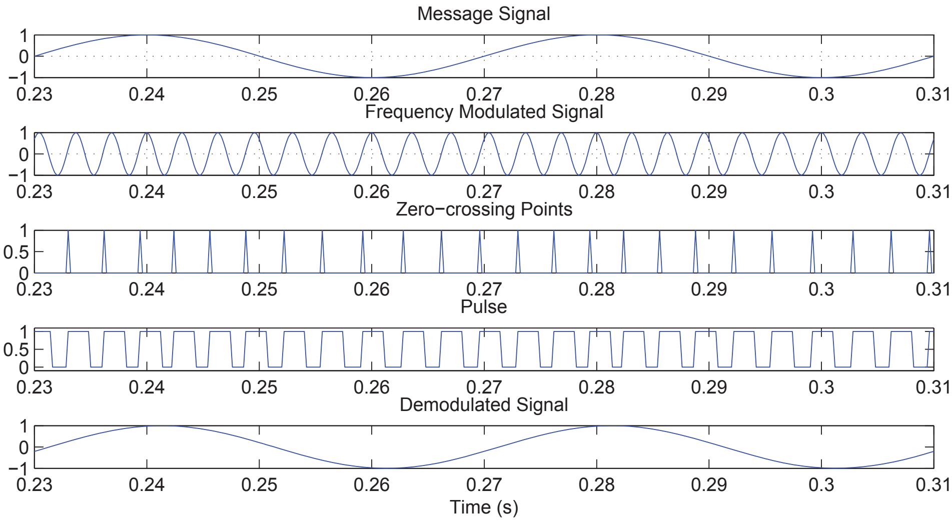

Figure 2.9: Process of zero-crossing demodulation

Figure 2.9 shows the process of zero-crossing demodulation. A 25 Hz sinusoidal wave is used as the message signal, the carrier is a 300 Hz sinusoidal wave. The maximum frequency deviation is 20 Hz. At first, the carrier is frequency modulated by the message signal. Then the zero-cross detector outputs the positive zero-crossing points of the FM signal, shown as triangular wave. After that, the pulse generator converts every zero-crossing point to a pulse with fixed width and amplitude, shown as rectangular wave. After the low-pass filtering, the message signal is recovered.

2.2.3 Quadrature Demodulation in FM System

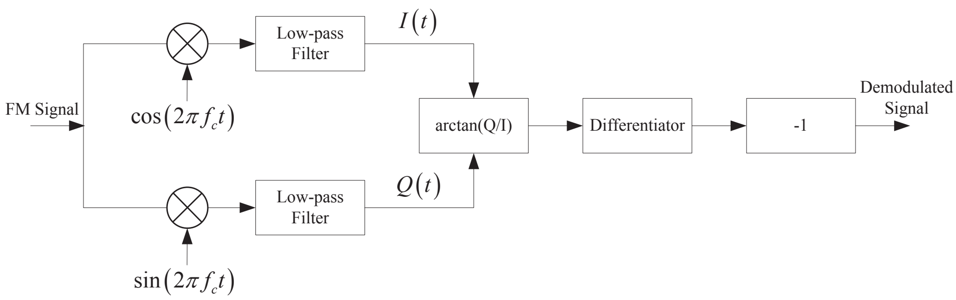

Figure 2.10: Block diagram of quadrature demodulation for FM signal

There is one difference between the quadrature demodulation method in AM signal demodulation and the one in FM signal demodulation. In AM signal demodulation, quadrature demodulation is used to extract the amplitude information of the message signal from the AM signal. In FM signal demodulation, quadrature demodulation is used to extract the angel information from the FM signal, and then the angle information is converted to message signal.

As in Figure 2.10, the $\arctan(Q/I)$ module is used to obtain the phase information of the modulated signal. Then the differentiator outputs the instantaneous frequency of the modulated signal. Thus the FM signal is demodulated [8].

Chapter 3

Simulation Design

This chapter shows the simulation details of AM signal demodulation and FM signal demodulation.

3.1 AM Signal Demodulation

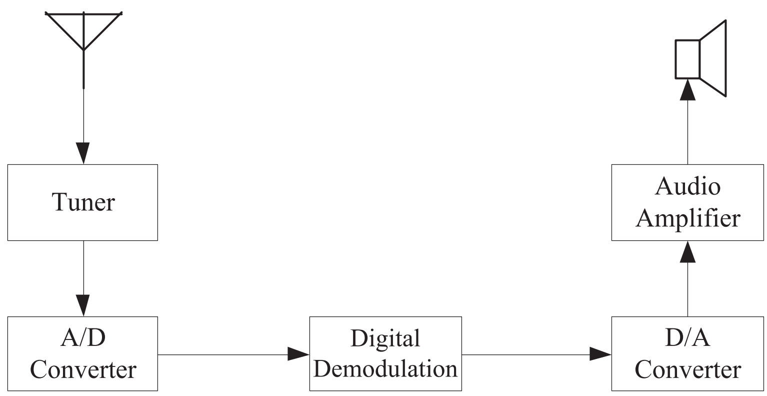

Figure 3.1: Block diagram of an AM signal receiver

As shown in Figure 3.1, the antenna is used to receive the Radio Frequency (RF) signal. The tuner converts the received RF signal to an Intermediate Frequency (IF) signal with carrier frequency of 455 kHz. Then the A/D converter samples the tuner output, converts the IF signal from an analog signal into a digital signal, with the sampling frequency at 4 MHz. After the digital demodulation, the recovered signal is obtained. At last, the recovered signal is converted to sound through the D/A converter, the audio amplifier and the speaker. The digital demodulation module is simulated in MATLAB.

Figure 3.2 shows the simulation design chart. First, the message signal modulates the carrier with three different methods: double side band transmitted carrier amplitude modulation, double side band suppressed carrier amplitude modulation and single side band amplitude modulation. Then the double side band transmitted carrier AM signal is demodulated by five methods: envelope demodulation using filter, envelope demodulation with Hilbert transform, coherent demodulation, square-law demodulation and quadrature demodulation. Other two kinds of AM signal are both demodulated by the coherent demodulation. At last, all the demodulated signals are normalized, and then the normalized signals are compared with the message signal in both time domain and frequency domain.

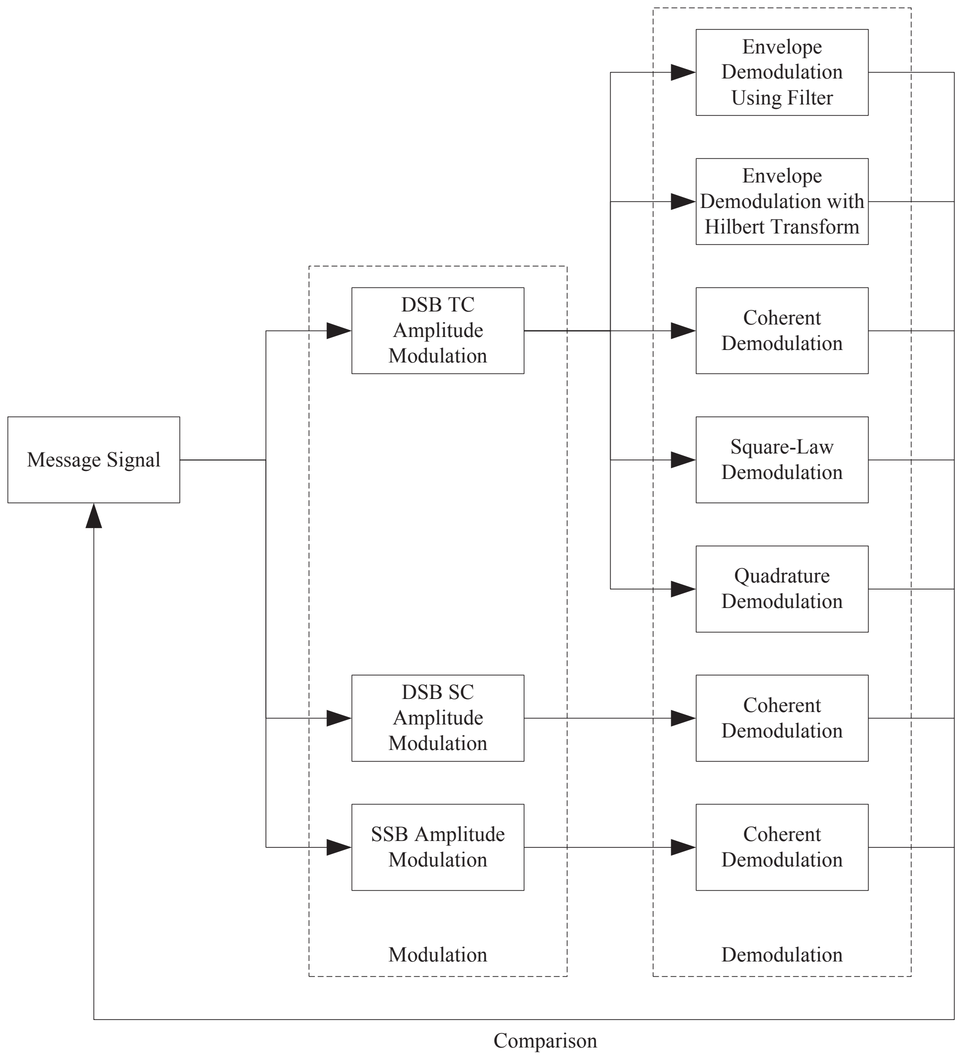

Figure 3.2: Design chart of simulation of AM signal demodulation

In this simulation, the amplitude modulation is used to simulate the output of the A/D converter. For single side band amplitude modulation, the lower side band is chosen. Simulation details are specified in Table 3.1.

Table 3.1: Parameters in the simulation of AM signal demodulation

Parameter	Value
Carrier frequency	455 kHz
Sampling frequency	4 MHz
Carrier amplitude	1
Carrier initial phase	0

3.2 FM Signal Demodulation

As shown in Figure 3.3, the block diagram of the FM signal receiver is similar to the block diagram of the AM signal receiver, with only a few differences. The carrier frequency of the IF signal, which is the tuner output, is 10.7 MHz. Then the IF signal is sampled by the A/D converter at the sampling frequency of 64 MHz. After that, the digital local oscillator and the mixer are used to shift the carrier frequency of the IF signal to 455 kHz. However, the carrier shifting produces an image frequency at 2.0945 MHz, thus the low pass filter is used to reject the image frequency. After low-pass filtering, the down sampling module decreases the sampling frequency of the pre-processed signal from 64 MHz to 4 MHz before digital demodulation. At last, the demodulated signal is converted to sound through the D/A converter, the audio amplifier and the speaker. The modules in the dash box are simulated in MATLAB.

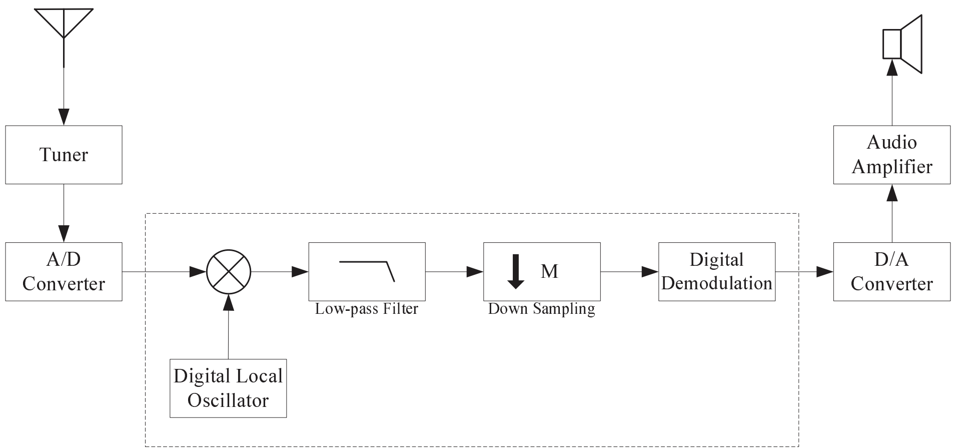

Figure 3.3: Block diagram of a FM signal receiver

Figure 3.4 is the design chart of simulation of FM signal demodulation. Message signal modulates the carrier into a FM signal. After carrier shifting and down sampling, the FM signal is demodulated by four methods: FM to AM conversion with Hilbert transform, FM to AM conversion using filter, quadrature demodulation and zero-crossing demodulation. At last, all the demodulated signals are normalized, and then the normalized signals are compared with the message signal in both time domain and frequency domain.

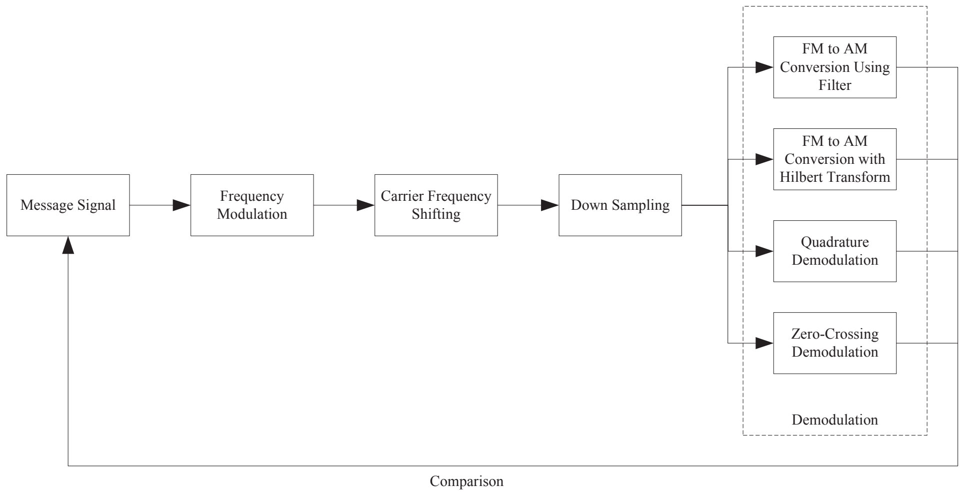

Figure 3.4: Design chart of simulation of FM signal demodulation

Frequency modulation in the simulation is used to simulate the output of the A/D converter. Simulation details are specified in Table 3.2.

Table 3.2: Parameters in the simulation of FM signal demodulation

Parameter	Value
Carrier frequency	10.7 MHz
Frequency deviation	±75 kHz
Carrier amplitude	1
Carrier initial phase	0
Sampling frequency	64 MHz
Carrier shifted to	455 kHz
Down sampled frequency	4 MHz

Chapter 4

Results and Analysis

4.1 Simulation of AM Signal Demodulation

In this simulation, envelope demodulation using filter, envelope demodulation with Hilbert transform, coherent demodulation, square-law demodulation and quadrature demodulation are simulated via MATLAB.

In AM broadcasting standard, the frequency range of the message signal is vary from 150 Hz to 4500 Hz. Thus a chirp signal in the frequency range from 150 Hz to 4500 Hz, which is generated by the function chirp, is used as the message signal. Figure 4.1 shows the message signal in time domain and frequency domain. Frequency of the message signal increases with time.

Figure 4.2 shows a 455 kHz sinusoidal wave, which is used as the carrier.

After amplitude modulation, three kinds of AM signal are obtained.

Figure 4.3 shows the double side band transmitted carrier AM signal in time domain and frequency domain.

Figure 4.4 shows the double side band suppressed carrier AM signal in time domain and frequency domain.

Figure 4.5 shows the single side band AM signal in time domain and frequency domain.

4.1.1 Envelope Demodulation

Figure 4.6 shows the result of the double side band transmitted carrier AM signal processed by the envelope demodulation with Hilbert transform. It shows that only a few negligible ripples at the beginning of the demodulated signal. The frequency spectrum is also almost the same as the frequency spectrum of the message signal in Figure 4.1.

Figure 4.7 demonstrates the result of the double side band transmitted carrier AM signal processed with envelope demodulation using filter. Similarly, the message signal and the demodulated signal are almost the same. The frequency spectrum of the result is the same as the frequency spectrum of the message signal in Figure 4.1.

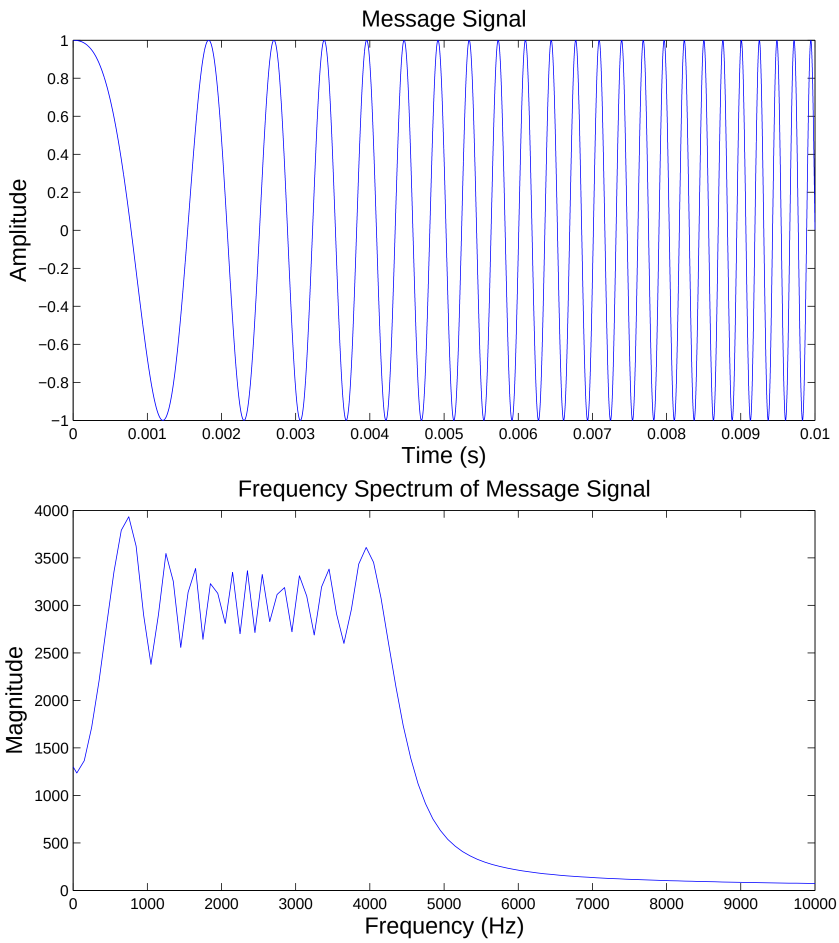

Figure 4.1: Message signal in time domain and frequency domain

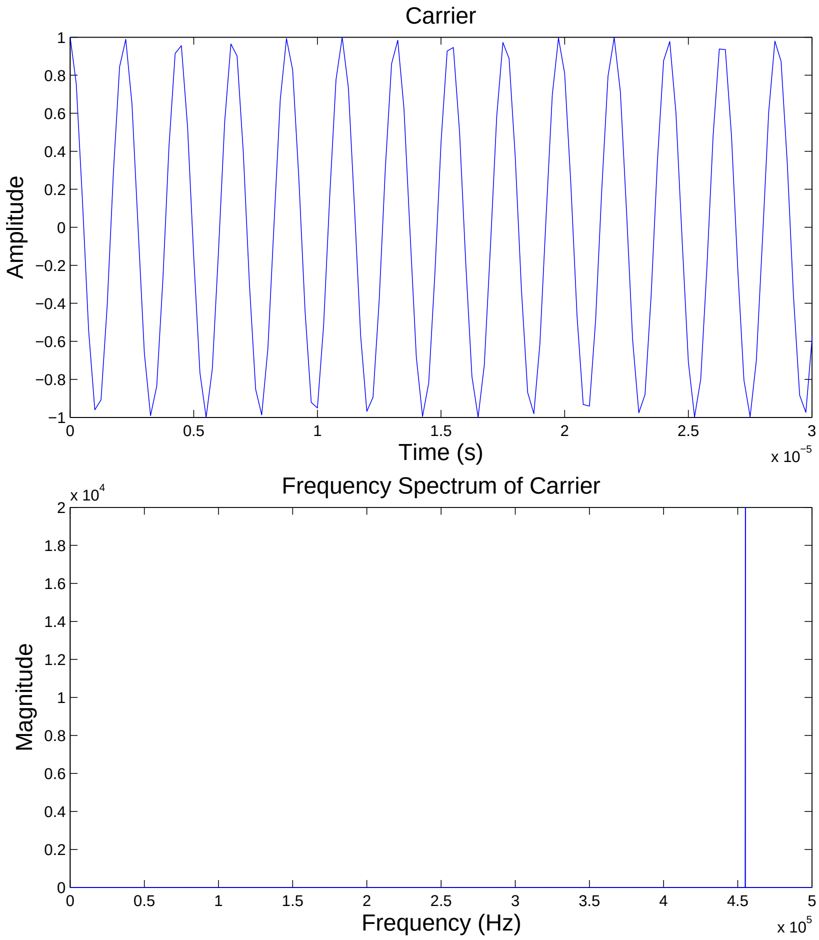

Figure 4.2: Carrier in time domain and frequency domain

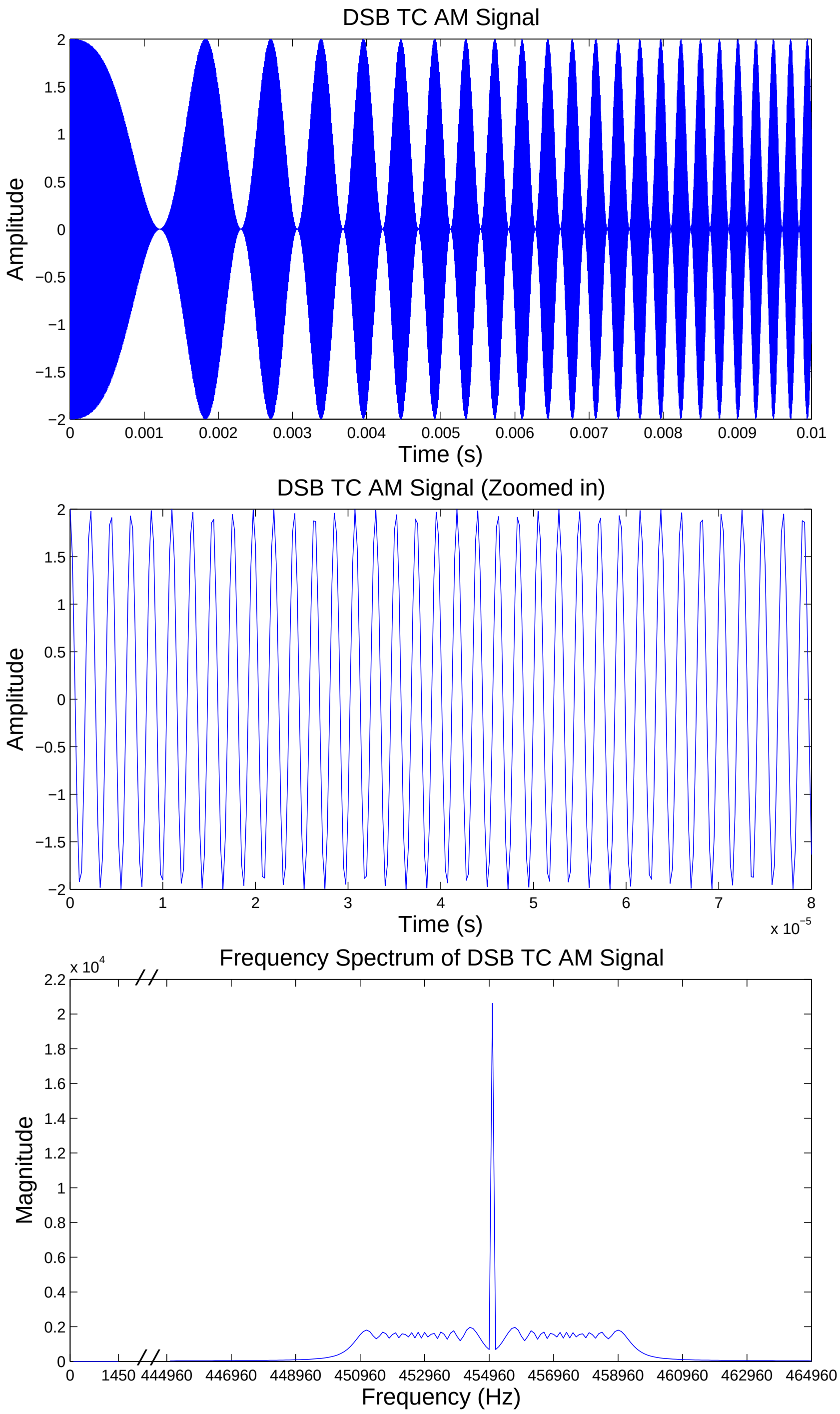

Figure 4.3: Double side band transmitted carrier AM signal in time domain and frequency domain

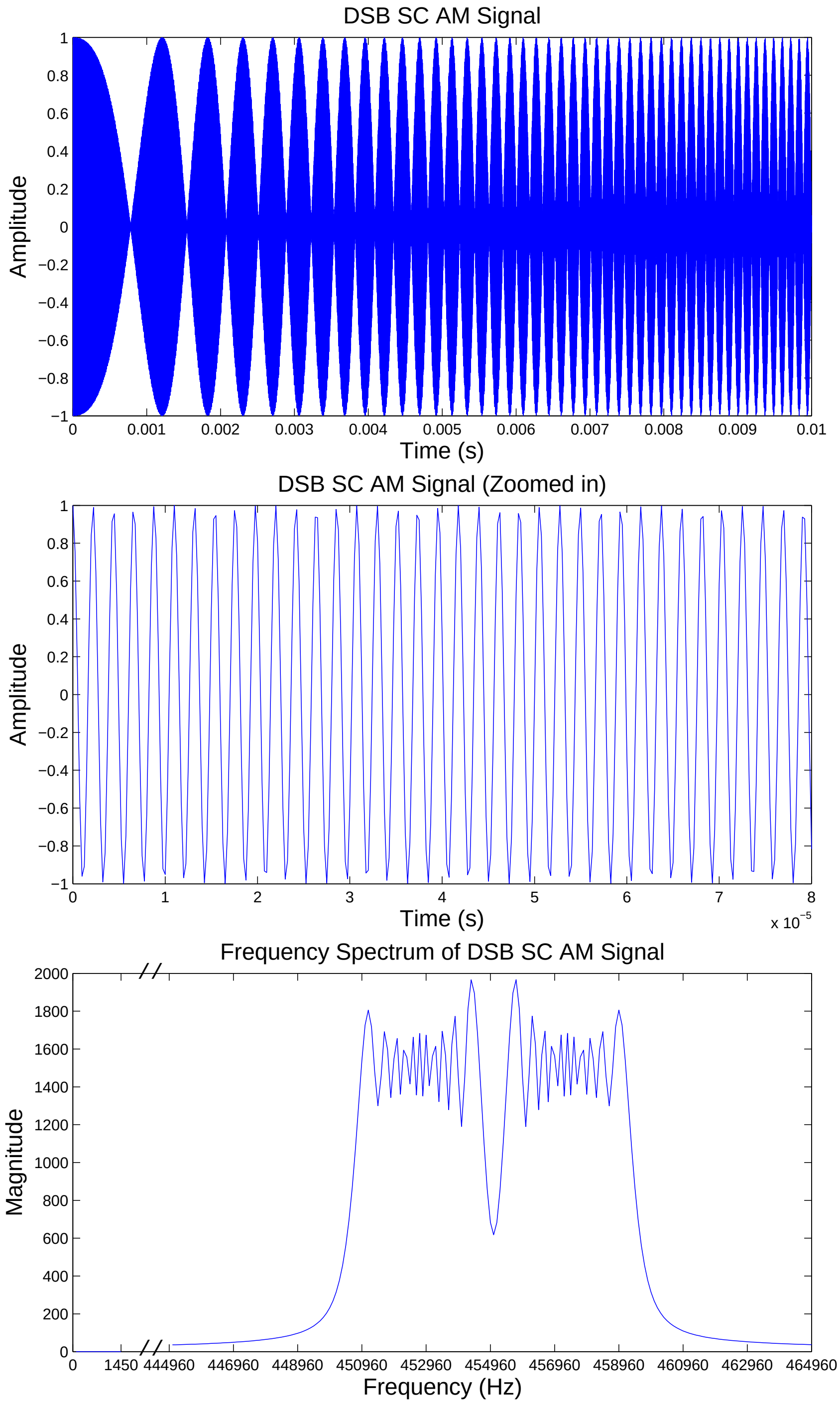

Figure 4.4: Double side band suppressed carrier AM signal in time domain and frequency domain

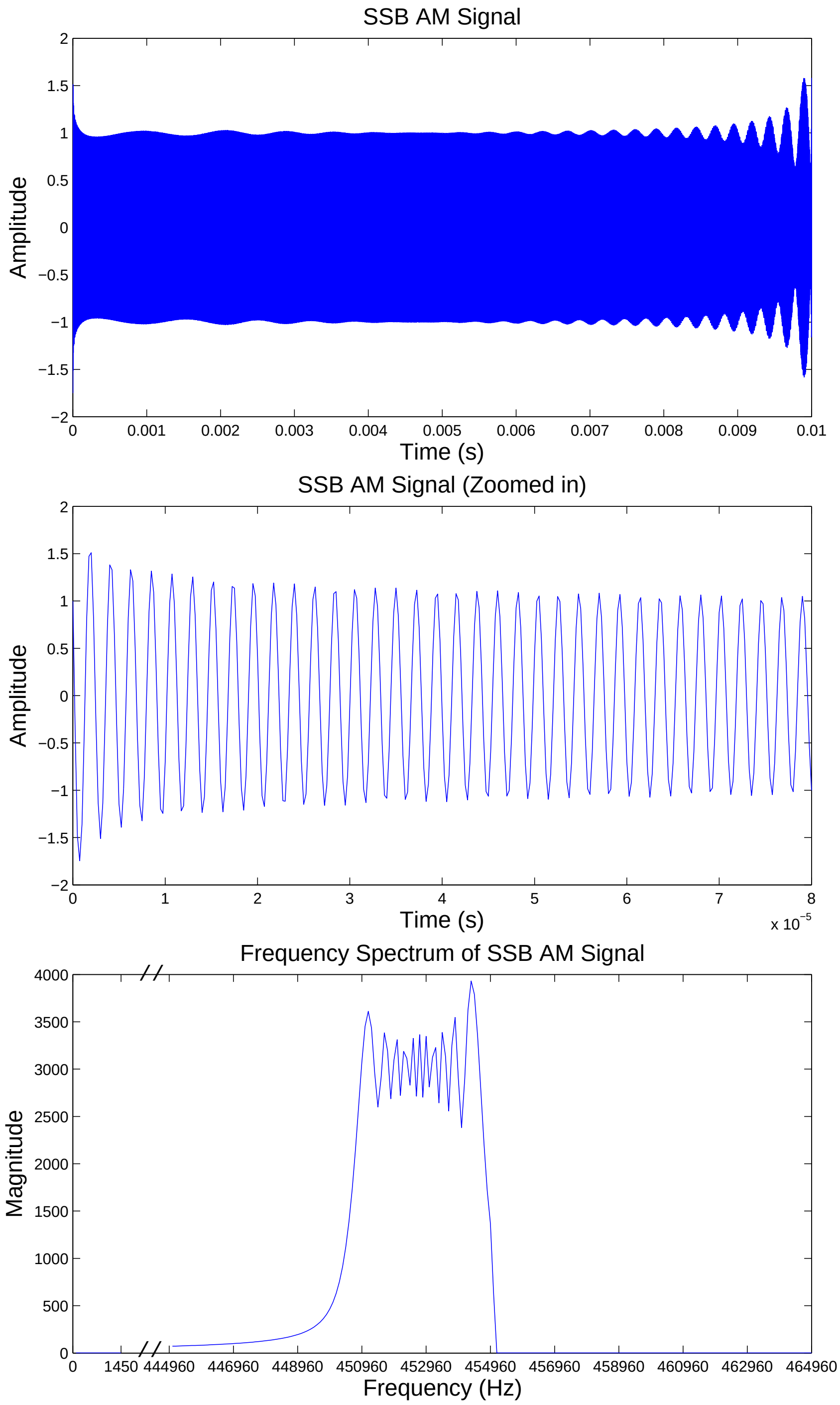

Figure 4.5: Single side band AM signal in time domain and frequency domain

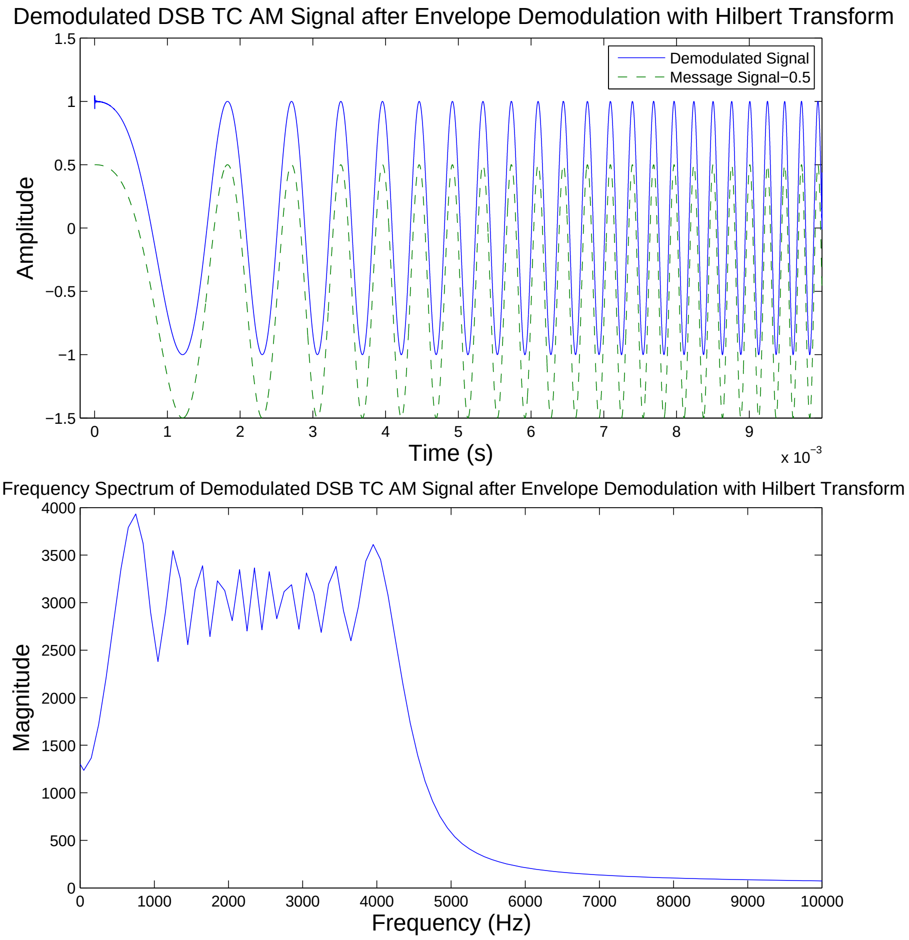

Figure 4.6: Result of demodulating double side band transmitted carrier AM signal after envelope demodulation with Hilbert transform in time domain and frequency domain

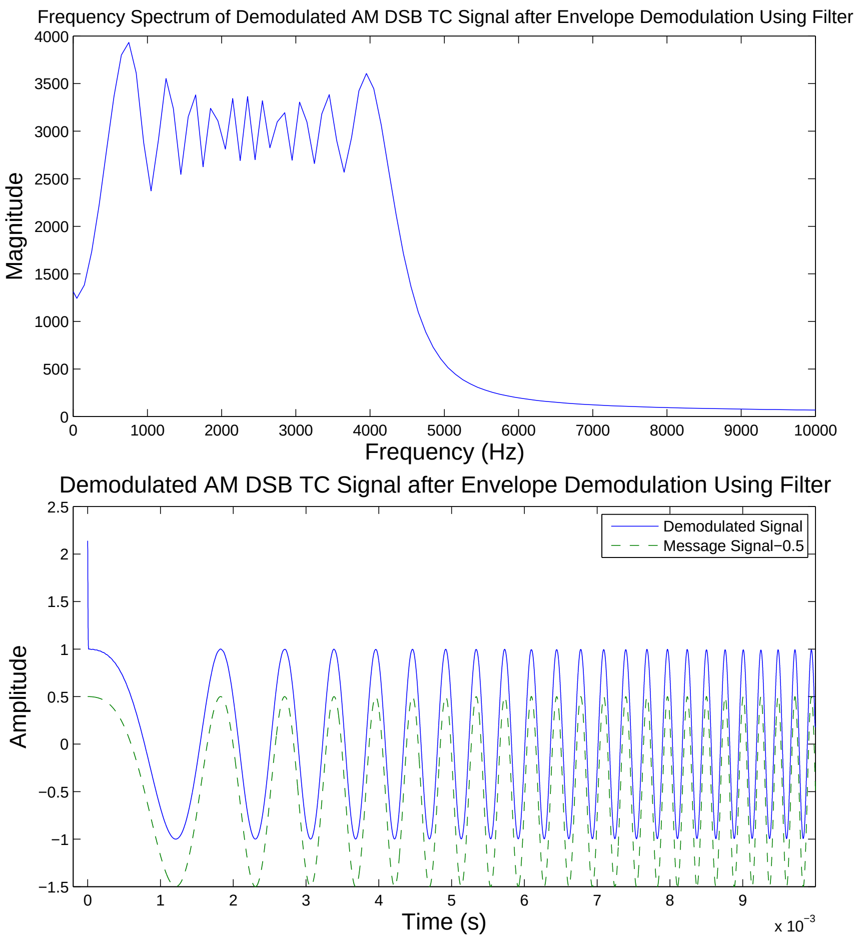

Figure 4.7: Result of demodulating double side band transmitted carrier AM signal after envelope demodulation using filter in time domain and frequency domain

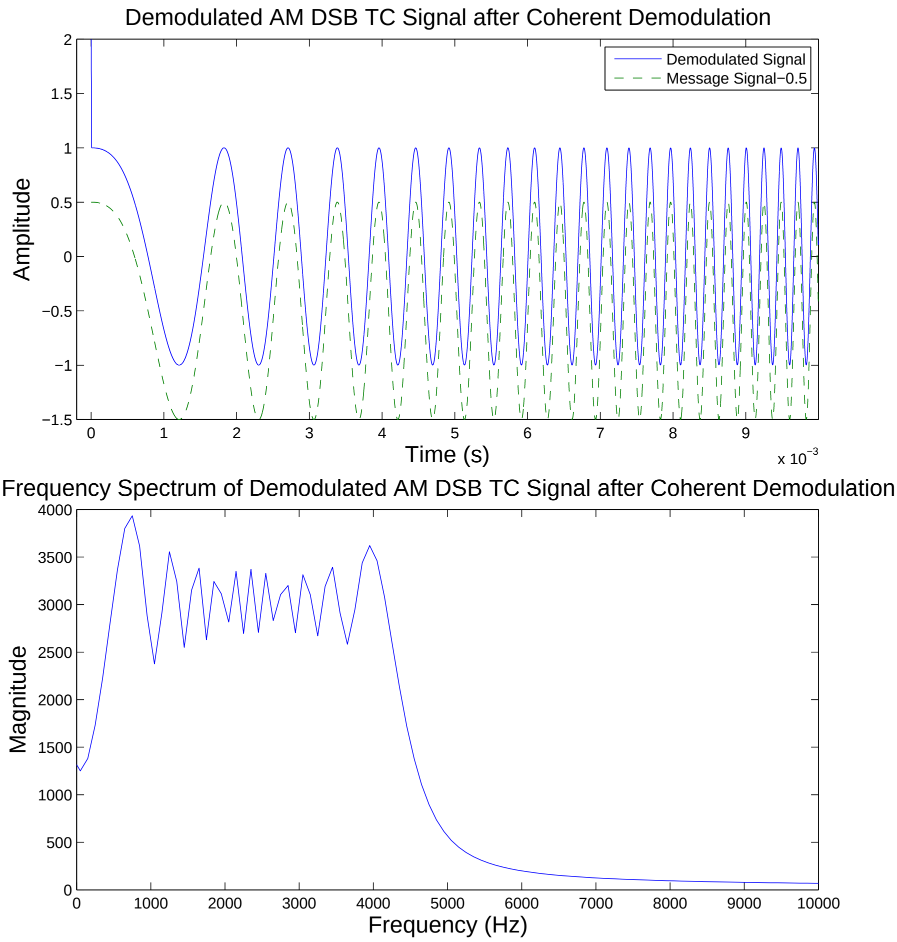

Figure 4.8: Result of demodulating double side band transmitted carrier AM signal after coherent demodulation in time domain and frequency domain

This shows that two envelope demodulation methods work as desired.

4.1.2 Coherent Demodulation

The result of demodulating the double side band transmitted carrier AM signal after the coherent demodulation is shown in Figure 4.8. In this figure, the demodulated signal is the same as the message signal. The frequency spectrum is identical to the one of the message signal in Figure 4.1.

Figure 4.9 is the result of demodulating double side band suppressed carrier AM signal after the coherent demodulation. In time domain, the demodulated signal reflects the message signal. While in frequency domain, the simulation of the coherent demodulation also works well.

In Figure 4.10, the demodulated signal is almost the same as the message signal in time domain. It also shows the same frequency spectrum as the

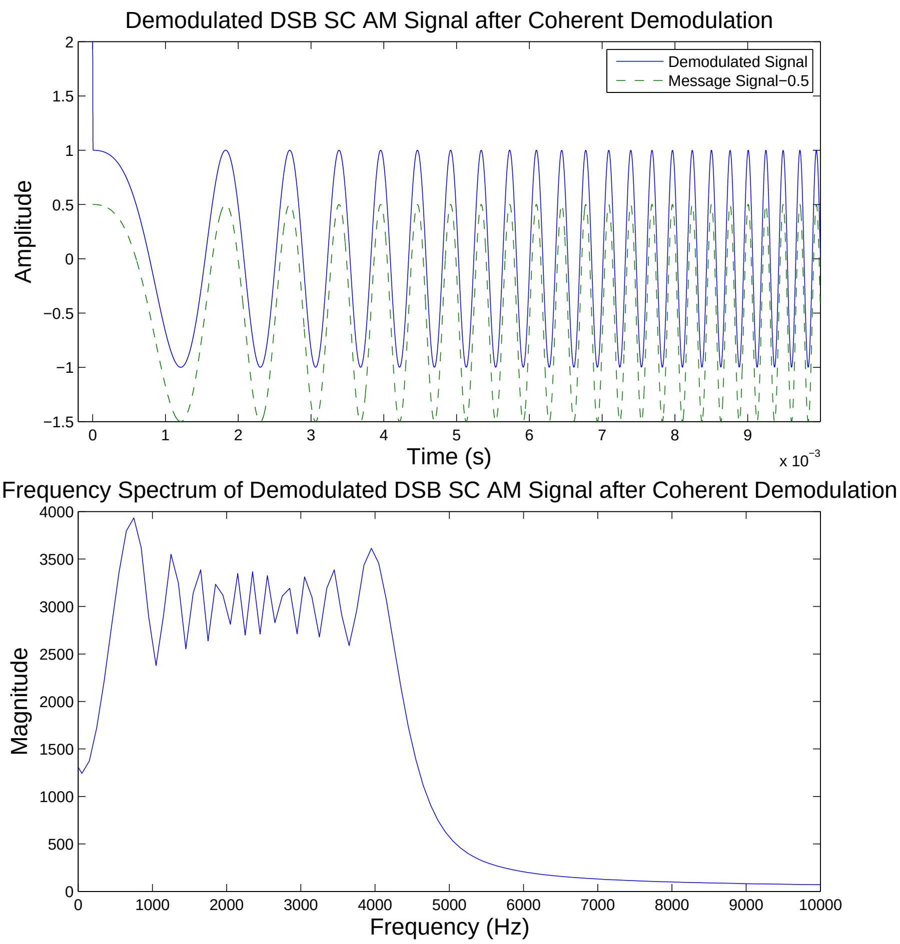

Figure 4.9: Result of demodulating double side band suppressed carrier AM signal after coherent demodulation in time domain and frequency domain

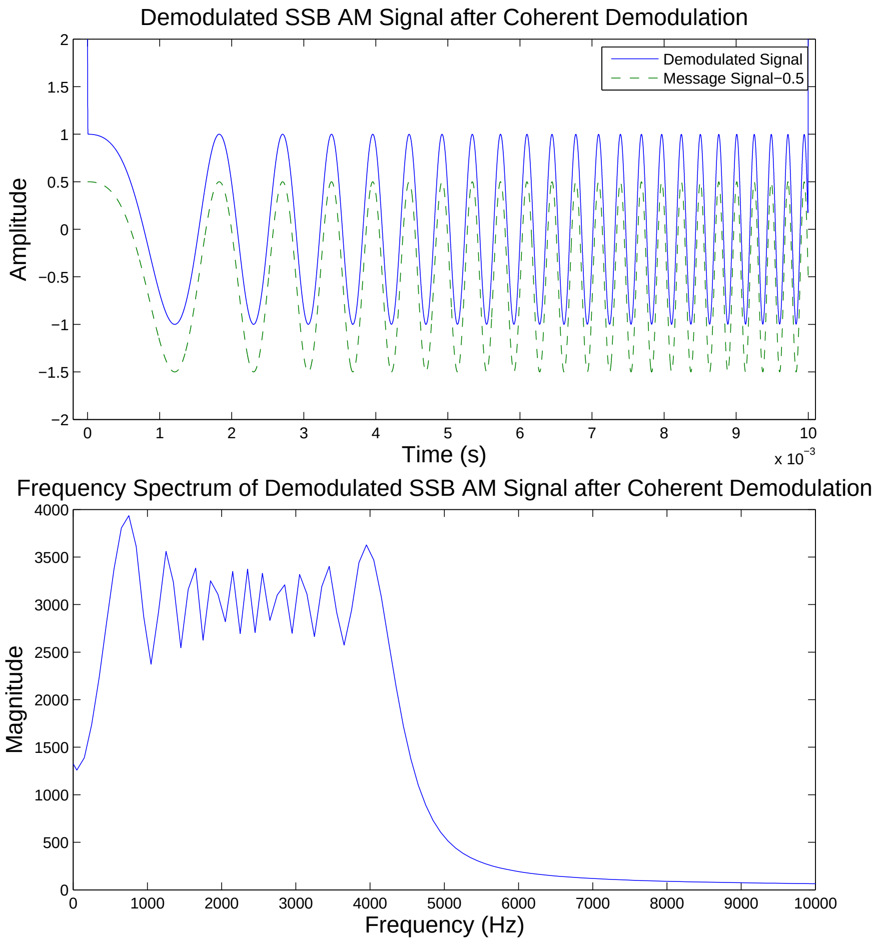

Figure 4.10: Result of demodulating single side band AM signal after coherent demodulation in time domain and frequency domain

one in Figure 4.1.

It shows that the coherent demodulation works as desired.

4.1.3 Square-Law Demodulation

Figure 4.11 shows the result of the square-law demodulation, the demodulated signal has the same shape as the message signal. They also share the same frequency spectrum.

This figure shows that the square-law demodulation demodulates the signal as the desired.

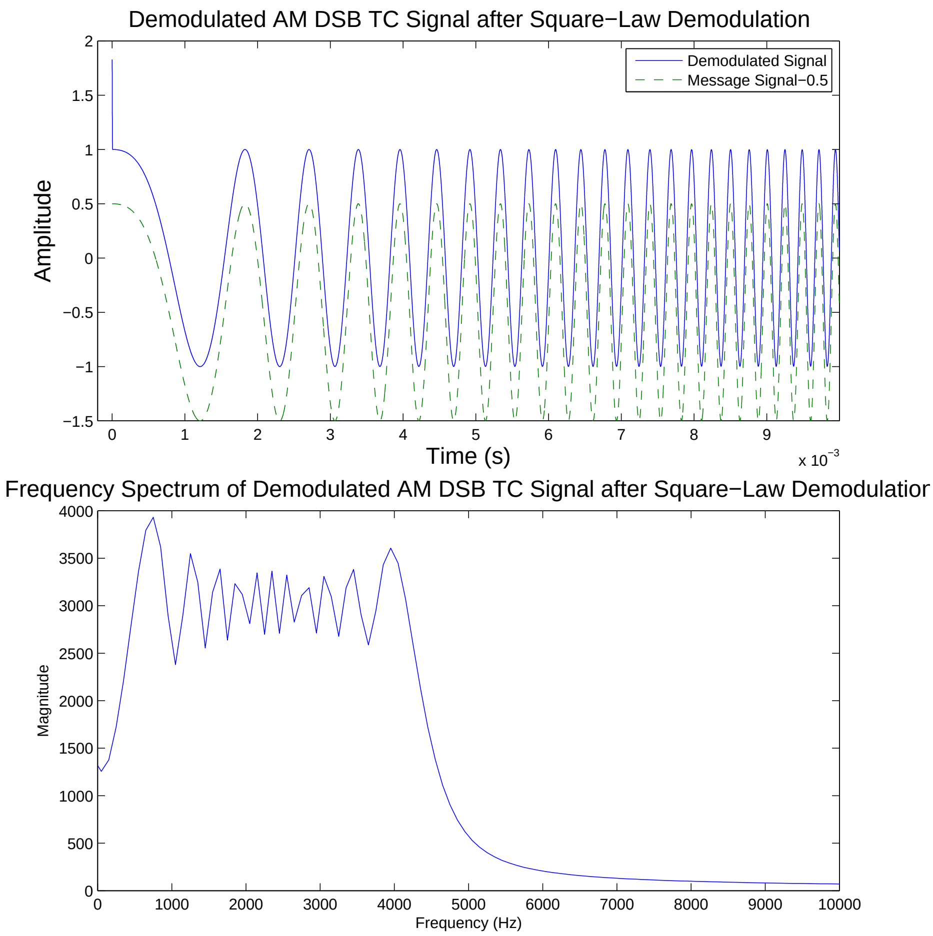

Figure 4.11: Result of demodulating double side band transmitted carrier AM signal after square-law demodulation in time domain and frequency domain

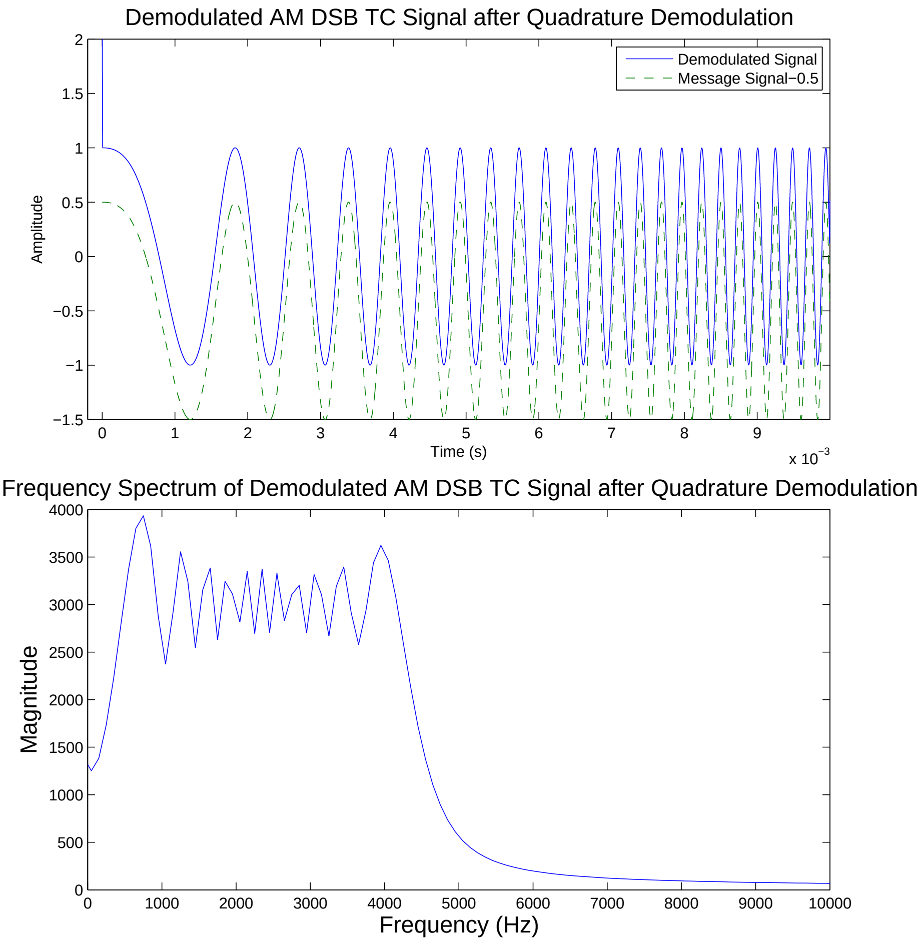

Figure 4.12: Result of demodulating double side band transmitted carrier AM signal after quadrature demodulation in time domain and frequency domain

4.1.4 Quadrature Demodulation in AM System

Figure 4.12 shows the simulation of the quadrature demodulation working on the double side band transmitted carrier AM signal. The demodulated signal has the same shape and frequency spectrum as the message signal.

The simulation of AM signal demodulation shows that these methods can demodulate the AM signal perfectly.

4.2 Simulation of FM Signal Demodulation

For the FM signal, four demodulation methods are simulated: FM to AM conversion using filter, FM to AM conversion with Hilbert transform, zero-crossing demodulation and quadrature demodulation.

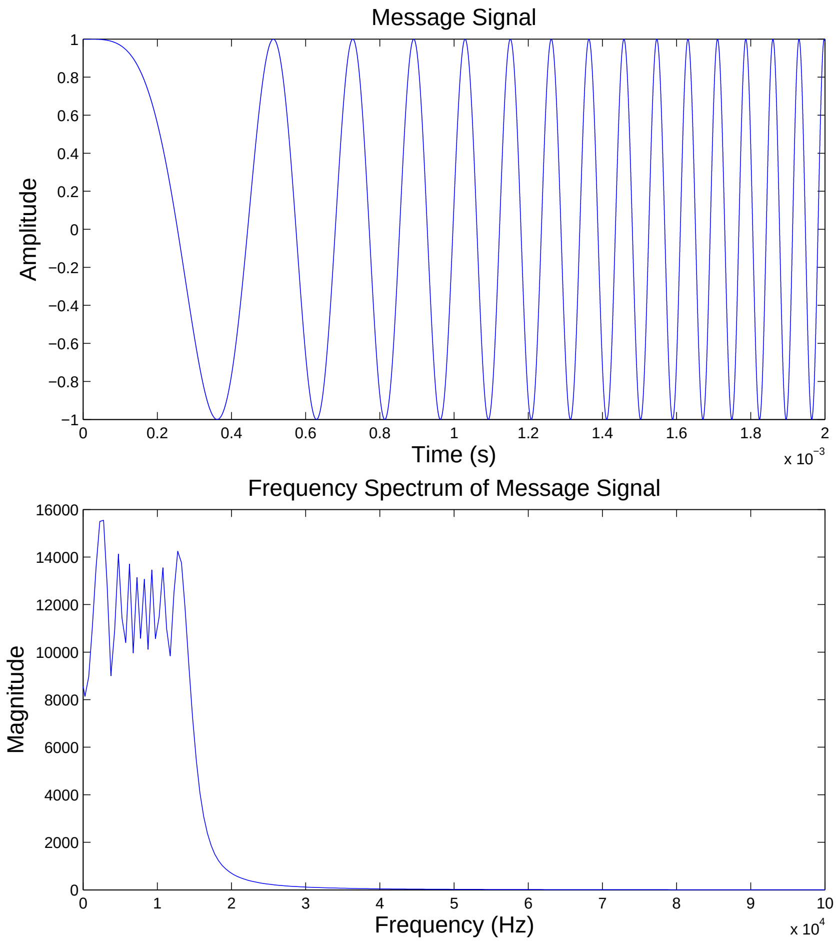

Figure 4.13: Message signal in time domain and frequency domain

In FM broadcasting standard, the frequency range of the message signal is vary from 30 Hz to 15 kHz. A chirp signal in the frequency range from 30 Hz to 15 kHz is used as message signal. Figure 4.13 shows the message signal in time domain and frequency domain. Frequency of the message signal increases with time. Sampling frequency of the message signal is 64 MHz.

After frequency modulation, the FM signal is obtained, as shown in figure 4.14. Carrier frequency of the FM signal is 10.7 MHz and sampling frequency of the FM signal is 64 MHz. Then the carrier frequency of the FM signal is shifted from 10.7 MHz to 455 kHz, the result of carrier shifting is shown in Figure 4.15. However, carrier shifting leads to the image frequency at 2.0945 MHz. Thus a low-pass filter is used to reject the image frequency. The low-pass filter has two functions: one is to remove the image frequency; the other is used as an anti-aliasing filter in the down sampling. Figure 4.16

is the result of low-pass filtering. Then the down sampling is performed. The sampling frequency of the FM signal is decreased from 64 MHz to 4 MHz, using the downsample function in MATLAB. Figure 4.17 illustrates the result of down sampling. This FM signal is demodulated in the following simulation.

The sampling frequency of the message signal is also decreased from 64 MHz to 4 MHz, Figure 4.18 shows the down sampled message signal in time domain and frequency domain. All the demodulation results are compared with the down sampled message signal.

4.2.1 FM to AM Conversion

Figure 4.19 shows the result of the FM to AM conversion with Hilbert transform. In time domain, the demodulated signal is nearly the same as the message signal. While in frequency domain, for frequency less than 15 kHz, the frequency spectrum of the demodulated signal is close to which of the message signal; in the frequency range from 18 kHz to 33 kHz, the demodulated signal shows a few ripples.

In Figure 4.20, the demodulated signal reflects the message signal, with a few ripples at the beginning of the demodulated signal. The amplitude of the demodulated signal decreases with time. And in frequency domain, the demodulated signal in the frequency range from 30 Hz to 15 kHz is distorted compared with the frequency spectrum of the message signal; for the frequency range large than 18 kHz, the magnitude of the demodulated signal is higher than that of the message signal.

4.2.2 Zero-Crossing Demodulation

Figure 4.21 shows that the zero-crossing demodulation cannot recover the message signal very well from 0 to 0.6 millisecond. The amplitude of the recovered signal is decreasing with time. In frequency domain, the magnitude of the recovered signal is higher than that of the message signal. There exist ripples in the frequency range between 15 kHz to 65 kHz.

4.2.3 Quadrature Demodulation in FM system

Figure 4.22 is the result of quadrature demodulation. In time domain, the demodulated signal has a steep rise at the beginning. And the demodulated signal also shows a slightly delay for a few samples. In frequency domain, there are a few ripples in the frequency range from 20 kHz to about 55 kHz.

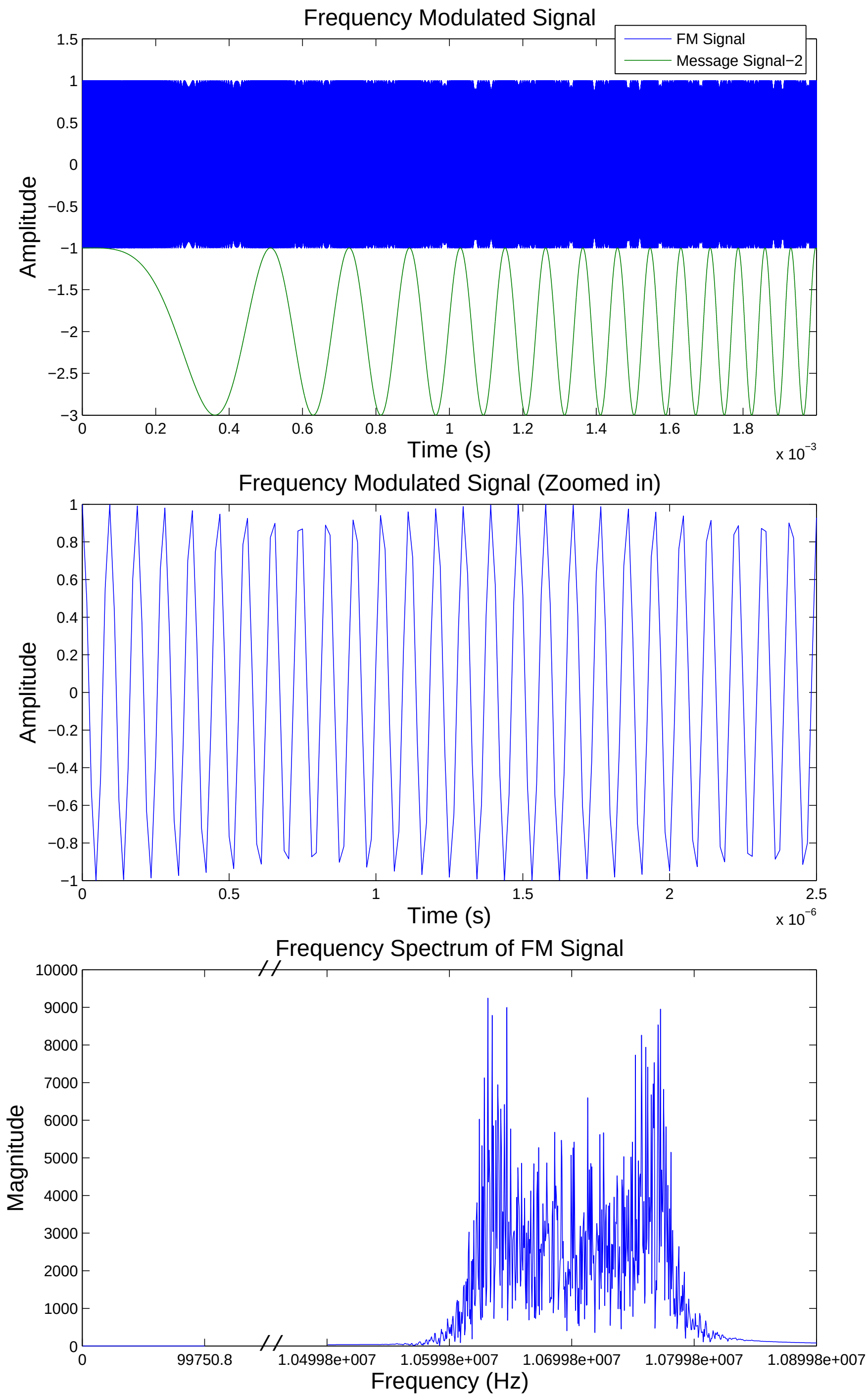

Figure 4.14: FM signal in time domain and frequency domain

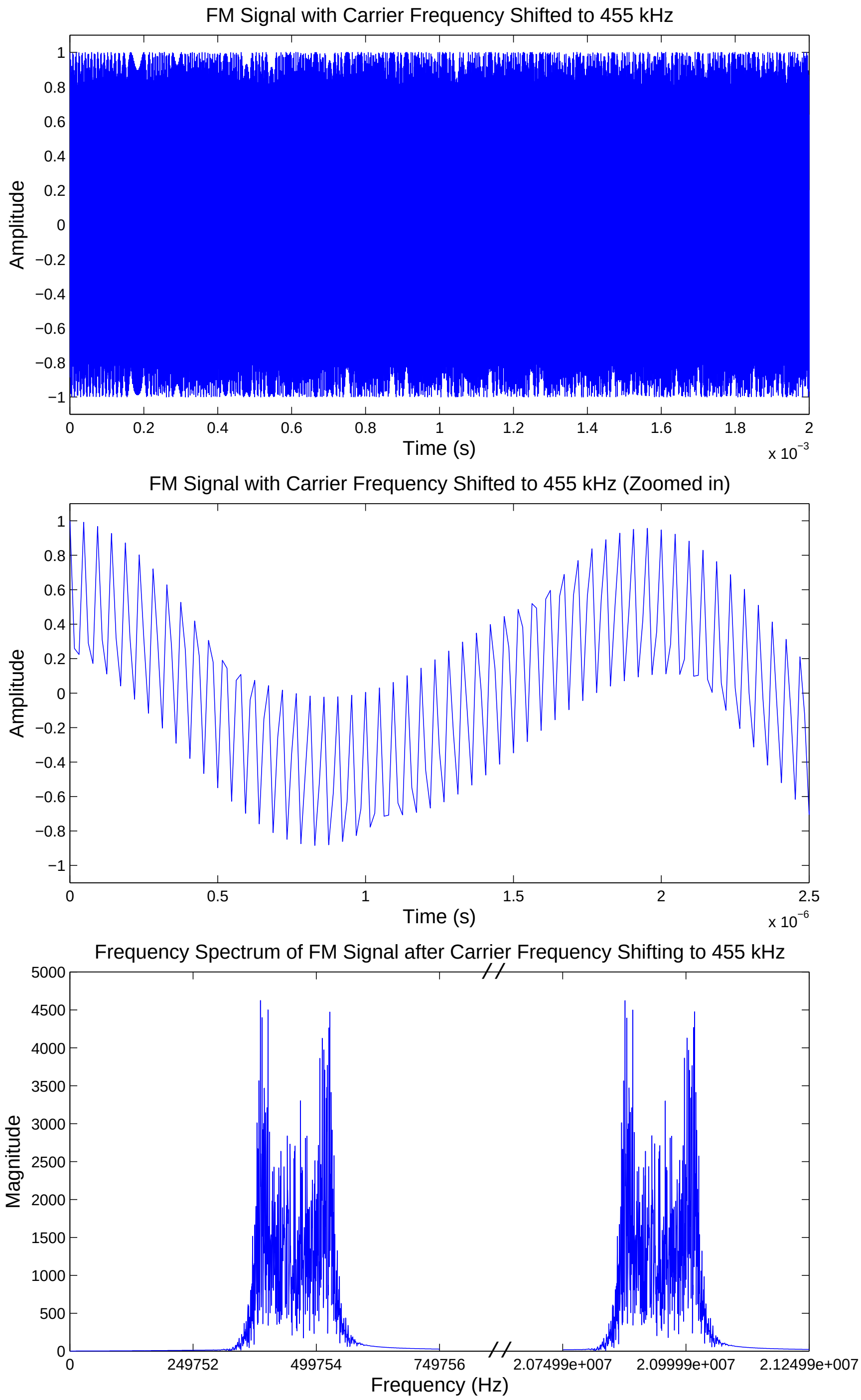

Figure 4.15: Result of carrier shifting the FM signal in time domain and frequency domain

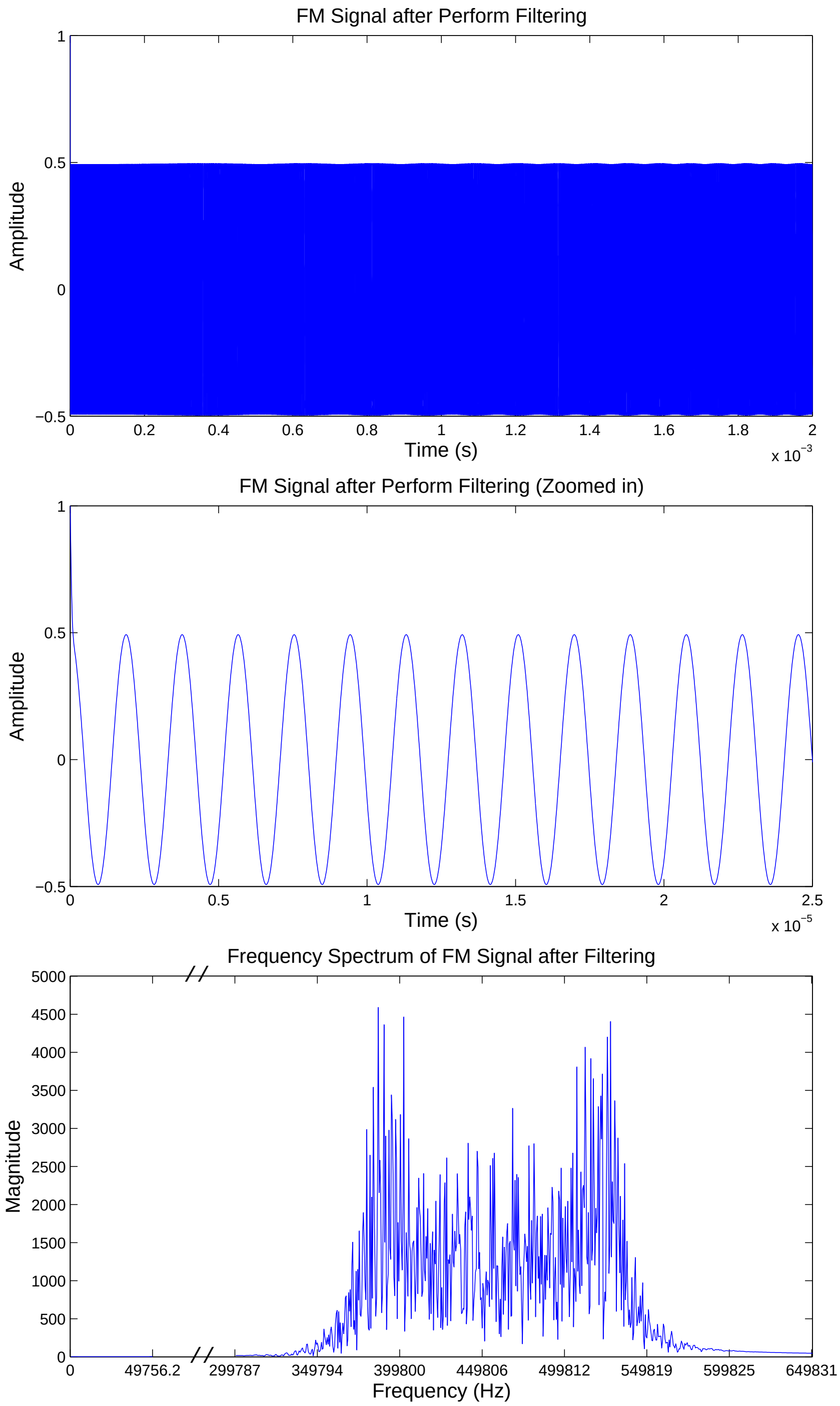

Figure 4.16: Result of filtering the FM signal in time domain and frequency domain

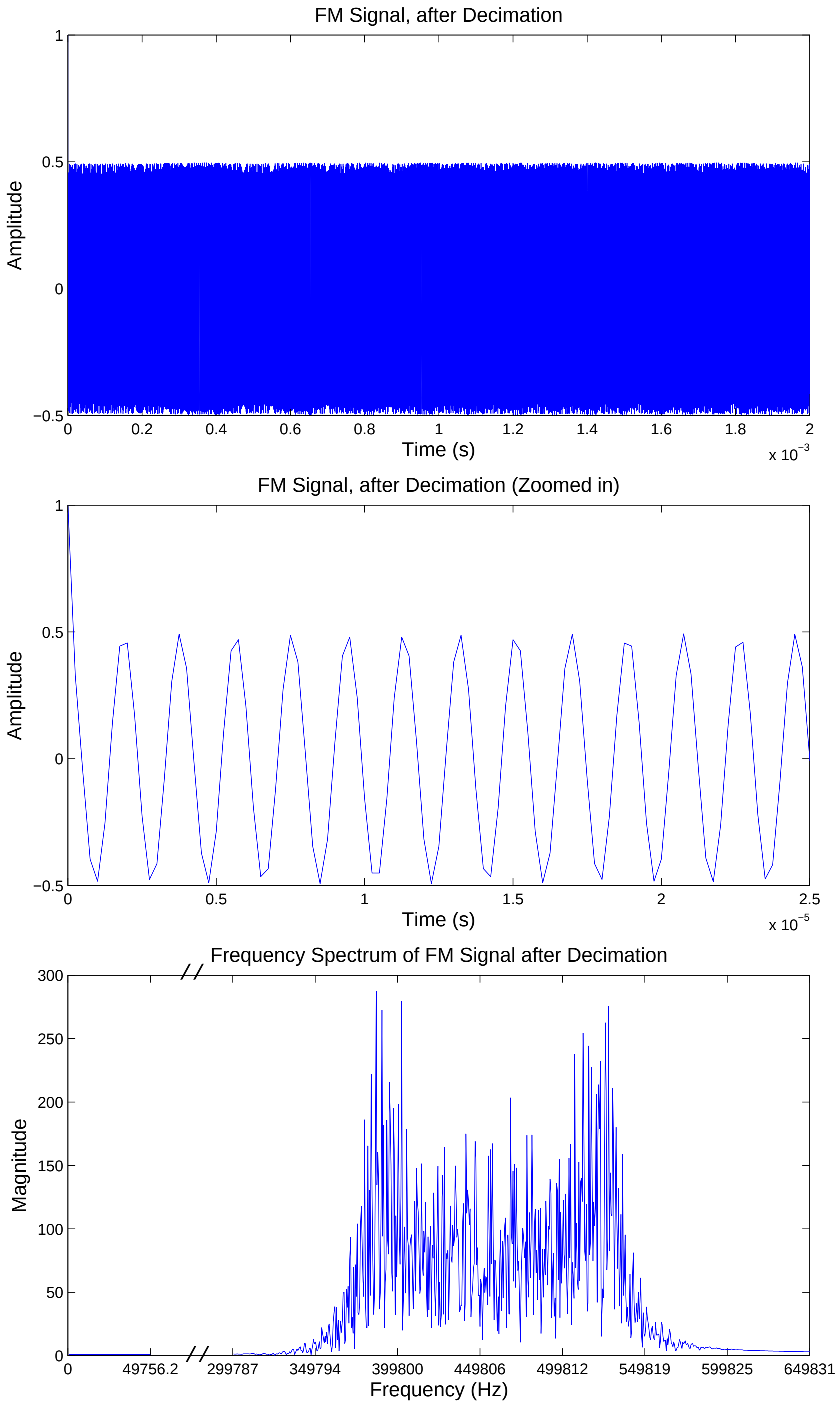

Figure 4.17: Result of down sampling the FM signal in time domain and frequency domain

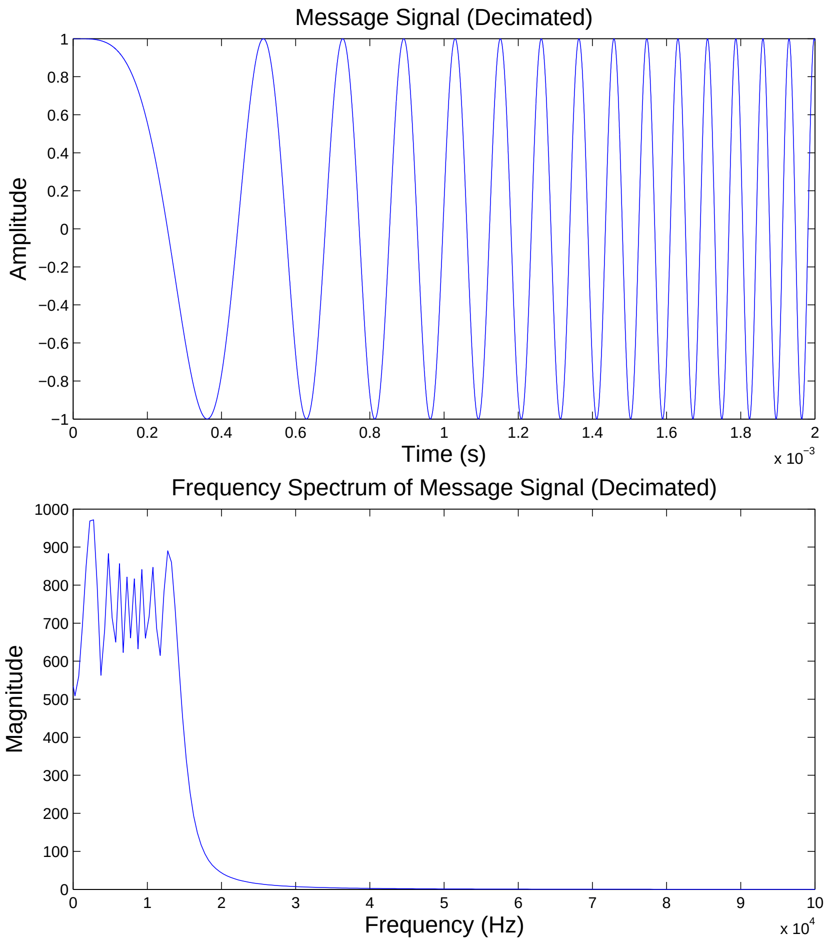

Figure 4.18: Result of down sampling the message signal in time domain and frequency domain

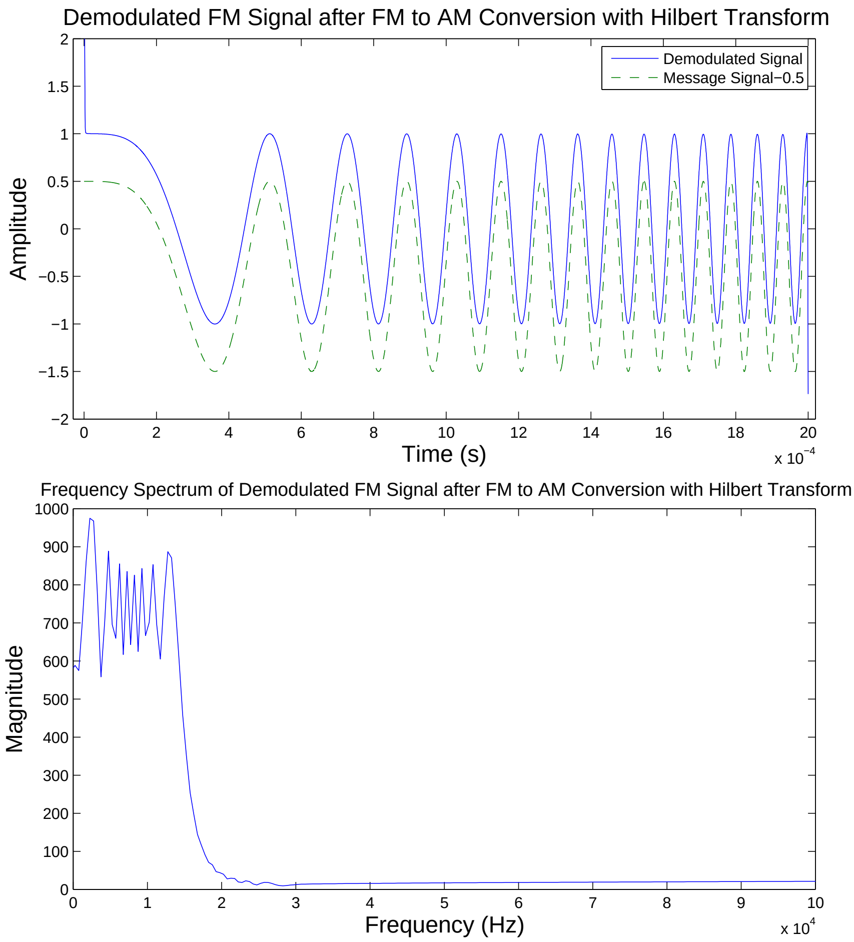

Figure 4.19: Result of demodulating FM signal after FM to AM conversion with Hilbert transform in time domain and frequency domain

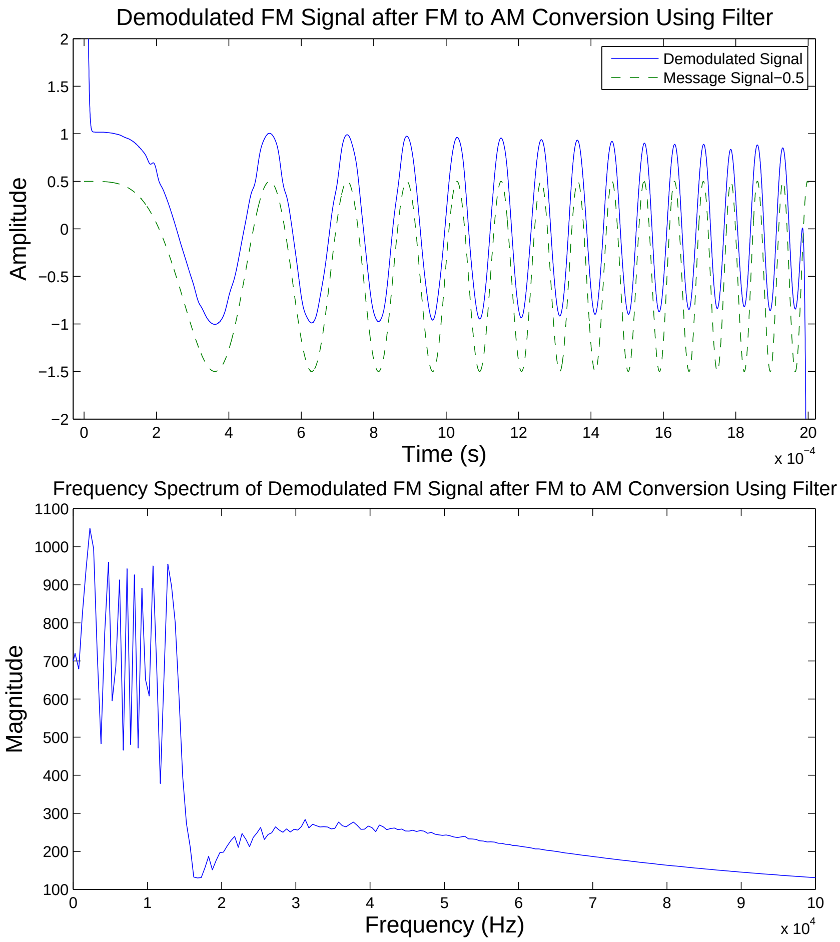

Figure 4.20: Result of demodulating FM signal after FM to AM conversion using filter in time domain and frequency domain

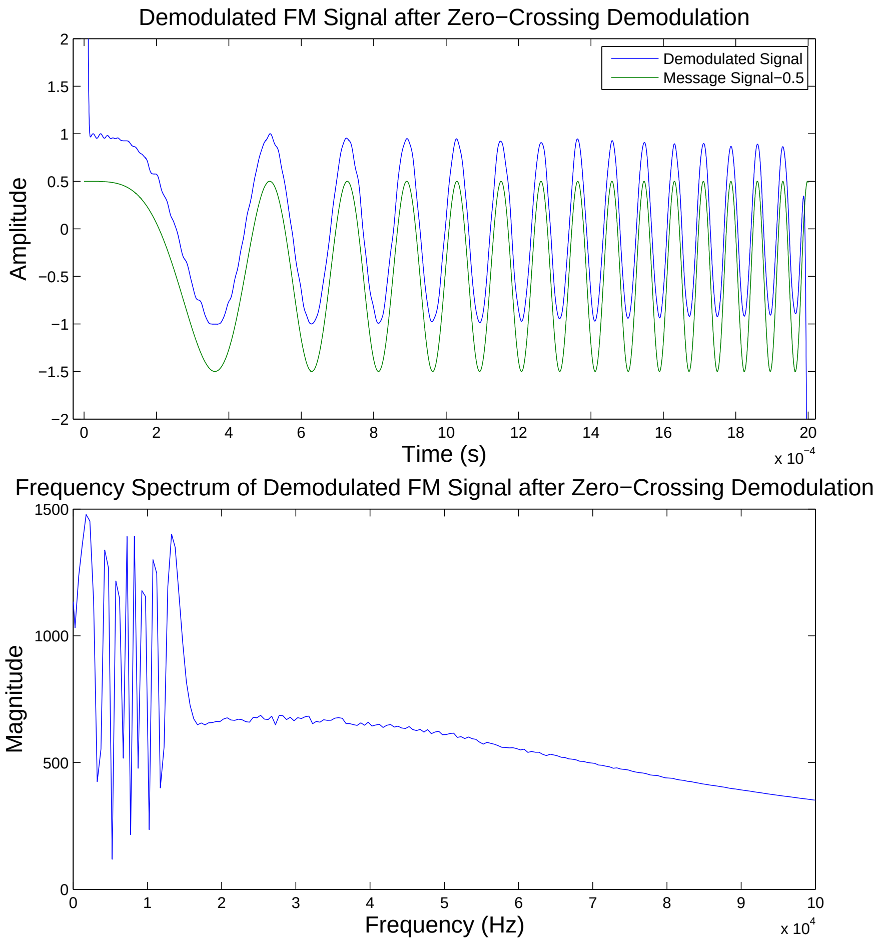

Figure 4.21: Result of demodulating FM signal after zero-crossing demodulation in time domain and frequency domain

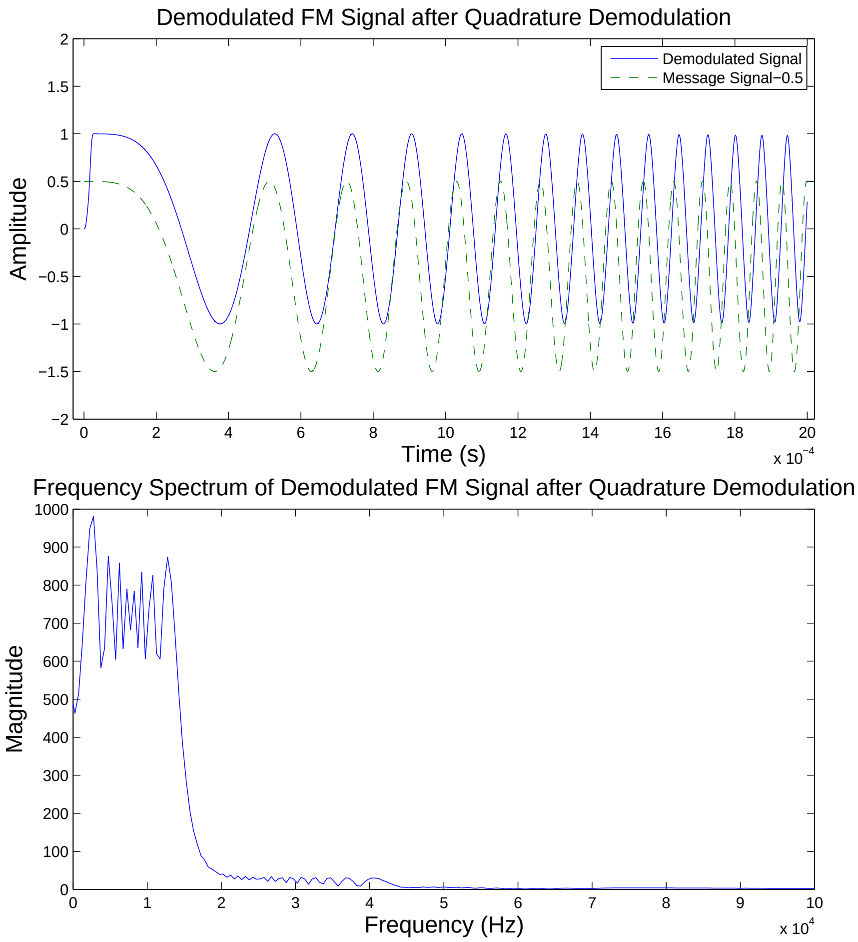

Figure 4.22: Result of demodulating FM signal after quadrature demodulation in time domain and frequency domain

Chapter 5

Reference Tables

This chapter provides the reference tables for different AM and FM demodulation methods.

5.1 Reference tables for AM signal demodulation

Table 5.1 describes the different AM signals and their demodulation methods.

Table 5.1: Different AM signals and their demodulation methods

Demodulation method	DSB TC AM signal	DSB SC AM signal	SSB AM signal
Envelope demodulation using filter	$\surd$	$\times$	$\times$
Envelope demodulation with Hilbert transform	$\surd$	$\times$	$\times$
Coherent demodulation	$\surd$	$\surd$	$\surd$
Square-law demodulation	$\surd$	$\times$	$\times$
Quadrature Demodulation	$\surd$	$\surd$	$\times$

$\surd$: the AM signal in the corresponding column can be demodulated by the method in the corresponding row.

$\times$: the AM signal in the corresponding column cannot be demodulated by the method in the corresponding row.

The principles of different AM signal demodulation methods are discussed in the Chapter 2, their computational complexity, pros and cons are summarized in Table 5.2 and Table 5.3.

Table 5.2: Computational Complexity of AM signal demodulation methods

Demodulation method	Computational Complexity
Envelope demodulation using filter	1 time absolute value taking; 1 time filtration
Envelope demodulation with Hilbert transform	1 time Hilbert transformation (filtration); 1 time absolute value taking
Coherent demodulation	1 time multiplication; 1 time filtration
Square-law demodulation	1 time multiplication; 1 time filtration; 1 time square root taking
Quadrature Demodulation	2 times multiplication; 2 times filtration; 2 times square taking; 1 times addition; 1 time root taking

Table 5.3: Pros and cons of AM signal demodulation methods

Demodulation method	Pros	Cons
Envelope demodulation using filter	1. It is easy to realize; 2. The receiving end doesn't need to know the carrier frequency; 3. It is the most commonly used in classical demodulators.	It can only demodulate the DSB TC AM signal.
Envelope demodulation with Hilbert transform	The receiving end doesn't need to know the carrier frequency.	It can only demodulate the DSB TC AM signal.
Coherent demodulation	It can demodulate the DSB TC AM signal, DSB SC AM signal and SSB AM signal.	1. The receiving end needs to know the carrier frequency; 2. If there exits phase difference between the carrier in the AM signal and the local carrier, the amplitude of the demodulated signal is multiplied with a cosine term.
Square-law demodulation	The receiving end doesn't need to know the carrier frequency.	It can only demodulate the DSB TC AM signal.
Quadrature Demodulation	1. The decreasing amplitude caused by phase difference in the coherent demodulation can be compensated; 2. It shares a similar design scheme with the FM quadrature demodulation.	1. The receiving end needs to know the carrier frequency; 2. The computational complexity of the quadrature demodulation is more than two times when compared with which of the coherent demodulation

From the simulation results in Chapter 4, the performance of the AM signal demodulation methods are concluded in Table 5.4.

Table 5.4: Comparison of AM signal demodulation results

Demodulation method	Low frequency	Medium frequency	High frequency	Out of information frequency
Envelope demodulation using filter	good	good	good	good
Envelope demodulation with Hilbert transform	good	good	good	good
Coherent demodulation	good	good	good	good
Square-law demodulation	good	good	good	good
Quadrature Demodulation	good	good	good	good

good: the demodulated signal can approximate the message signal very well.

satisfied: the demodulated signal shows a few ripples and / or a few distortion when compared with the message signal.

fair: the demodulated signal shows a lot of ripples and / or a severe distortion when compared with the message signal.

5.2 Reference tables for FM signal demodulation

The principles of different FM signal demodulation methods are discussed in the Chapter 2, their computational complexity, pros and cons are summarized in Table 5.5 and Table 5.6.

Table 5.5: Computational Complexity of FM signal demodulation methods

Demodulation method	Computational Complexity
FM to AM conversion with Hilbert transform	1 time differentiation; 1 time Hilbert transformation (filtration); 1 time absolutely value taking
FM to AM conversion using filter	1 time differentiation (subtract); 1 time absolutely value taking; 1 filtration;
Zero-crossing demodulation	1 time zero-cross detection; 1 time pulse generation; 1 time filtration;
Quadrature demodulation	2 times multiplication; 2 times filtration; 1 time division; 1 time arctan value taking; 1 time differentiation (subtraction);

Table 5.6: Pros and cons of FM signal demodulation methods

Demodulation method	Pros	Cons
FM to AM conversion with Hilbert transform	1. The receiving end doesn't need to know the carrier frequency; 2. It shares a similar design scheme with the Envelope demodulation with Hilbert transform for AM signal.	
FM to AM conversion using filter	1. It is easy to realize. 2. The receiving end doesn't need to know the carrier frequency. 3. It shares a similar design scheme with the Envelope demodulation using filter for AM signal; 4. It is the most commonly used in classical demodulator.	
Zero-crossing demodulation	The receiving end doesn't need to know the carrier frequency.	The higher demodulation accuracy needs a higher sampling frequency.
Quadrature demodulation	1. It shares a similar design scheme with AM quadrature demodulation. 2. It is the mostly commonly used in digital demodulator.	1. The receiving end needs to know the carrier frequency. 2. High computational complexity.

From the simulation results in Chapter 4, the performance of the FM signal demodulation methods are concluded in Table 5.7.

Table 5.7: Comparison of FM signal demodulation results

Demodulation method	Low frequency	Medium frequency	Medium frequency	Out of information signal frequency
FM to AM conversion with Hilbert transform	good	good	good	good
FM to AM conversion using filter	satisfied	fair	satisfied	fair
Zero-crossing demodulation	satisfied	fair	fair	fair
Quadrature demodulation	good	good	good	satisfied

good: the demodulated signal can approximate the message signal very well, only contain a slightly ripples.

satisfied: the demodulated signal shows a few ripples and / or a few distortion when compared with the message signal.

fair: the demodulated signal shows a lot of ripples and / or a severe distortion when compared with the message signal.

Chapter 6

Conclusions

In this thesis, two experimental systems for both AM signal demodulation and FM signal demodulation are designed and implemented. Five demodulation methods for AM signal and four demodulation methods for FM signal are selected. This thesis provides the researchers an easier reference of demodulation methods. As an integration of dispersed demodulation methods in literature, different demodulation methods for AM and FM are compared respectively, and their pros and cons are also stated in one composition.

In the simulation of AM and FM signal demodulation, two chirp signals are used as the message signal. Different demodulation methods are used to recover the message signal, and then the recovered signals are compared with the message signal.

For AM signal demodulation, all the demodulation methods do have good performance in both time domain and frequency domain.

For FM signal demodulation, all the demodulated signals approximate the message signal but with different precisions. In time domain, the results of FM to AM conversion with Hilbert transform and quadrature demodulation follow the message signal well. The results of FM to AM conversion using filter and zero-crossing demodulation have some ripples at the beginning of the demodulated signals, and their amplitude decrease with time. In frequency domain, the results of FM to AM conversion with Hilbert transform and quadrature demodulation are close to the message signal in the frequency range of the message signal. The results of FM to AM conversion using filter and zero-crossing demodulation show distortion in the frequency range of the message signal.

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