Lab 3: DSB Modulation

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Introduction

In this lab, we embarked on a comprehensive study of Double Sideband (DSB) Modulation and Demodulation. The experiment demonstrates how DSB modulation, by varying the amplitude of a high-frequency carrier wave according to the message signal, encodes information for transmission. We ventured into the realms of coherent demodulation, emphasizing the Square transformation method for carrier recovery, and delved into the precision of carrier synchronization, highlighting the critical role of Phase-Locked Loops (PLL) in ensuring the local oscillator's frequency and phase align with that of the received carrier signal.

Units

Time: s

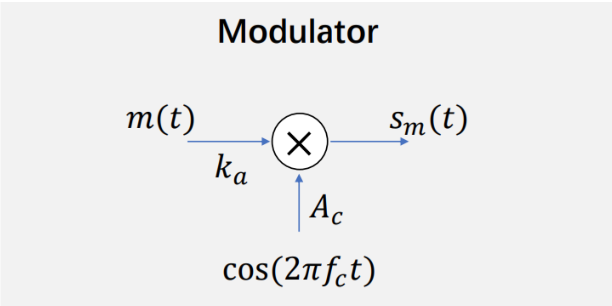
Frequency: Hz

Power: dBW

Amplitude: V

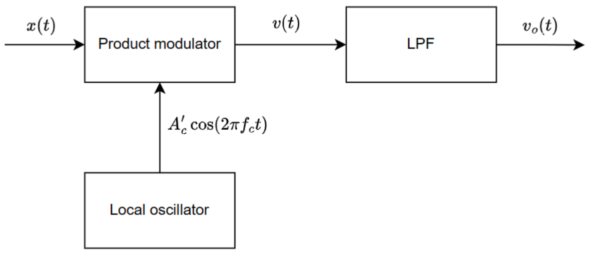
Basic Knowledge

DSB modulation



Double Sideband (DSB) modulation is a form of amplitude modulation where the amplitude of a high-frequency carrier wave is varied in accordance with the amplitude of a message signal. The principle can be described with the following equation: $s_m(t) = A_c(k_a m(t))\cos(2\pi f_c t)$, where $s_m(t)$ is the DSB modulated signal, A_c is the amplitude of the carrier, $m(t)$ is the message signal, f_c is the carrier frequency, and $\cos(2\pi f_c t)$ is the carrier wave.

Coherent demodulation

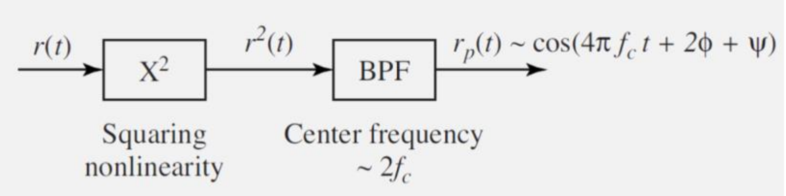


Coherent demodulation is a process where a received modulated signal is multiplied by a locally generated carrier that matches the original carrier in frequency and phase. This product is then passed through a low-pass filter (LPF) to extract the original message signal. The coherence refers to the precise synchronization of the local oscillator with the carrier signal.

Carrier synchronization

Carrier synchronization is the process of aligning the phase and frequency of a local oscillator with the phase and frequency of the received carrier signal to accurately demodulate the transmitted information.

Square transformation method



The square transformation method is a technique used for carrier recovery, a part of the demodulation process. It involves squaring the received modulated signal $r(t)$ which has the effect of doubling its frequency. The squared signal $r^2(t)$ contains components at twice the carrier frequency.

This signal is then passed through a band-pass filter (BPF) centered around $2f_c$ to isolate the component at this doubled frequency. The output $r_p(t)$ can be represented by a cosine function at the doubled frequency, $\cos(4\pi f_c t + 2\phi)$, where ϕ is the phase of the original signal.

Phase-locked loop

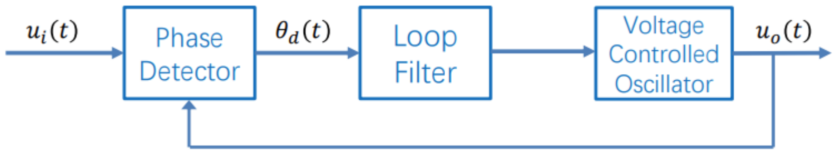
A Phase-Locked Loop is a control system that can synchronize the phase and frequency of an oscillator (local oscillator) with the phase and frequency of a reference signal or reference oscillator. The principle of PLL operation involves several key components and steps:

1.Phase Comparator: This component compares the incoming reference signal with the output signal of the local oscillator to detect any difference in phase between them.

2.Loop Filter: The output from the phase comparator, which is a voltage proportional to the phase difference, is passed through a loop filter. This filter removes high-frequency noise and produces a smooth control signal.

3.Voltage-Controlled Oscillator (VCO): The loop filter's output controls the frequency of the VCO. When the control signal varies, the frequency of the VCO adjusts in response, to minimize the phase difference with the reference signal.

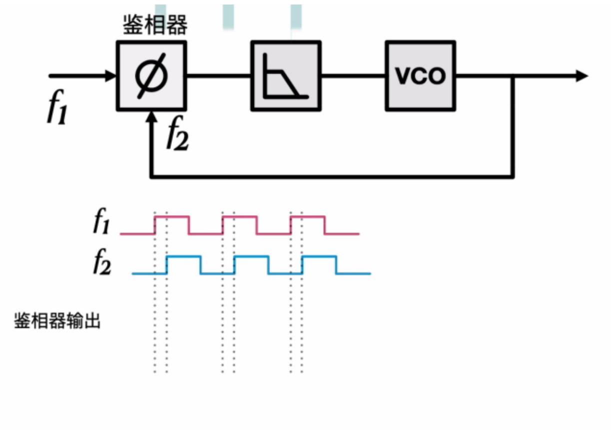
4.Feedback Loop: The output from the VCO is fed back into the phase comparator for a new round of comparison, thus creating a closed-loop control system.



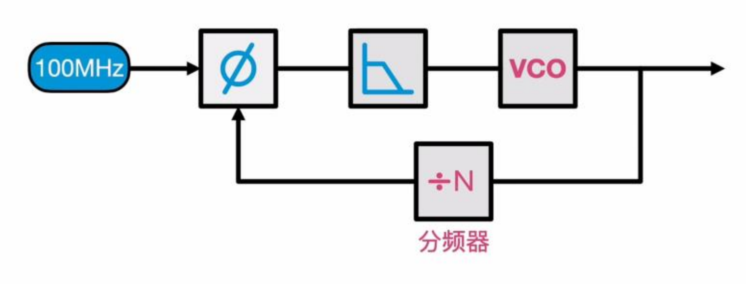
$$\theta_d(t) = (\omega_i - \omega_o)t + \theta_i - \theta_o$$

$$\omega_i - \omega_o = 0$$

The main purpose of a PLL is to synchronise the output of the oscillator with a reference signal. Even if the two signals have the same frequency, their peaks and troughs may not be in the same place, in other words, the two signals may have different phases. A PLL reduces the phase error between the output and input frequencies. When the phase difference between these signals is zero, the system is said to be "locked".



We can use a frequency divider to control the output frequency in relation to the input frequency.



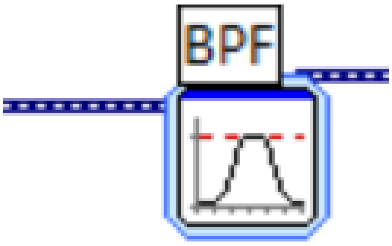
$$f_{ref} = \frac{f_{out}}{N}$$

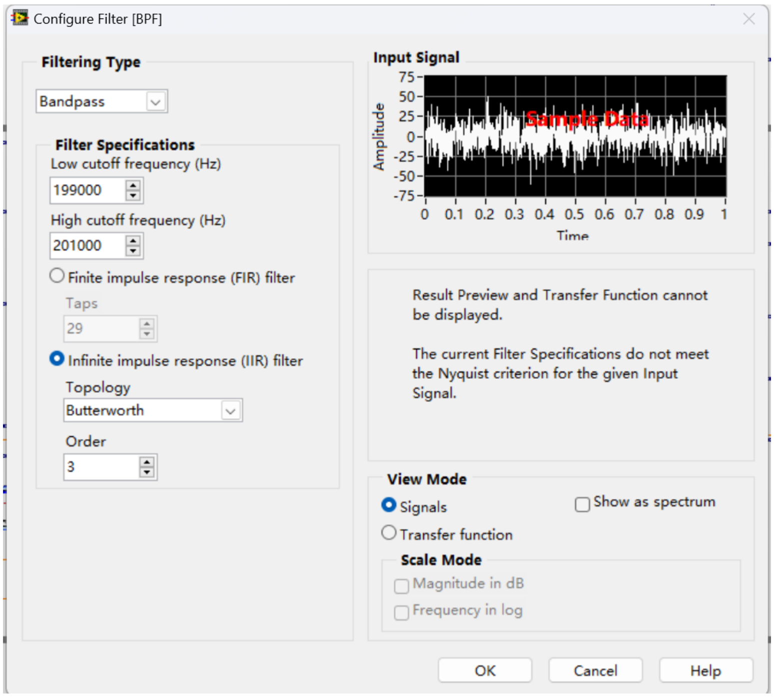
LabVIEW Express Modules

Frequency doubler

By using the square operation module, we transform the modulated signal $s_m(t) = m(t)\cos(2\pi f_c t)$ to $s(t) = \frac{1}{2}m^2(t) + \frac{1}{2}m^2(t)\cos(4\pi f_c t)$. This operation doubles the carrier frequency, and it also makes the power at frequency $2f_c$ become the highest(similar to autocorrelation) , making it easier to distinguish the carrier frequency using Array Max & Min module.

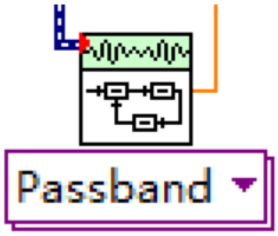
Bandpass filter





The band-pass filter (BPF) is centered around $2f_c$ to isolate the component at this doubled frequency.

Phase Lock Loop (PLL)



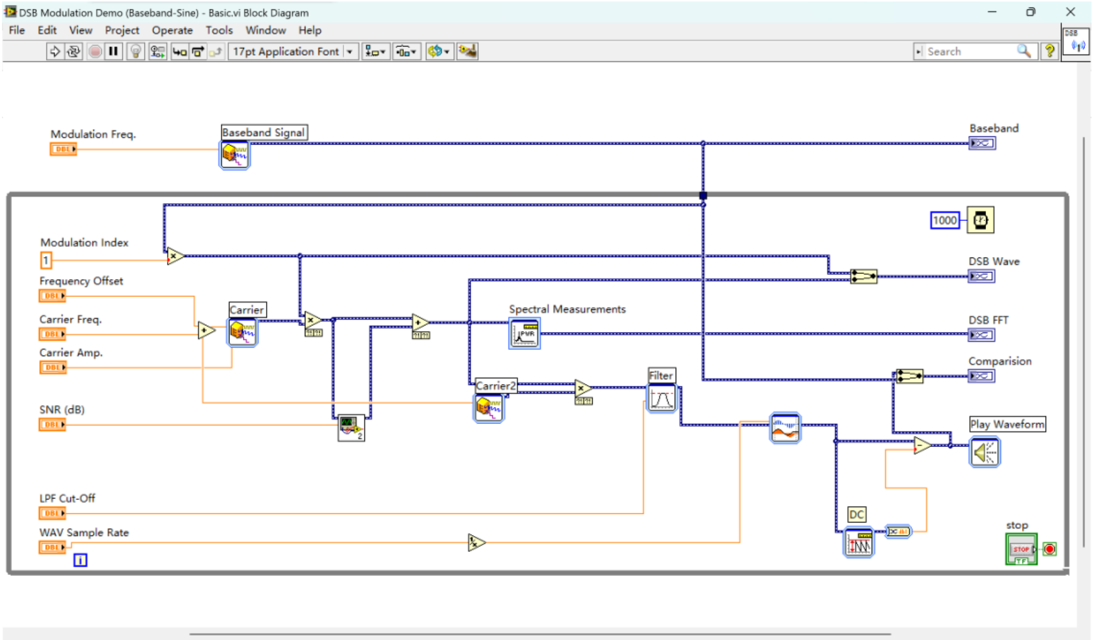
Simply input the signal after the bandpass filter, and the module will output the carrier frequency, facilitating carrier synchronization.

Lab results & Analysis:

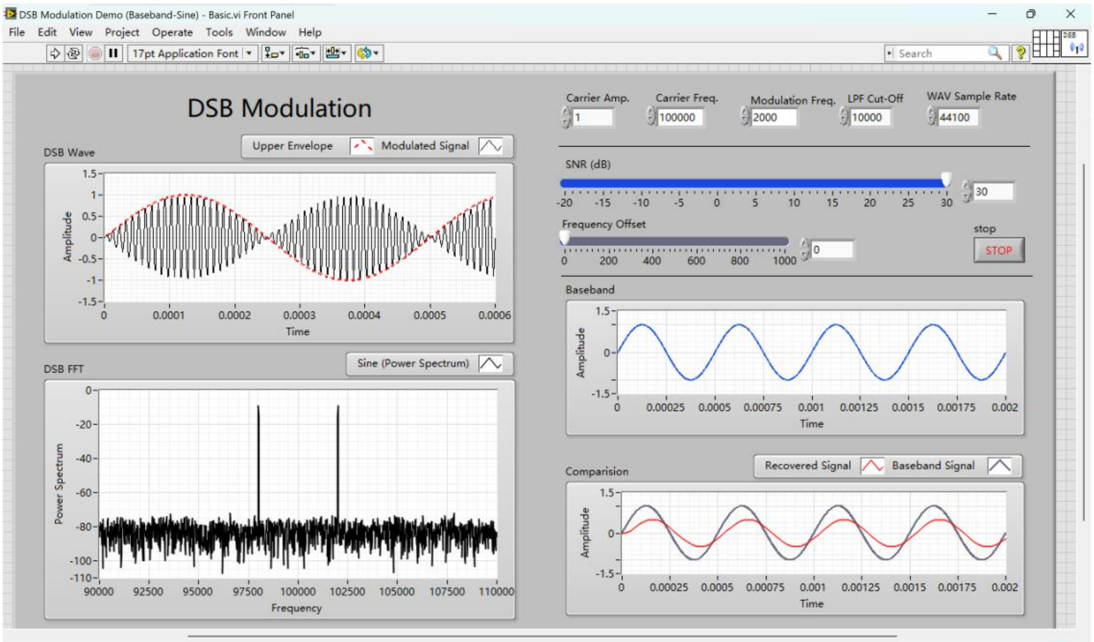
DSB modulation / demodulation simulation (block diagram, programming process,

simulation results), single-tone test and voice test

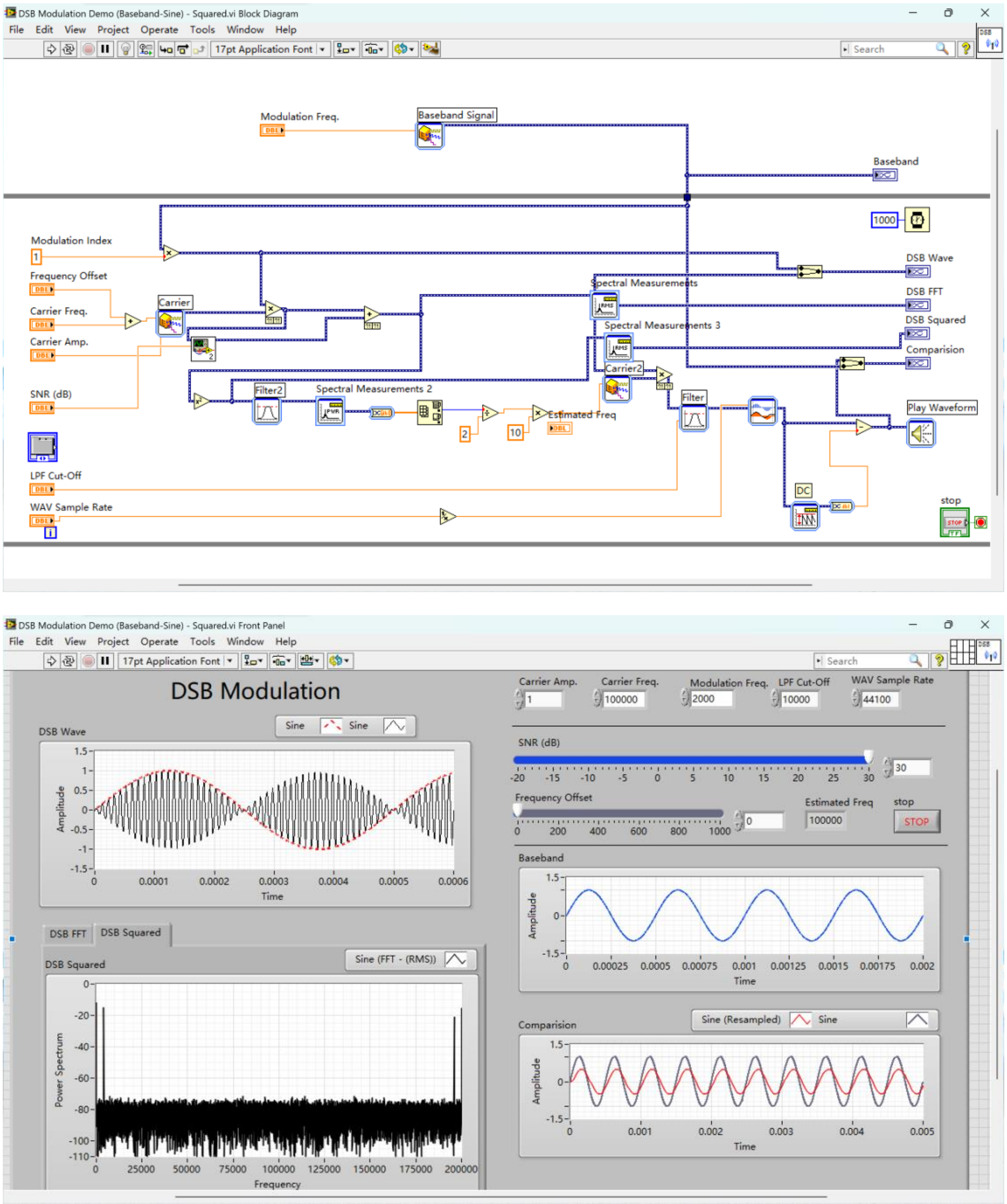
Basic modulation and demodulation: Block Diagram



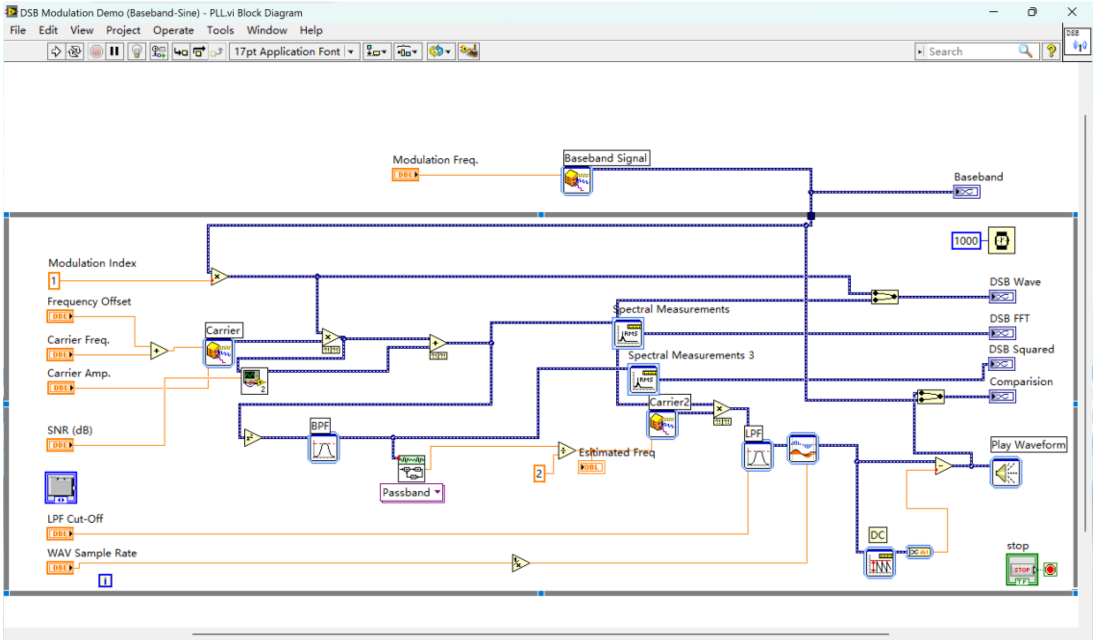
Basic modulation and demodulation: Front Panel(Result)

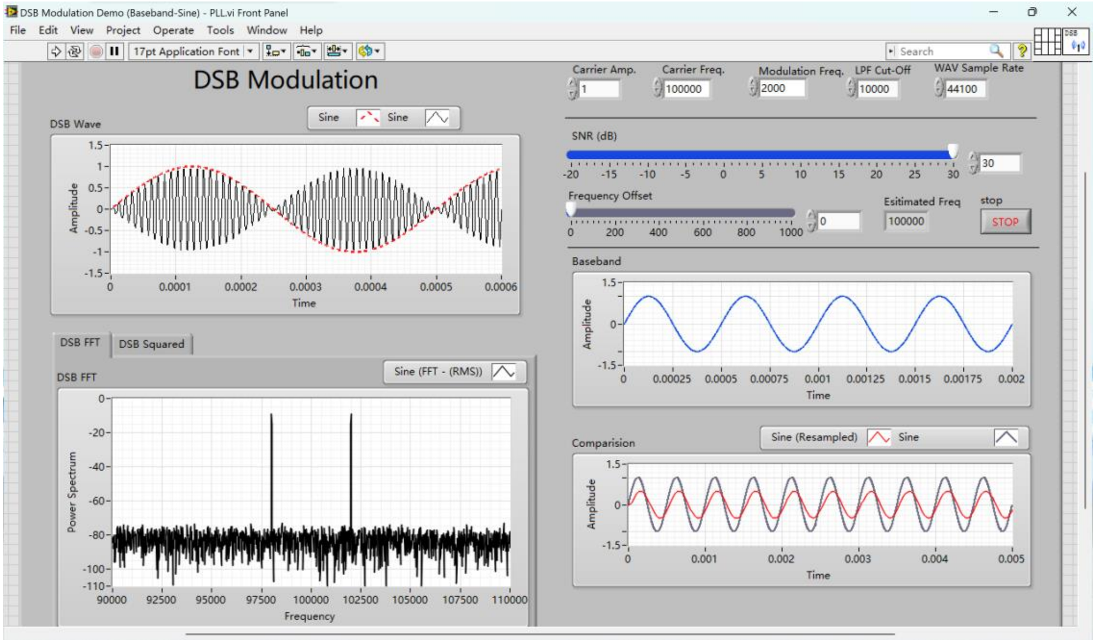


Modulation and Demodulation, by square method to determine demodulating frequency



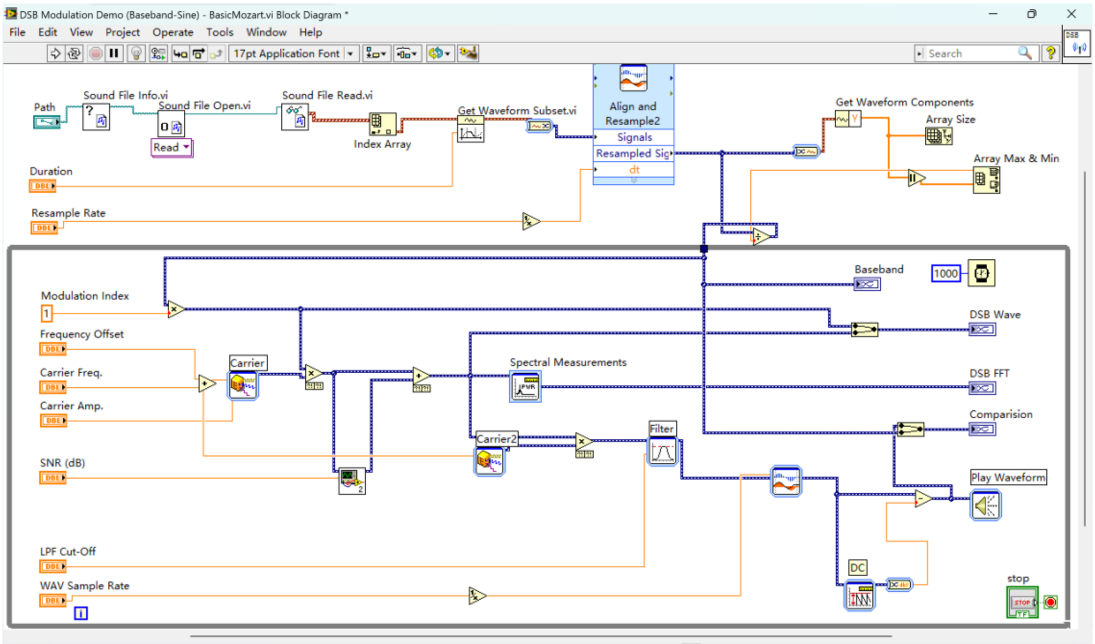
Modulation and Demodulation, by Phase Locked Loop

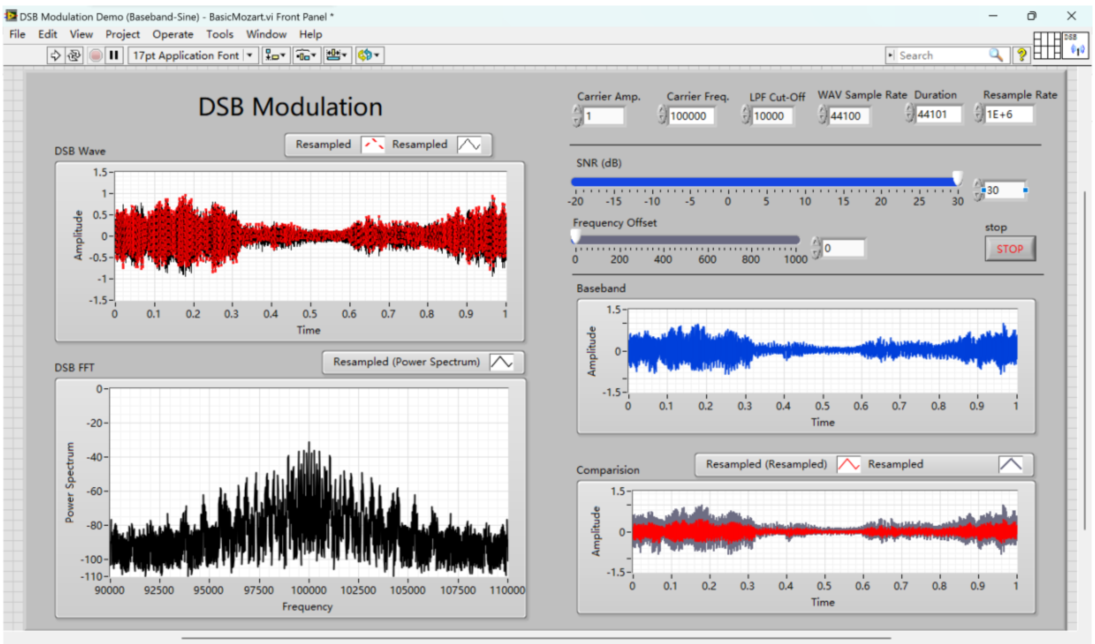




Above result are all single tone test.

Voice Test(Mozart) Modulation and Demodulation

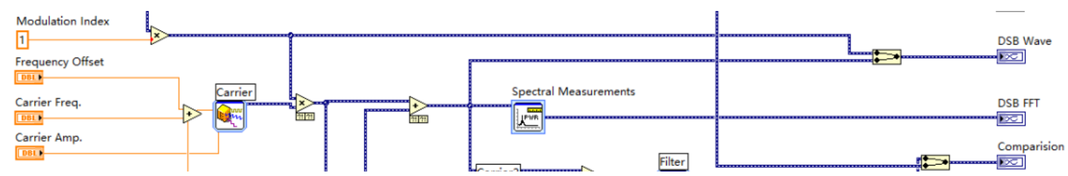




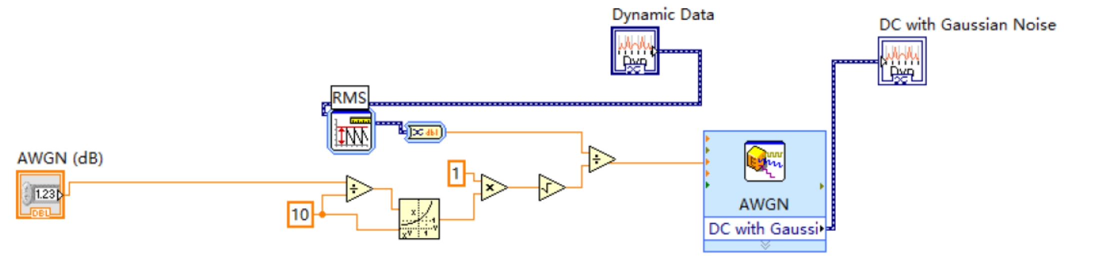
Programming Process:

Construct single tone

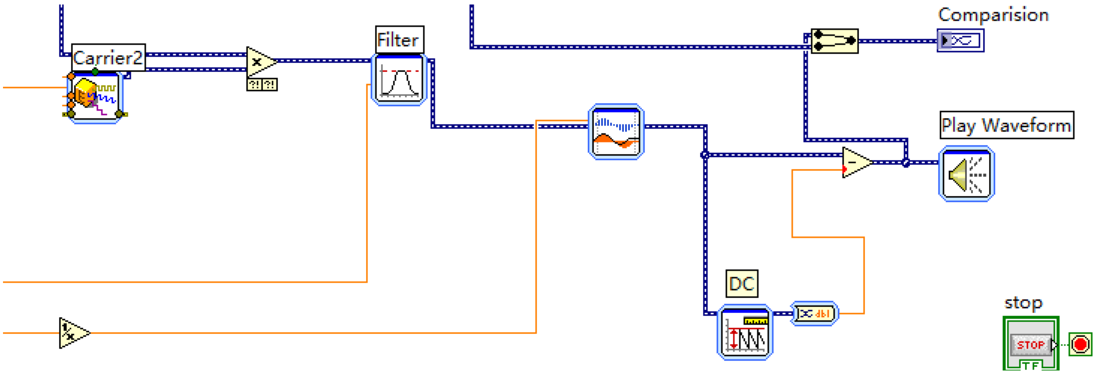
Modulation



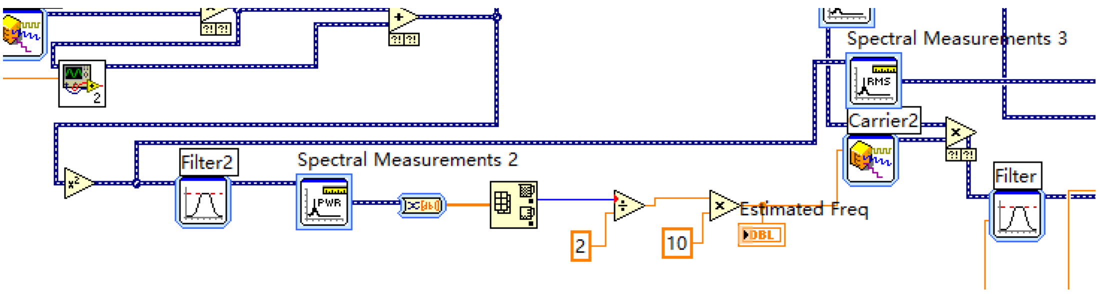
Generate noise(in subvi)



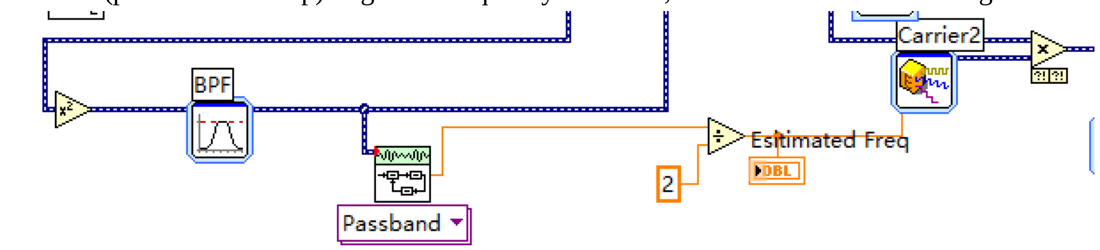
Demodulation



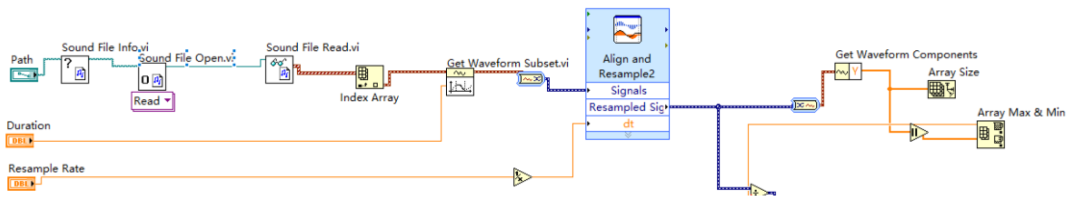
Use squared method to get the frequency of carrier, and then demodulate the signal



Use PLL(phase locked loop) to get the frequency of carrier, and then demodulate the signal



Generate voice signal



Analyse the factors affecting DSB modulation system (frequency deviation, cut-off

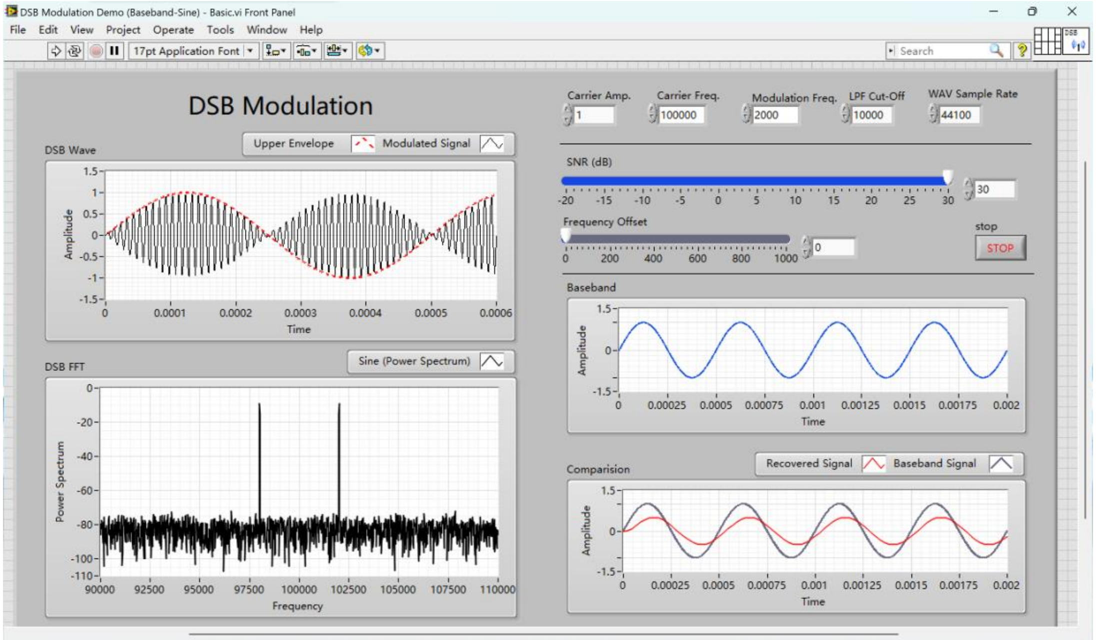
frequency of low-pass filter, order)

For convenience, use single tone signal to compare the impact of each parameters.

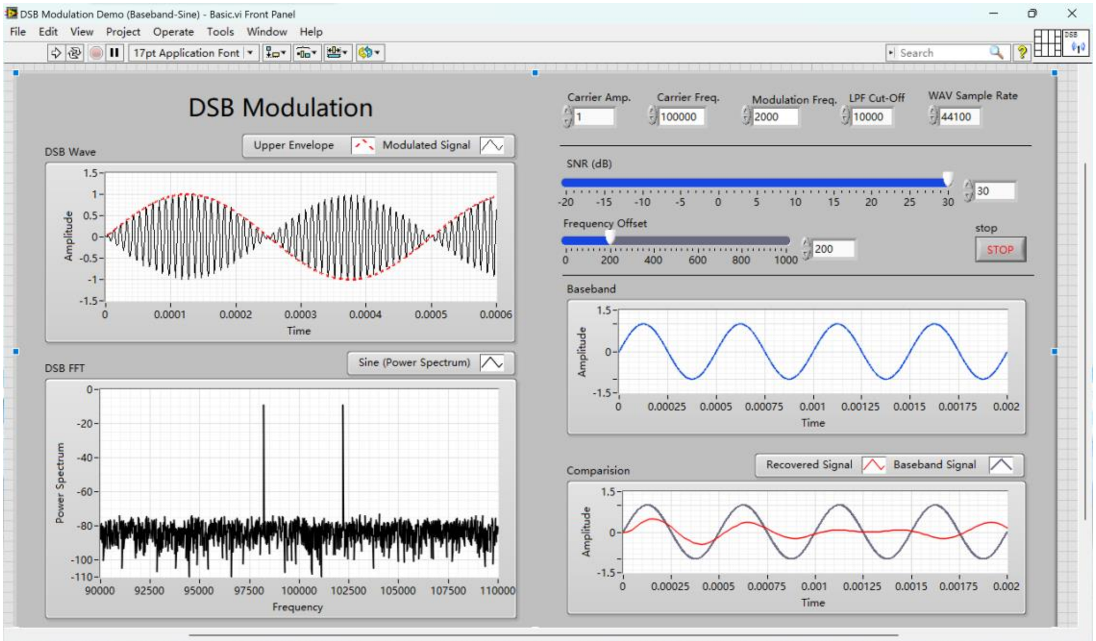
Frequency Deviation:

First, show the result of traditional way (directly use carrier frequency without offset)

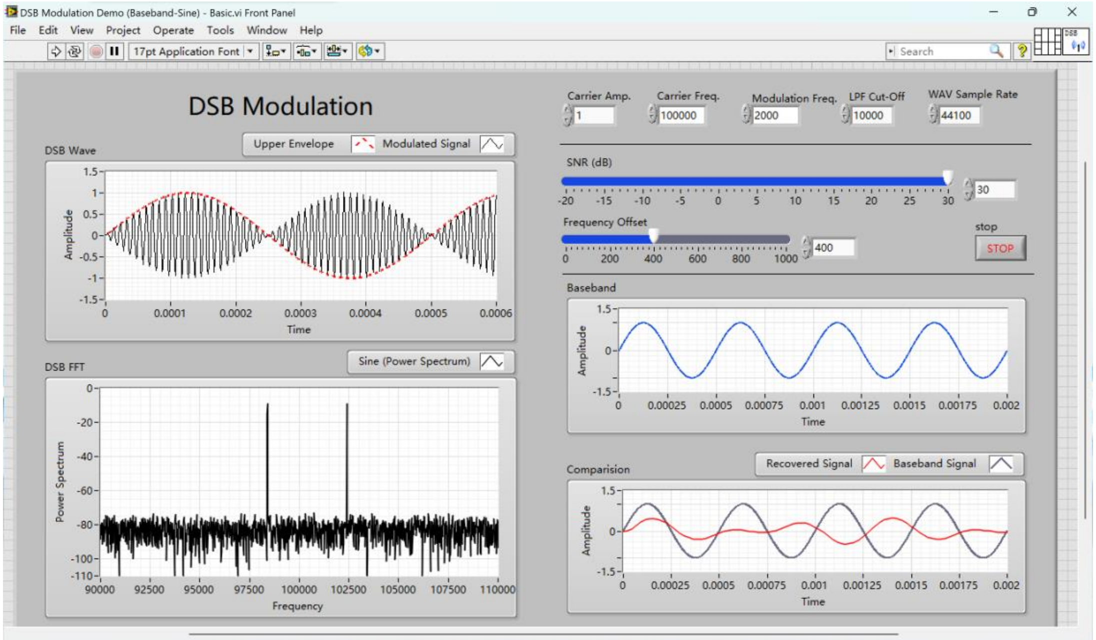
Frequency Offset = 0Hz



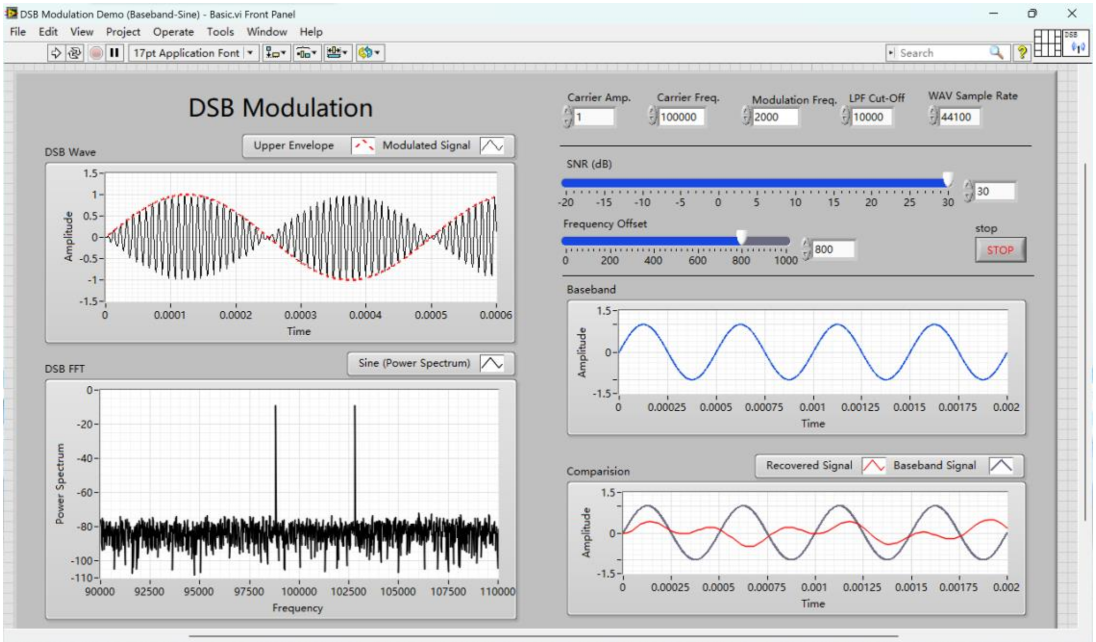
Frequency Offset = 200Hz



Frequency Offset = 400Hz



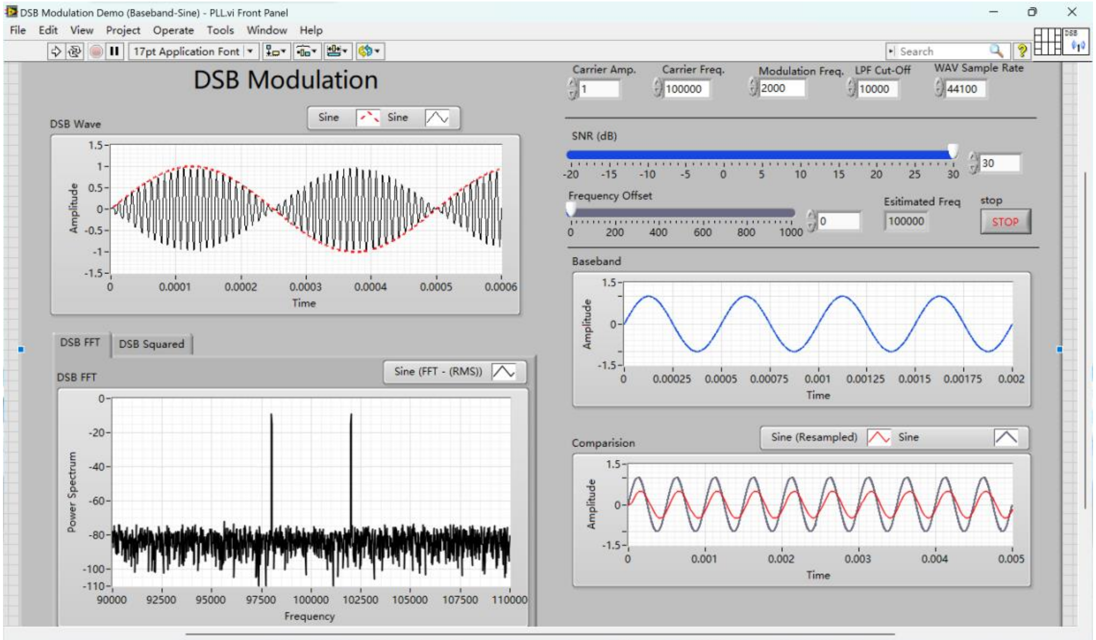
Frequency Offset = 800Hz



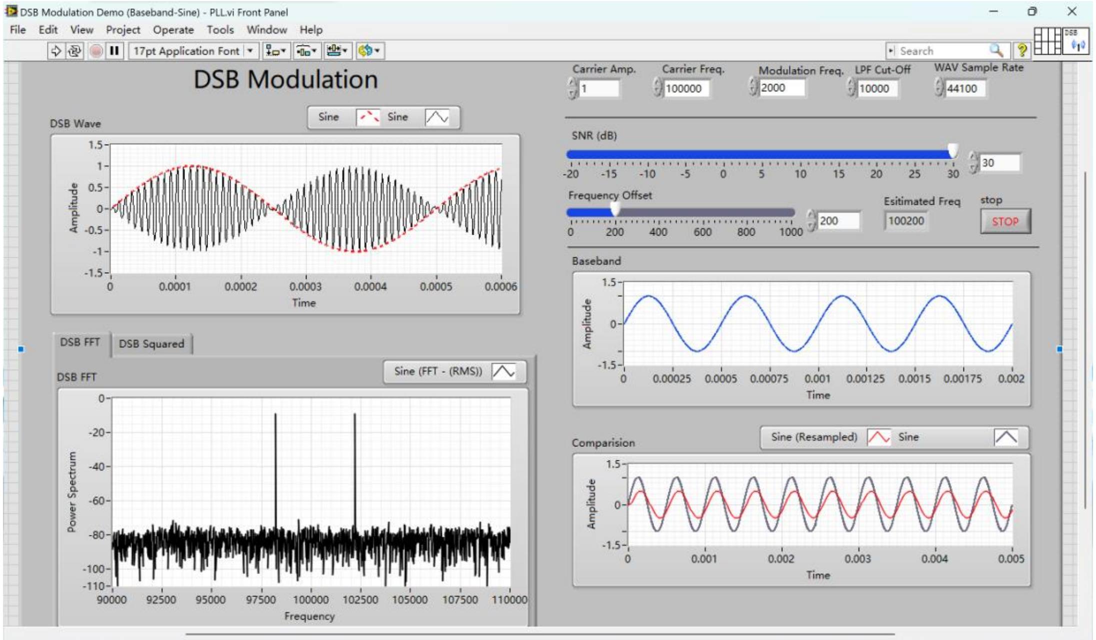
We can find that it’s hard to demodulate the signal if there exist considerable frequency offset

Then, we can try the method of estimating frequency(take PLL for example)

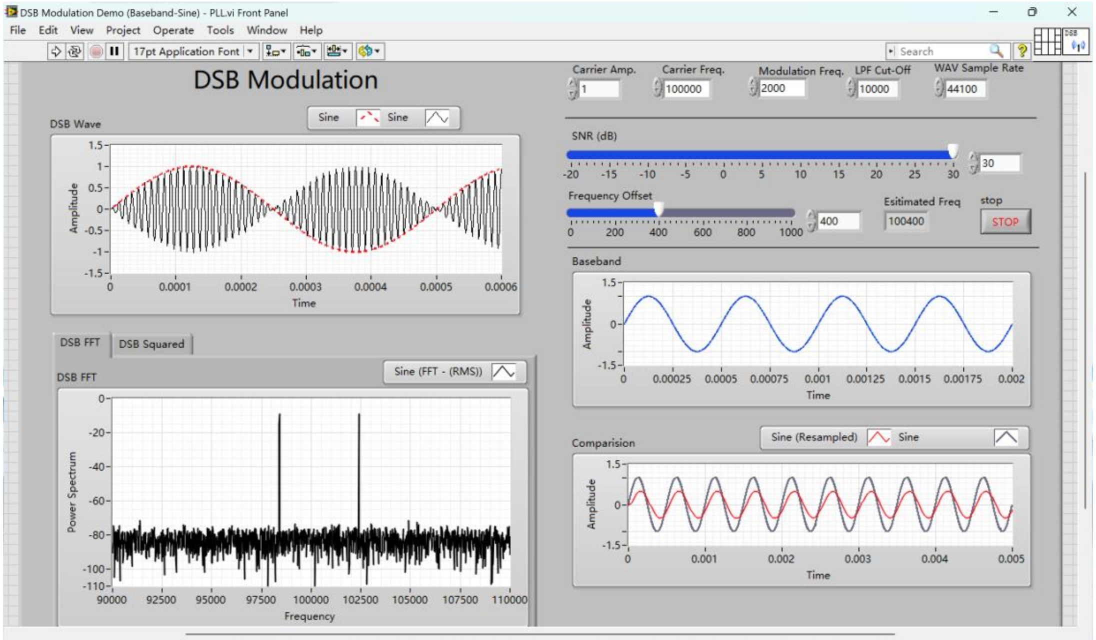
Frequency Offset = 0 Hz



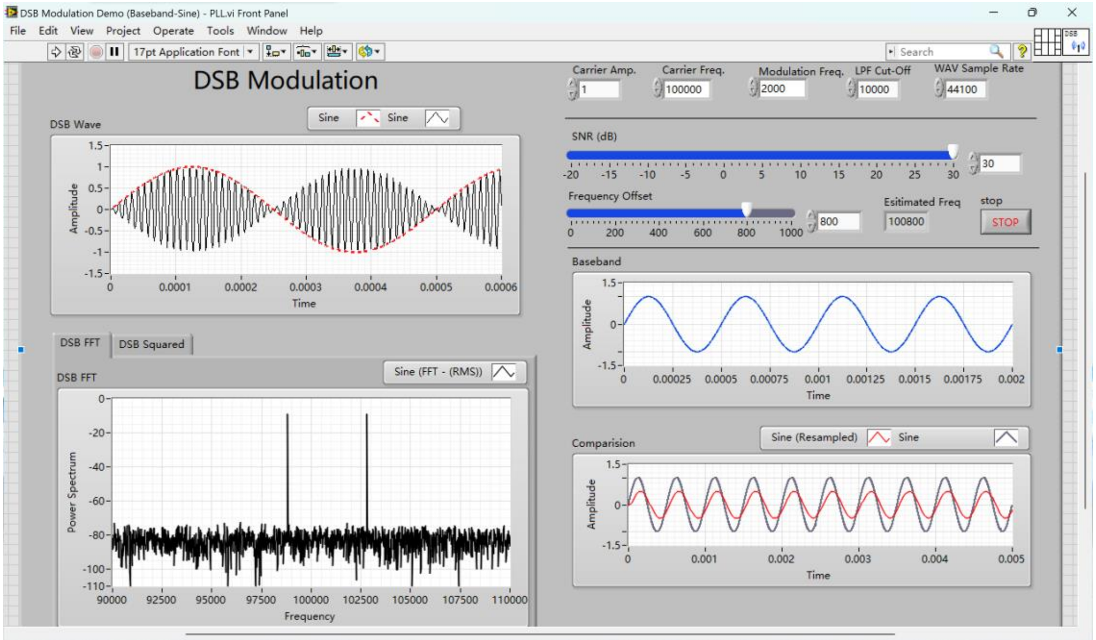
Frequency Offset = 200 Hz



Frequency Offset = 400 Hz



Frequency Offset = 800 Hz

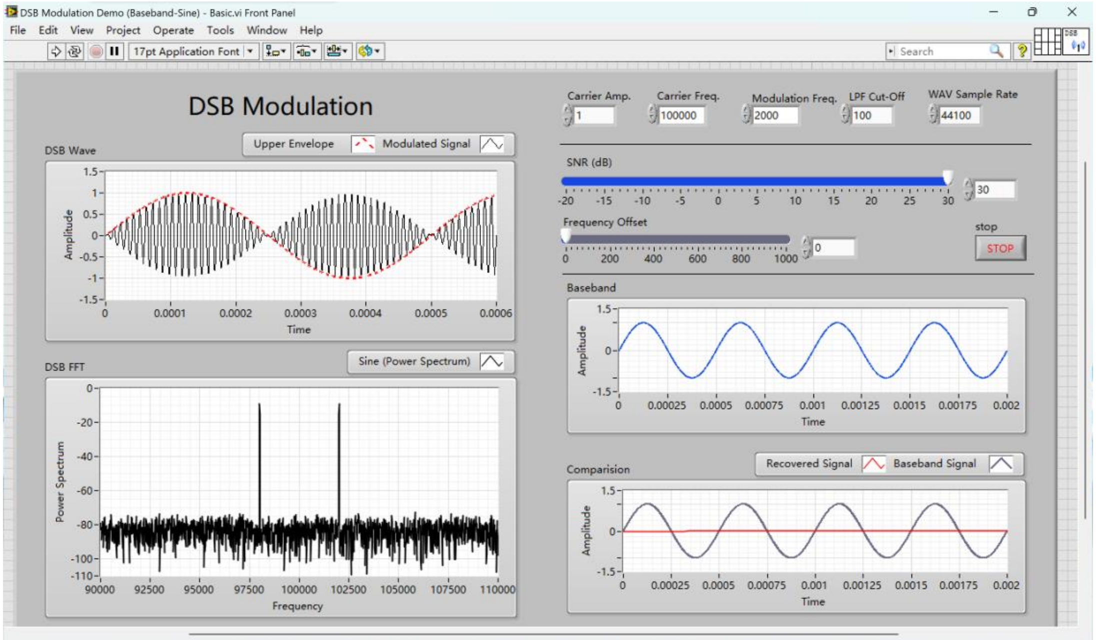


Using such method, we can still demodulate the signal, because the estimated frequency can take frequency offset into consideration.

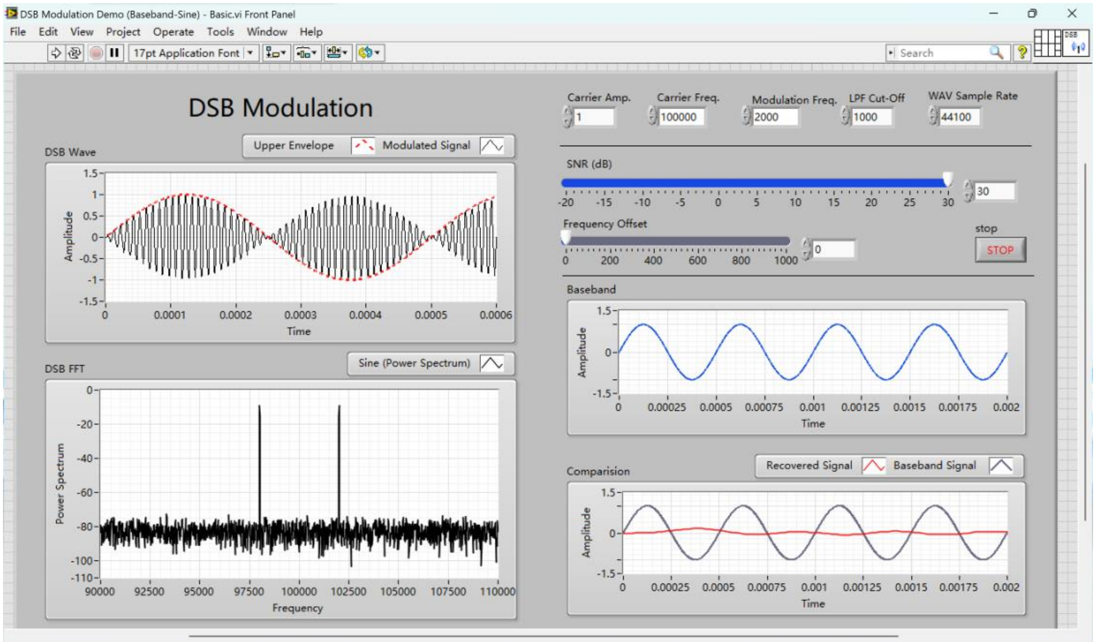
Cut-off frequency of LPF

Again, use single tone signal and basic way of demodulation.

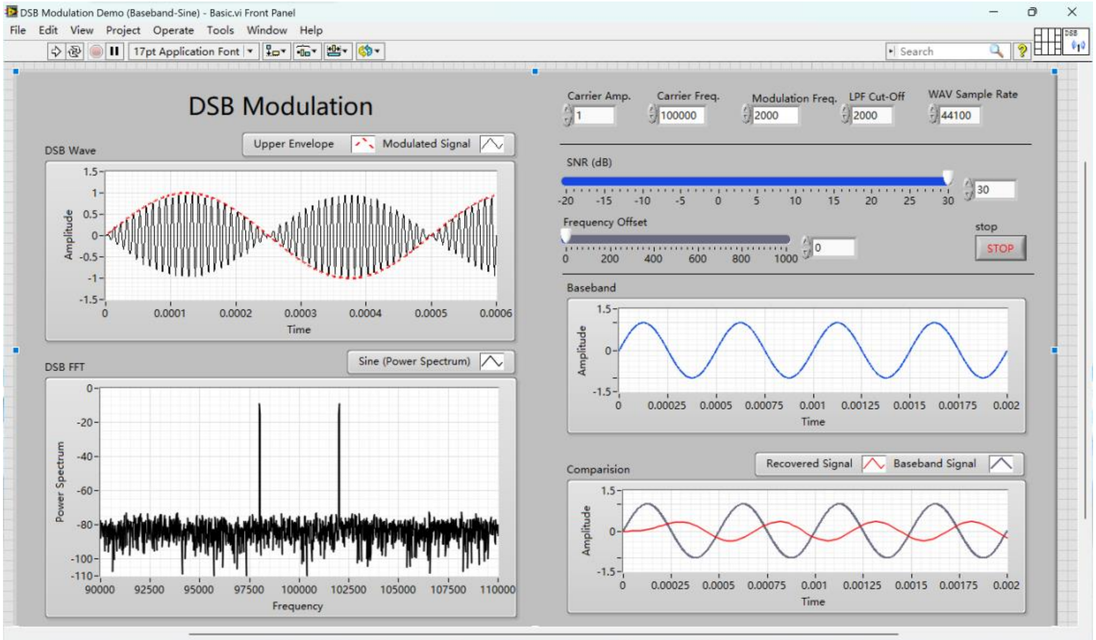
LPF Cut-off = 100Hz



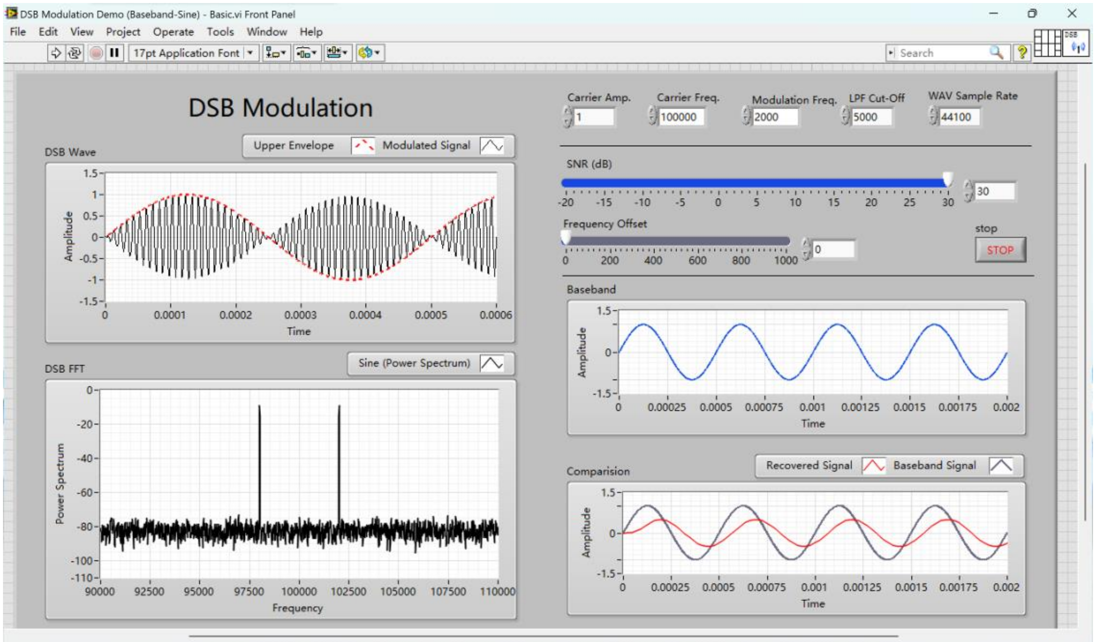
LPF Cut-off = 1000Hz



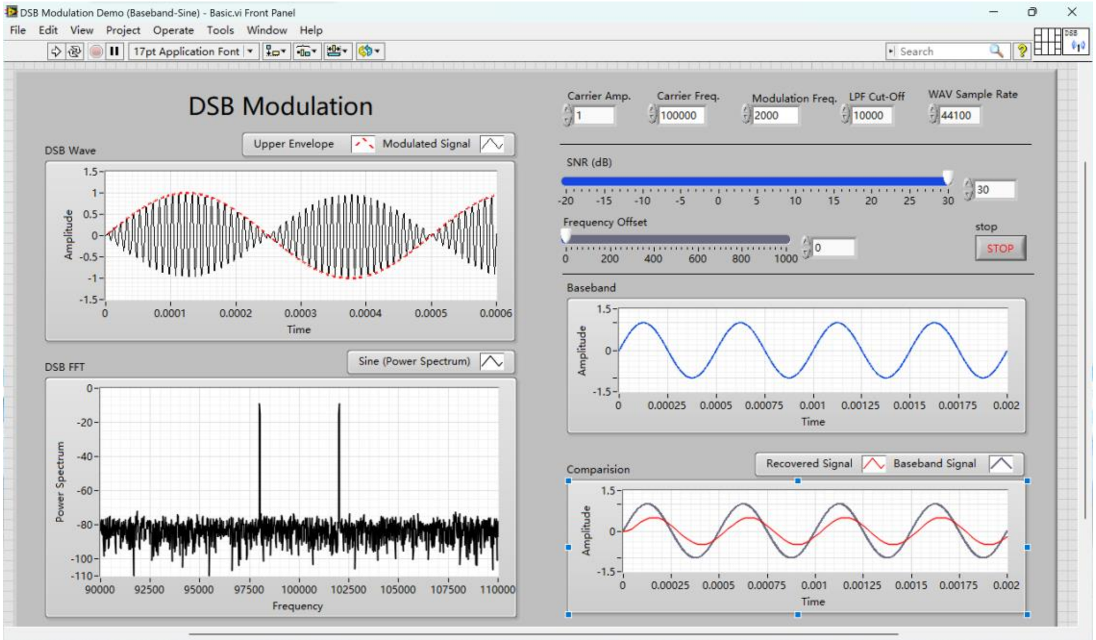
LPF Cut-off = 2000Hz



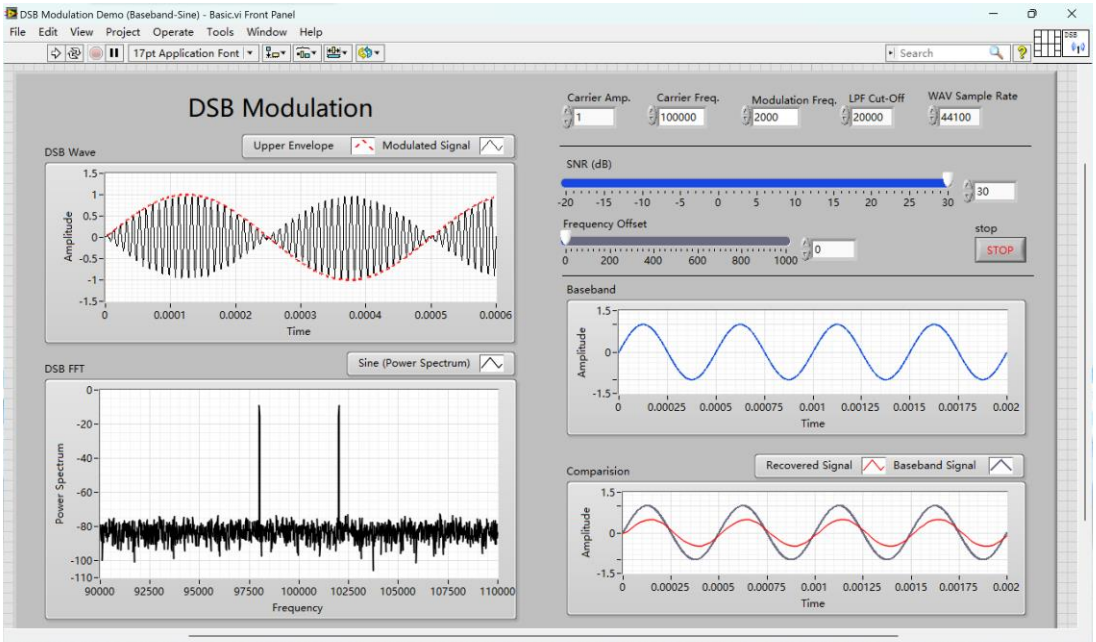
LPF Cut-off = 5000Hz



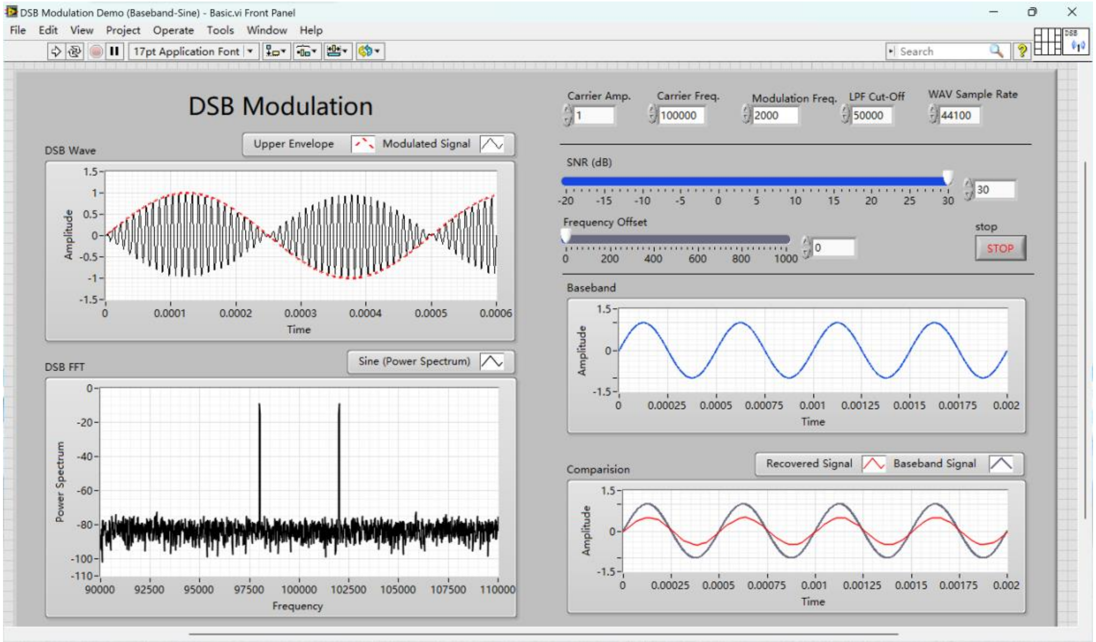
LPF Cut-off = 10000Hz



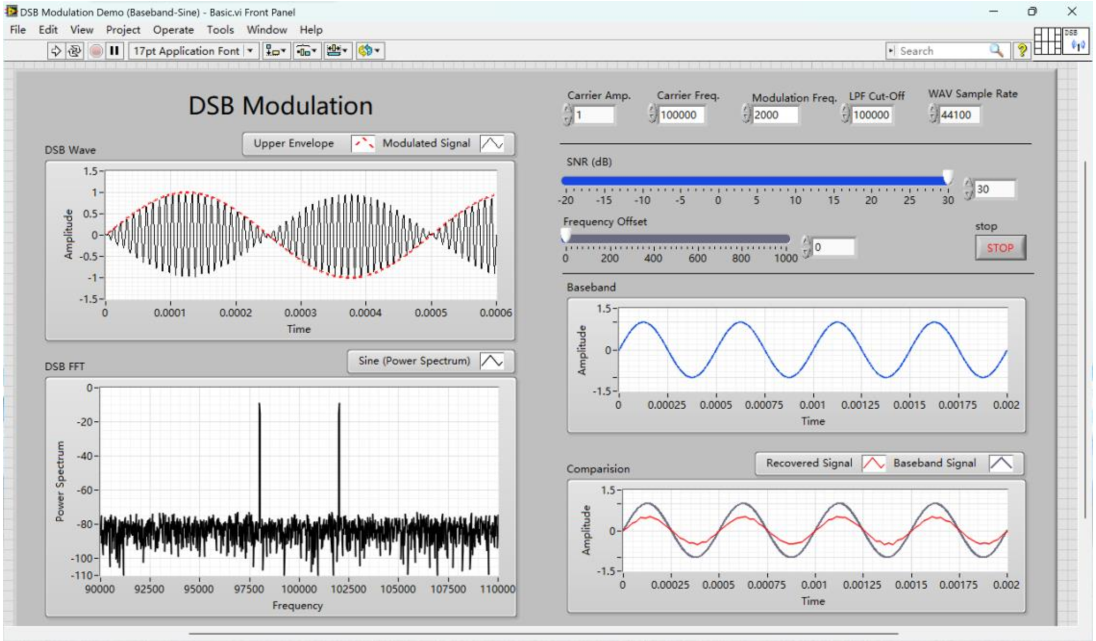
LPF Cut-off = 20000Hz



LPF Cut-off = 50000Hz



LPF Cut-off = 100000Hz

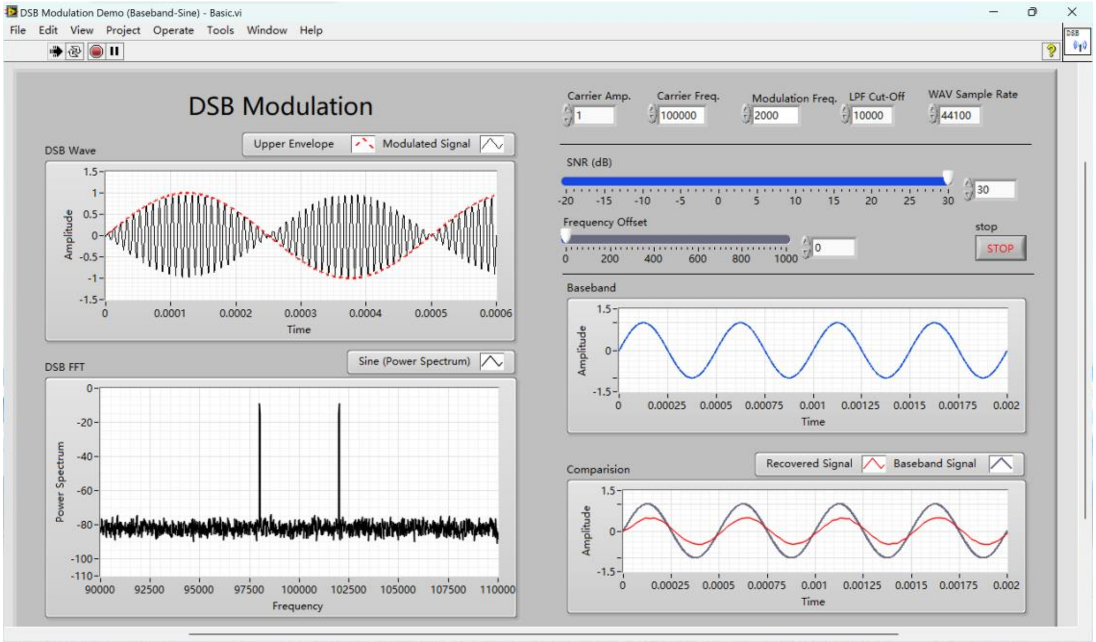


We can find that the demodulation works best when LPF Cut-off frequency is about 5000-20000Hz. If LPF Cut-off frequency is too small, the signal will be filtered as well, so it can't recover the signal. If LPF Cut-off frequency is too large, the noise can also pass the filter, so the result may not performs so well.

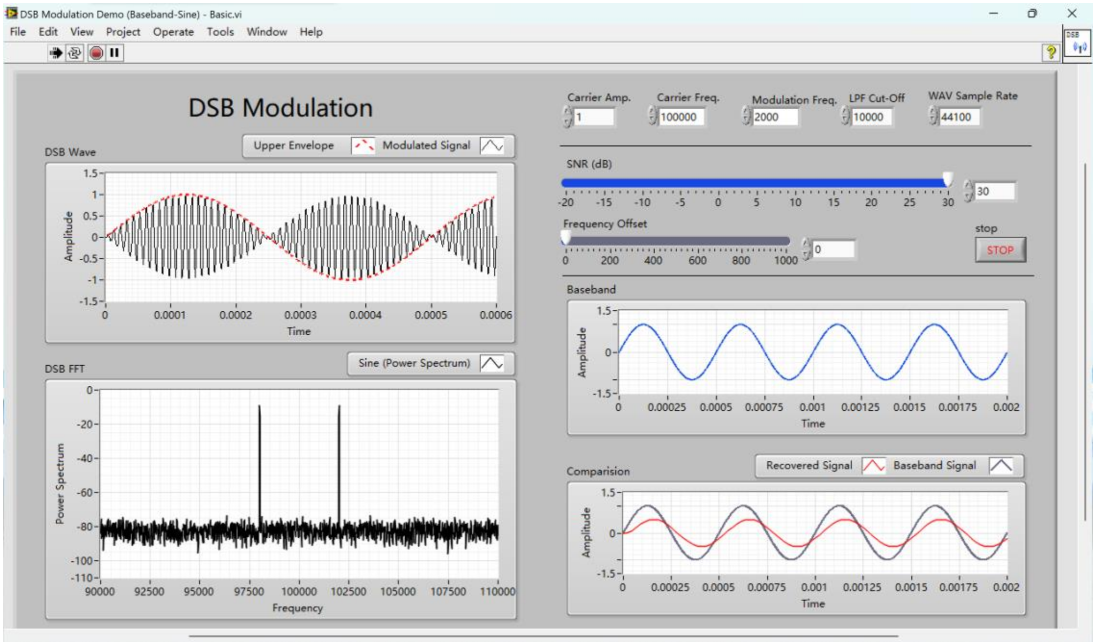
Order of LPF

We still use single tone signal, basic way of demodulation, and we choose Cut-off frequency of LPF to be 10000Hz this time to analyze the impact of order of LPF.

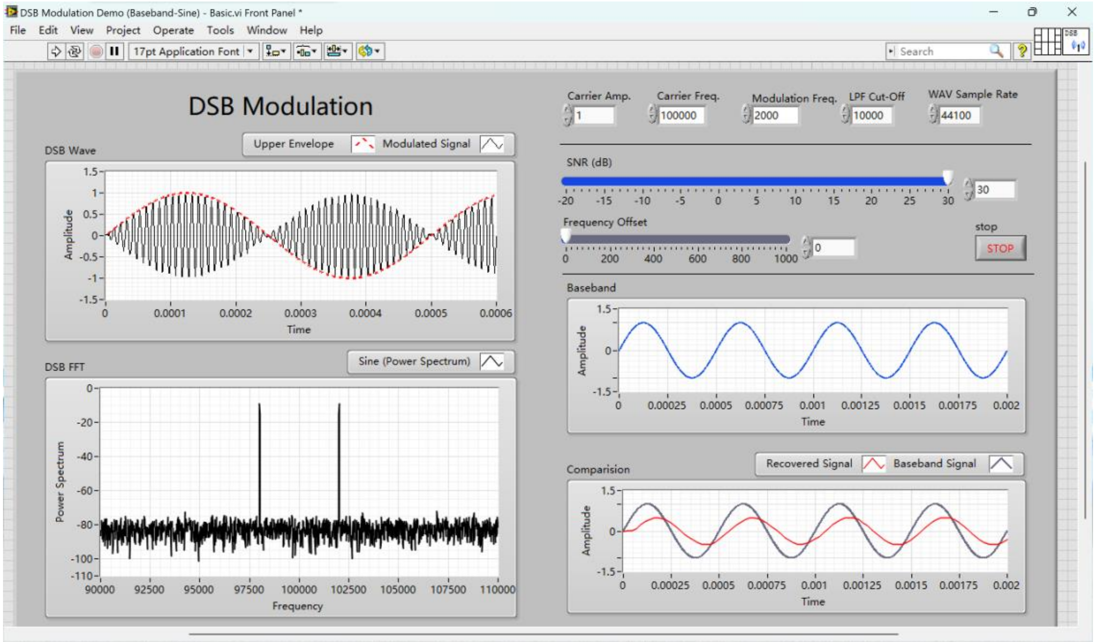
Order = 1



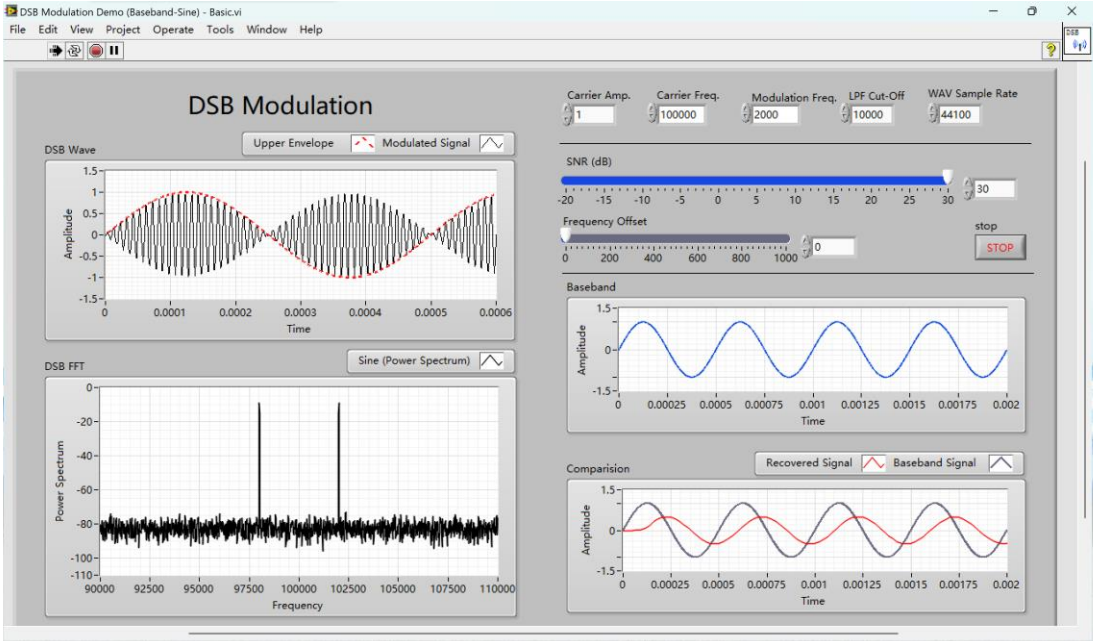
Order = 3



Order = 5



Order = 10



We find little difference when order is larger than 3, so for cost concern, we can choose order to be 3 in application.

Analyse the advantages and disadvantages of DSB modulation system, and explain its social application value.

Advantages:

DSB modulation is relatively inexpensive to implement.

Compared to AM modulation, DSB modulation does not transmit a carrier and therefore has a higher spectrum utilization.

The demodulator of DSB modulation is relatively simple and the original information can be easily extracted from the signal.

Disadvantages:

DSB modulation transmits two identical sidebands, which makes it less spectrally efficient than other modulation techniques.

Since DSB modulation transmits two complete sidebands, it requires more power as compared to single sideband modulation.

DSB modulation is not as effective as other modulation techniques (e.g. FM modulation) for long distance transmission due to low transmission efficiency and power wastage.

Application:

The biggest advantage of DSB modulation is that it is cheap, simple and easy to demodulate, so it is still used in some low-cost signal processing scenarios. Before SSB and FM modulation were widely used, DSB modulation was heavily used in radio and TV.

Experience

Final Block Diagram

It can be seen in the result & analysis part.

Gains

Engaging with this experiment on DSB modulation has deepened our practical understanding of signal manipulation and reinforced theoretical principles discussed in academic settings. Our experimentation highlighted the pivotal influence of parameters like frequency deviation and the low-pass filter's cutoff frequency on the integrity of signal transmission. Through hands-on experience, we observed how these factors affect the modulation process and the effectiveness of demodulation techniques. Furthermore, the project honed our analytical skills, particularly in employing LabVIEW's sophisticated modules for signal creation and examination. These competencies, including the insightful analysis of modulation systems' strengths and limitations, are invaluable assets for our future endeavors in the realm of communication systems and signal processing.

Contributions

Chen Weiyu: completed the basic modulation and demodulation process, wrote the introductory section.

Ying Yiwen: completed the basic modulation and demodulation process, as well as the modulation and demodulation process using mozart audio, wrote the conclusion and analysis

sections.	
Score	100

字体：英文 Times new Roman；中文宋体，正文五号

文件名统一命名方式：**LabX+姓名+学号，例如：Lab1+张三+00001 （正式报告删除此行！）**