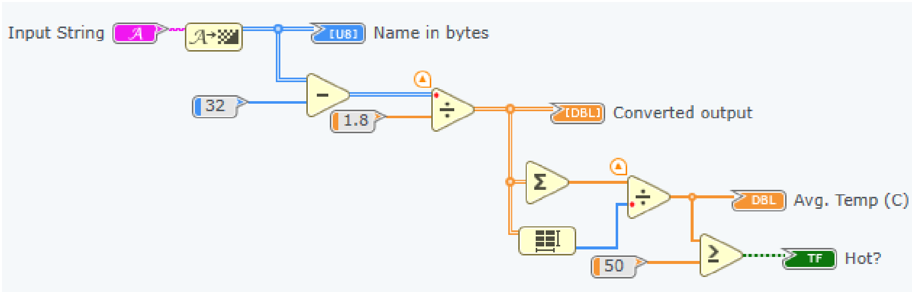
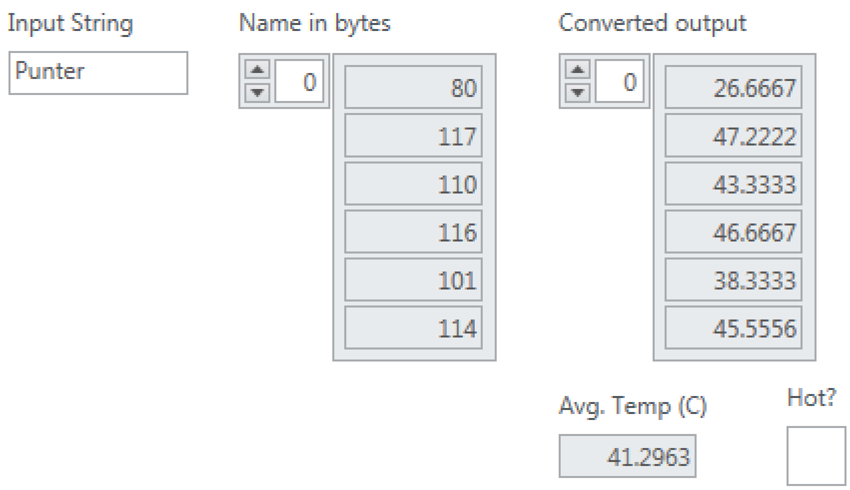
Lab 1 - Introduction to LabView

Exercise 1 - Data Types in LabView

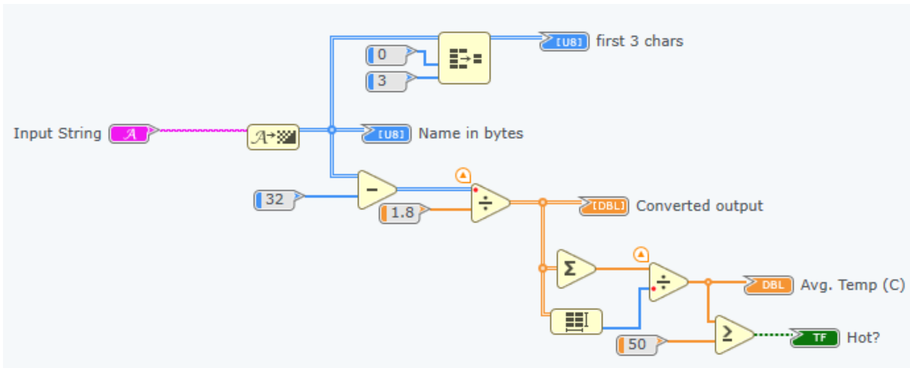
Block Diagram - Convert name into byte value, then convert from Fahrenheit to Celsius



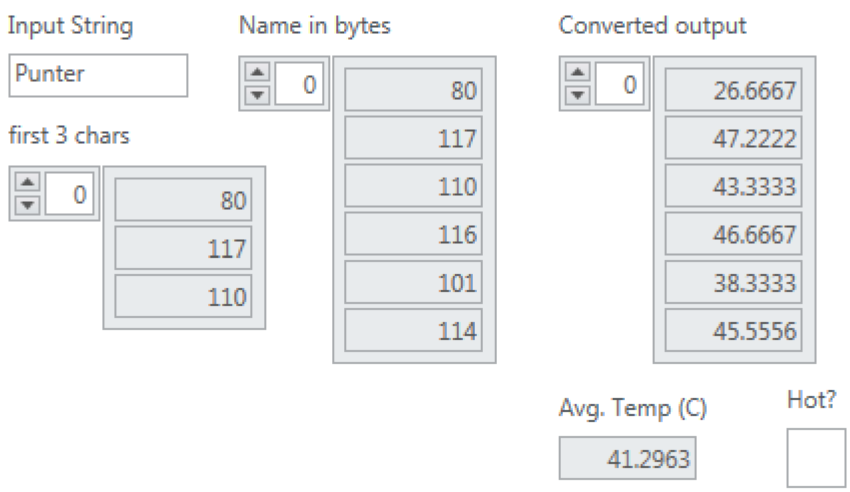
Panel - Running conversion of name from Fahrenheit to Celsius



Block Diagram - Get integer value of first three characters



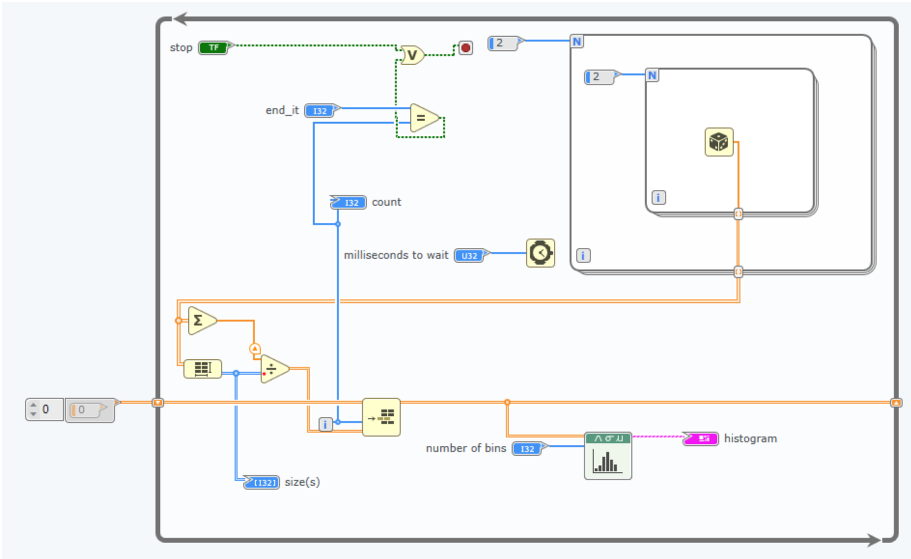
Panel - Running integer value of first three characters



Exercise 2 - Implementation of the Central Limit Theorem

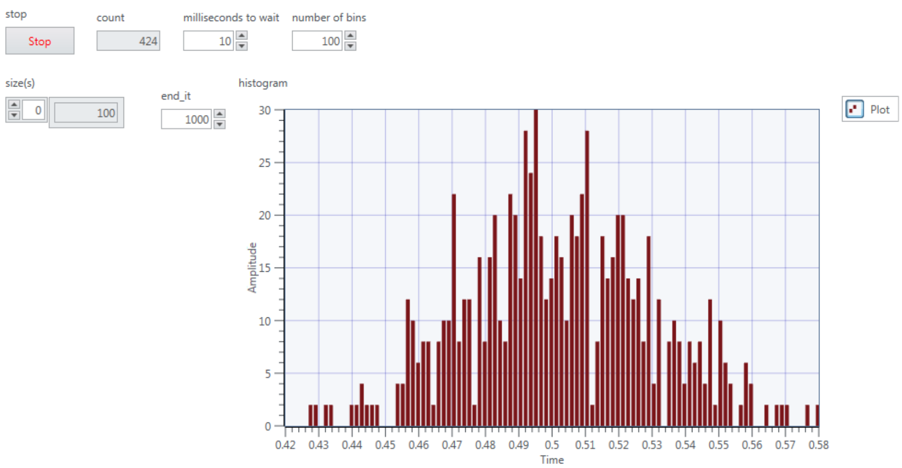
- First, we created a uniform distribution generator.
- By repeatedly generating uniformly distributed random numbers, we observed the sample mean converging to a normal distribution.

Diagram 1



Output:

Histogram 1



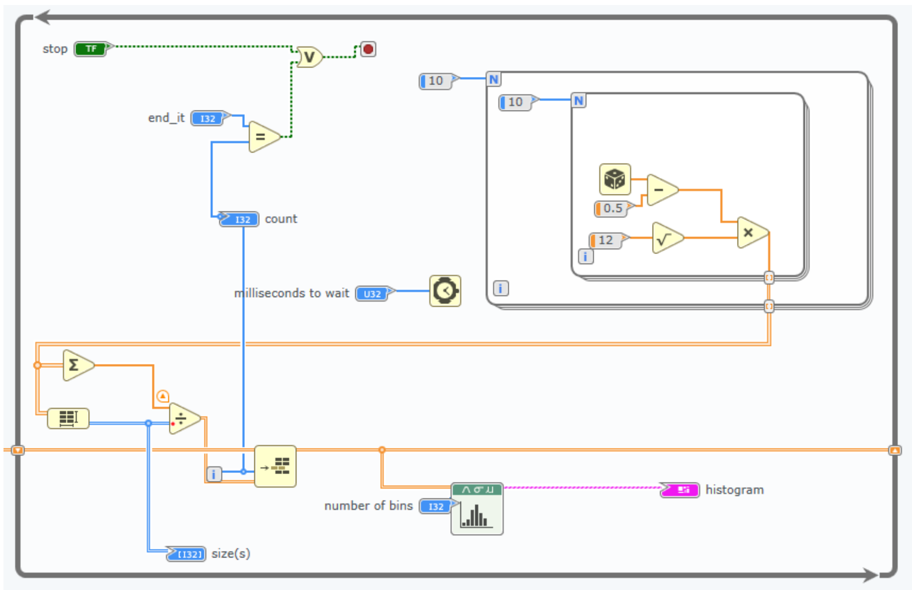
Task: to standardize the normal distribution produced by the Central Limit Theorem

- Subtracted the mean from the random variable, and then dividing by the standard deviation.

$$z = \frac{x - \mu}{\sigma}$$

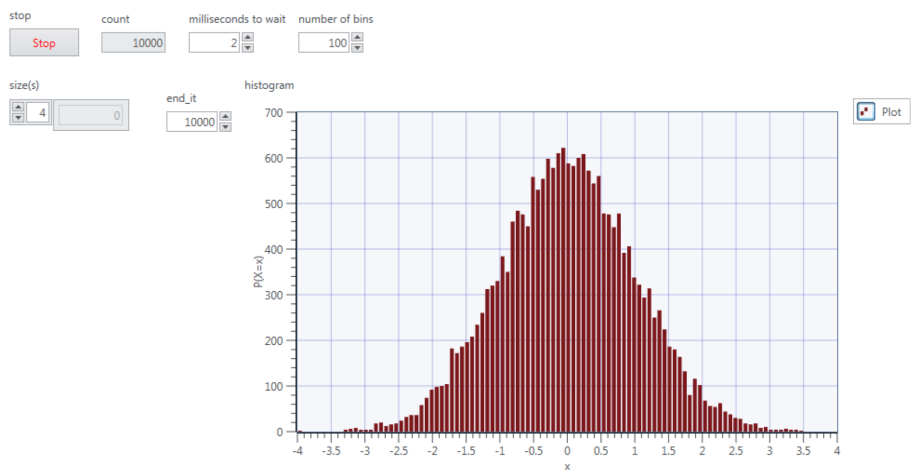
$\mu =$ Mean
$\sigma =$ Standard Deviation

Diagram 2



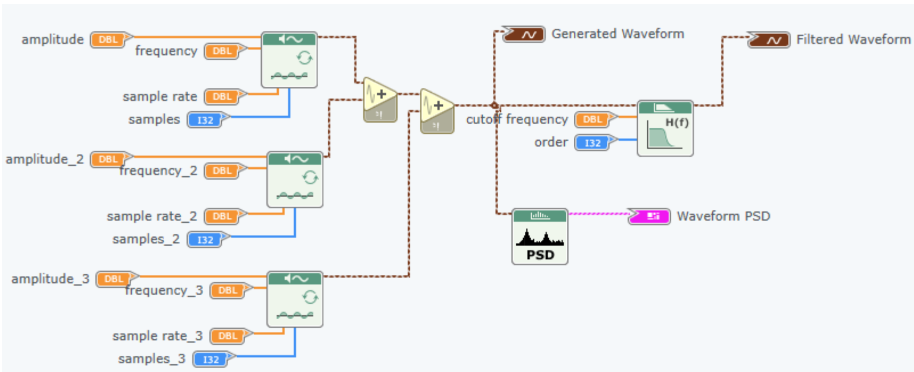
New output was as follows:

Histogram 2

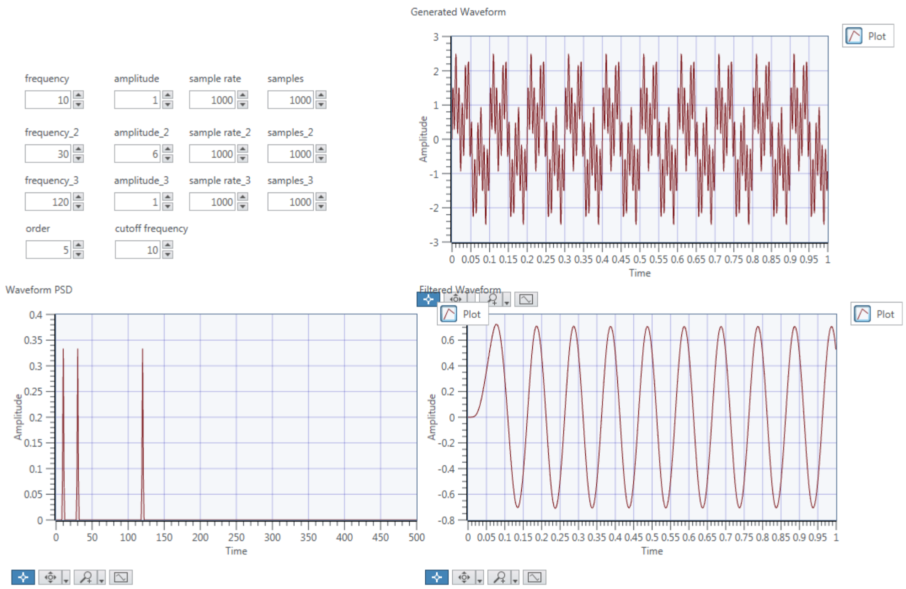


Exercise 3

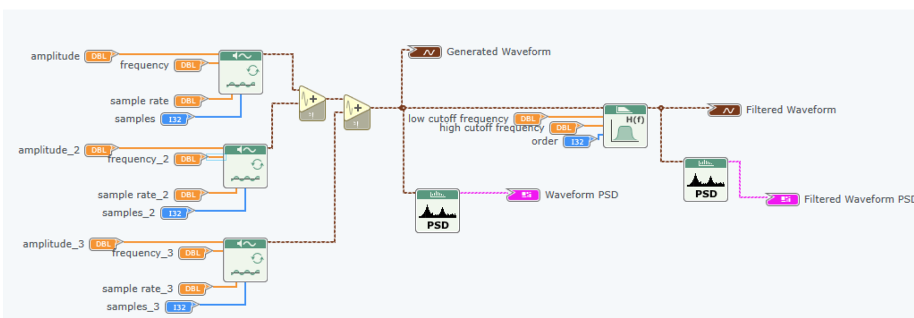
Block Diagram - Just lowpass filter



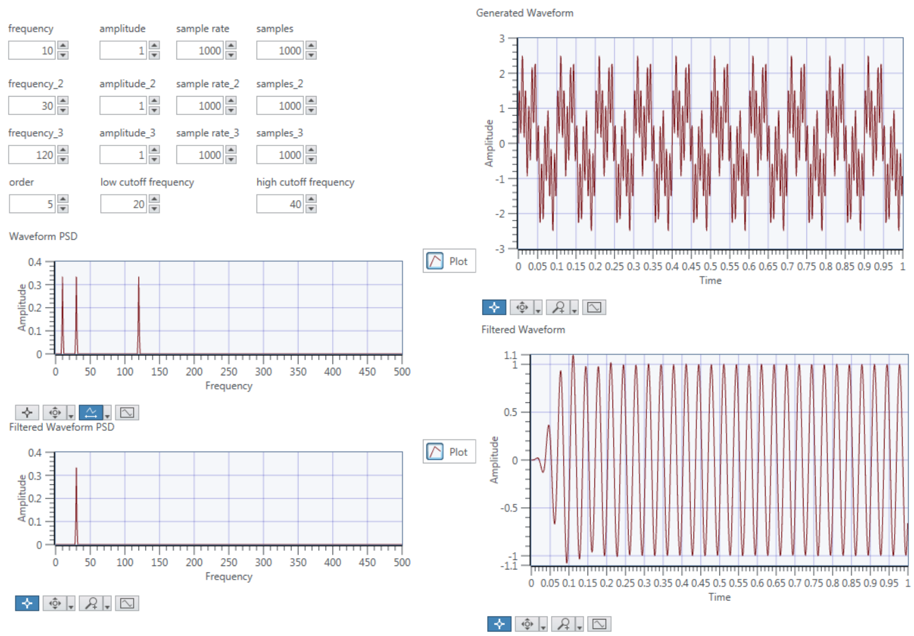
Front Panel - Just lowpass filter output



Block Diagram - Bandpass filter



Front Panel - Bandpass filter output



frequency
amplitude
sample rate
samples
frequency_2
amplitude_2
sample rate_2
samples_2
frequency_3
amplitude_3
sample rate_3
samples_3
order
low cutoff frequency
high cutoff frequency
Waveform PSD
Filtered Waveform PSD
Generated Waveform
Filtered Waveform
Plot
Amplitude
Frequency
Time

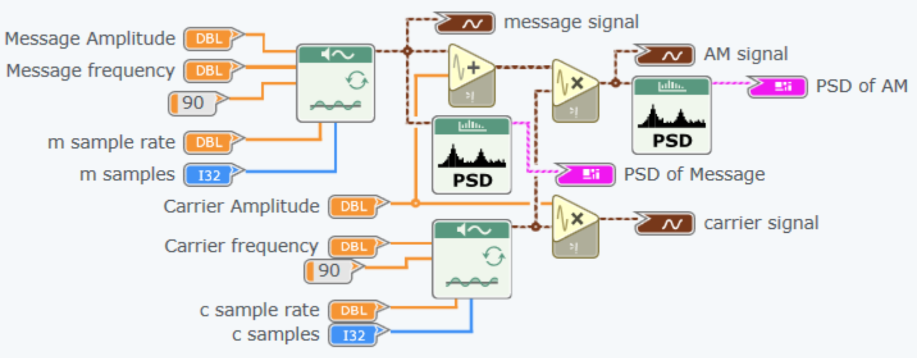
Lab 2 - AM Simulation and USRP

Exercise 1 - AM Modulation

The standard equation for AM signals:

$$s(t) = [A_c + A_m \cos(2\pi f_m t)]\cos(2\pi f_c t)$$

In this first exercise, we built an AM modulater following the standard equation where the carrier wave's amplitude is changed with respect to the message signal. The fully built diagram is as follows:



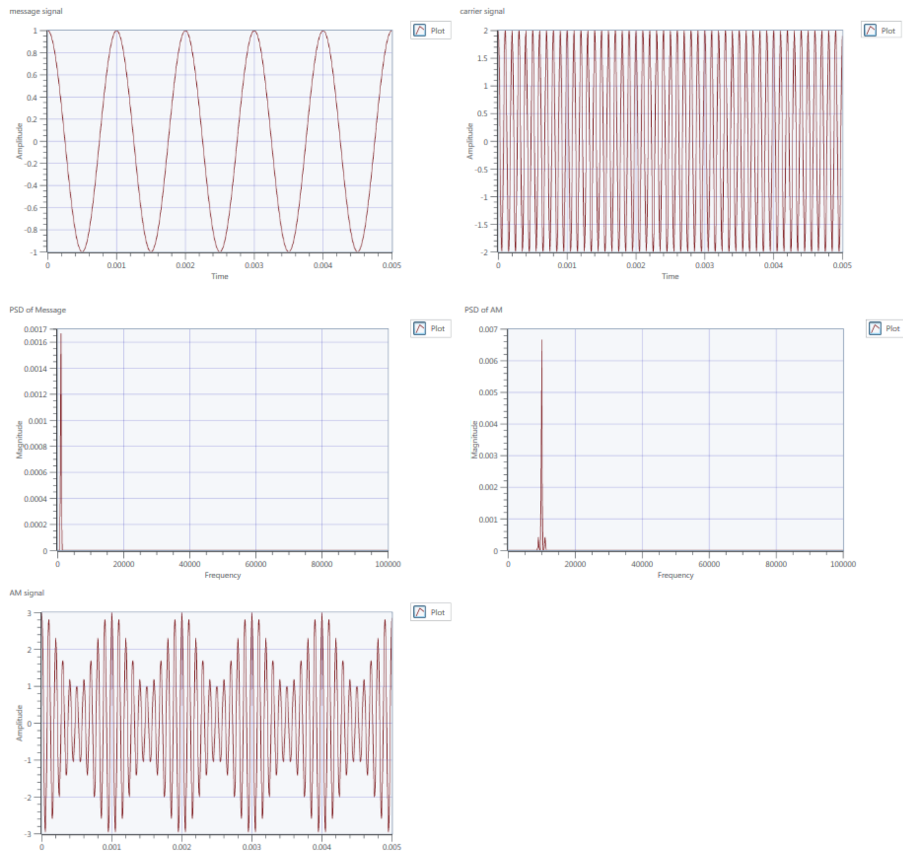
We set the parameters for the message and carrier signal to the following:

	Message Signal	**Carrier Signal**
Frequency	1kHz (Control)	10kHz (Control)
Phase	90° (Constant)	90° (Constant)
Amplitude	Message Amplitude (Control)	Carrier Amplitude (Control)
Sample rate	200k (Control)	200k (Control)
Samples	1000 (Control)	1000 (Control)

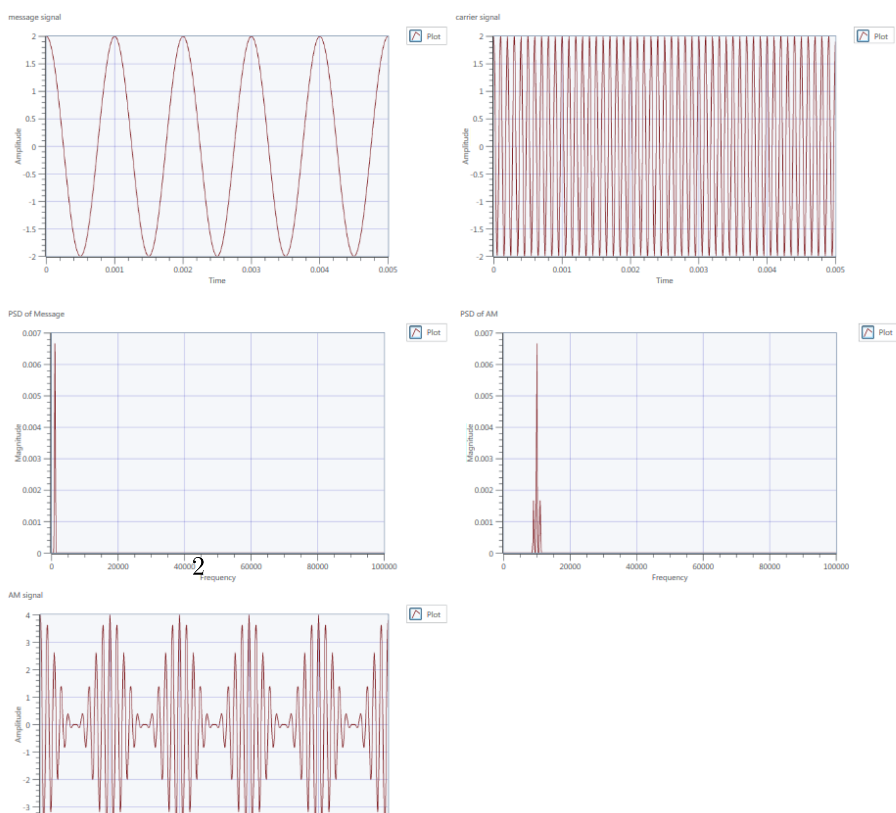
We then altered the modulation index and looked at the changes that appeared:

Modulation index = $\mu = \frac{A_m}{A_c}.$

Effect of Modulation Index

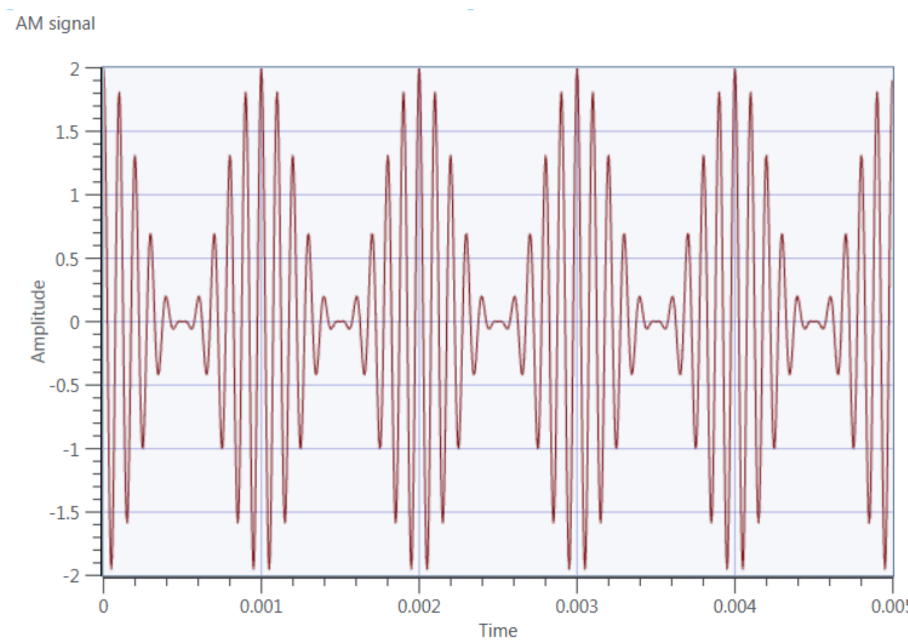


Mod Index = 0.5:

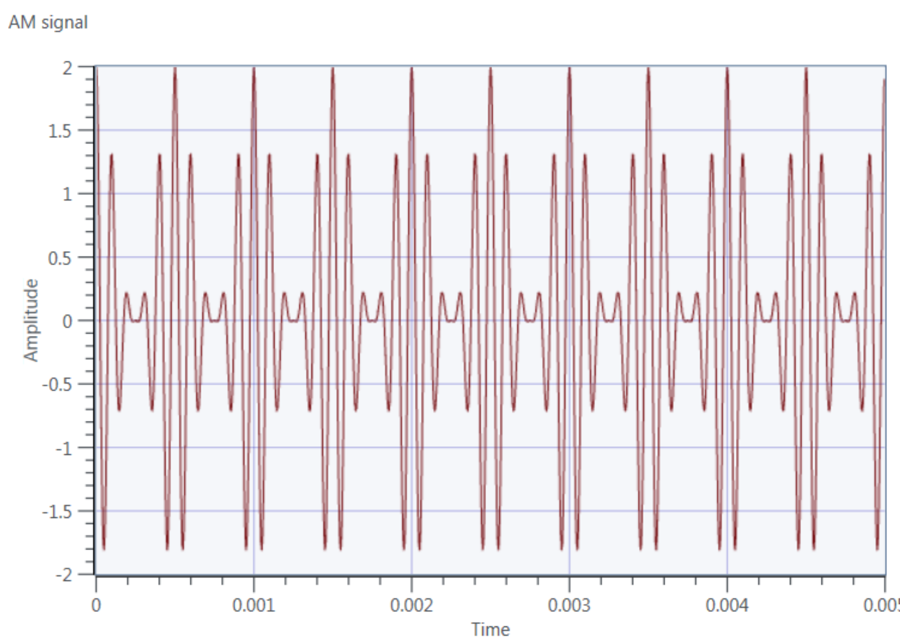


0.5, the signal is undermodulated, has low modulation depth and therefore does not utilise the carrier efficiently. At 1, the modulation depth reaches zero and at index of 1.5 over modulation occurs. The carrier signal goes below the zero point and phase reversal is exhibited. The phase reversal caused the sidebands to stretch out - this may cause intereference and must be filtered.

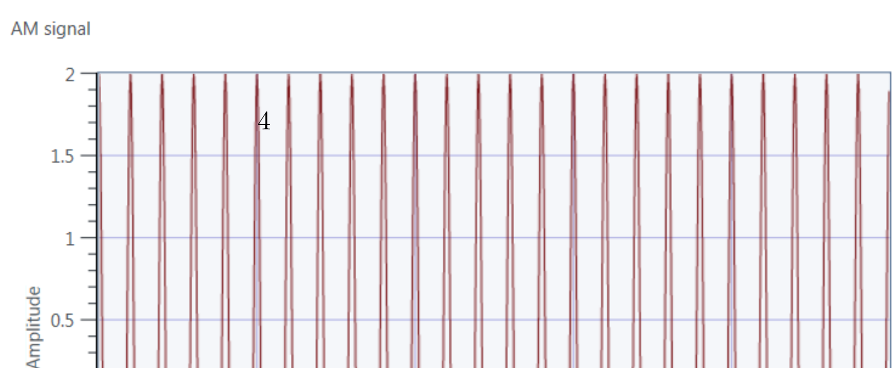
Effect of increasing frequency of the message



Fm = 1kHz:



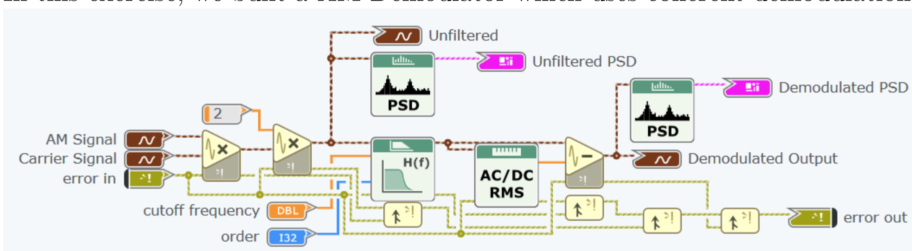
Fm = 2kHz:



- At 5k, the waveform becomes malformed because there is an operlap between -fc+fm and fc-fm.
- “Nyquist Rate” - message bandwidth must be at most half the carrier bandwidth.

Exercise 2a) - Coherent Demodulation

In this exercise, we built a AM Demoulator which uses coherent demodulation.



Theory

The carrier signal m(t)cos(2pifct) when received by the receiver is multiplied by cos(2pifct).

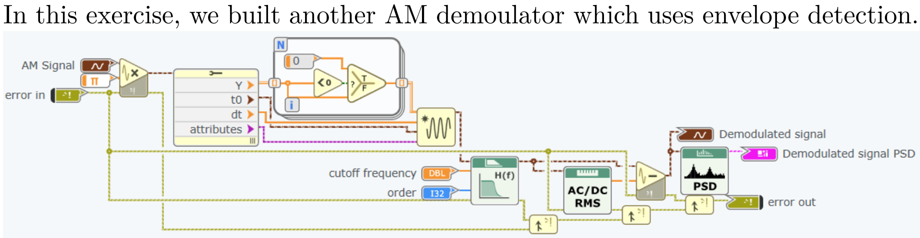
$$m(t)\cos(2\pi f_c t)^2 = m(t)\left({}^{1}\!/_{2}\,(1+\cos(4\pi f_c t)\right)$$

Using a low pass filter, centered around the baseband, the output signal will be 1/2m(t).

Therefore to retrieve the original signal amplitude, we multiply the output by 2.

Exercise 2b) - Envelope Detection

In this exercise, we built another AM demoulator which uses envelope detection.



Theory

The envelope detector outputs the envelope of one half of the signal. The high frequency element is then filtered out.

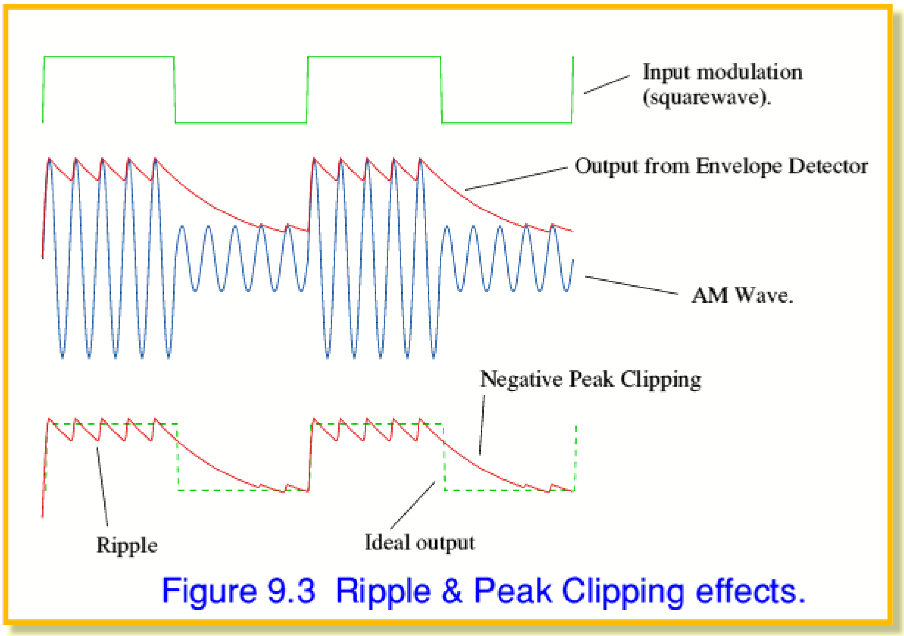
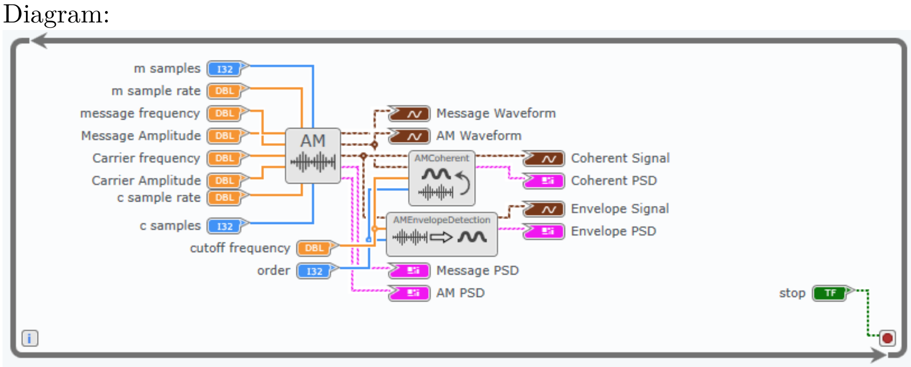


Figure 9.3 Ripple & Peak Clipping effects.

Exercise 3

Here we simulated the AM Modulator and Demodulator working together.

Diagram:



- Envelope detection is better at high frequencies.
- Envelope detection works until the modulation index goes above 1; at that point only coherent detection works correctly because during modulation

the envelope signal has parts that are left negative.
- Coherent detection is fine at >1 modulation indexes, but requires the transmitter and receiever to be in phase and at the same frequency.

Exercise 4

Universal Software Radio Peripheral components:

1. niUSRP Open Tx Session
2. niUSRP Configure Signal
3. niUSRP Write Tx Data
4. niUSRP Close Session
5. niUSRP Open Rx Session
6. niUSRP Initiate
7. niUSRP Fetch Rx data
8. niUSRP Abort

If the baseband discrete time signal is expressed as:

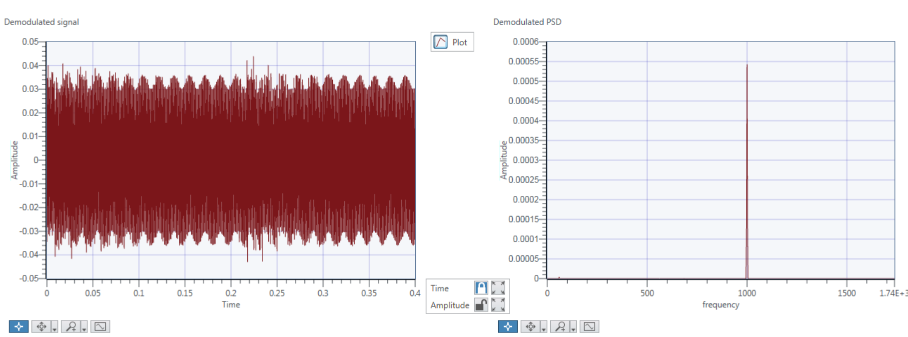
$$\tilde{g}[nT_x] = g_I[nT_x] + jg_Q[nT_x]$$

then the continuous time transmitted signal from the USRP is

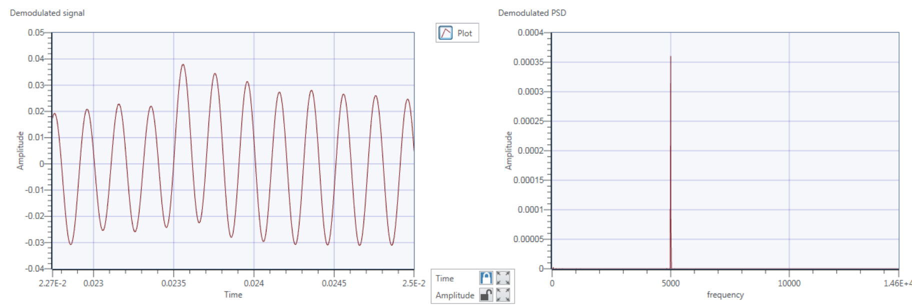
$$g(t) = Ag_I(t)\cos(2\pi f_c t) - Ag_Q(t)\sin(2\pi f_c t)$$

Explain how the transmitter and receiver work.

- The modules provide the USRP with complex data representing the signal.
- The USRP modulates the signal, giving the above continuous time signal if given a baseband time signal as above.
- At the receiver the USRP demodulates the signal and returns the demodulated signal.
- The signal is very noisy.



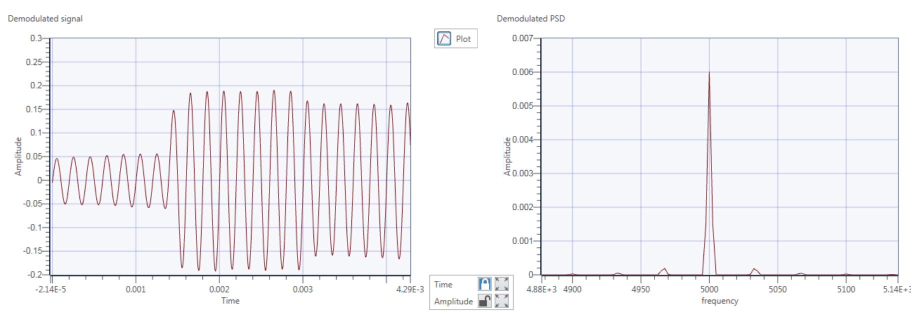
Single tone 5kHz



To observe the effect of noise in the demodulated signal, increase the receiver's gain to 20
dB (your receiver will start to detect other weaker signals in addition to the transmitted signal), and
adjust the X axis of the demodulated message (in time domain) to show values between 0 second

(s) and 0.004 s. Then, change the modulation index (mu) value and observe the effects on the plots
of the demodulated signal in both time and frequency. From what value of can noise be clearly
noticed in the plots?

- We increased the receiver gain to 20, and the modulation index all the way up to 100, and received the following noise:



Lab 3: FM Simulation and USRP

Exercise 1: FM Modulator

Generalised function for FM is:

$$g(t) = A_c \cos\left(2\pi f_c t + \theta_m(t)\right), \text{ where } \theta_m(t) = 2\pi\Delta_f \int m(\tau)\mathrm{d}\tau,$$

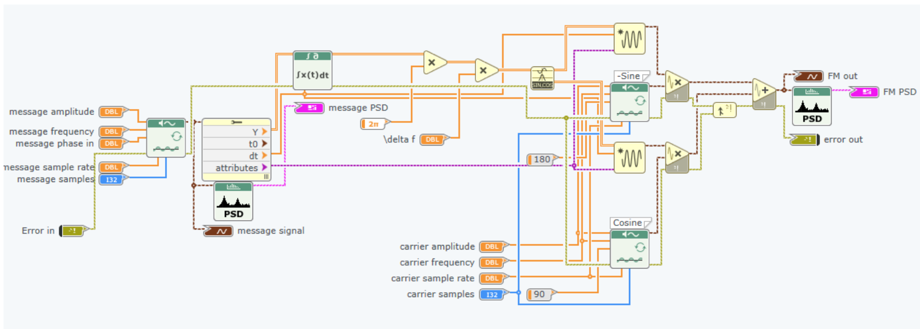
Instantaneous frequency is:

$$\omega = 2\pi f_c t + \theta_m(t)$$

Equivalent form we will be using:

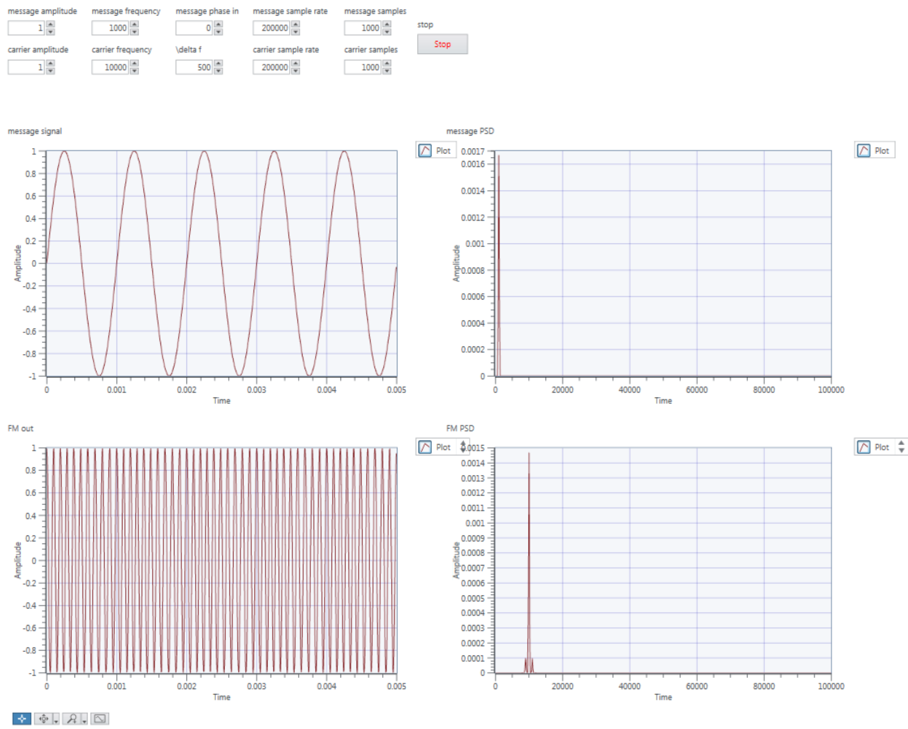
$$s(t) = A_c \cos\left(2\pi f_c t\right)\cos\left(\theta_m(t)\right) - A_c \sin\left(2\pi f_c t\right)\sin\left(\theta_m(t)\right)$$

Diagram

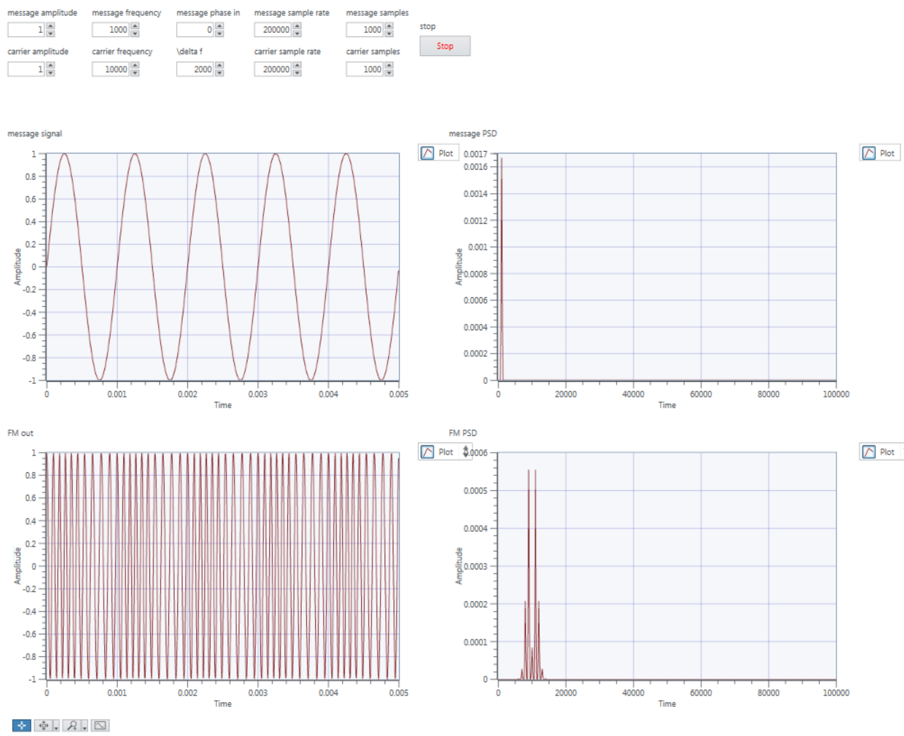


Varying *delta f*

delta f=500

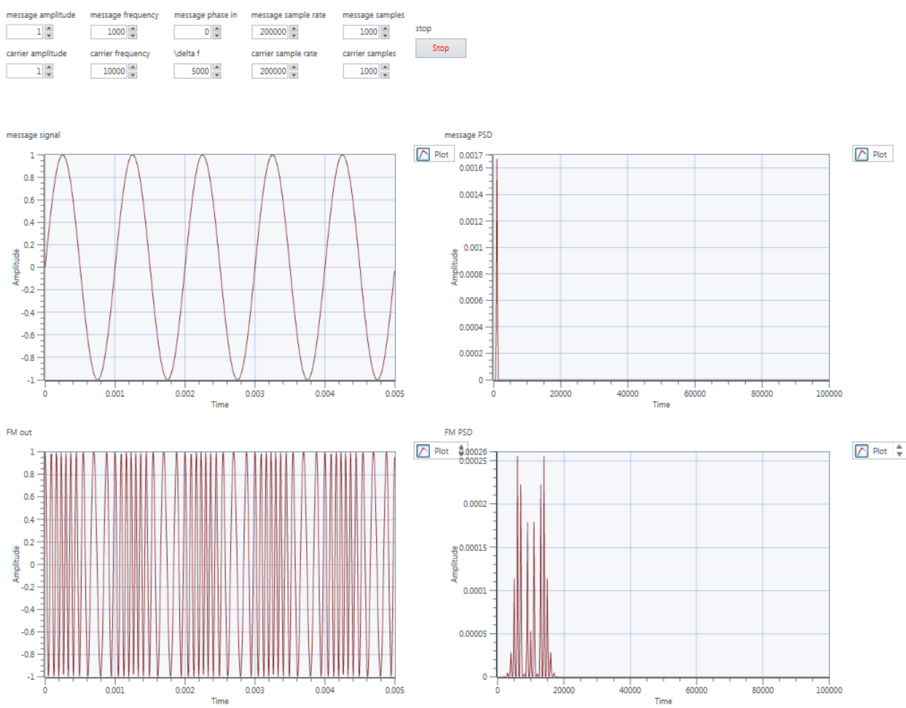


delta f=2000



message amplitude
message frequency
message phase in
message sample rate
message samples
stop
Stop
carrier amplitude
carrier frequency
\delta f
carrier sample rate
carrier samples
message signal
message PSD
FM out
FM PSD
Amplitude
Time
Plot

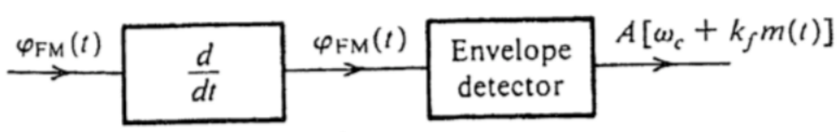
delta f=5000



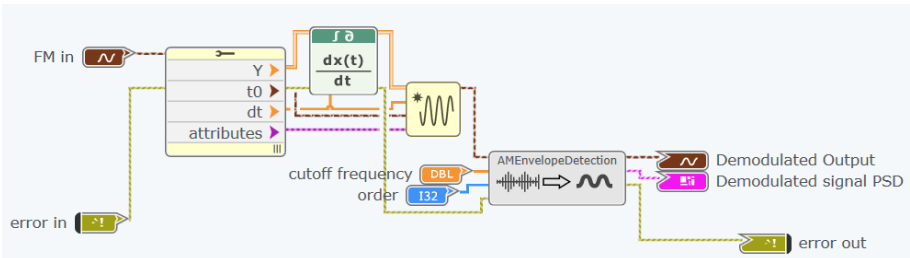
- As delta f increases, you can more easily see the variations in frequency of the FM signal.
- In the frequency domain, you can see the message signal spread across a larger bandwidth.

Exercise 2: FM Demodulator

- Theory is that the derivative provides a sinusoidal signal which has amplitude proportional to the message signal.
- This is just like AM modulation, so the envelope detection method works to retrieve the signal from the differentiated signal.
- Coherent detection would not work because we do not know the phase of the resultant signal

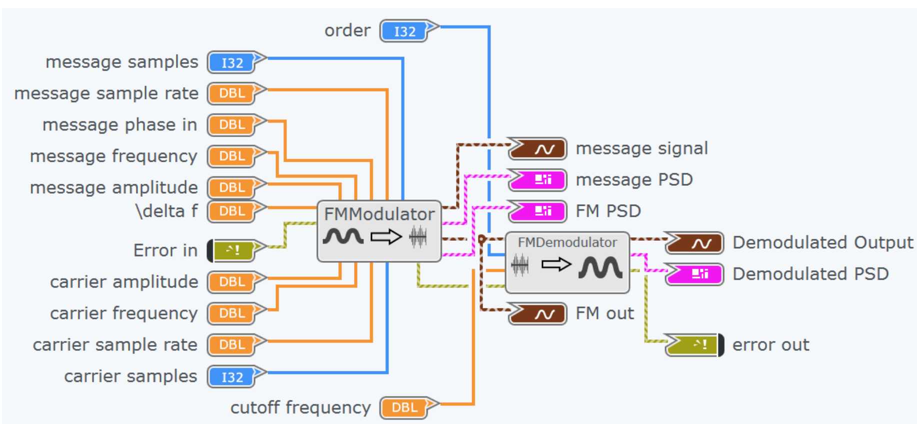


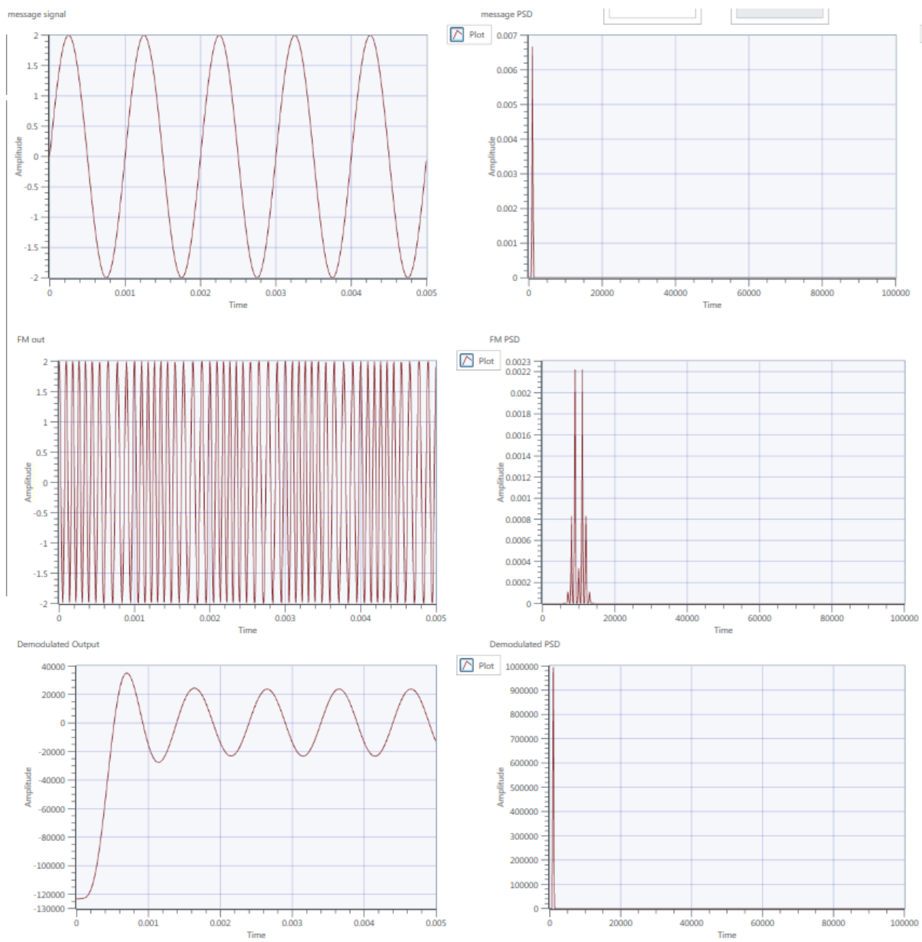
Envelope Detection

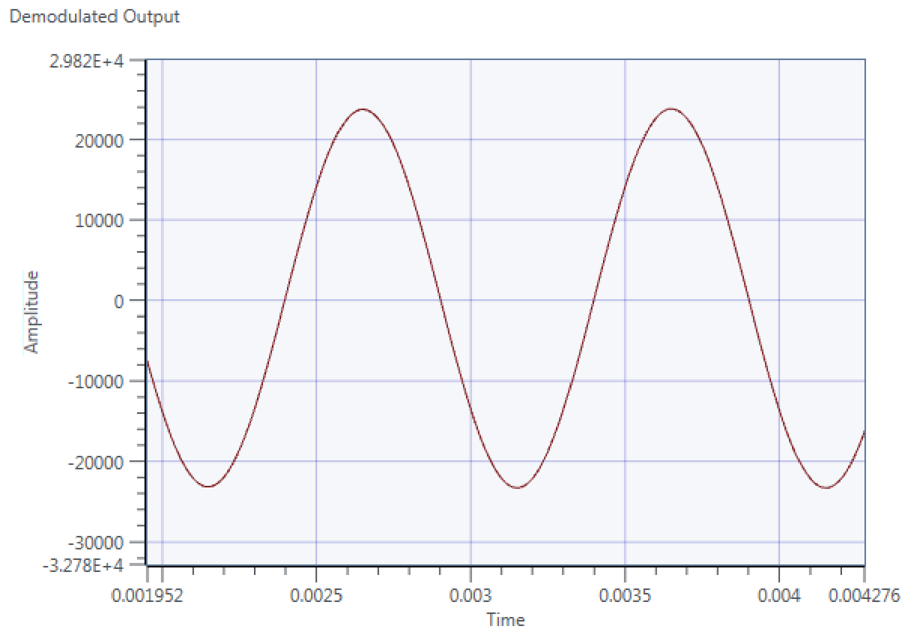


Exercise 3: FM Simulation

Top level Diagram

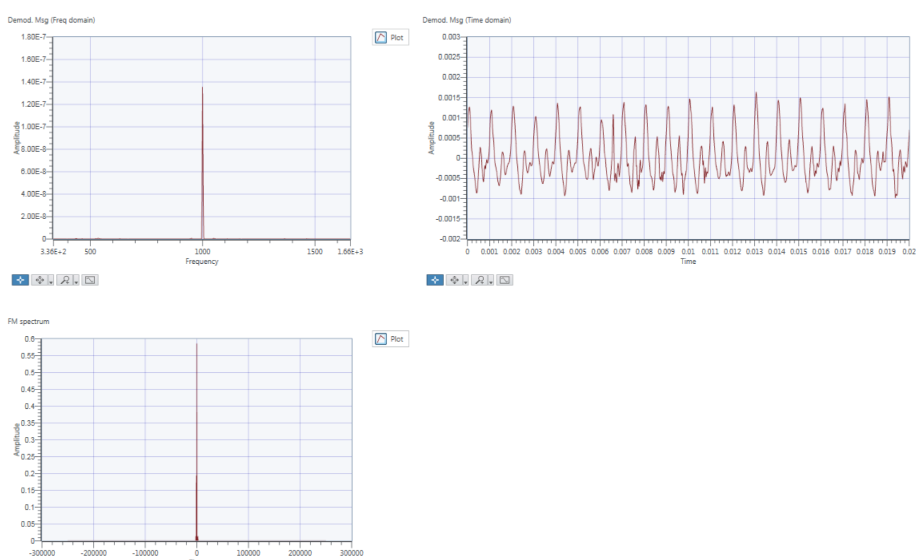



message signal
message PSD
FM out
FM PSD
Demodulated Output
Demodulated PSD
Plot
Amplitude
Time

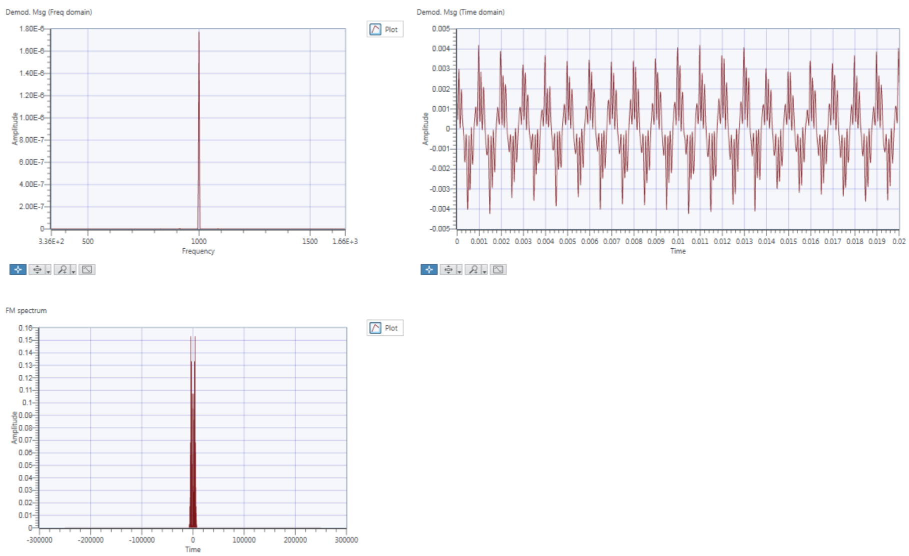


Exercise 4: FM USRP

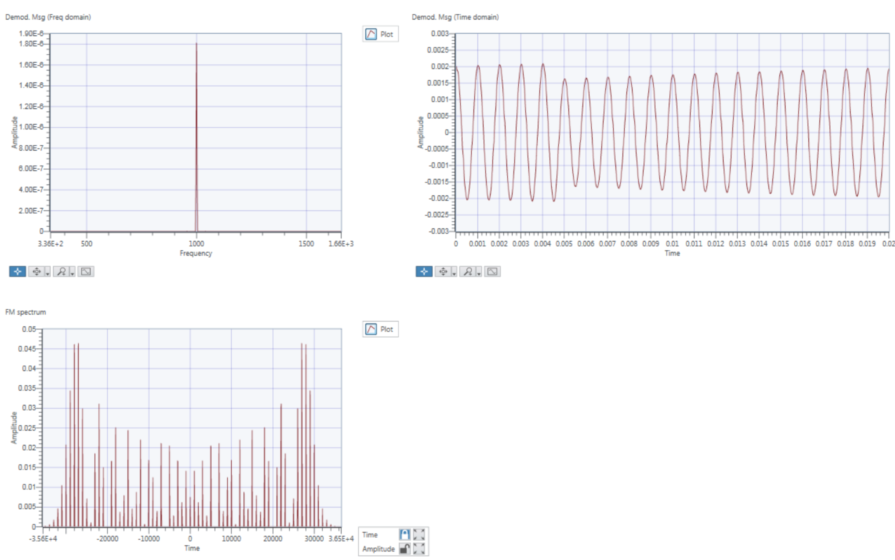
Delta f = 1kHz



Delta f = 5kHz



Delta f = 30kHz



- Note that the bandwidth of this signal is 62kHz, which agrees with Carsons rule for delta f = 30kHz and B = 1kHz:

$$B_{FM} \cong 2(\Delta f + B) = 62\text{kHz}.$$

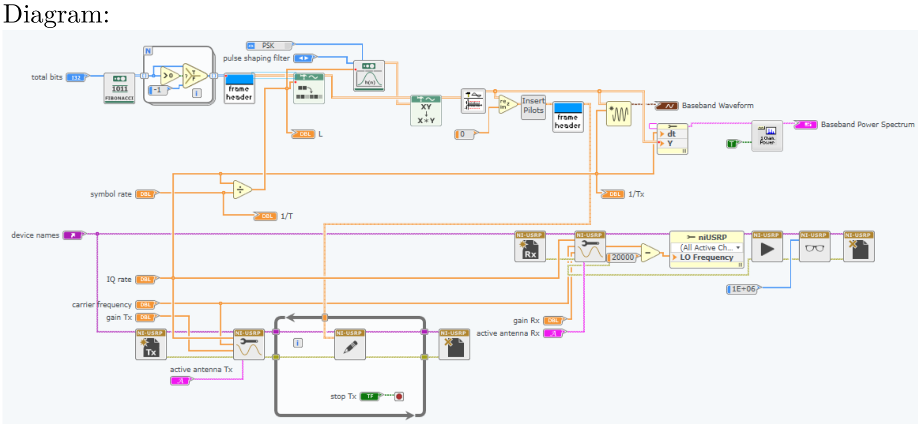
Lab 4 - Binary Phase Shift Keying (BPSK) Via USRP

Exercise 1 - BPSK Transmitter

Aim

In this exercise, we constructed a BPSK Transmitter

Diagram:

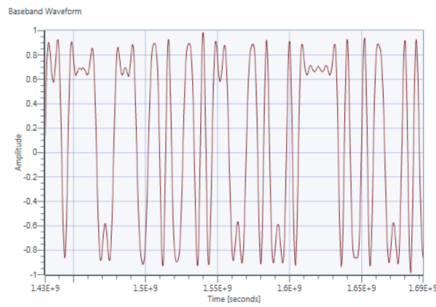


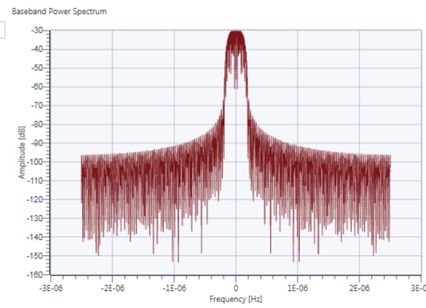
Observation

Values:

Carrier frequency	400 MHz
IQ Rate	200 kHz (Note: This sets the value of $1/T_x$)
Gain	0 dB
Active Antenna	TX1
Symbol rate	10,000 symbols/s
Message Length	1000 bits
Pulse shaping filter	Root Raised

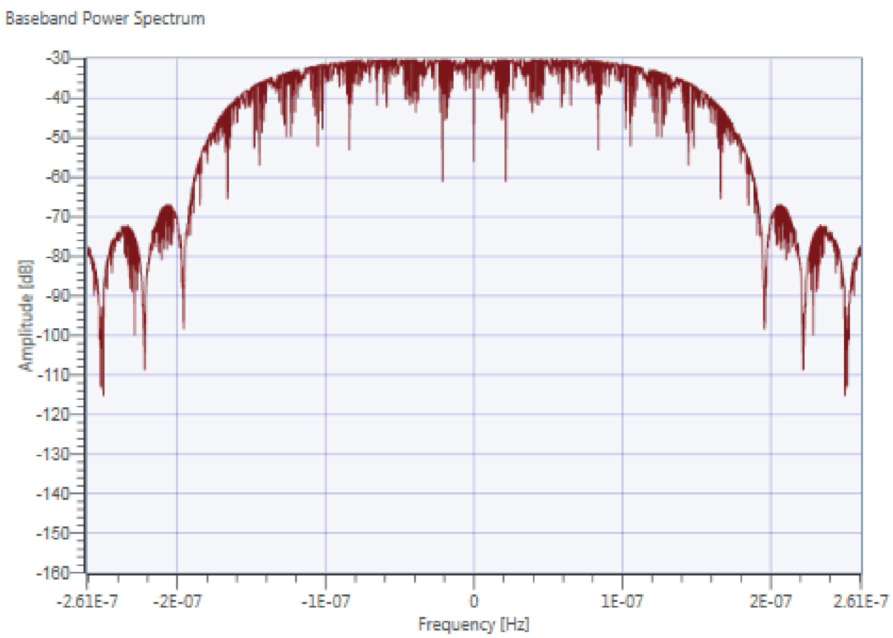
For a IQ Rate of 200k and a symbol rate of 10k the number of samples per symbol is 20.





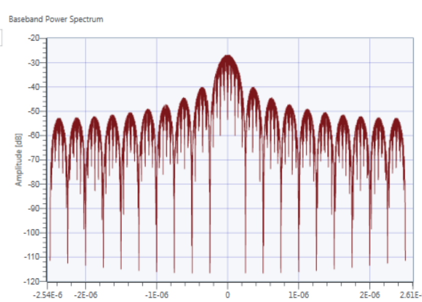
Graphs for values above:

Main lobe:

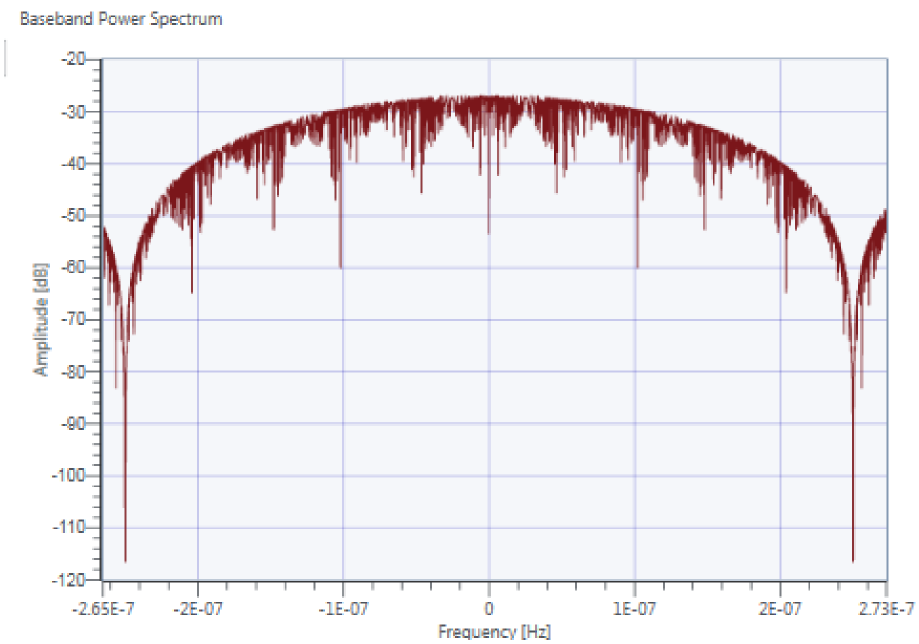


Main lobe bandwidth: 4e-07 Hz

No pulse shaping filter graphs:



No pulse shaping filter main lobe:



Main lobe bandwidth: 5.2e-07 Hz

- We observed that without a pulse shaping filter, the sideband lobes continued for a larger frequency range. This is because without a pulse shaping filter, higher frequency elements are required due to the rapid change in the message signal.
- Spectral rolloff is much faster using the pulse shaping filter.

Exercise 2 - BPSK Receiver

Aim

The steps needed to obtain the transmitted signals are:

1. Channel Estimation: This is required to remove phase ambiguity caused by the channel and the USRP oscillators. Because the Tx and Rx channels are both using free running oscillators, there will be some ambiguous phase offset that is highly dependent on the drift and skew of the oscillators themselves. The channel estimator attempts to correct this problem by reading the received pilots which you inserted in the Tx signal then performing the LSE channel estimation to find the channel transfer function. The channel transfer function is then inverted on the Rx signal to remove phase offset.

2. Matched Filtering: We will use a root-raised-cosine receiver filter. This filter's impulse response $g_{RX}[n]$ is matched to the pulse shape $g_{TX}[n]$ of the transmitted pulses. The matched filter gives optimum performance in the presence of additive white Gaussian noise.

3. Pulse Synchronization: The matched filter output is an analog baseband signal that must be sampled once per symbol time, i.e. once every *T* seconds. Because of filtering, propagation delays, and distortion caused by the communication channel, it is necessary to determine the optimum time to take these samples. A sub-VI called **PulseAlign** has been provided to align the baseband signal.

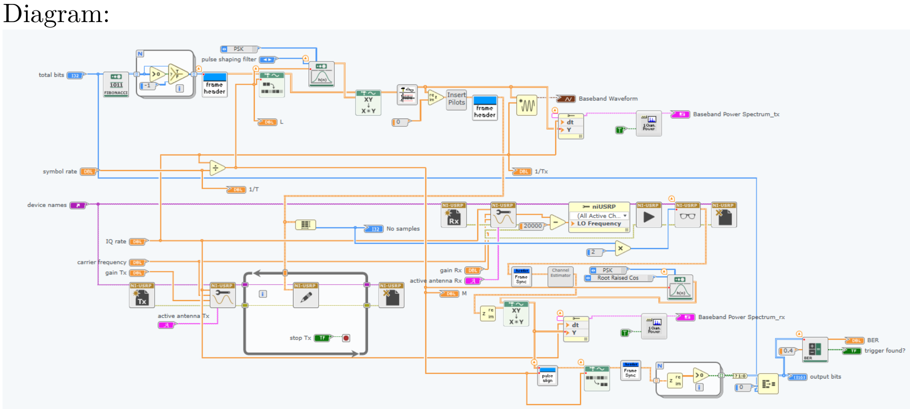
4. Sampling: The **Decimate** function will sample the aligned baseband waveform at index 0 and every *T* seconds thereafter.

5. Detection: Once the baseband waveform has been sampled, each sample must be examined to determine whether it represents a symbol of value 1 or a 0.

6. Symbol Mapping: The detected symbol values must be converted to bits. For binary PSK, this step is easily included in the detection step.

In this exercise, we constructed the BPSK Receiver

Diagram:



Observation

Tx Gain (dB)	Rx Gain (dB)	BER1	BER2	BER3	BER4	BER5	Average BER
0	0	0.499	0.508	0.002	0.002	0.003	0.203

Tx Gain (dB)	Rx Gain (dB)	BER1	BER2	BER3	BER4	BER5	Average BER
-35	-15	0.505	0.470	0.002	0.496	0.536	0.389
-37	-15	0.190	0.506	0.463	1	0.480	0.528
-40	-15	0.487	0.466	1	0.470	0.469	0.587

Exercise 3 - Error Correction Coding

Aim

Observation

- BER Module does not allow the program to compile.
- After doing all the steps, the program still did not work until the following steps were done:

1. Needed to turn the output from the final pulse align module to real from complex.
2. Needed to put a large constant into the number of samples port of the niUSRP Fetch Rx Data port so that it doesn't stop receiving data before the transmitter finishes transmitting.

Finally, when it was working, the BER was much lower:

Tx Gain (dB)	Rx Gain (dB)	BER1	BER2	BER3	BER4	BER5	Average BER
0	0	0	0	0	0	0	0
-35	-15	0.001	0	0.438	0.369	0	0.162
-37	-15	0.452	0.256	0.353	0.490	0.457	0.402
-40	-15	1	0.457	0.480	0.506	0.501	0.589

Exercise 4 - Differential Phase Shift Keying (DPSK)

What is DPSK?

In DPSK, the transmitted sends the difference between two adjacent bits and not the bits themselves. The table below shows how the difference is obtained for possible pairs of sumbols. Encoded sequence is obtained by bn=b(n-1)xan

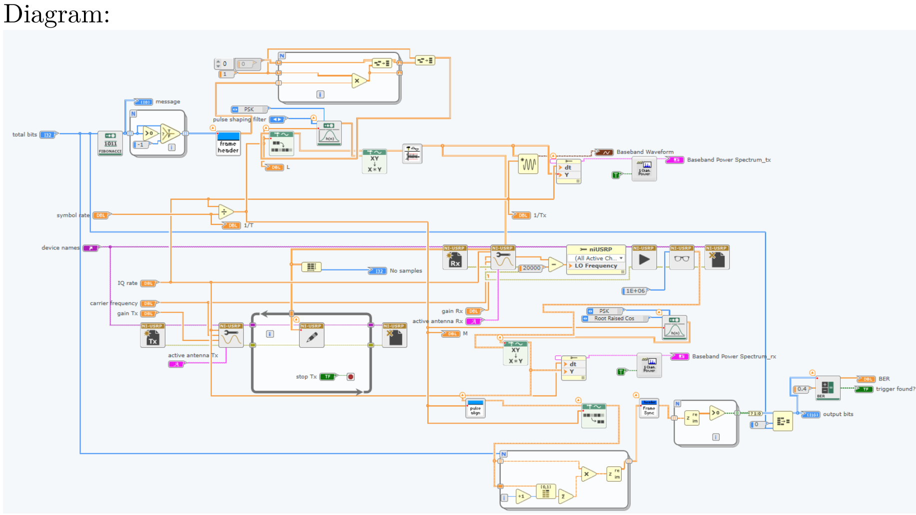
Information Symbols	{bn}	Decision
-	1 (reference bit)	1
1	1	-1
-1	-1	-1
-1	1	1

Information Symbols	{bn}	Decision
1	1	-1
-1	-1	-1
-1	1	1
1	1	1
1	1	-

Aim

In this exercise, we constructed a DPSK Encoder and Decoder and added it to the transmitter and receivers

Diagram:



Observation

Tx Gain (dB)	Rx Gain (dB)	BER1	BER2	BER3	BER4	BER5	Average BER
0	0	0	0	0	0.507	0.449	0.191
-35	-15	0.491	0	0	0.489	0	0.196
-37	-15	0.060	0.481	0.022	0.456	0.001	0.204
-40	-15	1	0.171	0.501	1	0.495	0.633

Comparison between DPSK and BPSK

With the data we collected, DPSK performed better until the Tx Gain was reduced to -40. In theory, DPSK is just sending the differences, so has protection from any phase flipping effect that may occur between the receiver and transmitter.

However, since DPSK only transmits differences, it is possible that errors get propogated forward which is why it performed worse at very low gain.