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Communication Lab Report

Experiment No – 04

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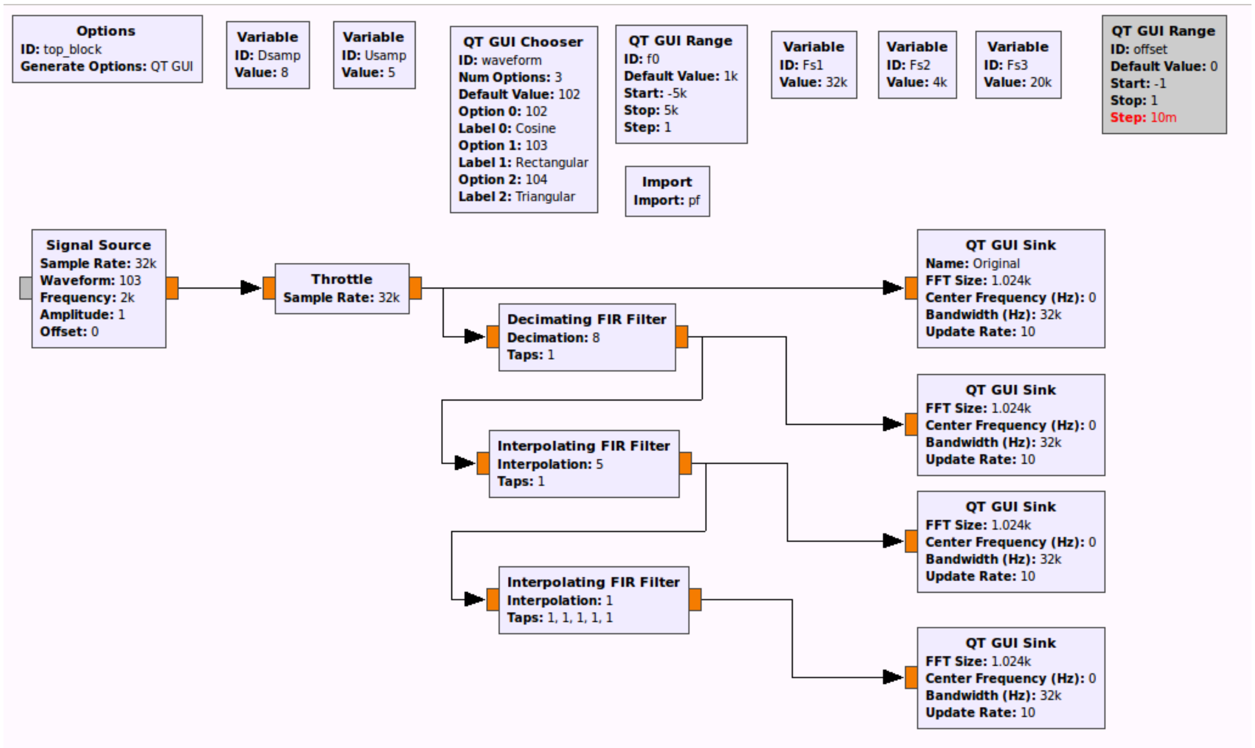
1. Sampling Theorem and Baseband Transmission

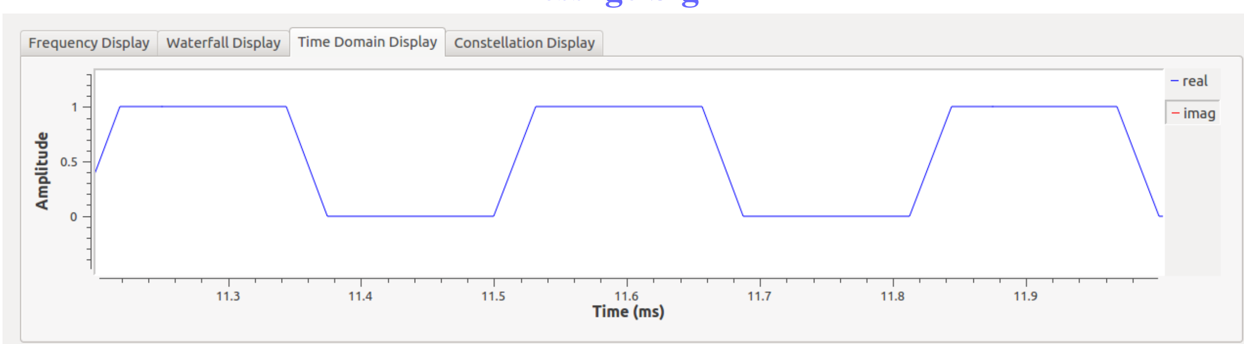
"Nyquist-Shannon Sampling Theorem" is the fundamental base over which all the digital processing techniques are built. The sampling operation samples the incoming signal at regular interval called "Sampling Rate". Sampling Rate is determined by Sampling Frequency.

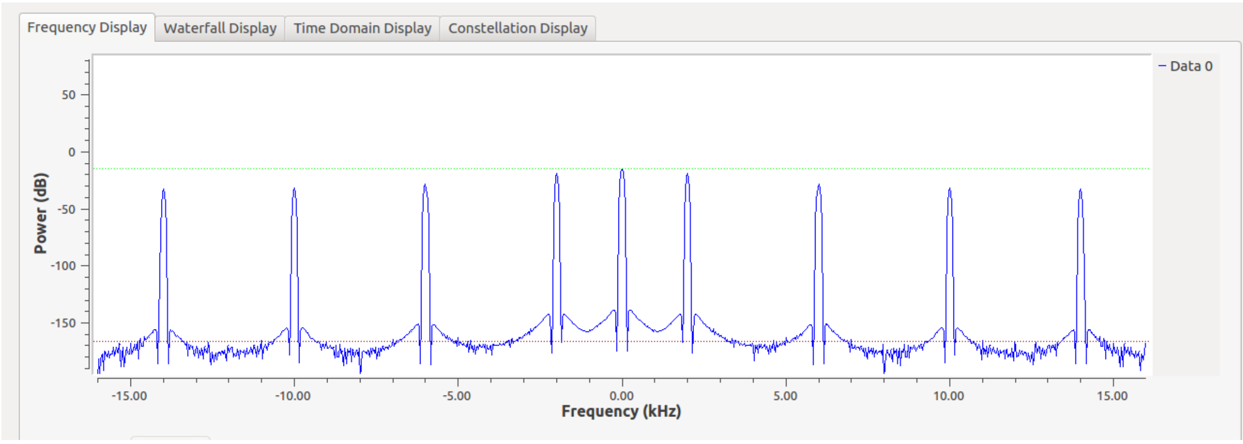
- There is a signal source of frequency 2000 Hz and amplitude 1 & it is sampled at 32kHz.
- The maximum frequency present in this signal is Fm=2k Hz .
- Now, to satisfy the sampling theorem to have a faithful representation of the signal in digital domain, the sampling frequency can be chosen as Fs >=4k Hz.
- That is, we are free to choose any number above 4k Hz.
- Higher the sampling frequency higher is the accuracy of representation of the signal.
- In time domain, the process of sampling can be viewed as multiplying the signal with a series of pulses ("pulse train) at regular interval.

Flowgraph



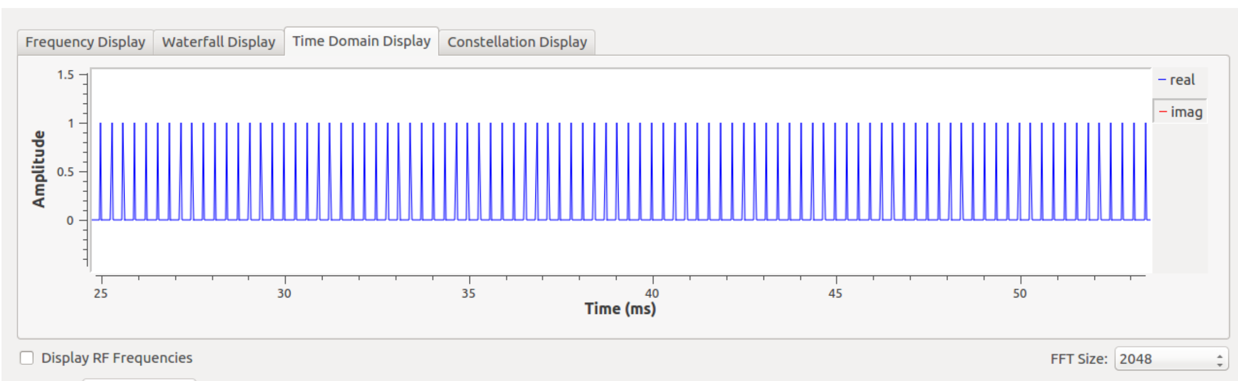
Message Signal

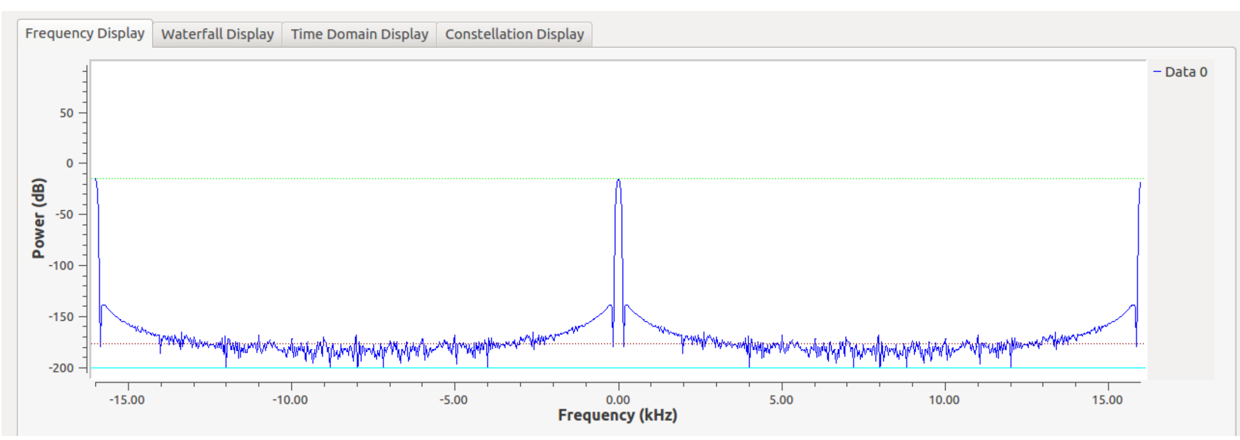




Sampling

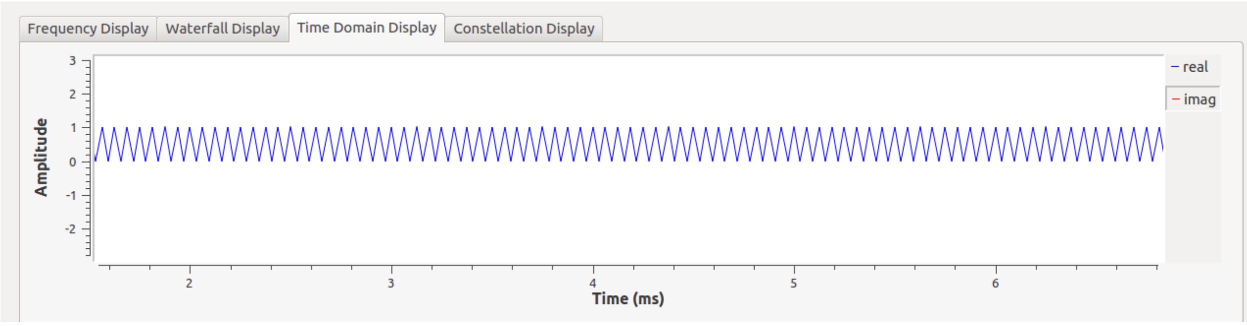
Decimation is the process of reducing the sampling rate of a signal. Decimating FIR filter block is used to samples the signals. Value of Decimation is 8 times.

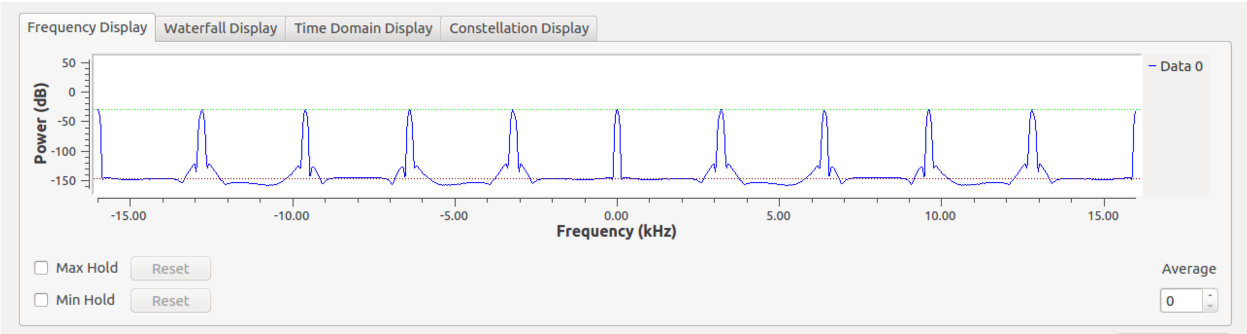




Interpolation

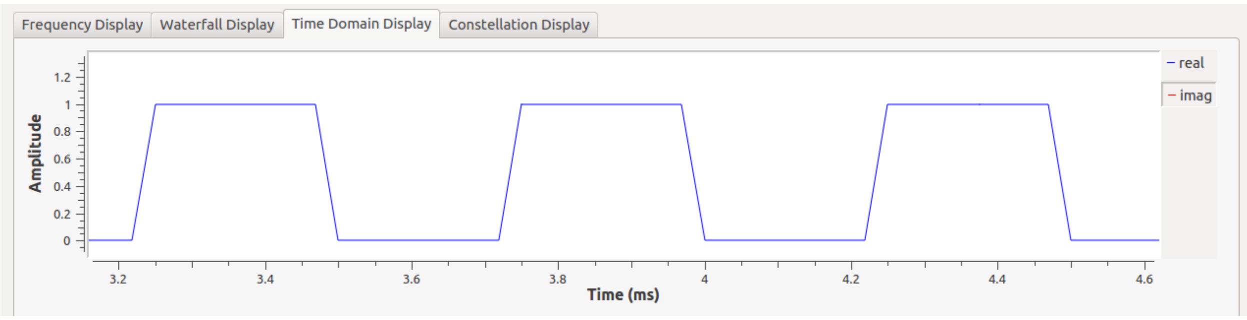
The Interpolating FIR fileter create finite impulse response (FIR) filters that perform the convolution in the time domain. The value of Interpolation is 5

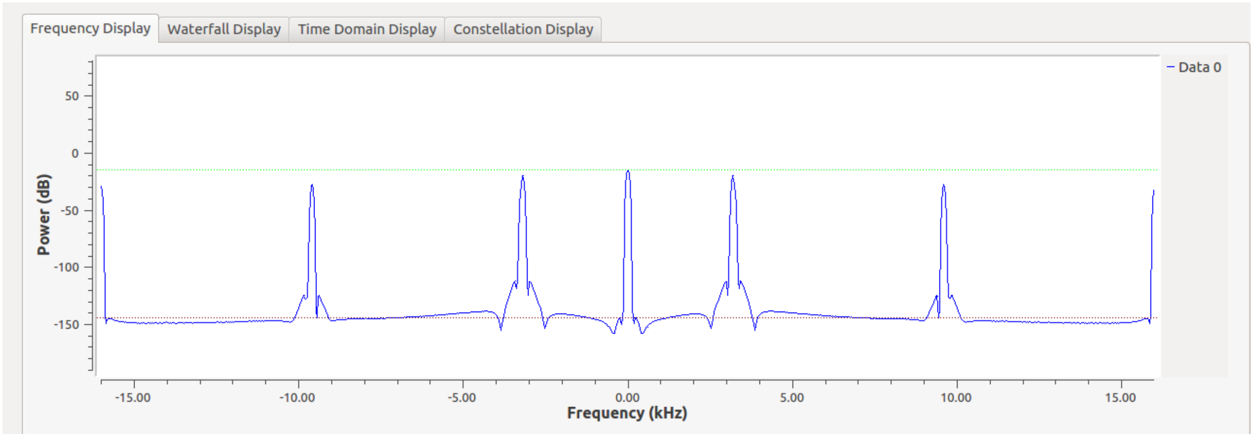




Reconstructed Signal

The reconstructed signal spectrum is little bit different from the original message signal.

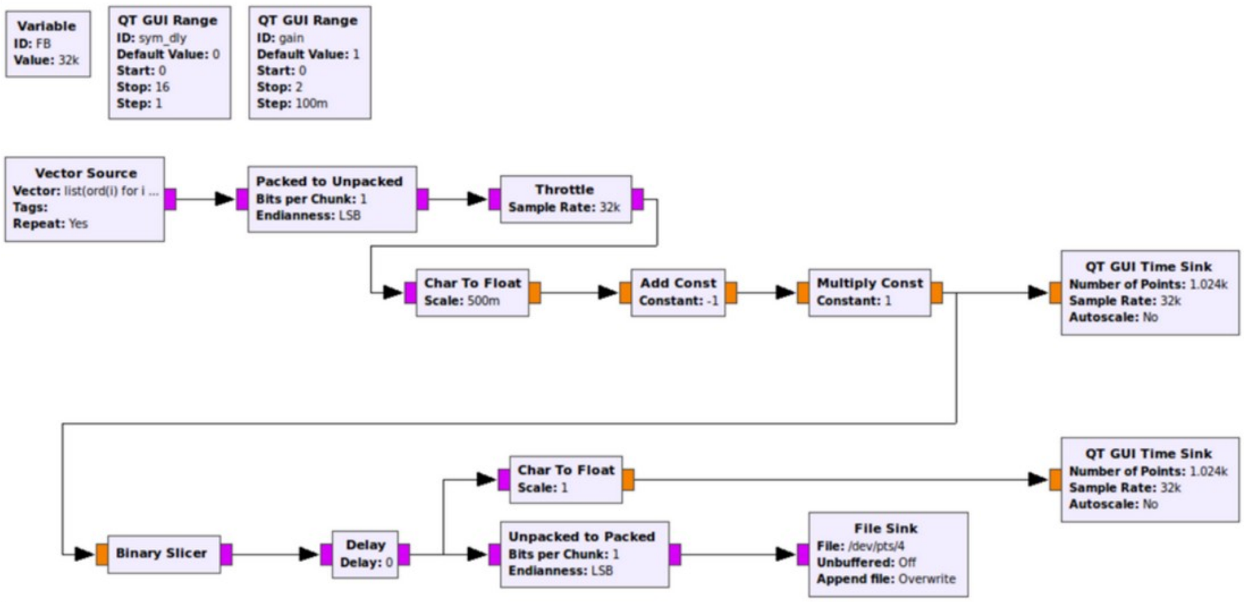




Results:
Signal is successfully sampled at 4K and reconstructed. But we observed distortion when cutoff frequency is lower than bandwidth of signal .

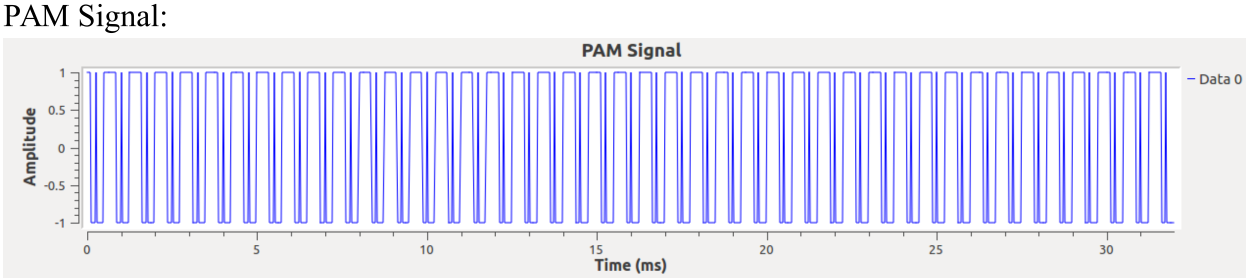
2. Generation of PAM

In this experiment, we need to modulate and demodulate PAM generated wave through a distortionless channel. To study this, we use a vector source for generating input signal. At the receiver, we demodulate it and store the message received in a text file. We use the following flowgraph for generating PAM signal.

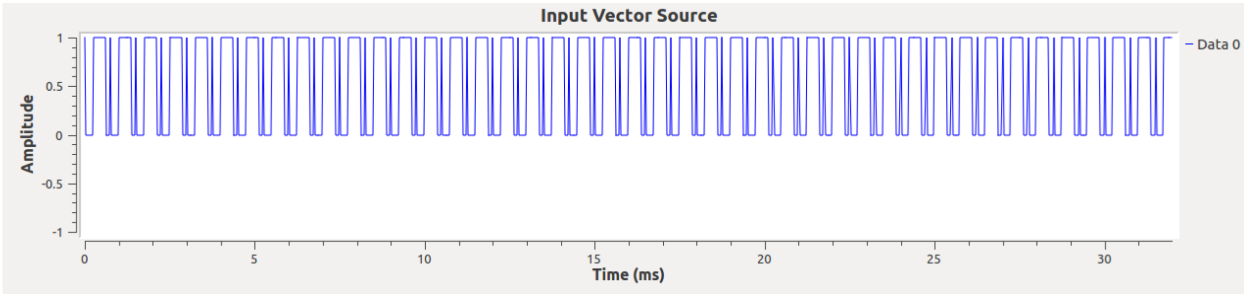


1. ***Vector source*** is used to generate an array of numbers. We can also send a string using the vector source but first we need to create an array of ASCII values of each character that we are sending.
2. ***Packed to Unpacked*** is used to unpack the binary of each ASCII value into one bit before sending it. And before demodulation, we use ***Unpacked to Packed*** to combine/pack the binary bits of every ASCII value we have sent**.**
3. ***Char to Float*****:** Converts a stream from char (byte) data type to float (real) data type.
4. ***Binary Slicer*****:** Slice float binary symbol producing 1 bit output as follows:
    a. *Input[0][n]* $< 0 \rightarrow 0$
    b. *Input[0][n]* $\geq 0 \rightarrow 1$
5. After binary slicing, the output is packed using unpacked to packed and then we store it in a file.

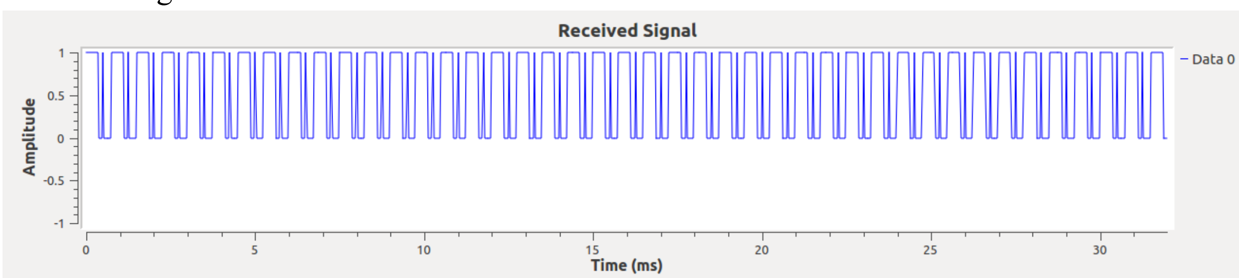
PAM Signal:



Vector Source:

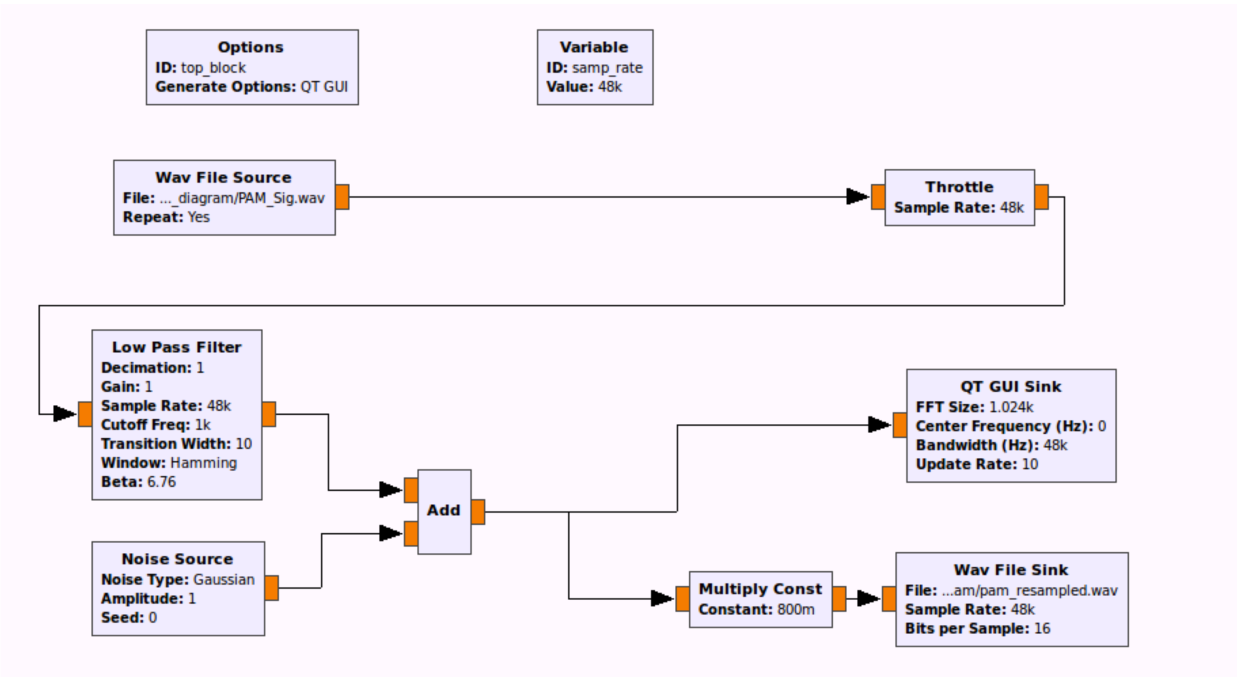


Received Signal:



3. Transmission of PAM through bandlimited channel in presence of noise. Demonstrating the Eye pattern.

The objective of this experiment was to study the transmission characteristics of a PAM signal through a band-limited channel using eye-diagram. For this purpose a PA modulated audio signal was used as source and transmitted through the band-limiting channel modelled in the flow diagram below.

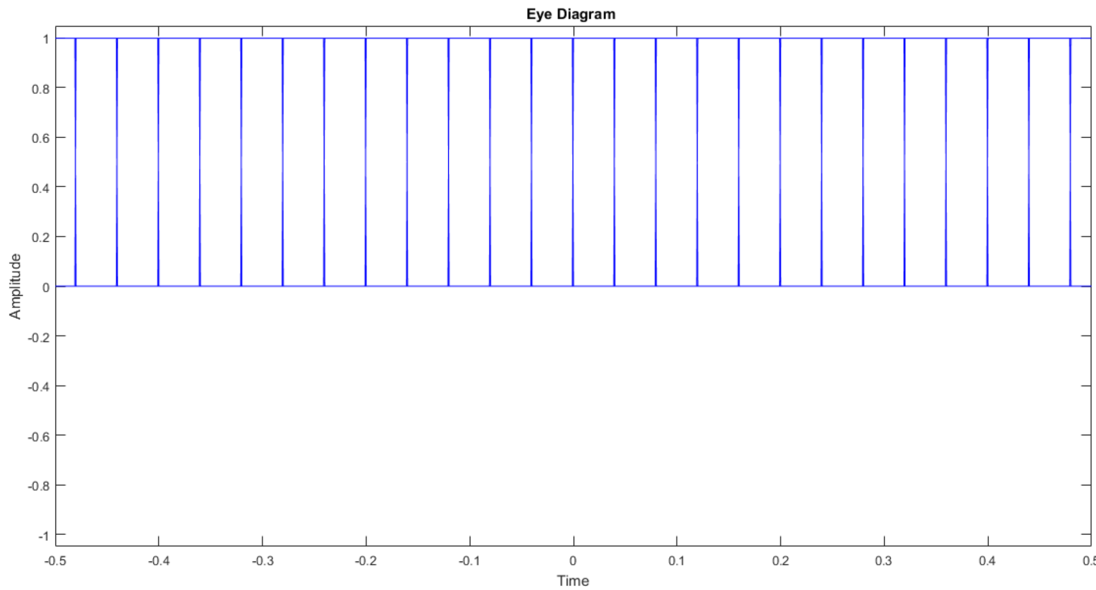


1. **WAV File Source** block is used as the input block for the PAM audio file. **Throttle** is used to regulate computational power.
2. The channel is modelled as a band-limiting (implemented using the **Low Pass Filter** with cutoff frequency 1kHz) and noise in the channel modelled as Additive White Gaussian Noise (AWGN) which is in turn modelled using the **Noise Source** and **Add** blocks.
3. The output signal is stored to an audio (.wav) file in a directory using **Wav File Sink**. Its time response and frequency spectrum is observed using **QT GUI Sink**.
4. Using the following MATLAB script the eye-diagram of the original audio file and output audio file is obtained.

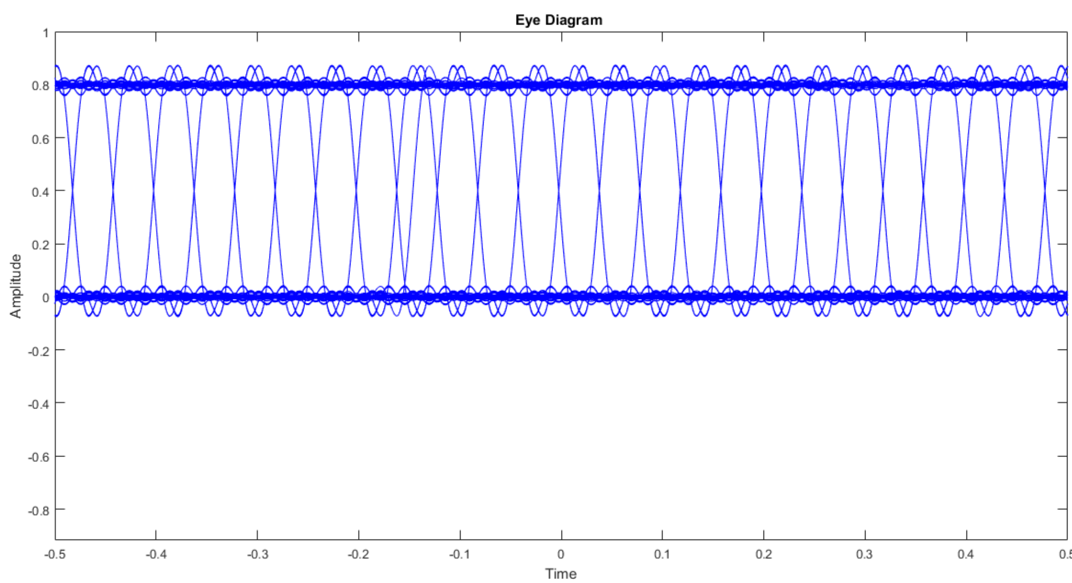
```
[y1,Fs] = audioread('pam_resampled.wav');
[y2,Fs] = audioread('PAM_Sig.wav');
info = audioinfo('pam_resampled.wav')
info = audioinfo('PAM_Sig.wav')
eyediagram(y1,1000,1)
```

```
eyediagram(y2,1000,1)
```

5. The ***audioread*** function loads the audio file into MATLAB as an array. ***audioinfo*** returns the sampling rate of the signal. The eye-diagram is finally plotted using the ***eyediagram*** function. It is observed as follows.



Eye-diagram of original audio file at 1000 samples/frame.



Eye-diagram of received audio file at 1000 samples/frame.

Result:

The PAM signal when transmitted through the channel is band-limited along with additive noise. The eye diagram gets more compact as SNR is decreased or alternatively cut-off frequency is reduced.