A Basic Example of Python

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interactive(children=(FloatSlider(value=0.0, description='awgn', max=1.0, step=0.01), Fl
oatSlider(value=0.0, d…
```

Background

Digital Signals

Digital signals use symbols to represent one or more bits of information.

Generaly, we want to transmit the most data with the least spectrum.

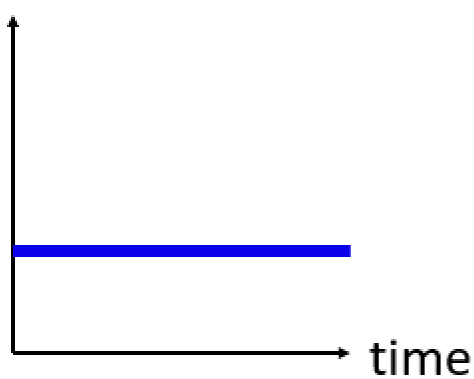
The frequency domain tells us how much spectrum/bandwidth our signal will use.

We usually want to maxamize throughput and minimize spectrum use.

Why can't we just transmit binary data at a faster speed?

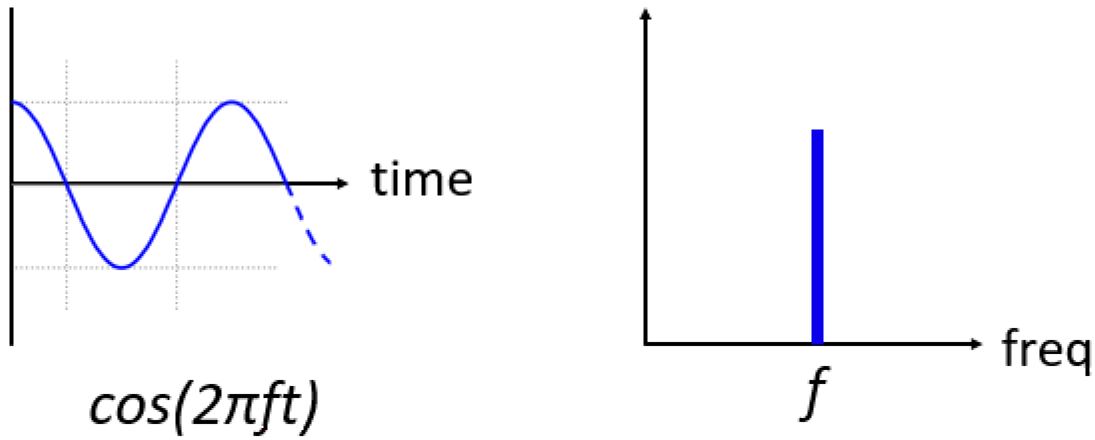
Time-Frequency Pairs

Constant Signal



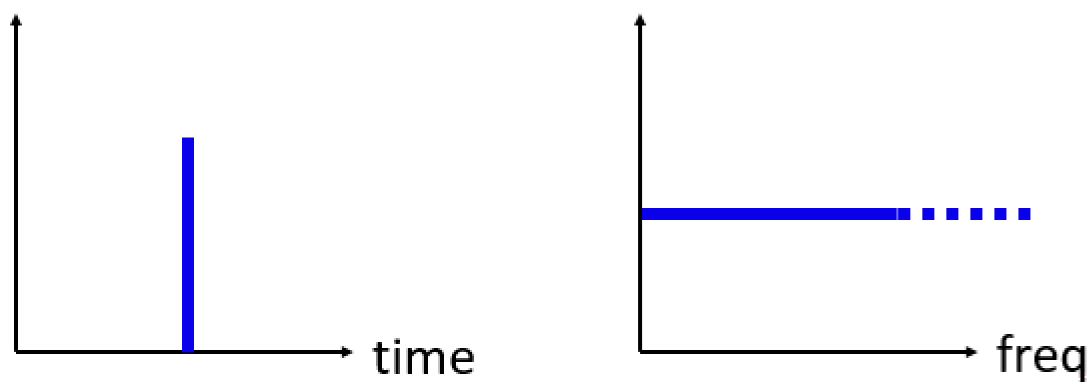
Because there is no frequency, in the frequency domain we have a spike at 0 Hz ("DC").

Sinusoidal Wave



Key Point: The signal has a single frequency, which is why we see a single peak(impulse) in the frequency domain.

Unit Impulse



An impulse in the time domain is a horizontal line (all frequencies) in the frequency domain.

Note that the signal theoretically contains every frequency, although it would have to be infinently short in the time domain for this to occur.

Key Point: Quick changes in the time domain result in many frequencies occurring.

Fourier Properties

Time Scaling Property

Scaling in the time domain causes inverse scaling in the frequency domain.

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$\mathcal{F}\left(x(at)\right) = \frac{1}{|a|}X\left(j\frac{\omega}{a}\right)$

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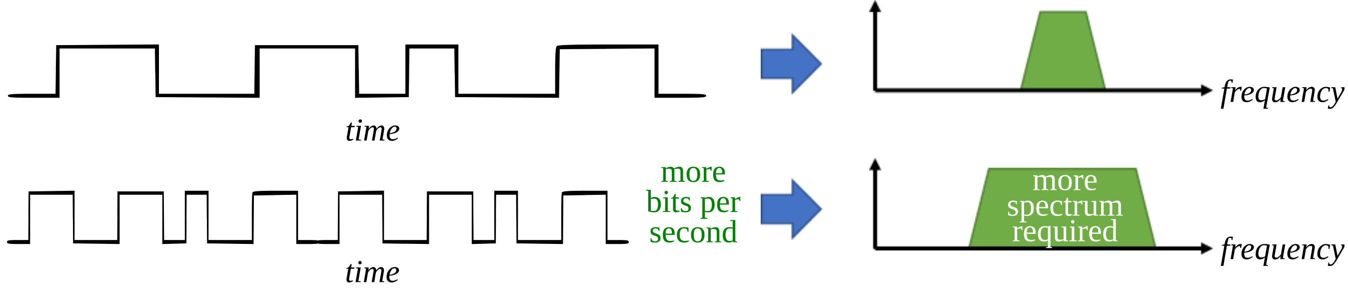

Transmitting 1’s and 0’s faster will increase the bandwidth of our signal and create undesireable frequency components.

Modulation

Our goal: to maximize “spectral efficiency” in units of bits/sec/Hz.

We do this with modulation.

Modify the properties of a sinusoid(the carrier) to transmit information.

- Amplitude
- Phase
- Frequency

Using schemes such as:

- ASK
- PSK
- FSK
- QAM

The spectral characteristics (frequency domain) of the baseband symbols do not change when we modulate a carrier.

Carrier modulation just shifts the baseband up in frequency while the shape stays the same.

The amount of bandwidth we use stays the same.

Inter-Symbol-Interference (ISI)

Definition

ISI occurs when a symbol interferes with subsequent symbols.

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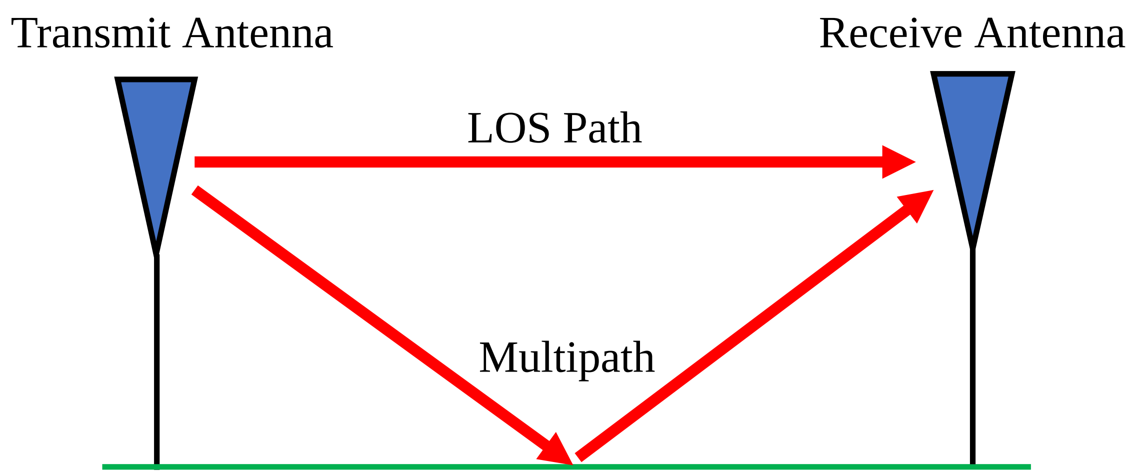
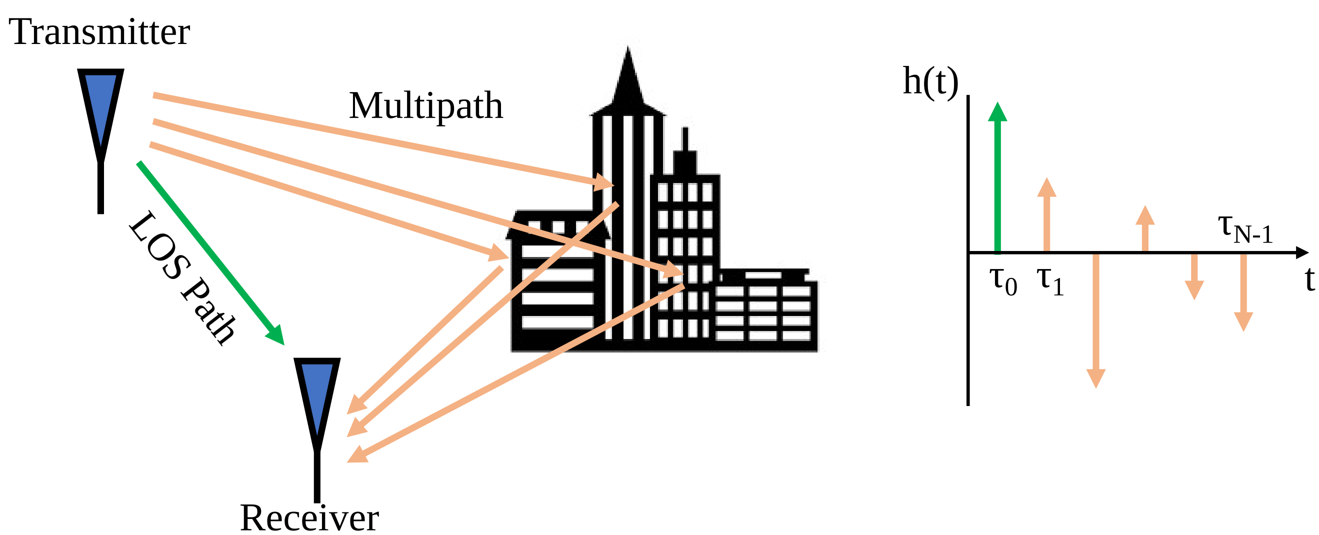
This causes distortion similar to noise and potentially introduces error in the receiver decision device.

Successive symbols appear to "blur" or "bleed" together.

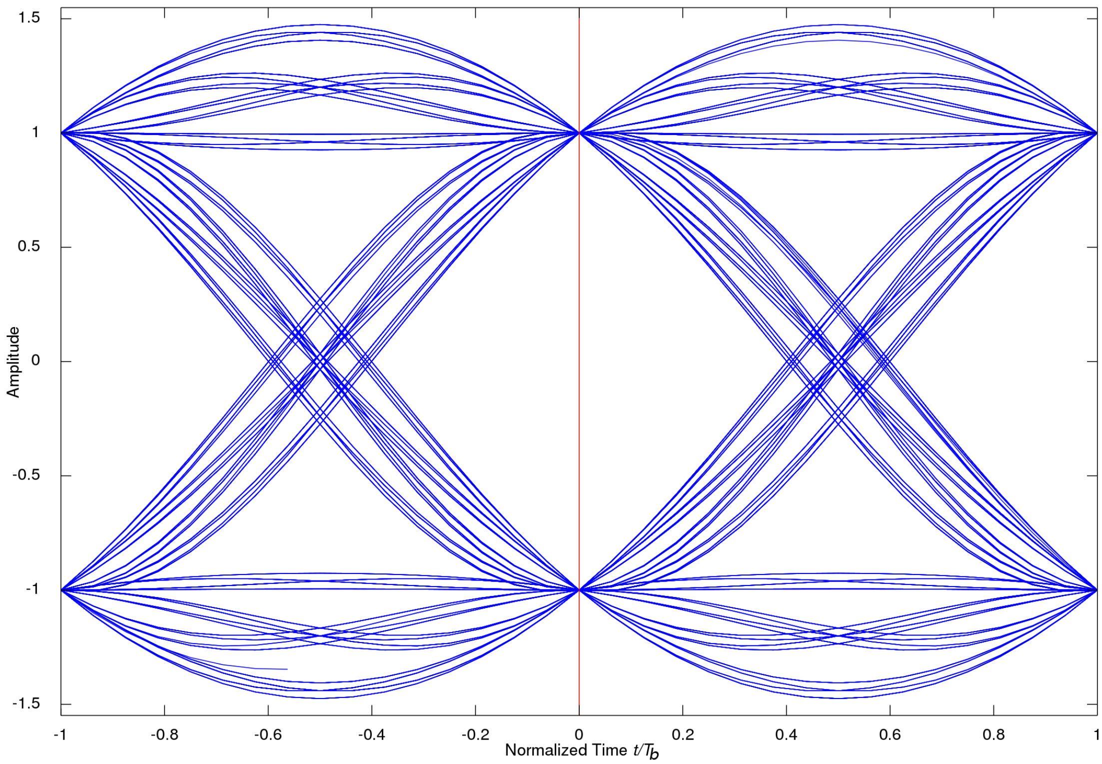
Causes

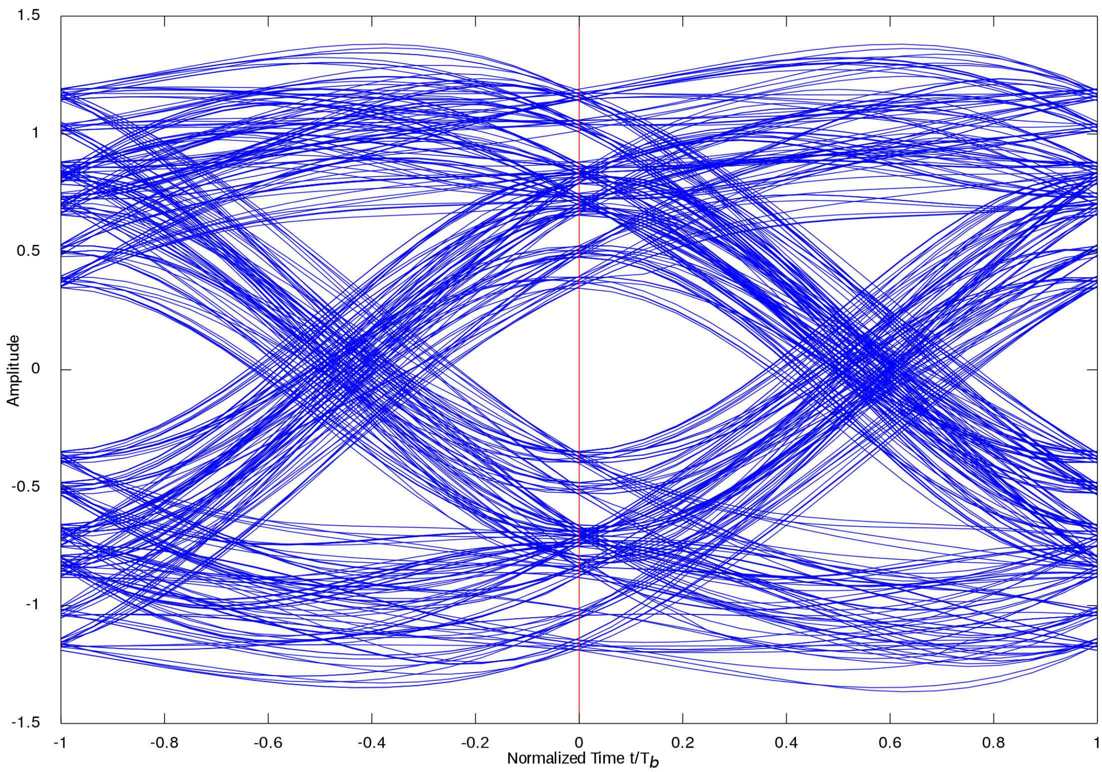
- Bandlimited Channel
  - Channel Frequency Response Characteristics
  - Artificial Restrictions (spectrum allocation)
- Multipath Propogation
  - Reflection
  - Refraction
  - Atmospheric Ducting
  - Ionospheric Reflection





BPSK Multipath Eye Diagram



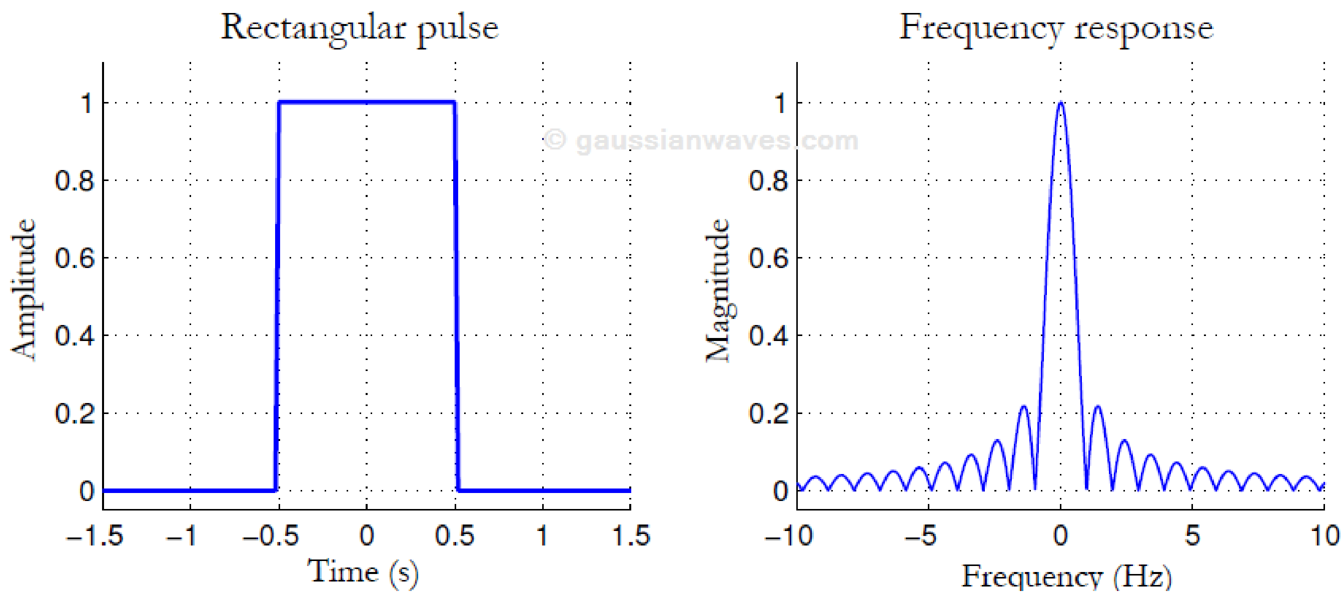


Nyquist ISI Criterion

To achieve zero intersymbol interference, samples must have only one non-zero value at each sampling instant.

The Rectangular Pulse

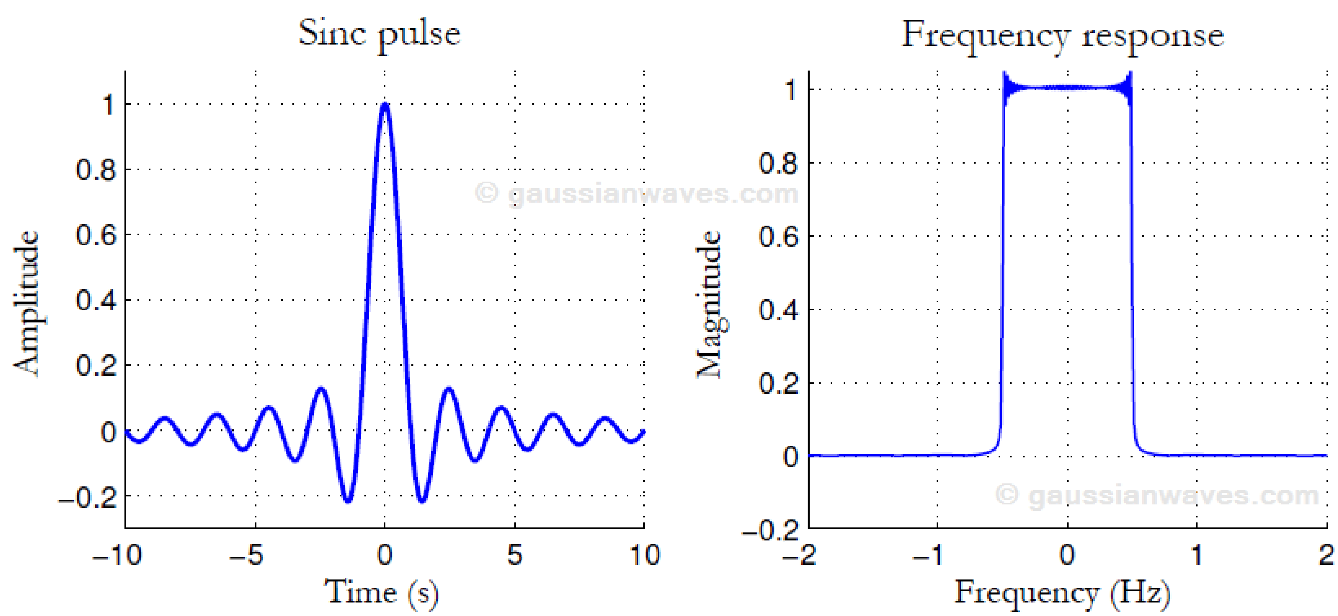
The simplest signal that avoids ISI is the rectangular pulse with width $T_{sym} = 1/F_{sym}$, but it consumes infinite bandwidth. This is because its Fourier transform (the sinc function) extends infinitely on either side of the spectrum.



The Sinc Pulse

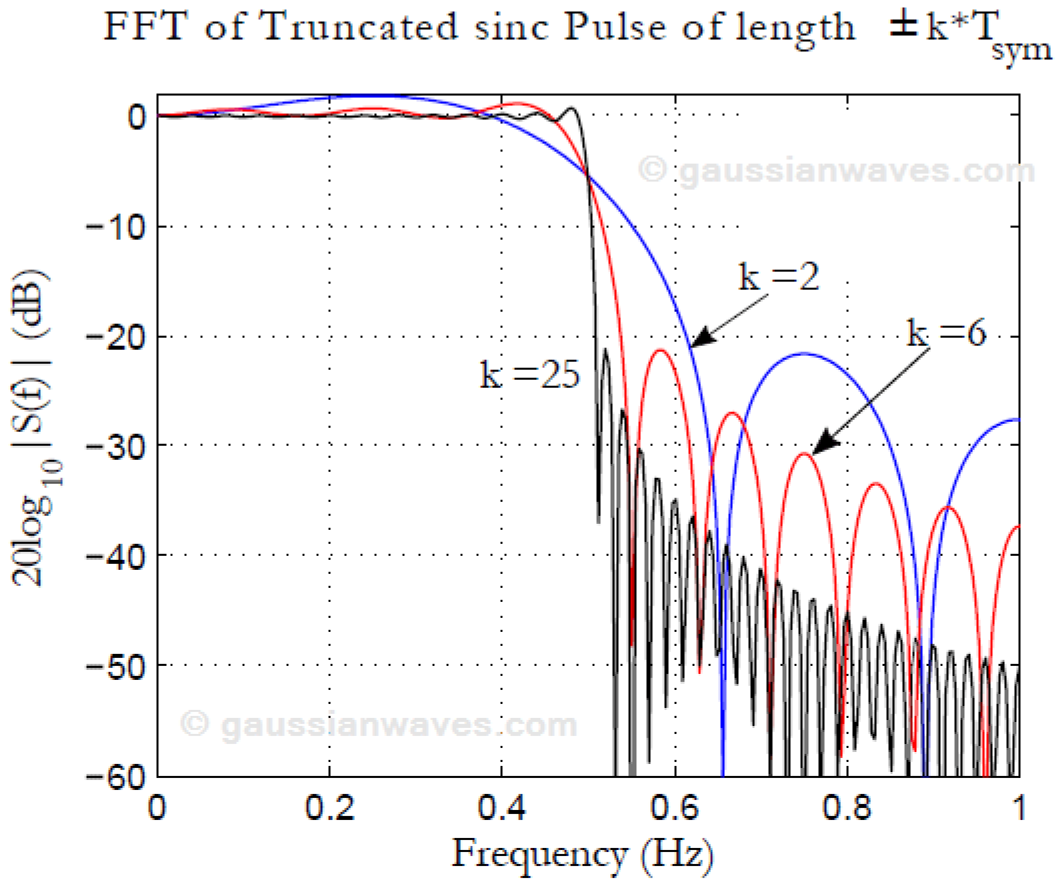
The signal that avoids ISI with the least amount of bandwidth is the sinc pulse with bandwidth $F_{sym}/2$

$$p(t) = \frac{sin\left(\pi t/T_{sym}\right)}{\pi t/T_{sym}} \quad \Leftrightarrow \quad P(f) = \begin{cases} T_{sym}, & \frac{-F_{sym}}{2} \leq f < \frac{F_{sym}}{2} \\ 0, & \text{otherwise} \end{cases}$$



The major drawback of the sinc function is the slow decay rate, which can lead to ISI between symbols that are far apart.

The infinite impulse response makes it practically unrealizable.



Solutions

- Error Correcting Codes
- Adaptive Equalization
- Symbol Guard Periods
- Receiver Sequence Detector

Basic Pulse Shaping Using BPSK

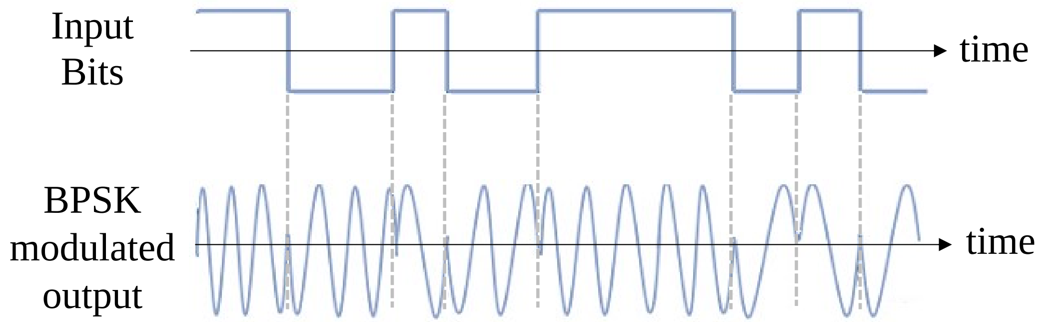
BPSK Fundamentals

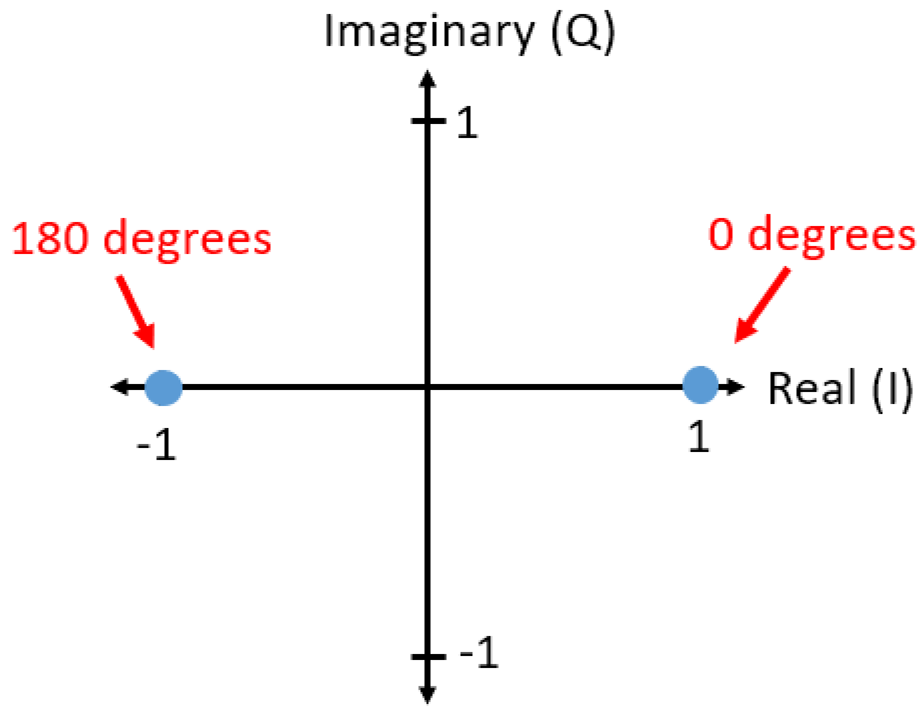
Using 1 bit per symbol is equivalent to transmitting square pulses.

BPSK is a square wave consisting of 1’s and 0's.

There are two phase levels:

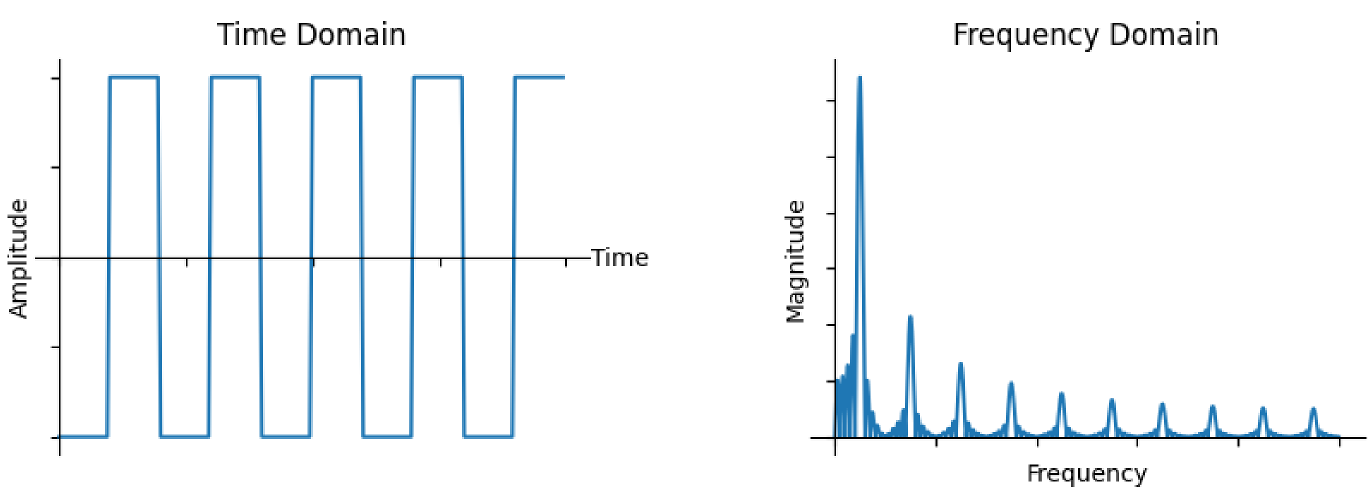
- 0° for binary 1
- 180° for binary 0





But square pulses are not efficient because they use an excess amount of spectrum.

Square wave Fourier Transform

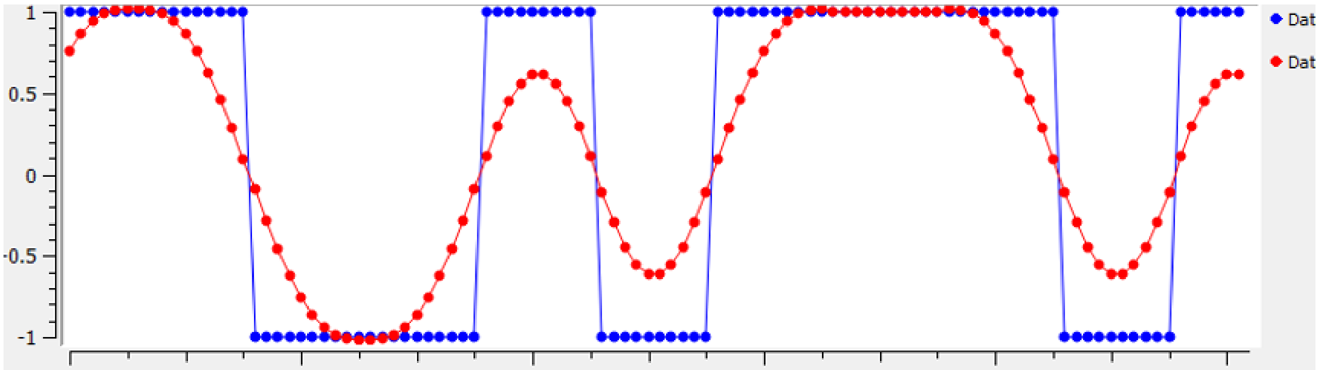


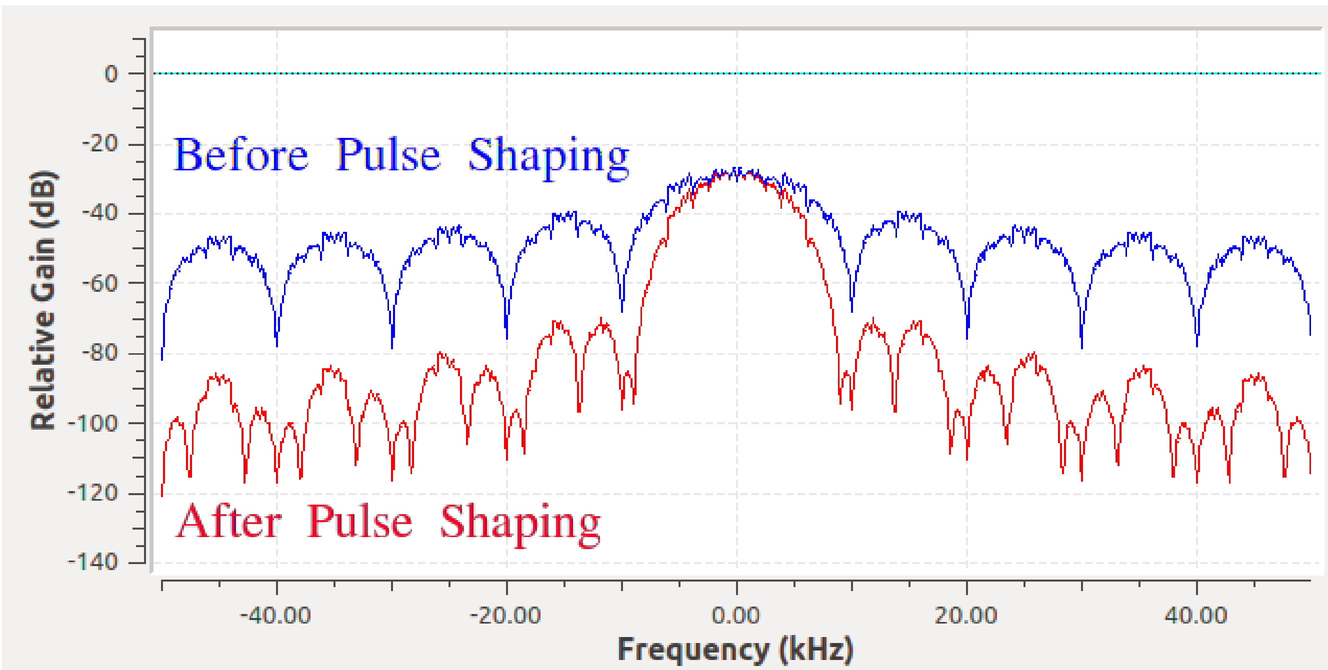
We want to "shape" the blocky-looking symbols so that they take up less bandwidth in the frequency domain.

How?

The Solution

Use a lowpass filter to discard the higher frequency components of our symbols.





The sidelobes are ~30 dB lower after pulse shaping.

The main lobe is narrower, so less spectrum is used for the same amount of bits per second.

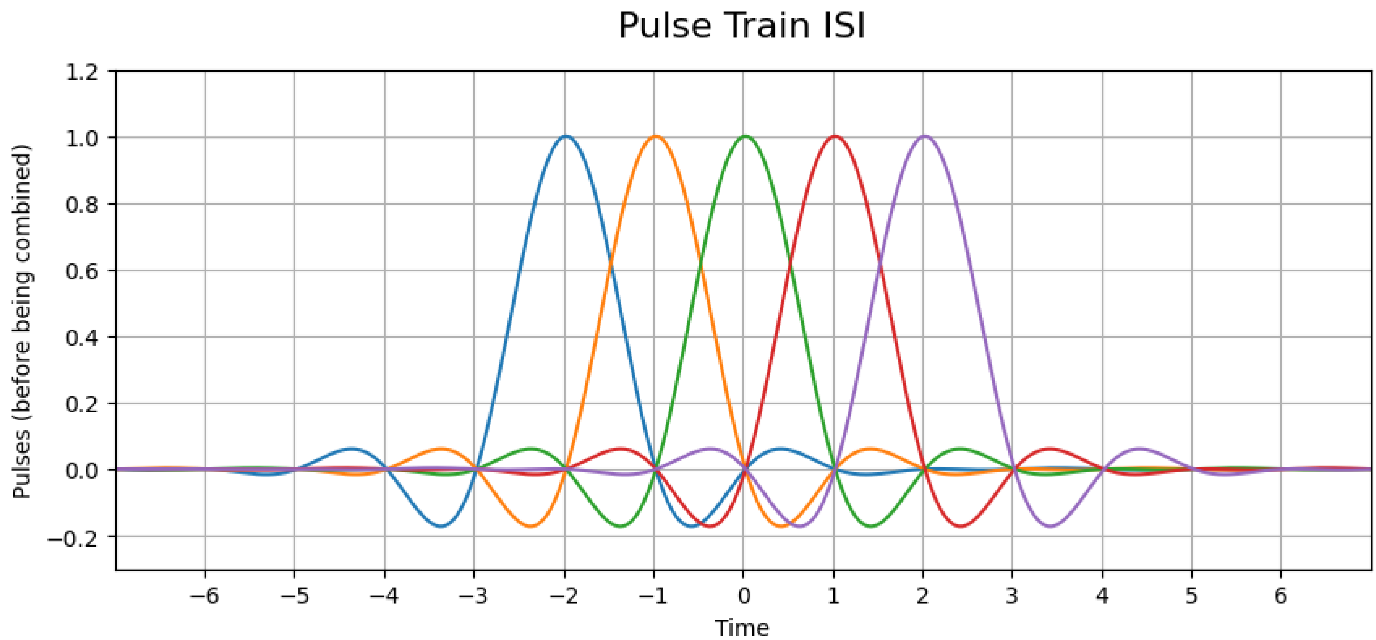
Pulse Shaping Theory

Filters and Symbol Timing

When we apply a pulse-shaping filter, it elongates the pulse in the time domain (in order to condense it in frequency).

This causes adjacent symbols to begin to overlap one another.

This is okay ***if*** all pulses but one add up to zero at every multiple of our symbol period *T*.



At every interval *T*, one pulse peaks while the other pulses are at 0.

We want the receiver to sample the signal at the perfect time (at our pulse peaks), so it must synchronize with the symbol period (somehow).

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*We only care about symbol alignment **at the receiver** when we perform our sampling.*

Matched Filters

Basic Concept

We (normally) want to low pass filter at the transmitter to reduce the amount of spectrum we use.

We also want to low pass filter at the receiver to elimate as much adjacent noise/interference next to the signal as possible.

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Theoretically, the optimal linear filter for maximizing the SNR in the presence of AWGN is to use the same filter at both the Tx and Rx.

</br> The receiver correlates the received signal with the known template signal. </br> The template signal is essentially the pulses the transmitter sends, irrespective of the phase/amplitude shifts applied to them. </br> Filtering is performed via convolution, and, in this case, correlation.

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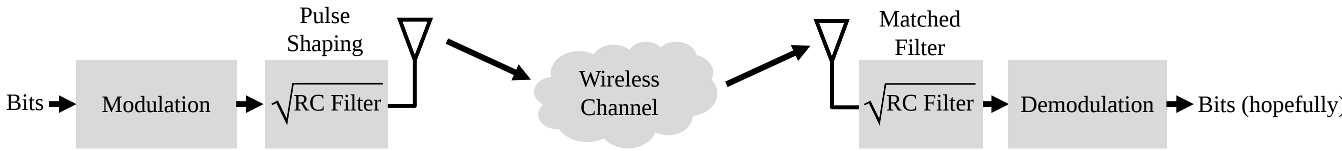
In matched filtering, we convolve the unknown signal with a conjugated time-reversed version of the template.

Filter Splitting

So we split the pulse shaping filter between the transmit and receive sides.



Filter Design

Requirements:

- Reduce our signal bandwidth (to use less spectrum)
- All pulses except one should sum to zero at every symbol interval
- Split the filter into TX and RX portions

Common Pulse-Shaping Filters

Common pulse-shaping filters include:

- Sinc filter
- Raised-cosine filter
- Root raised-cosine filter
- Gaussian filter

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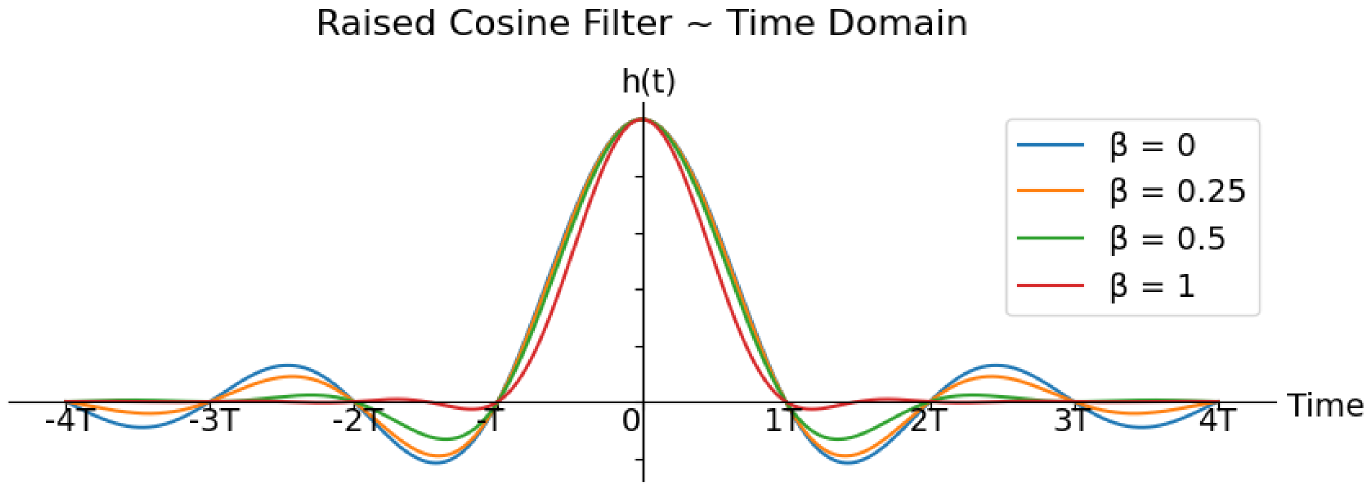
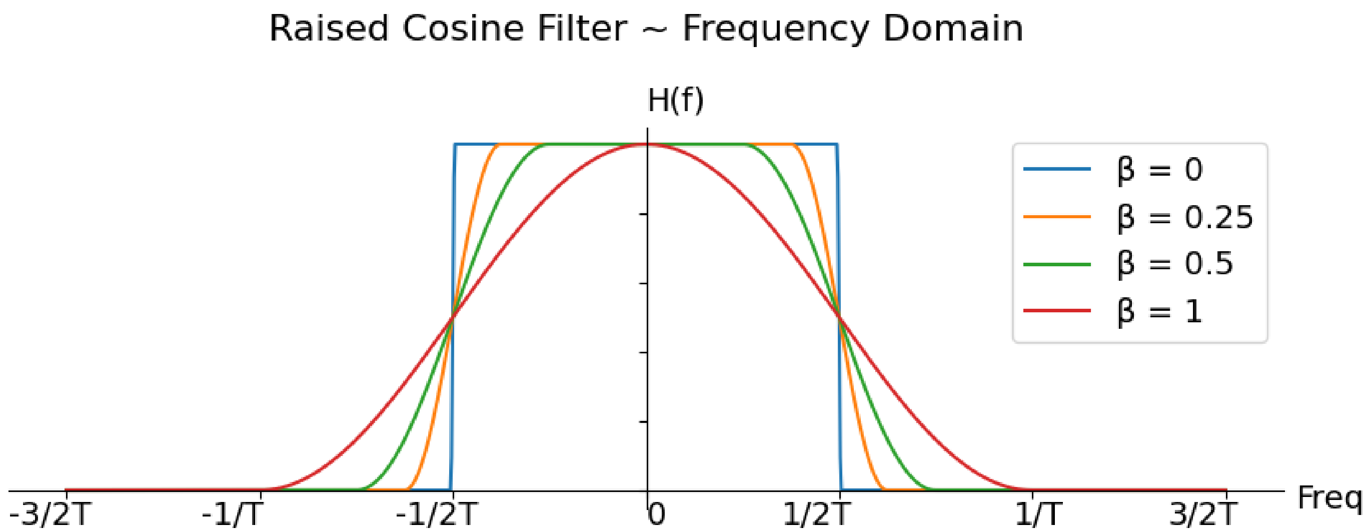
</br> Note that the sinc filter is equivalent to the raised-cosine filter when $\beta = 0$. The sinc filter is an ideal filter as there is no transition region.

Raised-Cosine Filter

The most popular pulse-shaping filter

$$h(t) = \frac{1}{T}\mathrm{sinc}\left(\frac{t}{T}\right)\frac{\cos\left(\frac{\pi\beta t}{T}\right)}{1-\left(\frac{2\beta t}{T}\right)^2}$$





The β parameter is the only parameter for the raised-cosine filter, and it determines how quickly the filter tapers off in the time domain, which will be inversely proportional with how quickly it tapers off in frequency.

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The reason it's called the raised-cosine filter is because the frequency domain when $\beta = 1$ is a half-cycle of a cosine wave, raised up to sit on the x-axis.

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interactive(children=(FloatSlider(value=0.35, description='beta', max=1.0, step=0.01), I
ntSlider(value=101, de…
```

Square-Root Raised-Cosine Filter

The equally split filter portion of the TX/RX raised-cosine filter.

$$h(t) = \begin{cases} \frac{1}{T_s}\left(1+\beta(\frac{4}{\pi}-1)\right) & t=0 \\ \frac{\beta}{T_s\sqrt{2}}\left[\left(1+\frac{2}{\pi}\right)\sin\left(\frac{\pi}{4\beta}\right)+\left(1-\frac{2}{\pi}\right)\cos\left(\frac{\pi}{4\beta}\right)\right] & t=\pm\frac{T_s}{4\beta} \\ \frac{1}{T_s}\frac{\sin\left[\pi\frac{t}{T_s}(1-\beta)\right]+4\beta\frac{t}{T_s}\cos\left[\pi\frac{t}{T_s}(1+\beta)\right]}{\pi\frac{t}{T_s}\left[1-\left(4\beta\frac{t}{T_s}\right)^2\right]} & \text{otherwise} \end{cases}$$

Raised-Cosine Roll-Off Factor

β is called the "roll-off" factor or "excess bandwidth."

It determines how fast (in the time domain) the filter rolls off to zero.

It is a number between 0 and 1.

More taps are required to lower β.

The lower the roll-off, the more compact the frequency of the signal for a given symbol rate.

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Bandwidth can be approximated for a given symbol rate and roll-off factor as:

$BW = R_s(\beta+1)$, where R_s is the symbol rate in Hz.

Advanced BPSK Pulse Shaping

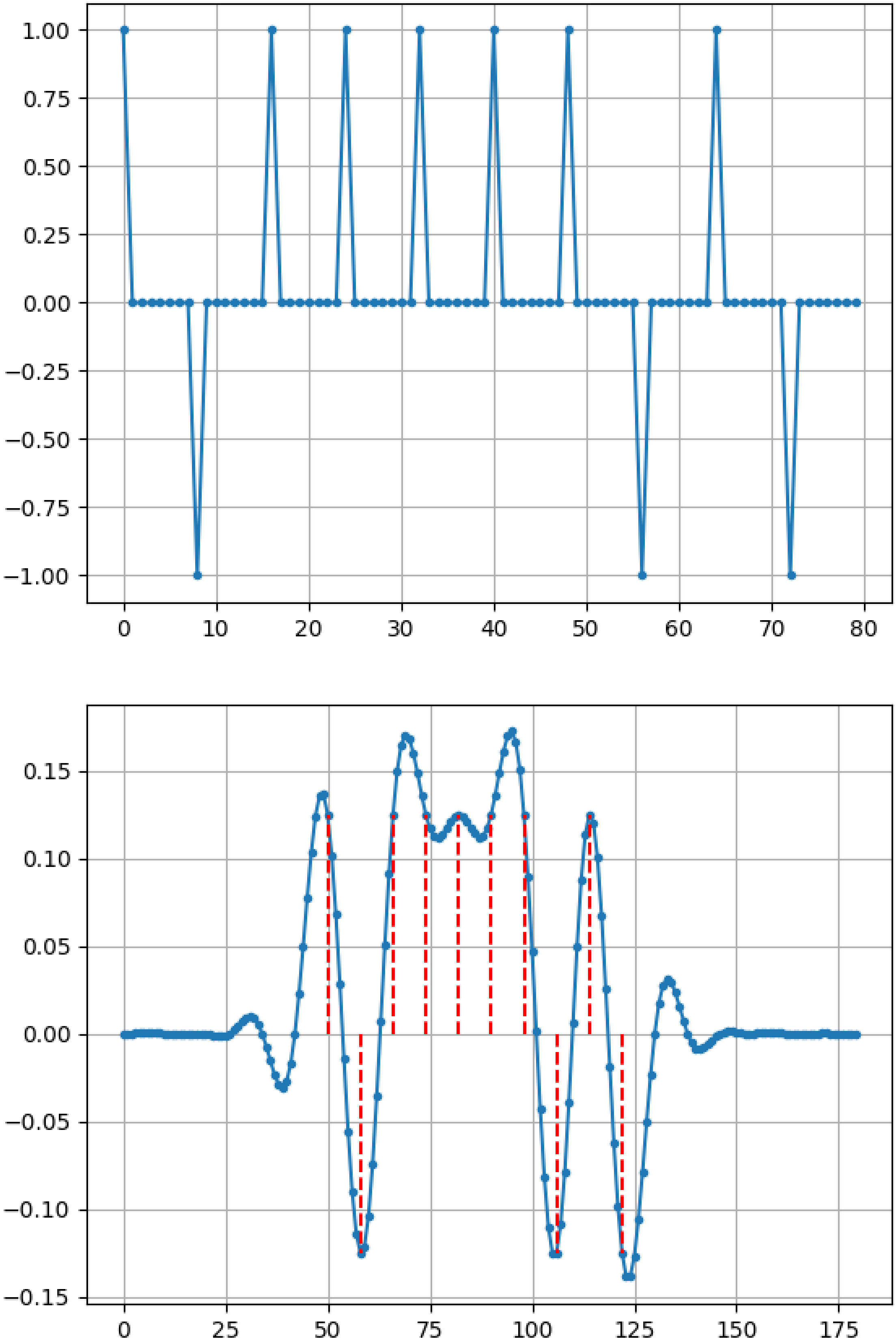
BPSK involves transmissitng 1's and -1's with the "Q" portion equal to zero, so we only need to plot the in-phase portion.

This is equivalent to bipolar 2-ASK, as a 180° phase shift is equivalent to multiplying the sinusoid by -1.

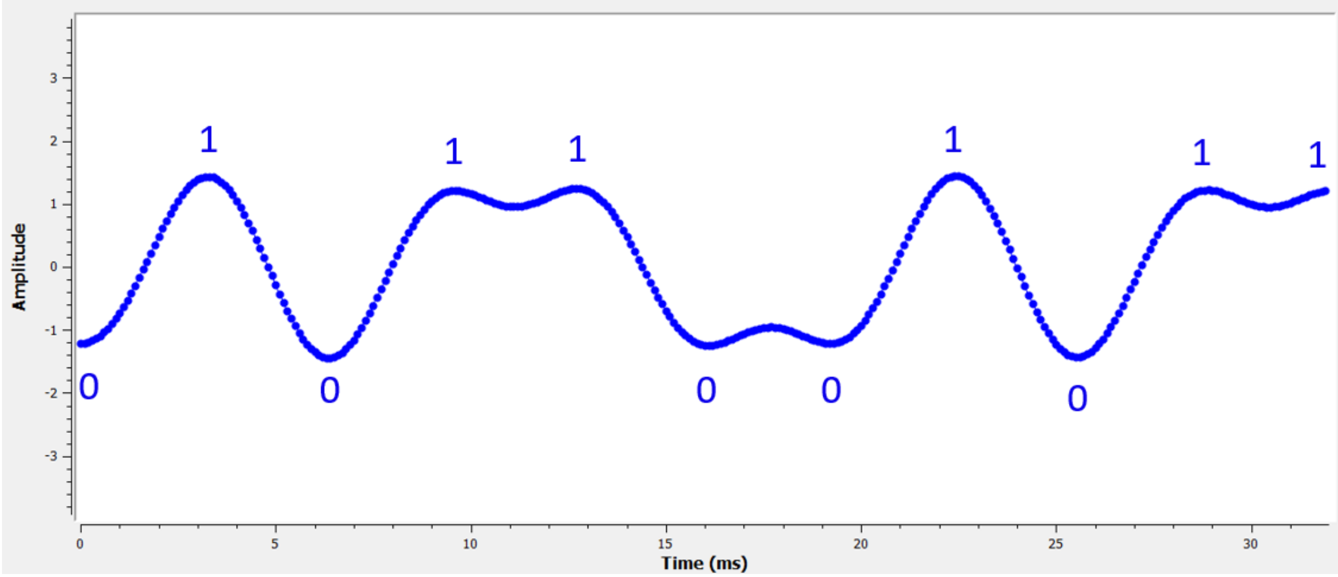
The Impulse Train

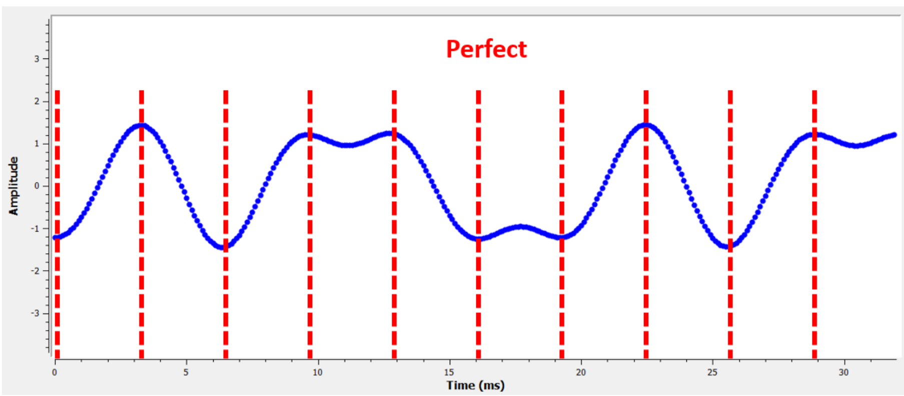
An Example Generated Setup:

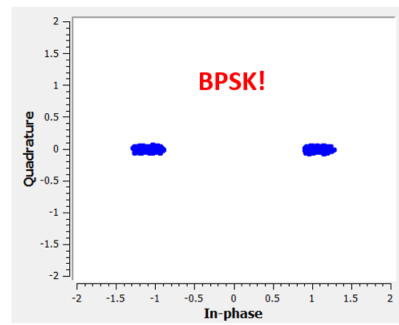
```
bits: [0, 0, 1, 1, 1, 1, 0, 0, 1, 1]
BPSK symbols: [-1, -1, 1, 1, 1, 1, -1, -1, 1, 1]
Applying 8 samples per symbol: [-1, 0, 0, 0, 0, 0, 0, 0, -1, 0, 0, 0, 0,
0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, ...]
```

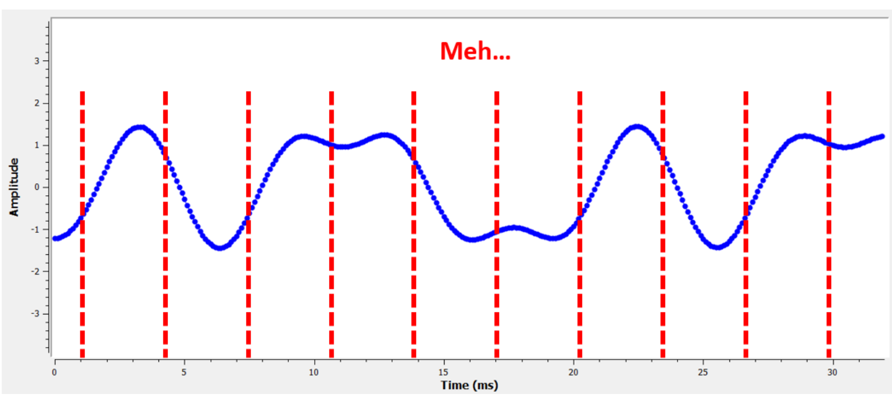


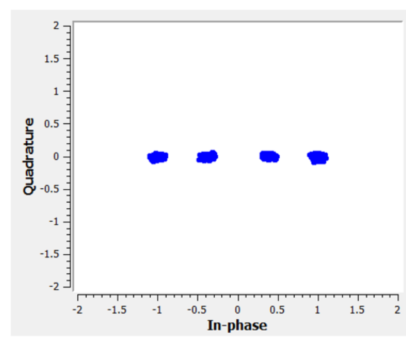
BPSK in GNU Radio

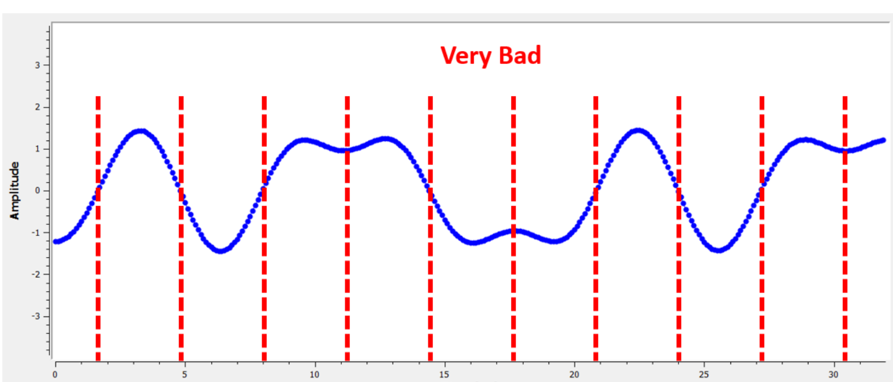

Amplitude
Time (ms)
0 1 0 1 1 0 0 1 0 1 1

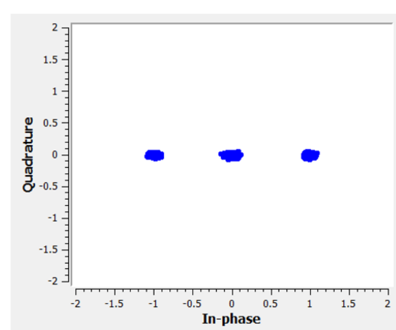

Perfect
Amplitude
Time (ms)


BPSK!
Quadrature
In-phase


Meh...
Amplitude
Time (ms)


Quadrature
In-phase


Very Bad
Amplitude
Time (ms)


Quadrature
In-phase