Improving Spectral Efficiency in Space Communications using SRRC Pulse-Shaping Technique

Simon Wasinya Pallam, Hikma Shabani, Dayang Azra Awang Mat, Norhuzaimin Julai
Department of Electrical and Electronics Engineering, Faculty of Engineering,
Universiti Malaysia Sarawak (UNIMAS), 94300 Kota Samarahan, Sarawak, Malaysia.
hikmash@hotmail.com

***Abstract*—This paper proposes an improved spectral efficiency in space communications using square root raised cosine (SRRC) pulse-shaping technique. The proposal is necessitated by the global shortage of bandwidths confronting the wireless communication system. The major contribution of this technique is its ability to reduce the channel bandwidth and eliminate inter-symbol interference and spectral seepages with ease. This is achieved by tightly constraining wider channel bandwidths in a given frequency spectrum using SRRC pulse-shape filtering, thus eliciting more channels, higher data rates per channel and more users in the system. To authenticate the rationality of the proposed technique, MATLAB simulations are carried out under varying SRRC filter roll-off factors. The proposed technique is capable of achieving 43.76% channel bandwidth improvement, which is close to an ideal case of 50%. This is a great achievement in band-limited communication system. It is envisioned that this technique will be very helpful to designers of high performing and bandwidth efficient space communication system.**

***Index Terms*—Inter-Symbol Interference; Pulse-Shaping; Raised Cosine Filter; Spectral Efficiency; Square Root Raised Cosine Filter.**

I. INTRODUCTION

Pulse-shape filtering is a vital phenomenon in wireless communication system due to its ability to reduce overall carrier transmission power and channel width as well as eliminate spectral seepages and ISI. The grand purpose of pulse-shaping in digital communication is to make the transmitted signal's bandwidth match that of the communication channel, by limiting the effective bandwidth of the signal. In multi-channel communication systems, limiting all the power of the modulated carriers to just the carrier bandwidth leads to more concentrated frequency band with reduced overall transmission power and improved spectral efficiency. The spectral efficiency of a transmission system, which is the amount of transmitted information that can fit into a given channel bandwidth is expressed in bits/s/Hz. By limiting a channel to the specific frequency band, therefore, helps in eliminating adjacent channel interference [2, 3].

In bandwidth-limited digital data transmission systems, pulse-shaping technique is mostly employed to constrain the signal bandwidth and minimize the likelihood of decoding errors at the receiver. This technique can be achieved by using root raised cosine (RRC) filtering. The RRC spectrum manifests a good characteristic of odd symmetry about $1/2\tau$, where τ is the transmission symbol period. Although pulse-shaping technique has been used in commercial communication systems such as cellular technology over the years, it is still relatively new in space communications [4]. With the global congestion of frequency spectrum in space communications, it is imperative to look into diverse techniques that can improve the spectral efficiency. It is pertinent to note that space communication systems have stringent requirements of high data rates per channel and narrow channel bandwidth. However, as the size of the channel bandwidth increases in the course of providing high data rates, the number of channels allotted in a fixed spectrum must be proportionally limited. In order to address these two scenarios (i.e. generating band limited channels and reducing ISI without compromising each other, the RRC filtering technique needs to be employed. The technique involves tightly packing wider channel bandwidths in the frequency spectrum to achieve high data rates per channel as well as more channels in the system. The filtering efficiency can be further improved by adjusting the filter order. Using RRC filtering in space communication transmitter and receiver can improve spectral efficiency and bit error rate (BER) performance of the system [5].

Theoretically, high data rates in communication systems imply that more information can be transmitted through the channel [6]. When the channel bandwidth is reduced to a narrower band by compressing the signal bandwidth, there will be more channels, more users and less noise in the system. However, if the channels are too narrow, the symbols will be too wide and will lead to ISI. To correct this effect, an ideal low-pass filter (ILPF) can be used to filter the transmitted signal. The spectrum of the transmission can, therefore, be determined by the pulse-shaping filter.

II. PULSES IN DIGITAL COMMUNICATIONS

A. Rectangular Pulse Energy

In digital communications, data can be transmitted in bits (binary digits) or symbols (group of bits) in form of pulses of energy. The most fundamental of these pulses is the rectangular pulse, which is synonymous with binary data (0 and 1) [7]. The '1' bit turns on the energy source for a duration of one pulse interval (τ seconds), while the '0' bit turns off the energy source during one pulse interval. The rectangular pulse shown in Figure 1 has a defined amplitude A=1 and a defined duration $\mathrm{T}=\tau$ (spanning from $-\tau/2$ to $\tau/2$) with the pulse centered at $\tau = 0$. The transmitted data is often encoded in the amplitude of the pulse. Since the pulse width is equivalent to $\mathrm{T}=\tau$, the pulse rate at A=1 is $1/\tau$ pulses per second, which implies that the data rate is $1/\tau$ bits per second. In complex data transmission systems, multiple bits can be encoded into the pulse as well as multiple pulses can be transmitted simultaneously. A group of multiple bits representing a unit of data is referred to as a symbol. When a pulse of width τ is used to transmit a symbol, the symbol rate is $1/\tau$ symbols per second. Generally, the problem associated with rectangular pulse is that the bandwidth extends to infinity, and therefore its unbounded frequency response makes it unsuitable for bandwidth-limited digital transmission systems [8]. In bandwidth-limited data transmission systems, the transmitted signals must be confined to certain bandwidth. This is where pulse-shaping technique plays an important role in shaping the rectangular waveform to a smooth shape and matching the signal bandwidth to channel bandwidth.

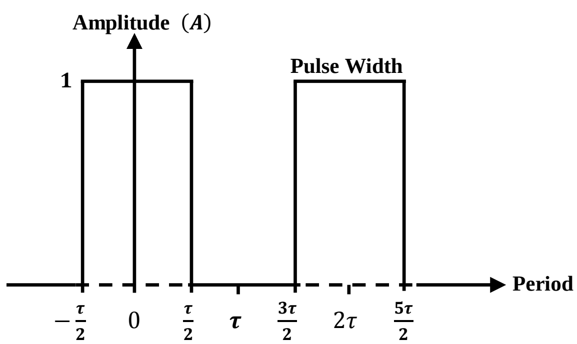

Figure 1: Rectangular Energy Pulse

B. Rectangular Pulse Spectrum

The application of Fourier Transform to time-domain rectangular waveform usually produces the spectrum response (frequency content) of the rectangular pulse. The spectrum response plot of a rectangular pulse is shown in Figure 2.

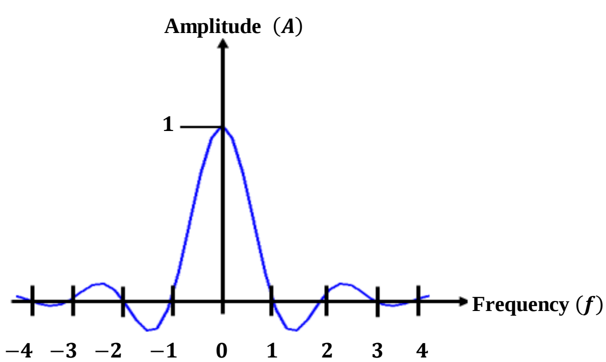

Figure 2: Spectrum Response of a Rectangular Pulse

The spectrum $h(t)$ of the rectangular pulse, often referred to as sinc response, is given as:

$$h(t) = \frac{sin(\pi t/\tau)}{\pi t/\tau} \tag{1}$$

where: t = sampling period
τ = symbol period

The function h(t) is a continuous Inverse Fourier Transform (IFT) of the rectangular pulse of width 2π and unity height. The sinc pulse is, therefore, given by:

$$sinc(t) = \frac{1}{2\pi}\int_{-\pi}^{\pi} e^{i\omega\tau}\, d\omega \tag{2}$$

To increase data rates in a data communication system, the value of pulse width τ must be decreased, which invariably leads to proportional increase in the frequency bandwidth. The data rates and bandwidth relationship poses problems for band-limited transmission systems. The challenge of every data transmission system, therefore, is to achieve the highest possible data rate in the allotted bandwidth with minimal or no errors. Pulses are usually sent by transmitters and eventually detected by the receivers in data transmission systems. The received signal is normally sampled at an optimal point in the pulse interval to maximize the probability of an accurate binary decision. This process eliminates the interference of pulse shapes with one another.

III. Applications of Pulse-Shaping

A. Digital Pulse-Shaping Filters

Pulse-shaping can be implemented after line coding and modulation of a transmitted signal using different types of sinc pulses imposed on the modulated signal. Passing the time-domain rectangular pulse signal through the digital pulse-shaping filter eliminates the sharp rectangular edges of the pulse to produce a resultant frequency-domain smooth shape as shown in Figure 3.

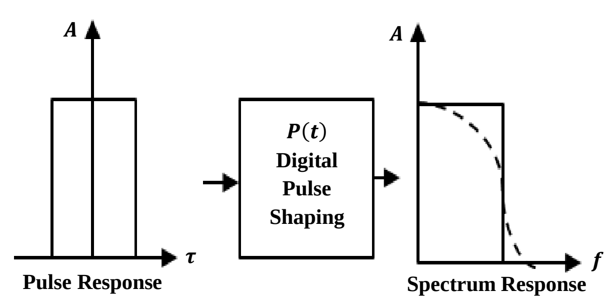

Figure 3: Pulse-Shaping using Digital Filters

In signal processing, a sinc filter is an idealized filter that removes all frequency components above a given cutoff frequency without affecting lower frequencies, and has linear phase response. The sinc filters commonly used in pulse-shaping technique are as follow:

i. The raised cosine filters,
ii. The square root raised cosine filters, and
iii. The Gaussian filters

Pulse-shaping using raised cosine filter is very common in communication systems. Usually, the starting and the end portions of the symbol period often create interference from multipath distortions [9]. Attenuating these portions using the raised cosine filter can significantly reduce the ISI. The block diagram of typical data communication system is shown in Figure 4.

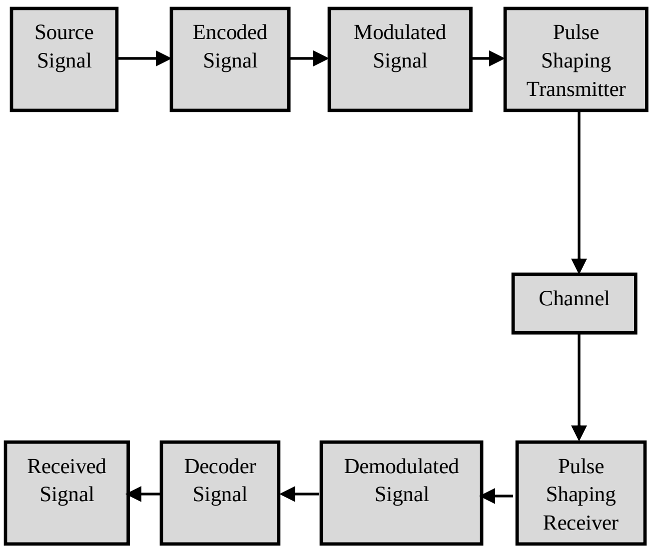

Figure 4: Block Diagram of a Data Communication System

B. Pulse-Shaping using the Raised Cosine Filters

The unbounded frequency response of rectangular pulse makes it unsuitable for band-limited digital transmission systems, which case leads to the quest for pulse-shaping technique. The bandwidth of the rectangular pulse can be limited by constraining it to pass through a low-pass filter. This action changes the shape of the pulse from purely rectangular to smooth contour devoid of sharp edges. As a result of the convolution of the rectangular pulse with raised cosine response, damped ripples are created before and after the pulse intervals. This scenario causes ISI and decoding errors at the receiver of the communication system. The ripples can be attenuated by a special filter which also matches the transmission bandwidth to the channel bandwidth. Pulse-shaping is therefore a process of filtering the rectangular pulses in time domain. As depicted in the raised cosine filter impulse response shown in Figure 5, time-domain ripples occur at various amplitudes with varying filter roll-off factor α, where $0 \leq \alpha \leq 1$. These ripples must occur as a result of bandwidth containment in digital communication systems. Hence, a tradeoff between bandwidth reduction and increase in ripple amplitude must be an issue of special filter design.

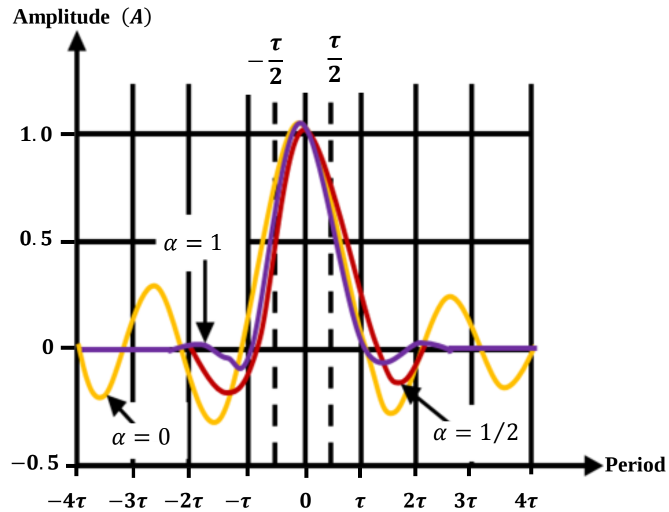

Figure 5: Raised Cosine Time-Domain Response

Raised cosine response is a very vital technique in pulse-shaping in data transmission systems because it exhibits time domain ripples crossing through zero point at the middle of the adjacent pulse intervals, thus minimizing ISI or decision making errors in the receiver. The filter-block at the transmitter shown in fig. 4 up-samples and filters the input signals using a normal raised cosine finite impulse response (FIR) filter. The impulse response h(t) of a normal raised cosine (RC) filter with a roll-off factor α, a sampling period t and a symbol period τ is given as:

$$h(t) = \frac{sin(\pi t/\tau)}{\pi t/\tau} * \frac{cos(\pi \alpha t/\tau)}{\left(1 - \frac{4\alpha^2 t^2}{\tau^2}\right)} \tag{3}$$

To achieve the most efficient (minimum) bandwidth, the roll-off factor α must be equal to zero, although the time domain ripple amplitude will be non-zero at this instant. Therefore, at α=0, Equation (3) will elicit an ideal low-pass rectangular spectrum given in Equation (1). Conversely, increasing the roll-off factor α from 0 to 1 increases the bandwidth from the minimum to maximum, while the amplitude of the time domain ripple is decreased to zero level respectively. As depicted in Figure 5, when roll-off factor varies from 0 to 1, the bandwidth varies from ½f to f respectively, where f is the sampling frequency. Raised cosine response has three notable frequency points, namely: Nyquist frequency, Stopband frequency and the Passband frequency. The Nyquist frequency, which occurs at ½f or ½ the symbol rate, is the ideal and minimum possible channel bandwidth used for transmitting data without distortion [10]. At Nyquist frequency, the filter sampling rate must be twice the input signal bandwidth in order to avoid aliasing. Therefore, bandwidth efficient modulation devoid of ISI inducement is always achieved at Nyquist bandwidth. This is to say that:

$$Bandwidth_{\ Nyquist} = ½ * \text{sampling rate} \tag{4}$$

The stopband frequency is the point at which the response first reaches the zero magnitude (A=0). Mathematically:

$$f_{stopband} = ½(1 + \alpha)f \quad (5)$$

The passband frequency is the frequency at which the response first deviates from the peak magnitude (A=1). The equation for the passband frequency is given as:

$$f_{passband} = ½(1 - \alpha)f \quad (6)$$

The dotted line in the raised cosine frequency response of figure 6 refers to the trace of rectangular pulse spectrum.

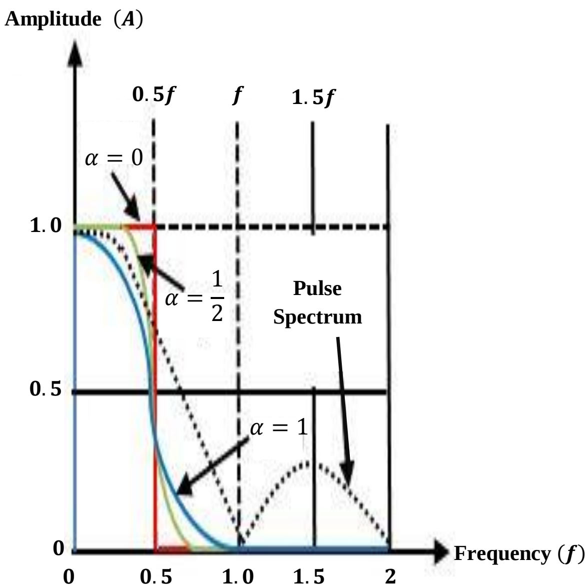

Figure 6: Raised Cosine Frequency Response

C. Pulse-Shaping using the SRRC Filters

Due to the global dearth of spectrum and stringent bandwidth requirements of space communication system, different techniques of improving the spectral efficiency are being explored. Square root raised cosine (SRRC) pulse-shaping technique is one of the most viable techniques in improving the spectral efficiency in space communication system. The technique will enable optimum usage of spectrum by the spacecraft and earth stations. A typical example of frequency band allocation by ITU to space operation (SO), space research (SR) and earth exploration-satellite (EES) services is shown in Table 1. The maximum occupied bandwidth for spacecraft in the 2.2GHz-2.29GHz band (space-earth direction) does not exceed 6MHz. The occupied bandwidth is just sufficient to transmit data at a specified rate and quality under predefined conditions [11].

Table 1
Frequency Allocation to Space Services

Frequency Band (MHz)	Allocated Services	Communication Directions
2025 – 2090	SO, SR and EES	Earth – space
2110 – 2120	SR	Earth – space
2200 – 2290	SR, SO and EES	Space – earth
2290 – 2300	SR	Space – earth
7145 – 7190	SR	Earth – space
7190 – 7235	SR	Earth – space
8925 – 8400	EES	Space – earth
8400 – 8450	SR	Space – earth
8450 – 8500	SR	Space – earth
25500 – 27000	SR	Space – earth
31800 – 32300	SR	Space – earth
34200 – 34700	SR	Earth – space
37000 – 38000	SR	Space – earth
40000 – 40500	SR	Earth – space

To reduce the signal bandwidth without exacerbating the ISI, a special digital filter known as SRRC filter is used to filter and down-sample input and received signals respectively [12]. SRRC filter is implemented by the convolution of two identical root raised cosine responses at both the transmitter and the receiver (i.e. a product of two responses, such that their combined response equals that of Nyquist filter). At the transmitter, the out-of-bound signal is prevented from being transmitted, and at the receiver, the out-of-bound signal is rejected. In this scenario, SRRC filter provides matched filtering, with a net response of zero ISI at the receiver. The impulse responses of a SRRC filter with a roll-off factor α, symbol period τ and sample time t is given as:

$$g(t) = 4\alpha \left\{ \frac{cos((1+\alpha)\,\pi t/\alpha) \; + \; \dfrac{sin((1-\alpha)\,\pi t/\tau)}{(4\alpha t/\tau)}}{\pi\sqrt{\tau}\left(1 \; - \left(\dfrac{16\alpha^2 t^2}{\tau^2}\right)\right)} \right\} \quad (7)$$

The convolution of two identical impulse responses of an SRRC filter will produce an impulse response of a normal raised cosine filter. The group delay (number of symbol periods between the start and the peak of the filter's response) and the oversampling rate determine the length (number of taps) of the filter's impulse response. This is shown in Equation (8).

$$length = 2 * N * group\ delay + 1 \quad (8)$$

where: N = Oversampling rate

The ground delay is calculated as:

$$Group\ delay = \frac{Filter\ Order}{2*N} \quad (9)$$

The filter order, which is an even number of samples, is determined by the Equation (10).

$$Filter\ Order = N * No.of\ Symbols \tag{10}$$

The data sampling frequency F_S, of the digital filter is given as:

$$Fs = data\ rate * upsampling\ factor \tag{11}$$

The roll-off factor which is a real number between 0 and 1, determines the filter's excess bandwidth. This is to say that when a filter has a roll-off factor of 0.125, its bandwidth is 1.125 times the input sampling frequency. Therefore, the overall gain for the normal raised cosine (RC) filter is given as:

$$Gain = 20\log(N * linear\ amplitude\ gain) \tag{12}$$

Similarly, the gain for the SRRC filter is given as:

$$Gain = 20\log\,(N * \ linear\ amplitude\ gain)^{1/2} \tag{13}$$

The roll-off factor, which ranges from 0 to 1, is a direct measure of the occupied bandwidth (excess bandwidth beyond the Nyquist bandwidth) of the transmission system. At Nyquist frequency, the occupied bandwidth is equal to the symbol rate and it is the minimum bandwidth required to transmit a signal without distortion. Furthermore, the maximum bandwidth required to transmit a signal through the channel is twice the symbol rate. The occupied bandwidth of the bandpass QAM modulation is calculated as follows:

$$Occupied\ Bandwith = Symbol\ rate\ (1 + \alpha) \tag{14}$$

Where symbol rate is the number of symbols transmitted in a unit time (s). Symbol rate (symbols/s), also known as modulation rate, is calculated as follows:

$$Symbol\ rate = \frac{Data\ rate}{Bits\ per\ Symbol} \tag{15}$$

From Equation (14), it infers that in an ideal system (α=0), the occupied bandwidth is the same as the symbol rate. Notwithstanding, it is practically impossible to achieve an ideal situation in transmission system. At a roll-off factor α=1**,** the occupied bandwidth is twice the symbol rate. In order to achieve a near ideal situation, the filter roll-off factor α=0.125 is selected, making the occupied bandwidth to be 1.125 times the symbol rate. It should be noted that in order to prevent aliasing, a digital filter used for pulse-shaping must operate at a sampling rate of at least twice the data rate. The filter must, therefore, oversample by a factor of 2 in order to obtain a good response characteristic. Nevertheless, in order to avoid a half-symbol delay phenomenon in digital communication system, it is technically advisable to select an odd oversampling rate. The number of taps in a filter is equal to the impulse response duration (i.e. the number of samples the response must span). This is given in Equation (16):

$$Taps = N * No.of\ symbols + 1 \tag{16}$$

IV. Bandwidth Efficiency

Channel bandwidth is a vital phenomenon in the design of transmission system. It is impossible to design a transmission system without channel bandwidth specification. The channel bandwidth determines the type of signal to be transmitted in the system. In order to avoid signal distortion, channel bandwidth is usually greater than signal bandwidth. The global telecommunication regulatory bodies make spectrum available to service providers, mostly with a common divisor of 5MHz as shown in the chart of Figure 7. Sub- or super-channelized bands can be in multiple of Nx5MHz where N may be an integer or fraction.

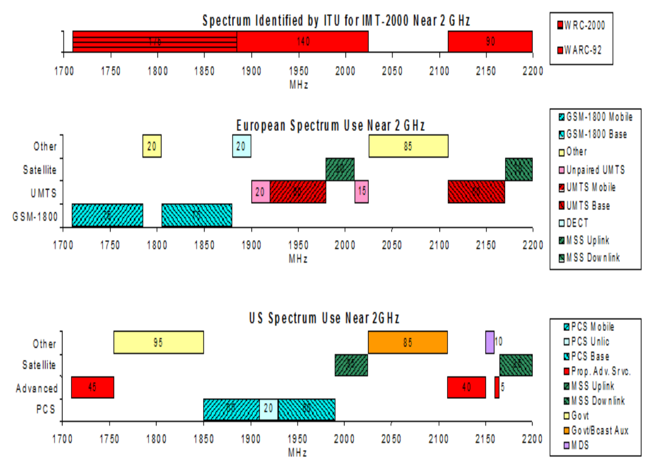

Figure 7: Spectrum Chart by Regulatory Bodies

In a transmission system, the key design quantities: bandwidth B_w, Transmitted power P and data rate $\mathcal{R}$, satisfy the following relationships [13].

$$\mathcal{R}\frac{E_b}{N_0} = \frac{P}{N_0} \tag{17}$$

where: E_b = transmitted energy per information bit
N_0 = noise spectral density

The spectral efficiency of the transmission system, also known as Bandwidth efficiency ή (in bps/Hz) is expressed as the ratio of data rate $\mathcal{R}$ to the required channel bandwidth B_w. It is also expressed as a product of Shannon or channel capacity C (in bps) and E_b/N_0 as shown in Equation (18).

$$ή = \frac{\mathcal{R}}{B_w} = C\left(\frac{E_b}{N_0}\right) \tag{18}$$

The maximum data rate that can be supported by a channel depends on three factors, namely: channel bandwidth, signal level and noise level. The noise level in transmission channel is determined by the channel quality. There are two types of channel qualities in the system: the ideal or noiseless channel and the realistic or noisy channel. The theoretical maximum data rate $\mathcal{R}$ for ideal or noiseless channel as defined by the Nyquist bit rate formula is given as:

$$\mathcal{R} = 2 * B_w * \log 2L \qquad (19)$$

where: L= Number of signal level

Similarly, the theoretical maximum data rate for realistic or noisy channel as defined by Shannon capacity is given by:

$$C = B_w * \log 2\,(1 + \mathrm{SNR}) \qquad (20)$$

where: SNR= Signal-to-Noise Ratio

Thus, when the ratio E_b/N_0 is known, the SNR can be calculated as follows:

$$SNR = E_b/N_0 + 10log10\,(K * coderate) - 10log10(nsamp) \qquad (21)$$

where: k = Bits/symbol
$nsamp$ = Oversampling rate
$coderate$ = Coding rate of filter

It should be noted that Shannon capacity does not consider the number of levels of the transmitted signals as done in Noiseless channel. The required bandwidth to transmit information of data rate $\mathcal{R}$ with power P is given by:

$$B_w = \frac{\mathcal{R}}{C\left(\frac{1}{\mathcal{R}} * \frac{P}{N_0}\right)} \qquad (22)$$

V. Simulation Results and Discussions

The SRRC pulse-shaping technique can be simulated using Matlab 2013a tools. However, this design may not be possible without specifying the filter parameters, as shown in Table 2.

To design filter impulse responses spanning 4 symbols using 64-QAM modulation of 6bits/symbol, with 53 taps and oversample rate of 13, the SRRC filter roll-off factor will be varied from 0 to 1. Matlab 2013a is used to simulate and visualize the impulse and magnitude responses using fdesign and (fvtool) tools as shown in Figure 8 and Figure 9 respectively.

Table 2
Filter Design Specifications

S/N	Parameters	Specification
1	Pulse shape	SRRC
2	Modulation scheme	64-QAM
3	Bits per Symbol	6
4	Symbol rate	5.057MBaud
5	Sampling frequency	Normalized
6	Samples per Symbol	13
7	Number of Symbols	4
8	Filter order	52
9	Roll-off factor	0.125
10	Transmission channel	AWGN
11	Eb/No	20dB
12	Code rate	2/3
13	Channel bandwidth	6MHz

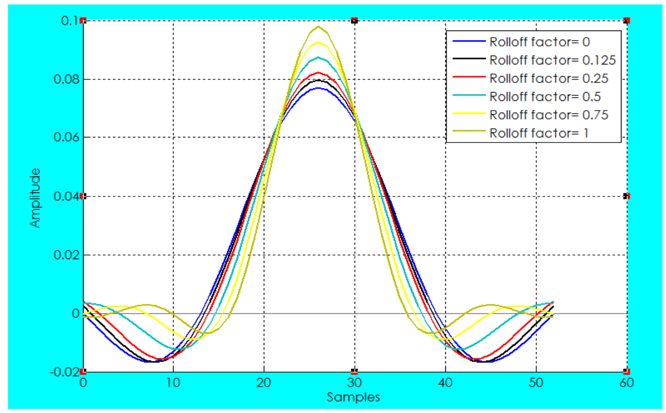

Figure 8: Impulse Responses of Roll-Off Factors 0 to 1

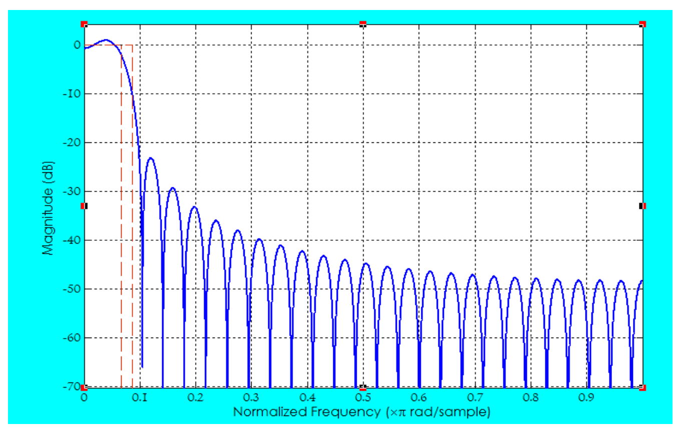

Figure 9: Magnitude Responses of SRRC Filter at $\alpha = 0.125$

From the impulse responses shown in fig. 8, it can be seen that the higher the roll-off factor, the steeper the response, implying a larger bandwidth. Conversely, as the roll-off factor decreases, the time-domain ripple level increases, implying that the bandwidth is reduced but at the expense of elongated impulse response. At a roll-off factor of 1, the time-domain pulse tails decay is maximized. Since the symbol rate of 64-QAM modulation is 5,057,000sym/s or 5.057Mbaud, operating the filter at a roll-off factor of 0.125 using Equation (14) will yield the occupied bandwidth as:

$$Occupied\ bandwidth = 5.057\ (1 + 0.125) = 5.69\ MHz$$

Similarly, when the filter is operated at a roll-off factor of 1, the occupied bandwidth will be:

$$Occupied\ bandwidth = 5.057\ (1 + 1) = 10.11 MHz$$

This shows that the occupied bandwidth has been reduced by (10.11-5.69) = 4.42MHz, which denotes bandwidth improvement of 4.42/10.11 = 43.76%. Knowing that there is 6bits/symbol in 64-QAM modulation, and using Equation (19), the raw data rate is calculated as follows:

$$Data\ rate = 5057000 * 6\ Mb/s = 30.34\ Mb/s$$

Specifying the channel bandwidth at 6MHz, the bandwidth efficiency of the system can be calculated as follows, using Equation (18)

$$Bandwidth\ efficiency = (30.34/6) bps/Hz = 5.1\ bps/Hz$$

Given the ratio E_b/N_0 the value of $20\ dB$ in AWGN transmission channel, the channel capacity can be calculated using Equation (20) as follows:

$$C = \frac{\dot{\eta}}{(E_b/N_0)} = \frac{5.1}{20} = 0.26\ bits/channel \quad (23)$$

The summary of simulation and calculated results is given in Table 3.

Table 3
Summary of Simulations and Calculated Results

S/N	Parameters	Measurement values					
		$\alpha = 0$	$\alpha = 0.125$	$\alpha = 0.25$	$\alpha = 0.5$	$\alpha = 0.75$	$\alpha = 1$
1	Sampling freq.	Normalized	Normalized	Normalized	Normalized	Normalized	Normalized
2	Passband edge	0.7692	0.0673	0.0577	0.0385	0.0192	0
3	Stopband edge	0.0769	0.0865	0.0769	0.1154	0.1346	0.15385
4	Passband ripple	7.4582dB	3.0798dB	1.2011dB	0.1324dB	0.1353dB	0dB
5	Stopband Atten.	6.4576dB	10.0369dB	14.0924dB	22.221dB	25.2188dB	28.2701dB
6	3dB point	0.0680	0.0704	0.0726	0.0766	0.0782	0.0770
7	6dB point	0.0760	0.0788	0.0819	0.0886	0.0955	0.1030
8	Transition width	0	0.0192	0.0385	0.0769	0.1154	0.1539
9	Group delay	2.0	2.0	2.0	2.0	2.0	2.0
10	Filter length	53	53	53	53	53	53
14	Data rate	30.34Mbps	30.34Mbps	30.34Mbps	30.34Mbps	30.34Mbps	30.34Mbps
11	Occupied bandwidth	5.057MHz	5.689MHz	6.321MHz	7.586MHz	8.850MHz	10.114MHz
12	Bandwidth Efficiency	6.0bps/Hz	5.33bps/Hz	4.80bps/Hz	4.0bps/Hz	3.43bps/Hz	3.0bps/Hz
13	% Bandwidth improvement	50%	43.76%	37.50%	24.97%	12.50%	0.00%
16	Channel Capacity	0.26bits/channel					

VI. Conclusion

The SRRC filter was designed in Matlab 2013a and used for simulating the pulse-shaping process. The filter roll-off factor played an important role in determining the occupied bandwidth and spectral efficiency of the system. It could be seen from the impulse responses in fig. 8 that the higher the roll-off factor, the more the occupied bandwidth. Higher bandwidths are always accompanied with lower or zero time-domain ripples and vice versa. This is shown in Table 3, where the roll-off factor of 0 exhibited passband ripple of 7.4582dB, whereas the roll-off factor of 1 exhibited a passband ripple is 0dB. Conversely, stopband attenuation increases as the roll-off factor increases. Improved spectral efficiency of a communication system requires low filter roll-off factors, just a little above the Nyquist frequency point. From Table 3, it can be seen that lower roll-off factors generated higher spectral efficiencies than the higher roll-off factors. Therefore, implementing a roll-off factor of 0.125 in this project improved the spectral efficiency up to 43.76%, which is close to ideal situation. This is a great improvement in space communication spectrum as higher data rates and more channels can be used in the system without inter-symbol interference.

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