UNIVERSITY OF ZAGREB
FACULTY OF ELECTRICAL ENGINEERING AND COMPUTING

MASTER THESIS No. 2028

BASEBAND SIGNAL PROCESSING CHAIN FOR A SATELLITE DIGITAL TRANSMITTER

Mario Šimunić

Zagreb, June 2020

UNIVERSITY OF ZAGREB
FACULTY OF ELECTRICAL ENGINEERING AND COMPUTING

MASTER THESIS No. 2028

BASEBAND SIGNAL PROCESSING CHAIN FOR A SATELLITE DIGITAL TRANSMITTER

Mario Šimunić

Zagreb, June 2020

UNIVERSITY OF ZAGREB
FACULTY OF ELECTRICAL ENGINEERING AND COMPUTING

Zagreb, 13 March 2020

MASTER THESIS ASSIGNMENT No. 2028

Student:	**Mario Šimunić (2401028282)**
Study:	Electrical Engineering and Information Technology
Profile:	Electronic and Computer Engineering
Mentor:	prof. Davor Petrinović

Title: **Baseband Signal Processing Chain for a Satellite Digital Transmitter**

Description:

It is necessary to describe and simulate the digital signal processing chain for use in a digital satellite transmitter for transmission rates of up to 16 Mbps and BPSK, QPSK and O-QPSK modulations together with the transmitter filter. The carrier signal frequency is 10.45 GHz. The input to the system is a 16 bit (or less) parallel data stream. Show under what conditions, and with respect to the sampling frequency, amplitude resolution, number of filter states and memory size, unwanted emissions outside the transmission band of less than -40 dBc can be achieved. Define a digital-to-analog converter (specify and select the chip) whose output will be IQ signals representing the input of the modulator in the next stage of the transmitter. The processing chain shall be implemented in the Matlab software environment, to evaluate the computational complexity of signal processing and its feasibility on the STM32 or Zynq-7000 platform. By simulation, estimate the bit error rate for a given noise power at the receiver.

Submission date: 30 June 2020

SVEUČILIŠTE U ZAGREBU
FAKULTET ELEKTROTEHNIKE I RAČUNARSTVA

Zagreb, 13. ožujka 2020.

DIPLOMSKI ZADATAK br. 2028

Pristupnik:	**Mario Šimunić (2401028282)**
Studij:	Elektrotehnika i informacijska tehnologija
Profil:	Elektroničko i računalno inženjerstvo
Mentor:	prof. dr. sc. Davor Petrinović

Zadatak: **Lanac za obradu signala u osnovnom pojasu u satelitskom digitalnom predajniku**

Opis zadatka:

Potrebno je opisati i simulirati lanac digitalne obrade signala za primjenu u digitalnom satelitskom predajniku za brzine predaje do 16 Mbps i modulacijske postupke BPSK, QPSK i O-QPSK te odašiljački filtar. Signal nosioc je frekvencije 10.45 GHz. Ulaz u sustav je paralelni podatkovni tok širine 16 bita (ili manje). Pokazati pod kojim uvjetima, a s obzirom na frekvenciju očitavanja, amplitudnu rezoluciju, broj stanja filtra i veličinu memorije, se mogu ostvariti neželjene emisije izvan pojasa propuštanja manje od -40 dBc. Definirati digitalno analogni pretvornik (specificirati i odabrati čip) na čijem izlazu će biti IQ signali koji predstavljaju ulaz modulatora u sljedećem stupnju predajnika. Lanac je potrebno implementirati u programskom sustavu Matlab, procijeniti računalnu složenost obrade signala te izvodivost na platformi STM32 ili Zynq-7000. Simulacijom estimirati vjerojatnost pogreške bita u prijenosu (engl. bit error rate) za zadanu snagu šuma na prijemniku.

Rok za predaju rada: 30. lipnja 2020.

UNIVERSITY OF ZAGREB
FACULTY OF ELECTRICAL ENGINEERING AND COMPUTING

MASTER THESIS no. 2028

Baseband Signal processing Chain for a Satellite Digital Transmitter

Mario Šimunić

Zagreb, September 2020.

Hvala

mentoru prof. dr. sc. Davoru Petrinoviću
na iznimnom strpljenju i povjerenju

prof. dr. sc. Dubravku Babiću na podršci i
motivaciji.

Ovaj rad posvećujem voljenoj Evi
koja mi je tako mnogo pomogla.

CONTENTS

LIST OF FIGURES

List of Tables

1. Introduction

Satellites are man made objects, ie. technical systems which orbits our planet. Essentially, there are four types of orbits: high Earth orbit (HEO), medium Earth orbit (MEO), and low Earth orbit (LEO). Many weather and some communications satellites tend to have a high Earth orbit, farthest away from the planet surface. Most scientific satellites, including NASA's Earth Observing System fleet, have a low Earth orbit [4]. Over last twenty years LEO have become increasingly interesting for academic satellite projects. Hundreds of nano-satellites, called Cubesat, with weight about one kilogram and volume of one liter are launched in LEO space over last ten years.

FERSat is academic project with objective to build own nano-satellite and launch it into LEO at about 600 km altitude. Nano-satellites such as FERSat have a few subsystems and one of them is Communications subsystem. Sensor and imaging data will be transferred over a high-bandwidth digital radio-frequency (RF) link. For that purpose FERSat need to have high-bandwidth digital RF transmitter onboard.

To achieve FERSat objective of building own RF transmitter subsystem it is necessary to understand practical limits of the signal processing communications system.

First part of this Thesis focuses on defining digital signal processing chain for M-PSK modulator. Simulation of 2-PSK (BPSK), 4-PSK (QPSK) and Offset-QPSK (O-QPSK) modulator is made in Matlab. Simulations gives an answer to question what is lower limits of processing complexity under which requirements are still satisfied. Simulation results of In-phase and Quadrature signal spectrum for a few different transfer rates will be compared. Second part of Thesis focuses on effects of limited digital word length, ie. limited memory register width.

Next chapters gives phase modulation and pulse transmission theory overview, disseminated modulator requirements and constraints, chosen modulator arhitecture, simulation explaination and results comparison.

2. Digital Communications Theory Overview

In digital communications information is represented using binary coded words. This means that information or data in a computer is represented as discrete states of binary bits stored in memory registers. In digital computers, time and signals are discrete. Signals which are discrete in time and amplitude are called digital signals. During transmission of information trough communication channel, information need to be represented with kind of signals suited for particular communication channel.
The need to transfer information from eg. satellite to earth station computer basically involves series of data transformation. First of all, communication channel is analog RF signal, which means continuous-amplitude and continuous-time signal. In order to transmit digital signals, representing data, trough analog communications channel they need to be converted into continuous time signals. During transmission it is necessary to conform regulations. Radio-frequency bandwidth is very constrained and limited good. One of the most important parameters of RF transmitter is RF bandwidth used for transmission.

2.1. Continuous and discrete time signals

Continuous-time signal $u(t)$ can be represented with it's spectrum $U(\omega)$. Connection between continuous-time signal and it's spectrum is given by Continuous-Time Fourier Transform (CTFT) [5].

$$\mathrm{CTFT}\big[u(t)\big] = U(\omega) = \int_{-\infty}^{+\infty} u(t)e^{-j\omega t}\,dt \tag{2.1}$$

Inverse Continuous-time Fourier Transformation is given by:

$$\mathrm{ICTFT}\big[U(\omega)\big] = u(t) = \frac{1}{2\pi}\int_{-\infty}^{+\infty} U(\omega)e^{j\omega t}\,d\omega \tag{2.2}$$

Digital signal $u[n]$, a series of data points describing time evolution of a real world phenomena, is obtain by taking samples of continuous signal at equidistant time points T_s. Sampling, a process of taking samples, is a discretization in time. After sampling, discrete time signals are quantized, that is discretized in amplitude, to B bits. B is word length of analog to digital converter.

Nyquist-Shannon sampling theorem establishes a sufficient condition for a sample rate that permits a discrete sequence of samples to capture all the information from a continuous-time signal of finite bandwidth. C. E. Shannon made sampling theorem in it's famous work [6] as Theorem 13. It states that if sampling of a bandlimited signal is done with sampling frequency $f_s \geq 2f_M$ where f_M is a maximum frequency component of the signal, then signal can be perfectly reconstructed. Sampling is written as in Eq. (2.3) [7, Eq.2.6] where $x^*(t)$ is sampled function, $f_s = \frac{1}{T_s}$ is a sampling frequency and $\delta(t)$ is a Dirac delta function.

$$x^*(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\delta(t - nT_s) \tag{2.3}$$

Spectrum of sampled function x^* is given in Eq. (2.4).

$$X^*(\omega) = \frac{1}{T_s} \sum_{k=-\infty}^{\infty} X(\omega - k\omega_s) \tag{2.4}$$

It can be seen that sampled faction spectrum X^* have original spectrum $X(\omega)$ scaled by $\frac{1}{T_s}$ and periodized with frequency period ω_s. If Nyquist-Shannon sampling theorem is satisfied there is no overlapping between replicas of $X(\omega)$. Discrete time signal is $x_d[n] = x(nT_s)$

Discrete-time signal $u[n]$ can be represented with it's spectrum $U(\omega)$. Connection between discrete-time signal and it's spectrum is given by Discrete-Time Fourier Transform (DTFT) [5].

$$\mathrm{DTFT}\big[u(t)\big] = U(\omega) = \sum_{n=-\infty}^{+\infty} u[n]e^{-j\omega n} \tag{2.5}$$

Inverse Discrete-time Fourier Transformation is given by:

$$\mathrm{IDTFT}\big[U(\omega)\big] = u[n] = \frac{1}{2\pi} \int_{-\pi}^{+\pi} U(\omega)e^{j\omega n}\, d\omega \tag{2.6}$$

DTFT and IDTFT operates at infinite series signals. Infinite series signal practically

isn't achievable. In reality, computers work with limited amount of memory and limited number of signal samples. Thus, we must use different tool for transformations of finite-length discrete-time signals. This is Discrete Fourier Transform and Inverse Discrete Fourier Transform Eq. (2.7), [5].

$$\mathrm{DFT_N}\big[u[n]\big] = U[k] = \sum_{n=0}^{N-1} u[n] W_N^{nk}, 0 \leq k \leq N-1 \tag{2.7}$$

Inverse Discrete-time Fourier Transformation is given by:

$$\mathrm{IDFT_N}\big[U[k]\big] = u[n] = \frac{1}{N} \sum_{n=0}^{N-1} X[k] W_N^{-nk}, 0 \leq n \leq N-1 \tag{2.8}$$

$$W_N^{nk} = e^{-j\frac{2\pi nk}{N}} \tag{2.9}$$

Fast Fourier Transform is a group of algorithms for efficient calculating DFT transform. It's is important to notice few details about FFT and DFT to understand signal spectrum in simulations. First, discrete frequency have range from $-\pi$ to π. Whole frequency range $[-\frac{f_s}{2}, \frac{f_s}{2}]$ of continuous-time signal is mapped into range $[-\pi, \pi]$.
DFT assumes periodic signal, ie. it assumes that finite-length signal $u[n]$ is exactly one period of a infinite-length periodic signal. If this isn't true, spectrum isn't perfectly clean.
DFT can be observed as array of matched filters, each of which is matched to k-th frequency. Input signal excites more or less each of the filter, and each filter output is proportional to how much signal frequency is matching that filters frequency.
Using finite-length sequence of infinite-lenth signal is a windowing operation. FFT by default operates with rectangular window. Windowing the signal produces amplitude and frequency effects in calculated spectrum of the signal. Rectangular window with amplitude of A and width T have frequency spectrum $X(\omega) = 2ATsinc(\frac{\omega T}{\pi})$. Thus multiplication of infinite-length signal withwindow in time domain produces convolution of signal spectrum with window spectrum in frequency domain. Eq. (2.10) is modified DFT equation which includes time window samples $w[n]$.

$$U[k] = \sum_{n=0}^{N-1} w[n] u[n] W_N^{nk} \tag{2.10}$$

From Eq. (2.10) it can be seen that amplitude of each spectral component k calculated by DFT is increased by a sum of window samples $w[n]$. To retain physical meaning of

spectrum amplitude, FFT need to be scaled by a sum of window samples. Corrected FFT transform is shown in Eq. (2.11).

$$U[k] = \frac{1}{\sum_{n=0}^{N-1} w[n]} \sum_{n=0}^{N-1} w[n]u[n]W_N^{nk} \tag{2.11}$$

There are three discrete-time signals for which it is important to know their spectra. Those signals are rectangular impulse, $\delta[n]$ (single non-zero sample) and sine wave signal.

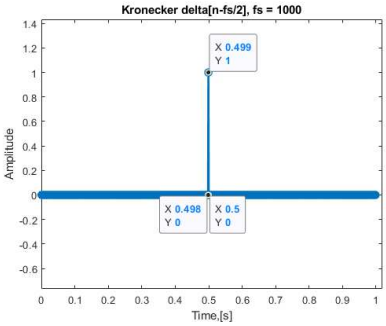


Figure 2.1: $\delta[n - \frac{f_s}{2}]$ signal

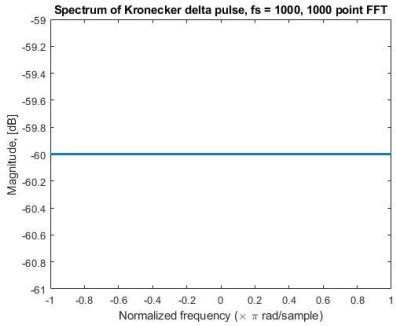


Figure 2.2: Spectrum of $\delta[n - \frac{f_s}{2}]$ signal

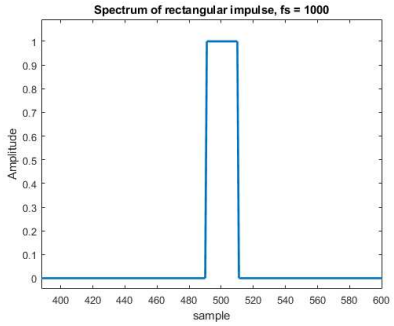


Figure 2.3: Single rectangular pulse signal

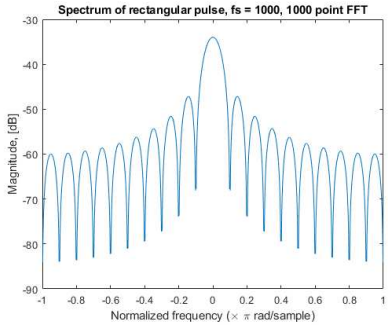


Figure 2.4: Spectrum of rectangular pulse

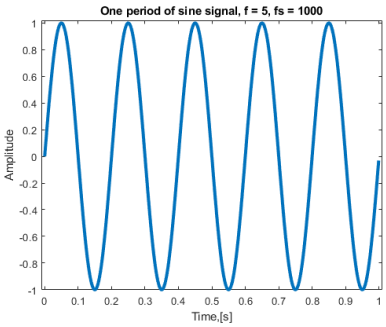


Figure 2.5: Sine wave signal

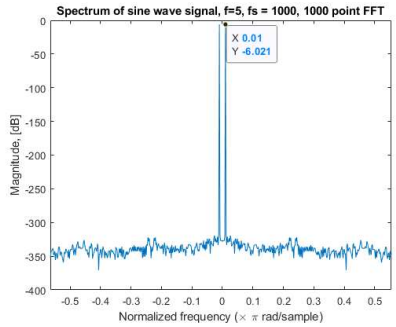


Figure 2.6: Spectrum of sine wave signal

2.1.1. Modulation and complex envelope of bandpass signal

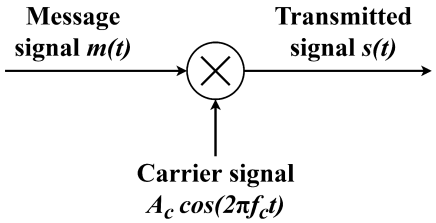


Figure 2.7: Block diagram of modulation

Modulation represented with block diagram Fig. 2.7 is a procedure in which modulation signal $m(t)$ is mixed with carrier signal. Modulation or mixing is mathematically multiplication. Modulation signal is carrying information or message. Carrier signal is RF signal tuned for RF channel frequency. Multiplication of carrier signal with $m(t)$ produces signal $s(t)$ which contains spectrum of $m(t)$ centered at carrier frequency, thus whole information signal spectrum is shifted into RF channel band.
Modulation can affect carrier signal frequency, amplitude or phase. Modulation formats which uses phase of carrier signal as information carrier are called phase modulations. Keying is a modulation format in which message signal is pulse coded, for example Amplitude Shift Keying in which pulse can have amplitude of 0 or 1. These pulses are multiplied with carrier signal to produce ASK modulated signal. A passband is the range of frequencies that can pass through a filter. For example, a radio receiver contains a bandpass filter to select the frequency of the desired radio signal out of all the radio waves picked up by its antenna. The passband of a receiver is the range of frequencies it can receive when it is tuned into the desired frequency. A bandpass-filtered signal (that is, a signal with energy only in a passband), is known as a bandpass signal, in contrast to a baseband signal [8]. Baseband is a signal that has a near-zero frequency range. In telecommunications and signal processing, baseband signals are transmitted without modulation, that is, without any shift in the range of frequencies of the signal [9]. Signal $m(t)$ is basically baseband signal.

Direct Conversion modulation is one of the technique which gives modulated signal using complex mixing or complex modulation. Fig. 2.8 represents basic block diagram of direct conversion modulator. Signal $u_{LP}(t)$ is a complex signal consisting of two signals. Real part of $u_{LP}(t)$ is signal $u_{LP\_I}(t)$ or In-phase signal. Imaginary part of $u_{LP}(t)$ signal is $u_{LP\_Q}(t)$ or Quadrature signal.

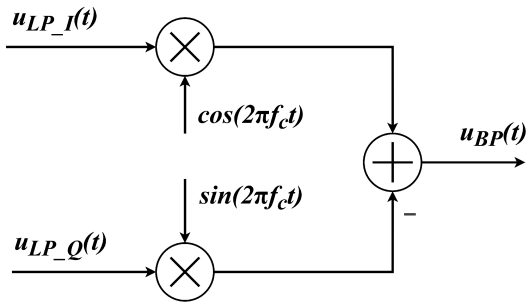


Figure 2.8: Basic block diagram of direct conversion modulator

$$u_{BP}(t) = Re[u_{LP}(t) \cdot e^(j\omega_0 t)] \tag{2.12}$$

$$u_{BP}(t) = Re[u_{LP\_I}(t) + ju_{LP\_Q}(t)] \cdot [(cos(\omega_0 t) + jsin(\omega_0 t)] \tag{2.13}$$

$$\boldsymbol{u_{BP}(t) = u_{LP\_I}(t)cos(\omega_0 t) - u_{LP\_Q}(t)sin(\omega_0 t)} \tag{2.14}$$

Equation Eq. (2.14) is showing that any modulated signal can be achieved by using adequate I and Q signals mixed with complex exponential, that is cosine and sine signals with carrier frequency. Summation of I branch signal with negative Q branch signal results in modulated bandpass signal. Signal $u_{LP}(t) = u_{LP\_I}(t) + ju_{LP\_Q}(t)$ is a complex envelope of signal $u_{LP}(t)e^{j\omega_0 t}$.

2.2. Digital modulation formats

Among many keying formats this Thesis focuses on M-ary Phase Shift Keying (M-PSK). M in M-PSK represents order of modulation. For M=2 modulation is 2-PSK or binary PSK (BPSK), and for M=4 modulation is Quadriphase PSK or QPSK.
Modulation states are represented with symbols instead of bits and symbols are actual information, that is signals which are modulated on carrier signal. Depending on order of modulation, one or more bits are packed into one symbol. This is done by mapping bits into symbols and representing symbols using vector space signals. Parameter $K = log_2(M)$ defines how much message bits is packed into one M-ary modulation symbol. For example, 8-PSK, with M=8 transmits or modulates K=3 bits per symbol.

Assigning signals to symbols.
Functions $\phi_j(t)$ form the orthonormal basis of vector space. Signals $s_i(t)$ can be shown

as linear combination of $\phi_j(t), j = 1, ..., M$.

$$s_i(t) = \sum_{j=1}^{N} s_{ij}\phi_j(t), 0 \leq t \leq T_b \tag{2.15}$$

Equation Eq. (2.15) for $N = 2$ resembles the direct conversion modulator in which basis functions are $\phi_1 = cos(\omega_0 t)$ and $\phi_2 = sin(\omega_0 t)$. Signals representing symbols are $s_{i1} = u_{LP\_I}$ and $s_{i2} = u_{LP\_Q}$.
Basis functions are chosen using equations Eq. (2.16), Eq. (2.15), and the facts that basis functions must be orthogonal and signals $s_i(t)$ are linear combination of products of multiplication basis functions with coefficients s_{ij}.

$$\phi_i(t) = \frac{g_i(t)}{\sqrt{\int_0^{T_b} g_i^2(t)dt}} \tag{2.16}$$

2.2.1. BPSK modulation

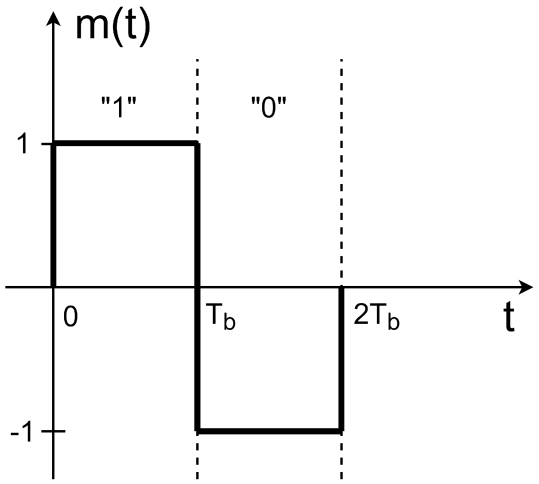


Figure 2.9: Message signal pulses representing binary "1" and "0"

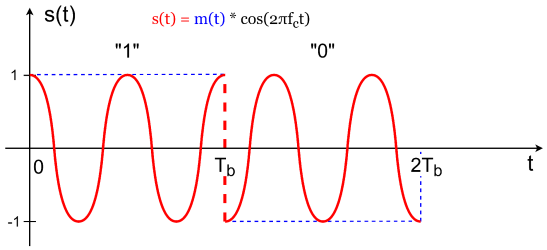


Figure 2.10: BPSK - modulated symbols "1" and "0"

Coherent BPSK is modulation with $M = 2$, thus two signals are used, s_1 and s_2. Signals are given with the equations Eq. (2.17) and Eq. (2.18).

$$T_b = \frac{1}{SR}$$

T_b is time duration of the symbol and SR is symbol rate at which symbols are transmitted.

$$s_1 = \sqrt{\frac{2E_b}{T_b}} cos(2\pi f_0 t) \tag{2.17}$$

$$s_2 = \sqrt{\frac{2E_b}{T_b}} cos(2\pi f_0 t + \pi) = -\sqrt{\frac{2E_b}{T_b}} cos(2\pi f_0 t) \tag{2.18}$$

BPSK have one basis function $\phi_1(t)$ thus signal vector space is one-dimensional.

$$\phi_1(t) = \sqrt{\frac{2}{T_b}} cos(2\pi f_0 t) \tag{2.19}$$

Basis function amplitude $\sqrt{\frac{2}{T_b}}$ ensures that basis is orthonormal. ˜

Vector which describes first signal:

$$s_1 = s_{11} = \int_0^{T_b} s_1(t)\phi_1(t)dt = \sqrt{E_b}$$

Vector which describes second signal:

$$s_2 = s_{21} = \int_0^{T_b} s_2(t)\phi_1(t)dt = -\sqrt{E_b}$$

Signal space or constellation of BPSK modulation is shown in Fig. 2.11. Probability

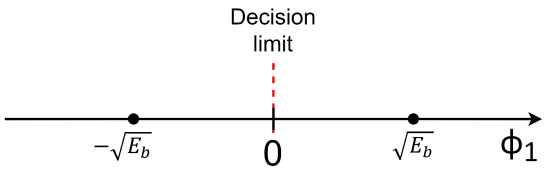


Figure 2.11: BPSK signal space

of transfer error for Additive White Gaussian Noise (AWGN) communication channel:

$$P_{e,BPSK} = \frac{1}{2} erfc(\sqrt{\frac{E_s}{N_0}})$$

where E_s is symbol energy and N_0 is channel noise energy. BPSK bit to symbol mapping maps one bit of message into one symbol, that is number of bits $K_{BPSK} = 1$. In case of BPSK modulataion symbol energy and symbol duration is same as bit energy and bit duration, $E_b = E_s, T_b = T_s$.

2.2.2. QPSK modulation

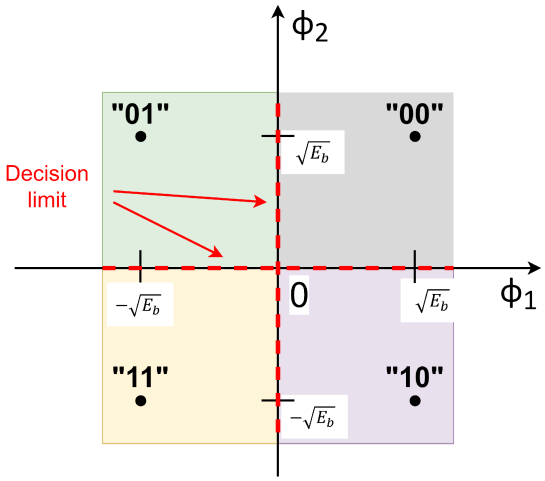


Figure 2.12: QPSK signal space

Coherent QPSK is modulation with $M = 4$, thus four signals are used, s_1, s_2, s_3, and s_4. Signals are given with the equations Eq. (2.20) - Eq. (2.23).

$$s_1 = \sqrt{\frac{2E_s}{T_s}} cos(2\pi f_0 t + \frac{1\pi}{4}) \tag{2.20}$$

$$s_2 = \sqrt{\frac{2E_s}{T_s}} cos(2\pi f_0 t + \frac{3\pi}{4}) \tag{2.21}$$

$$s_3 = \sqrt{\frac{2E_s}{T_s}} cos(2\pi f_0 t + \frac{5\pi}{4}) \tag{2.22}$$

$$s_4 = \sqrt{\frac{2E_s}{T_s}} cos(2\pi f_0 t + \frac{7\pi}{4}) \tag{2.23}$$

$$T_s = \frac{1}{SR}$$

T_s is time duration of the symbol and SR is symbol rate at which symbols are transmitted.

QPSK have two basis function $\phi_1(t)$ and $\phi_2(t)$ thus signal vector space is two-dimensional.

$$\phi_1(t) = \sqrt{\frac{2}{T_s}} cos(2\pi f_0 t) \tag{2.24}$$

$$\phi_2(t) = \sqrt{\frac{2}{T_s}} sin(2\pi f_0 t) \tag{2.25}$$

Signal space is shown in Fig. 2.12. Adjacent symbols have Hamming distance of 1, which means that the are different in 1 bit. Different constellation mappings are possible It is beneficial for reducing transfer error probability that signals which are close have small Hamming distance.

BPSK and QPSK have same probability of transfer error, P_e. This is because QPSK modulation uses one bit of message to modulate In-Phase signal and second bit of message to modulate Quadrature signal. These bits, and these quadrature component modulations are independent. For that reason QPSK is basically two independent BPSK modulations thus P_e is the same for both modulations. QPSK modulation sends two bits of message at the same time and uses twice the energy of BPSK to mentain same Signal To Noise ratio.

Rectangular signals $u_{LP\_I}(t)$ and $u_{LP\_Q}(t)$ from Fig. 2.8 represents coordinates in signal space, signal $u_{BP}(t)$ is a QPSK modulated modulated signal.

2.2.3. Offset QPSK (O-QPSK) modulation

QPSK modulation modulates In-Phase and Quadrature signals with message bits at the same instant of time. Symbol transition from one symbol to another passes trough IQ plane origin, that is zero amplitude. This variation of amplitude produces unwanted effectes in RF power amplifier and reduces amplifier efficiency.
O-QPSK is a variant of QPSK which modulates In-Pgase and Quadrature signals with time delay, or offset, of half a symbol duration, $\frac{T_s}{2}$. This procedure eliminates symbol transitions trough zero amplitude.

2.3. Nyquist filter and pulse shaping

Every real communication channel is band-limited. For example, FERSat expected channel bandwidth is 5 to 10 MHz. If we consider mixer input signals u_{LP_I} and u_{LP_Q} as rectangular pulses, for example as in Fig. 2.3 than magnitude spectrum of that signals has shape as in Fig. 2.4. This spectrum is unlimited with side lobes which

decays slowly (sinc function). Besides bandwidth issues with rectangular pulses, impulse response of the channel causes a transmitted symbol to be spread in time domain. This causes Inter Symbol Interference (ISI) where the previously transmitted symbols affects the currently received symbol, thus reducing tolerance for noise.

H. Nyquist came up with criterion for distortionless transmission which states that height (amplitude) of the middle of each signal element (symbol) should be undistorted. This means that each pulse $p(t)$ should be zero-valued in every sampling time instance T_s except for time instance $t = 0$, that is Nyquist criterion for zero-ISI:

$$p(t) = \begin{cases} 1; & t = 0 \\ 0; & t = nT_s \ (n \neq 0) \end{cases} \tag{2.26}$$

and

$$\sum_{n=-\infty}^{\infty} P\left(f - \frac{n}{T_s}\right) = T_s \tag{2.27}$$

A minimum bandwidth system has a rectangular shape from $-\frac{1}{2}T$ to $\frac{1}{2}T$. Where $\frac{1}{2}T$ is Nyquist frequency. Any filter that has excess bandwidth with odd symmetry around Nyquist frequency and even symmetry around zero frequency also satisfies zero-ISI criterion. Any such filter is called Nyquist filter.

Brick-wall filter or sync filter is ideal Nyquist filter. Frequency response and impulse response of the filter are illustrated in Fig. 2.13 and Fig. 2.14. Impulse response of

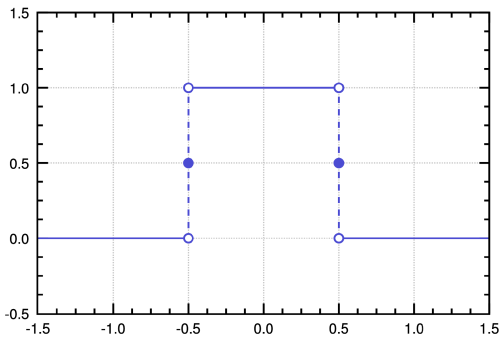


Figure 2.13: Rectangular function - frequency response of a brick-wall filter

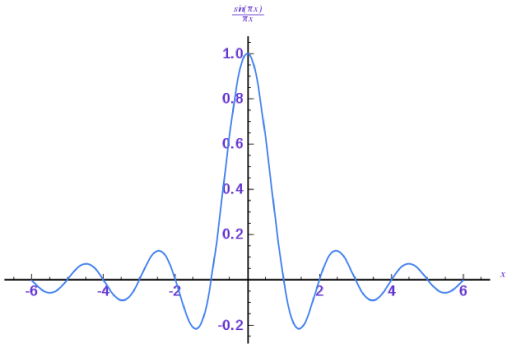


Figure 2.14: Graph of the normalized sinc function - impulse response of a brick wall filter

brick-wall filter is a Sinc function. It is slowly decaying signals which lasts from $t = -\infty$ to ∞ and because it decays slowly, time shifting and time-limiting such signal produces significant errors in spectrum. Therefore, it is practically unachievable. For

that reason we would like to use filter with different frequency response shape and fast decaying impulse response which could be approximated.

Raised Cosine filter is popular Nyquist filter. It's pulse has signal bandwidth $(\frac{1}{2}T_s)(1+\beta)$. Frequency response of Raised Cosine filter [10] is given in Eq. (2.28) and plotted in Fig. 2.15.

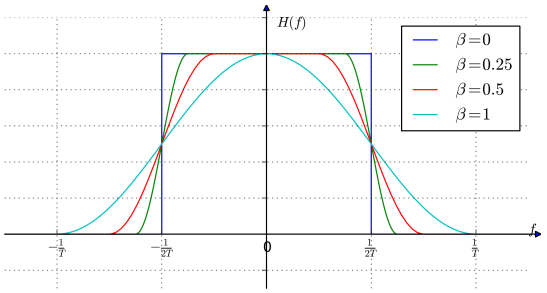


Figure 2.15: Frequency response of raised-cosine filter with various roll-off factors β [1]

$$H(f) = \begin{cases} 1, & |f| \leq \frac{1-\beta}{2T} \\ \frac{1}{2}\left[1+\cos\left(\frac{\pi T}{\beta}\left[|f|-\frac{1-\beta}{2T}\right]\right)\right], & \frac{1-\beta}{2T} < |f| \leq \frac{1+\beta}{2T} \\ 0, & \text{otherwise} \end{cases} \tag{2.28}$$

For each impulse at filter input filter output is one impulse response. This procedure is called pulse shaping. Using Raised Cosine (RC) shaped pulses, zero-ISI is achievable. Impulse response of RC filter decays quickly than Sinc pulses, and could be much better approximated. This type of filter is most widely in use as Nyquist transmit filter. Impulse response of Raised Cosine filter is given in Eq. (2.29) and plotted in Fig. 2.16 for various roll-off factors. There is an trade-off between spectral bandwidth and decaying slope.

$$h(t) = \begin{cases} \frac{\pi}{4T}\,\mathrm{sinc}\left(\frac{1}{2\beta}\right), & t = \pm\frac{T}{2\beta} \\ \frac{1}{T}\,\mathrm{sinc}\left(\frac{t}{T}\right)\frac{\cos\left(\frac{\pi\beta t}{T}\right)}{1-\left(\frac{2\beta t}{T}\right)^2}, & \text{otherwise} \end{cases} \tag{2.29}$$

2.4. Peak to Average Power Ratio (PAPR)

Theoretical modulation use rectangular impulses as symbols. Perfect rectangular impulses have instantaneous level transitions and maximally sharp edges. Using theoretical modulation, modulated signal wave would have shape as in Fig. 2.10. If we

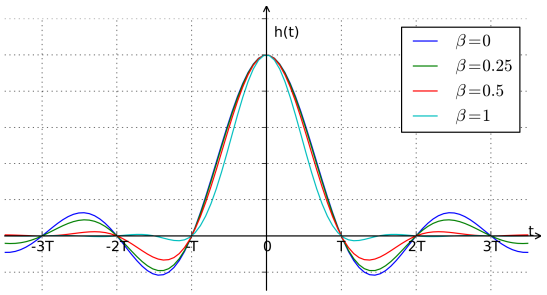


Figure 2.16: Impulse response of raised-cosine filter with various roll-off factors β [1]

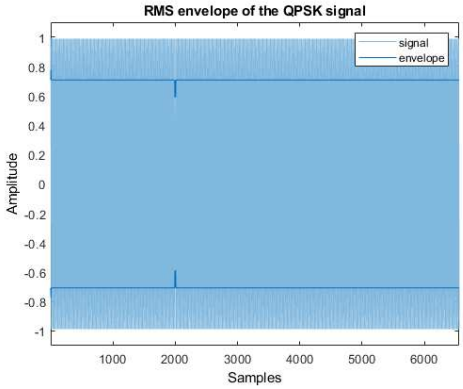


Figure 2.17: Envelope of QPSK modulation with ideal rectangular pulses

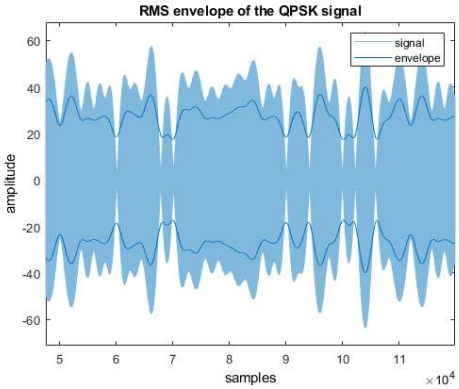


Figure 2.18: Envelope of QPSK modulation with RC shaped pulses

plot envelope of that signal Fig. 2.17, it can be seen that peak power to rms or average power is constant. When RC filtered pulses are used as symbols then symbol transition are not instantaneous. Sum of impulses produces envelope with peeks and drops. In that case PAPR is lower than ideal. Such transients behavior of signal envelope is a problem for linear RF power amplifier [11] because amplifier must have enough dynamic range to convey envelope peaks, but most of the time it operates at power lower than optimal. Envelope with peaks and drops produces unwanted nonlinear effects in the amplifier and reduces power efficiency of linear RF amplifier. Power amplifier efficiency is a trade-off for spectral efficiency of higher-order modulation formats. O-QPSK modulation format is variant of Q-PSK in which delay of $\frac{T_s}{2}$ between I and Q transitions produces modulated signal envelope which never drops to zero, that is signal transition in signal space never pass trough origin and envelope is more flat than compared with ordinary QPSK modulated envelope.

3. Proposed system architecture

3.1. System requirements

The goal is to have satellite communicating in Amateur-Radio X-band 10.450 - 10.500 MHz with dedicated channel bandwidth of 10 MHz. For that type of RF communication channel, International Telecommunication Union documentation was studied under FERSat project. Resulting maximum allowed spectral emissions are illustrated on the spectral mask given in Appendix A. This thesis is focused on more stringent spectral requirement of -40 dBc.

Analog part of transmitter is already defined. Mixer that will be used is Analog Devices's HMC1056LP4BE. This mixer receives analog IF input signals (In-phase and Quadrature component) in frequency range from DC up to 4 GHz. It's reasonable to have stringent requirements for digital processing part until transmitter prototype is field tested because of unknown parasitic parameters in analog part of transmitter, which can affect resulting unwanted spectral emissions.

System should transmit using one or all of modulation formats: BPSK, QPSK nad O-QPSK. Line transmition rate need to be up to 16 Mbit per second.

3.2. Digital modulator architecture

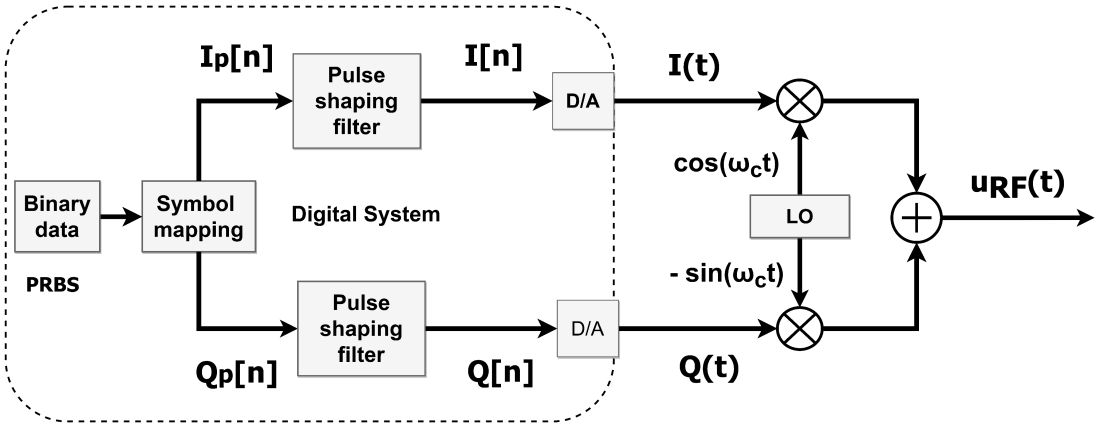


Figure 3.1: Block diagram of digital system for baseband signal processing

Fig. 3.1 illustrates proposed Direct Frequency Conversion mixer with complex mixing. Modulator is consisting of symbol mapper, transmit filters and digital to analog converters (DAC). Output of DACs are two continuous-time analog signals, $I(t)$ and $Q(t)$ which together makes complex envelope signal.
Pseudo-random Bit-Stream (PRBS) is not part of modulator. PRBS is used for testing and simulation purposes, which will be explained in the next chapter.

4. Matlab simulation

4.1. Pseudo-random Bit-Stream generator

PRBS generator is based on Linear Feedback Shift Register architecture. Basically it is a shift register with some flip-flop outputs (taps) connected to XOR gate. Output of the XOR gate is connected back to the input of the first flip-flop in the shift register [2]. Table in Fig. 4.2 contains listed polynomials for maximum length LFSR counters. For example, if PRBS-9 is used then potentions of x in the polynomial for the PRBS-9 defines which taps of LFSR are connected to XOR gate. Output of the PRBS is a sequence of random bits, that is sequence of the output from the last flip-flop in the LFSR. During first steps of writing simulation algorithm, it would be beneficial if

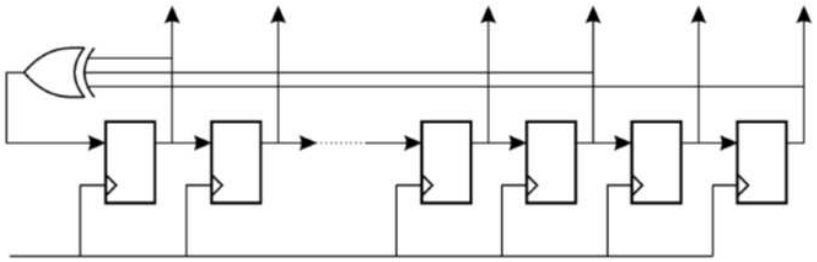


Figure 4.1: LFSR architecture [2]

resulting FFT spectrum of complex envelope would be clean and flat as possible. DFT expects that input sequence contains exactly k periods in the periodic signal, where $k \in \mathbb{N}$. To minimize spectral estimation errors due to non-integer number of periods, periodic sequence of data is used. PRBS generator of order n generates random bits with repetition cycle of $2^n - 1$ bits. In other word, PRBS-n is cyclic in $2^n - 1$ bits. This bits are mapped into symbols, thus period sequence of bits becomes periodic sequence of symbols. Because every symbol is represented with L samples, if FFT is calculated on a sequence of length $2^n - 1$ samples it will always be periodic sequence which will produce clean spectrum. This way it's easier to spot changes in spectrum

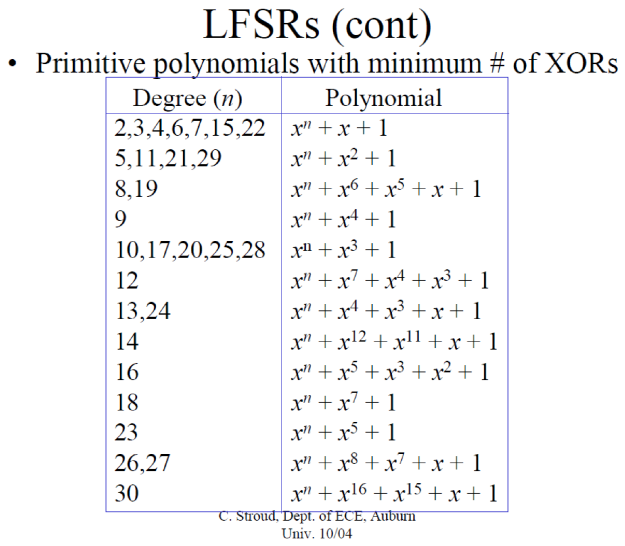


Figure 4.2: LFSR polynomials [3]

due to simulated system parameter change. The concept of creating periodic sequence illustrated in Fig. 4.3. Depending on modulation type, more than one bit is packed into symbol and PRBS-n doesn't produce even-lenght sequence. If just one period of PRBS is used as sequence of data bits then last symbol would not be complete for modulation with order M > 2. This would break the periodicity of FFT sequence. Solution is to use at least two periods of PRBS as data sequence. Then, it is possible calculate FFT on a sequence of symbols with length equal to PRBS period. This sequence can be shifted left and right for any number of symbols. FFT sequence will be periodic, no mather the shift. This shift is used to avoid calculating FFT from zero-valued samples which exist at the beginning of simulation due to filter delay.

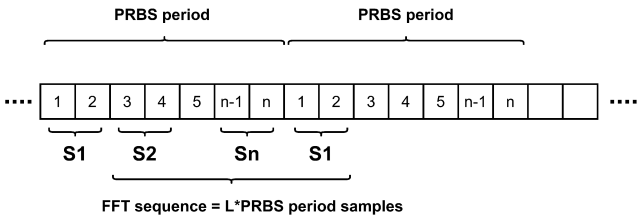


Figure 4.3: PRBS and sequence periodicity [1]

4.2. PRBS program implementation

PRBS generator is called in simulation by calling function
[symbol, nextseed] = f_prbs(seed, bits).

Function will determine order of PRBS-n from length of the parameter *seed*. Second paramter *bits* defines how many bits function will return as value of *symbol*. Next block of code will generate periodic pseudo-random message bits stored in variable *data* .

```
K = log2(M);                % number of bits per symbol
% ...
seed = '100000000';
bits = length(seed);
prbs_period = int16(2^bits)-1;     % LFSR RNG periodicity
sequence_period = lcm(int16(K), prbs_period);   % symbols to
    repetition: symbols for cyclical sequence
m = 2*sequence_period;  % need to have enough data for cyclic fft
    capture

for k=1:m
[symbol,seed] = f_prbs(seed,K);
switch symbol
%QPSK
case '00'
data = [data,0,0];
case '01'
data = [data,0,1];
case '10'
data = [data,1,0];
case '11'
data = [data,1,1];
%BPSK
case '0'
data = [data,0];
case '1'
data = [data,1];

otherwise
warning('Unsupported symbol type.');
end
end
```

4.3. Symbol mapping

Simbol mapper is implemented inside function *f_mappsym_ms*. Functions is called from main simulation script *mpsk_sim_ms.m* within next block of code:

```
% ############################################################
% SYMBOL MAPPING
% ############################################################

[symbols, I, Q, SYMmapped] = f_mappsym_ms(modulation, data);
```

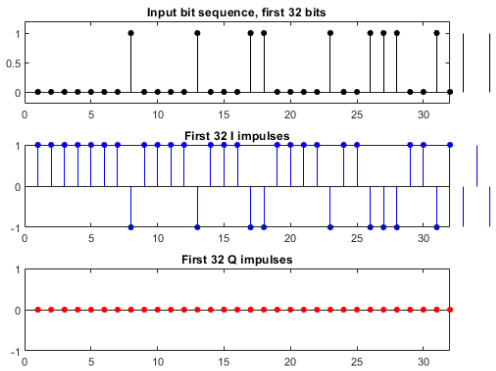


Figure 4.4: BPSK maped symbol pulses

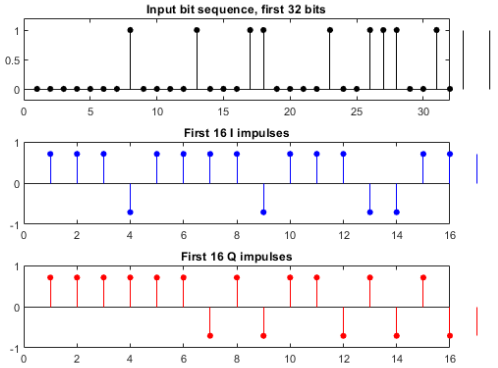


Figure 4.5: QPSK and O-QPSK maped symbol pulses

Symbol mapping is implemented in function
[symbols, I, Q, SYMmapped] = f_mappsym_ms (modulation, bitstream).
Parameter *modulation* defines modulation format (BPSK, QPSK, O-QPSK). Function reads sets of K bits from input message (array of bits) and maps them into output vectors I and Q which contains impulses (one sample is one impulse) representing coordinates of symbols in signal space. Every k-th symbol $S(k)$ have coordinates in vector space $S(k) = (I(k), Q(k))$. Output array $SYMmapped$ is array of k complex numbers where each complex number is representing one symbol $S(k)$ where $Re\big[S(k)\big] = I(k)$ and $Im\big[S(k)\big] = Q(k)$. Function also returns array $symbols$ which is array of numbers (bits) with columns length of K numbers. Each column is representing K bits, that is, one symbol. Fig. 4.4 and Fig. 4.5 shows first 32 bits of message sequence and corresponding mapped symbol pulses. The next block of code is part of function f_mapsym_ms.m. Constellation lookup tables and I, Q mapping is listed.

```
%% Constellations (Look-up Tables)
% vector-row format!!!

```

```
% BPSK constellation
constell_BPSK =  [ 1 , ...   % 0
-1 ];      % 1

% QPSK contellation, Gray code, Hamming distance = 1
constell_QPSK =  [ exp( 1i *   pi/4 ), ...   % 00 = 0d
exp( 1i * 3*pi/4 ), ...   % 01 = 1d
exp( 1i * 7*pi/4 ), ...   % 10 = 3d
exp( 1i * 5*pi/4 ) ];     % 11 = 2d

% Other constellations

% ...
% My own reshape and mapping
SYMmapped = zeros(1,len/K);          % memory allocation for mapped
    symbols
symbols = zeros(K,len/K);            % symbols, binary
symbol = zeros(1,K);                 % one symbol, binary
addr = zeros(1,len/K);               % memory allocation for symbol
    addressing
col = 0;
idx = 0;
for ii = 1: K : len
  col = col +1;
  idx = idx+1;
  for kk = 0 : K-1
    row = kk+1;
    symbols(row,col) = bitstream(ii+kk);    % All symbols
    symbol(kk+1) = bitstream(ii+kk);        % One symbol
    addr(idx) = addr(idx) + ((2^(K-1-kk)) * bitstream(ii+kk));
    SYMmapped(idx) = constell (addr(idx)+1);
  end
end

% ...
I = real(SYMmapped);
Q = imag(SYMmapped);
```

4.4. Raised Cosine filter

Raised Cosine filter is a family of Nyquist filters with frequency response shape of raised cosine function with flat top. It is used for zero-ISI transmission.
Filter is designed using function h = f_raisedcosine_ms(alpha, span, sps, shape, implementation) as in the next code block:

```
% h_rc is a impulse response coeficients of FIR nyquist filter
% h = f\_raisedcosine\_ms( alpha, span, sps, shape, implementation )

h_rc = f_raisedcosine_ms( rolloff, RCF_span, L, RCF_shape, 'matlab')
    ;
%
    ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
```

Function receives parameter *alpha* which is frequency rolloff in range 0 to 1. In this simulation filter is designed with rolloff factor value of 0.22. Second parameter is *span*, that is the duration or length of designed filter impulse response in number of symbols. Third parameter, *sps*, is the number of samples per symbol or symbol sampling rate. Fourth parameter, *shape* can have values of *'normal'* or *'sqrt'* and it defines shape of frequency response to be normal Raised-Cosine filter or Square-Root Raised Cosine (SRRC) filter. The last parameter *implementation* is used to define how filter coeficients will be designed - using My own code or using Matlab function rcosdesign. Filter for this simulation is designed using Matlab function rcosdesign. Basically, f_rcosdesign_ms is a wrap-around Matlab function for some special cases. Fig. 4.6 shows impulse response samples and Fig. 4.7 shows magnitude response of the designed RC filter using parameters:
alpha = 0.22, span = 128, L = 8, shape = 'normal', implementation = 'matlab'.

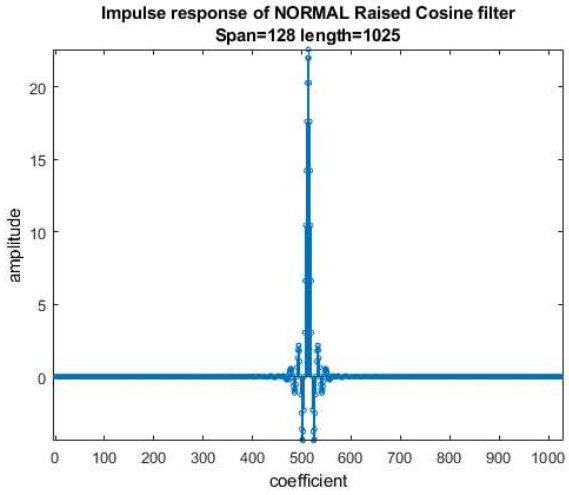


Figure 4.6: Impulse response samples of the designed RC filter

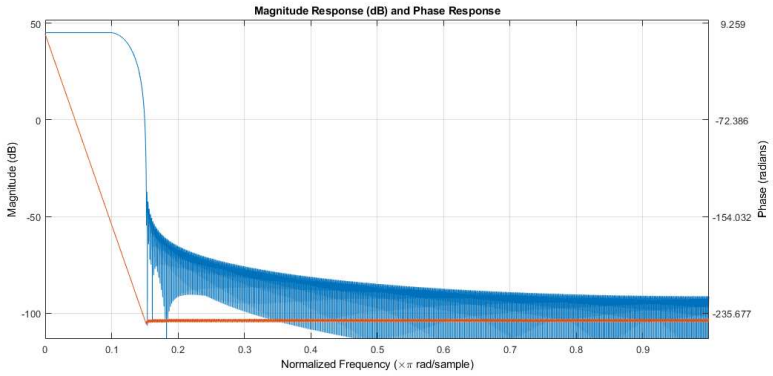


Figure 4.7: Magnitude and Phase response of the designed RC filter

4.5. Pulse shaping

Pulse shaping is a procedure in which pulse are 'shaped' by the Nyquist filter, that is RC filter. Each pulse is a impulse response of the filter caused by input stimulus. In order to produce correct impulse response, input stimulus cannot be mapped impulses but the need to be upsampled for a factor of 2, at least. Basically, filter need enough samples (discrete time) to produce response and enough samples containing information which describe each pulse. Upsampling by factor L is done by inserting $L-1$ zeros between original impulses. Next block of code produces upsampled I and Q signals (I_up and Q_up):

```
% ### UPSAMPLE
I_up = upsample(I, L);
Q_up = upsample(Q, L);
```

Pulse shaping is a filtering process and it is done using Matlab function filter by calling:

```
% Signals are concatenated with zeros at the end to allow filter
    response
% to decay
I_up = [I_up, zeros(1,L*(RCF_span/2))];
Q_up = [Q_up, zeros(1,L*(RCF_span/2))];

I_f_state = zeros(1,length(h_rc)-1);    % I filter state variable
Q_f_state = zeros(1,length(h_rc)-1);    % Q filter state variable

[I_filt, I_f_state] = filter(h_rc, 1, I_up, I_f_state);     % In-
    phase branch filtration
[Q_filt, Q_f_state] = filter(h_rc, 1, Q_up, Q_f_state);     %
    Qadrature branch filtration
```

Impulse response to each impulse is affected by impulse response of previous $\frac{span}{2}$ impulse responses. For this reason filter state is keept in a _state variable. At the beginning state variable is empty (all state are zero). Output signal delay is half the filter span. This is why the signal is concatenated with zeros at the end, to allow the filter to settle down after all data symbols are filtered.

Resulting upsampled signals for BPSK and QPSK are shown in Fig. 4.8 and Fig. 4.9. When using O-QPSK modulation then phase (time) delay of $\frac{T_s}{2}$ is inserted between I

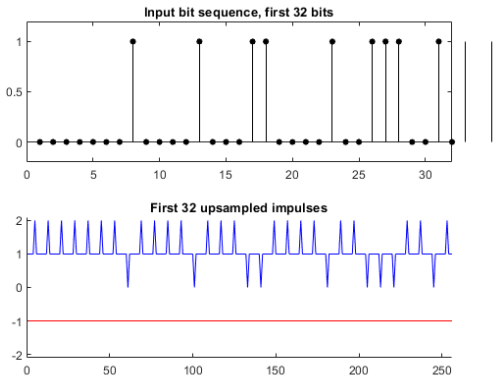


Figure 4.8: BPSK upsampled symbol pulses

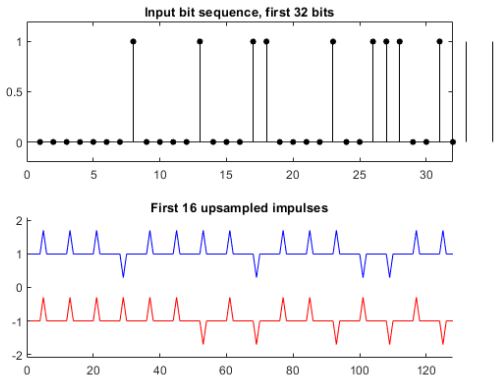


Figure 4.9: QPSK uppsampled symbol pulses

and Q impulses. This is show in Fig. 4.10. Effect of upsampling factor L is signal shift in discrete frequency band. Increasing L pushes signal spectrum closer to 0 frequency. It is easier to filter signal which is more 'packed' around low frequencies, thus it look that it is better to have larger upsampling factor L. There is a trade off in increased data rate and need for increased processing power.

Another question is delay of half the symbol time T_s when O-QPSK modulation is used. If symbol is sampled with even number of samples, for example 8, it not pos-

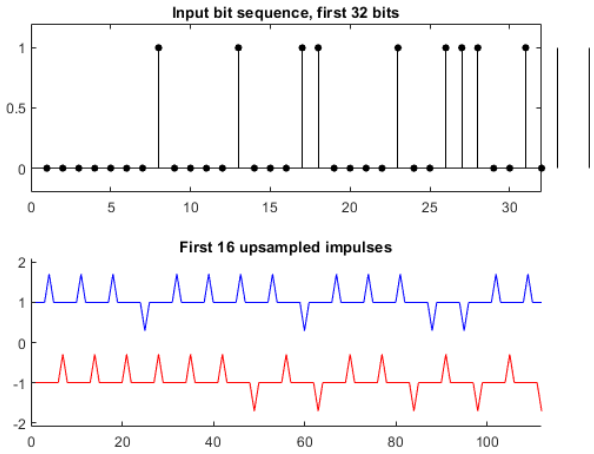


Figure 4.10: O-QPSK uppsampled symbol pulses

sible to delay impulses for a exactly half the symbol time. It can be delayed for a $floor[\frac{L-1}{2}]$ samples. In case of $L = 8$ that would be 3 samples. In case of even number of samples per symbol, L, to make correct delay of impulses interpolation would be needed. To avoid interpolation, thus reducing processing power, it is favorable to chose odd number of samples per symbol. Resulting shaped impulses are shown in

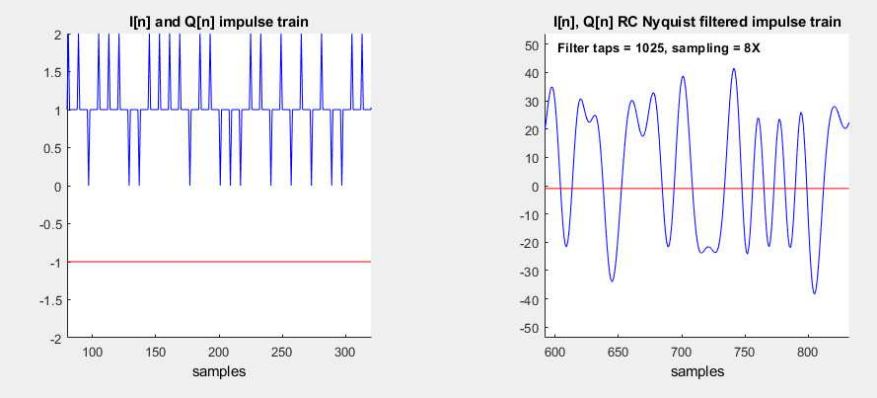


Figure 4.11: BPSK filtered and shaped I and Q impulses

Fig. 4.11, Fig. 4.12 and Fig. 4.13.

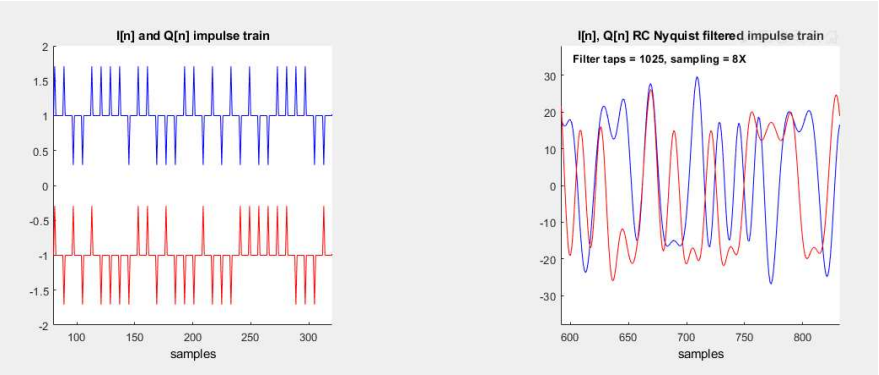


Figure 4.12: QPSK filtered and shaped I and Q impulses

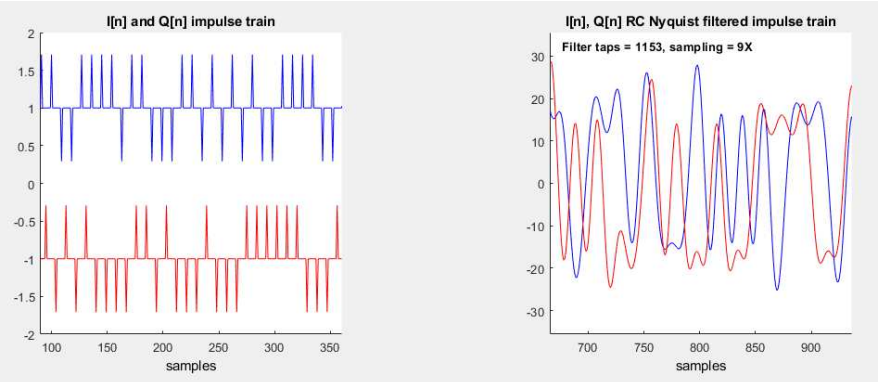


Figure 4.13: O-QPSK filtered and shaped I and Q impulses

5. Simulation results

5.1. Long RC filter span results

Using filter span 128 symbols and sampling factor L=8 next signal transition in vector space and resulting spectra are simulated. Shown in next sections.

5.1.1. BPSK results

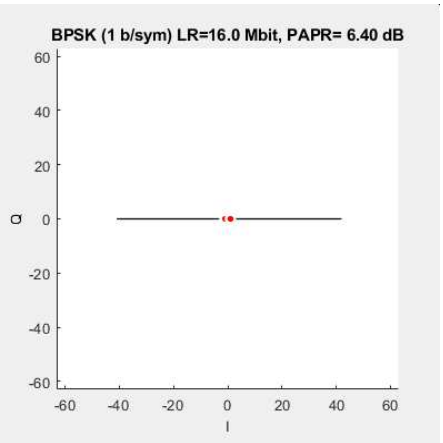


Figure 5.1: BPSK signal trasnition in signal space

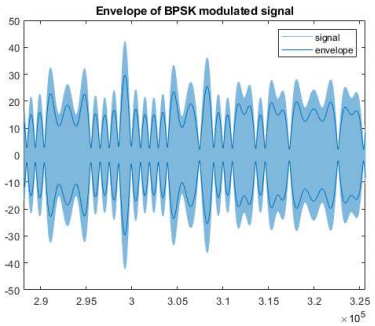


Figure 5.2: Enevlope of the BPSK signal

Fig. 5.1 shows that BPSK modulation have one-dimensional signal space. Envelope drops to zero in each symbol state change. Bandwidth of BPSK signal from Fig. 5.3 is wider than symbol rate bandwidth and conforms to rule for RC filter which is $SR(1+\alpha) = 16 \times 10^6 \times (1+0.22) = 19.52\ MHz$

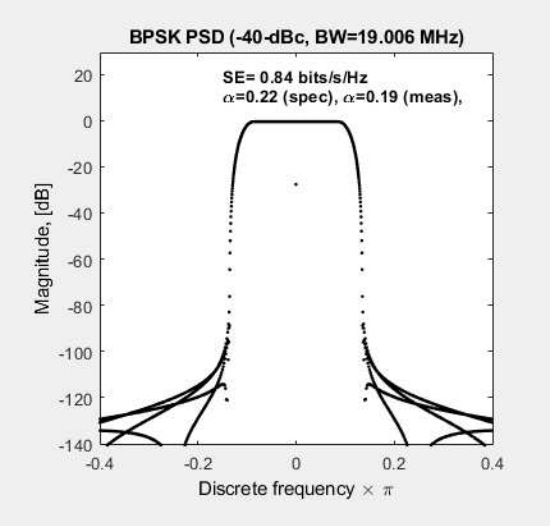


Figure 5.3: BPSK signal spectrum

5.1.2. QPSK results

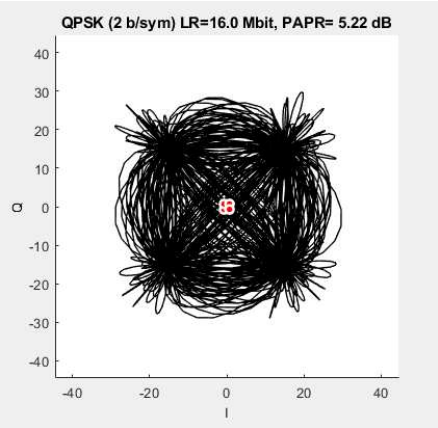


Figure 5.4: QPSK signal trasnition in signal space

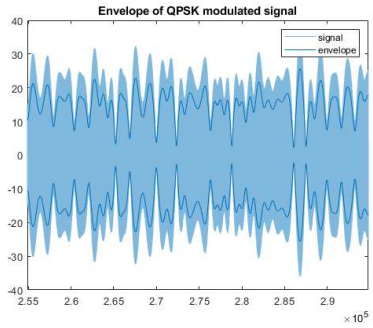


Figure 5.5: Enevlope of the QPSK signal

Fig. 5.4 shows that QPSK modulation have two-dimensional signal space. Envelope drops to zero, but not as in BPSK modulation. QPSK modulation produce signal with better Peak to Average Power Ratio. This modulation conforms the channel bandwidth requirement.

5.1.3. O-QPSK results

Fig. 5.7 shows that O-QPSK modulation have two-dimensional signal space. Envelope does not drop to zero amplitude and PAPR is beter than PAPR from QPSK modulation. Bandwidth is the sam as for QPSK modulation.

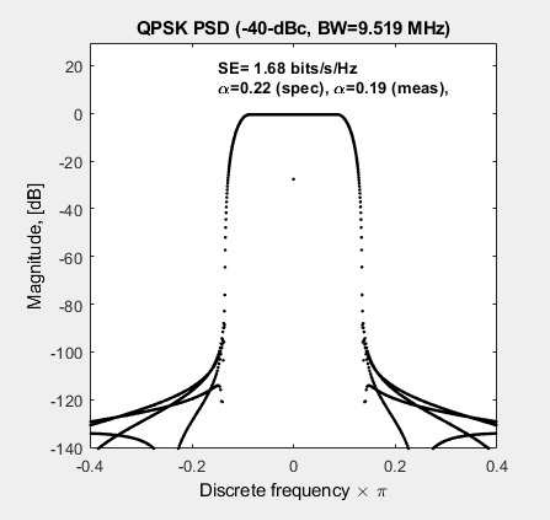


Figure 5.6: QPSK signal spectrum

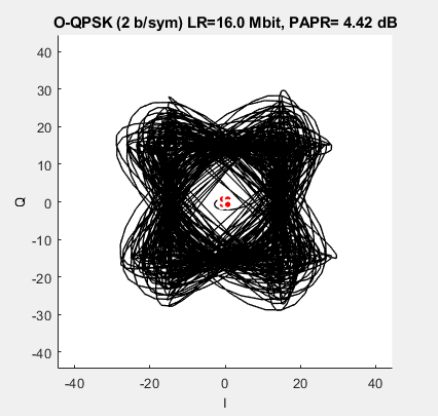


Figure 5.7: O-QPSK signal trasnition in signal space

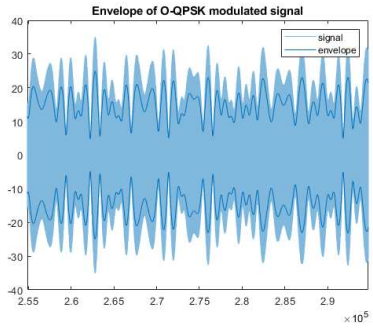


Figure 5.8: Enevlope of the O-QPSK signal

5.2. Shortening the RC filter span

When filter have smaller number of taps it's have greater error. Result is spectrum which stop band is not attenuated as much as with the longer filter. This simulation is concentrated on BPSK ind QPSK modulations.

Using the same rolloff factor alpha = 0.22 as in previous simulations, but shortening the filter span to 16 symbols and increasing the sampling factor to L=16 result with the filter response, as in Fig. 5.10.

Changing the rolloff factor from alpha=0.22 to alpha=0.30 widens passband bandwidth but stop-band attenuation is increased. Magnitude response of the RC filter with 257 tap, 16symbol span, 16 samples per symbol and rolloff=0.30 is shown in Fig. 5.11. Resulting spectra for BPSK and QPSK are shown in Fig. 5.12 and Fig. 5.13. From the

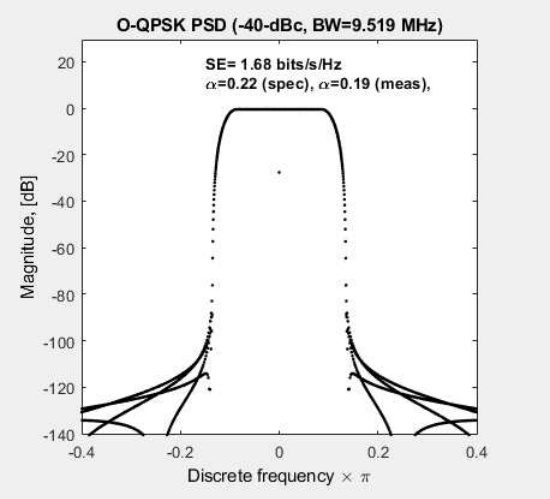


Figure 5.9: O-QPSK signal spectrum

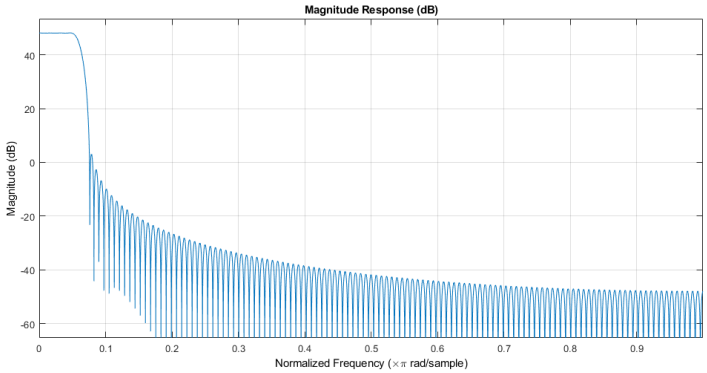


Figure 5.10: Magnitude response of 257 tap RC filter, span 16 symbols, L=16, α=0.22

simulation results it is clear that RC filter rolloff factor will make a trade off between excess bandwidth in passband and attenuation in stop-band. BPSK modulated signal violates channel bandwidth, thus using BPSK maximum attainable line rate is up to 8 Mbps. With this filter design, unwanted spectra is still under -40 dBc.

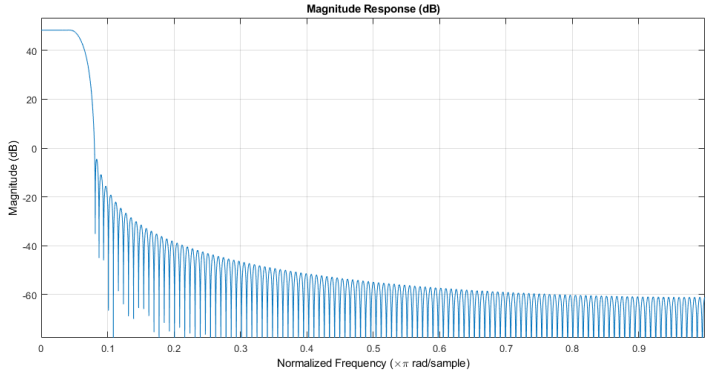


Figure 5.11: Magnitude response of 257 tap RC filter, span 16 symbols, L=16, α=0.30

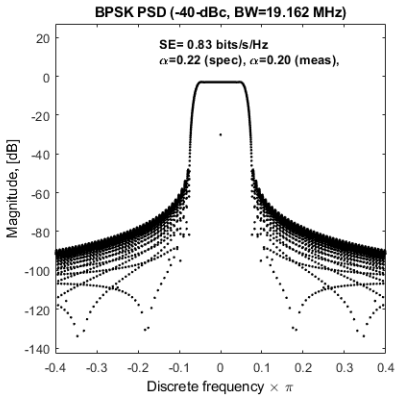


Figure 5.12: BPSK spectrum using 257 tap RC filter

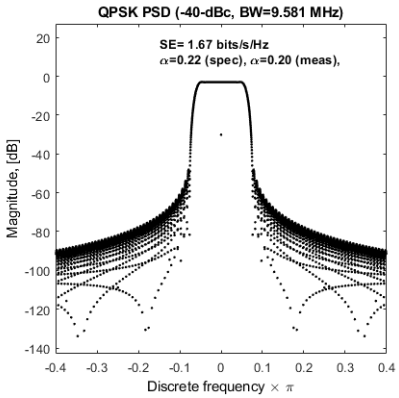


Figure 5.13: QPSK spectrum using 257 tap RC filter

5.3. Further shortening the RC filter span

Using the same rolloff factor alpha = 0.22 as in previous simulations, but shortening the filter span to 16 symbols and *decreasing the sampling factor to L=8* result with the spectra as in Fig. 5.14 and Fig. 5.15. From the simulation it stands out that the impact on stop band attenuation and first side lobe magnitude is greater by the filter span than by the sampling factor L. Minimum passband width is determined by the line rate, that is, symbol rate. If the only criterion would be first side lobe in spectrum below -40 dBc than filter span 16 symbols with 5 time oversampling would be marginally enough. But for better results, oversampling by a factor of 8 is needed. Thus resulting filter span using L=8 is 129 taps.

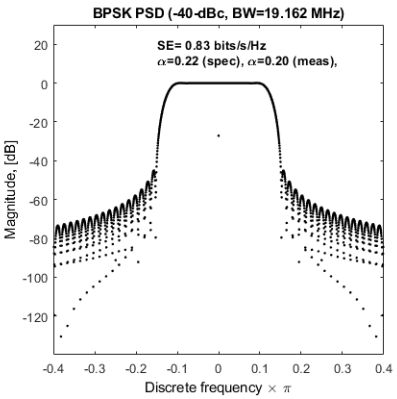


Figure 5.14: BPSK spectrum using 129 tap RC filter

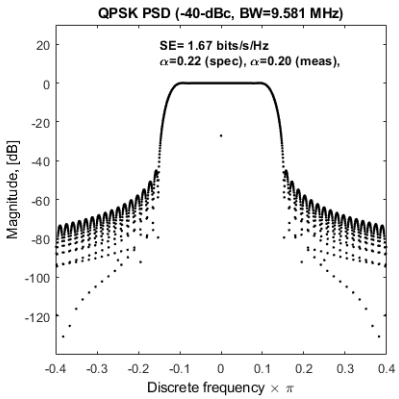


Figure 5.15: QPSK spectrum using 129 tap RC filter

6. Conclusion

This Thesis focus was on analysis digital signal processing steps for a baseband signals and how complex envelope signal is generated.
It is simulated how Nyquist filter affects complex envelope signal bandwidth. What are effects of rolloff factor, that is excess bandwidth and how filter impulse response is affecting spectral content of complex envelope. Trough multiple runs of simulation it is come to conclusion that to satisfy 10 MHz channel bandwidth using BPSK modulation maximum line rate is up to 8 MHz and spectral efficiency of BPSk is 0.84 bits/s/Hz. QPSK modulation has the spectral efficiency twice the BPSK modulation. To satisfy requirement of -40 dBc filter with at least 16 symbol span is needed. Oversampling factor from L=5 up is producing satisfying results. Thus resulting minimum filter span is 81 taps.
Whole processing system need to have two such filters. System output in case of L=5 oversampling would be 25 Msps for each of I and Q branches.
Next steps would be to consider decimating filter for lowering data throughput and analysis of digital to analog converter amplitude resolution effects.

BIBLIOGRAPHY

[1] User:Krishnavedala. (2011) Frequency response of raised-cosine filter with various roll-off factors. [Online; accessed 18-September-2020]. [Online]. Available: https://commons.wikimedia.org/wiki/File:Raised-cosine_filter.svg

[2] M. Vučić. (2020) Bilješke s predavannja iz predmeta obrada signala u komunikacijama. [Online; accessed 18-September-2020]. [Online]. Available: https://www.fer.unizg.hr/predmet/osuk/predavanja

[3] C. E. Stroud. (2011) Linear feedback shift registers. [Online; accessed 18-September-2020]. [Online]. Available: http://www.eng.auburn.edu/~strouce/class/elec6250/LFSRs.pdf

[4] H. Riebeek and R. Simmon, "Catalog of earth satellite orbits," Sep 2009. [Online]. Available: https://earthobservatory.nasa.gov/features/OrbitsCatalog

[5] "Pregled formula iz digitalne obradbe signala," 2019. [Online]. Available: http://www.fer.hr/predmet/dos/ispiti

[6] C. E. Shannon, "A mathematical theory of communication," *The Bell System Technical Journal*, vol. 27, no. 3, pp. 379–423, 1948.

[7] D. Petrinović, *Ekvivalencija vremenski kontinuiranih i diskretnih signala i sustava*, Fakultet elektrotehnike i računarstva, 2008.

[8] Wikipedia contributors, "Passband — Wikipedia, the free encyclopedia," https://en.wikipedia.org/w/index.php?title=Passband&oldid=972761523, 2020, [Online; accessed 20-September-2020].

[9] ——, "Baseband — Wikipedia, the free encyclopedia," https://en.wikipedia.org/w/index.php?title=Baseband&oldid=972377414, 2020, [Online; accessed 20-September-2020].

[10] ——, “Raised-cosine filter — Wikipedia, the free encyclopedia,” https://en.wikipedia.org/w/index.php?title=Raised-cosine_filter&oldid=948688776, 2020, [Online; accessed 18-September-2020].

[11] E. McCune, *Practical Digital Wireless Signals*, ser. The Cambridge RF and Microwave Engineering Series. Cambridge University Press, 2010.

Baseband Signal processing Chain for a Satellite Digital Transmitter

Abstract

Baseband signal processign chain for a Satellite Digital Transmitter is Master Thesis at the faculty of Electrical Engineering and Computing, University of Zagreb, Croatia. Digital processing procedures and steps for the baseband signal were analyzed. System for processing complex envelope signal was simulated. Minimal parameters of the digital processing system which satisfies assigned parameters of of signal quality were specified. Simulations were amde for digital modulation formats BPSK, QPSK and O-QPSK.

Keywords: BPSK, QPSK, O-QPSK, DSP, COMPLEX, MIXING, MODULATOR

Lanac za obradu signala u osnovnom pojasu u satelitskom digitalnom predajniku

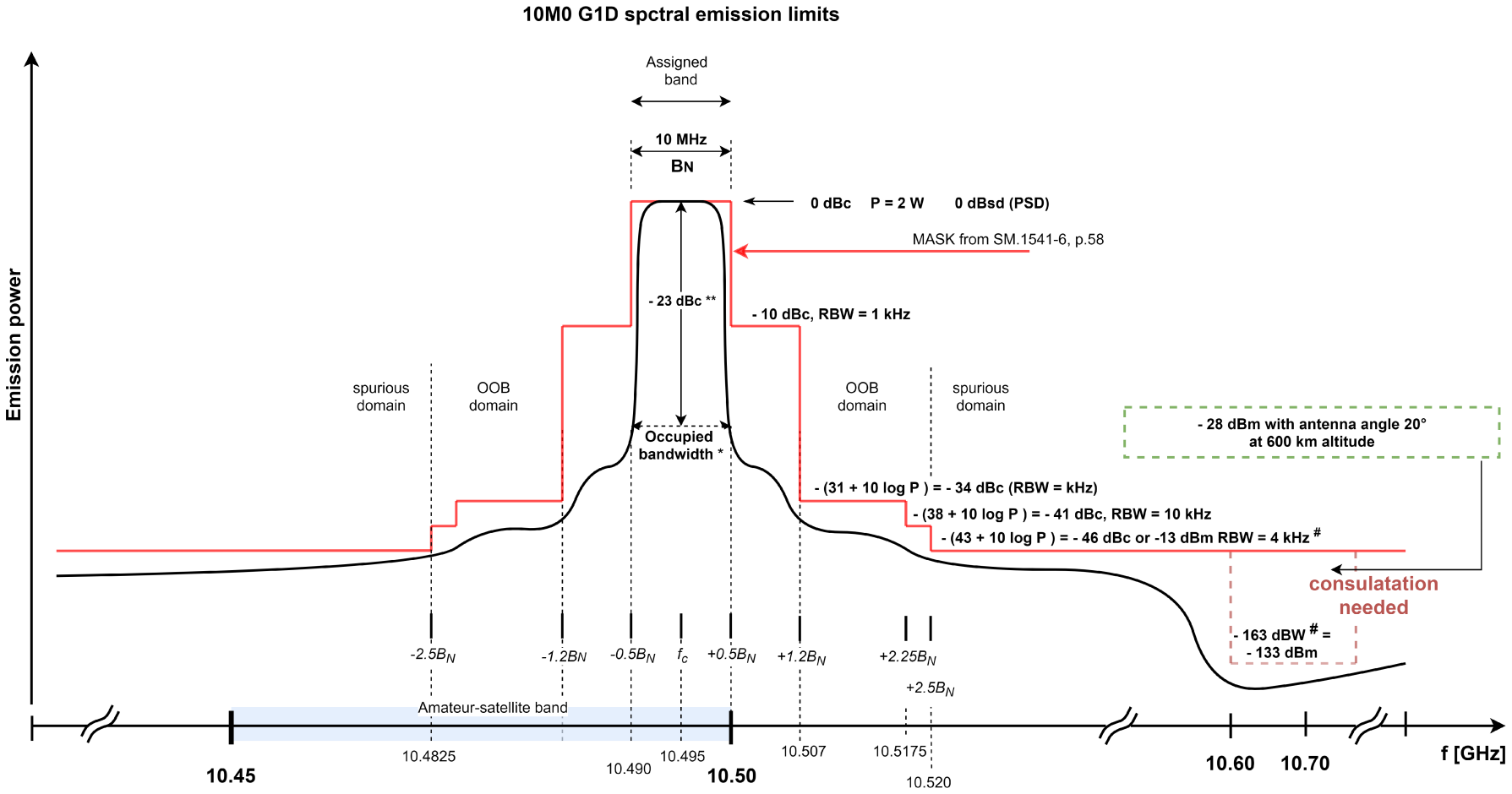
Sažetak

Lanac za obradu signala u osnovnom pojasu u satelitskom digitalnom predajniku je Diplomski rad na Fakultetu elektrotehnike i računarstva u Zagrebu. Analizirani su postupci digitalne obrade signala u osnovnom pojasu. Simuliran je sustav za obradu signala kompleksne ovojnice i određeni su minimalni parametri digitalnog sustava koji zadovoljavaju zadane kriterije kvalitete signala. Simulacije su provedene za tipove digitalnih modulacija BPSK, QPSK i O-QPSK.

Ključne riječi: BPSK, QPSK, O-QPSK, DSP, MODULATOR, KOMPLEKSNO, MIJEŠANJE

Appendix A
Spectral emission limits for Radio-Amater Band in X-Band radio frequencies

UTU Emission designator for 10 MHz bandwidth channel, digital phase modulation with single carrier (single channel communication) is 10M0 G1D.



(*) ITU-R SM1541-6, 1.153

(**) 99% power occupied bandwidth permitted by a particular emission mask can be determined by calculating the frequency difference between the 23 dB attenuation levels

(***) Ask J. Vuković - How is necessary bandwidth defined, depending on Link Budget or something else?

(***) Assigned bandwidth ≠ Necessary bandwidth ≠ Occupied bandwidth

(#) ITU-R SM.329-10, Table 9, p. 31

(##) ITU-R SM.329-10, Table 10, p. 34

Figure A.1: Spectral emission limits for 10 MHz amateur RF channel in X-Band using single channel communication and digital phase modulation

Appendix B
Source code

B.1. Source code of main Matlab simulation script

```
% File: MPSK_SIM_MS.m
%
% Master Thesis
%
% Work in progres
% Simulating an 4-PSK (QPSK) modulation
%
% Version:    0.200909-0
% Date:       September 9, 2020
% Student:    Mario Simunic
%   email:    mario.simunic@fer.hr
%   email1:   mario.simunic@gmail.com

% close all
% clear all
clc
colordef white

% ############################################################
% SIMUATION PARAMETERS
% ############################################################

use_prbs = 1;
% use_prbs = 0;

% -------------------------
% MODULATION
% -------------------------
```

```
29 modulation = 'BPSK';
30 M=2;                          % modulation order
31 % modulation = 'QPSK';
32 % M = 4;                        % modulation order
33 o_qpsk = 0;
34
35 % modulation = 'O-QPSK';
36 % M = 4;                        % modulation order
37 % o_qpsk = 1;
38
39
40 K = log2(M);                   % number of bits per symbol
41
42 % ---------------------------------
43 % TRANSMIT AND SAMPLE PARAMETERS
44 % ---------------------------------
45 LR = 8*10^6;                   % LR = line rate, Transfer speed, bits/
       second
46 SR = LR/K;                     % SR = symbol rate, symbols/second
47
48 Tsymbol = 1/SR;                % Symbol period, from link speed, br (
       bits per second)
49
50
51 % ------------------------------------------------
52 % TRANSMIT FILTER (Nyquist) PARAMETERS - Spectral Raised Cosine
       filter
53 % ------------------------------------------------
54 % RCF_shape = 'sqrt';         % Square-Root Raised Cosine filter
55 RCF_shape = 'normal';         % Square-Root Raised Cosine filter
56 RCF_span =16;               % Filter impulse response span
57 rolloff = 0.22;               % beta factor, spectrum rolloff
58 % ##### SYMBOL SAMPLING
59 L = 3;                        % symbol upsampling factor, samples per
       symbol
60
61 % ========= print parameters
62 starttime = char(datetime('now','Format','yyyy-MM-dd''_''HHmmss'));
63 log_filename = ['M-PSK_sim_',starttime,'.log'];
64 diary(log_filename);                  % diary on
65
66 fprintf("\nM-PSK SIMULATION by Mario Simunic\n");
67
```

```
68 fprintf("\n\nLog filename:\t\t\t M-PSK_sim_%s.log\n",starttime);
69
70 fprintf("\n\nSimulation parameters:\n");
71
72 fprintf("\nModulation:\t\t %s (M = %d)\n", modulation, M);
73
74 fprintf("\nBit rate:\t\t\t\t\t %d,\n", LR);
75 fprintf("\nSymbol rate:\t\t\t\t %d,\n", SR);
76 fprintf("\nSamples per symbol:\t\t\t %d,\n", L);
77 fprintf("\nSample rate:\t\t\t\t %d,\n", L*SR);
78 fprintf("\nTransmit filter shape:\t\t %s,\n", RCF_shape);
79 fprintf("\nTransmit filter span:\t\t %d symbols,\n", RCF_span);
80 fprintf("\nTransmit filter alpha:\t %1.2f\n", rolloff);
81
82 % ------------------------------------------------------------
83
84
85
86
87
88
89 % ############################################################
90 % BINARY DATA - pseudorandom bit vector
91 % ############################################################
92 % This binary data vector represents a synchronization header (ie.
       Barker
93 % code) plus payload data with protective coding. All together
       scrambled
94 % for uniform distribution of zeros and ones. Uniform distribution
       of zeros
95 % and ones (without contonous blocks) helps to estimate carrier
96 % signal at receiver.
97
98 fprintf('\n>BEGIN ====== f_prbs.m
       ==================================>> \n\n');
99
100 if use_prbs == 1
101 %
       ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
102 % Generate Ciclic repetitive pseudo-random sequence - for a nice FFT
       :-)
```

```
103 %
    ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
104
105 seed = '100000000';     % if it has nine digits, its PRBS9, if 11
    its PRBS11.
106 % seed = '10000';       % 5 bits
107 % It can also do 15, 21, 31, but those will be too long for the
    memory
108 bits = length(seed);
109 prbs_period = int16(2^bits)-1;     % LFSR RNG periodicity
110 sequence_period = lcm(int16(K), prbs_period);   % symbols to
    repetition: symbols for cyclical sequence
111 m = 2*sequence_period;  % need to have enough data for cyclic fft
    capture
112
113 fprintf(' Generating %d bits of random data \n', m);
114 data = [];
115 for k=1:m
116     [symbol,seed] = f_prbs(seed,K);
117     switch symbol
118         %QPSK
119         case '00'
120             data = [data,0,0];
121         case '01'
122             data = [data,0,1];
123         case '10'
124             data = [data,1,0];
125         case '11'
126             data = [data,1,1];
127         %BPSK
128         case '0'
129             data = [data,0];
130         case '1'
131             data = [data,1];
132
133         otherwise
134             warning('Unsupported symbol type.');
135     end
136 end
137
138 else  % use random number
139     data1 = f_rbs_ms(m*K);
```

```
140
141 end % end if
142
143 % Print data sequence
144 fprintf('\n data = \n');
145 for k=0:ceil(length(data)/40)-1
146     for(n=1:40)
147         if k*40+n <= length(data)
148             fprintf(" %d", data(k*40+n));
149         end
150     end
151     fprintf("\n");
152 end
153 fprintf('\n\n');
154 fprintf('>==== f_prbs_ms.m ========================================
      END\n\n');
155
156 % ------------------------------------------------------------
157
158
159
160
161
162
163 % ############################################################
164 % SYMBOL MAPPING
165 % ############################################################
166
167 [symbols, I, Q, SYMmapped] = f_mappsym_ms(modulation, data);
168
169 fig = figure('Name',['I and Q impulses of ', modulation,' modulation
    ']);
170     subplot(3,1,1);
171     stem(data,'k','MarkerSize',4,'MarkerFaceColor','auto');
172     axis([0, length(data)+1, -0.2, 1.2]);
173     title(['Input bit sequence']);
174
175     subplot(3,1,2);
176     stem(I,'b','MarkerSize',4,'MarkerFaceColor','auto');
177     axis([0, length(I)+1, -1.2, 1.2]);
178     title(['In-phase (I) impulses']);
179
180     subplot(3,1,3);
```

```
    stem(Q,'r','MarkerSize',4,'MarkerFaceColor','auto');
    title(['Quadrature (Q) impulses']);
    axis([0, length(Q)+1, -1.2, 1.2]);

fprintf("\nFigure %d: I and Q impulses\n", fig.Number);



if length(data) > 32
    N_disp = 32;
else N_disp = length (data);
end

N_IQ_disp = N_disp/log2(M);

fig = figure('Name',['First ',num2str(N_disp),' I and Q impulses of
   ', modulation,' modulation']);
    subplot(3,1,1);
    stem(data,'k','MarkerSize',4,'MarkerFaceColor','auto');
    axis([0, N_disp, -0.2, 1.2]);
    title(['Input bit sequence, first ',num2str(N_disp),' bits']);

    subplot(3,1,2);
    stem(I,'b','MarkerSize',4,'MarkerFaceColor','auto');
    axis([0, N_IQ_disp, -1, 1]);
    title(['First ',num2str(N_IQ_disp),' I impulses']);

    subplot(3,1,3);
    stem(Q,'r','MarkerSize',4,'MarkerFaceColor','auto');
    title(['First ',num2str(N_IQ_disp),' Q impulses']);
    axis([0, N_IQ_disp, -1, 1]);

fprintf("\nFigure %d: First %d I and Q impulses\n", fig.Number,
   N_disp);

%------------------------------------------------------------




% ############################################################
% SYMBOL FILTERING
```

```
% ############################################################

% ### UPSAMPLE
I_up = upsample(I, L);
Q_up = upsample(Q, L);

if o_qpsk == 1                % ##### Using O-QPSK
    Q_up = circshift(Q_up, floor(L/2));     % Equivalent to time
   delay
                                            % of half a sybol
end

fig = figure('Name',['Upsampled I and Q signals']);
    subplot(2,1,1);
    stem(data,'k','MarkerSize',4,'MarkerFaceColor','auto');
    axis([0, length(data), -0.2, 1.2]);
    title(['Input bit sequence']);

    subplot(2,1,2);
    hold on
    plot(1 + I_up,'b');
    plot(-1 + Q_up,'r');
    hold off
    axis([0, length(I_up), -2.1, 2.1]);
    title(['Upsampled impulses']);

fprintf("\nFigure %d: %s input bit stream, I and Q signals upsampled
    with %d samples per symbol.\n", fig.Number, modulation, L);

if length(data) > 32
    N_disp = 32;
else
    N_disp = length(data);
end
N_IQ_disp = N_disp/K;


fig = figure('Name',[modulation,' I and Q upsampled signals']);
    subplot(2,1,1);
    stem(data,'k','MarkerSize',4,'MarkerFaceColor','auto');
    axis([0, N_disp, -0.2, 1.2]);
    title(['Input bit sequence, first ',num2str(N_disp),' bits']);

```

```
263     subplot(2,1,2);
264     hold on
265     plot(1 + [zeros(1,floor(L/2)),I_up(1:end-floor(L/2))],'b');
          % [zeros(1,L),I_up] inserted zeros are for plot alignment of
        impulses with stem(data)
266     plot(-1 + [zeros(1,floor(L/2)),Q_up(1:end-floor(L/2))],'r');
267     hold off
268     axis([0, N_IQ_disp*L, -2.1, 2.1]);
269     title(['First ',num2str(N_IQ_disp),' upsampled impulses']);
270
271 fprintf("\nFigure %d: %s input bit stream, I and Q signals upsampled
        with %d samples per symbol.\n", fig.Number, modulation, L);
272
273
274
275 % ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
276 % NYQUIST FILTER - design using L samples per symbol
277 % ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
278
279 % simulation parametrs - at the begining
280
281 % #### Ideal Nyquist filter is brick-wall filter or rectangular
        spectral
282 % pulse. Problems with ideal filter - Impulse response is decaying
        slowly,
283 % it is non-causal, can't be realized, etc.
284 % Paramters of Nyquist filter ----
285 % We will be using Raised Cosine and/or Square-Root Raised Cosine
        filter
286
287 % h_rc is a impulse response coeficients of FIR nyquist filter
288 h_rc = f_raisedcosine_ms( rolloff, RCF_span, L, RCF_shape, 'matlab')
        ;
289 %
        ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::

290 % Filter Gain, [dB]
291 FG = 40-4.139;  % To get PSD at 0 dB
292 % rcosdesign and f_raisedcosine_ms returns filter coefficients with
293 % energy = 1, ie. h = b2/sqrt(sum(b2.^2));
294 % It would be beneficial to use filter with heigher gain, eg. +20db,
        but
295 % not to exceed dynamic range of variable (register).
```

```
G = 10^(FG/20);
                % In case of using R bit wide register and fixed
    point
                % arithmetic. Number in 2's complement format 1.(R
    -1) can
                % have maximum positive value 1 - 2^(-(R-1)).
                % To avoid saturation use gain K < 1.0
                % For 13 bit register and number in fractional 1.12
    format, maximum postive
                % number is 0.99975586

% impulse reponse with gain
h_rc = G * h_rc;
% ------------------------------------------------------------

% plot RC filter

fig = figure('Name','f_raisedcosine_ms: Designed Nyquist filter');
    stem(h_rc,'MarkerSize',3,'markerFaceColor','none');
    axis([-5, length(h_rc)+6, -0.1 0.3]);
    title({['Impulse response of ',upper(RCF_shape),' Raised Cosine
    filter'],['Span=',num2str(RCF_span),' length=',num2str(length(
    h_rc))]});
    xlabel('coefficient');
    ylabel('amplitude');
fprintf("\nFigure %d: Impulse response of %s Raised Cosine filter,
    span = %d symbols with %d samples per symbol.\n",...
    fig.Number, RCF_shape, RCF_span, L);

% Magnitude frequency characteristics
H_rc = 20*log10((1/length(h_rc))*abs(fftshift(fft(h_rc))));
f_step = 2/(length(H_rc)-1);          % 2*pi/pi = 2;
f_axis = -1:f_step:1;                 % f_axis = -1*pi/pi:f_step/pi
    :1*pi/pi;

fig = figure('Name','Magnitude frequency response of nyquist filter'
    );
    plot(f_axis,H_rc);
    axis([-1.05 1.05 max(H_rc)-120 max(H_rc)+10]);
    title(['Magnitude frequency response of ',upper(RCF_shape),'
    Raised Cosine filter']);
    xlabel('discrete frequency \times \pi');
    ylabel('magnitude, [dB]');
```

```
330 fprintf("\nFigure %d: Frequency response of %s Raised Cosine filter,
        span = %d symbols with %d samples per symbol.\n",...
331     fig.Number, RCF_shape, RCF_span, L);
332 % ----------------------------------------------------------------
333
334
335
336 % ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
337 % PULSE SHAPING (filtration) using Raised Cosine Nyquist filter
338 % ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
339
340 [D, state_D] = f_set_fir_delay( floor(L/2)); % used to prepare the
       FIR for delaying one half of the symbol
341 [BI, state_BI] = f_set_fir_delay( 1); % used for FIR for syncing the
        response with the inpyut impulses
342 [BQ, state_BQ] = f_set_fir_delay( 1); % used for FIR for delaying
       one half of the symbol
343
344 % Signals are concatenated with zeros at the end to allow filter
       response
345 % to decay
346 I_up = [I_up, zeros(1,L*(RCF_span/2))];
347 Q_up = [Q_up, zeros(1,L*(RCF_span/2))];
348
349 I_f_state = zeros(1,length(h_rc)-1);   % I filter state variable
350 Q_f_state = zeros(1,length(h_rc)-1);   % Q filter state variable
351
352 [I_filt, I_f_state] = filter(h_rc, 1, I_up, I_f_state);    % In-
       phase branch filtration
353 [Q_filt, Q_f_state] = filter(h_rc, 1, Q_up, Q_f_state);    %
       Qadrature branch filtration
354
355
356 fig = figure('Name',[modulation,' - filtered I and Q signals']);
357     hold on
358     plot(0.5 + I_filt,'b');
359     plot(-0.5 + Q_filt,'r');
360     hold off
361     axis([0, length(I_filt), -(max(I_filt)+0.5)*1.1, (max(I_filt)
       +0.5)*1.1]);
362     title(['Filtered impulses']);
363     xlabel('sample');
364     legend('I signal','Q signal','location','east');
```

```
365
366 fprintf("\nFigure %d: %s quadrature signals filtered using %s Raised
        Cosine filter, span = %d symbols with %d saples per symbol.\n
    ",...
367     fig.Number, modulation, RCF_shape, RCF_span, L);
368 % ------------------------------------------------------------
369
370
371
372
373
374
375
376 % ############################################################
377 % SPECTRAL ANALYSIS
378 % ############################################################
379
380 % ### FFT
381 ss_period = L*sequence_period;
382 ssp = L*(RCF_span);                 % we will make FFT somewhere in
        signal. This beginng is SSP
383 Nfft = double(ss_period);           % ss_period == speriod
384 % -----
385
386 I_fft = 1/Nfft * fftshift(fft( I_filt( ssp : ssp+Nfft ), Nfft)); %
        FFT of filtered I signal
387 Q_fft = 1/Nfft * fftshift(fft( Q_filt( ssp : ssp+Nfft ), Nfft)); %
        FFT of filtered Q signal
388
389 % P_fft = 10*log10(abs( I_fft.^2 + Q_fft.^2));    % PSD - here lies
        a problem - sidelobes in spectrum
390 P_fft = 10*log10(abs( I_fft .* conj(I_fft) + Q_fft .* conj(Q_fft)));
          % PSD - here lies a problem
391 I_fft = 20*log10(abs(I_fft));
392
393 fstep = 2/Nfft;
394 f_axis = -1:fstep:1-fstep;
395
396 %find max magnitude
397 mag0dBc = max(P_fft);                   % Carrier magnitude or for M
        -PSK it's max magnitude in passband
398 % mag0dBc = max(I_fft);
399 mag_limit_dBc = -40;                    % 60 dB below carrier
```

```
400 bw_limit = mag0dBc + mag_limit_dBc;     % Bandwidth limit - I choose
        to set at 60 dB below carrier (-60 dBc)
401 for k = Nfft:-1:floor(Nfft/2)           % search index where FFT is
       gretaer than magnitude given in bw_limit
402 %     if P_fft(k) > bw_limit
403     if P_fft(k) > bw_limit
404         leftk = k;
405         break;
406     end
407 end
408
409 % #### bandwidth
410 bw_desc = 2*f_axis(leftk);      % descrete domain bandwidth
411 fsample = SR*L;                 % symbol sample rate
412 bw_cont = bw_desc * fsample/2;  % continous domain bandwidth
413
414 % Efefctive bandwidth  = SPECTRAL EFFICIENCY
415 % bweff = SR*K/BW_cont;     % LR = SR*K
416 bweff = LR/bw_cont;             % spectral efficiency, [bits/s/Hz]
417
418
419 % #### PAPR
420 E = I_filt.^2 + Q_filt.^2;
421 P = sum(E)/length(E);
422 Pavg = 10*log10(P);
423
424 Ppeak = 10*log10(max(E));
425
426 PAPR = Ppeak - Pavg; %It is Ppeak/Pavg, but this is logarithmic
427 %-----------------------------------------------------------------
428
429
430
431
432
433
434 % ###########################################################
435 % RESULTS PLOTING
436 % ###########################################################
437 % plot labels
438 textlabel2 = sprintf('Filter taps = %d, sampling = %dX', RCF_span*L
        +1, L);
439
```

```
440 textlabel3 = sprintf('%s (%d b/sym) LR=%3.1f Mbit, PAPR=%5.2f dB',
    modulation, K, LR*1.0e-6, PAPR);
441
442 bwlabel1 = sprintf(' PSD (%d-dBc, BW=%5.3f MHz)', mag_limit_dBc,
    bw_cont*1.0e-6 );
443
444 bwlabel2 = sprintf('SE=%5.2f bits/s/Hz', bweff );
445 bwlabel2_pos = [-0.15, mag0dBc+20];
446
447 bwlabel3 = sprintf('\\alpha=%4.2f (spec), \\alpha=%4.2f (meas), ',
    rolloff, bw_cont/SR-1 );
448 bwlabel3_pos = [-0.15, mag0dBc+12];
449
450
451 % plot all
452 fig = figure('Name',[modulation,' simulation results'], 'Position'
    ,[50 50 1100 900]);
453     subplot(2,2,1)
454     hold on
455     plot(1 + I_up,'b');
456     plot(-1 + Q_up,'r');
457     hold off
458     axis([10*L 40*L -2 2]);
459     axis square;
460     title(['I[n] and Q[n] impulse train'],'FontSize', 11);
461     xlabel('samples');
462
463     subplot(2,2,2)
464     hold on
465     plot(1 + I_filt,'b');
466     plot(-1 + Q_filt,'r');
467     hold off
468     axis([L*(10+RCF_span/2) L*(40+RCF_span/2) -max(I_filt)*1.2 max(
    I_filt)*1.2]);
469     axis square;
470     % Kako je prof. Babic dobio vecu (istu) amplitudu na izlazu
    filtra??? -
471     % Pogledaj energiju filtra - skaliranje koeficijenata
472     title(['I[n], Q[n] RC Nyquist filtered impulse train'],'FontSize
    ', 11);
473     xlabel('samples');
474     text(L*(10+RCF_span/2)+10,max(I_filt)*1.1,textlabel2,'FontSize',
     10,'FontWeight', 'bold');
```

```
475
476     ax = max(I_filt);
477     subplot(2,2,3)
478     hold on;
479     plot( I_filt, Q_filt, 'k-','LineWidth', 1);
480     plot( I, Q, 'wo','LineWidth', 2,'MarkerFaceColor','r');
481     hold off
482     axis([-1.5*ax 1.5*ax -1.5*ax 1.5*ax]);
483     xlabel('I','FontSize', 10 );
484     ylabel('Q','FontSize', 10 );
485     title(textlabel3,'FontSize', 11 );
486     axis square;
487
488     subplot(2,2,4)
489     plot(f_axis, P_fft,'k.');
490     axis([-0.4 0.4 max(P_fft)-140 max(P_fft)+30]);
491     axis square;
492     title([modulation, bwlabel1],'FontSize', 11);
493     xlabel('Discrete frequency \times \pi');
494     ylabel('Magnitude, [dB]');
495     text(bwlabel2_pos(1), bwlabel2_pos(2), bwlabel2,'FontSize', 10,'
    FontWeight', 'bold' );
496     text(bwlabel3_pos(1), bwlabel3_pos(2), bwlabel3,'FontSize', 10,'
    FontWeight', 'bold' );
497
498 fprintf("\nFigure %d: %s simulation results.\n",...
499     fig.Number, modulation);
500
501 fprintf("\n\nRESULTS:\n");
502 fprintf('%3d-dBc BW = %2.3f MHz, SE = %2.2f bits/s/Hz\n', floor(
    mag_limit_dBc), bw_cont*1.0e-6, bweff );
503 fprintf('\\alpha=%4.2f (spec), \\alpha=%4.2f (meas), PAPR=%5.2f dB\n
    ', rolloff, bw_cont/SR-1, PAPR );   % PAPR - to explain !!!
504
505 % Just spectrum figure
506 figure('Name',[modulation,' - resulting spectrum'])
507     plot(f_axis, P_fft,'k.');
508     axis([-0.4 0.4 max(P_fft)-140 max(P_fft)+30]);
509     axis square;
510     title([modulation, bwlabel1],'FontSize', 11);
511     xlabel('Discrete frequency \times \pi');
512     ylabel('Magnitude, [dB]');
```

```
513     text(bwlabel2_pos(1), bwlabel2_pos(2), bwlabel2,'FontSize', 10,'
        FontWeight', 'bold' );
514     text(bwlabel3_pos(1), bwlabel3_pos(2), bwlabel3,'FontSize', 10,'
        FontWeight', 'bold' );
515
516
517 % Envelope of the signal
518 %
519 fs_RF = 100*SR*L;
520 Tc_RF = 1/fs_RF;
521 fc = fs_RF/20;
522
523 % take peace of filtered pulses and resample the to RF sampling rate
524 % (simulation)
525
526 I_filt_block = I_filt(1: L*RCF_span*5);
527 Q_filt_block = Q_filt(1: L*RCF_span*5);
528 I_rf = resample(I_filt_block,100,1);
529 Q_rf = resample(Q_filt_block,100,1);
530
531 t_step = 1/fs_RF;
532 tc = 0:t_step:(length(I_rf)-1)*t_step;
533
534 uc_sine = -sin(2*pi*fc*tc); % cosine signal, frequency fc, duration
        = I_rf
535 uc_cosine = cos(2*pi*fc*tc);
536
537 I_t = uc_cosine .* I_rf;
538 Q_t = uc_sine .* Q_rf;
539
540 u_rf = I_t + Q_t;
541
542 figure();
543 envelope(u_rf,20*L,'rms');
544 title(['Envelope of ',modulation,' modulated signal']);
545
546
547
548 fprintf("\n\n============================================== END
        =========");
549 fprintf("\n\n");
550 diary off
```

B.2. Source code of PRBS generator

```
function [symbol, nextseed] = f_prbs(seed,bits)
% seed in string form, the length of the seed determines n
% output is a bit for bits=1, but you can get any number of bits out
    with
% bits

n = numel(seed);
switch n
    case 1
        symbol = '';
        for k=1:bits
            if rand > 0.5
                symbol = strcat(symbol, '1');
            else
                symbol = strcat(symbol, '0' );
            end
        end
        nextseed = '0';
        return;
    case 2
        tap1 = 2;
        tap2 = 1;
    case 3
        tap1 = 3;
        tap2 = 1;
    case 4
        tap1 = 4;
        tap2 = 1;
    case 5
        tap1 = 5;
        tap2 = 2;
    case 6
        tap1 = 6;
        tap2 = 1;
    case 7
        tap1 = 6;
        tap2 = 7;
    case 9
        tap1 = 5; % 4
        tap2 = 9;
    case 10
```

```
        tap1 = 10;
        tap2 = 3;
    case 11
        tap1 = 9;
        tap2 = 11;
    case 15
        tap1 = 14;
        tap2 = 15;
    case 17
        tap1 = 17;
        tap2 = 3;
    case 18
        tap1 = 18;
        tap2 = 7;
    case 20
        tap1 = 20;
        tap2 = 3;
    case 21
        tap1 = 21;
        tap2 = 2;
    case 22
        tap1 = 22;
        tap2 = 1;
    case 23
        tap1 = 18;
        tap2 = 23;
    case 25
        tap1 = 3;
        tap2 = 25;
    case 28
        tap1 = 28;
        tap2 = 3;
    case 29
        tap1 = 29;
        tap2 = 2;
    case 31
        tap1  = 28;
        tap2 = 31;
    otherwise
        fprintf('prbs unrecognized... exit\n');
        return;
end
verb = 0;
```

```
for k=1:n
    rg(k) = str2num(seed(k));
    if verb
        fprintf('%d', rg(k));
    end
end
if verb
    fprintf('\n');
end
symbol = '';
for k=1:bits
    if (rg(tap1) == 1) && (rg(tap2) == 0)
        dummy = 1;
    elseif (rg(tap1) == 0) && (rg(tap2) == 1)
        dummy = 1;
    else
        dummy = 0;
    end
    for ii=n:-1:2
        rg(ii) = rg(ii-1);
    end
    rg(1) = dummy;
    dum  = num2str(rg(n));
    symbol = strcat( symbol, dum );
end
nextseed = '';
for k=1:n
    nextseed = strcat( nextseed, num2str(rg(k)));
end
```

B.3. Source code of symbol mapper

```
%   F_MAPPSYM_MS() will mapp input data vector of bits (bitstream) to
%   M - point (M-ary) constelation based on Gray code.
%   modulation - type of modulation. Supported types are 'BPSK', 'QPSK'.
%   bitstream - input vector (length power-of-2) of integers or boolean
%   representing bits.
%
%   f_mappsym_ms(modulation, bitstream) outputs binary symbols in form of
%   K x N matrix, where K is number of bits in constellation symbol and N
%   is a length(bitstream)/K constellation points.
%
%   [symbols, I, Q, SYMmapped] = f_mappsym_ms(modulation, bitstream)
%   outputs binary symbols, mapped constellation points in vector of
%   In-Phase componenets, I, and quadrature components, Q, and the vector
%   of complex points, SYMmapped, consisting of I + 1i*Q points.
%
%
% Version:    0.200909-0
% Date:       September 9, 2020.
% Student:    Mario Simunic
%   email:    mario.simunic@fer.hr
%   email1:   mario.simunic@gmail.com

function [symbols, I, Q, SYMmapped] = f_mappsym_ms ( modulation, bitstream )
version = '0.2009090-0'; % mod(2006050,11) = 2
filename = 'f_mappsym_ms.m';
file = dir(filename);

fprintf('\n\n>BEGIN ===== f_mappsym_ms.m ===== ver. %s ================>\n',version);
if isempty(file) == 1
    warning('File f_mappsym_ms.m is not in the current folder.');
else
%     fprintf("%s, file version: %s, last change: %s\n", file.name, version, file.date);
```

```
33 end
34
35
36 %% Constellations (Look-up Tables)
37 % vector-row format!!!
38
39 % BPSK constellation
40 constell_BPSK =  [ 1 , ...   % 0
41                   -1 ];      % 1
42
43 % QPSK contellation, Gray code, Hamming distance = 1
44 constell_QPSK =  [ exp( 1i *   pi/4 ), ...   % 00 = 0d
45                    exp( 1i * 3*pi/4 ), ...   % 01 = 1d
46                    exp( 1i * 7*pi/4 ), ...   % 10 = 3d
47                    exp( 1i * 5*pi/4 ) ];     % 11 = 2d
48
49 % Other constellations
50
51
52
53 %% Symbol Mapping
54 % M = modulation order
55 % K = number of bits per symbol
56
57 % convert type just for the logic operations (if, switch, etc.)
58 bitstream = boolean(bitstream);
59
60 switch modulation
61
62     case 'BPSK'
63         constell = constell_BPSK;
64         M = 2;
65         K = log2(M);
66
67     case 'QPSK'
68         constell = constell_QPSK;
69         M = 4;
70         K = log2(M);
71
72     case 'O-QPSK'
73         constell = constell_QPSK;
74         M = 4;
75         K = log2(M);
```

```
76
77      otherwise
78          error('Unsupported modulation type.');
79  end
80
81  % Check in bistream length is correct for the modullation type.
82  % len need to be divisible by the K
83  len = length (bitstream);
84  if (mod(len,K) ~= 0)
85      error('Unsupported length of input data vector.\n');
86  end
87
88  fprintf("\n\nUsing modulation: %s ( M = %d)\n", modulation, M);
89
90
91      % symbol mapping using matrix operation and block data
92      % syms = reshape(data,K,L/K);      % Vector-row convert to K-
        elemnt columns.
93      %                                  % Each column is a symbol.
94      % addr = (2.^((K-1:-1:0))*syms);   % Symbols converted to look-up
         table
95      %                                  % address.
96      %
97      % Ss = constell (addr+1);          % Symbol stream
98
99
100 % Symbol mapping using elemnt operation and elemnt-by-element
        operation
101     %     Parameters for testing
102     %     data = [1 0 1 1 0 0 0 1];
103     %     K = 2;
104     %     L = 8;
105
106 % My own reshape and mapping
107 SYMmapped = zeros(1,len/K);          % memory allocation for mapped
        symbols
108 symbols = zeros(K,len/K);            % symbols, binary
109 symbol = zeros(1,K);                 % one symbol, binary
110 addr = zeros(1,len/K);               % memory allocation for symbol
        addressing
111 col = 0;
112 idx = 0;
113 for ii = 1: K : len
```

```
    col = col +1;
    idx = idx+1;
    for kk = 0 : K-1
        row = kk+1;
        symbols(row,col) = bitstream(ii+kk);    % All symbols

        symbol(kk+1) = bitstream(ii+kk);        % One symbol

        addr(idx) = addr(idx) + ((2^(K-1-kk)) * bitstream(ii+kk));

        SYMmapped(idx) = constell (addr(idx)+1);

    end
end



%% plot constellation points
fig = figure('Name','f_mappsym_ms: Symbols mapped to constellation')
    ;
hold on;
    % unit cicrcle definition
    r = 1;
    theta = 0:2*pi/50:2*pi;
    x = r*cos(theta);
    y = r*sin(theta);
    plot(x,y,'b.');         % Unit circle

% constellation points from lookup table
plot(real(constell),imag(constell),'ro', 'MarkerFaceColor','r');
axis([-1.5 1.5 -1.5 1.5]);
% axis equal
axis square
grid on

% constellation point labels
if(0)
    for k = 1:length(constell)
        text(real(constell(k))-0.2, imag(constell(k))+0.2, dec2bin(k
    -1, K),...
            'Color','red','FontSize',13);
    end
end
```

```
155
156 % plot symbol point labels with number of counted symbols from data
    stream
157 if (1)
158     % count number of occurrence of each symbol
159     PointCnt = zeros(1,M);
160     for k = 1:length(addr)
161         PointCnt(addr(k)+1) = PointCnt(addr(k)+1) + 1;
162     end
163     % For each constellation point print number of occurrences
164     for k = 1:length(constell)
165         msg = [dec2bin((k-1),K),' \times ',int2str(PointCnt(k))];
166         pos_x = real(constell(k))-0.2;
167         pos_y = imag(constell(k))+0.2;
168         text(pos_x, pos_y, msg,'Color','red','FontSize',13);
169     end
170 end
171
172 hold off;
173 xlabel('In-Phase'); ylabel('Quadrature');
174 title([modulation,' constellation with number of occurrences']);
175 % ax = gca;
176 % ax.XAxisLocation = 'origin';
177 % ax.YAxisLocation = 'origin';
178 % set(ax, 'FontSize', 12);
179 fprintf("\nFigure %d: Symbols mapped to constellation\n", fig.Number
    );
180
181 I = real(SYMmapped);
182 Q = imag(SYMmapped);
183
184 % fprintf('\n Input data = \n');
185 % disp(bitstream);
186 %
187 % fprintf('\n Symbols mapped (each column is one symbol):\n');
188 % disp(symbols);
189 %
190 % fprintf('\n Symbols mapped to constellation = \n');
191 % disp(SYMmapped);
192
193 fprintf('\n>===== f_mappsym_ms.m
    ======================================= END\n\n');
194
```

```
195 end
196
197 %EOF
```

B.4. Source code of filter design

```
%F_RAISEDCOSINE_MS Spectral Raised Cosine Filter design.
%   h = f_raisedcosine_ms(alpha, span, sps, shape, implementation) returns spectral raised
%   cosine FIR filter coefficients, h, with rollof factor alpha. The filter
%   is truncated to SPAN symbols and each symbol is represented by sps
%   samples. F_RAISEDCOSINE_MS designs a symetric filter. Therefore, the
%   filter order which is span*sps, must be even. The filter energy is 1.
%   mode will define how FIR filter will be designed, 'mtlb' - using matlab
%   rcosdesing() function, or, 'ms' - using my own implementation. This is
%   for testing purposes and results should be identical.
%
%   alpha is rolloff factor,
%   span is filter impulse response duration (symbols),
%   sps is number of samples per symbol,
%   shape defines which spectral shape of the filter will be designed. When
%         shape is set to 'normal' function will return spectral Raised
%         Cosine filter coeficients. When shape is set to 'sqrt' function
%         will return spectral Square-Root Raised Cosine filter coefficients,
%   implementation is a string value 'matlab' or 'ms' where 'ms' represents
%                 My implementation.
%
% Version:    0.200909-0
% Date:       Spetember 9, 2020
% Student:    Mario Simunic
%   email:    mario.simunic@fer.hr
%   email1:   mario.simunic@gmail.com
%
function h = f_raisedcosine_ms( alpha, span, sps, shape, implementation )

```

```
29 version = "0.200909-0";
30 filename = 'f_raisedcosine_ms.m';
31 file = dir(filename);
32
33 fprintf('\n\n>BEGIN ====== f_raisedcosine_ms.m ==== ver. %s
    ============>\n', version);
34 if isempty(file) == 1
35     warning('File f_mappsym_ms.m is not in the current folder.');
36 else
37 %     fprintf(" %s, file version: %s, last change: %s\n", file.name,
     version, file.date);
38 end
39
40 % Debugging
41 % fprintf('\n Using implementation = %s\n', implementation);
42
43
44 %% Take One - Using Matlab function rcosdesign()
45 %  ================================================================
46 %
47 %  Spectral Rised Cosine filter design using rcosdesign()
48 %
49 %   b = rcosdesign(beta,span,sps) returns the coefficients, b, that
50 %   correspond to a square-root raised cosine FIR filter with
    rolloff
51 %   factor specified by beta. The filter is truncated to span
    symbols,
52 %   and each symbol period contains sps samples. The order of the
    filter,
53 %   sps*span, must be even. The FILTER ENERGY is 1.
54 %
55 %   b = rcosdesign(beta,span,sps,shape) returns a square-root raised
     cosine
56 %   filter when you set shape to 'sqrt' and a normal raised cosine
    FIR
57 %   filter when you set shape to 'normal'.
58
59 if (1 == strcmp(implementation,'matlab'))
60     fprintf('\n Using ''matlab'' implementation for computing
    coeffcients.\n\n');
61     % Raised cosine filter
62     b1 = rcosdesign(alpha, span, sps, shape);  % design impulse
    response
```

```
63
64      h = b1;
65  end
66
67  %  ==================================================================
68
69
70
71  %% Take Two - My implementation of raised cosine filter impulse
        response
72  %  ==================================================================
73  %
74  if (1 == strcmp(implementation,'ms'))
75
76      fprintf('\n Using ''ms'' implementation for computing
        coeffcients.\n\n');
77
78      FIRord = span * sps;         %
79      % Check if filter order is even
80      if ( mod(FIRord,2) ~= 0)   %
81              error('    Filter order of Raised Cosine filter must be
        even!');
82      end
83
84      FIRlength = FIRord + 1;
85      b2 = zeros(1,FIRlength);     % allocate memory for filter samples
86
87      for k=1 : FIRlength  %filter length L = filter_order+1, odd.
88          % irad = (double((k-1)-nfs/2)*(double(fs)/double(nfs)))/2;
89  %         idx = (((k-1) - FIRord/2)*(span/FIRord))/2;
90              % idx represents t/Tsymbol in raised-cosine equation
91              % idx is index in range [-6.0, 6.0]
92              % (span/FIRord) equals (1/L);
93          time = ((k-1) - FIRord/2)*(span/FIRord);
94
95          if time == 0
96              b2(k) = 1;  % sinc for t=0;
97          else
98  %             b2(k) = sin(2*pi*idx)/(2*pi*idx);   % sinc
99              b2(k) = sin(pi*time)/(pi*time);   % sinc
100         end
101
102         d = 2*alpha*time;   % part of denominator
```

```
103
104         if abs(d) == 1
105             b2(k) = 0;      % prevent division with zero
106         else
107             b2(k) = b2(k) * (cos(pi*alpha*time)/(1-d*d));
108         end
109
110     end
111
112     % Normalization to unit energy so that FILTER ENERGY would be 1
113         h = b2/sqrt(sum(b2.^2));
114 end
115
116
117 fprintf('>====== f_raisedcosine_ms.m
        ==================================== END\n\n');
118 end
119
120 % EOF
```