Communication theory Assignment-1

Report and results for MATLAB questions:

A)

We have taken B_c to be a vector of length N and it consists of -1, 1 generated randomly.

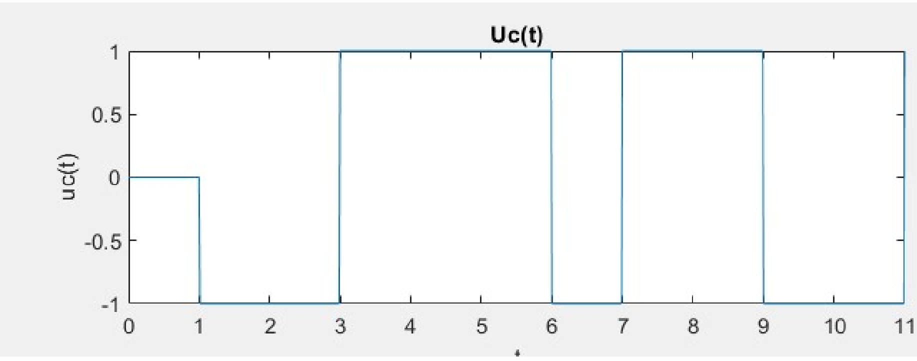
We need to plot $u_c(t)$ which is B_c multiplied with shifted rectangular pulses p(t-n).

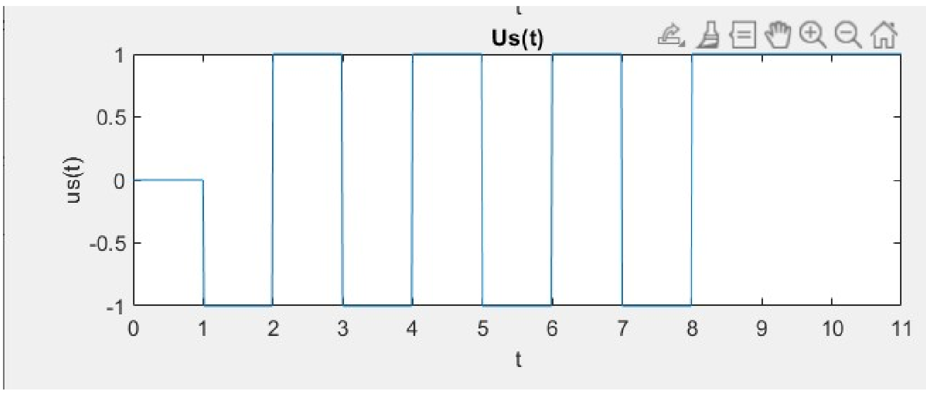
Parameters:

N=10

Fs=80 Hz

Plots:





Explanation:

Clearly, the above graph depicts {1, -1} being multiplied with shifted rectangular pulses. We have taken the sampling frequency as 80 Hz(multiples of 20) so that the graph actually looks like a rectangular pulse. Note that even for lesser frequency also graph can be obtained.

B), C)

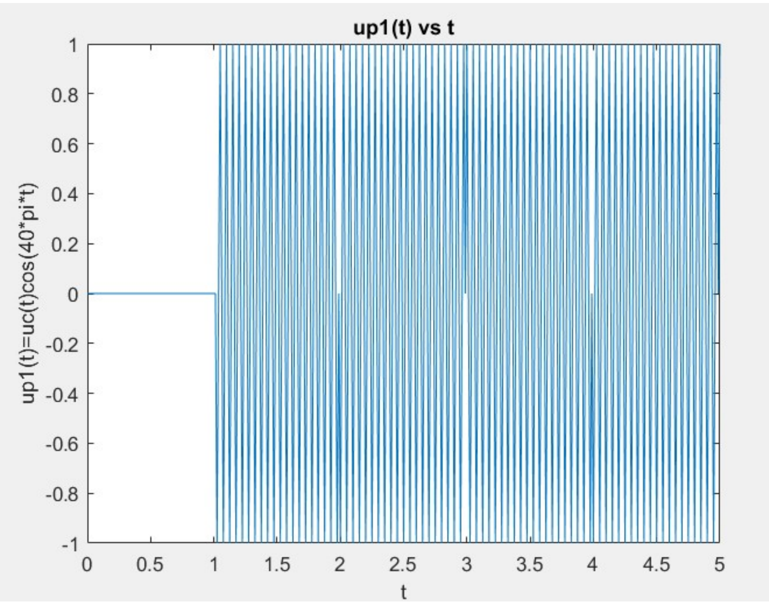
Here, we are required to upconvert the baseband form by multiplying uc(t) with cos(40πt) to get passband signal. We are expected to take a proper sampling rate so that we don't get a distorted plot.

Parameters:

N=4

Fs=80 Hz

Plots:



Explanation:

The above plot depicts the passband signal of uc(t). One cannot exactly tell what uc(t) after the upconversion happens by looking at up1(t). However, we find that the recognition when we have **a phase change** i.e. from -1 to 1 or 1 to -1, we notice a gap in the graph that is caused due to the change in phase of uc(t). So the name Binary Shift Keying(BPSK) is justified. Also the amplitude is 1 as expected.

Regarding the frequency, we need to be careful that when we take lesser frequencies, we get a bad graph (ie phase shift is not clear) and the sampling frequency needs to be slowly increased in multiples of 20Hz and when we get the expected graph we can stop.

D)

Here, we need to add the other upconverted component, and finally we generate the complete passband signal up(t) = uc(t)cos40πt − us(t)sin40πt.

Parameters:

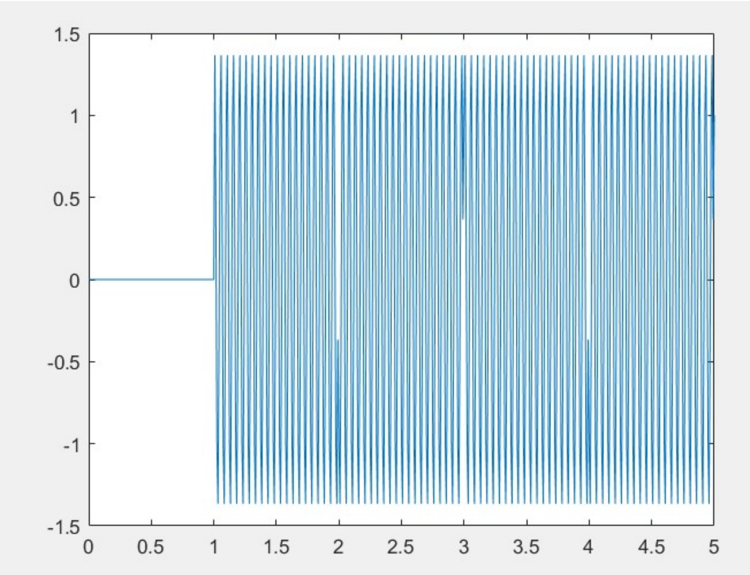
N=4

Fs=120 Hz

Plots:

X-axis- Time

Y-axis-up(t)



Explanation:

Here, we have finally got the passband signal up(t). It is impossible to tell what uc(t) and us(t) are from looking at the plots. Here, we notice that there is a **phase shift** whenever either **Bc(n) changes or Bs(n) changes.** Since there are 4 such possible cases, we say that this is QPSK. Also since we get a combination of sin and cos, the maximum value of their sum root(2) is the amplitude expected.

With regard to the sampling frequency, 80Hz is not sufficient to prevent the distortion of shapes we increase it to 120Hz so that the graph is neatly formed and looks nice. Any higher values are also fine as long as we clearly are able to observe the change in phase.

E)

Here, we need to downconvert the signal by multiplying the signal with cos(2πfct+ theta) and sin(2πfct+ theta), and then pass it through a low pass filter I(0, 0,25). We take phase to be 0.

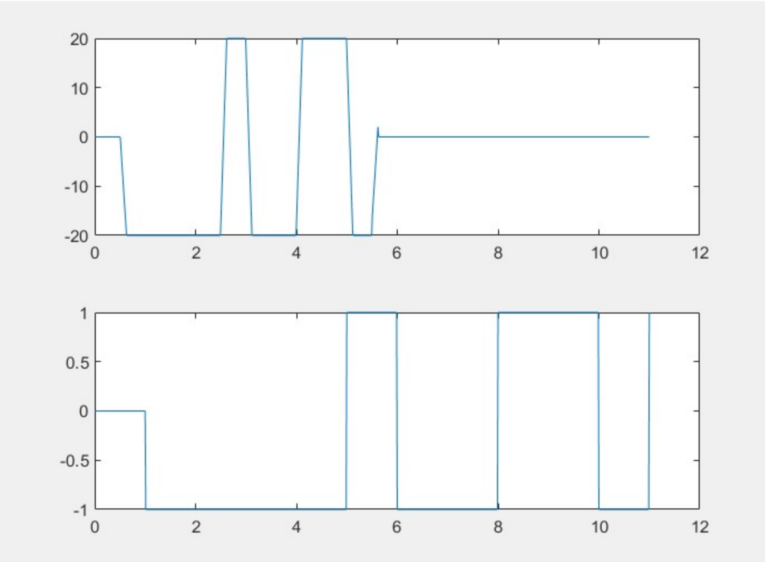
Parameters:

N=10

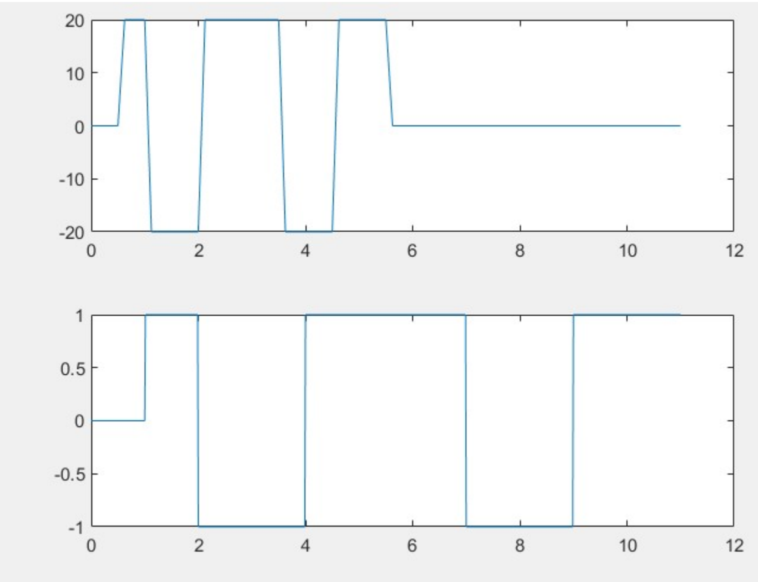
Fs=80 Hz

Plots:

Vc(t) -1st subplot; Uc(t) -2nd subplot



Vs(t) -1st subplot vs Us(t) -2nd subplot



Explanation:

Clearly we notice that in both the cases, the Vc, Vs obtained are similar to the actual waveform but are scaled. The scaling in x axis is due to defining new time variable for convolution.

We can see that we **can actually identify** what bit is sent using our eyeballs and the chance of error is almost 0.

F)

Here, we need to downconvert the signal by multiplying the signal with cos(2πfct+ theta) and sin(2πfct+ theta), and then pass it through a low pass filter I(0, 0,25). We take phase with which we multiply the values to be π/4.

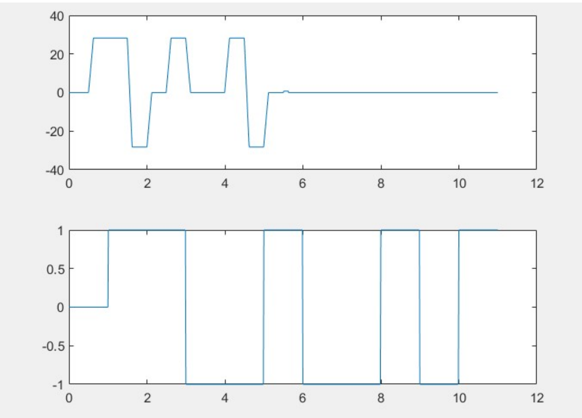
Parameters:

N=10

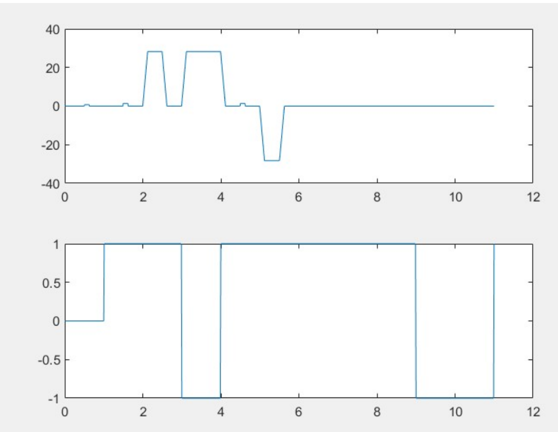
Fs=80 Hz

Plots:

Vc(t) -1 st subplot; Uc(t) -2nd subplot



Vs(t) -1st subplot; Vc(t)-2nd subplot



Explanation:

Clearly we notice that in both the cases, the Vc, Vs obtained are different from each other. The scaling in x axis is due to defining new time variable for convolution.

We **cannot actually identify** what bit is sent using our eyeballs and the chance of error is extremely high because of the initial phase in the value of uc and us.

G)

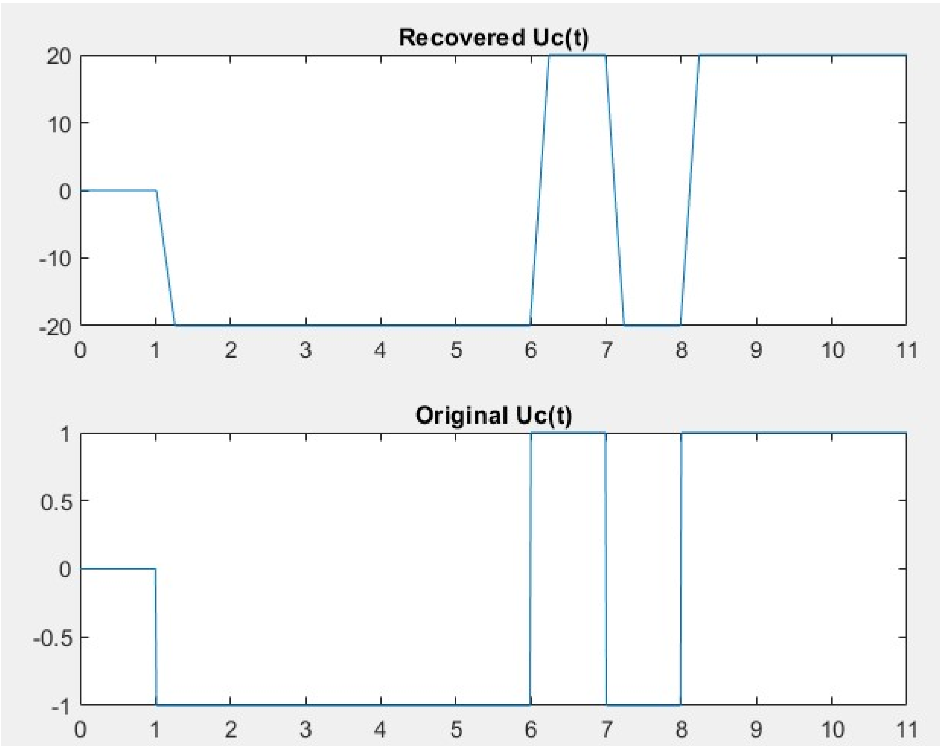
Here, we need to downconvert the signal by multiplying the signal with cos(2πfct+ theta) and sin(2πfct+ theta), and then pass it through a low pass filter I(0, 0,25). We take phase with which we multiply the values to be π/4. After that we manipulate the components using certain formula to get uc(t) and us(t)

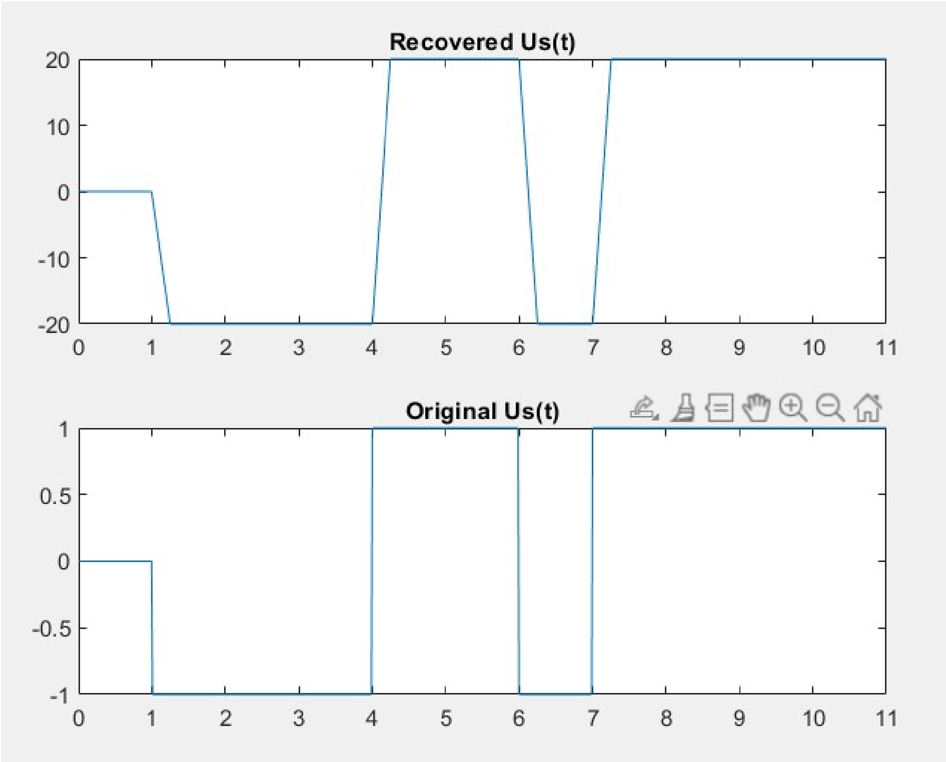
Parameters:

N=10

Fs=80 Hz

Plots:





Explanation:

We are able to recover uc(t) and us(t) from vc(t) and vs(t) by using the following formula

us(t)= vc(t)* cos(theta) – vs(t) *sin(theta);
uc(t)= vc(t) *sin(theta) + vs(t) *cos(theta);

We can simply derive this formula from the basic formulas we know during the process of downconversion.

Clearly we can observe that we are able to arrive at the initial signal and be able to tell the values of Bc and Bs clearly by looking at the following graph. So this process can be applied when we downconvert with a nonzero phase.