Communication Protocols - For connected objects – TP Report

Introduction to Software-Defined Radio: Analog demodulation of signals using GNURadio

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A Kind Note: This is an Individual Report. I was the only person who worked on this report.

Question 1

According to Fig. 2, considering that the received signal is similar to the transmitted one $r_{RF}(t) = s_{RF}(t)$and using (2) and trigonometric formulas, express the signal $\underline{r}_R(t)$ and $\underline{r}_R(t)$ in function of $S_R(t)$, $S_I(t)$, f_0,f_c

Solution 1

Given that $r_{RF}(t) = s_{RF}(t)$, which is a perfect transmission medium and operation of IQ demodulator, we need to multiply $cos(2\pi f_c t)$ and $-sin(2\pi f_c t)$ to the received signal to get the real & imaginary parts respectively. As a result,

The real part is

$\underline{r}_R(t) = S_{RF}(t) \cdot cos(2\pi f_c t)$ $\underline{r}_R(t) = \frac{1}{2}[S_R(t)(cos(2\pi f_0 t + 2\pi f_c t) + (cos(2\pi f_0 t - 2\pi f_c t)) - S_I(t)(sin(2\pi f_0 t + 2\pi f_c t) + (sin(2\pi f_0 t - 2\pi f_c t))]$

The imaginary part is

$\underline{r}_I(t) = -S_{RF}(t) \cdot sin(2\pi f_c t)$ $\underline{r}_I(t) = \frac{1}{2}[-S_R(t)(sin(2\pi f_c t + 2\pi f_0 t) + (sin(2\pi f_c t - 2\pi f_0 t)) + s_I(t)(cos(2\pi f_c t + 2\pi f_0 t) - (cos(2\pi f_c t - 2\pi f_0 t))]$

Question 2

If we take $f_c = f_0$ -translation in the baseband by heterodyning, what should be the characteristics of the h filters to get $r_R(t) = s_R(t)$ and $r_I(t) = s_I(t)$? You have to make a representation of $\underline{R}_R(f)$and $\underline{R}_I(f)$, after using the Fourier transforms. If you want to work in the time domain, a hypothesis must be clearly stated and checked.

Solution 2

Let, $\alpha = 2\pi f_c$ and $\beta = 2\pi f_0$

Given that $f_c = f_0$, henceforth it allows to isolate the real and imaginary part of the RF signal. In either cases, the translation of signal occurs at

1. At $2f_c$, high frequency harmonics occurs
2. At $0H_z$, harmonics occurs

If $\alpha = \beta$, then $cos(\alpha - \beta) = 1$ which implies that the real and imaginary part of the signal is duplicated. It is reduced at the baseband and also at $2f_c$ mentioned above.

Now, during the duplication, the signal is divided by 2 (With reference to Trigonometry formula), So to find the imaginary or real part, it is enough to pass a low filter with a cut-off frequency at $F = \frac{B}{2}\ Hz$ and a gain of 2 - between 0 and $\frac{B}{2}\ Hz$

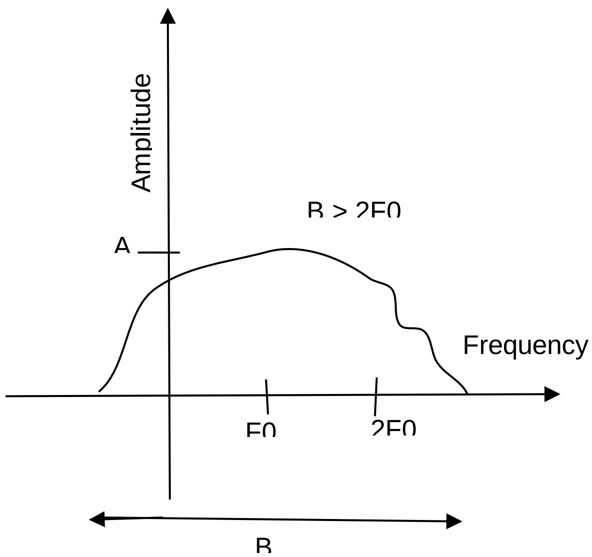
Question 3

Can the receiver presented in Fig. 2 work with wide-band signals? Explain.

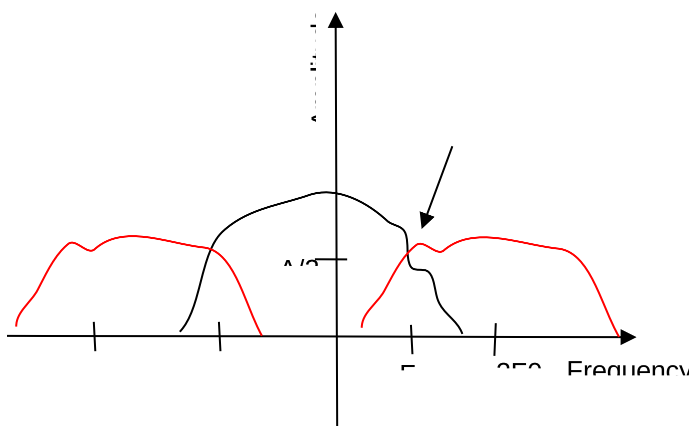
Solution 3

If the signal is of wide-band, there is a band overlap. During the frequency translation, the modulating signals in the case of modulation, is multiplied by a higher frequency $f_0$which is the carrier.

If $f_0$is not greater than B, the signal transmitted in the channel looks as depicted in the below image.



After the frequency translation, IQ demodulator looks as depicted in the below image. This is the very first stage.



When the signal is reduced to the baseband, the overlap occurs around f_0 between the two signals. This overlap makes it impossible to decode the signals hence with these theory it is evident that the narrow band signal is necessary.

Question 4
How must the sampling period T_e be chosen in order to recover $r_R(t), t \epsilon R$ from $r_R(k \cdot T_e), k \epsilon Z$ *?*

Solution 4

As per Shannon's theorem, sampling frequency must be at least 2 times higher than the maximum frequency of the signal considered. Therefore, we have $f_e = \frac{1}{T_e}$.

Reduction in baseband signal leads the signals to occupy a spectrum ranging $\frac{-B}{2}$ to $\frac{B}{2}$. Hence the maximum frequency $f_{max} = \frac{B}{2}$, which means the f_e must be greater than B for a functional sampling.

Therefore, $f_e > B$ which is equivalent to $T_e < \frac{1}{B}$.

Question 5
Why do not we interchange the stages of frequency transposition and analog to digital conversion?

Solution 5
Super heterodyne receiver is used to reduce the baseband signal by multiplying it with an adequate frequency to simplify the processing chain. It also includes the signal sampling provided by the Analog-to-Digital-Convertor(ADC).

If the maximum frequency of the RF signal is $f_0 + \frac{B}{2}$, the sampling frequency must be at least [2 * $f_0 + \frac{B}{2}$] which is considerate when listening on the VHF/UHF bands.

Hence the baseband can be achieved before sampling when the baseband is not reached in the initial stages.

Question 6
Supposing a real narrow-band signal

$$s_{RF}(t) = A(t) \cdot cos(2 \cdot \pi \cdot f_0 \cdot t + \varphi(t))$$
$$s_{RF}(t) = s_R(t) \cdot cos(2 \cdot \pi \cdot f_0 \cdot t) - s_I(t) \cdot sin(2 \cdot \pi \cdot f_0 \cdot t) , t \epsilon R$$

Expressing in frequential, then in temporal, its analytic signal and its complex envelop in function of f_0, knowing that $S_{RF}(f) = {S_{RF}}^*(-f)$

Solution 6
On Combining equations (4) and (5), we have

$$S_a(f) \ = S_{RF}(f) \ + \ sgn(f) \cdot S_{RF}(f)$$
$$S_a(f) \ = S_R(f - f_0) \ + \ j \cdot S_I(f - f_0)$$
$$S_a(f) \ = [S_R(f) + j \cdot S_I(f)] * \delta(f - f_0)$$
$$S_a(f) \ = (S_R(t) + j \cdot S_I(t)) \cdot e^{\ j2\pi f_0 t}$$

Assuming spectral envelop as $\phi \ = 2\pi f_0$ and the modulant as $A(t)$,

$$S_a(f) \ = A(t) \cdot e^{\ j\phi t}$$

The spectral envelop is as wide as the frequency deviation range configured in the VCO, the modulation block of the transmitter.

Second part:

Reception of frequency modulation(FM) broadcasting

Question 7
Present the role of each block used in the processing chain. Help: use the online manual.

Solution 7
QT GUI Frequency Sink :
According to the configured bandwidth, shows the frequency of the incoming signal.
File Source :
Opens stream samples from the source file.
Throttle :
Controls the number of samples that flows throughout the graph. One throttle suffices as it cascades the settings across all the flowgraph. Audio Sink block & sound card also acts as throttles and helps in controlling the frequency in which the sample files are being read.

Question 8
Specify the values of the missing variables in the characteristic function of the recording file.

Solution 8
As per the given theory, we set the given sampling frequency & frequency center values in the following variables

Sample_rate = 1.5MHz
Freq_center = 99.5MHz

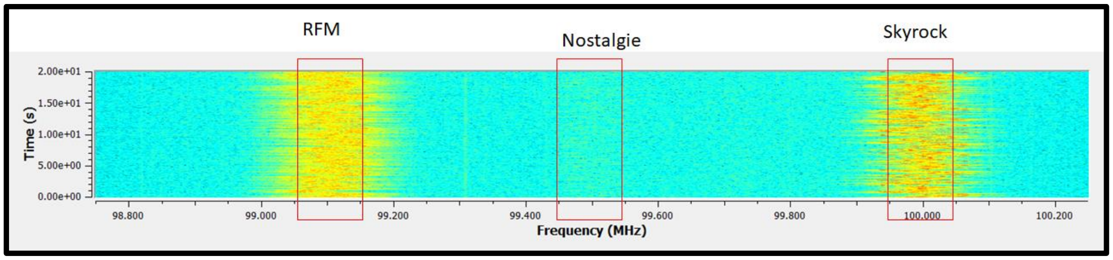
Also to note, variables of the sink blocks are purely visual and inappropriate settings of the sampling frequency and frequency center variables does not affect the spectrum. It's only the appropriate values entered gives the consistent display.

Question 9
How many frequency channels –to be noted L- do you observe? According to the allocation of frequencies in the FM band near Toulouse which stations are observed?

Solution 9
There are 3 frequency channels noted using waterfall - the names of the channels are RFM, Nostalgie, Skyrock.

Apart from these 3, exactly at 99.300 we see a low amplitude, no frequency signal which may correspond to some other signals unlike the FM.

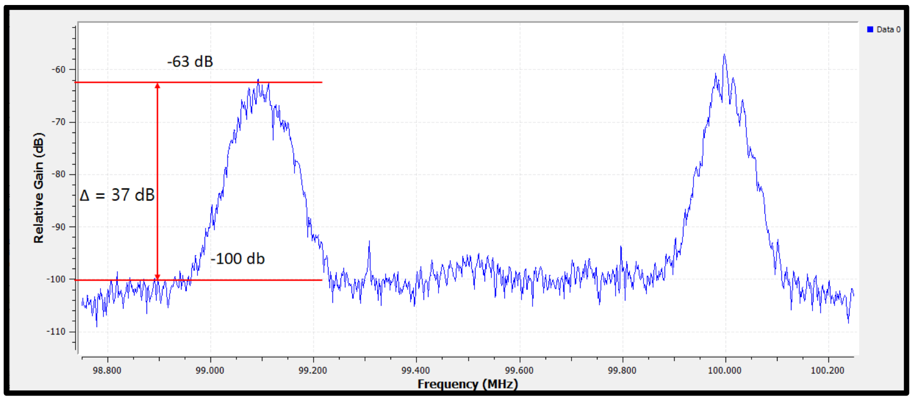
Both RFM & Skyrock looks alike and the Nostalgie looks less effect, with lower values. Nevertheless, they would be easily distinguishable by listeners.



Question 10
What is the measured signal-to-noise ratio in decibel? Do you think that is enough to be able to demodulate the signal?

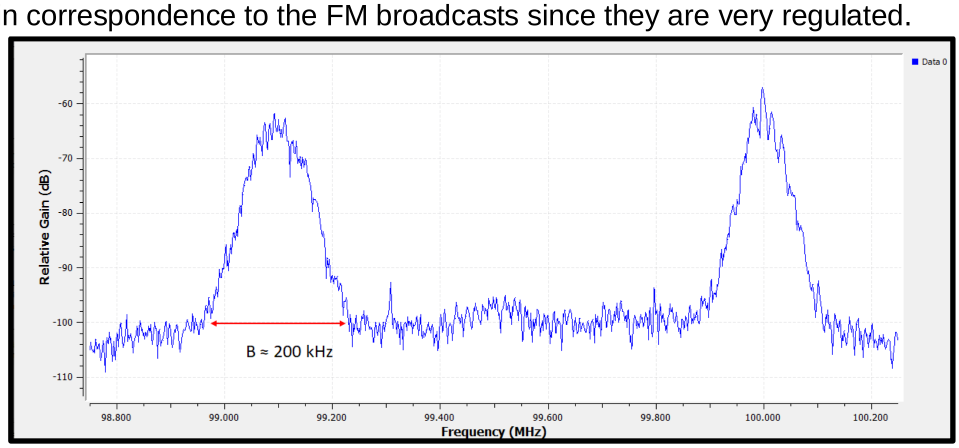
Solution 10
From the below image, for the RFM, we could see that the Signal-to-noise ratio(SNR) is 37decibel from the noise floor of the receiver to the maximum amplitude of the FM signal. This suffices to demodulate in most cases. Since we have an analog signal, demodulation can always be done as seen in the below image. There is no sampling error happened during decoding. The final block of the flowgraph is human ear, the overall sound will be greater provided the SNR value is higher.

Hence, there is no actual “too low” SNR value such that the demodulation cannot happen for any of the FM signal.



Question 11
What is the approximate bandwidth of a channel?

Solution 11
As per the theory, the conventional bandwidth at -3dB should be very low. But in practical, this is not true in correspondence to the FM broadcasts since they are very regulated.



Question 12
What are the frequency offsets needed to center each channel?

Solution 12
Since Nostalgia station is the center frequency, it is brought back to the baseband.

FM Channel Name	***FM Channel Frequency***	***Difference(Δ)***
RFM Toulouse	99.1 MHz	+400 kHz
Nostalgia Toulouse	99.5 MHz	
Nostalgia Toulouse	99.5 MHz	-500 kHz
Skyrock	100.0 MHz	

Question 13
What happens if the frequency offset is higher than the sampling frequency F_e?

Solution 13
The 2π- periodicity of the complex potential explains the circular permutation of the spectrum which can be seen in the sink blocks. The signal sent by the local oscillator can be expressed as follows:

$$S_{local}[k] = S_{local}(\frac{k}{f_e}) = e^{j(2\pi f_{local}\frac{k}{f_e})}$$

We can also see that,

$$f_{local} = f_e \preccurlyeq=\succ S_{local}[k] = e^{j(2\pi f_e\frac{k}{f_e})} = e^{j(2\pi k)} = 1$$

Hence, the local frequency of the oscillator is equal to the sampling frequency, the transmitted RF signal remains unchanged. If the local frequency is higher, a cycle is started again.

The final real frequency displacement which is based on the f_{local} module on the f_e is

$$f_{shift} = f_{local}\ [f_e]$$

Question 14

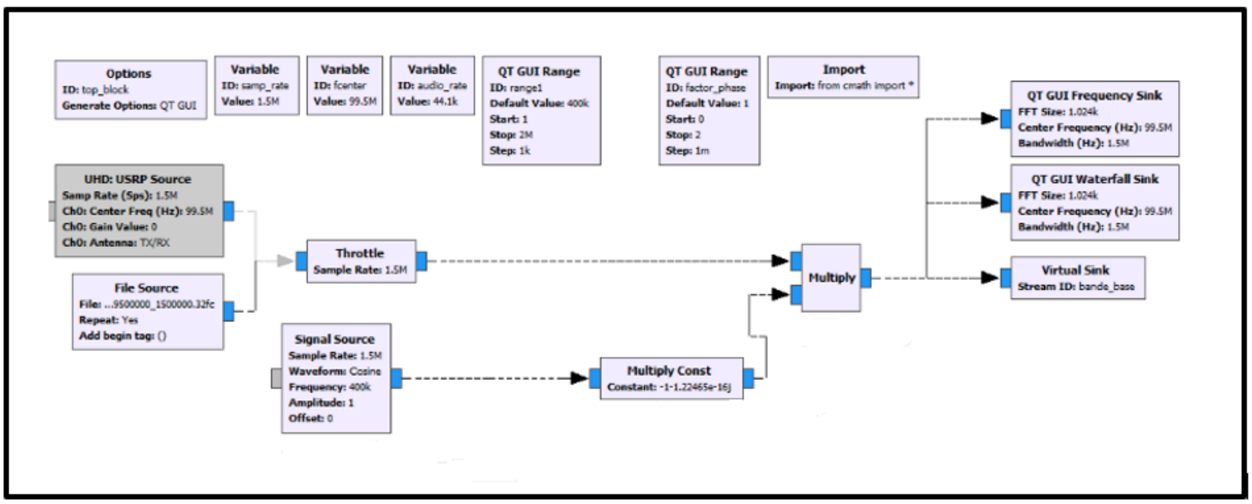
What are the low-pass filter parameters, as well as those of the frequency analyser at the output of the filter?

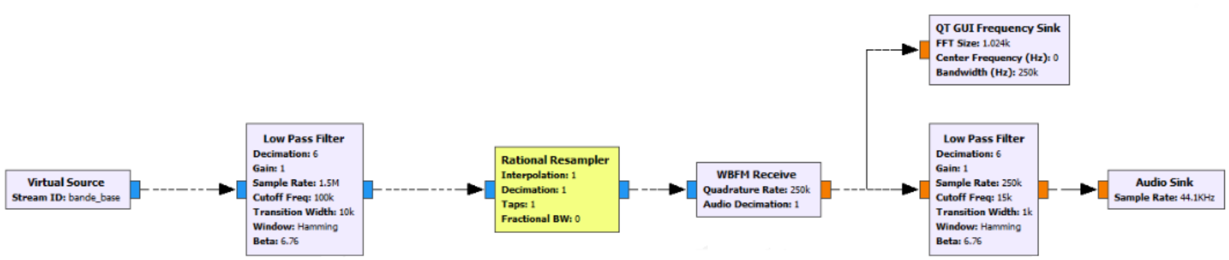
Solution 14

With the signal at its center 0, the maximum deviation is +/- 100kHz.

Now, we choose 10kHz which is a 10% of the cutoff frequency with width transition - to get a decent slope value, which has also a reasonable number of taps, considering the speed of calculation and efficiency of a filter.

We need to remove the decimal places for the incoming signal by 6 which would not degrade our payload. This is required because we need to represent a discrete way without losses in a signal. Also, increasing the decimal places furthermore will increase the processing speed but has practical difficulties.





Here we need not add a rational resampler block, which is skipped here as we can see two decimations of 6 following each other which gives 36 samples at the input of the audio sink.

3. Frequency demodulation and restitution

Question 15
Using the Carson rule, check that the bandwidth of the channel measured in the previous part confirms the theory.

Solution 15
Maximum deviation = 75kHz
Maximum frequency of the modulant = 53kHz
By applying the Carson rule, B = 2 * (53+75) = 256kHz

This band has 98% of the signal energy still, it is not the traditional band with 3dB bandwidth. As a Rule of thumb, it is normal to underestimate the true width of the useful bank. Here, compared to approx 200kHz, which has the same order of magnitude. Hence the rule is confirmed.

Question 16
Define the value of k_f, with b[k] a complex noise term introduced by the propagation channel as well as by the transceiver itself.

Solution 16
From the previous derivation of $S_a(f)$in the question 6 and , $\phi(t)$ from the variant of equation(10), we have the following:

$$S \quad (f) = A(t) \cdot e^{\,j\phi t}$$

$$\frac{\Delta f}{max(|m(t)|)} \cdot \int_{-\alpha}^{t} m(u) \cdot du$$

In discrete mathematics, integral is equal to sum, equation discrete yields

$$s[k] = A[k] \cdot e^{\,j\phi t}$$

$$s[k] = A[k] \cdot e^{\,j\frac{\Delta f}{max(|m[k]|)}\Sigma_{i=0}^{k} m[i]}$$

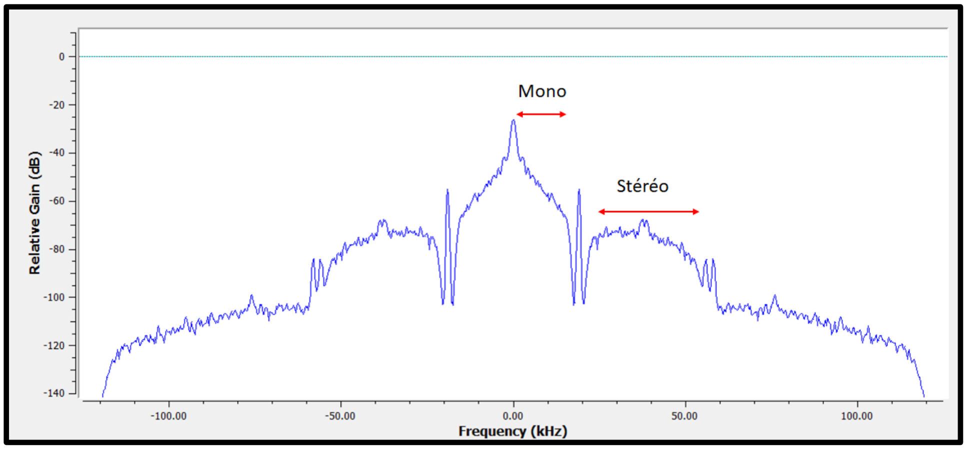
we now have, $k_f = \dfrac{\Delta f}{max(|m[k]|)}$

To get the equation(12) from the statement, we need to add the complex noise $b[k]$

Question 17

Plot the spectrum of the demodulated channel and compare with the Fig. 6.

Solution 17



This is between the mono and stereo band and it is the same as seen in the figure 6 (Stereophonic composite signal before the frequency modulation)