ELEC6021 (EZ619) Communications Assignment I

October 7, 2005

Submission Details

You should work through the given exercises and provide documentation that your implementations work satisfactorily. This includes annotated graphs that clearly indicate your results to the questions below, and the provision of Matlab m-files on a DOS formatted floppy disk (or list of Matlab m-files).

Matlab provides very good on-line help facilities. You should spend sufficient hours of work on your assignment, of which a good proportion is set aside for learning about how to use Matlab. The written assignment should be **concise** and contains a set of **clearly labelled figures**.

1 AM/FM Analogue Modulation

1.1 AM Modulation

Figure 1 shows the schematic of a transmission using amplitude modulation (AM) of an input signal $x(t)$ and demodulation to obtain a received signal $\hat{x}(t)$.
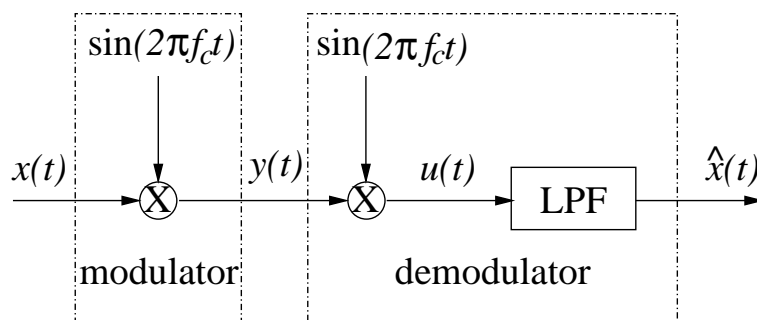

Figure 1: AM modulation and demodulation.

a) **Input Signal.** Simulate a normally distributed random signal $x(t)$ bandlimited to an upper frequency of $f_{\text{upper}} = 10\text{kHz}$. The simulation of the analogue signal $x(t)$ can be performed using a sufficiently high sampling rate of e.g. 2MHz, over a simulation interval of here $T = 0.1\text{s}$. Thus, the first steps are

```
>> fs = 2000000;
>> t = (0:1/fs:0.1);
>> xw = randn(1,length(t));
```

Look up any unknown Matlab commands and their parameters using the online help facility,

e.g.

```
>> help randn
```

to obtain information on how to create a normally distributed variable. The signal `xw` resulting from above is white. You may check this by viewing the spectrum using the power spectral density (PSD)

```
>> psd(xw,8192);
```

Now, using the command `fir1`, design a lowpass filter that bandlimits the signal to the upper frequency component of approximately 10kHz. Hint: as a filter order, $N = 1023$ is a suitable choice. The filtering itself can be performed using the command `filter`, in order to obtain a simulated version `x` of the signal $x(t)$.

Use a PSD plot to provide evidence that the obtained spectrum of `x` is as desired. To save a Matlab figure into a file for printing or inclusion in a word processor, the command `print` can be used.

In written assignment, show in graphs the PSD of the white noise (signal) xw and the PSD of the bandlimited transmitted signal $x(t)$.

b) Modulation. Create a carrier with frequency $f_c = 250$kHz and modulated the signal $x(t)$, thus obtaining $y(t)$ as shown in Fig. 1. Observe the PSD of $y(t)$ and compare it to the PSD of $x(t)$.

In written assignment, show in graph the PSD of the modulated signal $y(t)$.

c) Demodulation. Implement a demodulation of the signal $y(t)$ to $u(t)$ as given in Fig. 1 and define an appropriate lowpass filter (LPF) in order to retrieve the input signal, $x(t)$. When designing the LPF, a filter order of $N = 23$ should be sufficient for your purposes.

Specify your selected filter parameters such as the cut-off frequency, and provide a PSD of the output, $\hat{x}(t)$.

In written assignment, show in the same graph the PSD of the signal $u(t)$ and the magnitude response of your designed LPF, and show in graph the PSD of the demodulated signal $\hat{x}(t)$.

d) Received Signal. Verify that the output $\hat{x}(t)$ is essentially a delayed copy of the input signal $x(t)$ with delay τ, i.e. $\hat{x}(t) = x(t - \tau)$. Verify this experimentally, and determine and justify the incurred delay τ.

Hint: The overall system gain (from $x(t)$ to $\hat{x}(t)$) should be kept to unity. For example, the factor $\frac{1}{2}$ in demodulation of Figure 1 should be compensated.

In written assignment, show in the same graph the time waveforms of the transmitted signal $x(t)$ and the received and demodulated signal $\hat{x}(t)$. To see that a small delay is incurred, you should zoom in a small segment of the time waveforms.

1.2 FM Modulation

Implement a frequency modulation according to

$$y(t) = \sin\left\{2\pi(f_c + \Delta F \cdot x(t)) \cdot t\right\}$$

with $\Delta F = 1$Hz and carrier frequency $f_c = 250$ kHz as above.

a) Spectra. Plot the PSD of your FM modulated signal $y(t)$ and compare it to the results for the AM modulated signal. What are the differences and similarities?

In written assignment, show in graph the PSD of the FM modulated signal $y(t)$, and show in another graph the PSD of the FM modulated signal superimposed on that of the AM modulated signal with a properly chosen normalized frequency range.

2 Digital Modulation

Below, the transmission by means of quadrature amplitude modulation (QAM) of a digital signal in the form of a bit stream is to be developed step by step. In a first step, this bit stream is converted to a stream of QAM-symbols, as seen in Fig. 2. This signal is upconverted and filtered in the transmitter prior to the actual modulation, as given in Figs. 3 and 4. With inverse operations also shown in Figs. 3 and 4, a demodulation and retrieving $x[n]$ can be achieved.
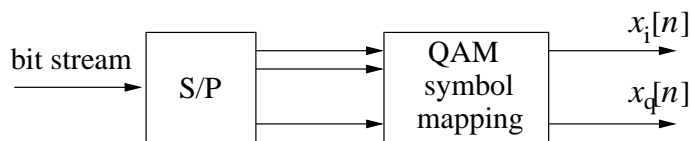

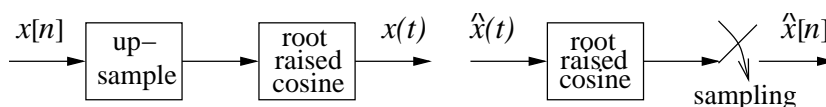Figure 2: Mapping a bit stream onto a QAM-symbol stream.

Figure 3: Transmit/receive filtering and "digital-to-analogue"/"analogue-to-digital" conversion.
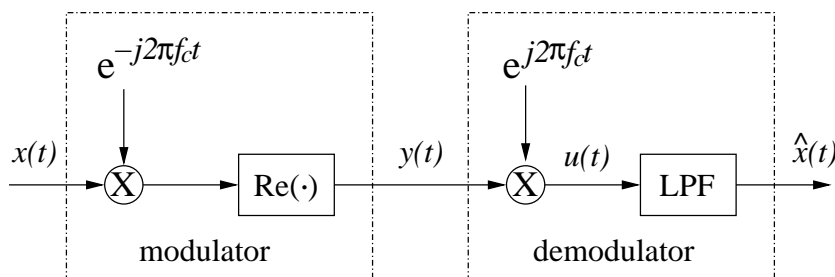

Figure 4: QAM modulation and demodulation.

The carrier frequency is again $f_c = 250$ kHz, and symbol rate 10 ksymbols/s.

a) Binary Input. Create a binary input sequence with 16000 bits.
Hint: this can be easily achieved by appropriately rounding the samples of a uniformly distributed random signal, obtained with the function `rand`.
Mapping to 4-QAM symbols. Write a function that turns a binary input sequence into a stream of QPSK (4-QAM) symbols, $x[n] = x_i[n] + j \cdot x_q[n]$. with $j = \sqrt{-1}$, as shown in Fig. 2.
In written assignment, show in graph the constellation diagram of the 4-QAM symbols, and show in graph the time waveform of $x_i[n]$ (or $x_q[n]$).

b) Transmit and Receive Filtering. We simulate the conversion to an analogue signal as depicted in Fig. 3 by selecting a high sampling rate of 2MHz, to which the signal can be upsampled using

```
>> x = upsample(xqam,100);
```

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if the vector `xqam` contains the samples of the signal $x[n]$ and the symbol rate is 10kHz. Using the command `firrcos`, design a square root raised cosine filter with cut-off frequency $f_{cut} = 10$kHz, bandwidth 10kHz, and order $N = 255$. Using the `filter()` function, filter your upsampled signal. What does the PSD of the filtered signal look like?

The same square root raised cosine filter can be used as a receive filter.

In written assignment, show in graphs the magnitude responses of the transmit filter, and the combined transmit and receive filter, and the PSD of the filtered signal $x(t)$.

Hint: The overall system gain (from $x[n]$ to $\hat{x}[n]$) should be kept to unity.

c) Modulation and Demodulation. Implement a QAM modulation and demodulation as shown in Fig. 4. The real part operation can be performed using `real`, and the lowpass filter LPF needs to be defined appropriately and can thereafter be designed by `fir1`.

In written assignment, show in graphs the PSDs of the modulated signal $y(t)$ and the demodulated signal $\hat{x}(t)$.

d) Receive Filtering and Symbol retrieval. After filtering with the receive filter, show that the original data can be retrieved by appropriate sampling of the signal (you may for example show a short data segment, highlighting at which time instances you would sample the signal).

In written assignment, show in graph the time waveform of the transmitted real part of symbols $x_i[n]$, and in another graph the time waveforms of $\hat{x}_i(t)$ and $\hat{x}_i[n]$. Alternatively, you may show the waveforms of imaginary parts of the signals.