Submission deadline is Sunday, 13.01.2019, 23:59

Mandatory oral presentation of results is Monday, 14.01.2019, 14:00 (max. 5 slides)

Signal Processing – Digital Part

Assignment 2018-19

Remarks:

- In order to submit the assignment, the soft copy report and LabView files have to be submitted via email to Prof. Dr. Ulf Witkowski (email: witkowski@fh-swf.de) within the due date.
- Email subject has to be stated with your surname as 'SP-Digital_2018_Surname'.
- This assignment score is 30% of Signal Processing module.
- If a student gets less than 40% of the assignment score, the student will fail.
- This assignment is prohibited to do in team.
- If plagiarisms have been found, both the original and the copied versions will be noticed as fail, automatically.

Digital Filter Design – Introduction

Different types of digital filter can be realized by placing poles and zeros in the z domain. As an example putting a zero on unity circle at zero angle will cancel out the DC frequencies which results in having a high-pass filter. Likewise a zero on unity circle at 180° will cancel out higher frequencies resulting in a low-pass filter. Having zeros on both points generates a band-pass filter. The poles positions decide about the cut-off frequency and bandwidth.

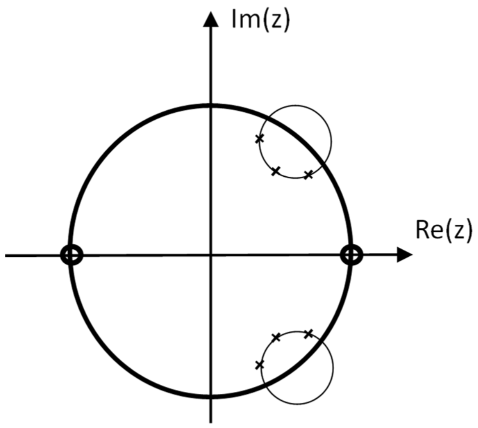


Figure 1: Example of a band-pass filter. Poles are close to the unity circle, and a zero pair is located on the real axis.

Phase modulation / demodulation – Introduction

Part of this assignment is the demodulation of a binary phase-modulated signal. This method is also known as binary phase shift keying (BPSK). This method is very suitable for transmission of binary messages and it has been used in digital devices such as wireless telephones for many decades. For a phase modulated signal, the carrier phase is modified according to the value of a (low frequency) binary message signal. The process of the modulation and demodulation is shown in Figure 2.

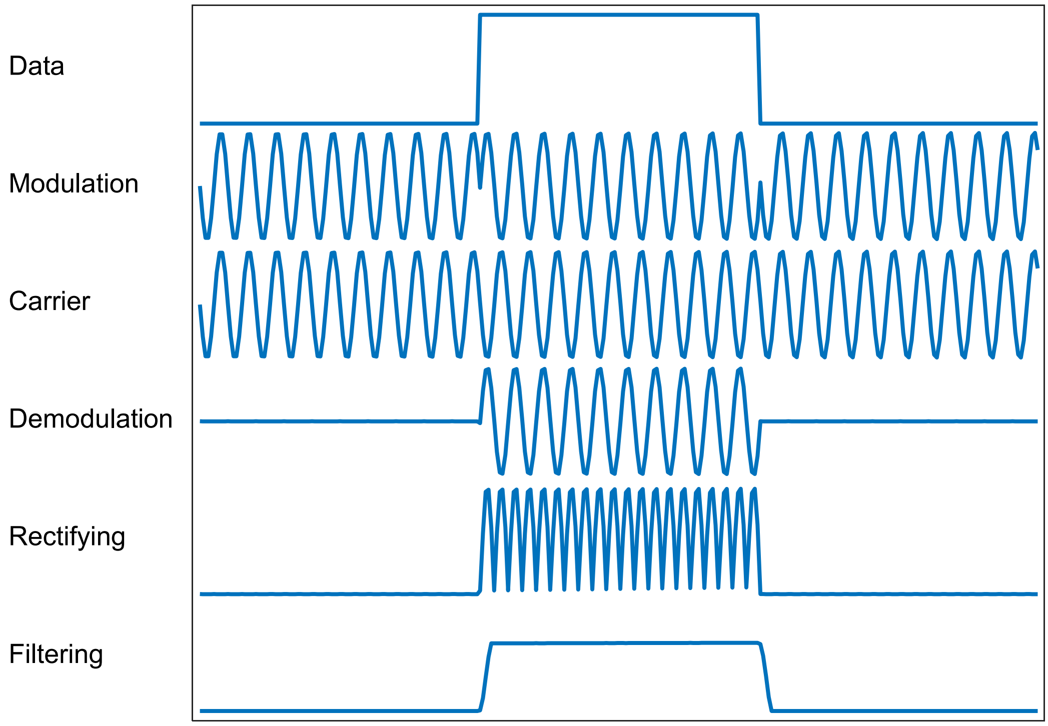


Figure 2: Process of BPSK modulation and demodulation: A message signal (top) is shown after modulation (2nd), demodulation (4th), rectifying (5th) and finally filtering (Bottom)

In order to modulate a signal according to BPSK method, a data containing bit zero is represented as no shift in carrier signal and a data containing bit 1 is represented as shift of 180° (Fig. 2.b).

In order to demodulate this signal, many methods are available. In its simplest form, you can use a sine wave as carrier signal with the same frequency as the modulated signal and mix the carrier signal with the modulated signal. This results in extraction of the data part in the form shown in Fig. 2.d. Typical challenges of this part is synchronisation of the two signals as any phase difference would ruin the demodulation concept.

Using common methods of rectification and filtering, one is able to reconstruct the data signal from the results of previous step. A suitable filter type for this step is a low-pass filter which retains the envelope of the signal and removes the high frequency elements.

The data signal used in this assignment is Morse code which is representing numbers in the form of dots and lines. The dots are simulated using a short pulse and the lines with long pulse. You can also find more information about the Morse code in the lecture slides.

Tasks

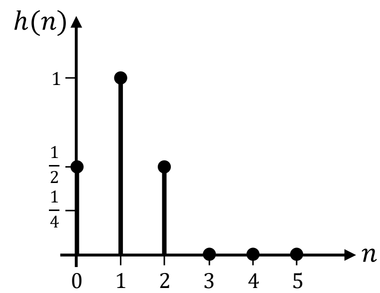
In this first part sampled signals and time discrete systems have to be analysed.

Task 1: Calculate the frequency response (transfer function $H(z)$ in z domain) of the following systems. The systems are characterized by their unit pulse response $h(n)$.

a) $h(n) = \frac{1}{2}\delta(n) + \frac{1}{3}u(n) - u(n-1) \cdot \left(\frac{1}{2}\right)^n$

b) $h(n) = \delta(n)\left(\frac{1}{4}\right)^{n-1} + \frac{1}{2}u(n-2)\frac{1}{2^{n-1}}$

Task 1c: The pulse response of a system has been recorded as shown:



a) Calculate the transfer function $h(n)$.

b) Calculate the transfer function $H(z)$ in z domain.

c) Calculate the poles and zeros of the transfer function and draw the pole-zero diagram.

d) Calculate the magnitude transfer function $|H(\omega)|$

e) What is the DC gain of the related filter?

e) Draw the magnitude diagram for $|H(\omega)|$ in range of 0 … π.

f) What type of filter is realized by the given system?

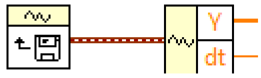
Task2: In the second part of this assignment, a BPSK modulated signal has to be demodulated and decoded. At the first step of this assignment you need to analyse the given signal and evaluate the major frequencies present in signal. The sample frequency of the signal is 40KHz and the signal is modulated using a 3KHz carrier signal. Another task is to demodulate the signal but before that you may need to filter the noises available in the signal with a filter. For removing the noise, a digital filter has to be designed using theoretical calculations.

Use a simple signal carrier for demodulating the signal. As you are implementing the carrier signal in LabView it is expected that no phase shift will occur therefore no complex synchronization technique is required for demodulation. The message signal that has been used for signal modulation is a Morse code in the form of a digital signal. The used Morse codes only include some numbers so you don't need to spend any effort to implement the whole Morse codes such as alphabets. After demodulation a decoding of the resulting signal has to be realized. The whole process should be implemented using LabVIEW and the resulted signal should be demonstrated at each step of demodulation and decoding. A final step is the decoding of the Morse coded binary signal. The content of the data stream is ASCII codes, i.e. the output result at the end is a string.

As the message signal is a binary signal, just two dominant phases are available in the BPSK modulated signal. The 180° shift of phase codes a '1' and the 0° of phase codes a '0'.

Detailed tasks:

Task 2a: Analyse the frequency modulated signal in LabVIEW. The required file "Assignment.lab" is available in the download area. You can read this file in LabVIEW using the "read waveform" block:



The file contains both the time data and the amplitude data. Evaluate the signal in both time and frequency domains and determine the dominant frequencies and phases.

Task 2b: Determine the frequency specification and required characteristics of a suitable filter that should be used to improve the noisy signal. Briefly explain your approach and comment your findings.

Task 2c: Design the digital filter according to previous step using either of the two following techniques:

1) In this approach, the filter has to be designed with pole and zero placement and the coefficients have to be calculated based on the poles and zeros. The transfer function should be in the form of $H(z) = \frac{(z-z_1)\dots(z-z_n^*)}{(z-p_1)\dots(z-p_n^*)}$.

2) As a second approach, the bilinear transform has to be used to design a filter. Start with the design of an analogue filter. Afterwards it should be converted to digital form.

Task 2d: Now the designed filter has to be implemented in LabView. For implementing the filters, a predefined building block "IIR Filter.vi" can be used. The manually calculated coefficients have to be applied as reverse and forward coefficients to the building block.

Task 2e: Simulate the frequency response of the designed filter using LabVIEW. From the simulated frequency response: Are the filter characteristics (slope) sufficient for the noise reduction?

Task 2f: Generate a carrier clock using a sine wave with the same frequency as the modulated signal. Use this signal to demodulate the signal into the form of Fig. 2.d.

Task 2g: Use a rectifying method to reach to the signal given in Fig. 2.e.

Task 2h: Use a second digital filter according to the methods provided in the task 2c in order to reconstruct the Morse signal. Evaluate the required filter characteristics for this design and implement the filter in LabView and finally evaluate the results.

Task 2i: Decode the demodulated signal visually (manually, i.e. no coding or implementation of automatic detection is required). The underlying coding of the digital signal follows the Morse code (Only numbers but not alphabets). Provide the final coded number as solution.

Task 2j: (Optional task with extra bonus) Develop a decoder in LabView that automatically looks for the Morse code and extracts all the frames. The code block should present the final number as a series of characters (string) visible in a text box.

Task 2k: Document all calculations and code implementations and briefly discuss your results.