Teaching Signals to Students: a Tool for Visualizing Concepts in Digital Signal Processing (DSP)

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Abstract

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Few Engineering students at Uppsala University choose to pursue their studies in signal processing, due to the level of difficulty within the subject. The main underlying problem is the students’ trouble with understanding the behaviour of analog and digital filters, as well as the concept of stability. In this project, a system that allows the teacher to visualize multiple concepts in signal processing was developed. The system consists of analog filters, digital filters implemented in a digital signal processor, and software for simulating unstable filters. The goal with this project is to give teachers the possibility to make students intuitively understand the subject, and improve the overall learning experience for the students.
Sammanfattning

Acknowledgements

First and foremost, we would like express our deepest gratitude to our external stakeholder Steffi Knorn, who guided us; who was always available to answer our questions and exchange ideas. Secondly, we would like to give a special thanks to Ping Wu, who so generously allowed us access to his framework for signal processing, and to Pouya Ashraf, Linnar Billman and Adam Wendelin for paving the way and helping us avoid the obstacles that they had faced. Lastly, we would like to thank Uwe Zimmerman, for his expert help with soldering.
DT: Discrete Time
FIR: Finite Input Response
GPIO: General Purpose Input Output
HP: High-pass (filter)
IDE: Integrated Development Environment
IIR: Infinite Input Response
LP: Low-pass (filter)
MCU: Microcontroller Unit
OP-Amp: Operational Amplifier
ZOH: Zero-Order-Hold
1 Introduction

Signal processing is an important part of modern technology, and a crucial topic in many diverse areas such as seismic data processing, communications, speech and image processing, defence electronics, consumer electronics, and consumer products [3]. Consequently, aspiring engineers will not succeed in developing new technology, or even further develop existing one, if they are not sufficiently taught within the area of signal processing.

At Uppsala University efforts have been made to ameliorate the learning process for the students, by employing a teaching tool for analog filter visualization (as well as some digital signal processing) [6]. Indeed, teaching tools of different kinds have proven to be beneficial for students when it comes to understanding the behaviour of signals [7, 14, 19, 20]. The intention of this project is to further develop the tool created by P. Ashraf et al [6], so that it handles digital signal processing and stability as well as analog signal processing.

To display the behaviour of digital signals we will be visualizing the two different types of digital filters: Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters. In addition to this, we have visualized filter instability; how a digital signal behaves when a filter becomes unstable.

2 Background

Signal and systems is a progressive area that is continuously used in the development of modern technology. People may encounter signal processing in their everyday life, for example: electrical signals representing sound or music and image processing describing images in digital cameras and computers.

Signal processing is thereby an important part in most engineering educations. Despite the importance of this topic, many students find this a very difficult subject [7, 11]. Few students choose to pursue their education within this area, because of the perceived level of difficulty [8, 20]. A random sample at Uppsala University, where this project was done, can be seen in Figure 1. These courses were chosen at random, from all Engineering courses a student may add to their Masters education\(^1\) in addition to the Master program specific courses at Uppsala University.

\(^1\)Specifically, Masters in Engineering
2 Background

Figure 1: A bar chart displaying the statistics of applications to engineering courses at Uppsala University, Sweden, autumn 2016. There is a significantly lower number of students applying to courses involving signal processing, the three first bars from the left in the diagram.

A tool for illustrating how analog signals behave within signal processing already exists at Uppsala University[6]. However, this is only half the content of signal processing. We would like to further develop this so that this tool is also able to illustrate digital signals.

2.1 The external stakeholder

Steffi Knorn, head of several courses involving signal processing, requested a further development of this teaching tool. When using the tool for analog signals the user may observe how an analog signal behaves when different filters are applied to it. Knorn has requested that the first system would maintain all functionality and that our project is built as an extension to the tool. She also requested a simulation for instability to built, in order to be able to show students how different unstable filters behave.
2 Background

2.2 Filters

Filters are used in signal processing to remove unwanted parts of a signal. In other words, filtering is the removal of some frequency components in order to remove noise and interference.

Ideally, a filter has the gain of one (also known as unity gain) for the band of frequencies that it passes, and zero for the ones it does not (see Figure 2). These two band of frequencies are known as pass band and stop band, respectively. In this ideal scenario there would also not be any ripples i.e. fluctuations of the gain. Unfortunately, it is impossible to design ideal filters, since the impulse response would have to be infinitely long in time, as well as non-casual [15, chapter 2].

![Figure 2: The transfer function $H(w)$ (frequency domain) of ideal low-pass filter.](image)

Generally, there are at least a few ripples in the stop and/or pass band, and there is a slope from the pass band to the stop band (see Figure 3a). The steepness of this transition (of the slope) is called roll-off. The roll-off and the amount of ripples can be influenced by the design type and the order of the filter. The design type simply describes a method (usually a polynomial function) that can be used to design a filter, and the order of a filter is determined by the number of poles or taps (more on taps in section 2.2.2).

There are mainly four different passband shapes, as can be seen in Figure 3: low-pass (LP), high-pass (HP), band-pass (BP) and band-stop (BS). All filters are said to have at least one cut-off frequency; this is defined as the frequency for which the gain is $\frac{1}{\sqrt{2}}$ [20, chapter 4.03, p 85]. It is easy to compare how the different filters behave if one imagines that they all have the same cut-off frequency (see Figure 3).
2 Background

Figure 3: LP filters pass frequencies below the cut-off frequency, HP filters above it. BP and BS filters passes frequencies and stops frequencies within a specific range (band), respectively.

In both Figure 2 and 3, the magnitude of the transfer function $H(\omega)$ is plotted. When talking about filters, one usually talks about transfer functions, but the impulse response $h(t)$ (which is $H(\omega)$ in time domain) is important as well, particularly when it comes to digital filters (more in section 2.2.2) and stability (more in section 2.4).

In this project, all filters are implemented as Butterworth filters. This is because the Butterworth filters do not have ripples in the passband, something that is thought to make them more accessible to students who are not familiar with filters. Elliptic filters and Chebyshev filters were considered, but were dismissed as they both have ripples in the pass band and/or stop band [17, chapter 15, p. 676-677].

2.2.1 Analog filters

Historically, analog filters were the first kind of filter designed, and were as such implemented with the use of analog electrical components. In general, analog filters are implemented using passive components like resistors, capacitors and inductors, as well as active components such as operational amplifiers (OP-amps) [6]. Two different analog filters, filter A and filter B, can be seen in Figure 4, with the first consisting of only passive components, and the second consisting of both passive and active components.
The main advantage of filter B is the filter’s ability to have unit gain in the whole pass band, something that filter A cannot do. This is because, while the gain of filter B can be set with a resistor \( R \) in parallel to a capacitor \( C \), the gain of filter A is dependent on the voltage drop across \( R \). This makes unity gain for filter A impossible if any load is applied, which it always is when the filter is used, since its output is then connected to something else.

Additionally, filter B has a lower output impedance, meaning that one can drive circuits that require more current with the output from filter B than from filter A. All current supplied from filter A has to pass through the resistor \( R \), while in filter B the output current is supplied by the OP-amp. Consequently, filter B will have a lower output impedance, and can supply more current.

### 2.2.2 Digital filters

The main difference between analog and digital filters is that analog filters work with signal in continuous time (CT) while digital filters work in discrete time (DT). In addition to this, there are some characteristics that differ; these can be seen in table 1.

<table>
<thead>
<tr>
<th>Analog</th>
<th>Digital</th>
</tr>
</thead>
<tbody>
<tr>
<td>Always none-linear phase shift</td>
<td>Can have linear phase shift</td>
</tr>
<tr>
<td>Drift due to component variation</td>
<td>No drift due to component variation</td>
</tr>
<tr>
<td>Adaptive filters difficult</td>
<td>Flexible, adaptive filters possible</td>
</tr>
</tbody>
</table>

Table 1: A short summary of how digital and analog filters are different [18, chapter 2, p. 19].
2 Background

There are two major digital filter types: finite impulse response (FIR) and infinite impulse response (IIR). FIR filters are determined by having an impulse response of finite width, while IIR filters by having an impulse response of infinite width\(^2\). In other words, FIR filter expresses each output as the weighted sum of the last N inputs, while the IIR filter expresses the output as a linear combination of both previous inputs and outputs. The recursive characteristic of the IIR filter is what gives it its name.

The order of IIR filters are determined by the number of poles, while the order of FIR filters are simply the number of inputs used. These inputs are usually called taps, and FIR filters are often talked about in terms of number of taps rather than order.

2.3 Reconstruction filter

The process of recovering a signal from a number of samples is called reconstruction, and is in general done using a digital to analog-converter (DAC). A common method of reconstruction is called zero-order-hold (ZOH) reconstruction, in which the DAC holds the value of a sample in the beginning of a sampling period for the entire period, until the next period \([18, \text{chapter 7, p. 350}]\).

2.4 Stability

Another important aspect of this project is stability, in particular BIBO stability. “BIBO” stands for “bounded input, bounded output”, with “bounded” meaning that the magnitude of the signal is less than some final value \(M < \infty\). BIBO stability indicates that for any given bounded input, the output will be bounded as well. This can be written as:

\[
\int_{t=0}^{\infty} |h(t)|dt < \infty \quad (1)
\]

\[
\sum_{n=1}^{\infty} |h[n]| < \infty \quad (2)
\]

With (1) being the requirement for CT, and (2) for DT. Two consecutive requirements for this are:

\(^2\)Note that this is not the same as an impulse response being infinitely long in time
Where $H(s)$ and $H[z]$ are transfer functions in CT and DT, respectively.

Unstable systems are unreliable, e.g., when a system is unstable, the output of the system may be infinite even though the input to the system was finite.

Since the output of FIR filters is always finite, FIR filters are guaranteed to be stable. However, they need to be of significantly higher order than IIR filters to achieve similar performance.

### 3 Purpose, aims, and motivation

A summary of the project’s aspirations and objectives are presented in this section, along with the motivation behind it.

#### 3.1 Purpose

Last year (2016), P. Ashraf et al created a teaching tool for visualization of the behaviour of signals put through analog filters (LP, BP and HP) [6]. The tool could also be used to visualize reconstruction of a digital filter by ZOH. The purpose of this project is the same as that project: “to help teachers in courses containing signal processing and analysis”.

This project further develops this teaching tool to include digital filters, to ensure that teachers are able to fully visualize digital signal processing (DSP), and instability.

#### 3.2 Aim

This project aims to add digital filters as well as visualization of instability to the tool built by P. Ashraf et al [6]. The digital filters are intended to match the analog filters in

3 The degree of the numerator does not exceed the one of the denominator.
terms of gain when the same cut-off frequency is used. Furthermore, just as the analog filters, the digital filters can be toggled. This is so that the teacher can both demonstrate DSP through digital filtering, as well as comparing the behaviour of analog and digital filters.

3.3 Motivation

The concepts in signals and systems are all built on abstract mathematical constructs, and studies [7, 11, 12, 13, 14, 20] show that students struggle with completely understanding them. In general, students find convolution and the relation between time/frequency domain the most difficult - both of which are prevalent when it comes to filtering.

The kind of organic understanding that can be achieved when it comes to abstract concepts such as accelerated reference system is usually built on the use of everyday examples (in this case “you are sitting in an accelerating car”). This is hard to attain when it comes to concepts in the subject Signals and Systems, as the average student does not encounter this on an everyday basis. A teaching tool such as the one in this project gives the students a kind of real life example, as well as a more hands-on learning experience, which is something multiple studies [7, 14, 19, 20] have shown benefits the students.

During our research we found that many of the available systems focus on digital signal processing, with only one including analog (more details in section 4). Furthermore, while many tools present the poles/zeros of the systems, stability is rarely actually visualized. This is relevant since the relation between a system’s poles and its behaviour in time domain is something that students find hard to grasp.

This project, as previously stated, is based on a teaching tool that contains analog filters and some DSP. Theoretically, the difference between analog and digital filters is clear, but in practice they can look very similar. Visualization of digital filters, and comparison of them to analog ones allows for a more nuanced picture of signals in different domains, and as such further deepens the students’ understanding of signals and different domains.

Another concept in signals and systems is that of stability/instability, something that is very important for a real life system, but that a lot of students find difficult [13]. Therefore, the tool in this project has been developed to contain digital filters and visualization of instability as well as the previously mentioned analog filters and DSP.
3.4 Practical limitations

The tool from last year implemented three kinds of analog filters: LP, BP and HP, all second order filters. The digital filters that will be created in this project will match these filters both in kind and order. In contrast to the previous project, the cut-off frequencies available for the filters in this project will be static, predetermined ones rather than freely chosen on a continuous scale. This is done due to the many calculations that the embedded system (a microcontroller unit) will otherwise have to make as soon as the cut-off frequency is changed even by 1Hz. By using predetermined cut-off frequencies, the focus can lie on matching the analog and digital filters rather than optimising the calculations.

4 Related work

There has been multiple tools made for visualizing different aspects of signal processing, most of them focusing on digital signal processing [9, 21, 24].

The most obvious one is last year’s previously mentioned project, a teaching tool for visualization of analog filters (and some DSP) by P. Ashraf et al [6]. The system implements some DSP in the form of sampling and reconstruction, that can be used to show discretization of signals and different methods of reconstruction.

There are several tools [21, 24] that make use of MATLAB to create a simulated system in order to visualize signal processing. One application (hereafter called SSUM), by B. L. Sturm and J. D. Gibson [21], was made with media arts students in mind, in order to teach the students complex concepts while avoiding diving into advanced mathematics [21]. SSUM could be used to demonstrate aliasing, filtering (IIR and FIR), sampling, etc. Unfortunately, the program’s last update was in 2005, and is no longer compatible with the newer version of MATLAB [6]. Furthermore, it also seems that the program is nowadays unavailable, as the downloading page mentioned in the article is unusable. This might be a consequence of the fact that the author no longer works for Media Arts and Technology at University of California, Santa Barbara, and they have removed his page.

Another application that make use of MATLAB integrates MATLAB and a fixed-point processor TMS32OC5x DSP Starter Kit (DSK), to create a teaching tool for DSP that allows the user to “compare the theoretical filter performance with the real-world performance” [24]. Stability problems are mentioned, but only the mathematical definition is shown. While this tool was well-received by the students according to the authors,
the DSK in question is old, and it is hard to find a retailer that sells it.

The tool that is the closest to what we are aiming to achieve is the teaching tool by H. Ping et al [16]. It is a MATLAB-based tool that simulates both analog and digital filters, as well as impulse response and zeroes and poles. It also has a module for audio manipulation that handles real-time signals, with features such as echoing, equalisation, mixing, etc. Sampling, reconstruction and stability are not touched upon. Even so, this tool is the one out of the ones we found that covers the most concepts.

In none of the found systems are analog and digital filters compared, and stability is only mentioned as pole zero plot [24] and not shown how the output signal behaves. Moreover, even though some of them contain real-time signals, the majority of the tools are simulations. Nonetheless, studies show that the use of “real world systems applications creates a degree of credibility and relevance that is not possible with software simulations” [20]. As we aim to make a tool that concretizes the concepts students find difficult, we have concluded that the inclusion of a real-life system (the circuit) is beneficial for the students’ understanding.

5 Method

This section includes the general methods for the design and development of the system, and what instruments were used during building and testing the system.

5.1 Filter implementation

**Analog filters** All of the analog filters are Butterworth filters of the second order, modelled of the schematics found in P. Karantzalis [10, pp. 42]. A LTC6911 (Dual Matched Amplifiers with Digitally Programmable Gain) was selected as an alternative to the LTC6912 specified in the schematic by P. Karantzalis [10, pp. 42] because the LTC6911 does not use a serial interface to program its gain. P. Ashraf et al used analog potentiometers in place of the LTC6912 [6]. The schematic for this project’s analog filters can be found in Figure 5.

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4except the tool from Uppsala University
5 Method

Figure 5: A digitally tunable second order active RC filter. Blue (thick) lines are buses, and green (thin) lines are single signals. This schematic can be found in the appendix.

The system was implemented using through hole-mounted components with the following tolerances: resistors $\pm 1\%$, plastic film capacitors $\pm 10\%$ and electrolytic capacitor $\pm 20\%$. All components were soldered onto a perfboard\(^5\), and connected with wires; the full schematics for the analog system are included in the appendix. All passive components used were selected on merit of being standard components, and therefore cheap and readily available. The TL974 was selected as it was one of the cheapest OP-Amps available to us that has audio circuits as a recommended application, and is specified to operate at 5V or less [22].

Digital signal processing The embedded system used for digital signal processing in this project is an MCU of brand and model “Atmel UC3-A3 Xplained”. All DSP were implemented in C, and the code was written using Atmel’s own IDE “Atmel Studios”. The digital filters were designed using MATLAB, to match the characteristics of the FIR and IIR filters as closely to the ones of the analog filters as possible. We chose to work

\(^5\)A perfboard is a thin, rigid sheet with holes pre-drilled at standard intervals across a grid, usually a square grid of 2.54 mm (0.1 in) spacing
with Atmel because we are familiar with the system due to previous work in the course “Signal processing and embedded systems”. We were also able to use the framework which we are familiar with that we used in this course, written for Atmel studios by Ping Wu.

The FIR filter was designed using the windowing method (which is essentially taking a subset of a larger dataset) [23, chapter 7.2, p. 170]. For the windowing, the Hamming window was chosen as it is one with the least ripples (a characteristic it shares with the Butterworth filter). The IIR filter in turn was designed by applying the bilinear transform: mapping an analog filter in s-plane into z-plane. The mapping is done by substituting $s$ in the transfer function $H(s)$ with the function $B(z) = \frac{2f_s z - 1}{z + 1}$ (where $f_s$ is the sampling frequency).

**Source code** This project’s DSP makes use of a source code for an embedded system. The source code was created by the head of practical work for courses in embedded systems, Ping Wu, for assignments with the intention to teach signal behaviour during the courses.

### 5.2 Computer application

**Oscilloscope** The visualization of the filters was done through a digital oscilloscope, based on the one by P. Ashraf [6].

**Instability simulation** The simulation of instable filters was done using MATLAB, and visualizes three different kind of instable filters. The simulation shows an unstable filter’s pole-zero plot, its transfer function’s magnitude and phase, and the input signal and output signal in time domain.

### 5.3 Additional equipment

Testing of the analog circuit was done mainly using a multimeter (Fluke 189 [2]) to measure voltages and impedance, and a frequency generator (Metex MS-9150 [5]) to generate signals. A digital oscilloscope (Rhode&Schwartz - HMO1002 [1]) was used to observe signals.
6 System structure

The system can be divided into three main parts: a hardware filter system and two computer applications. The hardware filter and one of the computer applications are connected via an audio-out and an audio-in connector on the computer. The second computer application is separate from the rest of the system.

6.1 Filter system

The filter system is the hardware that contains both the filter implementation as well as the interface that allows the user to select between different passband shapes (LP, BP, HP) and filter implementations (analog, IIR, FIR). The basic structure of the system can be seen in Figure 6 (note that only signal paths are shown in the figure).

Figure 6: System Block Diagram. Blue (thick) lines are buses, and green (thin) lines are single signals. The dotted lines marks are borders between stages. HV and LV stands for high voltage and low voltage, respectively. This and more detailed schematics for the different parts can be found in the appendix.

An input signal is generated using the computer application (Computer Audio Out) and sent either through an analog filter or the digital system, after which it goes back into the computer (Computer Audio In). The user interface (UI) consists of buttons, switches and potentiometers that enables the user to make choices regarding filter passband, cut-off frequency, filter gain and sampling frequency. The digital system consists of a DSP part, with digital filters, as well as an analog reconstruction filter.
6.1.1 Hardware user interface

The hardware UI consists of two multipositional switches, one rotary encoders, three pushbuttons, three LEDs and one potentiometer. The UI is connected to the MCU, which handles all choices made by the user, and sends out signals to different parts of the system depending on the user’s choices. The reconstruction filter is completely separated from the rest of the system, which is why some parts of the UI have multiple purposes. The possible choices are these:

(i) LP, HP or BP filter
(ii) FIR, IIR, analog or reconstruction filter
(iii) Order/number of taps of digital filter
(iv) Cut-off frequency for analog/digital filter (discrete)
(v) Cut-off frequency for reconstruction filter (continuous)
(vi) Toggle frequency and granularity

The filter type (i, ii) is chosen through multipositional switches. The filter order, as well as the toggle frequency and granularity, is chosen using buttons. The cut-off frequencies for the analog and digital filters are chosen using rotary encoders, and cut-off frequency for the reconstruction filter is chosen using a potentiometer. The hardware components are connected to different pins on the MCU, and to handle bouncing a time delay is used.

6.1.2 Analog filters

The analog filter system consists of a set of adjustable filters: a low, high and band pass filter as described by P. Karantzalis [10]. This system is based on the one used by P. Ashraf et al [6]. One difference is that in this system there are a fixed number of cut-off frequencies, as opposed to a continuous scale of frequencies. To be precise, there are seven different cut-off frequencies (see Table 2 on page 18). There are also additional features, mostly switches, added to the user interface part (more on this in section 6.1.1).
6.1.3 Digital filters

The digital filter system consists of a microcontroller unit (MCU) that can be used as an IIR or FIR filter. That can implement a low, high or band pass filter with matching cut-off to the equivalent analog filter. The digital system retains the functionality that was implemented in the base design by P. Ashraf et al [6].

In order to enable fast computing, all filter coefficients have been calculated beforehand, with a three different available orders of filters.

6.1.4 Reconstruction filter

This part of the system is a copy of the hardware implementation of the reconstruction filter used in P. Ashraf et al [6]. The evaluation of this is not in the scope of this project but is needed to maintain the functionality of the previous system and to be able to repurpose the digital resampling system for the digital filters.

6.2 Computer applications

The two computer applications are two graphical user interfaces (GUIs): one for the signal processing visualization, and one for the simulation of instability. The first one is a computer based oscilloscope created by P. Ashraf et al [6], and the second one is an application for simulation of instability in filters. The instability simulation shows the input signal, output signal, bode plot and pole-zero plot of the chosen unstable filter. There are currently four different inputs and three different unstable filters available. The instability simulation is separate from the rest of the system.

7 Requirements and evaluation methods

The end product should be an aid in understanding signal processing, and as such has both technical and non-technical requirements.
7 Requirements and evaluation methods

7.1 Requirements

**Analog system**  The analog system should be able to process analog signals, adhering to the user’s choices of e.g. cut-off frequency. There are seven static cut-off frequencies available, and the cut-off frequencies should increase with each step, such that the cut-off frequency for setting 2 is lower than the cut-off frequency for setting 4. In addition to this, it should be possible to illustrate the manipulated signals using an oscilloscope or a computer. Both in terms of different domains and with respect to human senses, in particular vision and hearing; a person with average vision and hearing ability should be able to use our system.

Since the system is implemented as hardware, component values are not exact. Errors due to e.g. tolerances and noise picked up from adjacent wires are to be expected. However, the main focus of this project is a tool that shows the difference between analog and digital filters, so the cut-off frequencies only need to be audible. Consequently, the tool tolerate a much higher variance in cut-off frequency than the components would suggest; it tolerate up to ±50% of the calculated cut-offs. However, the filter behaviour should be recognisable (i.e. there should be a clear distinction between LP and HP filters for the same frequency), and reproducible.

Furthermore, one request from Knorn is to have a system that is compatible with head phones or speaker. This is also one of the improvements proposed by P. Ashraf et al [6].

**Digital system**  Both digital filters, IIR and FIR, should match the analog characteristics: being available as low, high and band pass, with various different cut-off frequencies. The simulation of instability should show the behaviour of an unstable filter, with at least three different filters being available.

**Non-technical requirements**  The target user group for this project is teachers that hold courses in the subject Signals and Systems, that is, people that have theoretical knowledge within the subject but not necessarily experience with hardware and embedded systems. Therefore, the finished product should be user friendly, such that people not involved in the development of the product should be able to use the system without difficulties. Consequently, it is important that the system is documented thoroughly, mainly in form of a user manual. Another facet of the requirement of user friendliness is that students new to the subject should be able to interpret the results.

To ensure that the project is reproducible, system documentation beyond the user manual is needed, in the form of i.e. schematics. Lastly, it should be possible to use the product without access to lab equipment such as function generators, oscilloscopes, etc;
only a power outlet and a computer running MATLAB R2016a [4].

7.2 Evaluation and testing

Both the analog and digital filters were evaluated using an oscilloscope in conjunction with a function generator. This was done by comparing input and output signals with regards to amplitude in order to find the cut-off frequency of the filter. The analog filter sets the baseline and the digital filters should have the same cut-off frequency ±10%. The system’s compatibility with headphones and speakers was evaluated by having a person with unimpaired hearing listen to audio through the system, to ensure that it is audible through headphones and speakers.

To ensure compatibility with the oscilloscope software made by P. Ashraf et al [6] the analog filter and reconstruction filter of their system and ours was compared. The parameter that was compared was the output voltage swing, measured with an oscilloscope.

Ideally we would like to evaluate the system by letting several teachers test our system on their students over a period of time. The teachers should be teaching courses involving signal processing, comparable to the external stakeholder of this project, Steffi Knorn. We would also like to view statistics from the courses, to see if there would be an increase in the number of students passing courses involving signal processing. However, this evaluation method is not possible for us to perform due to lack of time; the results from this kind of study would be available in a couple of years.

8 Results and discussion

In this section, the results for the different parts of the system are presented, along with a discussion of them.

One thing that created a lot of problems was interference and noise. The computer is a source of noise in itself, as is the MCU. Although having a separate voltage source is not actually necessary for the system (all needed voltage can be supplied via the computer), it helps with the noise.
8.1 User Interface

The implemented user interface is able to switch between the different functionalities as described. The three pushbuttons can be used to chose IIR filters of order 1,2 and 3, and FIR filters with 16, 32 and 64 taps. They can also be used to toggle the sample rate and granularity as in the system by P. Ashraf et al [6]. However, the method used by the system to handle bouncing is different than the one in the system by P. Ashraf et al [6]. More specifically, the time delay was lowered from about 750ms to 100ms, ensuring a quicker response and a less noticeable delay.

The rotary encoder with which the user selects cut-off frequency has a delay of 20ms. A consequence of these delays is that the system sometimes does not register the impulse from the rotary encoder, or in some cases registering in rotation the wrong direction. This problem can be avoided by rotating the rotary encoder slowly (at least 1s between each complete turn)\(^6\).

8.2 Analog system

The results of the analog system were measured by using an input signal from a function generator, and then observing the output signal using an oscilloscope. The filters behave as expected, with clear difference in attenuation between the filters.

In addition to this, the cut-off frequency was also measured using an oscilloscope. This was done for all possible cut-off frequencies, see table 2. The cut-off frequency increase with every step, as expected.

In comparison to the previous project by P. Ashraf et al, the system is now optimised space wise [6]. The previous system was quite difficult to recreate due to lack of documentation of the building process. To make better documentation for our system, we have attached the schematics (see appendix), for our system.

<table>
<thead>
<tr>
<th>Filter type / Setting</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
</tr>
</thead>
<tbody>
<tr>
<td>LP</td>
<td>102</td>
<td>203</td>
<td>409</td>
<td>818</td>
<td>1640</td>
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<td>1990</td>
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<td>50/130</td>
<td>99/262</td>
<td>199/522</td>
<td>396/1050</td>
<td>789/2110</td>
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</tr>
</tbody>
</table>

Table 2: Cut-off frequencies in Hz, for all settings for filter types LP, HP and BP

\(^6\)This, and other special cases, is mentioned in the user manual.
8.3 Digital filters

Since analog input and output signals are necessary when communicating with a computer, it is impossible to completely separate the analog and digital system. The digital system makes use of DACs and ADCs to convert the signal from analog to digital (input), and back again (output). As a consequence of the analog components, there will be small, non-constant phase shift in the input to the MCU.

This is relevant for the FIR filter, since the FIR filter otherwise has a constant phase shift.

FIR The creation of FIR filters to match the analog filter was successful to some extent; the FIR LP filter works as expected. However, the HP and BP filters both work as LP filters. This is most likely due to a programming error during the calculations of the filter coefficients.

IIR A framework for all IIR filters has been created. However, at the time of writing it is not yet functional. This probably has to do with the coefficients, which were not correctly calculated.

8.4 Instability simulation

The application shows the desired plots (input signal, output signal, bode plot, pole-zero plot) for the chosen instable filter, with the possibility to choose between three unstable filters:

   (i) A second order filter
   (ii) A quantized IIR filter of very high order
   (iii) A non-proper filter

The application works as desired, and further unstable filters are easy to add. An example of a simulation displaying plots for the non-proper filter can be seen in Figure 7.
9 Future work

Because the project integrates several different fields of study there are many possible ways to continue development.

**Circuit board**  The current system’s greatest weakness is interference. This interference could be significantly reduced by designing the circuit on a printed circuit board instead. Apart from reducing reduces noise and interference, this would also allow for an overall improved performance of the whole system.

**Application that simulates digital filters**  An interactive teaching tool that the students themselves can use to explore and observe digital filter behaviour. This would be similar to the digital oscilloscope, except the digital filters would be simulated in MATLAB instead of implemented on an embedded system. We did find tools such as this during our research, but none that were available to the public and/or up to date with the current versions of programs.

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**Figure 7:** The user interface for the filter instability simulation, displaying plots for the non-proper filter with input signal $x(t) = \sin(300t)$. 

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Filter instability An improvement to visualization of filter instability would be to make it easier to add more instable filters, mayhaps having transfer function as a possible user input. Also, it would be convenient for future users if the system was compatible with future MATLAB versions as well.

Digital filters An improvement would be to add all the correct filter coefficients so all functionality is in place for the digital filters.

10 Conclusions

The goal of this project was to create a tool that visualize analog and digital filters, as well as reconstruction and instability. This tool’s purpose would be to for teachers to teach students intuitively how signals behave when a filter is applied.

Hardware The developed system gives the user the possibility to choose a filter implementation and passband shape by turning the two switches to different positions on the board. The user may also change sampling frequency from the three different buttons placed on the system board. Furthermore, rotary encoder on the board allows the user to choose the cut-off frequency for the reconstruction filter.

Software The finished project do also contain a simulation software showing filter instability. The simulation software is made in MATLAB. The user may choose an input signal for the system and which type of instable filter to simulate. It is also possible to simulate a stable filter, a second order Butterworth, for comparison.

Embedded software The embedded system is the updated framework, written for the Atmel board in C code. The software is working fairly well but is currently very unstable and the MCU must be power cycled. This leads us to affirm our decision to use static cut-off frequencies and precalculated coefficients. Expecting the MCU to handle all coefficient calculations in addition to everything else would have been unreasonable.

Both digital filters, IIR and FIR, match the analog filter characteristics to some extent. However, while the framework is close to complete, the calculations for filter coefficients are not.
In conclusion, while the project did not fulfill all of its requirements entirely, a great foundation for future improvements were made.

References


[22] *TL97x Output Rail-To-Rail Very-Low-Noise Operational Amplifiers*, Texas Instruments, JANUARY 2015, sLOS467H.


A Appendix

Components:

- OP-Amp (TL974)
- Rotary encoder (EC12E24204A2)
- Rail splitter (TLE2426ILP)
- Diskret OPAnp (LTC 6911-2)
- Voltage regulator for 5V (L7805CV)
- Switch two pole for Reconstruction
- Switch 2x3 pole for filter choice
- Switch 2x4 pole for tool choice
- 3,5mm contact
- torque resistor
- Adapter (MSOP-10)
- Potentiometer (dual) 85kΩ

Capacitors:

- 2 x 22nF
- 220μF
- 220pF

Resistors:

- 1 Ω
- 1,5kΩ
- 10kΩ
- 49,7kΩ
- 2 x 220kΩ
Header J1
- TWI0_SDA
- TWI0_SCL
- USART3_RXD
- USART3_TXD
- SPI1_CS0
- SPI1_MOSI
- SPI1_MISO
- SPI1_SCK
- GND
- VCC_P3V3

Header J2
- ADC0
- ADC1
- ADC2
- ADC3
- ADC4
- ADC5
- ADC6
- ADC7
- GND
- VCC_ANAL
- VCC_PWR

Header J3
- GPIO0
- GPIO1
- GPIO2
- GPIO3
- GPIO4
- GPIO5
- GPIO6
- GPIO7
- GND
- VCC_P5V0

Header J4
- TWI1_SCL
- TWI1_SDA
- USART1_RXD
- USART1_TXD
- SPI0_CS3
- SPI0_MOSI
- SPI0_MISO
- SPI0_SCK
- GND
- VCC_P3V3

+3.3V
S_LP
S_HP
S_BP
S_IIR
S_FIR
S_Resamp

S1
S2
S3
D0
D1
D2
RotA
RotB

UI_Passband
Filter_Imp

UL_Ctr
UL_Passband
UL_I0

Shematic by Tobias Holmberg

Title: Filter Visualization, Atmel UC3-A3 Xplained Connections

Size: A4
Date: 2017-05-13
File: Atmel_UC3-A3_Xplained.sch
Rev: B
Id: 3/9
The switch bypasses the reconstruction filter.
Rail-Splitter
VREF = 2.5V

CON701
BARREL_JACK

1N5406
D701

C701
0.33u
GND

C702
0.1u
GND

D701
1N5406

+5V
OUT

1
COM
2
IN

U701
7805

C703
1u
GND

U703
TLE2426

+5V
GND

VREF

GND

GND

GND
Modified version of "A Simple Digitally Tunable Active RC Filter" by Philip Karantzalis, Published in LT Magazine March 2006

Schematic by Tobias Holmberg

Title: Filter Visualization, Adjustable Analog Filter

Sheet: /Analog Filter/
File: analog_filter.sch

Size: A4 Date: 2017-05-13 Rev: B
KiCad v8.4 Head 40.3.6 ID: 8/9
Terminated Unused OP amps
Could be used for multiple outputs