Teaching Signals to Students: a Tool for Visualizing Signal, Filter and DSP Concepts

Pouya Ashraf
Linnar Billman
Adam Wendelin
Abstract

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Students at Uppsala University have for some years been given the opportunity to take courses in subjects directly, or indirectly, related to the fields of signal processing and signal analysis. According to the directors of these courses, a considerable number of students are recurrently having difficulties grasping different concepts related to this field of study. This report covers a tool that easily allows teachers to visualize and listen to different manipulations of signals, which should help students get an intuitive understanding of the subject. Features of the system include multiple kinds of analog filters, sampling with variable settings and zero-order hold reconstruction. The finished system is flexible, tunable and modifiable to the teachers every need, making it usable for a wide variety of courses involving signal processing. The system meets its requirements even though individual components’ results deviate slightly from ideal values.
Sammanfattning

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List of Abbreviations

AC: Alternating Current
ADC: Analog-to-Digital Converter
AUX: Auxiliary input jack, generally for audio
DAC: Digital-to-Analog Converter
DC: Direct Current
DSP: Digital Signal Processing
FPGA: Field-Programmable Gate Array
GPIO: General Purpose Input Output
IDE: Integrated Development Environment
OP-Amp: Operational Amplifier
Q-factor: Quality-factor
ZOH: Zero-Order-Hold
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1 Introduction

Students are having recurring difficulties grasping signal processing at the university. According to the teachers of the courses Embedded Signal Processing Systems and Signals and Systems at Uppsala University among others [22, 21, 20, 19], this is a problem. Signal processing requires rigorous analysis of signals and has strong ties to mathematical transform theory, which for many can seem quite abstract. This makes it difficult for the students to build any kind of intuition for the subject at hand.

The purpose of the system described in this report is to provide a tool, which can demonstrate and visualize different concepts related to the fields of signal processing and analysis. It is to be used by teachers in signal processing and analysis courses as an apparatus that provides a frame of reference for the students and a more intuitive understanding of signal processing. The system is designed to be as flexible as possible to allow multiple concepts to be demonstrated with the flick of a switch or press of a button. The signals are shown before and after the manipulation by the system on an easy to use digital oscilloscope developed in MATLAB [3], where the signals can also be generated. Teachers also have the option of playing the signals in speakers to let students hear the difference between a filtered and unfiltered signal.

The areas of signal processing that are covered by the system are: active analog filters (low-pass, high-pass and band-pass), sampling, reconstruction, and frequency domain analysis of signals. Filters and sampling parameters are configurable during run time and everything is contained within a circuit board, an embedded system and a computer compatible with MATLAB. The implementation deviates slightly from the design due to non-ideal analog components, which could be improved upon in a future revision, but overall the system works as intended.

The system can be used as a teaching assistance in any course involving signal processing and analysis. A local implementation of the system with this report as a guideline should work with a variety of wall outlet adapters and filter requirements. The filters’ frequency ranges can be easily be modified, allowing teachers to adapt the system to their specific need. Any embedded system that features general-purpose input/output (GPIO) pins, an analog-to-digital converter (ADC) and a digital-to-analog converter (DAC) can be used. Overall this means that the system is usable with a wide range of components and settings, which should make it accessible to teachers with a varying amount of resources.
2 Background

A short introduction to the cause of the problem is discussed, followed by basic signal processing concepts. Limitations of real world filters are portrayed and applications are brought up in the form of reconstruction filters and anti-aliasing filters.

2.1 Why This Project?

Throughout the years that the course *Embedded Signal Processing Systems* and *Signals and Systems* has been running at Uppsala University, it has become evident that students are having problems grasping the concept of signals, in particular the concept of different domains. Without this crucial understanding of the basic characteristics of signals, numerous students have struggled to pass the course and as a result, have been discouraged from the subject itself.

The external project stakeholders are the teachers of *Embedded Signal Processing Systems* and *Signals and Systems*, who have realized a demand for a tool to visualize the difficult concepts involved in the course. They have supervised the project and provided feedback on the usability of the system.

2.2 Signals

The definitions of signals discussed in this paragraph are described in *Introductory Signal Processing* [15, chapter 0, p. 1]. A signal is a function that conveys information about the behavior of a system or attributes of some phenomenon. Depending on the field of study, there are more specific definitions. In the field of electrical theory, it often refers to a way of representing different physical quantities of interest, using a voltage or current.

In this project we studied electrical signals. In particular we studied and presented the properties of periodic signals when manipulated (i.e. filtered, sampled, reconstructed) with the help of analog and digital systems.

2.3 Analog Filters

A filter is a system that according to its design characteristics mathematically maps an input signal to an output signal [23, chapter 1.3, p. 4]. This mapping could include dampening of a disturbance, or rejecting a specific band of frequencies. An analog filter is often implemented using components like resistors, capacitors, inductors and OP-Amps.
Ideally, a filter has an output gain of exactly one (1) in the band of frequencies it is designed to pass, and an output gain of zero (0) in the band of frequencies it is designed to reject [9, chapter 2.1]. This also implies that the ideal filter has no rippling in either band (meaning there will be no fluctuations of the gain).

It is, however, not possible to implement an ideal filter, since they would require infinite impulse responses, meaning they would have an impulse response $\geq 1$ for all time $t$ between $-\infty$ and $\infty$. This would also imply non-causality, which in reality is impossible to implement [12, chapter 5.1, p 65].

All filters have at least one cutoff frequency, $\omega_c$. We use the definition as the frequency at which the output gain of the filter is $\sqrt{2}^{-1}$ [25, chapter 4.03, p 85]. There are four main filter types, with their own corresponding characteristics:

- **Low-pass**: passes $\omega < \omega_c$, rejects $\omega > \omega_c$ (Figure 1a)
- **High-Pass**: passes $\omega > \omega_c$, rejects $\omega < \omega_c$ (Figure 1b)
- **Band-pass**: passes $\omega_c < \omega < \omega_c$, rejects $\omega < \omega_c < \omega$ (Figure 1c)
- **Band-Stop**: passes $\omega < \omega_c < \omega$, rejects $\omega_c < \omega < \omega_c$ (Figure 1d)

As mentioned above it is not possible to implement an ideal filter. To be able to implement a filter, we would have to modify the specifications to allow for a transition-band to exist in the filter, meaning the output gain would not instantly switch from pass-band to stop-band, but instead allow for a continuous transition.
2.4 Sampling, Reconstruction, and Reconstruction Filters

This section is about a basic concept in signal processing and based on *Signals and Systems: Analysis Using Transform Methods and MATLAB* [16, chapter 3,10].

Continuous-time data consists of an infinite number of data points, which makes direct processing by a digital system not feasible. Digital computing cannot handle unlimited data points due to memory and processing constraints. Sampling discretizes continuous signals by choosing a finite number of data points, that can be processed and manipulated by a computer. A discretization of a sinusoid is depicted in Figure 2.

![Figure 2: Visualization of continuous-time and discrete-time signals.](image)

A well sampled signal retains its information and would allow one to recreate it in ideal conditions perfectly. Two important parameters are needed to sample a signal well: sampling frequency – the rate at which you obtain the samples from, and the number of samples – the amount of data points taken from the signal.

A requirement for the signal to not lose information from sampling is that it is band-limited, meaning that the signal is bounded to a limited range of frequencies e.g. a signal of sinusoids ranging from 0–200 Hz. A signal is said to be recoverable exactly from the samples if the sampling rate is more than twice the bandwidth of the signal, often called the Nyquist rate. Sampling with a frequency below and above the Nyquist rate is called undersampling and oversampling respectively.

In practice, sampling of electrical signals is done via an ADC, which accepts an electrical voltage (continuous-time) as input and outputs a set of binary bits. An ADC needs to output a limited number of bits, which corresponds to the input signal’s different amplitude values.

The process of recovering a signal from a number of samples is called reconstruction, and is in practice done using a DAC. A common method of reconstruction is called...
zero-order hold (ZOH) reconstruction, in which the DAC holds the value of a sample for an entire sampling period $T_s$, until the next sampling value occurs.

![Diagram of discretized signal and ZOH reconstruction](image)

Figure 3: Visualization of a discretized signal and ZOH reconstruction of said signal

The result of ZOH reconstruction is a staircase-like signal with sharp edges (Figure 3). These edges are caused by high-frequency components of the rectangles used in the convolution. To smooth these edges, we can use a low-pass filter with an appropriate cutoff frequency adapted to the sampling frequency, and thus dampen the higher frequencies of the signal. This will result in a smoother version of the reconstructed signal.

### 2.5 Aliasing and Anti-aliasing Filters

Aliasing is thoroughly discussed in *Signals and Systems: Analysis Using Transform Methods and MATLAB* [16, chapter 10], which the following paragraphs are based on.

Studying the frequency domain of a sampled signal, we see that copies of the frequency spectrum occurs periodically with the sampling frequency $f_s$. The copies are called aliases. A signal sampled at a frequency less than twice the bandwidth of the signal will have overlapping aliases, which is called aliasing.

The overlapping is due to the signal being undersampled and will make high-frequency components of the signal appear like low-frequency components. Undersampling can, as stated above, lead to incorrect identification of frequencies in the signal and is visualized in Figure 4.

To minimize the magnitude of the aliases when sampling, an anti-aliasing filter can be used. Filtering the signal before the sampling with a low-pass filter to remove higher frequencies will limit the noise from aliasing, which is seen in Figure 4c.
3 Purpose, Aim and Motivation

A summary of the projects goals and desired results are presented in this section, together with the proposed demand for the results.

Purpose
This project is supposed to help teachers in courses containing signal processing and analysis. With a proper tool for teaching, the subject could be taught with more ease. Students’ intuition for signal processing/analysis should be increased, as well as their motivation to consider further studies in the subject.

Aim
The project strived towards designing and implementing analog filters and digital sys-
tems, which would show different signal processing concepts. An analog system consisting of a low-pass, high-pass and band-pass filter with tunable cutoff frequencies was developed to visualize the general properties of filter types. The three mentioned filters can be toggled. The low-pass filter of the system can also be utilized as an anti-aliasing filter. An embedded system samples and reconstructs, with various settings, the signal gathered from the aforementioned analog system. After the embedded system, the signal is propagated to an analog system with a reconstruction filter that can be toggled. This gives multiple filter options that the teacher can use to display a variety of concepts including, but not limited to, filtering, sampling and ZOH reconstruction.

A complete system includes distinctly displaying the signals on a computer independent of the operating system. A digital oscilloscope was developed in MATLAB to accommodate for such a need.

Motivation
Multiple papers [22, 21, 20, 7, 19] find that students have a hard time grasping the abstract concepts of courses involving signal processing and analysis, but through using visualization tools and real applications, a greater interest in the subject can be attained. The currently available systems mainly deal with either analog or digital systems, but not both according to our research, discussed in Section 4. The concluded problem is the lack of an easily obtainable system with a combination of flexible analog and digital systems. The system in this report puts an emphasis on using general components and simple subsystems to make it obtainable, while still providing a reasonable set of functionality to make it flexible.

3.1 Delimitations
We did not develop a tunable band-stop filter. Complexity of the circuit would have increased and a band-pass filter was deemed to show the same fundamentals as a band-stop filter. We were able to build flexible analog filters that were able to cover a small range of cutoff frequencies and filter types. Due to the frequency range limitations of the analog filters, the sampling frequency had to be in a specific range to allow demonstration of aliasing and ZOH reconstruction.

4 Related Work
For the analog filters, we used related work on digitally tunable analog filters [14]. The referred system features a low-pass, high-pass and band-pass filter combined into one
system. All filters’ cutoff frequency can be tuned with the help of Serial Peripheral Interface [10], a digital communication protocol. We desire multiple tunable filters and therefore used the work as a basis. However, we modified this to be able to tune the filters analogically instead of digitally to make the analog system independent of the digital system.

The subsystem concerning sampling and reconstruction was derived from a general digital signal processing system [11] consisting of anti-aliasing, ADC, DAC and reconstruction, but with modifications to allow components to be toggled and configured. The number of systems similar to ours were sparse and the systems mentioned below are the ones which were found during our research. There may be a number of systems, which were not published, providing comparable tools.

An application for MATLAB, "Signals and Systems Using MATLAB" [21], was developed to teach media arts students complex concepts without the use of advanced mathematics. Similar to our proposed system, they valued a flexible, fast system with different mediums to obtain information, such as sound and distinct visuals. The application provided tools to demonstrate sampling, frequency analysis, filtering etc.. The authors of the article, Bob Sturm and Jerry Gibson, concludes that the application made students approach the subject with enthusiasm. The program’s latest update was in 2005 and is no longer compatible with newer versions of MATLAB, which inhibited us from using and testing it.

Xiaoyan Tian et al. have developed a teaching system [22] for digital signal processing (DSP), based on MATLAB and Field-Programmable Arrays (FPGA). The system was supposed to reduce the difficulty for students by showcasing real DSP on an FPGA. They claim that the system has stimulated learning, improved the quality of teaching and to be more in line with engineering practices in comparison to simple MATLAB simulations. We strived toward similar goals and hoped to achieve comparable results, but used analog implementations instead of an FPGA to showcase analog filters rather than digital filters. A system consisting of an FPGA is fully modifiable in the realm of digital filters. However, it lacks capabilities in analog filtering, most notably anti-aliasing and reconstruction filtering. Our system can only implement limited tuning due to the physical realizations of the analog filters, but does provide demonstrations of common analog filters, which the FPGA system is incapable of.

Quantization, sampling, reconstruction, aliasing and a digital bandpass filter have been demonstrated by a system [18] at the University of Reading. It was based on a digital signal processor and provided an anti-aliasing filter and a reconstruction filter, both of which could be toggled digitally. Quantization noise was demonstrated by variable
masking of the least significant bits in the ADC. The aforementioned system has basic analog filters with set cutoff frequencies and a DSP with fixed sampling frequency. Our system showcased quantization noise similarly, but also provided a greater variation in flexible analog filters and a sampling frequency that could be modified.

The system we developed is expected to be beneficial to a teacher of signal processing compared to the aforementioned systems, since the whole process of filtering, sampling and reconstruction can be demonstrated with a large combination of filter properties, sampling frequencies and ADC granularities. Not being limited to only analog or digital signal processing means that our system can be used in a variety of courses.

5 Methods

In this section, general methods of design and development for the different analog and digital systems are presented, along with the essential components used to implement them.

How close the system adheres to its reference design can be improved by choosing components with a stricter tolerance. Embedded systems with an ADC, DSP, GPIO pins and a DAC can be used for the digital part of the system, with varying results based on ADC/DAC resolution and sampling frequency capabilities.

Atmel UC3-A3 Xplained

The embedded system in our design consists of a microcontroller of brand and model "Atmel UC3-A3 Xplained". To write programs for this embedded system, we used Atmel Studio [1], which is an IDE developed by Atmel specifically to write programs for their embedded systems. The code for the embedded system was written in C, and the more vital components used were the ADC, DAC, and GPIO pins.

Analog Systems

The analog systems were implemented using surface-mounted components with the following tolerances: resistors ±1%, plastic film capacitors ±10% and electrolytic capacitor ±20%. Operational amplifiers (OP-Amps) of model TL082CP were seated in DIP 8 sockets. Components were soldered onto a perfboard, where they were connected with solid wires.

The filters were designed to adhere to Butterworth specifications and characteristics as far as possible. The tunable filter circuit was based on a design presented in an article [14, p. 42-43] and the reconstruction filter was implemented in the Sallen-Key
topology [17], as opposed to the Cauer topology [24, chapter 2], since the latter requires inductor coils.

**Facilitating equipment**
Evaluation and testing of designed subsystems were carried out using a frequency generator (Metex MS-9140) to generate signals, an oscilloscope (Hitachi V-252) to display signals and a multimeter (UNI-T UT33D) to measure voltages. The aforementioned set of specialized equipment facilitates the evaluation of the analog systems.

**Digital Oscilloscope**
To present the output signal of the system, a digital oscilloscope program was designed. The program was designed and implemented using MATLAB, a powerful tool for scientific modeling and calculation. The digital oscilloscope displays two plots: one to display the signal in time domain, and one to display the frequency spectrum of the signal.

# 6 System Structure

The complete system which has been created consists of the following components:

- Digital oscilloscope created in MATLAB to generate and display signals
- Analog tunable filter circuit consisting of three types of filters with tunable properties
- Analog ADC offset which prepares the signal for digital processing
- Atmel UC3-A3 Xplained, an embedded system that has been programmed to perform ADC and DAC with various settings
- Analog reconstruction filter which processes the signal received from the embedded system
- Analog rail splitter which adapts the input voltage from an outlet to provide supply voltage to the analog circuits.

These systems will be explained further in Sections 6.1 and 6.2.

A typical signal in the system will start by passing through either the tunable low-pass, high-pass or band-pass filter. It will continue through the flexible ADC of the embedded system and exit out of the digital system by the DAC in a ZOH form. The signal is then propagated into the reconstruction filter, which smooths the jagged ZOH reconstruction
into a signal similar to the original. For a graphical representation of the system, see Figure 5.

Figure 5: Block-diagram of the complete system, which is comprised of two main parts: the analog system (Tunable filters, rail splitter and reconstruction filter), and the digital system (ADC, DAC, and ZOH Reconstruction on the Atmel Board.)

Because of the flexibility of the system it has more than one combination of input- and output-signals. One can either choose the output coming from the tunable filters, or the output coming from the reconstruction filter.

6.1 Analog Systems

All analog filters are implemented as Butterworth filters due to the low fluctuations, which is thought to be easier to grasp for people not familiar with signal processing. A Chebyshev- or an Elliptic filter were possible candidates for analog filters, but they have ripples in the pass-band and/or stop-band [16, chapter 15, p. 676-677], which is why they were disregarded in the implementations.

Rail Splitter

A rail splitter turns positive voltage and ground into positive voltage, virtual ground and negative voltage. This is needed for systems utilizing OP-Amps and manipulating AC signals.

We wanted to make our system as portable as possible. This, in our case, meant not having to rely on equipment described in Facilitating Equipment in Section 5. This included being able to power it from a regular wall socket. However, to power our analog system, we needed positive as well as negative voltages in the ranges of $\pm 2.5 \text{ V}$ since
multiple possible input signals to the analog systems will be AC. As outlets only provide 230 V AC and a ground, we initially converted the supply voltage from 230 V AC, to approximately 10 V DC with the help of a simple wall adapter. Secondly, we created a virtual ground to simulate negative voltages. This will henceforth be referred to as a rail splitter.

An alternative to a wall outlet would be to power the circuit using batteries. The same principle, with a virtual ground, would be applied. Batteries do have the drawback that they get drained, which might make the circuit unreliable when it is needed.

**Tunable Reconstruction Filter**

As mentioned in Section 2.4, the role of a reconstruction filter is to filter away high frequency components from a reconstructed signal, to smoothen the jagged edges. For this we used a second order low pass filter with a tunable cutoff frequency. A switch is present to allow the user to enable and disable the reconstruction filter to easily show the effects of reconstruction filtering.

A Butterworth filter was desired, but due to the required flexibility and the wanted unity gain, the filter was slightly more dampened than a standard Butterworth filter. The filter has a Q-factor of 0.5 compared to a Butterworth filter’s Q-factor of approximately 0.707. The Q-factor directly corresponds to the amount of ripples in the filter. The designed filter has the same Q-factor in all frequencies, but could have been designed to meet a Butterworth filter’s requirements on a specific frequency. Such a filter would have a varying Q-factor, which would make the ripples hard to predict when tuning the filter.

**Tunable Filter Circuit**

As described in Section 4, we created a filter circuit consisting of three second order Butterworth filters of types low-pass, high-pass and band-pass. These have tunable cutoff frequencies. The cutoff frequencies are tuned using operational amplifiers and potentiometers instead of an integrated circuit and digital signals. This decision means that the user of the system has direct control of the system through a potentiometer, which makes the tunable filter circuit usable without an embedded system and allows more independent use of the different subsystems. A switch lets the user choose the desired filter.

**ADC Offset**

The ADC of the used embedded system [2] has a range of 0 V to 3.3 V. Therefore the AC signal provided by the Tunable Filter Circuit needs a DC offset to prevent losing the information stored in the negative part of the signal. An OP-Amp was setup as a Differential Amplifier, which switched polarity of the input, and added a DC offset of
1.2 V. The offset was chosen to maximize the potential amplitude of the signal without saturating the ADC or the previous circuits.

### 6.2 Digital Systems

The digital part of the system consists of an embedded system and a software implemented digital oscilloscope. Their capabilities will be discussed in this Section.

**Atmel UC3-A3 Xplained**

As mentioned in Section 5, the digital systems used in this project consist of an embedded system of brand and model "Atmel UC3-A3 Xplained" [2]. The Atmel Board is equipped with many different hardware components that will be used in this project. Among those are:

- **ADC:** The ADC on the Atmel Board will be used to take continuous signals as input, and sample them, which results in a discrete signal that can be manipulated by the DSP.

- **DAC:** The DAC is used to take discrete signals that have been processed internally on the board, and convert them back into continuous signals using the Zero-Order Hold reconstruction method discussed in Section 2.4.

- **GPIO pins:** The GPIO pins are used as external triggers to send interrupts to the microprocessor. This will allow for the control of different parameters during run time, like the sampling frequency and quantization resolution of the ADC.

In addition to these components, the Atmel board has a very capable microprocessor which makes it a good choice for many kinds of signal processing like digital filtering, etc.

**Digital Oscilloscope**

To display the resulting output signals of our system, a digital oscilloscope was developed. It features a simple function generator where one can generate a sine-wave, square-wave or sawtooth-wave at a wide range of frequencies to use with the system. With the oscilloscope one is able to select an input device, and simultaneously display both input and output signals in both time- and frequency domain.

The oscilloscope was developed as a script in MATLAB. Additionally, for the purpose of portability, the script was compiled into a standalone application, which is able to run on computers installed with MATLAB Compiler Runtime.
7 Requirements and Evaluation Methods

The resulting system is to be used as an aid in visualizing and understanding signals and systems, therefore it needs to fulfill some requirements that are not purely technical.

**Analog System**

The analog system should be capable of manipulating analog signals, using different kinds of filtering. One should also be able to illustrate how these manipulations are perceived in different domains (time- and frequency- respectively) and how the human body registers them (i.e. with respect to vision and hearing) with the help of a computer or oscilloscope.

All filters should comply well enough with the theoretical implementation described in Section 8.1 to be able to collaborate with the rest of the system. However errors are expected since analog components are used, which means that the values are not definite and there is a possibility of picking up noise from adjacent wires in the circuit. The behavior of the filters however should be recognizable and be precise enough to allow demonstration of the filters’ properties. The rail splitter should also adhere similarly to theoretical values. The analog subsystems are deemed precise enough if the implementations have a maximal deviation of 10% each.

**Digital System**

The digital system should be capable of sampling a signal, and subsequently reconstructing said signal using ZOH reconstruction. More importantly, the system should refrain from any reconstruction filtering, since that will be handled by a separate analog system. Additionally, just as with the analog system, one should be able to illustrate how sampling and reconstruction via ZOH is perceived in different domains, and how the human body registers them.

The sampling frequency and the granularity of the ADC should be able to be modified by hardware switches during run time without requiring a restart of the embedded system.

**Nontechnical Requirements**

Because the system will be used mainly by lecturers, professors and teachers for educational purposes, it is of great importance that the system itself is documented in the shape of user manuals, such that someone who was not directly involved in developing the system is able to use it without difficulties. Furthermore, the results it produces should be understandable by students who do not have a lot of experience in the field of signal analysis/processing.
The system should also be designed with portability and modularity in mind. One should be able to use the entire system, without access to heavy-duty lab equipment (e.g. oscilloscopes, voltage/current suppliers, function generators, etc.). Optimally, the only thing that should be required to run the system would be a computer running Windows, Mac OS X, or Linux to display and generate signals.

7.1 Evaluation Method

**Analog Circuits**
Evaluating the analog circuitry was simply a matter of comparing our implementation with the specifications provided by reference designs and calculations that are derived in Section 8.1, and seeing if the implementation matched the reference designs within an acceptable margin of error of 10%.

The Rail splitter was evaluated with respect to its ability to set a stable virtual ground at the midpoint of the supplied voltage and the actual ground (i.e. if the rail splitter is supplied with 10 V and the ground is at 0 V, the virtual ground should be set at approximately 5 V).

For the tunable low-pass, band-pass and high-pass filters, their respective cutoff-frequency ranges were evaluated, and how closely they adhered to the original design presented in [14].

**Digital Oscilloscope**
The evaluation of the oscilloscope was made mainly by observing how it performed, compared to an analog oscilloscope confirmed to be functioning correctly. The developed oscilloscope’s ability to correctly do the following was evaluated:

- **Time scaling:** The ability to change the scale of the time axis is elementary to understanding what signals look like, and how they behave. For signals with high frequencies, one would want a short time scale in the range of microseconds, whereas for signals with lower frequencies, one would want a time scale in the range of milliseconds. This setting is often referred to as 'Time/Div' on oscilloscopes.

- **Voltage Scaling:** Just as with the scaling of the time axis, being able to change the scaling of the voltage axis is an important feature of an oscilloscope, since one would want the ability to observe signals with different amplitudes. This setting is often referred to as 'Volts/Div' on oscilloscopes, since the amplitude of an electrical signal is measured as its voltage.
8 Implementation

- **Triggering**: The trigger function is one of the key elements of any oscilloscope, the presence of which allows one to observe a stable image on the screen. Without a trigger, or other form of synchronization, it would be nearly impossible to display a steady signal on the screen.

- **Spectrum plotting**: The ability to display the frequency spectrum of a signal is a feature present in many modern digital oscilloscopes. Being able to display the spectrum of a signal makes for easier identification of what frequencies a signal is composed of, and can give a visual explanation of phenomena like aliasing.

**Embedded System**
The evaluation of the Atmel board’s correct functioning was based on visual experiments to determine if the wanted effect was achieved. The correctness of the sampling frequency was evaluated by observing the aliasing effect of a sinusoid in the frequency spectrum, where half the sampling frequency is a colliding point for the aliases of the signal.

ZOH was evaluated by inputting a sinusoid into the Atmel’s ADC and observing the visualized output from the DAC of the Atmel board. If the output featured distinct squares without any kind of compensation, ZOH was deemed to have been used. The change of ADC granularity that was required for the system to be deemed functional was evaluated by monitoring the number of distinct levels the output featured in an oscilloscope. Only low ADC resolutions were controlled due to the amount of levels that had to be manually counted.

8 Implementation

Schematics for the analog systems and the general structure of the digital systems are introduced in this section.

8.1 Analog Systems

Each of the analog systems are soldered onto a single perboard and connected with wires. The different filters can be toggled with switches and the potentiometers of both filters are dual to allow easier manipulation of the frequency ranges.

**Rail Splitter**
One major problem that occurred was how to supply the correct voltages to the OP-Amps used in the rest of the analog system (±2.5 V). To solve this problem, we developed a circuit that takes an input voltage of 8 V, and brings the voltage down to 5 V with
the help of a voltage divider. Finally, it sets a virtual ground at 2.5 V.

![Circuit diagram for the rail splitter.](image)

This gives us the ability to supply electricity to our analog system directly from a wall outlet, as opposed to needing a dedicated power supply.

**Reconstruction Filter**

As mentioned in Section 2.4, reconstruction via ZOH results in a signal with jagged edges, which are caused by high frequency components. To smooth out these jagged edges, we will be using a low-pass filter to eliminate these high frequency components. The design is based on a Sallen-Key low pass filter [8].

![Circuit diagram for the reconstruction filter.](image)

The reconstruction filter used in our system consists of a second order low-pass filter, with a tunable cutoff frequency from 55 Hz to 4.3 kHz. The transfer function for this filter is given by:

\[
H(s) \big|_{s=(\sigma+j\omega)} = \frac{1}{R_1 R_2 + \frac{1}{s C_1} (R_1 + R_2) + \frac{1}{s^2 C_1 C_2}} \quad R_{1,2} \in [1 \, \text{k}\Omega, 85 \, \text{k}\Omega] \quad C_{1,2} = 22 \, \text{nF}
\]
Tunable Filter Circuit

This circuit is the more substantial part of our analog system. As can be seen in Figure 8 below, the circuit takes an input signal, and passes it through a set of filters. Depending on what kind of filtering is desired, one chooses the low-pass, high-pass or band-pass output. The transfer functions for each respective output is also presented below.

The main advantages of this circuit are not only the fact that it contains three different types of filters, but also that the cutoff-frequencies of these filters are tunable with the use of a potentiometer (variable resistor).

This circuit is largely based on the circuit presented in "A Simple Digitally Tunable Active RC Filter" [14]. The original design used a digital circuit with an SPI-connection to tune the cutoff-frequencies of the filters, while our slightly modified design uses analog potentiometers to achieve the same result. This essentially eliminates the need for a separate computer to operate the system.

Figure 8: Circuit diagram for filters with tunable cutoff frequencies.
Table 1: Cutoff frequency (Hz) and gain of the tunable filters

<table>
<thead>
<tr>
<th>Cutoff/Gain</th>
<th>45</th>
</tr>
</thead>
<tbody>
<tr>
<td>Low-pass 100</td>
<td>4650</td>
</tr>
<tr>
<td>High-pass 100</td>
<td>4650</td>
</tr>
<tr>
<td>Band-pass 50/200</td>
<td>2350/8750</td>
</tr>
</tbody>
</table>

\[
H_{LP}(s)_{s=(\sigma+j\omega)} = -\frac{\frac{R_4}{R_3}4\pi^2f_0^2}{s^2 + \frac{2\pi f_0}{Q} s + 2\pi^2 f_0^2} \quad R = 1.58 \, \text{MΩ}
\]

\[
H_{BP}(s)_{s=(\sigma+j\omega)} = -\frac{\frac{R_3}{R_2}2\pi f_0}{s^2 + \frac{2\pi f_0}{Q} s + 2\pi^2 f_0^2} \quad C = 1 \, \text{nF}
\]

\[
H_{HP}(s)_{s=(\sigma+j\omega)} = -\frac{\frac{R_2}{R_1}s^2}{s^2 + \frac{2\pi f_0}{Q} s + 2\pi^2 f_0^2} \quad f_0 = \frac{G}{2\pi RC} \quad Q = \frac{R_3}{R_2}
\]

The cutoff frequencies of the different filters and maximal/minimal potentiometer settings are depicted in Table 1.

8.2 Digital Systems

The Atmel Board

The Atmel Board is configured to sample at 46875 Hz to allow multiple fairly precise lower sampling frequencies, which are obtained by only recording every X sample. For example, a sampling frequency of 2000 Hz is obtained by only registering every 23rd sample, effectively giving a sample rate of 2038 Hz. A set of sampling frequencies are rotated while the system is running. Rotation is triggered with a press on a momentary switch, which sends a logical high signal to a GPIO pin on the Atmel Board.

Granularity of the ADC can be set with a similar momentary switch to a different pin. Different resolutions of the ADC are obtained by clearing a number of least significant bits in the sampled value, which is maxed out at 10-bit resolution. The system is therefore capable of conveying 1-bit to 10-bit resolution.

The set of chosen sampling frequencies is \{997, 2038, 3906, 5859\} which allows signals that are pleasant for the ear to be demonstrated. The set of resolutions that can be chosen is \{2, 3, 10\}. They were deemed the most recognizable and therefore most suitable for teaching purposes.
9 Evaluation Results

The momentary switches were implemented with 10 kΩ pull down resistors to ensure a logical low in the idle mode. They are mounted on an add-on module, which is attached directly to the pins of the board. A third button was added in hardware and software to accommodate for further development on the system with digital filters.

To avoid bouncing of the switches, which would make them trigger uncontrollably, a delay between presses is shared between all buttons. The delay is noticeable if multiple consecutive presses are to be done. An alternative implementation that would allow faster pressing of the buttons contains a Schmitt-trigger [6, Chapter 7.3.2] to reduce noise in the button circuit and to provide clean logical transitions of the voltage. The interrupt delay could then have been reduced or completely avoided. This option was not feasible due to limitations on the physical space the button circuit must adhere to.

Digital Oscilloscope

The digital oscilloscope was written as a platform independent MATLAB script, that is capable of plotting the outgoing and incoming signals multiple times per second, together with the frequency spectrum for each corresponding signal.

The scales for the time and frequency axes can be set independently of each other as well as the voltage scale. The program also has the capability of generating sinusoidal-sawtooth- and rectangular signals with different frequencies.

9 Evaluation Results

The evaluation of the system is based on the methods in Section 7.1. The analog- and digital systems are separately evaluated and compared to the requirements.

9.1 Analog Systems

During evaluation, we found the analog systems were performing in a satisfactory manner. The tunable filters, the reconstruction filter and the rail splitter behaved largely as expected, and were in line with the original specifications. The tunable filters’ deviations are presented in Table 2. The reconstruction filter differs 0 % on the lowest setting and 1 % on the highest setting. The rail splitter’s virtual ground deviates by only 1 %.

All deviations of the analog subsystems are within 10 %, which satisfies the requirements. The measured values for cutoff frequencies and the rail splitter voltages are presented in Section 10.1, with discussions about the possible causes of the deviations.
10 Results and Discussion

Table 2: Deviations of the tunable filters

<table>
<thead>
<tr>
<th>Deviations/Gain</th>
<th>1</th>
<th>45</th>
</tr>
</thead>
<tbody>
<tr>
<td>Low-pass</td>
<td>0 %</td>
<td>8.6 %</td>
</tr>
<tr>
<td>High-pass</td>
<td>0 %</td>
<td>8.4 %</td>
</tr>
<tr>
<td>Band-pass</td>
<td>0 %/0 %</td>
<td>6 %/3%</td>
</tr>
</tbody>
</table>

9.2 Digital Systems

Software for the Atmel Board
The sampling frequency is able to be set to multiple frequencies, with varying accuracy due to the implementation method. The achieved sampling frequencies, in Hz, are \{997, 2038, 3906, 5859\}, which have been confirmed visually and should be adequate for all visual demonstrations. Distinct squares can be distinguished when the input is a sinusoid, which concludes that a pure ZOH has been attained. The multiple ADC granularities have been observed and the granularities of one, two and three bits have been counted to assure that the number of levels are correct, which makes us assume that also the higher bit resolutions are correct. The full amount of levels of the ADC can not be realized with our system due to the signal not fully saturating the voltage range of the ADC.

Digital Oscilloscope
The oscilloscope has the functionality mentioned in Section 7.1, namely scaling of the time/div and frequency axes, as well as voltage scaling, software triggering, and the ability to plot the frequency spectrum of a given signal. The different signal processing concepts that are to be demonstrated by the system are all deemed to be visible and distinct when displayed on the oscilloscope. Most of the text is bold and colors are contrasting to make everything distinguishable when displayed using a projector. Thus it performs as expected and does not deviate in any noticeable way.

10 Results and Discussion

The results for each subsystem are presented, together with a brief discussion around said results. The section starts with the analog systems, continues with the digital systems, and is concluded with an overview of what has been accomplished in relation with the goals set at the beginning of the project.

10.1 Analog Systems
Rail Splitter
The results of the rail splitter were measured with a multimeter on the output, with an input voltage of 8.40 V from the outlet. The rail splitter brought down the voltage to 4.94 V and placed the virtual ground at 2.50 V. This provided a supply voltage of 2.50 V and −2.44 V. We also noted, from observations on an oscilloscope, that the virtual ground was sufficiently stable for our purposes, even with the rest of the circuit connected.

These results were satisfactory when compared to our initial goals and complied well with the theory. Ideally, the rail splitter should provide ±2.5 V, however the deviations from these values are small enough to not affect the system in any noticeable way.

Reconstruction Filter
The results of the reconstruction filter were measured by applying a signal from a function generator at two tunable modes and measuring the cutoff frequency of the filter on an oscilloscope. At each of these, a sinusoidal signal was fed into the system. After establishing the maximum amplitude of the signal (where the filters have no damping), the frequency of the signal was altered until reaching the cutoff frequency.

![Figure 9: The frequency response of the reconstruction filter when the resistance in the potentiometer is set to 1 kΩ (red) and 85 kΩ (blue).](image)

The two settings that were measured, were the left most and the right most. This is due to the fact that, when using this filter, one would tune the cutoff until the appearance of the signal satisfies one’s desire. Therefore, specific settings are not interesting.

Observing Figure 9 above, one can see that the designed filter has a cutoff frequency ranging from approximately 55 Hz to 4.3 kHz. The implementation was measured to have a cutoff range of 55 Hz to 4.35 kHz, which clearly is within expected deviations for analog circuitry (see Section 5) and Sallen-Key implementations [13, chapter 6, p. 14].
Tunable Filter Circuit
The results of the tunable filter circuit were measured the same way as with the reconstruction filter in Section 10.1, but with eleven (0-10) different settings.

The results comply reasonably well with the theoretical implementation described in Section 8.1, except for a few deviations of the cutoff ranges of the filters. The maximum cutoffs of the low-pass and the high-pass deviates with 400 Hz. The maximum cutoffs of the band-pass deviates with 150 Hz and 250 Hz. These deviations are small enough to be credited to tolerances in components. The minimum cutoffs of the filters did not deviate.

Figure 10: 10a-10c: The frequency responses of the tunable filters, based on the transfer functions provided in [14], when the gain factor is set to 1, 9, 17, 25, 33, 41 (left to right). 10d: The measured cutoff frequencies, \( f_c \), vs. the setting for the tunable filter circuit.
10.2 Digital Systems

Atmel Board
The subsystem that consists of the Atmel Board and momentary switches, features sampling in 46,875 Hz at 10-bit resolution and a DAC that operates in 46,875 Hz at 16-bit resolution. The resolution of the DAC won’t matter due to the ADC’s bit depth shortcomings. Three pins, wired to momentary switches, are set up to an interrupt when they are triggered by a rising edge in the voltage. The Atmel Board registers the pressed button and accordingly modifies the resolution of the ADC or the sampling frequency.

A larger button module with more functionality would help reduce delay as mentioned in Section 8.2. Three distinct button circuits would have to be enlarged, which would drastically increase the complexity and size of the current flexible button module.

Digital Oscilloscope
The oscilloscope software developed has, apart from an oscilloscope, a spectrum display and a simple function generator. All of these have various settings for customization. It works at a sampling rate of 44,100 Hz and 16-bit resolution.

The oscilloscope and the spectrum work as intended by displaying the input signal with its spectrum in real time. One can also scale the axes of the plots and trigger the incoming signal.

The function generator is slightly less effective than intended when changing the frequency. This is due to the large amount of data being processed with each adjustment of frequency.

10.3 Results in Relation to Initial Goals

The goal for this project was to develop a series of analog and digital systems which could interact well with one another. These systems should also be portable and fairly straightforward to use. With these systems, a computer and a power source, one should be able to display some characteristics of signals and simple signal manipulations. This includes filter behavior, ADC, aliasing, ZOH reconstruction and how these interact with each other.

The developed systems satisfy these goals and comply rather well with the theoretical implementation with some minor deviations, which have been described. These deviations however, does not impair the ability to illustrate signal behavior, and therefore does not go against the initial goals.
11 Conclusions

This report presents the design and implementation of a series of analog and digital systems. The main purpose of these systems was to, in an intuitive manner, illustrate the effects of different kinds of signal manipulation.

The systems had to be operable by someone who had not been directly involved with the design and development of said systems, and thus needed to be, to some extent, user friendly.

The systems that were ultimately developed are discussed below.

A Rail Splitter, which supplies the analog circuitry with voltage and allows for the operation of the system without the use of any specialized equipment like dedicated voltage suppliers (Section 8.1). A set of filters, which are capable of displaying concepts related to filtering, and are easily tunable to the needs of the user (Section 8.1). Lastly, in regards to analog systems, a reconstruction filter was developed, (Section 8.1). This allows the user to showcase why reconstruction filtering is needed.

Software was written for our embedded system, which allows the user to show the effects of using different bit-depths when quantizing the signal, the effects of oversampling and undersampling, as well as the risks involved in the latter (Section 8.2). Additionally, the system is set to reconstruct the signal using ZOH.

At last, a digital function generator/oscilloscope was written using MATLAB. The function generator is capable of generating basic signals like sine-, square-, and sawtooth-waves, with different frequencies. The oscilloscope is able to plot ingoing and outgoing signals simultaneously, as well as the frequency spectrum of both signals. Just like a ‘real’ oscilloscope, the user can change the scale of the different axes (Section 8.2).

To Conclude: The system that has been designed and implemented satisfies all of the initial requirements set by our external project stakeholders. Additionally, the digital system is able to change the sampling frequency and quantization resolution without recompiling the software.

12 Future Work

Because this project is related to more than one field of study, there are many ways to continue the development. Some examples of possible future development include:
• **Analog Electronics:** Some examples of future development are aiming for optimization of the circuit with respect to the space required for the components. One could also redesign the circuit in a way that a fewer number of components are required for the implementation. It is also, with very high probability, possible to redesign the circuits all together to accomplish both of the previously mentioned points, by utilizing different components (particularly in the case of the rail splitter).

• **Digital Electronics:** A good way to continue the development on the Atmel board would be to implement digital filters, so that the possibilities of Digital Signal Processing can also be illustrated.

• **Digital Oscilloscope:** During the development of the digital oscilloscope, we found that, while having useful libraries and overall good performance when manipulating fairly large collections of data (arrays with 450000 indices), its performance is lacking when doing function calls.

  Because of this, it would be desirable to rewrite the oscilloscope in a language like Python (which also has open source libraries specialized for scientific computing, like SciPy [5] and NumPy [4]). Another possibility is to only rewrite performance critical parts of the code in C/C++, which can be compiled into external functions with the MATLAB compiler and called from within the program.

• **User Friendliness:** As of now, no part of the system is actually shielded from its surroundings. One could, for instance, design a container for the system, which apart from shielding sensitive components, holds all parts of the system securely in place.

While the system, in its current state, is fully capable of manipulating and illustrating signals, future development and continued work is possible on many aspects of the system.
References


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