

# Project Planning

Project Team: 1

Version 0.2

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0.1		First draft		
0.2	102/14/2014	Minor Changes		

# **1 Introduction**

## 2 Problem

## 3 Theory

### 3.1 System overview

### 3.2 Audio

### 3.3 Digital signal processing

The digital signal differ from its analogue counter part in two fundamental ways. Firstly, in digital applications a signal is represented as a discrete stream of values called samples. The rate, the difference in time, between two samples are called the sampling period  $T$ . The period can be no smaller than the maximum resolution of time offered by the system wich is determined by the system clock. The reciprocal of the sampling period is called the sampling frequency  $f_s$ . In accordance with the sampling theorem, the sampling frequency should be chosen to be twice of the signal bandwidth component of the sampled signal  $f_s = f_{max}$  to avoid aliasing. Secondly, the amplitude of the signal is quantized to a discrete value with the resolution set by the number of available bits (3). Take for instance a system using twos-complement representation and a  $w$ -bits it is possible to represent values between  $-2^w$  to  $2^w - 1$ . Consequentially, a digital signal is discrete in both time and amplitude.

#### 3.3.1 Filters

In general, the task of filters are to remove or separate components within an object. In signal processing applications filters are used to attenuate, retain or enhance frequency components within a target signal (3). The selection of which components and what change e.g in amplitude and phase is done according to frequency. Filters can be built to target different frequencies and in different ways. A myriad of different filters can be implemented, each changing the amplitude or phase of selected frequency components. Filters can be classified as shown in the bulletlist below (3).

- Lowpass: Retains frequencies up to a set cut-off frequency  $f_c$  and attenuates components above.
- Highpass: Attenuates frequencies up a set  $f_c$  and retains components above.
- Bandpass: Retains components between the low cut-off  $f_{cl}$  and high cut-off frequency  $f_{ch}$ , attenuates other components.
- Bandreject: Attenuates frequencies between  $f_{cl}$  and  $f_{ch}$  and retains the rest.
- Allpass: Retains the amplitude of all components but changes the phase of the signal.

**I**n general signal processing the filters are selected in such a way that unwanted frequency components are removed from the signal. What remains after the filtering is thus the components of interest.

**F**ilters can also be used in audio applications as the audio signal is just like any other signal, it contains components of different frequencies. In contrast to other signals, the frequency components in an audio signal is normally divided into three categories, the bass (low frequencies), mids (middle high frequencies) and treble (high frequencies). By applying filters to the audio signal the audio signal can be given a distinctly different sound. If a highpass filter with a  $f_c$  of about 800 Hz, the bass is effectively removed, leaving a signal that sounds colder than unfiltered signal. In audio applications however, the ordinary filter properties are not desired. Rather than attenuate or retain frequency components, selected components are amplified or attenuated and remaining components retained. Filters that accomplish this feat is shelving filters and peak filters where the former exist in two different versions (1).

- Lowpass shelving filter: Attenuates or amplifies frequency components up to a selected  $f_c$ .
- Highpass shelving filter: Attenuates or amplifies frequency components from a selected  $f_c$ .
- Peak filter: Attenuates or amplifies frequency components (How to describe?)

**B**y applying different variations of these filters the audio signal can be shaped and the sound altered in desirable ways. Processing an audio signal with respect to frequency is called equalizing, which is a topic that will be presented in later section.

**I**n audio applications, filters can be viewed in a different light. Instead of filtering out different frequencies it could instead filter components according to time (3). What you get then is the delay based audio effects such as delay, tremolo and chorus all of which is presented in later sections.

**FIR filters** flera tappar  
minimal fäspåverkan och används med fördel i applikationer som kräver minimal fäspåverkan.  
Många tunga beräkningar....

**IIR filters** IIR filters is an acronym for infinite impulse responses. The name comes from the fact that IIR filters include a feedback network. Consequentially, if an impulse is applied at the input the response will never end (?). Because of this feedback network, higher order IIR filters easily become unstable due to rounding errors in the system. In practice, higher order IIR filters are instead realized by connecting several lower order filters (first or second) together to form a series. The second order links in the series are called biquad links. The term biquad refers to a general second order filter. IIR filters offers tight control of the magnitude response at the expense of phase. Thus, IIR filters are used when high requirements are posed on the magnitude and phase matters less (2). Like the FIR filters, all the aforementioned filters can be realized as IIR filters by using appropriate filter coefficients. The general equation for a first and second order IIR filter is shown below:

$$H[z] = \frac{b_0 + b_1 * z^{-1}}{1 + a_1 * z^{-1}} \quad (1)$$

$$H[z] = \frac{b_0 + b_1 * z^{-1} + b_2 * z^{-2}}{1 + a_1 * z^{-1} + a_2 * z^{-2}} \quad (2)$$

This form is called canonical or direct form 1. It is suitable for fixed-point systems due to know delay states.

There exist another form, called direct form 2 shown below.

$$y[n] = b_0 w[n] + b_1 w[n-1] + b_2 w[n-2] \quad (3)$$

where

$$w[n] = x[n] - a_1 w[n-1] - a_2 w[n-2] \quad (4)$$

This form is called direct form 2 and is suitable for floating point based systems. Make transition to the other text by uno where higher order filters are used due to their superior frequency control...

Färre beräkningar

Direct form 1

Canonical Direct form 2

Led in på de olika formerna och varför biquad är speciellt intressanta...

### Discrete structures

- Digital signa processing (short)
- General on filters
- IRR and FIR
- direct forms 1 and 2
- Introduction, What is an EQ?
- Shelving filters
- Peak filters
- Putting it together (conclusion)

#### 3.3.2 Delay

- Your stuff goes here Philip

### 3.4 Interface

### 3.5 Softcore

## **4 Design**

### **4.1 System overview**

### **4.2 Audio**

#### **4.2.1 Equalizer**

- Fixed point for calculations.
- Direct from 1 due to fixed
- Relying on software FP for coefficient calculations
- Interpolation to reach a comprimize between load and precision
- 3-band but flexible
- Internal precision 32 bits, crushed while maintaining sign

#### **4.2.2 Delay**

### **4.3 Interface**

### **4.4 Softcore**



## **5 Evaluation - Reflection**

## **6 Conclusion**

## References

## References

- [1] F. Keiler and U. Zolzer, “Parametric second-and fourth-order shelving filters for audio applications,” in *Multimedia Signal Processing, 2004 IEEE 6th Workshop on*. IEEE, 2004, pp. 231–234.
- [2] R. Storn, “Differential evolution design of an iir-filter,” in *Evolutionary Computation, 1996., Proceedings of IEEE International Conference on*. IEEE, 1996, pp. 268–273.
- [3] U. Zölzer, X. Amatriain, and J. Wiley, *DAFX: digital audio effects*. Wiley Online Library, 2002, vol. 1.