Project Planning

Project Team: 1

Version 0.2

Status	Reviewer	Date
Reviewed		
Approved		

Project identity

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Contents

Version	Date	Changes	Signed	Reviewed
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 aplications 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$fs = fmax$ to avoid aliasing. Secondly, the amplitude of the signal is quantizized to a descrete value with the resolution set by the number of avilable bits (2). 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 atenuate, retain or enhance frequency components within a target signal (2). 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 be implemented, each changing the amplitude or phase of selected frequency components. Filters can be classified as shown in the bulletlist below (2).

- 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 frequeny 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 frequecy components in an audio signal is normaly divided into three categories, the bass (low frequencies), mids (middle high frequencies) and trembel (high frequencies). By applying filters to the audio signal the audio signal can be given a distincly 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 accomplishes 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 aselected f_c.
- Peak filter: Attenuates or amplifies frequecy 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 persented in later section.

I n audio appliations, filters can be viewed in a different light. Instead of filtering out different frequencies it could instead filter components according to time (2). What you get then is the delay based audio effects such as delay, tremolo and chorus all of which is persented in later sections.

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

IIR filters IIR filters is an acronym for infinite impuls respons. The name comes from the fact that IIR filters include a feedback network. Consequeltially, if an impuls is applied at the input the respons will never end (**?**). Because of this feedback network, higher order IIR filters easily become instabil due to rounding errors in the system. In practice, higer order IIR filters are insted realized by connecting several lower order filters (first or second) together to form a series. The second order links in the seires are called biquad links. The term biquad referes to a general second order filter. IIR offers tight cotrol of the magnitude respons at the expense of phase. Thus, IIR filters are used when high requirements are posed on the magnitude and phase matters less (**?**).
Make transition to the other text by uno where higher order filters are used due to their superior frequency control...
Färre beräkingar
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] U. Zölzer, X. Amatriain, and J. Wiley, *DAFX: digital audio effects.* Wiley Online Library, 2002, vol. 1.