Slovak University of Technology in Bratislava
Faculty of Electrical Engineering and Information Technology
Department of Telecommunications

Dipl.-Ing. (FH) Alexander S. Treiber

Design and Benchmarking of Acoustic Feedback in Human Machine Interface

Dissertation

Reference Number: FEI-10849-55317
Supervisor: Doc. Ing. Gregor Rozinaj, PhD.
Co-Supervisor: Prof. Dipl.-Phys. Gerhard Gruhler
Study Program: Telecommunications
Study Field: 5.2.15, Telecommunications
Revision 1.01
Bratislava, May 2011

Abstract

The number of functions offered by modern cars has constantly risen over the past years. On the other hand, the expectations of the drivers regarding usability and user experience have increased as well. These trends have led to today's complex user interfaces which commonly use very few input devices which have to be used frequently.
These control elements should transport the impression of high quality since they are the actual connection between the driver and the vehicle. A high-quality control element should *feel* expensive. This feeling can be divided into four impressions:

- Appealing visual design
- Expensive materials (metal feels different than plastic
- High tactile quality (low slackness, precise)
- High acoustic quality (Pleasant auditory feedback)

This work focuses on the high acoustic quality, more specifically on the definition of high-quality target sounds and methods for acoustical benchmarking of control elements.
As of now it is common practice to specify the acoustic feedback of a switch only based on absolute maximum values for it's sound pressure. This means that only the amplitude of the feedback but not it's actual *sound* is specified. Furthermore the specifications are often based on the opinions of very few experts.
This project introduces a methodical approach to define target sounds for electromechanical control elements. The method uses subjective trials with naïve subjects as a basis for the specification of an optimized acoustic feedback stimulus. A suitable synthesis algorithm along with purpose-built hardware for sound evaluation within a meaningful context is presented as well.
This work also introduces a method to evaluate the acoustic feedback of control elements for conformity with the methodically designed target sounds. The evaluation model uses both the overall amplitude as well as correlation of spectral and temporal features. Allows more reliable prediction of the acoustic feedback than with the current consideration of sound pressure alone.
Finally, results of trials are presented which show that subjects are able to work faster and more precise with well-designed acoustic feedback than they are able without feedback or with unpleasant feedback.

Anotácia

Počet funkcií ponúkaných užívateľovi v súčasných automobiloch neustále Na druhej strane rastú aj nároky vodičov a cestujúcich na užívateľský komfort. Tento trend podporuje komplexné užívateľské rozhranie, ktoré využíva pomerne obmedzené vstupné zariadenia používané pomerne často. Riadiace prvky musia spĺňať okrem iného aj požiadavku na pocit vysoko kvalitného zariadenia. Tento pocit môžeme rozdeliť do 4 kategórií:

- Pôsobivý vzhľad
- Luxusné prevedenie (drahé materiály)
- Pocit komfortného dotyku (pohodlie, presnosť)
- Kvalitná akustická spätná väzba (príjemný zvuk)

Predkladaná práca sa zameriava na kvalitnú akustickú spätnú väzbu, presnejšie na definovanie vysoko kvalitných zvukov a metód pre akustické vyhodnotenie prvkov. Doteraz bolo bežné definovať kvalitu spätnej väzby spínača iba podľa intenzity zvuku. To znamená, že sa sledovala iba amplitúda akustickej spätnej väzby ale nie typ zvuku. Okrem toho bola špecifikácia kvality založená na subjektívnom hodnotení iba niekoľkých expertov.

Tento projekt sa venuje metodickému prístupu na definovanie zvukov pre elektromechancké riadiace prvky. Predkladaná metóda používa subjektívne testy jednoduchých prvkov ako základ pre špecifikáciu optimalizovanej akustickej spätnej väzby. V práci je predstavený aj vhodný algoritmus syntézy spolu s návrhom technického vybavenia pre vyhodnotenie kvality zvuku.

Práca ďalej predkladá metódu na vyhodnotenie akustickej spätnej väzby riadiacich prvkov pre porovnanie s požadovanými a navrhovanými zvukmi. Vyhodnocovací model používa porovnanie obálky ako aj koreláciu spektrálnych a časových vlastností. Tento prístup ponúka kvalitnejšiu predikciu a vyhodnotenie spätnej väzby ako doteraz známe metódy založené iba na intenzite.

V neposlednom rade, výsledky testov ukázali, že užívatelia sú schopní pracovať rýchlejšie a presnejšie s prvkami s kvalitnou akustickou spätnou väzbou ako s prvkami so žiadnou alebo nepríjemnou spätnou väzbou.

Contents

List of Figures

List of Tables

Chapter 1

Introduction

The number of functions offered by modern cars has constantly risen over the past years. On the other hand, the expectations of the drivers regarding usability and user experience have increased as well. This trends have led to today's complex user interfaces which commonly use very few input devices which have to be used frequently. These control elements should transport the impression of high quality since they are the actual connection between the driver and the vehicle. A high-quality control element should *feel* expensive. This feeling can be divided into four impressions:

- Appealing visual design
- Expensive materials (metal feels different than plastic
- High tactile quality (low slackness, precise)
- High acoustic quality (Pleasant auditory feedback)

This work focuses on the high acoustic quality, more specifically on the definition of high-quality target sounds and methods for acoustical benchmarking of control elements.
Apart from the perceived quality of a switch, the acoustic feedback also provides the user with the information that the control element has been operated without the need to look at a display. This in turn increases the security of operation since the distraction of the driver is lowered.

Chapter 2

Objective of this Work

In this chapter the current methods of specifying and benchmarking acoustic feedback stimuli are briefly introduced, followed by a discussion of their strenghts and weaknesses. Subsequently an approach to improve the current method is proposed in a generalized form followed by a synopsis of the concrete improvements.

2.1 Current Methods of Specifying and Measuring Transient Sounds

As explained in the previous chapter, the specification of transient sounds is important in a number of industrial applications. It is common to simply limit the maximum value of the allowed sound pressure level. In the area of luxury cars desired sounds are usually defined by a small number of experts. Typically the definition is based on the selection of a control element with the desired acoustic feeling from a range of physical prototypes. This prototype is henceforth defined as a benchmark against which samples from the production car are occasionally compared.

2.2 Discussion of Limitations and Shortcomings of Currently Used Methods

The current common practice of specification and evaluation of transient feedback signals involves mainly expert listeners. This section outlines the most important weaknesses of the current practice and hence provides the foundation for the derivation of improvements.

2.2.1 Specification of Maximum Sound Pressure

Depending on the demands a specification based on sound pressure level alone might be sufficient. However, this is only true if it is required that the transient sound is not noticed by a user. If it is required that a transient sound *can* be heared it becomes

obvious that the simple specification of a maximum is insufficient. Musical instruments can be used as obvious examples here:
A pizzicato string sound and the sound of a snare drum can both be recorded and normalized to the same level. Yet the two sounds are clearly not very similar, even if they both are transient. One can argue that the tonal nature of the string and the noiselike nature of the drum is the reason for this. But the same effect occurs for two different kinds of drums and even for the same drum played with different sticks or mallets.

2.2.2 Practical Implementation of Subjective Evaluation

The sounds of control elements are generally noiselike. The sounds depend on the mechanical construction of a certain control element as well as tolerances caused by the manufacturing process. In the musical instrument analogy the mechanical construction can be regarded as the type of drum and the tolerances can be seen as the sticks. Sound specifications - or sound preferences to be precise are typically stated by a single expert listener or a panel thereof. The panel which defines the desired sound selects a control element which they consider acoustically suitable. Usually, a subjectively very "good" example is picked as a reference. Furthermore one or more examples of "bad" sounds are picked as well. Samples which are taken from the production line for quality control are later on compared by either the same or other experts who judge the acoustic quality.

Expert Listener Panel

While this method usually leads to relatively exact classifications of sounds within a specific panel, the results can vary when the same sounds are compared by other people. This can be an issue in larger companies in which the manufacturing process takes place at other sites than the development. Furthermore, if the decision is only dependent on a single person, the desired sound of the average customer may or may not be in line with this person's taste. Finally, in current development processes prototypes equipped with every possible control element have to be built to enable the expert panel to test the sounds. Building prototypes costs both time and money, prohibiting too many iterations of prototypes to be built. It would therefore be an advantage if

- the specification / selection of a desired sound was done by as many people as possible
- simple yet effective rules existed to check, if a sound meets the specification without the help of all the people whose opinion the specification is based on
- a solution existed which enables designers to design specifications based on acoustic prototypes

Unwanted Multimodal Evaluation

Apart from the time and cost effect of the evaluation of prototypes there is another constraint when physical protoypes are used. It might be true that a specific control element which sounds best to the expert panel also feels best. But it also might be that an acoustically good sample is rejected because of poor haptic feeling or because of it's visual appearance as the impressions of other senses affect the rating. To eliminate those unwanted multimodal interactions it is therefore reasonable to isolate the acoustic impression. Basically, this can easily be achieved by recording the sound of a control element and playing it back. However, in this case the sound is isolated from it's context. However, there are recent works [9] which criticize the practice of judging a sound's impression out of context. There are approaches to eliminate these interactions in haptic research tools [1] by building an electronically programmable device. In acoustic research a team at NASA [2] conducted experiments with subjects which wore thick gloves and were blindfolded in order to impair there tactile and visual senses. Ideally there should be an acoustic rapid prototyping tool for control elements which is able to do the following:

- Play back either recorded or synthesized acoustic signals of a control element.
- Play back stimuli in an interactive way.
- Play back stimuli in a meaningful context.

2.2.3 Highly Subjective Target Sound Specification

Finally, the current method of defining a target sound is highly subjective as the size of an expert listener panel is typically relatively small and does not provide a sufficient statistical basis. Exchanging the panel members can affect the specifications defined by that team. Furthermore, specifications which are designed to fit the taste of a small panel or even one single individual can easily be criticized.

2.3 Proposed Improvements

In order to overcome the outlined shortcomings of the currently common approach to specification and analysis of transient signals the following improvements to the methodology for analytics and the approach to specifications are proposed:

2.3.1 Introduction of Spectral and Temporal Properties

As shown above the use of the overall sound pressure level is insufficient to describe the acoustic quality of a signal.

It basically is possible to record an entire sound, use it as future reference and reject all sounds which do not exactly match this reference without any tolerances.

This would however result in an over-specification of the signal as even sounds which cannot be distinguished by average listeners would be ruled out. Therefore it is proposed to take into account the temporal structure and the sound pressure level in several frequency bands for a relatively broad classification of sounds. It would be preferable if the used frequency bands are compliant with bands commonly used in acoustic analysis such as octave- or 1/n-octave filters.

2.3.2 Introduction of an Acoustic Rapid Prototyping Methodology for Sound Design

As an improvement to the general approach of selecting and specifying target sounds a toolchain for acoustic rapid prototyping of transient signals is proposed. It consists of four components:

- Recording Equipment
- Synthesis Algorithm
- Interactive Playback Device
- Methodology

The recording equipment is used to acquire the signals emitted by existing electromechanical control elements. The synthesis algorithm enables the user to create desired synthetic feedback signals from scratch. The interactive playback device is used to present the signals within subjective trials. The main feature of the playback device will be that it basically looks and feels like an actual control element but the acoustical character is interchangeable by software. Finally and most importantly a methodical approach to the design of a signal has to be specified.

2.4 Goals of this Work

As a conclusion of this chapter, one can formulate the following goals of this work:

- Introduction of a methodology for the acoustic rapid prototyping of transient acoustic feedback signals.
- Introduction of an analysis algorithm for the objective prediction of the acceptance of an acoustic feedback stimulus.
- Experimental verification of the positive influence of suitable acoustic feedback to speed and accuracy of operation.

The following chapter will explain the general approach to achieve this goals.

Chapter 3

Theoretical and Practical Approach

After the challenges of this project have been stated in the previous chapter the actual approach to the proposed solutions will be outlined. As the human sense of hearing and the emotions evoked by specific noises are highly subjective and vary from person to person it is necessary to involve several subjective experiments in the process. Since even a very short acoustic signal has an almost infinite number of degrees of freedom in terms of temporal structure and frequency, the meaningful parameters have to be identified. This will happen in an iterative series of experiments in which the results and conclusions from one experiment will be used as premise for the subsequent one.

3.1 Iterative Examination of Sound Properties in Subjective Trials

Literature [10] suggests that some parameters (i.e. amplitude) have a larger impact on the subjects' reaction than others and eventually mask the effects of other parameters completely. At the beginning of the project there were no published results about this exact type of sound. Basic research has been done regarding the influence of signal duration on the sensation of perceived loudness. This experiments however were done using signals dissimilar to realistic click sounds. In sound quality analysis it is a common starting point to use existing real sounds. For this reason the proposed suitable measuring/recording equipment had to be build before the start of the subjective trials. The concept is to use selected recordings in the first trial, extract at least one feature which influences the subjective rating of the sound and normalize this feature in the next iteration.

3.2 Isolation of Properties Relevant for Subjective Perception

The isolation of specific relevant signal properties happens after a series of subjective trials is finished and the subjective results of all probands for all sounds in the test have been statistically analyzed. This statistical analysis results in a preference ranking of the used stimuli. The stimuli itself will be acoustically analyzed and ranked in the order of the respective analysis feature (i.e. amplitude, peak/rms ratio, spectral peaks...). Ideally there will be a high correlation of one or more signal parameters to the preference of a sound.

3.3 Derivation of a Synthesis Model for Transient Sounds

A synthesizer for the creation of mathematically describable yet realistic sounding stimuli will be developed in parallel and eventually replace the use of recorded signals in later iterations. As the derivation of the synthesis model is an iterative process as well, it will lag behind the isolation of properties by one step. Results gained from a subjective trial which lead to a new identified meaningful signal property will be included into the synthesis model. As soon as the model is sophisticated enough to produce realistic sounding stimuli it can be used for adaptive test procedures in which the subjects have control of the details of the very sound they have to evaluate. This approach leads to statistical results of the ideal ranges which each of the identified parameters should have.

3.4 Derivation of an Analysis Algorithm

After a certain number of iterations the synthesis model will eventually be able to create realistic clicks from a relatively small number of control parameters. Based on the premise that these parameters are sufficient to describe a stimulus the analysis algorithm will be based on the same model. It shall analyze recorded signals in order to resynthesize them using the extracted analysis results fed into the synthesis algorithm.

3.5 Experimental Evaluation of Analysis Algorithm in a Subjective Trial

Finally the analysis algorithm has to be evaluated in a subjective trial. The proposed method is to

1. use the synthesizer to adaptively set the desired sound.
2. analyze the subject's desired sound.
3. select sound from the sound database which this subject is likely to like or dislike.
4. present the selected sounds to the subject.
5. compare the predicted acceptance ranking to the actual acceptance.

Chapter 4

Preparation

This chapter briefly outlines supporting work and preparation which had to be accomplished before actually beeing able to make valid measurements of control elements. It also explains the signal properties of the feedback of a large number of physical control elements, which provide insights on the variation of several parameters when dealing with mass-produced parts.

4.1 Measurement Equipment

The measurement equipment consists of an integrated measurement system for the acoustic feedback of control elements which has been designed and built during the course of this project on the one hand. On the other hand the industry-standard ArtemiS software by Head Acoustics has been used. This components will be briefly introduced in this section.

4.1.1 Robotic Actuation Unit

To acquire reliable and reproducible measurements of mechanical control elements, a bespoke measurement system has been designed and developed. It's main component is a robotic actuation unit which is placed in an anechoic test chamber (see *figure 4.1*).

The actuation unit can operate both rotary encoders and push-buttons using a stepper motor. The motor is driven by analog sine and cosine waves to minimize vibration and hence noise. The device is big enough for entire automotive components like radio head units.

The anechoic test box which surrounds the actuation unit provides free-field conditions as well as ambient noise rejection exceeding 30 dB for frequencies above 500 Hz. More detailled information can be found in [15].

The entire system is controlled by custom software implemented in Matlab. The software automatically detects and extrects the relevant signals for each measurement. The extracted signals are subsequently stored in the standard WAV-format for further analysis.

Figure 4.1: Final version of the actuation robot in it's anechoic test box

4.1.2 Analysis Tools

While custom analysis methods have been developed during the course of this project, the industry-standard ArtemiS software by Head Acoustics has been extensively used as well. This software package can perform a number of standardized analysis methods, including frequency analysis based on FFT or filter implementations, extraction of the psychoacoustic parameters mentioned above as well as analysis methods developed by Head Acoustics which are suitable to detect transient events in noisy backgrounds.

4.2 Description of Acoustic Feedback Signal

The acoustic feedback signal of control elements is typically generated when springs inside the switch snap into place. The signal is usually referred to as 'click'. In a typical microswitch the spring snaps when enough force for actuation is loaded onto the switch. The spring then connects the contacts of the switch. Sound is radiated upon impact of the spring on the contacts. In the rotary encoder which is used in this work the sound

is caused by a radial spring which generates both haptic and acoustic feedback. In this section, the properties of the acoustic feedback which occurs in typical rotary control elements is examined in detail, along with an inspection of the mechanical construction of one particular type of encoder.
In general, the sound is highly impulsive with only about 1 ms rise and typically 15 ms decay time. Depending on the type of the switch this values vary but the length of the signal is in any way shorter than 100 ms which is considered to be the value above which steady-state conditions apply [10] .
Due to the impulsiveness of the sounds, the short duration and the small dimensions of typical control elements, the feedback sounds lack frequency components below 500 Hz. A typical frequency over time plot of a rotary encoder can be seen in the following *figure 4.2*.

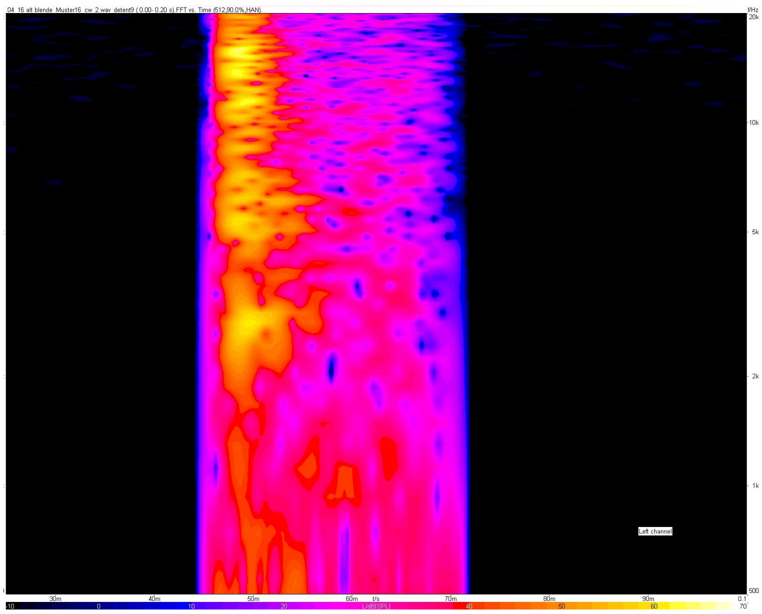

Figure 4.2: Typical acoustic feedback signal (”click”) in a spectrogramm

In this example the frequency components with the highest intensity are well above 10 kHz, the first local maximum is roughly at 3 kHz. As can be seen in the figure, the maxima are relatively broad in the frequency domain and that there is no structure observeable in the frequency domain. This means, that the feedback signal is not tonal, the spectrogram shows the properties of noise.
For the preparation of both the early stages of the subjective trials as well as the evaluation of the proposed analysis method a large number of recorded feedback signals of real control elements was required. For this reason, several hundred rotary encoders have been measured in both clockwise and counterclockwise direction. As each of this encoders features 16 or 20 clicks per rotation this lead to over 30000 recorded single clicks.
This extensive measurement series showed that encoders of the same type show variations in the amplitude of the acoustic feedback exceeding 15 dB. The amplitude envelope on the other hand is relatively stable, showing only slight variations within one

type of encoder.
Larger variations of course occur across different types which can happen due to several reasons:

- Different geometry / stiffness of the spring
- Different number of detents and hence changed geometry of the locking disk
- Other materials used
- Use of viscous dampers (grease)

In the actual use of the control element, there is the even bigger influence of the mounting point of the device. An encoder which is soldered on a very small PCB in a small plastic panel will sound different from an encoder which is screwed to the comparatively large metal case of a radio head-unit.

4.3 Interactive Acoustic Simulator

The main tool used in subjective experiments is the so called acoustic simulator. This is a control element with no inherent acoustic feedback. The feedback is generated artificially and played back using loudspeakers. The encoder itself has very little friction and no audible feedback. Furthermore it features a resolution of 256 increments, typical encoders in user interfaces feature 16 to 24 increments per rotation. The encoder is connected to the control PC via a microcontroller which translates the high resolution signal of the encoder to a resolution of 16 increments per rotation. The resulting signal is sent to the PC and triggers the sound playback. The high-resolution input is used to avoid multiple sound playback if a subject jitters.

In order to be perceive the acoustic feedback in a realistic way the playback of the stimulus has to occur within a certain time frame after the sound is triggered. The research of Adelstein [2] showed that the acoustic feedback has to occur within 25 ms after a haptic sensation. This constraint has to be kept even though in this study no haptic feedback is used since playback of sound longer than 25 ms after a subject has stopped turning the encoder has to be avoided.

For spatial congruence, the loudspeaker used for playback is placed horizontally over the rotation axis of the encoder so that there is no displacement on the horizontal plane. The displacement on the median plane is 7.5° and hence within the constraints presented by Altinsoy [16] .

The subjects operate the encoder with the right arm, the center axis of the loudspeaker is aimed at the subject's right ears. The stimuli are presented in a purpose-built quiet environment with a reverberation time of 0.2 s. The reverberation time is frequency independent in the relevant frequency band. The details of the room are explained in [17].

Chapter 5

Synthesizer

This chapter describes the basic structure and specific features of a synthesizer for acoustic feedback. As mentioned in the previous chapter, the development of both the synthesizer and the analysis model were iterative processes and ran in parallel over the course of the project. Only the relevant fundamentals and the final stage of the synthesizer are described in this chapter.

5.1 Basic Principle

The basic principle of the synthesizer has been proposed by Gaver [18]. Gaver stated in this paper that impact-like sounds are exponential decays of a sine wave and several hamonics as given in the following formula:

$$G(t) = \Sigma_n \phi_n e^{-\delta_n t} cos \omega_n t \tag{5.1}$$

$G(t)$ denotes the resulting waveform over time, consisting of n partials. ϕ_n is the initial amplitude, δ_n the damping constant and ω_n the frequency of partial n.
Gaver cites Freed's [19] work, in which the hardness of a mallet which is used to strike objects was examined regarding the effect on the resulting sound. The ratio of low to high frequency energy in the sounds as well as its change over time served as most powerful predictors for the subjects' hardness judgement.

5.2 Modification of Amplitude Envelope

The described algorithm synthesizes a sound which instantly starts at it's peak amplitude. As the examination of the physical control elements showed, this is not the case in natural feedback sounds. As the work of MacAdams [20] shows, the rise or *attack* phase of the sound of transient musical instruments is critical to their identification. For this reason, Gaver's original algorithm has been enhanced by an attack phase.

5.3 Introduction of Wavetable Oscillator

While Gaver's approach of using harmonic signals works well for impact sounds of i.e. a hard object against a metal bar or a drum head, it does not produce satisfactory results for control elements. One of the subjective experiments (see *chapter 6.2*) presented in this thesis showed that subjects prefer sounds based on noise over harmonic sounds. For this reason the sine oscillator of Gaver's model was replaced with a wavetable oscillator.
Wavetables may contain arbitrary waveforms and subject's prefer the feedback sounds to be based on noise. As explained in *chapter 6.3* , the spectrum is important to the impression of the sound. In order to gain control over the spectrum the model had to be expanded to multiple wavetable oscillators, each containing band pass filtered noise. To determine the width of the filters it is neccessary to understand that due to the limitations of the human sense of hearing only a limited number of frequencies can be perceived simultanously. Blauert [21] proposed 23 bands which cover the entire frequency range of human hearing. Especially at frequencies above 1 kHz these bands are similar to third-octave-bands which are common in acoustic analysis. It is therefore proposed to use the 15 third octave band filters which are defined in the DIN EN 61260 [22]. The amplitude for each of this wavetable oscillators can be controlled individually. It is now possible to use either one amplitude envelope for a sum of all oscillators or an individual amplitude envelope for each oscillator. The latter provides the possibility to model longer decay times at certain frequency bands which might occur as a result of resonances.

5.4 Control Parameters

While this synthesis model is generally suited to produce mathematically describeable synthetic feedback sounds it is relatively difficult to use, due to the number of parameters. Three parameters have to be set for each oscillator, which leads to 45 parameters in total. While this complexity can be handled by an automatic analysis tool which analyses an existing sound to extract it's properties for resynthesis, it can not be controlled manually by an untrained test subject. For use in subjective test procedures the number of control parameters therefore has to be reduced significantly. As the results from a subjective test (see *chapter 6.3*) have shown, the actual length of the attack phase, at least within bounds which exist in physical control elements, have very little impact on the acceptance of their sound. Therefore it has been decided to lock the attack time to a value of 2 ms, which is near the average of a wide number of measured buttons. Furthermore, subjects have frequently stated to like or dislike *dull* or *bright* sounds, rather than *low* or *high* sounds. This means that they do not want to hear very narrow band sounds of a certain pitch but broadband signals with a certain emphasis on low or high frequencies. Rather than allowing the subject to adjust the level for each oscillator it is proposed to simplify the settings by using a model for the sensation of brightness. This can be implemented as a low- or high-pass filter with adjustable slope or by adjusting the levels of each envelope accordingly (see *figures* 5.1 and 5.2).

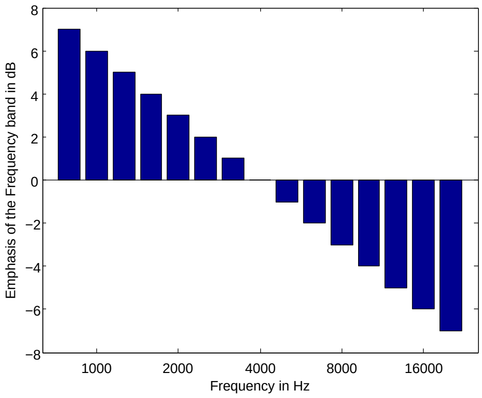

Figure 5.1: Example of a dull spectrum

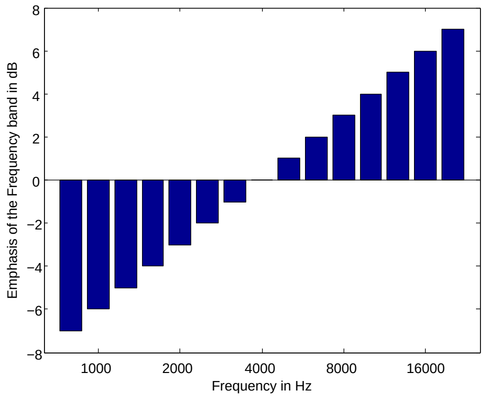

Figure 5.2: Example of a bright spectrum

If the spectrum is adjusted in this way a parameter for the overall amplitude is required, since chances in the spectrum parameter only lead to relative changes of the oscillator levels, not to general amplification or attenuation.

5.5 Synthesis Model

The intention of the synthesis model is to use the control parameters mentioned above to generate virtual click sounds which sound realistic but are based on a relatively small set of parameters.

5.5.1 Oscillator

As explained above, the synthesis model uses 15 fixed-waveform table-lookup oscillators [23]. The 15 wavetables are obtained by a filter-bank of 3rd-octave bandpass filters which is used to extract 15 3rd-octave signals from an oscillator which generates uniform noise. The 15 signals will subsequently be processed independently.

5.5.2 Envelope Generator

Each oscillator requires an independent amplitude envelope with a logarithmic attack and an exponential decay as explained above. Based on this two input variables an amplification factor for each sample of the according wavetable has to be calculated. Each of the 15 wavetables is subsequently multiplied with the respective amplitude envelope. Finally, each oscillator is attenuated according to the spectrum which has been set.

Chapter 6

Subjective Experiments

This chapter explains which subjective experiments have been conducted using the tools and methods explained in the previous chapters. As explained above, this work uses a set of subjective experiments to derive an analysis and synthesis model for definition and benchmarking of transient feedback signals. The process is iterative, the described experiments will therefore start with recorded stimuli of real, electromechanical control elements and then continue with the use of synthesized stimuli which are specifically designed for the respective experiment. The insights gained from each experiment will be used to develop the synthesis and analysis model one step further.

6.1 Paired Comparisons of Recorded Sounds

In the first trial, eight recorded sounds were used as stimuli. The sounds were selected from the sound database by expert listeners and were not amplified or filtered in any way. The trail was a paired comparison with a randomized sequence of pairs. All eight sounds were presented to the subjects as primers at the beginning of the experiment. The subjects were asked to select their preferred sound from each pair. At the end of the experiment
There was a strong correlation between amplitude and acceptance of the sound. The loudest sounds in the test were rated very poor, the quiet sounds received relatively good ratings. Interestingly however, the rating of the most quiet sound showed a very high standard deviation compared with the other stimuli. Two conclustions can be drawn from this experiment:

- Quiet stimuli are preferred over loud ones.
- If the stimulus is too quiet, many subjects reject it, so there seems to be an ideal range for the amplitude.

6.2 Paired Comparisons of Loudness Normalized Synthetic Sounds

Goal of the second test was to examine the influence of the spectrum of a click sound on it's acceptance. Again the test consisted of eight sounds which had to be compared. One sound was the sound which has been perceived best in the first series of tests. It was used as a reference. The other seven samples have been synthesized in Matlab. The stimuli were based on sine waves of 2; 4 and 8 kHz, a 2 kHz square wave, a 2 kHz sawtooth wave, white noise and filtered noise. The amplitudes of the signals were modulated with an envelope on order to match the amplitude envelope of the recorded click. The envelope uses linear attack and release slopes. The amplitudes of the stimuli were normalized using the sone-scale which takes the frequency dependance of the human sense of hearing into account.

The test subjects considered the recorded benchmark sound best. It was followed by the signals based on noise, the signals based on sine wave received very poor ratings, some subjects complained about the artificial sound of the stimuli based on harmonic signals. The insights gathered during the first two experiments were used to develop the synthesis algorithm described above.

6.3 Rating of Synthesized Sounds

The goal of the experiment was to determine the importance of each of the synthesis algorithm's control parameters for the acceptance of a stimulus. After the isolation of parameters which have great influence on the acceptance, less important parameters can be locked in further trials and hence reduce the number of degrees of freedom.
The stimuli which were used in this trial featured the 5 and 95 percentile values for attack and decay time based on the sound database as well as the averaged spectra for three different types of encoders. This leads to 24 stimuli which were rated using a nine-point scale ranging from extremely good to extremely bad. This method has been prefered over the paired comparison due to the high number of stimuli.

The experiment showed that the influence of the attack time within the limits which occur in reality can be neglected while the influence of the decay time is significant and the spectrum tendencially affects the acceptance as well.

6.4 Search for Just Noticeable Differences and Reaction Time Measurements

Goal of this experiment was to identify the detection threshold for changes of a signal parameter (also known as just noticeable differences of JND), the identification of suitable ranges for this parameters and the effect of acoustic feedback on the speed and accuracy of operation.

6.4.1 Results

50 percent of the subjects will not notice a level difference of about +/-1.5 dB. The highest recorded values for just noticeable differences are -2.5 dB and +5 dB.

Regarding the decay time, the subjects were able to notice changes as little as 1 ms, which is illustrated in the following *figure 6.1*:

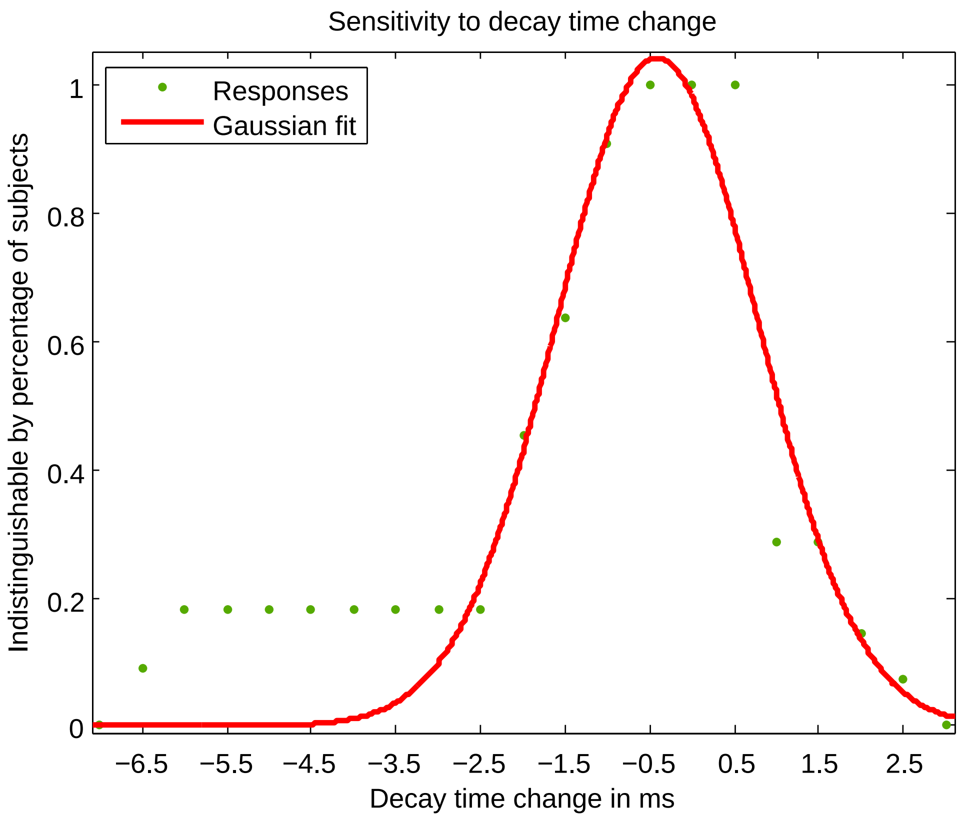

Figure 6.1: The figure shows the percentage of the test subjects who noticed a certain change of the decay time which is denoted on the x-axis. The dots are actual data points, the line is a gaussian fit

It can be seen that the peak of the gaussian fit is offset towards shorter decay times by about 0.5 ms and the raw data suggests that subjects are more sensitive towards increases of the decay time. This can be explained by regarding the logarithmic nature of the decay as relatively short increments of the decay time lead to noticeable increments in perceived loudness.

Evaluation of the data from the second part of the experiment, in which the subject's task was to design their individual feedback signal provides information on how an auditory feedback of a rotary encoder *should* sound like.

The majority of the subjects set the sound to decay within 30 milliseconds and set the peak amplitude within a 10 dB range. While 10 dB means a perceived doubling of the loudness of the stimulus it is in a similar magnitude than normal tolerances which occur in typical electromechanical encoders due to tolerances in the manufacturing process. Furthermore, two thirds of the subjects preferred one of the three available spectra while another spectrum was almost completely rejected by the subjects. This shows that even if the impact of the spectrum on the actual acceptance is small compared to the impact of the amplitude as was pointed out in experiment three, there are certain predominant preferences among all subjects regarding the spectrum.

Impact of the Acoustic Feedback Quality

13 out of 21 subjects were the quickest with their individually set-up sound while only 3 subjects were quickest without any feedback. On average, subjects were 74 ms quicker when feedback was present. Interestingly, the subjects are on average 33 ms quicker with their individual feedback signal than with the very loud but annoying sound. This proofs that it it not only interesting from a marketing point of view to provide pleasant feedback but from a usability perspective and ultimately from a safety perspective as well. Regarding the accuracy of the subjects it is also the self-defined stimulus that leads to the best performance. The subjects typically performed best using the self defined feedback while the clearly audible but unpleasant feedback increased the rate of error to almost twice the result using the individually optimized stimulus.
In total, the effects auditory feedback on the performance of subjects selecting values using a rotary encoder as an input device can be summed up in the following *table 6.1.*

Acoustic Feedback	Mean Error in Detents	Mean Reaction Time in ms
None	1	2200
Self-Defined	0.67	2141
4 kHz Burst	1.24	2170

Table 6.1: This table sums up the results of the experiment regarding the effect of pleasant auditory feedback on both accuracy and reaction time

This shows, that pleasant auditory feedback increases both reaction time (and hence security) and accuracy (and hence user-friendlyness) of a rotary encoder as an input device.

Chapter 7

Proposed New Method for Prediction of Acceptance of Acoustic Feedback

Since the perceived quality of a sound is an impression generated by the reflective part of the human perception it is an impression which varies greatly from listener to listener. Furthermore the quality of a sound is context dependant. Therefore it is not feasable to design a method which can automatically identify a sound which is good per se and hence the proposed method uses a subjectively selected reference sound. It is assumed that this reference sound is perceived as good by the subject, or at least as good as possible within practical limitations. This sound shall subsequently be used as a benchmark to compare arbitrary sounds of the same type against. It is important to understand that the sounds which are compared have to be of the same nature.

7.1 Outline

At the beginning of automatic feedback prediction the arbitrary reference sound has to be analyzed first. The analysis results are regarded as reference values for the subsequent comparison with sounds that have to be examined. The general structure is very similar to the approach of the proposed synthesis method: The analysis method is based on the extraction of three signal properties of the entire input signal as well as from 15 3rd-octave bands of the input signal. The input signal is supposed to be time-discrete and quantized.

- Peak Amplitude
- RMS Amplitude
- Amplitude Envelope (leading to attack and decay time)

The detection of the RMS amplitude is an addition to the parameters which are used in the synthesis model. It is necessary as the measured signals are subject to microphone and amplifier noise. If the peak amplitude is not at least 10 dB higher than the RMS amplitude, the recording is considered as too noisy and will therefore be discarded.

Envelope Detection

The results of the envelope detection are subsequently used to derive the attack and decay times of the signal in a two-step procedure.

First, the absolute peak value and the respective sample index i_{max} of the input signal are detected. The input signal shall be

$$f(i), 0 < i < f_s/5, i \in \mathbb{N} \tag{7.1}$$

with f_s being the sample frequency.

Beginning from this absolute peak of the signal the algorithm moves forward and backward through the dataset and marks the local absolute maximum for the interval from the first sample to the current sample for the attack portion:

$$e(i) = MAX(f[1:i]), 0 < i < i_{max}, i \in \mathbb{N} \tag{7.2}$$

and from the current sample to the last sample for the decay portion:

$$e(i) = MAX(f[i:f_s/5]), i_{max} < i < f_s/5, i \in \mathbb{N} \tag{7.3}$$

The result $e(i)$ is a raw amplitude envelope as seen in *figure 7.1* which is subsequently used to determine attack and decay times.

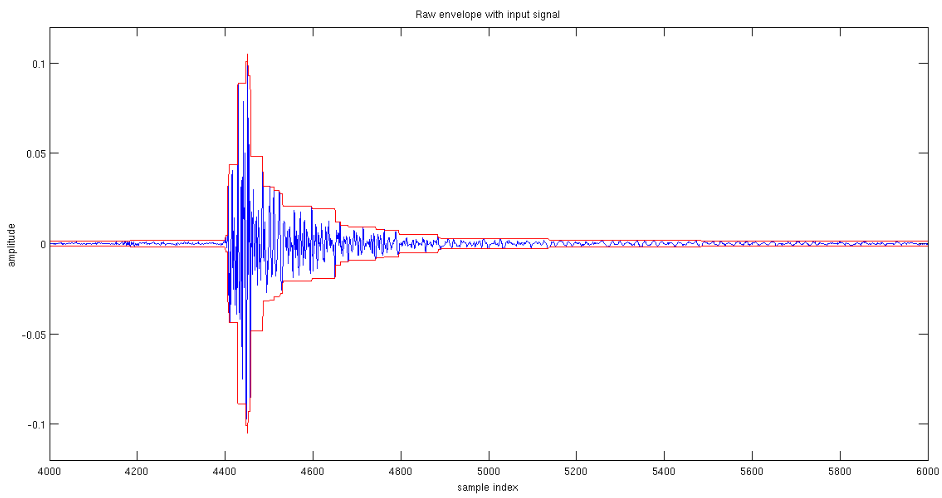

Figure 7.1: The blue line is the input signal, the red line the raw amplitude envelope as it is acquired using the described method, the image has been zoomed in to the actual click sound

Attack Time

First, the attack time is calculated. The peak value of the amplitude envelope $MAX(e(i))$ and sample index i_{max} of the peak value are used as reference. Furthermore, the mean value of the envelope is calculated. The detection algorithm starts at the sample index of the peak value and is then decremented until the value at the current sample index is either below the mean value or below 25 % of the peak value. The difference between this sample index and the index of the maximum is considered the attack time in samples which can easily be converted into seconds.

Decay Time

The subsequent calculation of the decay time is done in a similar way. It is important to understand that the decay phase of the amplitude envelope is considered to be exponential. Like the attack time detection the routine first detects the peak value of the amplitude envelope as well as the sample index i_{max} of the maximum. Every sample before the detected maximum will be discarded for further processing, so the maximum will be located at sample index $i = 1$. Starting from this position the index is incremented until the value at the current index position is less than the mean value of the decay phase. This index shall be called i_{noise}. At this point the measured signal disappears in the noise floor. Due to the exponential nature of the decay this can lead to significant errors in the identification of the decay time depending on the level of the noise floor. To overcome this, the following approach shall be used:

The general form of an exponential decay is given as:

$$x = x_0 e^{-\delta t} \cos(\omega t + \varphi) \tag{7.4}$$

with $\cos(\omega t + \varphi)$ being the oscillation and $x_0 e^{-\delta t}$ beeing the amplitude envelope of this decay. x_0 is the initial amplitude. Based on knowledge from the field of psychoacoustics, the human sense of hearing acts as an integrator and is less sensitive to relatively quiet sounds following a relatively loud stimulus. It is therefore necessary that the integral of the modelled decay

$$\int_1^{n_{noise}} x_0 e^{-\delta t}, dt \tag{7.5}$$

matches the integral of the raw decay which is taken fom the measured sample. As the sample is available in time-discrete form only, with $e(i)$ beeing the variable in which it is stored, the resulting equation is formulated as

$$\sum_{i=1}^{i_{noise}} e(i) = \sum_{i=1}^{i_{noise}} x_0 \cdot e^{i \cdot d}, i \in \mathbb{N} \tag{7.6}$$

Finally, the algorithm returns the t_{60} time.

7.2 Correlation

As justification by the overall amplitude is insufficient as explained above, the correlation coefficient of one or more signal parameters to a given reference are proposed as signal features to judge the subjective acoustic quality of a stimulus. The idea is that stimuli which are similar to a *good* reference stimulus are likely to be perceived good as well while differing from the reference decreases the chance of being well accepted. For the extraction of theoretically good sounds from the database a reference is needed. This can be acquired by one of the following three methods:

- Selection of an existing sound from the database
- Recording and subsequent analysis of a physical sample of a reference control element
- Design and subsequent analysis of a target sound using the synthesizer which is explained above

Since the third subjective trial showed almost no impact of the attack time and since the damping coefficient and the decay time describe the same phenomenon, the correlation of the damping coefficients and signal peaks are sufficient to compare the signals to a reference.

7.3 Experimental Validation of the Predictor

This chapter describes the validation of the method for the prediction of the acoustic feedback of a rotary control element as described above. The basic idea is that each subject will design an individual reference sound which is fed into the analysis system. The analysis algorithm shall then select sounds from the database which are likely to be perceived well by the subject as well as sounds which are supposed to be rejected.

7.3.1 Design of Experiment

The experiment consists of two parts, separated by a short break in which the calculation is performed. In the first part the subjects are asked to adjust the acoustic feedback of the simulator in a way they like. After the individually *good* sound is set up by a subject, the sound is stored and used as a reference which is compared with all the sounds in the database. The program selects a total of 29 sounds from the database, which are selected due to the following criteria:

- Increasing Correlation Coefficient of the Damping Coefficients (10 stimuli)
- Increasing Correlation Coefficient of the Peak Amplitudes (10 stimuli)
- Match of the Overall Amplitude

- 10 dB too quiet (3 stimuli)
- match (3 stimuli)
- 10 dB too loud (3 stimuli)

Since the sounds which are stored in the database are all relatively similar due to the fact that they are all based on physical control elements which are constructed in a similar way, their typical correlation coefficients were similar as well. *Figure 7.2* shows an example of how the typical distribution of correlation coefficients looks like:

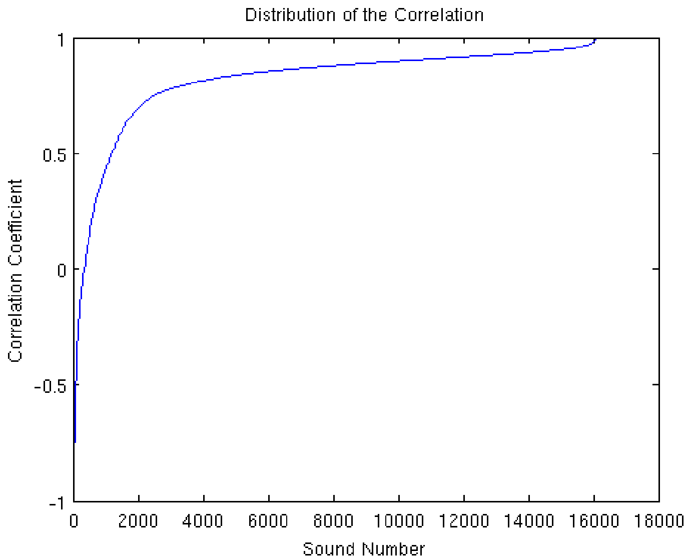

Figure 7.2: Note that over 75% of the sounds feature very high correlation coefficients

The figure shows the correlation coefficients for the peak amplitude parameter for the entire database. It can be seen that the curve is relatively steep for the first 2000 sounds and then becomes relatively flat, more than 75% of the sounds show high to very high correlation with the reference value in this example. The same is true if the damping coefficients are correlated.
The analysis and subsequent correlation and data extraction can be done in less than a minute so that the subjects can continue with the second part of the experiment right anway, since the time used for calculation is used to brief the probands with their task for the second part.
In the second part of the experiment it is the subject's task to rate the sounds which have been selected by the system in the step before. The subjects do this using a seven-point scale. The order in which the stimuli were presented to the subjects was randomized for every subject. The 20 sounds which are selected because of their correlation coefficients are normalized in terms of peak amplitude in order to prevent any influence through this parameter.

7.3.2 Analysis of the Results

The results of both the individually optimized feedback signals and the validation of the prediction algorithm will be examined in this section. The first provides further insight on the kind of feedback signals the subjects desire. The latter shall prove that the proposed methods for the improvement of the prediction of acceptance of acoustic feedback are valid.

Part 1 - Individually optimized feedback signals

After the subjects have configured and confirmed their individually desired feedback signal, these signals are subsequently analyzed and compared. The attack time of the signals was fixed, what they could adjust was the slope of the spectrum (i.e. emphasis of high frequencies, uniform distribution or emphasis of low frequencies), the overall peak amplitude and the damping coefficients.

The subjects typically preferred uniform spectra or spectra with over-emphasized high frequencies. Finally, *figure 7.3* shows the individual settings for the peak amplitude:

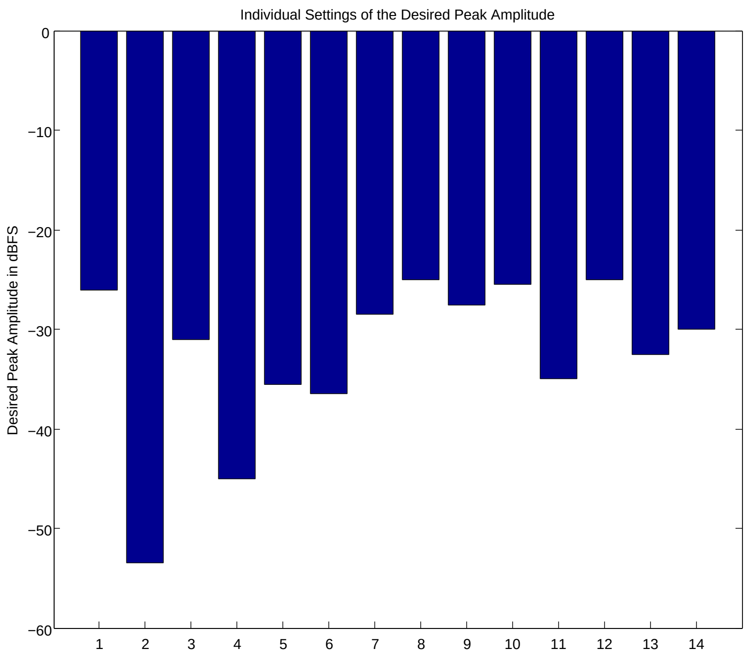

Figure 7.3: Individual desired peak amplitudes in dbFS, note that the majority of answers is within a 10 dB range.

This result supports the results from the previous experiment 4, since the majority of the subjects adjusted the desired amplitude to values within a 10 dB range, while only 2 of 14 subjects desired significantly lower values.

Part 2 - Rating of automatically selected sounds

The acceptance has been predicted based on three different methods:

- Correlation of the Damping Coefficients
- Correlation of the Peak Amplitudes
- Match of the Overall Peak Amplitude

The proposals for a prediction method which are based on correlation of signal parameters are supposed to have an effect on the acceptance of the stimuli, i.e. high correlation coefficients tend to higher acceptance ratings than low correlation values. A comparison of both methods of selection is shown in *figure 7.4* :

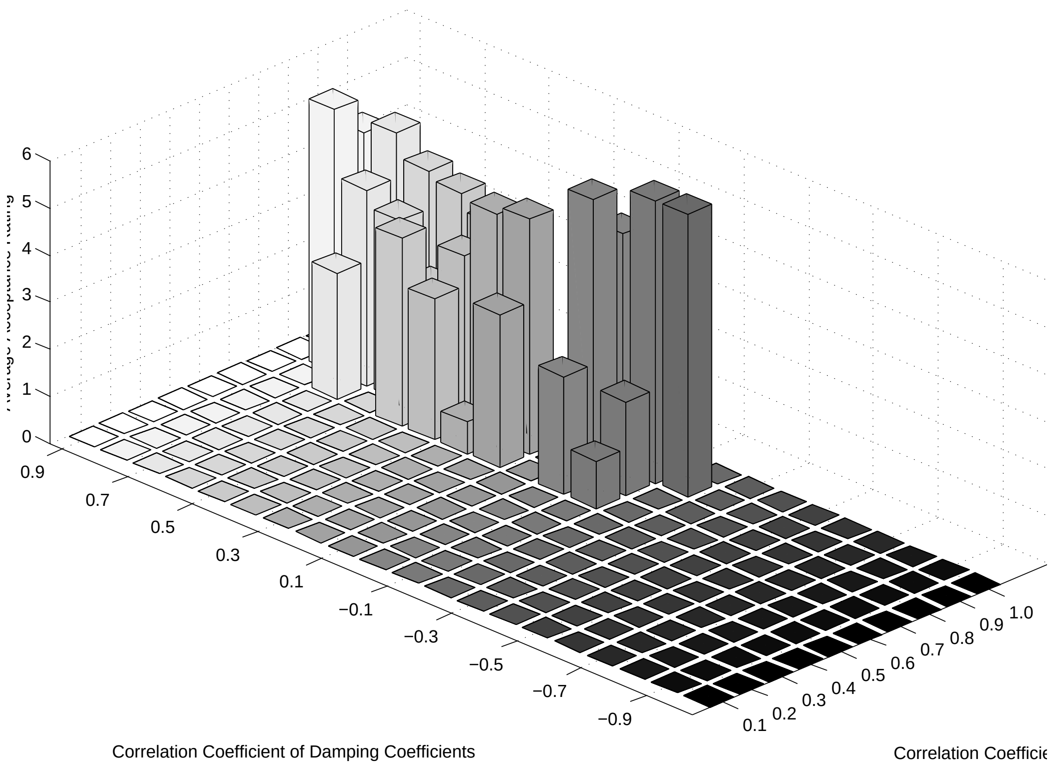

Figure 7.4: This plot shows the average acceptance values achieved with stimuli which showed the according combinations of the correlation coefficients of peak amplitudes per band and damping coefficients per band.

Since the variations in the actual correlation coefficients which have been calculated for each sound are very small the values have been quantized into 20 ranges and averaged within each range. The result is shown in the above figure. x- and y-axes denote the correlation coefficients of the damping coefficients and the peak amplitudes respectively. The z-axis denotes the average acceptance within a certain correlation coefficient bin. One would expect that the highest z-values occur in the upper ends of both x- and y-axis.

This is true for the correlation coefficient of the damping coefficients, in which relatively small deviations from the highest possible values lead to a significant reduction in the acceptance values. Regarding the correlation coefficients of the peak values it can be seen that even relatively low values lead to high acceptance ratings as long as the damping coefficients correlate well with the reference.
A possible explanation for this is that the human sense of hearing works as an integrator and as long as the damping coefficients correlate well, the integral of emitted sound per band should correlate relatively well, too.
As far as the influence of the absolute peak value on the acceptance is concerned, the results from the very first subjective experiment remain valid, as can be seen in *figure 7.5*

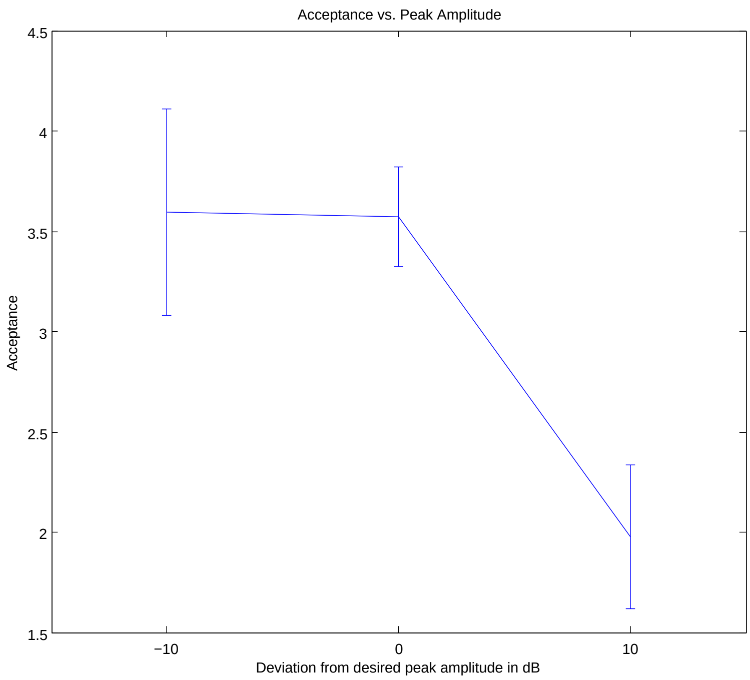

Figure 7.5: This figure shows that exceeding the desired amplitude leads to a significant decrease of the average acceptance of the sounds

Just like in the first subjective trial, the stimuli which exceed the desired amplitude receive significantly worse accaptance values. While the acceptance with the individually desired peak amplitude lead to a very similar acceptance value than the stimuli which were 10 dB too quiet, the standard deviation of the results is smaller with the stimuli which match the desired peak amplitude.

7.3.3 Conclusion

The experiment has shown that at least one of the two proposed methods for the improvement of the prediction of the acoustic feedback in fact leads to higher acceptance values in the subjectively rated sounds.

Chapter 8

Results

This section sums up what has been acheived within this project and formulates the improvements to the topics which habe been pointed out in *chapter 2*.

8.1 Method for Acoustic Rapid Prototyping

At first, a new methodology (see *figure 8.1*) for the definition of target sounds has been developed and successfully tested.

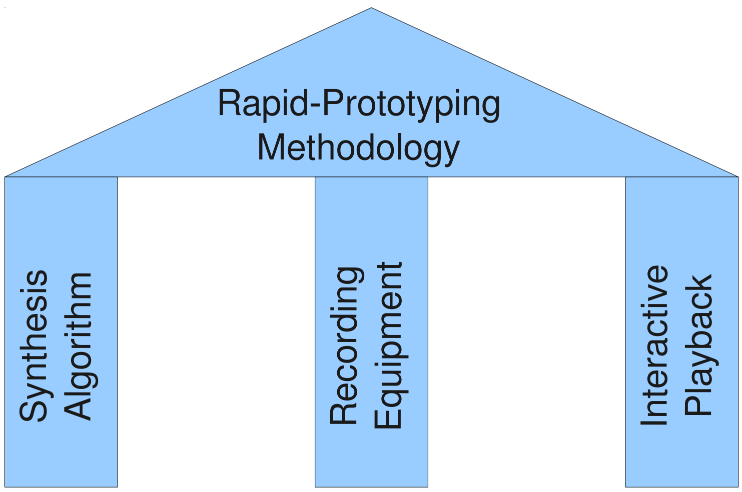

Figure 8.1: Visualization of the acoustic rapid prototyping methodology which uses synthesis and interactive playback as its foundations.

Basically the method consists of an adaptive approach to the target sound. This has been made possible by reduction of the parameters which are needed to synthesize a realistically sounding acoustic feedback signal. The final significant enhancement to traditional sound design approaches is the acoustic simulator. This device enables the designer of the sound to experience the sound in it's natural context. Optionally, the method can take measurements of physically existing control elements into account and use them as the basis for future sound design steps.

8.2 New Algorithm for Prediction of Acoustic Acceptance

Based on the results which were gathered in extensive subjective trials, an new algorithm for the prediction of the acceptance of the acoustic feedback of a control element has been formulated. The new idea of the proposed method is to acknowledge that the acoustic feedback cannot be predicted on an absolute scale but it is context sensitive and subjective. This leads to the conclusion, that the reference which is used as a standard for the prediction has to be context sensitive and subjective as well.
However, this insight in combination with the method for rapid prototyping which has been introduced in this work provides a possibility to objectively predict the acceptance of a sound within a given context. The extraction of the relevant acoustic features - which are the same as for synthesizing a convincing acoustic feedback stimulus - and the subsequent correlation with a reference has proven successful. The reference can be obtained from applying the introduced prototyping method to either a small number of experts or a larger number of naïve listeners and subsequent averaging of the obtained individual results.

8.3 New Proof for the Effects of Acoustic Feedback

Finally, the conducted subjective trials showed that the acoustic feedback influences the performance of subjects. This is both true from a speed and from an accuracy point of view. While the increases in speed are relatively small compared to the overall duration of an operation they are certainly not to be neglected in the context of a car or in the operation of critical machinery. The improvement in accuracy however is significant and certainly shows that acoustic feedback can in fact improve usability, especially in a context in which there are no haptic cues.

Chapter 9

Conclusion

As outlined in the introduction of this work, the perceived quality of today's products is determined to a large extend by the user experience. In the automotive market, this paradigm has been stated by Porsche [26] once as follows:

> "...the driver is always the center of attention. His experience is the measure of all things."

In today's complex producty high-quality user interfaces rapidly gained attention as a method of transporting the impression of general high quality. As the quoted works of Brewster [27], Altinsoy [28] and Reisinger [1] show, the feeling of the feedback of control elements gained importance in academic research and is still gaining importance in a number of applications.
The introduced new methods of designing and benchmarking provide tools which enable engineers and designers to design more precise specifications for subjectively more pleasant sounds in the future.
The success of touch-screen operated mobile phones or navigation devices in recent years as well as the advent of tablet PCs and new developments like camera-observed gesture control show that many appliances in the future are likely to be operated not by pyhsical control elements anymore but by virtual input devices in some way or another. Adding acoustic feedback to such a virtual input device always requires the playback of synthetic sounds. Results of this work are already used in this field.
Furthermore, the playback of synthetic acoustic feedback enables the designer of the appliance to adjust the feedback signal depending on the current conditions. In a car environment this means that a signal might be very quiet when the car is parked, but significantly louder at high speed and hence noise.
Finally, the usefulness of acoustic feedback has currently only be proven in a laboratory environment with an abstract task. Future work might deal with the question, how much the awareness of the driver is increased in for example a driving simulator. However, tools and methods developed in this work can be transferred for this purpose. Especially the intermodal effects which occour when several ways of feedback are used simultanously are as of now very little understood and provide further opportunities.
In general, one can state that the increasing complexity of today's technical products, the increased demand for comfort as well as the demographic change will set the future challenges in user interface design, and the concise and well planned use of acoustic cues can provide improvements for both usability and user experience in this field.

Bibliography

[1] J. Reisinger, H. Bubb, J. Wild, and G. Mauter, "Haptical Feeling of Rotary Switches", in *Proc. EuroHaptics*, 2006, vol. 6, pp. 49–55.

[2] B. Adelstein, M. Anderson, D.R. Begault, and B. McClain, "Thresholds for Auditory-Tactile Asynchrony", 2005.

[3] K. Hotting and B. Roder, "Hearing cheats touch, but less in congenitally blind than in sighted individuals", *Psychological Science*, vol. 15, no. 1, pp. 60–64, 2004.

[4] M. Kurosu and K. Kashimura, "Apparent usability vs. inherent usability: experimental analysis on the determinants of the apparent usability", in *Conference on Human Factors in Computing Systems.* ACM New York, NY, USA, 1995, pp. 292–293.

[5] N. Tractinsky, "Aesthetics and apparent usability: empirically assessing cultural and methodological issues", in *Proceedings of the SIGCHI conference on Human factors in computing systems.* ACM New York, NY, USA, 1997, pp. 115–122.

[6] D.A. Norman, *Emotional design: Why we love (or hate) everyday things*, Basic Civitas Books, 2004.

[7] P. Glanzmann, "Autopsychologie", Sat 1 Automagazin 31.03.2007, TV Show.

[8] D. Ali, "Identification of transient events from a hard disk drive using non-stationary loudness", *Acoustical Society of America Journal*, vol. 123, pp. 3160, 2008.

[9] J. Blauert and R. Guski, "Critique of Pure Psychoacoustics", in *NAG/DAGA 2009*, 2009.

[10] H. Fastl and E. Zwicker, *Psychoacoustics: facts and models*, Springer-Verlag New York Inc, 2006.

[11] A. Pohl, "Bau eines akustisch optimierten betätigungsroboters für bedienelemente", Diploma Thesis, Heilbronn University, 2005.

[12] P. Reiter, "Entwicklung einer ansteuerungssoftware für einen akustikprüfstand", Term Work, Heilbronn University, 2009.

[13] M. Frear, "pa-wavplay", `http://www.mathworks.com/matlabcentral/fileexchange/4017`, 2004.

[14] J. Huber, "Akustischer simulator", Term Work, Heilbronn University, 2007.

[15] A. S. Treiber and G. Gruhler, "Measurement and optimization of acoustic feedback of control elements in cars", in *Proceedings of the* 122^{nd} *AES Convention*, 2007.

[16] E. Altinsoy, "Perceptual aspects of audio-tactile asynchrony", in *10th International Congress on Sound and Vibration*, 2003.

[17] A.S. Treiber and G. Gruhler, "Simulation of Room Acoustics Design of a Listening Environment for Sound Quality Jury Tests", in *Proceedings of Radioelektronika 2006*, 2006.

[18] W.W. Gaver, "Synthesizing auditory icons", in *Proceedings of the INTERACT'93 and CHI'93 conference on Human factors in computing systems.* ACM New York, NY, USA, 1993, pp. 228–235.

[19] Daniel J. Freed, "Auditory correlates of perceived mallet hardness for a set of recorded percussive sound events", *The Journal of the Acoustical Society of America*, vol. 87, no. 1, pp. 311–322, 1990.

[20] S MacAdams, "Audition: Cognitive psychology of music", *In The Mind-Brain Continuum*, 1996.

[21] J. Blauert, *Communication acoustics*, Springer, 2005.

[22] *DIN EN 61260 - Bandfilter für Oktaven und Bruchteile von Oktaven*, DIN Deutsches Institut für Normung e.V., 2001.

[23] C. Roads and J. Strawn, *The computer music tutorial*, MIT press, 2000.

[24] D.A. Norman, *The design of everyday things*, Basic Books New York, 2002.

[25] H. Levitt, "Transformed up-down methods in psychoacoustics", *Journal of the Acoustical Society of America*, vol. 49, no. 2, pp. 467–477, 1971.

[26] Dr. Ing. h.c. F. Porsche AG, "Porsche cayenne catalog 2007", 2007.

[27] E. Hoggan, T. Kaaresoja, P. Laitinen, and S. Brewster, "Crossmodal congruence: the look, feel and sound of touchscreen widgets", in *Proceedings of the 10th international conference on Multimodal interfaces.* ACM New York, NY, USA, 2008, pp. 157–164.

[28] M.E. Altinsoy and S. Merchel, "Audiotactile Feedback Design for Touch Screens", in *Proceedings of the 4th International Conference on Haptic and Audio Interaction Design.* Springer, 2009, p. 144.