TEKNILLINEN KORKEAKOULU
Sähkö- ja tietoliikennetekniikan osasto

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ACOUSTIC TIMBRE ENHANCEMENT OF GUITAR PICKUP SIGNALS WITH DIGITAL FILTERS

Diplomityö, joka on jätetty opinnäytteenä tarkastettavaksi diplomi-insinöörin tutkintoa varten Espoossa 3.12.2002.

Työn valvoja Professori Matti Karjalainen

TEKNILLINEN KORKEAKOULU **DIPLOMITYÖN TIIVISTELMÄ**

Tekijä:	Henri Penttinen
Työn nimi:	Kitaramikrofonisignaalien akustisen äänenvärin parantaminen digitaalisilla suotimilla
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Tämän diplomityön tavoitteena on ollut tutkia ja kehittää tapoja, joilla voidaan digitaalisesti parantaa kitaramikrofonisignaalien äänenlaatua. Työn lähtökohtana on ollut luonnollisen kuuloinen tallamikrofonisignaalin toistaminen.

Tyypillisesti esiintymistilanteissa kohdataan ongelma ettei käytetty mikrofoni aisti akustisen kitaran onton kaikukopan vaikutusta. Seurauksena mikrofonin ulostulo kuulostaa epäluonnolliselta verrattuna ilmateitse säteilevään ääneen. Eräs ratkaisu tähän ongelmaan on käyttää tilanteeseen sopivalla tavalla suunniteltua digitaalista suodinta. Sopiva suodin pystyy syntetisoimaan onton kaikukopan vaikutukset tallamikrofonisignaaliin.

Aluksi tämä diplomityö käsittelee akustisen ja sähkökitaran toimintaa ja mikrofoneja, joita niissä tyypillisesti käytetään. Tämän perusteella esitellään systeemin signaalitiet. Tämän jälkeen kuvataan työssä käytetyt kitarankopan mallinnusmenetelmät, tekniikat ja mittausjärjestelyt. Sen jälkeen käsitellään suotimien toteutusmenetelmiä ja mallinnusaiheeseen liittyviä muita kysymyksiä. Lopuksi esitellään työn tulokset sekä pohditaan aiheeseen liittyviä kysymyksiä ja mahdollisia jatkotutkimusmahdollisuuksia.

Työssä käsitellyt menetelmät pystyvät mallintamaan onton kitarakopan ominaisuuksia. Menetelmiä voidaan käyttää tallamikrofonilla varustetulle akustiselle kitaralle. Lisäksi sähkökitara saadaan digitaalisuotimella kuulostamaan lähes akustiselta kitaralta. Kaikukopan havaittua kokoa voidaan myös muuttaa, niin että pienikoppainen kitara saadaan kuulostamaan suurikoppaiselta ja päinvastoin. Tapa, jolla tämä on toteutettu, mahdollistaa myös kaikukoppamallin käytön aikamuuttuvana ääniefektinä.

Avainsanat: akustinen kitara, sähkökitara, digitaalinen signaalinkäsittely, ääniefektit, soittimien mallinnus

HELSINKI UNIVERSITY OF TECHNOLOGY **ABSTRACT OF THE MASTER'S THESIS**

Author:	Henri Penttinen
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The objective of this thesis was to study and develop ways to digitally enhance the timbre of guitar pickup signals. The basis for this work was natural sounding reinforcement of an undersaddle pickup signal.

A problem typically encountered in a live performance situation is that the microphones often used in acoustic guitars do not capture the effect of the hollow body of the acoustic guitar. As a consequence, the audible timbre of the microphone output sounds unnatural, compared to the air-radiated sound. A solution to this problem is to use a properly designed digital filter. A proper digital filter is able to synthesize the effect of a hollow guitar body to the undersaddle pickup signal.

First, this thesis discusses the behaviour of the acoustic and electric guitar, and microphones typically used in them. Based on this, the signal paths of the system are described. Thereafter, the modeling methods, techniques, and measurement setups are described as used in this work to synthesize the hollow body of a guitar. Different ways to realize the modeling digital filters are discussed, with some further issues related to the subject. Finally, the work describes the results obtained, and discusses possible future work plans.

The methods discussed in this work are able to model the characteristic of a hollow guitar body. The modeling schemes can be used for acoustic guitars with an undersaddle pickup. In addition, the electric guitar can be filtered to sound more like an acoustic guitar. The perceived size of the modeled body can also be changed, i.e., a small-bodied guitar can be made to sound like a large one, and vice versa. The way this is implemented enables the use of the modeling filter as a time-varying sound effect.

Keywords:	acoustic guitar, electric guitar, digital signal processing, sound effects, instrument modeling

Preface

The presented work has been carried out in Helsinki University of Technology, at the Laboratory of Acoustics and Audio Signal Processing. This study is based on the collaborative work done between Matti Karjalainen and Vesa Välimäki from the Laboratory of Acoustics and Audio Signal Processing at Helsinki University of Technology and Heikki Räisänen and Harri Saastamoinen from EMF Acoustics. This work continued this collaboration. Part of this work was financed by the Academy of Finland.

During this work I have received a lot of support from numerous people in and out of the working environment. Here I try to thank and show my gratitude to those who I now can remember. First of all I would like to thank my supervisor Professor Matti Karjalainen for his great support and tremendous patience. The involvement of EMF Acoustics made this work possible, a big and warm thank you goes to Heikki Räisänen and Harri Saastamoinen, plus to the rest of the staff. I would like to thank Dr. Aki Härmä for helping me to warp my mind more efficiently and Ms. Riitta Väänänen for giving me insight on the reverberation business. I would also like to express my appreciation to the so far unmentioned co-authors in the publications related to this work Prof. Vesa Välimäki, Ms. Hanna Järveläinen, and Mr. Tuomas Paatero. Mr. Perttu Hämäläinen deserves a warm thank for the creative discussions on the subject, and Mr. Cumhur Erkut and Mr. Juha Merimaa for helping me out during the measurements and their support. I am grateful to Mr. Tero Tolonen for the supportive discussions we had. I also send my appreciation to Ms. Lea Söderman, the secretary of the Acoustics Lab, for helping me out with different practical arrangements.

All the crazy personnel at the Acoustics Laboratory have created a unique and great atmosphere to work in. I send them a humongous thanks. I also say 'Hi!' to all my friends.

A special expression of gratitude goes to my mother, Rea, and my sister, Irina.

Contents

List of Abbreviations

DECON	Deconvolution method
DFT	Discrete Fourier Transform
DSP	Digital Signal Processing
EMF	Electro Mechanical Film
FDN	Feedback Delay Network
FFT	Fast Fourier Transform
FEM	Finite Element Method
FIR	Finite Impulse Response
HRTF	Head Related Transfer Function
IIR	Infinite Impulse Response
IQR	Inter-Quartile Range
LMS	Least Mean Square
LP	Linear Prediction
LTI	Linear Time Invariant
MFDN	Multiple Feedback Delay Network
MIR	Modified Impulse Response (method)
SNR	Signal to Noise Ratio
STFT	Short Time Fourier Transform
WFIR	Warped FIR Filter
WGF	Waveguide Filter
WIIR	Warped IIR Filter

List of Symbols

a	Acceleration
a_n	De-nominator coefficients
$\alpha_k, \tilde{\alpha}_k$	Feedback and modified feedback coefficients of warped filter
$A(z)$	Transfer function of de-nominator coefficients
b_n	Nominator coefficients
$B(z)$	Transfer function of nominator coefficients
β_k	Feed-forward coefficients of warped filter
C_n	Amplitude of n^{th} harmonic
d	Diameter
d_m	Distance from the guitar to microphone
$D_1(z)$	Warped delay element
E	Youngs' modulus or elastic modulus of string
F	Force
f, f_1, f_n	Frequency, fundamental frequency, and frequency of harmonic n
f_s	Sampling frequency
G, G_0, G_B	Gain variables for parametric resonator
h	Displacement of string at plucking position
$h_\mathrm{eq}(n)$	Impulse response of equalization filter
$\tilde{h}(i)$	Warped time domain response
$H(\omega)$	Transfer function as a function of angular frequency
$H(z)$	Transfer function in the z-domain
$H_\mathrm{eq,ac/el}(z)$	Transfer function of equalization filter for acoustic/electric guitar
$H_c(z, \lambda_m)$	Transfer function of pre-filter
i	Discrete time variable of a warped time domain response
k	Index variable for de-nominator coefficients
K	Stiffness radius
l	Index variable for nominator coefficients
L	Length of string
L_d	Distance from guitar bridge
$\lambda, \lambda_o, \lambda_m$	Warping parameter, original warping parameter, and modified warping parameter
m	Mass
M	Order of de-nominator coefficients
μ	Mass per unit length
n	Discrete time variable or index variable of polynomial coefficients
N	Order of filter or order of nominator coefficients
ω	Angular/radian frequency
$p(t)$	Air radiation signal
$P(\omega)$	Air radiation of acoustic guitar
Q	Bandwidth
r	Pole radius
$R(z)$	Transfer function of termination reflection filter
ρ	Density
$s_\mathrm{dry}(n)$	Discrete signal from guitar pickup
$s_\mathrm{eq}(n)$	Discrete guitar pickup signal filtered with body model

S	Cross-sectional area
$\sigma_k, \tilde{\sigma}_k$	Feedback and modified feedback coefficients of delay-free warped filter
σ_z	Stiffness
t	Continuous time variable
T	Tension
v	Velocity
$V_{\mathrm{b}}(\omega)$	Vertical bridge excitation
$w(n)$	Windowing function
$x_{\mathrm{ac/el}}(t)$	Pickup signal from acoustic/electric guitar
$\tilde{x}(z)$	Discrete time domain response of modulated instrument body model
$X(\omega)$	Pickup output
z^{-n}	Delay of n samples
$\tilde{z}^{-1}$	Warped delay element
Z	Impedance

1 Introduction

This thesis studies and proposes solutions to the problem of enhancing guitar pickup signals with digital filters. Enhancing in this context means making a guitar pickup signal sound more like a natural acoustic guitar. The goal of this work is to achieve a response from a guitar pickup signal that is similar to that when standing in front of an acoustic guitar player. This work discusses methods for obtaining a more acoustic sound both for an acoustic and an electric guitar pickup signal.

The problem of amplifying an acoustic guitar, especially in a live performance situation, is to obtain a natural acoustic-like response. Often the audible response is very dry and almost resembles an electric guitar. Therefore, the perceived guitar playing looks more acoustic than it sounds like, i.e., the received visual information contradicts slightly with the perceived auditory information. The reason for this contradiction lies in the used microphone settings. More precisely, the microphones often used in acoustic guitars are not able to sense the presence of the guitar body. Hence, the sound is dry and contradicts with the visual signal.

To achieve a natural timbre from an acoustic guitar, external microphones are often preferred. With this setup the player is confined to the vicinity of the microphone. Usually the distance from the player to microphone, placed in front of the player, is a few decimeters at the most. Microphones placed to the sound hole are also used. External microphones sense the playing of the instrument, but they also sense unwanted noises such as movements of the player and other loud instruments, e.g., drums. In addition, the system is susceptible to acoustic feedback. As a result of this, the whole amplification system may produce very loud and disturbing ringing tones. To allow the player to move freely, and to reduce acoustical feedback and unwanted noises, internal microphones are frequently used. In this case the microphone is attached inside the body of the instrument, or beneath the bridge. Microphones used solely for amplifying string vibrations of a guitar are often called pickups. By using internal microphones, many of the problems encountered with external microphone settings are reduced, but with a negative effect on the timbre of the audible signal. The audible signal lacks naturality and liveliness of the acoustic guitar and sounds easily more like an electric guitar than an acoustic one.

The reason for the difference in the timbre between a real acoustic guitar and an acoustic guitar with an internal microphone is that the internal microphone cannot fully sense the body of the instrument. The acoustic body of the instrument adds coloration and reverberation to the audible response. Also the high frequency components of the radiation, straight from the strings, are missing. Since the response of an acoustic body is reverberant it is understandable that the pickup or microphone signal that misses this property is often described as dry, as previously mentioned.

Solutions to these kinds of problems exist. In the vanguard publication in this field (Mathews and Kohut, 1973) analog means were used to produce resonances that enhance the response of a violin. The resonances modeled the instrument body modes, i.e., resonances of the body. Instrument body modeling in general and in the context of instrument modeling and synthesis has also been studied (Smith, 1983; Karjalainen et al., 1991; Karjalainen et al., 1993a; Välimäki et al., 1996; Bradley et al., 1995; Karjalainen, 1996; Karjalainen and Smith, 1996; Tolonen, 1998; Välimäki and Tolonen, 1998; Cook and Trueman, 1998; Wong et al., 1999b; Wong et al., 1999a). The foundation of modeling an instrument body lies on the linear and time-invariant (LTI) characteristic of the signal path from a vibrating string to air radiation. This property allows modeling of the instrument body with linear filters. Mathews and Kohut used analog filters in (Mathews and Kohut, 1973) to implement resonators but the development of digital signal processing algorithms and hardware enables the use of digital filters and DSP processors.

The study presented in this thesis is based on the work done at Helsinki University of Technology, Laboratory of Acoustics and Audio Signal Processing, concerning the equalization of undersaddle pickup signals for acoustic guitars (Karjalainen et al., 1999). The author has had the privilege to collaboratively continue on the work (Karjalainen et al., 1999; Karjalainen et al., 2000a; Karjalainen et al., 2000b; Penttinen et al., 2000a; Penttinen et al., 2000b; Penttinen et al., 2001a; Penttinen et al., 2001b) and taking the initial concept to be useful also for electric guitars and moving into the field of sound effects. The simulations have been done with the MATLAB 6.0.0.88 software. All the filters have also been implemented to run in real-time on a SHARC 65-L DSP processor[1].

The outline of this thesis is as follows. In Chapter 2 the guitar is viewed from its structural and functional properties point of view. The discussion covers both the acoustic guitar and the electric counterpart. Microphones typically used in guitars are also dealt with. Based on the talk in Chapter 2, Chapter 3 analyzes the signal paths of the system from a model-based point of view. Chapter 4 introduces the body modeling methods and techniques used in this study. Chapter 5 discusses the realization issues related to the modeling methods, covering different filter structures and modifications that have been done for them. Modifications meaning the separation of isolated body modes and changing of the spectral structure. Chapter 6 covers further aspects concerning instrument body modeling by dealing with some general issues and some more detailed issues. Chapter 7 presents the results of the implemented instrument body modeling schemes. Chapter 8 includes discussion and suggestions for future work. Chapter 9 completes the report with conclusions.

[1]Excluding the further issues discussed in Chapter 6 (except the three-band equalizations), and body modeling done with reverb algorithms.

2 Overview of Guitar

Guitars are widely used in popular, jazz, and also classical music. It has been a fashionable instrument, on and off, since the 1700th century. And during the twentieth century it has become one of the most popular of all musical instruments (Bacon, 1991)(pp.9). Both the acoustic and the electric guitar come in a legion of different designs. Modern acoustic guitars are descendants of an ancient string instrument brought to Spain by the Romans in circa 470 AD. A more recent predecessor of the acoustic guitar is the vinhuela, which is a sixteenth-century Spanish folk instrument. Moreover, Spain was the focal point of the instrument's development and especially luthier [2] Torres (1817-1892) evolved the construction techniques of the classical guitar. The electric guitar is an offspring of the acoustic guitar. It started evolving in the U.S.A. in the 1930s and 40s. Figure 1 depicts an acoustic guitar (a), called the classical guitar, and an electric guitar (b) designed by Leo Fender. For a more thorough discussion on different acoustic and electric guitar designs see (Bacon, 1991; Denyer, 1982) and for historical treatments (Summerfield, 1991; Wheeler, 1993) are recommended.

In this chapter, the behaviour and structure both of the acoustic and the electric guitar are discussed, it also includes microphones used with guitars. When reduced to its simplest, the mechanics of a guitar is just slightly more than the amplification of vibrating strings. Therefore, this chapter first discusses the behaviour and properties of strings, and then moves to the guitar itself. The acoustic guitar is addressed first, by examining the anatomy and the acoustical properties of it. Then the electric guitar is dealt with, leaving out the discussion of the constructive parts and behaviour that are much alike with the acoustic guitar. After this microphones and pickups often used in guitars are presented. As a close up and clarification of this section a comparison is performed between the acoustic and the electric guitar.

2.1 Vibration of Plucked String

By simplifying the behaviour of a guitar: it is the amplification of vibrating strings which are stretched across a semi-rigid structure. A string in a guitar is excited by plucking it with a finger, a nail, or a plectrum. By plucking the string it is first displaced from its rest position by a distance d. Then the string is released and it starts to vibrate. As a consequence, two waves start to propagate to opposite directions in the string. At the boundaries (the ends) the displacement waves reflect and retain their phase if

[2] Luthier: one who makes stringed musical instruments as violins or guitars.

Figure 1: (a) acoustic guitar and (b) electric guitar.

the end is free and invert their phase if the end is fixed. In a string the reflecting waves result in standing waves. The vibratory motion of the string can be expressed as a sum of modes that are obtained by solving the wave equation for a string (Morse and Ingard, 1968). The freely vibrating quasi-periodic sinusoidal signals can be approximated as being harmonically related, i.e., the frequencies of the modes are in integral ratios. In this case, the mode with the lowest frequency is called the fundamental, f_1, and higher modes are called harmonic partials or harmonics, f_n, where $n > 1$. In real strings the vibrations gradually attenuate and the string returns to it rest position. The attenuation is caused by external and internal losses and transmission of energy to the guitar body.

A vibrating string has three vibrational degrees of freedom or vibrational modes: transversal, longitudinal, and torsional. Figure 2 depicts the three vibrational modes in respect to the top plate of a guitar. The transversal vibration mode (Fig.2 a) is divided into horizontal and vertical components. Horizontal vibrations occur along the top plate and vertical ones perpendicular to the first one. Longitudinal vibrations move about the equilibrium in the direction of the string (Fig.2 b). The third vibrational mode, torsional wave motion, takes place by rotating or twisting around the string (Fig.2 c). The longitudinal and torsional wave motions are not considered that important in the guitar or in the perspective of this study. Although, Conklin has reported (Conklin, 1999) the existence of so-called phantom-partials due to longitudinal string forces that are coupled to the body via the bridge. However, the longitudinal string vibrations can have a more important role in other string instruments, such as the kantele (Karjalainen et al., 1993a; Erkut et al., 2002). Furthermore, the torsional wave motions in the violin have an important role in the bow-string interaction (Cremer, 1984). In the case of the guitar, the horizontal and vertical components are the most important vibrational modes of the strings. Moreover, the vertical direction has a particulary important role in this study as will be discussed in Sec. 3.3.

The length and tension of a string are the most important factors that affect the resulting fundamental frequency. The fundamental frequency and its harmonics for an

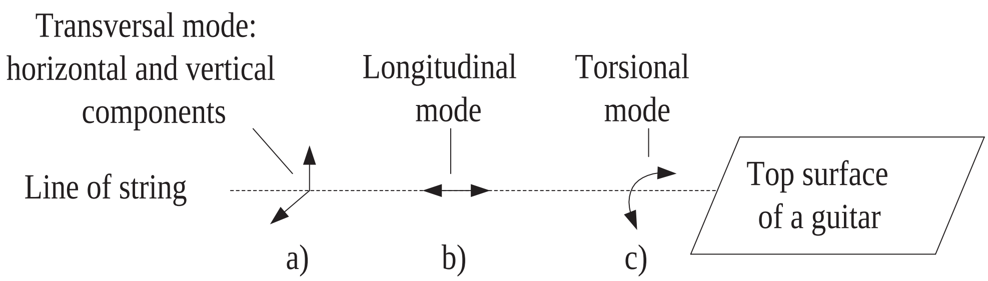

Figure 2: A string has three vibrational degrees of freedom: (a) transversal, (b) longitudinal, and (c) torsional. The transversal motion is divided into horizontal and vertical components.

ideal string can be calculated as follows

$$f_n = \frac{n}{2L}\sqrt{\frac{T}{\mu}}, \tag{1}$$

where where n is the number of the harmonic, L is the length of the string, T is tension, μ is mass per unit length.

2.1.1 Effect of Plucking Position

By changing the plucking position, a guitar player can control the produced tone. In the spectra of the tone harmonics that have a node at the plucking position are not excited and have a zero amplitude. Figure 3 depicts the four first standing waves created when the string is plucked one-fourth of the total string length from the bridge. The amplitude of a harmonic can be calculated in the following manner

$$C_n = \frac{2h}{(n\pi)^2}\frac{L^2}{L_{\mathrm{d}}(L-L_{\mathrm{d}})}\sin(\frac{n\pi L_{\mathrm{d}}}{L}), \tag{2}$$

where h is the displacement of the string at the plucking position, and L_{d} is the distance from the bridge. Figure 4 illustrates the effect of the plucking position in the frequency domain in two cases: when the string is plucked (a) in the middle of the string, and (b) at a point fourth of the string length from the bridge. However, in the excitation process the excitation point is not infinitely small, i.e., it has a finite width. Therefore, vibrational modes behaving like in the idealized case seldom exist.

2.1.2 Properties of Real Strings

Dispersion Waves corresponding to different modes of a string travel with different velocities, i.e., the velocity of each harmonic is frequency-dependent. As a result, the

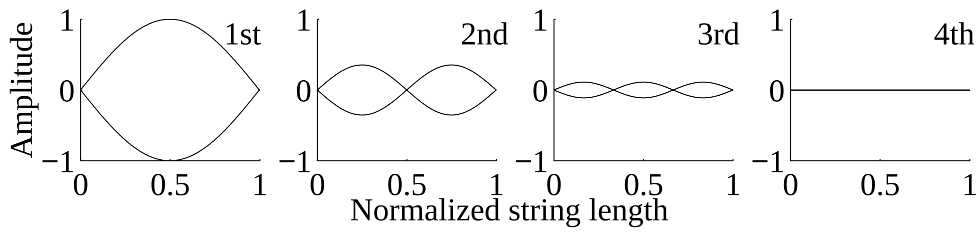

Figure 3: Standing waves created to a string when it is plucked one-fourth of the string length from the bridge.

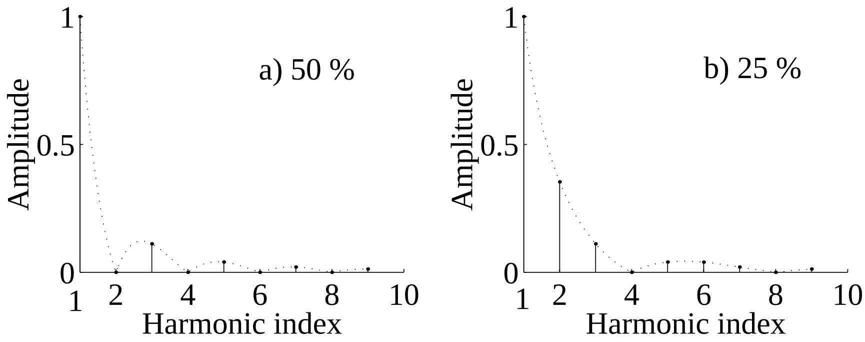

Figure 4: Effect of plucking position. Two spectra exemplifying the comb filter effect due to plucking position. String excited a) in the middle, and b) one fourth of string length from the bridge.

harmonics will not have exactly integral ratios, and the vibrational modes of the string will be slightly inharmonic. This property is called dispersion.

Dispersion is caused by the finite stiffness of a string and its non-uniform mass distribution. The frequency of each partial in a stiff string can be calculated as follows

$$f_n = nf_1\Big(1 + \beta + \beta^2 + \frac{n^2\pi^2}{8}\beta^2\Big), \tag{3}$$

where f_1 is fundamental frequency of a flexible string, and $\beta = (2K/L)\sqrt{ES/T}$, where E is the Youngs' modulus, S stands for the cross-sectional area, K is the stiffness radius, L is the length of the string, and T is the tension in the string. Figure 5 shows the frequency of partials plotted as a function of harmonic index, both for the ideal string (solid line) and stiff string (dotted line). Fig. 5 and Eq. 3 show that the frequency of a harmonic increases as the index of the harmonic increases when compared to the behaviour of an ideal string (Eq. 1).

Shifts in Mode Frequencies There are two main mechanisms that cause the mode frequencies to deviate from the frequencies obtained from the wave equation of a rigidly supported stiff string. (I) Tension modulation. A transversal displacement in the string makes a second-order change in its length, and thus in its tension (Legge and Fletcher,

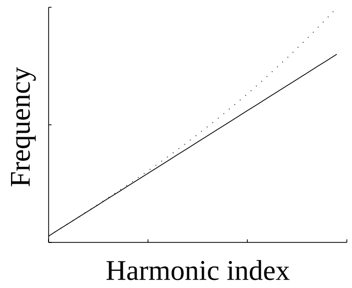

Figure 5: Inharmonicity in a stiff string increases with frequency. The relative frequency as a function of harmonic index is plotted for a stiff string (dashed line) and ideal string (solid line). The inharmonicity of a string depends on the physical properties of the string.

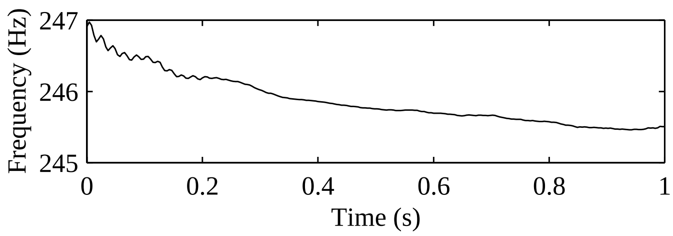

Figure 6: Shift in mode frequency of a classical guitar tone. The fundamental frequency is displayed as a function of time.

1984). Therefore, a stretched string is linear only in the first approximation. For this reason, the mode frequencies obtained from the wave equation are only valid when the string is at rest. (II) Coupling between modes via the bridge can cause beating of harmonics (Gough, 1981). Shifts in mode frequency can be exemplified by observing Figure 6. It displays the fundamental frequency of a plucked guitar tone as a function of time. A collective characteristic for guitar tones can be observed: The frequency of the fundamental and its harmonics decrease after the initial excitation.

Beating of Harmonics and Two-Stage Decay Amplitude modulation of partials and two-stage decay are caused by the frequency dependent bridge impedance. The finite bridge impedance causes a change to the effective length of the string. As mentioned previously in this chapter, transversal string vibrations can be divided into two polarization components, namely horizontal and vertical polarizations (see Fig. 2). As a consequence of the non-uniform bridge impedance in different directions, the two polarizations vibrate in a slightly different manner. In the case of a piano string these two polarizations have been studied by Weinreich (Weinreich, 1977). For the kantele, tanbur, and guitar the subject has been discussed in, e.g., (Välimäki et al., 1999), (Erkut et al., 1999), and (Tolonen, 2000), respectively.

Beating of harmonics means that the amplitude of a harmonic is modulated periodi-

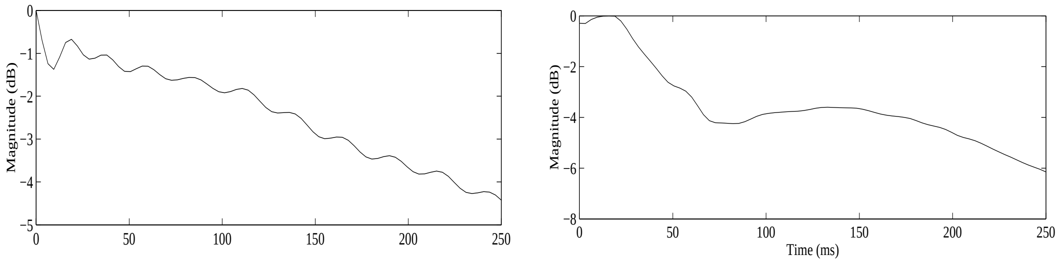

Figure 7: Beating of harmonics (left pane) and two-stage decay (right pane). The amplitude envelopes are for the second and fourth partials of note F_4 played on a classical guitar.

cally. This results from slightly different frequencies of the polarizations. This again is caused by the change of effective length altered by the bridge impedance. The left-hand side pane in Figure 7 shows the beating effect for the second partial of an F_4 tone played with a classical guitar.

Two-stage decay stands for the decay of amplitude in two stages with different decay rates. A two-stage decay is caused by deviating decay rates of different polarizations. A two-stage decay of a classical guitar partial (fourth for tone F_4) is shown in the right-hand side pane of Figure 7. The player is also able to control the overall amplitude envelope two-stage decay by changing the plucking direction, see (Fletcher and Rossing, 1991) for further discussion.

Generation of Weak Harmonics The plucking position affects the spectrum of a vibrating string as described in section 2.1.1. Partials of the vibrating string have different amplitudes and ideally decay exponentially after the initial excitation, with a beating or with a two-stage decay. However, the amplitude of a harmonic that is depressed, as an effect of the plucking position, can increase (or stay the same) in amplitude as time passes. This is caused by nonlinear coupling between vibrating modes (Legge and Fletcher, 1984). Figure 8 displays the fundamental (310 Hz) and two harmonics of a classical guitar tone when the b-string has been plucked in the middle. Theoretically the second harmonic should not be excited at all, but practically it is very difficult to achieve a fully symmetrical excitation situation that would prevent the harmonic not to be excited at all. In Figure 8 the second harmonic is clearly depressed, but the amplitude of the harmonic stays the same and increases slightly after the build-up. This exemplifies the generation of a weak harmonic in the case of a classical guitar. This effect can be more prominent if the coupling between different polarizations is stronger. This can be achieved by plucking the string very hard. For example in the the Turkish string instrument, the tanbur, the generation of a weak harmonic can be even more evident (Erkut et al., 1999).

Sympathetic Coupling Between Strings Sympathetic coupling is an important feature in the total sound of an acoustic guitar. This phenomenon is also present in the

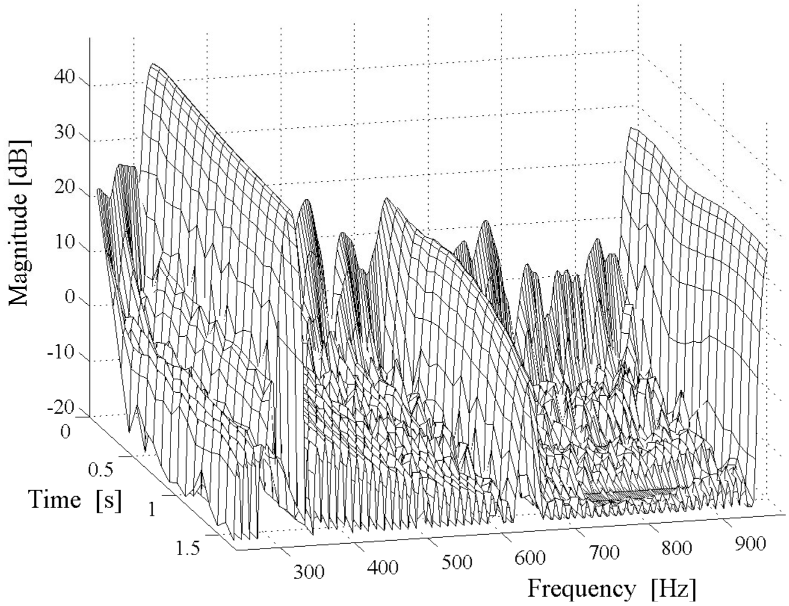

Figure 8: Time-frequency plot of fundamental (310 Hz) and two harmonics of a classical guitar tone when the b-string has been plucked in the middle. The second harmonic exemplifies the generation of a missing mode. The frequency components between the string partials are body resonances.

electric guitar, but does not have as important role as in the acoustic guitar. In practice sympathetic coupling can be noticed by plucking a string without damping the others and then damping the string that was plucked. After damping the plucked (initially excited) string, silent sounding of other string or strings can be perceived. Sympathetic coupling is brought about through mechanical coupling between the strings, and to a smaller extent via the air (Fletcher and Rossing, 1991).

Sympathetic coupling between two strings occurs at frequencies that are common resonance modes for both strings. The primarily vibrating string drives the secondary string at the fundamental or its harmonic frequencies. For example, the fourth harmonic of the lowest E-string (82 Hz) drives the fundamental and its harmonics of the highest e-string (329 Hz). Typically, the attack and decay of the amplitude envelope of the secondary string is slow. Sympathetic coupling adds a spacious feel to the playing that can be controlled by damping the unplucked strings.

2.2 Acoustic Guitar

There are basically two types of acoustic guitars: ones with nylon strings and ones with steel strings. Nylon strings are used in classical and flamenco guitars. Their names also indicate what kind of music is typically played with them. Steel strings are used in flat-top, arch-tops, and resonator guitars, to name a few. The flat-top is the most commonly used steel-stringed guitar and is often used in music styles like folk, blues,

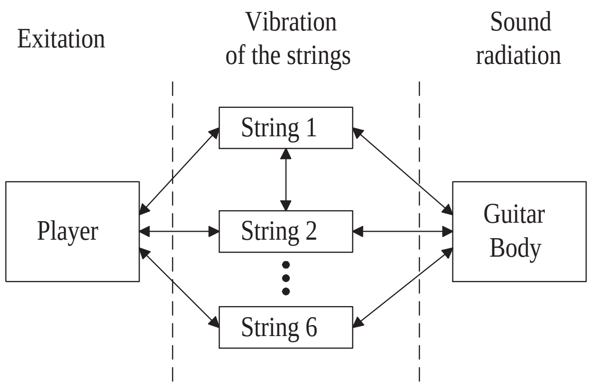

Figure 9: Block diagram of the sound production mechanism in the acoustic guitar, after (Karjalainen and Välimäki, 1993).

rock and even jazz. As a common denominator, all acoustic guitars have a hollow body that colors the response and amplifies the string vibrations.

The flat-top is an offspring of the classical guitar whose beginnings were taken in America during the early 1900^{th} century (Denyer, 1982). Since the classical guitar is the forefather of other acoustic guitars, the discussion concerning the acoustic guitar is revolved around it. Even if their timbres differ somewhat, this generalization is acceptable because their main principles of sound production and structure are the same.

In addition to the classical and flat-top guitars there are other designs that can be considered as acoustic guitars. Resonator guitars were developed during the 1920s and 30s to produce a louder response than regular acoustic guitars have. The resonator guitars, also known as dobros, have one or more resonating aluminum cones inside them that greatly increase their volume. Some of the models have even metal bodies (Denyer, 1982)(p.48). These features give the guitars a distinctly harsh sound. The so called arch-top guitars were also developed to give a louder response when compared to a regular acoustic guitar. In these designs the shape slightly departs from flat-tops, the top-plate has two f-holes rather than one round hole and the body plates are thicker in diameter (Denyer, 1982)(p.46).

To understand the behaviour of the acoustic guitar it is worthwhile to examine the functional components of the instrument. This way a better and deeper understanding can be reached for the need and influence of the equalization filtering introduced and discussed in this study. The sound production mechanism can be clarified by dividing it into three parts (Karjalainen and Välimäki, 1993) as illustrated in Figure 9. The parts consist of the string excitation, vibration of the string, and finally of the sound radiation. Next the discussion continues on the construction and acoustical properties of the acoustic guitar, while trying to give a better understanding of Fig. 9.

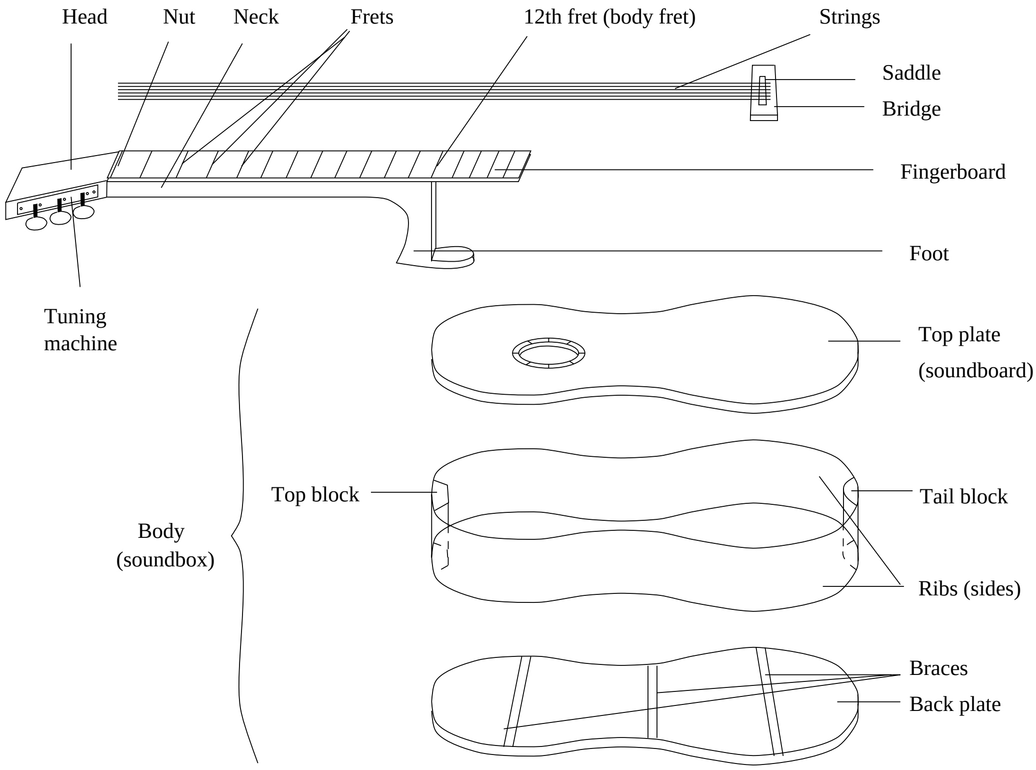

Figure 10: Structure of acoustic guitar.

2.2.1 Structure of Acoustic Guitar

The construction and design of the classical guitar has been almost the same for over 100 years (Denyer, 1982). Even if the design and use of materials may vary from a guitar-maker to another, the overall construction in the majority of acoustic (classical) guitars is the same. The structure of the acoustic guitar is shown in Figure 10, which displays an exploded view. The hollow guitar body, also called the soundbox, consists of a top plate, a back plate and two sides, also called the ribs. The bridge is situated on top of the soundboard and is the other element that holds the strings under tension. The guitar neck is situated at the other end of the soundbox. The fingerboard is glued above it and at the tuning machinery is located at the head of the neck. The strings, that are tuned with the tuning machinery, run just above the fingerboard. On the whole, all these parts work together and all affect and contribute, more or less, to the final timbre of the acoustic guitar. In the following sections the most important parts of the acoustic guitar are discussed more thoroughly. Figure 10 still depicts these parts, but is not referred to separately.

2.2.2 Guitar Body

The hollow guitar body has a crucial role in the acoustic guitar. The purpose of the guitar body is to amplify the vibrations of the strings. This amplification is needed since the string emits weakly its vibrations (Morse and Ingard, 1968). After the strings, the body is the most prominent part of the guitar that shapes the timbre of the instrument. The hollow and resonant guitar body functions also as an energy sink. It draws energy from the vibrations of the strings, especially at frequencies where string vibrations match with resonances of the guitar body. The choice of material used in the guitar body and its shape has a profound effect on the timbre of the guitar. The construction of the guitar body is shortly discussed here and its effect to the sound production of the acoustic guitar will be consider in more detail in Section 2.2.5.

The guitar body consists of the top and the back plate and the sides (also called ribs). They are connected together with linings. The two sides are also connected with the top and tail blocks that also reinforce the body. (Denyer, 1982)

The top plate, also called the soundboard, has a significant effect on the tone and quality of the instrument. The soundboard is made out of two pieces of wood and is braced (strutted) to strengthen its construction. The wood used for the soundboard is typically spruce or pine. The bottom plate is made out of hard wood, such as Brazilian or Indian rosewood, African walnut, or mahogany, to give the instrument better strength. It is also braced, but with a simpler pattern and design. (See (Fletcher and Rossing, 1991) (Fig. 9.2) for traditional bracing patterns.) Ideally the braces and the strutting pattern should only support and strengthen the soundboard allowing a uniform vibration pattern. However, the strutting pattern can have a dramatic effect on the sound of the guitar (Denyer, 1982). As the bottom plate, the sides are also made out of hard wood.

The soundboard of a guitar should be very light but stiff. Hence, the soundboard is usually slightly thinner than the sides and the back plate. The sides and the back plate being thicker, they somewhat prevent unwanted vibrations of the body that would decrease the sustain of the instrument. Traditionally, the material used for soundboard strutting is wood, but there are also guitar builders who use more modern synthetic materials like carbon fiber epoxy (Rossing et al., 2002; Mead, 1994). This way a high stiffness-to-mass ratio can be achieved and, hence, high-resonance frequencies. Also, this makes the instrument to have a longer sustain compared to a more traditional construction (Mead, 1994).

2.2.3 Neck and Fingerboard

The neck of the acoustic guitar comes in many different designs. The neck is made of hardwood such as mahogany, maple, or rosewood. The use of hardwood is necessary to keep it in shape, since the bending force introduced by the strings is relatively powerful. In nylon-stringed guitars the force is not as powerful as in steel-stung guitars. Therefore, in nylon string guitars, the neck can be constructed out of a single piece of wood or optionally out of three pieces. In a three-piece neck the head and the foot are made separately. In some designs a separate hardwood strip is set to a groove in the neck, underneath the fingerboard. In steel-stringed guitars the tension caused by the strings is greater than in nylon-stringed guitars. As a consequence, many designs have a steel truss rod to give additional support. In many cases, the tension created by the truss rod can be adjusted, so that the neck can be straightened if it becomes skewed (Denyer, 1982). The shape of the back of the neck also varies to some extent. In some of the more modern designs the neck is thinner than in older designs. The neck is also thinner in width at the nut than at the soundboard (Denyer, 1982).

The fingerboard, also called the fretboard, is glued above the neck. Part of the neck is typically on the soundboard, starting from frets above the twelfth one. Frets are made of metal and are mounted to the fingerboard. Hence, the additional name fretboard. The material of the fingerboard is typically ebony or some other hard and dark wood (Denyer, 1982). The fingerboard in classical and flamenco guitars is usually flat, whereas some steel-stringed guitars have a slightly curved fingerboard (Denyer, 1982). Frets are situated across the fingerboard at evenly changing distances from each other. This is to keep the guitar in tune while played anywhere on the neck. The nut is located at the end of the fingerboard at the head of the guitar. The nut has small grooves that guide the strings and sets the distance between them. Even if the strings run just above the fingerboard when a string is pressed against a fret, the angle between the saddle and the string changes. As a consequence the relative tension is increased when compared to an open string. This causes a slight mistuning which is compensated by making the open string length, i.e., the distance from the nut to the saddle, slightly larger. The tuning machinery is located at the head of the guitar. It is used to tune each string separately by changing the tension in the strings (Denyer, 1982).

2.2.4 Strings and Bridge

The modern guitar has typically six strings. Table 1 lists the frequencies, note names, and order numbers in a typical western tuning. Alternative tunings also exist and are often used in the style of slide playing (Denyer, 1982)(pp.158-159) In the case of a twelve-string instrument each of the additional six strings are placed close to the main string, hence forming string-pairs. The additional strings are tuned one octave higher than the main string, except the highest b- and e'-strings which are tuned in unison. As mentioned before, the core string materials are either nylon or steel. The three strings tuned to the highest are typically plain. The rest are wound, i.e., they have an extra wire wound around the core. Three different types of wires can be used for the wounding: round, flat, and half-flat. These names indicate the cross-sectional shape of the wire. In addition, the thickness of a set of strings (6-strings) vary. In the case of a thick set of strings the third or g-string is also wound. Moreover, both the type of wiring and the thickness of the used string-set effect the final timbre of the guitar (Denyer, 1982). Hence, different types of string-sets are used in different playing styles. Also the type of strings used affects the produced sound level. The sound level produced by flat-top guitars is about 5 to 10 dB higher than produced by classical guitars. This is because steel strings are under twice as much of tension as nylon strings and their mass is twice as large (Rossing, 1990).

Table 1: Typical tuning of six string guitars. The numbers refer to the often used order numbering, the note names and corresponding frequencies are also shown.

String no.	**Note**	f **[Hz]**
1	e'	329.63
2	b	246.94
3	g	196
4	d	146.83
5	A	110
6	E	82.41

The bridge has an important role in acoustic guitars since it transmits nearly all the string vibrations to the guitar body. In nylon string guitars the bridge is glued directly to the top plate. In this case, the bridge is often made of ebony, which has a high density and hence aids transference of the vibration energy and prolongs the sustain of the instrument. In some steel-stringed guitars, e.g., arch-tops, the bridge can be floating. This means that its place on the top plate can be changed. (Denyer, 1982) (p.40)

The saddle is a small item mounted in a groove carved in the bridge. The strings run over the saddle. In contrast with the nut, which is located at the head of the guitar, the saddle does not have grooves where the strings would run through. Although, some saddles have small notches or semi-circular grooves, to keep the strings in place during hard playing. Together, the saddle and the nut define the termination of the vibrating section of the strings. Some steel-stringed guitars have adjustable bridges, where the height of the bridge can be controlled. This way the distance of the strings to the fingerboard can be adjusted. The saddles in hand made guitars are generally made of ivory or bone and in other types of guitars synthetic materials are used more often. Even if the saddle is a small item mounted to the bridge, it has a definitive effect on the tone, volume, and tuning of the guitar (Denyer, 1982)(p.41). As a side-remark: The Elastic Electret Film (EMF) pickups used in this work are mounted under the saddle. The EMF pickups will be discussed in Sec. 2.4.2.

2.2.5 Sound Radiation from Guitar Body

The vibrational energy of the strings is mainly radiated to the surrounding air through the vibrations of the guitar body. The strings themselves have a poor impedance matching with the surrounding air (Fletcher and Rossing, 1991). Therefore, they do not radiate their vibrations sufficiently, although the straight air radiation of the string vibrations is also audible. To obtain a useful sound level of the acoustic guitar the bridge transfers the vibrations of the strings to the guitar body. Therefore, the body and the air enclosed within it vibrate and amplify the signal. The guitar body has a good amplifying property since it has a good impedance match with the surrounding

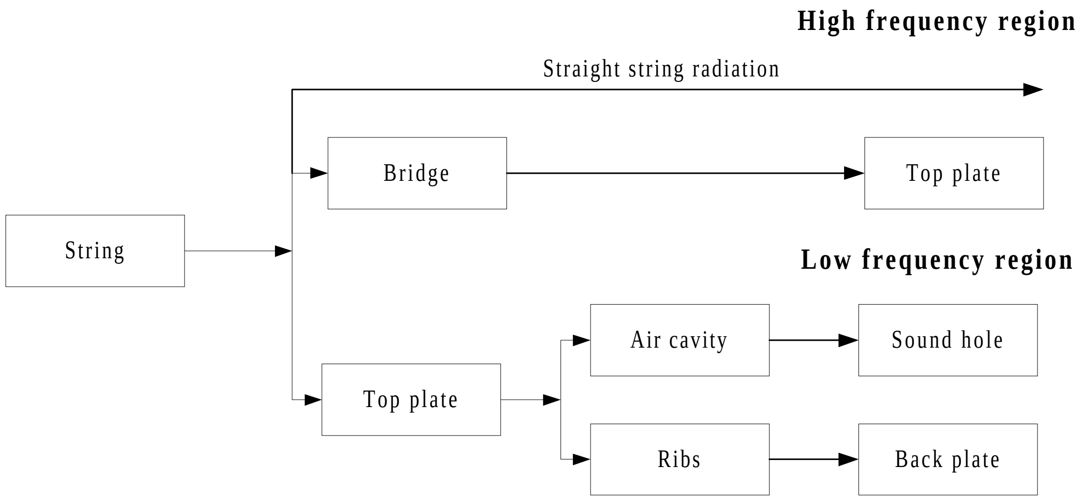

Figure 11: Schematic view of sound radiation of the acoustic guitar. The vibrational properties are divided into two frequency regions. After Fig. 9.3 in (Fletcher and Rossing, 1991).

air (Fletcher and Rossing, 1991). In addition to the amplifying property, the vibrating body and the air inside it color the response of the string vibrations. Coloring of a response means that some frequencies are amplified more than others. This also affects, what is called, the sound quality or tone of the guitar. Furthermore, the body also determines the directional properties of the guitar, since the air cavity and the plates do not radiate in a uniform manner to all directions. At low frequencies, i.e., frequencies up to circa 300 Hz, the sound of an acoustic guitar is radiated quite evenly to all directions (omni-directional). At frequencies above about 300 Hz the sound is more directional so that the directivity pattern depends strongly on the frequency. For polar radiation patterns see (Rossing, 1990)(Fig. 10.32).

The radiation process of the guitar body can be analyzed via examining a schematic presentation illustrated in Figure 11 (Fletcher and Rossing, 1991) where the vibrational properties are divided into low and high frequency regions. At the lowest mode, the soundhole and the back plate are mainly responsible for the created sound radiation. At high frequencies, the soundboard radiates most of the energy and the bridge has a significant part in this process. Also in the high-frequency region, the straight string radiation has a significance to the produced sound, especially in the near field. In the next section, a response of an acoustic guitar is examined in the time and frequency domains.

2.2.6 Response of Acoustic Guitar Body

The response of an acoustic guitar consists of a collection of resonances, also called body modes. The response is slightly reverberant. At low frequencies, the body modes can be distinguished from each other. At high frequencies, the mode density increases significantly and as a consequence the body modes cannot be distinguished as easily from each other. The lowest body modes are the strongest in magnitude. Moreover, their frequencies are size dependent, so that the larger the body the lower the frequency. The lowest resonance occurs typically at a frequency somewhere between 80 and 110 Hz and the second one between circa 170 and 250 Hz. They correspond to the first mode of the air cavity or the so called Helmholtz mode of the guitar body, and to the first mode of the top plate, respectively (Christensen and Vistisen, 1980).

A response of an acoustic guitar body can be obtained by performing measurements in an anechoic chamber. The anechoic environment diminishes the effect of possible reflections in the room. The bridge of an acoustic guitar is tapped with an impulse hammer and the radiation is recorded with a measurement microphone placed in front of the guitar at the level of the sound hole. The microphone should not be too close to the sound source to avoid the proximity effect. An often used distance from the sound hole of the guitar is one meter. The nylon-stringed classical guitar body response discussed below was recorded with the impulse hammer technique with a distance of one meter between the sound hole and the microphone [3].

Next the response of an acoustic guitar body will be examined in more detail. This way the need for guitar body simulation filters and their different responses can be understood more easily. Figure 12 shows an impulse response of a nylon-stringed acoustic guitar body. The attack part of the response is impulsive, hence the spectral content should be rich. The long decay part consist mainly of a few low-frequency resonances where a more clean waveshape of a sinusoid can be distinguished. Figure 13 displays the corresponding magnitude response, where the top pane displays the whole response and the bottom pane zooms into low frequencies. The strong and prominent low-frequency body modes can clearly be distinguished as they are circa 20 dB stronger than the overall level of the response at high-frequencies.

Figure 14 depicts a time-frequency representation of the same body response, where x-axis corresponds to frequency, y-axis to time, and z-axis to the magnitude (in each time window). This representation gives a good view of how fast the high-frequency resonances decay compared to the low-frequency ones. The low-frequency resonances take 100 to 300 ms to decay, at frequencies up to circa 1.5 kHz. The resonances above 1.5 kHz decay clearly faster and have decay times from 40 to 100 ms. Also, in the vicinity of 1.5 kHz a clear change in the decay rate can be noticed. In addition, the high-frequency resonances fall rapidly under the noise-floor. Therefore, at the end of the response (200 to 300 ms) the slowly decaying low-frequency resonances are audible and at high frequencies only the noise-floor can be perceived.

[3]The body response was recorded by Matti Karjalainen and Vesa Välimäki.

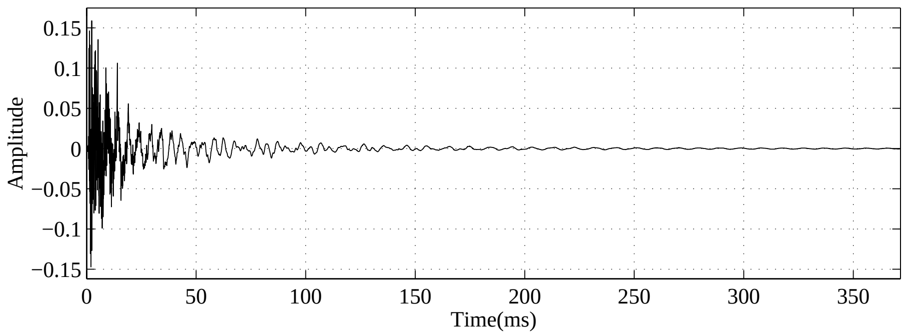

Figure 12: Impulse response of an acoustic guitar body.

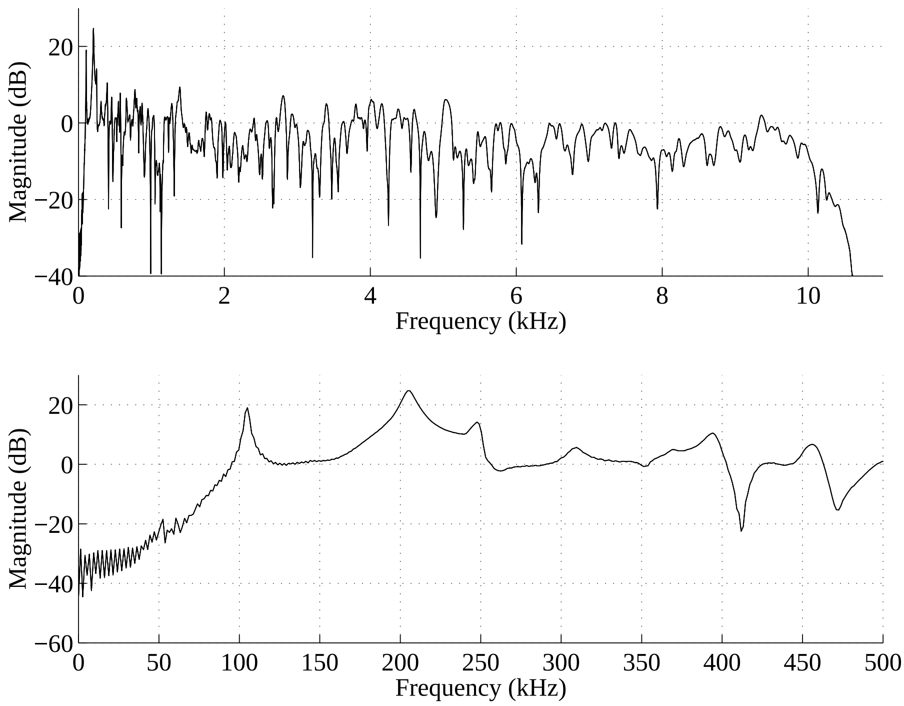

Figure 13: Top pane: magnitude spectrum of an acoustic guitar body. Bottom pane: zoomed to low frequencies. The sampling rate of 22050 Hz results in a frequency range up to 11.025 kHz.

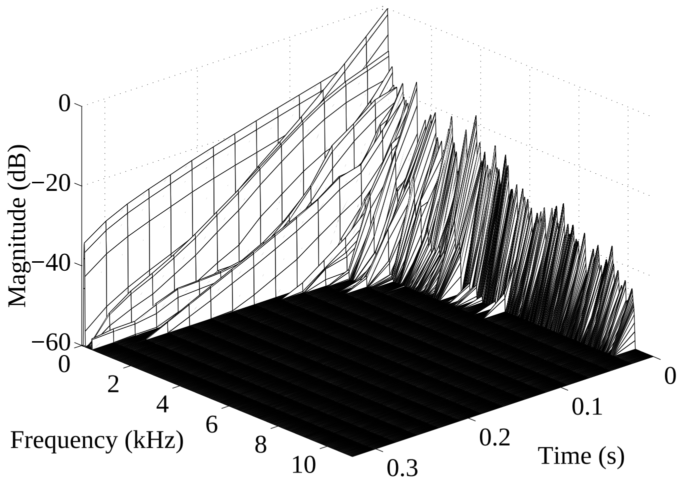

Figure 14: Time-frequency plot of acoustic guitar body response. Sampling rate is 22050 Hz, hop size is 1024, and overlapping is 50 percent.

2.3 Electric Guitar

The electric guitar differs inevitably from the acoustic one through its tone and structure. The electric guitar, in most cases, has a solid wooden body and is equipped with pickups to convert the string vibrations to an electric signal that is amplified and fed to a loudspeaker (Denyer, 1982)(p.49). These factors, among others, result in a different audible timbre. The first electric guitars with magnetic pickups and solid bodies were developed by Adolph Rickenbacker in the 1930's. More specifically, they were electric Hawaiian guitars (Denyer, 1982)(p.55). All in all, the solid-body electric guitar evolved from early amplified acoustic guitars through the need of higher volume levels.

Figure 15 shows two solid body electric guitars. Figure 15 (a) depicts a Stratocaster model manufactured by Fender. It is equipped with three single-coil pickups. The bridge is situated roughly in the same position as in its acoustic predecessor and it has a tremolo arm attached to it. The finger plate, also called the pickup guard, is positioned beneath the strings and on top of the body. The pickups and control units (gain and tone controls, and pickup selector) are mounted to the finger plate and the body. The body at these places is carved so that the required electronics fit beneath the finger plate. The neck is attached to the body and in this model the tuning machinery is located at the end of the neck (head). Figure 15 (b) shows a Les Paul model manufactured by Gibson, which has two humbucker pickups, and a static bridge. This Les Paul model does not have a pickup guard as the Stratocaster has, therefore the control units are mounted straight to the solid body. The output socket in electric guitars is usually a standard 1/4-inch jack plug socket. The pickup selector enables to switch between available pickups and their combinations. The tone controls are simple equalizers attached to the guitar and effect the timbre of the instrument.

Electric guitars are produced in a considerable number of imaginative designs. How-

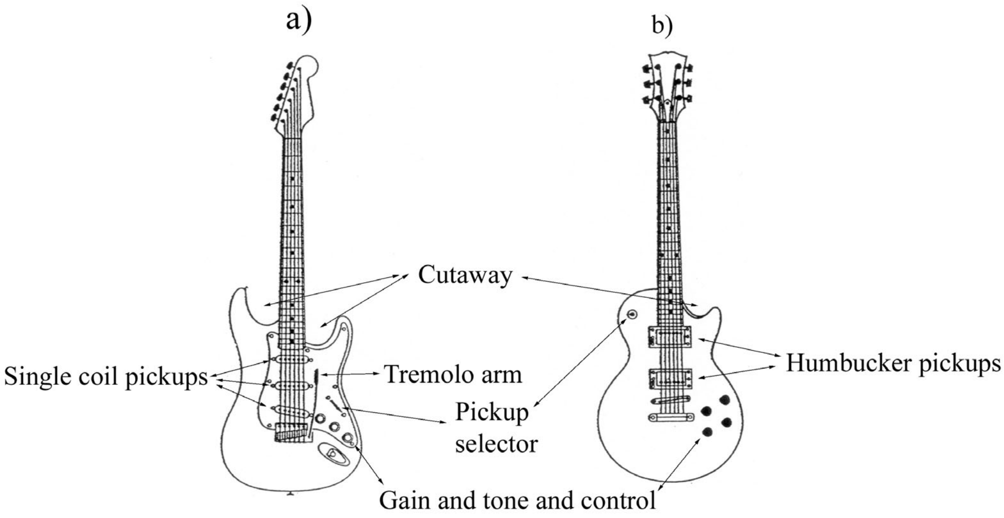

Figure 15: Two solid-body electric guitars: (a) Fender Stratocaster with three single-coil pickups and a tremolo arm, and (b) Gibson Les Paul with two humbucker pickups and a static bridge.

ever, the Fender Stratocaster and the Gibson Les Paul have established a solid foundation as electric guitar standards among musicians. Both were introduced in the 1950's. All electric guitars are not completely solid bodied: the hollow body and semi hollow electric guitars are electric ones even if they are not completely solid bodied. The numerous and imaginative designs of electric guitars are not discussed here, (Bacon, 1991; Denyer, 1982; Harmony Central, Inc., 2002) are referred to for pictures and discussion.

This subsection discusses the anatomy and the sound production of electric guitars. Pickups used in electric (and acoustic) guitars are briefly mentioned here and are discussed more thoroughly in Sec. 2.4. Again in the following discussion in electric guitar construction Figures 15 and 10 are referred to as a quick reference without separately mentioning them.

2.3.1 Electric Guitar Body, Fingerboard, Neck, Strings, and Bridge

The electric guitar resembles structurally the acoustic one in many ways. The differences are mainly due to the solid body, which in itself enables some new possibilities not possible in an acoustic guitar.

Solid Body The purpose of the body in electric guitars is to hold the pickups, mounted to the body, as stable as possible. This is because at high volume levels acoustic feedback problems arise when the pickups are able to vibrate. This problem is very common with hollow-bodied instruments. By having a high-mass body, the ability to receive and transmit vibrations, that cause acoustic feedback, is radically reduced compared to light and hollow bodies. A high-mass body can be obtained by using high-density wood, such as mahogany, walnut, ash, alder, and maple. As a consequence of a solid body, the design parameters for an electric guitar body are very relaxed, in contrast to the strict design of an acoustic guitar body, where most of the tone and amplification relies on the hollow body. The electric guitar has also evolved to a solo instrument played frequently above the 12th fret. Therefore, a single or double cutaway to the body is often preferred in designs. This allows easy access to high frets on the fingerboard. In Figure 15 the Fender exemplifies a double cutaway and the Gibson a single cutaway design.

Neck The necks in electric guitars are much like the ones in acoustic guitars. However, the relaxed design constrains give a chance to attach the neck in various ways to the solid body without constructing a poor quality electric guitar. There are mainly three neck designs: glued-in, bolt-on, and straight-through necks (Denyer, 1982). The glued-in neck is permanently glued to a slot in the body. In the bolt-on neck the neck is detachable, and is bolted or screwed to the body. In the straight-trough neck the body is comprised of two pieces that are glued to neck that runs in the middle. This way the strings are attached to one piece of wood. In addition to the varied neck attachment designs, a number of adjustable truss rods have evolved. As in steel-stringed acoustic guitars, many electric guitars have an adjustment screw of the truss rod at the head stock. Alternatively, the adjustment screw can also be located at the body. Some designs have double truss rods that allow the correction of a twisted neck. There is an ongoing debate about which neck design and truss rod option gives the longest sustain. Sustain is the length of the time a note continues to sound after a strings has been plucked. The neck is not the only part that effects the length of the sustain: the nut, the bridge, pickup output, the used materials also have an effect on the sustain. However, the arguments are not conclusive. On the other hand, some research has been made on the influence and vibrations of the electric guitar neck (Fleischer, 1998). Fleischer has found out that the neck vibrations result in so called dead spots that are frequency-dependent (Fleischer, 1998). As a consequence, some notes die out faster than others, i.e., all notes along the neck are not equal when the sustain is considered.

Fingerboard In most guitars the fingerboard is a separate strip of wood glued on the neck. It is often made out of rosewood or ebony. As in acoustic guitars the frets are glued to the fingerboard. In some designs, the frets are set directly to the neck (Denyer, 1982). In these necks, the material used is often rosewood. Some fingerboards are scalloped, which means that concave dips are between two frets by carving (Denyer, 1982). The gauges of frets in electric guitars vary more than in acoustic guitars. With gauges thicker than usually used the purpose is to gain more sustain and make playing easier.

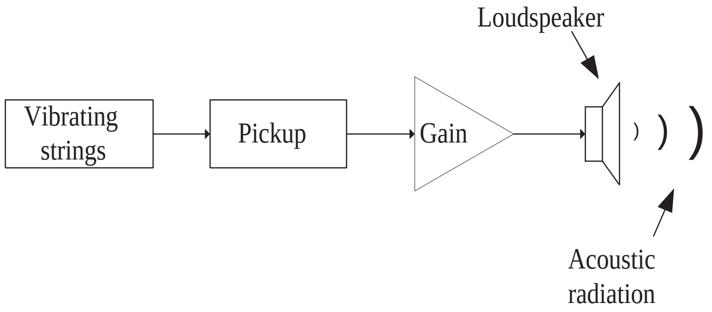

Figure 16: Schematic view of the sound production of an electric guitar.

Bridge and Strings Numerous different bridges can be used in electric guitars. One design solution is to have a static bridge, where the distance from the body to the whole bridge is adjustable. Some guitars have a tremolo arm integrated to the bridge (Denyer, 1982)(p.53). With a tremolo arm the tension of the strings can be modified. Hence, a tremolo arm enables to create vibrato by mechanical means. Vibrato is a rapid and regular variation in the pitch of a note. There are also many tremolo arm designs. It is often a problem to keep the strings in tune if the strings drag to a direction or another during playing. Improvements to this problem have been made by locking the strings at the nut with openable metal pieces and moving the actual tuning machinery to the bridge. Typically, in electric guitars each string sits on its own adjustable saddle. The adjustable saddle enables fine tuning of each string separately. However, some older designs of bridges with a tremolo arm do not have separate saddles. Strings in electric guitars are of the same construction as described for acoustic guitars, but with a few exceptions. Firstly, nylon strings are not used in electric guitars, since they do not induce an electric signal magnetic pickups used in electric guitars (the induction process will be discussed later in Sec. 2.4.3). Secondly, the wounding materials may differ from the ones used for acoustic guitars, so that the induction properties of the strings are of high quality.

2.3.2 Sound Production of Electric Guitar

Compared to acoustic guitars, electric guitars have a totally different approach to making its string vibrations audible. The electric guitar uses pickups to convert the vibrations of the strings to an electric signal. This signal is fed to an amplifier and then to a loudspeaker which converts the electrical signal to an audible acoustic signal. Figure 16 shows a simple diagram of the sound production procedure involved with an electric guitar. The pickups and their placements have a definite effect on the final timbre of the instrument. The most typical pickup in electric guitars is a magnetic pickup. Piezo-electric pickups are also used but are not as common as magnetic pickups. The effect of pickups on the timbre will be discussed here and the operation principles of pickups will be described in the next subsection 2.4.

2.3.3 Effect of Pickup Placement

The placement of a pickup effects the final timbre of the instrument. A pickup can be placed between the bridge and the fretboard. Electric guitars have typically one to three pickups. The position of the pickup changes the spectral content of the tone by creating a comb filtering effect, very similar in the way the plucking position affects the spectrum of the string. First we may look at the situation in the time domain by viewing the standing waves created in a string. Figure 17 displays the fundamental and the 5th harmonic in relation to the pickup positions. The amplitude of the standing wave is represented by the arrows at each pickup position, for both the fundamental and the 5th harmonic. The middle pickup is located under a node of the 5th harmonic. Hence, the pickup will not generate any voltage at the frequency of the 5th harmonic. In other words, the 5th harmonic will be missing in the spectrum.

Figure 18 displays the effect of pickup positioning in the frequency domain. It shows the occurring comb filtering effect for three pickup positions: (a) neck (b) middle and (c) bridge pickup position in the case of the lowest E string after Fender Stratocaster specifications [4]. The transversal line shows the frequency range of the fundamental of the lowest E string with a two-octave fingerboard. In this case, frequencies above the transversal line are present as harmonics. Figure 18 displays the effect of the pickup position in the frequency domain in respect to the displacement of the string. To obtain the behaviour in respect to velocity the displacement function is differentiated. In Fig. 18 the first peak and first notch in each case move to higher frequencies as the pickup is placed closer and closer to the bridge. The placement of the pickups and selection among them or their combination provides an easy way to control the timbre of the electric guitar.

In addition to detecting the string vibrations, a magnetic pickup initially functions as a low-pass filter (Jungmann, 1994). The low-pass filtering effect is partly caused by the impedance of a magnetic pickup. The other phenomena causing a low-pass effect is due to the width of a pole piece in a pickup, which is discussed in Chapter 2.4.3. The impedance, i.e., the resistive, capacitive, and inductive properties of the magnetic pickup create a passive low-pass filter. The cutoff-frequency of the low-pass filtering effect is pickup-dependent. Typically, the cutoff-frequency is between 2 and 5 kHz and the stop-band attenuation is about 10 to 20 dB. This low-pass effect results in a softer tone compared to an acoustic guitar.

The impedance of cables and amplifiers affects the transfer function characteristics of the whole system (Jungmann, 1994). The capacitance of a cable affects the resonance frequency of a magnetic pickup, so that as the capacitance increases the resonance frequency decreases in frequency. The input resistance of amplifiers that an electric guitar is connected to affects the transfer function of the system in the following manner: with low input resistance (< 300 kΩ, when capacitance is assumed to be 0) the output gets attenuated and low-pass filtered. With high resistance values the gain is

[4] Scale length is 25.5 in and the distances from the bridge for the neck, middle, and bridge pickups are 6.375 in, 3.875 in, and 1.625 in, respectively.

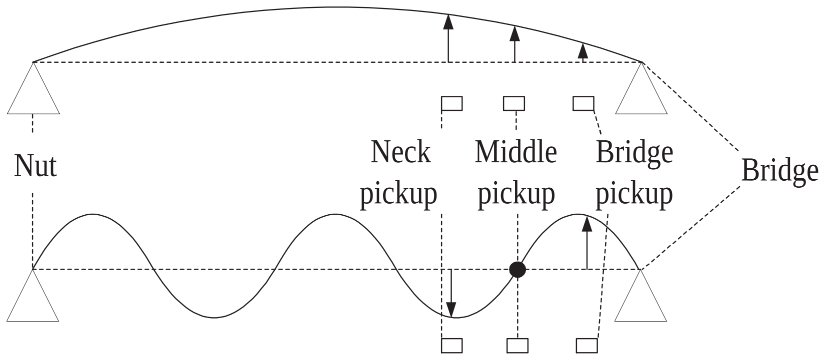

Figure 17: Effect of pickup positioning. Time-domain interpretation of the situation. The amplitudes of the fundamental and the 5th harmonic are displayed in relation with the positions of the pickups. In this case the 5th harmonic is missing when the middle pickup is used.

about 0 dB and a distinct resonance peak forms. A more detailed discussion on these effects can be found in (Jungmann, 1994)(pp. 65-67).

2.4 Microphones and Guitar Pickups

Microphones are used to capture and convert vibrations propagating in an intermediate agent (medium) to a corresponding electric signal. Typically, microphones are used to convert the changes in the air pressure that humans hear as sounds. In this case, the medium where the vibrations propagate is air. Sound vibrations can also propagate via many other mediums, and in this study, in addition to air, the most interesting ones would be the body, the saddle, or the bridge of a guitar. Microphones that are especially designed and used to convert vibrations produced by a guitar are often called guitar pickups. (Eargle, 2001) is a good reference on the electrical and functional properties of different microphones and Denyer (1982) discusses guitar pickups and magnetic pickups.

In this study microphones are discussed from the point of view of a guitar, but the discussion could still be understood on a general level. There are basically three ways to convert vibrations produced by a guitar to electric signals, by using (I) external (conventional) microphones, (II) internal (contact) microphones, or (III) magnetic pickups.[5] External microphones are often able to capture not only the vibrations of the string, but also the effect of the guitar body. In contrast, internal microphones and magnetic pickups capture mainly the string vibrations. Magnetic pickups are typically used for electric guitars, but they can also be used for steel strung acoustic guitars. Combining the signals from pickups on a guitar is an often used way to gain controllability over tone. In acoustic guitars, the signals from external and internal microphones are typically combined to gain adjustability. Also in electric guitars, signals from different

[5]External and internal microphones in this context mean that they are placed outside of the guitar (= external) or are in contact with the body (=internal).

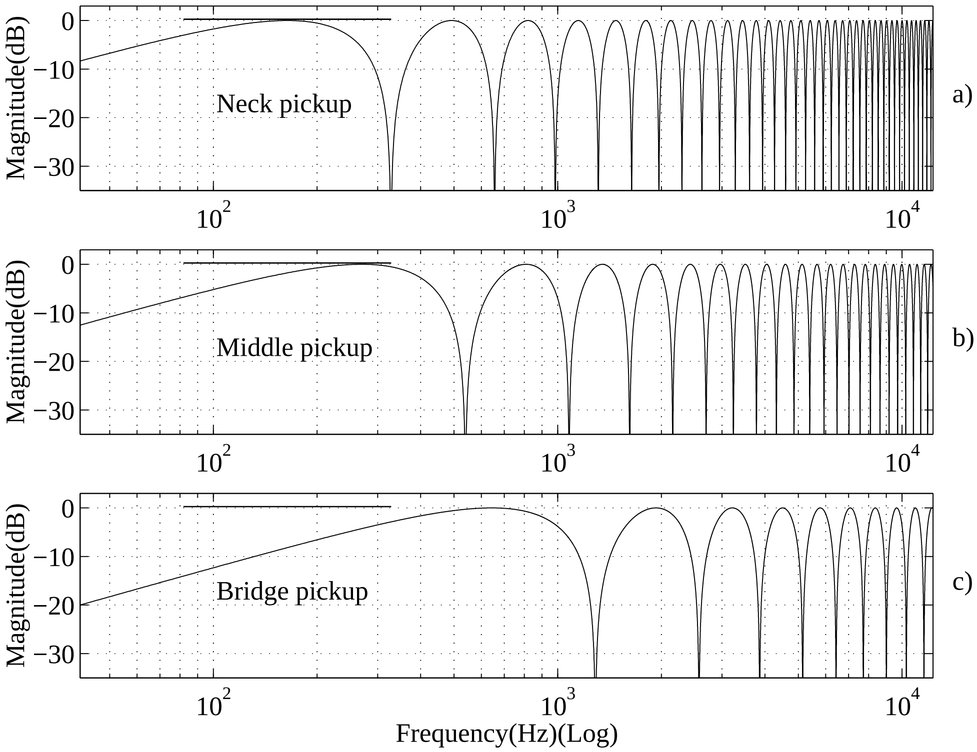

Figure 18: Effect of pickup positioning displayed in frequency domain in respect to string displacement. The effect is displayed for three cases: (a) neck (b) middle and (c) bridge pickup position. The transversal line shows the frequency range of the fundamental of the lowest E string with a two-octave fingerboard.

magnetic pickups can be summed.

This section describes the principles of microphones and pickups used for guitars. First external microphones are dealt via discussing crystal, dynamic, and condenser microphones. Then internal microphones are discussed through piezo microphones and electret microphones. Finally, magnetic pickups and their functioning principles are covered.

2.4.1 Crystal, Dynamic, and Condenser Microphone

Microphones react to changes in air pressure created by sound waves or to particle velocity as sound waves propagate. Microphones that react to air pressure are called pressure microphones and are in more common use than those who respond to particle velocity. A pressure microphone has a thin diaphragm that moves back and forth with the pressure changes in a sound wave. The diaphragm is connected to an electric generator, such as, a piezoelectric crystal (crystal microphone), a moving coil (dynamic microphone), and a variable capacitor (condenser microphone). The electric generator creates an electric signal that corresponds to the pressure variations created by a sound wave. The microphones discussed in this section can be used for acoustic guitars as

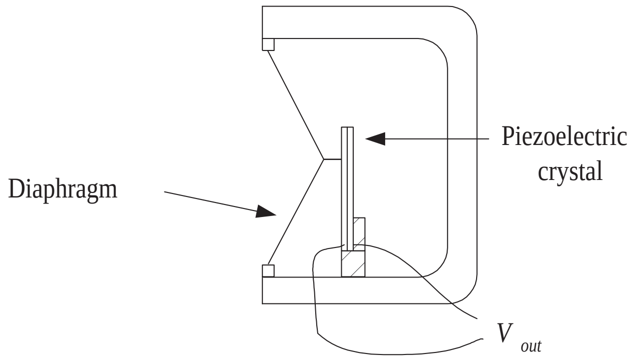

Figure 19: Side view of a crystal microphone.

external microphones. External in the sense that they are placed for example in front of the guitar at a certain distance or in some cases attached to the guitar near the sound hole with a clip.

Crystal Microphone Crystal microphones use piezoelectric crystals as a way to generate an electric signal. Figure 19 shows a cross-section view of a crystal microphone. When the crystal is deformed it creates a voltage as a consequence of the piezoelectric effect. A diaphragm is attached to a thin strip of piezoelectric material. The displacement of the diaphragm is deflected to the crystal which acquires opposite charges on the two sides of the crystal. The charges are proportional to the deformation of the crystal and disappear when the stress of the crystal disappears. Hence, the created voltage is proportional to the changes in air pressure created by a sound wave. Crystal microphones have a relatively large electrical output voltage. Therefore, they are practical in some sound applications. However, the frequency response is not of good quality due to the relatively high mass that has to be moved. As a result crystal microphones are not usually used in quality audio applications.

Dynamic Microphone In dynamic or magnetic microphones an output signal is produced as a result of the motion of a conductor in a magnetic field. Figure 20 shows the basic structure of a dynamic microphone. The microphone relies on electromagnetic induction, also called the dynamo principle, where a current is induced to the conductor when it moves in the magnetic field. In the most common dynamic microphone the coil functions as the conductor which moves in the magnetic field. The coil runs around the center piece of the magnet. Moreover, the coil is attached to a diaphragm which is moved by sound pressure. The voltage created by a dynamic microphone is relatively small. On the other hand, the impedance of the coil is low, which means that even if the output voltage of the microphone is low the power generated by the microphone is necessarily not small. The high-frequency response of a dynamic microphone can be good if the mass of the coil is kept small.

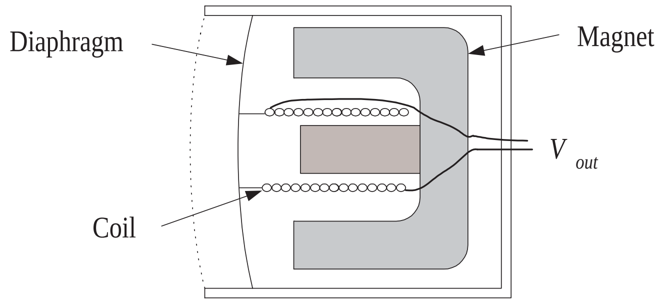

Figure 20: Cross sectional figure of a dynamic microphone. The coil runs around the center piece of the magnet.

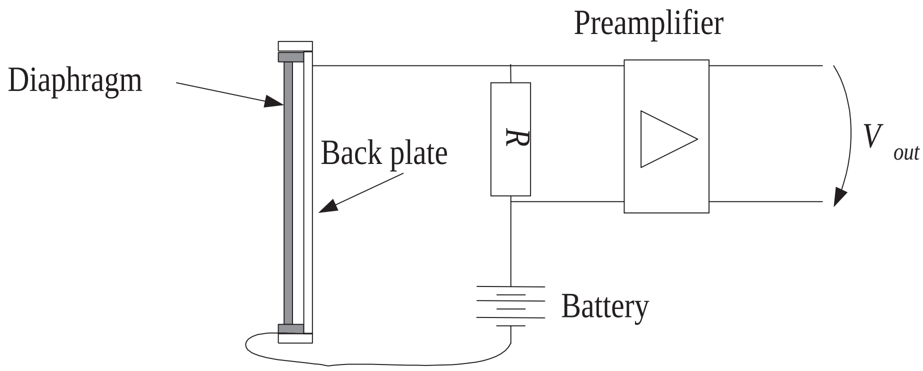

Figure 21: A condenser microphone with a battery to charge the metal plates and a preamplifier to improve the signal-to-noise-ratio.

Condenser Microphone In condenser microphones sound pressure changes the spacing between a thin metallic membrane and a stationary back plate. The plates are charged with a DC voltage source. When the plate spacing changes the charge creates a current through a resistor. Figure 21 depicts a simple setup of a condenser microphone. A condenser microphone has a very high source impedance. Therefore, a preamplifier is usually embedded to the microphone itself, placed in cascade after the resistor. The preamplifier decreases the output impedance and increases the signal level. Since the diaphragm can be very light the high-frequency response of the microphone can be of good quality. In addition, the overall frequency response of a condenser microphone can meet demanding specifications.

2.4.2 Contact Microphones: Piezo Pickup and Elastic Electret Film Pickup

The name contact microphone reveals that they are in contact to a vibrating surface. They are also often called piezo pickups, because the most common contact microphone relies on the piezoelectric effect. However, there are also other possibilities to create a contact microphone. In this study pickups used in the measurements were made out of elastic electret film.

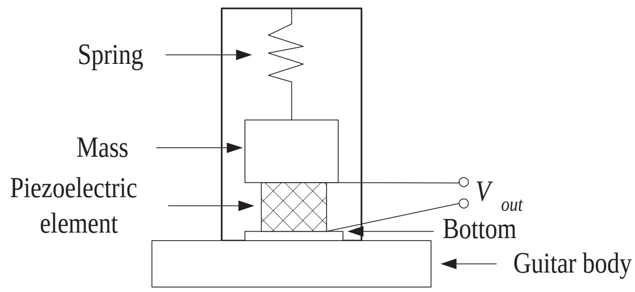

Figure 22: Side view of a piezo microphone attached to a surface, which in this case is the body of a guitar.

Contact microphones can be used both for an acoustic and an electric guitar. The placement of the microphone effects strongly the produced sound. When a contact microphone is used with an acoustic guitar it can be attached somewhere out- or inside of the body. Placing a microphone to the upper bout or upper part of the body enhances high-frequencies, while a placement to the lower bout enhances low-frequencies (Denyer, 1982)(pp.36). Placing the microphone under the saddle in the bridge is a good and commonly used option. This is because the vibrations of the strings are captured well. When contact microphones are used with electric guitars they are attached to the bridge. This inherits from the reason that the resonances of a solid body are weak compared to the ones in an acoustic guitar and therefore the string vibrations would not be sensed properly. However, connecting a contact microphone to the bridge of an electric guitar is not straightforward and can cause problems. First properties of the piezo pickup are discussed and then of the elastic electret film pickup.

Piezo Pickup A piezo pickup works on the same principle as a crystal microphone, i.e., the piezoelectric effect. It has a piezoelectric crystal attached to a mass which is connected to a spring. Figure 22 shows an arrangement that includes the surface it is attached to, the guitar body. As the body of the instrument vibrates the piezoelectric element is exerted by a tension or compression between the bottom piece and the mass. The force, F, exerted to the piezo element by the mass, m, is proportional to the acceleration, a, of the mass, and the weight of the mass after Newton's second law

$$F = ma. \tag{4}$$

The weight of the mass affects the force applied to the piezo element. Hence, the mass affects the level of the output volume. Furthermore, the output voltage is proportional to the acceleration. The time integral of acceleration is the velocity which is the preferred output variable. This means that an integrator should be cascaded with the pickup output signal to obtain a proper signal. In practice, a lowpass filter is used to implement the integration. Otherwise the spectrum of the output signal would be unbalanced and the level of low-frequency components would be too low. In practice piezo pickups also have a preamplifier to increase the output level, balance the frequency response, and improve the signal-to-noise ratio (SNR).

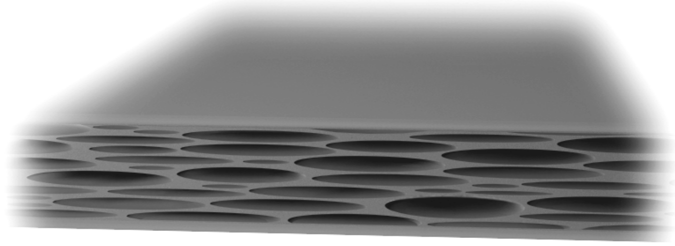

Figure 23: Computer model of EMFi film. The electret film material contains flat gas bubbles. The film is coated with metal electrodes on both sides and is about 50 μm thick.

Elastic Electret Film Pickup A novel technology, alternative to the traditional acoustic guitar pickups such as piezoelectric sensors, is based on the EMFi (electromechanical) film (Backman, 1990; EMF, 2002). The EMFi film can be used as an actuator as well as a sensor material. It has been found to be well suited to mechanical vibration sensing in general. The EMFi film pickup used in acoustic instruments acts as a contact microphone that reacts to velocity changes.

Figure 23 illustrates the structure of the EMFi film. It is shaped as a thin and light strip. The film consists of a permanently charged dielectric material that is filled with gas bubbles. The charged material is placed between two plastic sheets that are coated with electrically conductive layers and shields. Therefore, the structure behaves like a condenser microphone. Variations in the pressure acting on the film are converted to corresponding changes in film thickness. When a constant charge exists in the capacitance, the alteration of thickness is converted to voltage change between electrodes. Thus, when the dielectric material has a permanent charge, an electret type pressure transducer is obtained, without the need of an external polarization voltage source. The gas bubbles increase the sensitivity of EMFi as a pressure sensor due to higher compliance of gas compared to plastic materials. Furthermore, EMFi has good impedance match with wooden parts of an acoustic instrument.

The EMFi material can be easily manufactured into a thin and small strip that fits under the saddle of a string instrument. Figure 24 illustrates the B-Band$^{\mathrm{TM}}$ EMFi pickup manufactured by EMF Acoustics Ltd. It consists of the transducer strip part and the cable part that is a "passive" strip similar to the transducer part but without a charged dielectric.

Compared with piezo transducers, EMFi-based electret transducers are less sensitive and require more voltage gain in the first stage of the preamplifier to achieve a high SNR. Also, the capacitance of practical B-Band transducers is relatively small, thus requiring a preamplifier with a very high input impedance, located as close as possible to the transducer.

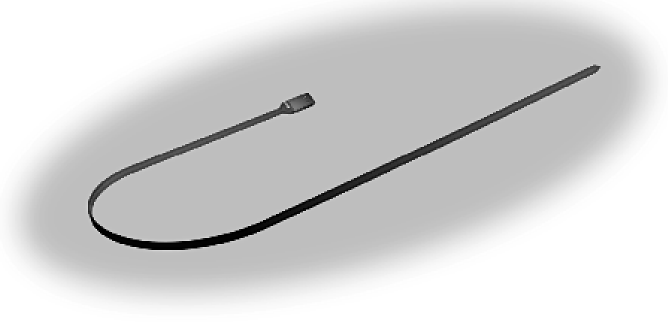

Figure 24: B-Band^{TM} EMFi-type under-saddle pickup element with cable. The thickness of the element is about 0.3 mm. The active sensor area is about 80 mm in length and 3 mm in width.

2.4.3 Magnetic Pickups

Electric guitars use magnetic pickups that respond to string vibrations in the magnetic field of the pickup. Magnetic pickups react to changes in their magnetic field. Strings used in electric guitars are made of steel. Therefore, when they vibrate in the magnetic field of the pickup they interact with it. As a result, an electric voltage signal is induced that is proportional to the velocity vibrations of the strings.

A magnet pickup consists of a coil with a magnetic core. It has a permanent magnet with a very thin copper wire wrapped around it several thousand times. The wrapped copper wire and the magnet form an electric coil. Next the principle of magnetic pickups is discussed.

Principle of Magnetic Pickups Magnetic pickups make use of electromagnetic forces, much in the same way that dynamic microphones do. In a dynamic microphone the vibrating air moves a voice coil in a magnetic field. In magnetic pickups, a substance (steel string) is brought to the magnetic field and it reacts with the magnetic field.

A magnet creates a static magnetic field around it which the string passes through (Fig. 25). In a static situation the magnetic field is more intense at the south pole of the magnet than at the north pole. This is caused by the ferromagnetic behaviour of the string. When the string starts to vibrate the field changes its shape according to the vibration of the string. Figure 25 depicts how a magnetic field is affected by a vibrating string in it. Figure 25 (a) shows how the string pushes the magnetic field towards the south pole as the string gets closer to the south pole. Figure 25 (b) depicts the situation when the strings pulls the magnetic field away from the south pole as the string moves away from the pickup.

The voltage induced in a magnetic pickup is proportional to the velocity of the string, i.e., derivative od the magnetic field changes. In addition the width of a pole piece

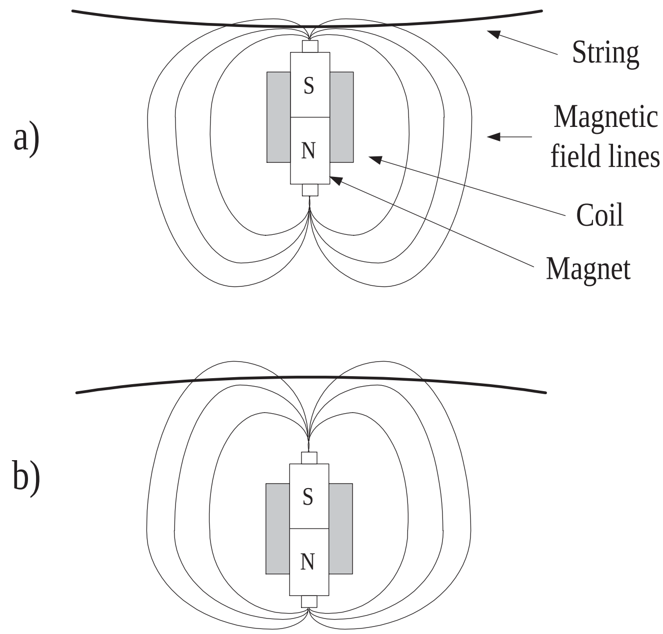

Figure 25: Force lines of magnetic field when the string is (a) close and (b) faraway of a pole-piece.

affects the final output. The pole pieces do not sense the string at a single point, rather over an area which is called the aperture of the pickup. For single coils the aperture is about 2.5 cm and for a humbucker circa 6 cm (Tillman, 2001). An approximation of the effect of the aperture is calculated by averaging over its length. This leads to an integral, which results in a 6 dB per octave low-pass comb filter (Tillman, 2001). The aperture of a pickup causes a low-pass comb filter, so that the -3 dB point is at a lower frequency for a humbucker pickup than a single coil pickup.

Single Coil Pickups Single coil pickups have typically six individual pole-pieces. Although there are pickups with a single pole-piece, individual pole-pieces are more commonly used than one pole-piece pickups. A pole-piece concentrates and directs the magnetic field to properly sense the string vibrations. Usually the south pole of each pole-piece is on top, close to the string. The height of a pole-piece can be adjustable. Adjusting a pole-piece closer to the string increases the output level. However, if the pole-piece is very close to the string the output will be distorted. The distortion is caused by the force of the magnetic field exerted to the string.

Humbucker Pickups A coil in a magnetic pickup works as an antenna and senses surrounding electromagnetic radiation. In practice this means that they are sensitive to magnetic fields produced by amplifiers, fluorescent lights, and other electrical appliances nearby. As a result the output signal of the electric guitar has an unwanted disturbing signal often described or called a hum. The hum typically has a 50 Hz cycle (at least in Europe), which results from the output cycle of power-distribution networks. Humbucker pickups are able to cancel the unwanted hum, hence the the name

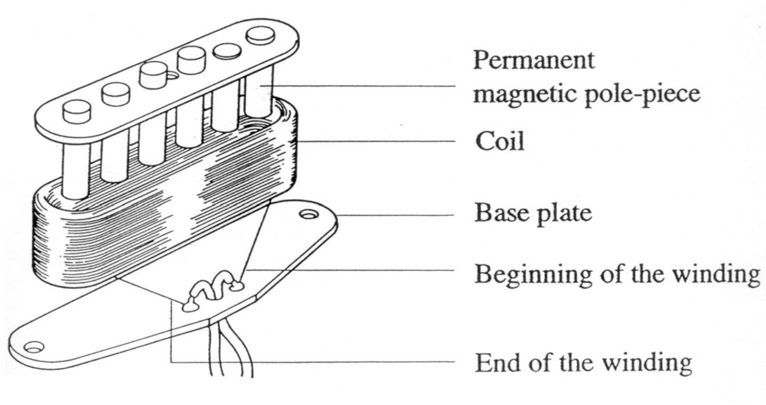

Figure 26: Structure of a single coil pickup. The single coil pickup has six individual pole-pieces with a copper wire wrapped around.

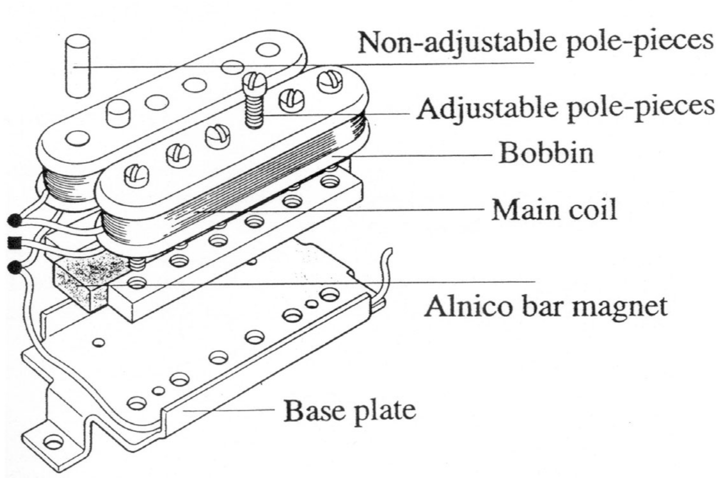

Figure 27: Structure of a humbucker pickup. The magnetic polarities alternate in the pole pieces to cancel out unwanted disturbances and hum.

of the pickup. Figure 27 shows the construction of a humbucker pickup. Humbucker pickups have two coils, and the one shown in Fig. 27 has both non- and adjustable pole-pieces. Humbucker pickups cancel the unwanted hum because they have two coils that are combined out-of-phase. The hum voltage induced to the coils by the magnetic field is cancelled. However, to preserve the signal created by the vibrating strings, each pole-piece has to have an opposite magnetic polarity when compared to the next pole-piece.

The two coils of a humbucker pickup can either be connected in series or parallel. Both alternatives affect significantly the timbre of the output. A series connection gives a high-levelled output, since the resulting impedance is the sum of impedances of the coils. Therefore, the output voltage is large. In a series connection high-frequencies are somewhat attenuated, but the SNR is good. A parallel connection gives a low output level, since the resulting impedance is $Z = (\sum_{k=1}^{i} 1/Z_i)^{-1}$. High frequency components are not attenuated, but the SNR is not as good as in a series connection.

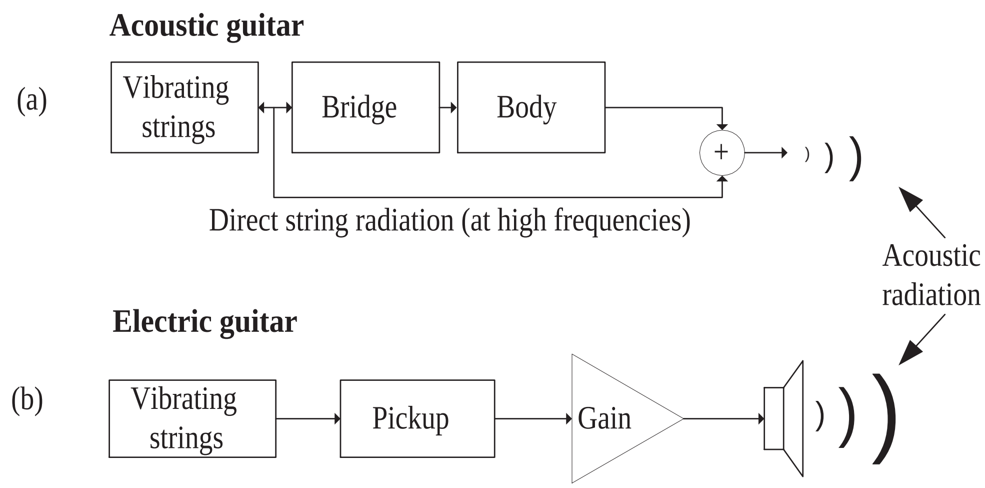

Figure 28: Simple schematic of sound production for (a) acoustic and (b) electric guitar.

2.5 Acoustic vs. Electric Guitar

The acoustic guitar differs from the electric one both structurally and in the produced timbre. This subsection sums up the differences described earlier and discusses differences not mentioned so far. It is beneficial to understand these differences, since one of the methods proposed in this report aims to make an electric guitar sound more like an acoustic one.

Differences Figure 28 shows a simplistic view of the sound production for (a) an acoustic and (b) an electric guitar. The fundamental difference is that the electric guitar does not have a hollow body that amplifies or colors the string vibrations. Therefore, the strings vibrate longer in an electric guitar than in an acoustic instrument, i.e., the electric guitar has a longer or better sustain. While the hollow body colors the timbre of an acoustic guitar, the timbre of an electric one is colored by the placement and the low-pass characteristics of the used magnetic pickup, the amplifier, and the loudspeaker. In addition, the airborne sound of an acoustic guitar contains the straight radiation of the strings. The straight radiation of the strings adds high-frequency components to the sound which are missing in the electric guitar.

The termination of the strings in acoustic guitars also differs somewhat from the electric ones. The difference is more in the way the bridge lets the strings vibrate. Therefore, beating of the harmonics is acoustic guitars occur more than in electric guitars. Beating of harmonics results when vibrations of different polarizations get summed. Also, the designs of bridges used in electric guitars vary more than in acoustic guitars.

Table 2 lists some of the main differences between an acoustic and an electric guitar: body, sound production, straight radiation, beating of harmonics, sympathetic vibrations, feedback sensitivity, output level, sustain. The style of the table is a simplifying

Table 2: Table of differences. Acoustic vs. electric guitar.

Difference	**Acoustic**	**Electric**
Body	hollow	solid
Sound production	bridge and body	pickup, amplifier, and loudspeaker
Resonant coloring	body resonances	comb-filter (pickup placement)
Straight radiation	existent	non-existent
Beating of harmonics	more	less
Sympathetic vibrations	more	less
Feedback sensitivity	more	less
Output level	smaller	larger adjustability
Sustain	shorter	longer

one, but gives compact view on some of the main differences.

Common properties and aspects Both the acoustic and electric guitar are plucked string instruments. The string vibrations are sensed and amplified a way or another. A player who can play an acoustic guitar can also play an electric one. So even if many differences between the acoustic and electric guitar have been pointed out, they are virtually the same instrument.

Guitar players use often effects to color the timbre of the produced signal. Effects are typically cascaded before the amplifier, but can also be set in parallel with each other or with the main amplification path. The order in which different effects are set has a significant role in the resulting timbre. Often used effects include reverb units, flangers, choruses, phaseres, wah-wahs, equalizers, and distortion, to name a few (Zölzer, 2002; Zölzer, 1997; GEO, 2002). Typically, electric guitar players use effects more extensively than acoustic guitar players.

3 Model-Based Analysis of System Signal Paths

Sound synthesis is an active field in computer music and audio research. The common goal is to create sound from scratch with a computer, without an acoustic instrument. There are numerous ways to synthesize sound and it depends on the goal, resources, usability etc. which one to be used. Musically, equivalence of a sound with an existing instrument or sound source is not necessarily very important. Whereas, instrument or sound source modeling strive for physically and perceptually accurate reproduction of a sound. Physical modeling and model-based sound synthesis are methods that attempt to do just that. Their goal is to simulate and duplicate the most important sound generation properties of a particular musical instrument, or more generally, a sound source. While, other synthesis methods model the produced sound, i.e., the waveform, spectrum etc.

The body modeling techniques and aspects discussed in this work are related to sound synthesis and sound source modeling. The body modeling methods can be addressed (at least) from two synthesis point of views: source-filter synthesis and physical modeling. Next source-filter synthesis and physical modeling, including string modeling, will shortly be discussed keeping in mind their connection/relation to this work. Then the signal paths in acoustic and electric guitars are viewed from a model-based view point. Sound synthesis methods are discussed more thoroughly, e.g., in (Roads, 1995; Tolonen et al., 1998) with references and see (Smith, 1991; Tolonen et al., 1998) for classifications of different methods.

3.1 Source-Filter Synthesis

Source-filter modeling, often also known as subtractive synthesis, is based on linear filtering of an input signal. The method is separated into two parts: an excitation signal (source) and a resonator (filter). The excitation or input signal has typically a rich spectrum, such as white noise or a periodic pulse. The filter block emphasizes some frequencies and attenuates others. Furthermore, both the source and the filter can be static or time-varying. The technique can be utilized to produce synthetic speech, e.g., (Moorer, 1985). For musical applications of the method see (Roads, 1995).

The methods discussed in this work can interpreted as source-filter synthesis or modeling. The playing of a guitar functions as a source signal and the body model acts as an LTI filter. As a result, the timbre of an acoustic guitar is modeled.

3.2 Physical Modeling

Physical modeling aims for mathematical formulations and signal processing algorithms that implement physically and perceptually the most prominent properties of a sound source (e.g. an instrument). Physical modeling investigates digital ways to model the physical phenomena rather than the waveform produced by the sound source. If the physical model is well structured the parameters controlling model have a correspondence with the real world and are physically meaningful. Hence, a physical model can be a more useful one to the end user than a model with purely abstract parameters. However, parameter estimation is a highly non-trivial task, that still has a lot of undergoing research. Here physical modeling is dealt at a principal level, with a relation to this work. For a deeper discussion on physical models see, e.g., (Tolonen et al., 1998; Välimäki and Takala, 1996).

Physical modeling is a combination of mathematics, physics, measurements, and digital signal processing. Often including using knowledge of the human auditory system (perceptual modeling) and creativity. The backbone of physical modeling often lies in physics and mathematics. The most important discovery or development was the formulation of the wave equation, also known as the Helmholtz equation. It describes the propagation of waves in a homogeneous medium. To solve the wave equation, as a function of time and place, the initial state of the system has to be known, the boundary conditions, acting external forces, and the internal losses. However, physical modeling would not be anything without measurements of the studied sound source. Measurement results give vital information of the physical characteristics of functioning components of the source. Furthermore, acoustical measurements give information about the produced sound field and hence information, e.g., mode or partial behaviour of an instrument. On the basis of measurement data parameters for a mathematical model can be derived. This is not always straightforward and can involve a lot of non-ideal factors and optimization.

Modeling the essential characteristics of an instrument can also involve modeling of the human auditory system and discarding features that cannot be perceived. The modeling of the human auditory system can be realized by using auditory models, e.g., (Zwicker and Fastl, 1990). By making perceptual assessments through listening tests non-perceivable phenomenon can be discarded in a model (Järveläinen, 2001). This way the computational load of an instrument model can be reduced. Furthermore, the computational load of a model is often optimized, so that real-time implementation of the model would be possible. Real-time implementation is often an important objective, but at least from an academic point of view it is not always the primary concern.

In physical modeling, a musical instrument is typically divided into parts with respect to their functional properties (Välimäki and Takala, 1996). Often an instrument model is divided into three parts: the excitation mechanism, the resonator, and the radiator. In many instruments the excitation and the resonator are connected with a non-linear feedback. When the functional parts are modeled separately it gives a possibility to combine models of different instruments and hence create new interesting timbres at the synthesis stage. In the case of the guitar, or string instruments, the resonator stands for the vibrating string and the body of the instrument acts as the radiator. Hence, the techniques discussed in this study concentrate on the radiator part of an instrument and more specifically on the guitar body.

Now that sound synthesis and source modeling has been approached on a general level a popular string modeling method will be approached as a part of a way to formulate the work of this thesis. After this the system signal paths in a guitar will be discussed from a model-based approach.

3.2.1 String Modeling

String modeling will be discussed through what is called waveguide synthesis. This is because it is so far the most important physical modeling method, both academically and commercially. The theory of wave guides was mostly formulated by Smith (Smith, 1986; Smith, 1987; Smith, 1992; Smith, 1993). Waveguides are well suited for modeling a vibrating string, a narrow acoustic tube, or a thin bar. The technique has been applied for modeling instruments such as the acoustic guitar, the violin, brass instrument tones, to name a few (Smith, 1996; Karjalainen et al., 1998). A great advantage of the waveguide approach is that it models a physical phenomenon directly in a digital discrete way. In other words, the need for creating a continuous-time model that has to be discretized is discarded. Discrete-time systems that implement a waveguide model are called waveguide filters (WGFs).

Waveguides are reported extensively in references mentioned above, here waveguides are shortly presented in the light of string behaviour both in the acoustic and the electric guitar. Figure 29 depicts a string model with two submodels. The submodels correspond to the vertical and horizontal vibration polarizations of the string (see Sec. 2.1 and Fig. 2). Longitudinal vibrations in a guitar string can be considered to only have a minor contribution to the final timbre and can therefore be left out from the model. As its simplest, the length of a delay line can be set only to an integer value. However, to meet a desired pitch of a string the length (of a delay line) can be set more accurately with the help of a fractional delay filter, e.g., with a first-order allpass filter (Jaffe and Smith, 1983), a linear interpolator (Sullivan, 1990), or a third order Lagrange interpolator (Karjalainen and Laine, 1991).

The plucked excitation propagates in the delay lines as a wave form to opposite directions. Losses of the string are concentrated to filters at the terminations of the waveguide. The termination reflections filters, $R(z)$, achieve the frequency dependent decay. The frequency dependent and over all decay is different in the acoustic and electric guitar, hence the reflection filters are different in these cases. The beating and two-stage decay phenomena of a string can be modeled by using two parallel delay lines with different parameter values (Karjalainen et al., 1998). Dispersion of harmonics can be modeled with cascaded first-order allpass sections as described in (Duyne and Smith, 1994). Modeling of tension modulation with WGFs is discussed in, e.g.,

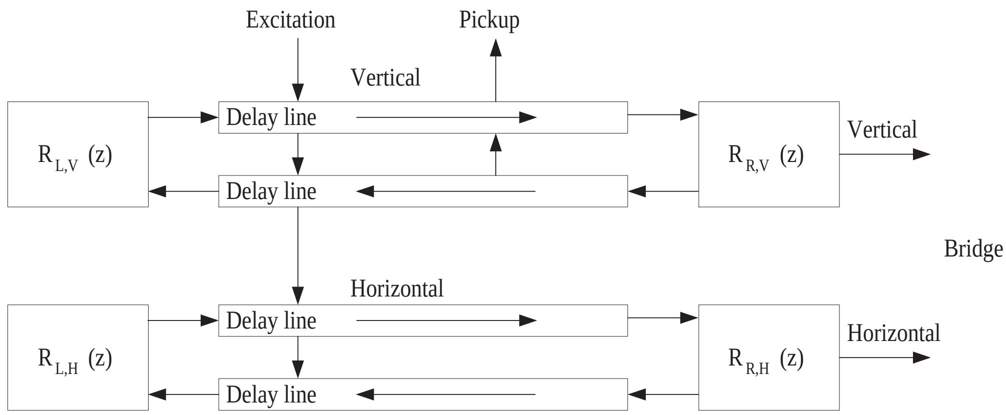

Figure 29: Dual-polarization string model with vertical (V) and horizontal (H) submodels, termination reflection filters $R(z)$ (L = left, R = right), and a shared excitation point. The outputs, however, can be thought as separate: one for a pickup of an electric guitar and another for the bridge of an acoustic guitar.

(Tolonen et al., 2000). On the whole, WGFs are a computationally efficient way to model many of the characteristics of a real string.

Now considering the role of the guitar body in the model in Fig. 29. The output for an electric guitar model (in Fig. 29) is only taken from the vertical submodel since the magnetic pickup only senses this vibrational polarization. Furthermore, the position of the output signal represents the position of the pickup. For an acoustic guitar model the string model output is taken from both submodels and directed to a bridge model and then to a body model. This kind of a formulation deviates from the popular and efficient commuted waveguide synthesis method, where the effect of the body is included in the excitation signal (Smith, 1993; Karjalainen et al., 1993b). But this kind of a formulation suits better the purposes of this work. For compact presentation and ease of comparison, the string model shown in Figure 29 combines the acoustic and electric guitar string, however, they still have to be understood as separate strings.

3.3 System Signal Paths

Enhancing the timbre of a guitar pickup signal to a more acoustic one means modeling the transfer function from the pickup signal to the air radiated sound. Next we look at the signal paths involved in the process of altering the timbre of a pickup signal. The acoustic guitar is considered to be equipped with an undersaddle electret bridge pickup and the electric one with a magnetic pickup (see Sec. 2.4: p. 28 and 29, respectively). By viewing the situation through functional block models the derivation of algorithms to alter a timbre to another can be more understandable. The algorithms and methods will be described in the next Chapter.

Figure 30 displays a block diagram of a generic guitar model. It is a model describing the signal paths in a guitar from string vibration to an output. The model is generic

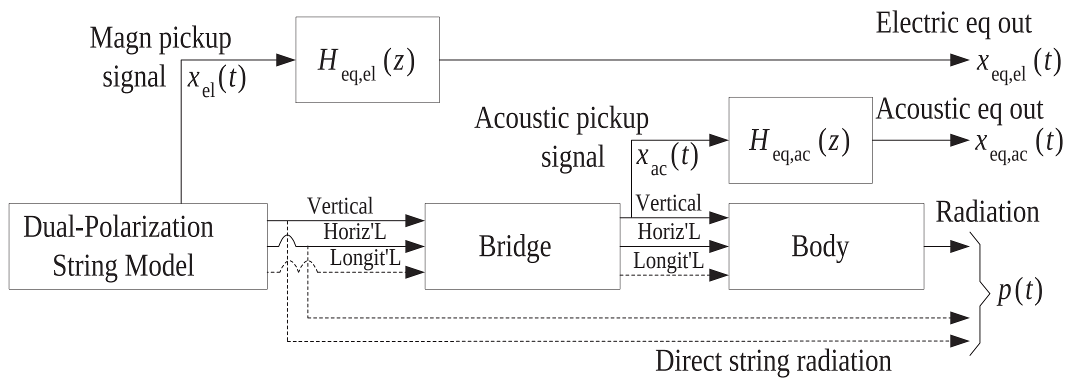

Figure 30: Block diagram of a generic guitar model. System signals paths both for an acoustic and an electric are shown. Details of the dual-polarization string model are depicted in Fig. 29.

in terms that it includes the important functional elements both for an acoustic and an electric guitar. As an useful simplification the models and blocks are assumed LTI.

The acoustic guitar (in Fig. 30) consists of the following chain:

$$\textit{excitation} \Rightarrow \textit{string} \Rightarrow \textit{bridge} \Rightarrow \textit{body} \Rightarrow \textit{radiation},$$

where *excitation* is the signal fed to the string model to simulate the pluck event. As mentioned before a guitar string vibrates in three directions: vertical, horizontal and longitudinal. The vertical motion through the bridge is the polarization an undersaddle pickup effectively sense. Fortunately, this is the direction that has the strongest coupling between the string and the bridge. After the bridge the signal flows are separated: (I) the pickup signal, $x_{\text{ac}}(t)$ (ac = acoustic guitar), which consists only of the vertical vibrations surpasses the body and (II) all polarizations are coupled, more or less, through the body. The air radiation, $p(t)$, consists of the body radiation and the direct string radiation. The filter block $H_{\text{eq,ac}}(z)$ (eq = equalization) stands for the body model filter that is cascaded with the acoustic guitar pickup signal.

In the electric guitar (in Fig. 30) the signal flow is the following:

$$\textit{excitation} \Rightarrow \textit{string} \Rightarrow \textit{pickup}.$$

In contrast to the acoustic guitar signal path, there are no parallel signal paths in this chain. The magnetic pickup signal, $x_{\text{el}}(t)$ (el = electric guitar), consists mainly of vertical vibrations. The position of a pickup results in a comb filter effect and the lowpass effect of a pickup are not visible in the chain. The magnetic pickup signal is fed to the $H_{\text{eq,el}}(z)$ filter block that emulates the missing body of an acoustic guitar. That is to say this block makes the electric guitar pickup signal, $x_{\text{el}}(t)$, sound more like the radiation of an acoustic guitar, $p(t)$.

4 Body Modeling Methods and Techniques

Physical modeling of the behaviour of an instrument body is rather complicated. At low frequencies the Finite Element Method (FEM) (Bretos et al., 1999) can be used to approximate the parameter values of a resonance, but at high frequencies it is only capable of a more statistical approximation. The radiation pattern of an instrument resonator is different for different vibration modes. These facts further complicate the use of a detailed physical model of an instrument body in real-time sound synthesis and audio effect applications. Therefore, this study uses digital filters as a mean to model guitar body responses.

This chapter discusses three, that were studied and used in this study, to model the response of a guitar body. In this context body modeling means simulation of an instrument body response with digital filters. In this work the objective of a body modeling filter is to synthesize the important characteristics of a resonant body that are missing from a pickup signal.

First, a method that could be called the deconvolution method (DECON) is discussed. It is based on frequency domain division, where a transfer function is obtained by dividing two spectra (Karjalainen et al., 2000b). The discussion also includes issues related to measuring instrument bodies in general. Secondly, a technique is viewed that relies on responses measured with the help of an impulse hammer. Here it is called the modified impulse response method (MIR). The principle of both these methods will be discussed and the necessary measurement setups will be described. Finally, issues related to modeling an instrument body with reverb algorithms will be shortly covered. This chapter covers the concepts of the themes and the next chapter discusses the implementational issues of them.

4.1 Deconvolution Method

This method is founded on determining the transfer function and the corresponding impulse response from a pickup signal to air radiated sound by calculating the deconvolution through frequency-domain division. First, the mathematical formulation of the method will be presented and then the required measurements will be described.

By recalling the signal paths in Figure 30, on p. 38, we can understand that the LTI transfer function from a vertical bridge excitation $V_{\mathrm{b}}(\omega)$, to the acoustic radiation $P(\omega)$

can be assumed as

$$H_{\mathrm{a}}(\omega) = \frac{P(\omega)}{V_{\mathrm{b}}(\omega)}, \tag{5}$$

where ω is the angular frequency and the variables with capital letters are Fourier transforms. The corresponding transfer function, $H_{\mathrm{b}}(\omega)$, from a vertical bridge excitation to the pickup output, $X(\omega)$, is then

$$H_{\mathrm{b}}(\omega) = \frac{X(\omega)}{V_{\mathrm{b}}(\omega)}. \tag{6}$$

Hence, the transfer function from a pickup signal to air radiation is

$$H_{\mathrm{eq}}(\omega) = \frac{H_{\mathrm{a}}(\omega)}{H_{\mathrm{b}}(\omega)} = \frac{P(\omega)}{X(\omega)}. \tag{7}$$

Moreover, the impulse response of the transfer function can be calculated by frequency-domain deconvolution as

$$h_{\mathrm{eq}}(n) = DFT^{-1}\left(\frac{DFT^{-1}p(n)}{DFT^{-1}x(n)}\right), \tag{8}$$

where $p(n)$ and $x(n)$ are discrete-time signals of the air radiation and bridge pickup responses, respectively, and n is the discrete time variable, and DFT is the Discrete Fourier transform. For the actual calculations of the Fourier Transform the Fast Fourier Transform, FFT, algorithm has been applied.

4.1.1 Measurement Setups in General

For the deconvolution method two slightly different measurement setups were used. They differ from each other based on the final utilization target of the body model filter. That is to say, the measurement setup is different when the body modeling filter will be used for an acoustic or an electric guitar pickup signal. All the measurements discussed in the following have been carried out in an anechoic chamber. This way the effect of possible reflections, present in a normal room, are minimized. Hence, the sound field will be created by the sound source itself and does not contain reflections from surfaces, such as, the floor and walls. Next, some general issues of the measurement setup will be discussed and then the setups applied for obtaining a filter for both guitar types are described. The measurement setup for acoustic guitars is described in more detail than for the electric guitar. However, most of the facts presented for the acoustic case apply also for the electric one and are therefore left out from the latter description.

The radiation pattern of an acoustic guitar is irregular in the sense that when the examination point (place of the measurement microphone) is changed even just a few centimeters the amplitudes and phases of corresponding frequency components (may) change. Therefore, calculating the transfer functions for several acoustic field points and taking their average could be an option to obtain a more general transfer function. However, slight changes in different high-frequency components and their phases for the acoustic field and the excitation signal can result in a corrupted body model filter. For this reason, the body modeling filters calculated in this work are estimating the transfer function from a pickup signal to the acoustic field for one point.

The measurement microphone used in these types of measurements is typically placed 1 m in front of the sound source. This way the setup approximates a free-field situation. The measurements were done in an in-situ fashion, i.e., the guitar was on the lap of the player as in a real-life situation. In addition to having a proper distance from the microphone, the player should not make additional unwanted sounds besides the playing sounds. Breathing and moving are typical such additional sounds. Also, the playing should be as correct as possible to avoid unwanted sounds. For example, when plucking a string too forcefully the string will hit some frets, and as a result produces high-frequency components (Rank and Kubin, 1997) that are not part of what the body of the instrument radiates. Meeting these criteria can sometimes be surprisingly difficult. One thing that can help this situation, is to have a possibility for the player to hear what is being recorded.

4.1.2 Ways to Excite Guitar Bridge

To obtain an adequate equalization filter the excitation signal that drives the bridge should be rich in its frequency content (Välimäki et al., 1996). The bridge can be excited in two different ways: (I) The bridge is hit with an impulse hammer or (II) the strings are plucked as in normal playing. In this work the bridge was excited with an impulse hammer for body responses measured with the MIR method. Playing was used as an excitation method for the deconvolution method, since as stated in (Karjalainen et al., 2000b), hitting the bridge with a hammer strongly emphasizes the lowest modes.

——— **Exciting the Bridge with a Hammer** ——— Hitting the bridge is a proper way to drive or excite the bridge with a spectrally broad signal. On the other hand, during the contact of the bridge and an impulse hammer the interaction and sluggishness of the masses involved result in a non-ideal excitation. That is to say the excitation is not an ideal impulse: in the time domain the impulse is spread and in the frequency domain the magnitude response has a lowpass characteristics. The degree of the lowpass characteristics depends, among other things, on the mass of the impulse hammer. Thus, the use of a light-weight impulse hammer, e.g. a pen, results in a flatter magnitude response compared to a heavy hammer. The lowpass filtering effect caused by a non-ideal excitation can also be partially reversed by measuring the force signal of the impulse hammer. The force signal is measured in parallel with the acoustic field and the corrected response is obtained by using Eq. (8). However, noise problems are always present in these kinds of methods, as will be seen later in this section. Another thing that has to be avoided during a measurement of this kind, is a so called double hit. A double hit means that the impulse hammer touches the bridge twice in a very short time period.

If the impulse hammer has audible resonant modes in itself the hammer should be discarded. For example, a wooden pencil produces additional unwanted sounds and should not be used. Mentioned as a counter part, pens do not usually have as distinct and audible resonances as pencils.

——— **Exciting the Bridge by Playing** ——— To obtain satisfactory measurement data the played notes should produce a spectrally rich signal. For this work the excitation signal consisted mainly of major chords moving chromatically upwards, from an open E-major to a D-major played on the 10th fret. After the 10th fret the strings easily hit the frets and thus made unwanted noises. The amount of energy, in acoustic guitars, at frequencies above circa 10 - 15 kHz is very small compared to lower frequencies. In a sense, this results in a worse SNR at high-frequencies than at low-frequencies, since the noise floor of a measurement setup is always present. Hence, reliability cannot always be guaranteed, or taken for granted, for high-frequency information.

By moving chromatically, a semitone at a time, frequencies between two notes are skipped. By using a slide, metal or glass tube, a spectral sweep can be achieved. This way the skipped frequencies can be obtained to the excitation signal. However, using a slide corrupts the resulting filter. This is because sliding the tube across the wound strings the portion of direct string radiation is radically increased, when compared to regular playing. Hence, the resulting filter has excessively strong high-frequency components that are not part of the body. Therefore, the resulting filter sounds unnatural.

Another problem when using a slide is that the termination is not as rigid and fixed as when the strings are pressed down and terminated by the frets. In this situation it is difficult to get a good excitation signal, i.e., a chord without unwanted noises. Such unwanted noises are buzzing sounds that are created when a string is mainly in contact with the tube but very shortly looses the contact. Therefore, it is not advisable to use a slide in this context.

Another solution for the problem of skipping frequencies between semitones would be repeating the measurement, e.g., two times and tuning the guitar a half semitone higher for the second measurement. The coverage of skipped frequencies can naturally be improved by doing more measurements and decreasing the tuning steps correspondingly. However, this can result in the phase and frequency cancelling problems discussed earlier. On the other hand, based on this work it seems to be adequate to use just chords from one tuning and a single measurement.

4.1.3 Measurement Setup for Acoustic Guitar

To obtain a body modeling filter for an acoustic guitar pickup signal, the undersaddle pickup signal and the acoustic response of the guitar are measured simultaneously. The pickup signal is obtained straight from the pickup output while the acoustic response is recorded with a measurement microphone placed 1 m in front of the guitar. The acoustic field is measured only at one point, hence, the corresponding body modeling filter simulates the transfer function form the pickup signal to the particular point.

The distance between the guitar and the microphone is often recommended to be at least one meter to approximate a far field situation. Because of the distance between the microphone and the guitar, it takes some time for the sound to reach the microphone. In contrast, the pickup signal can be considered to be captured instantly, since the electrical signal in the cable propagates at the speed of light. Therefore, a time domain difference between the signals occurs. This time-difference or in-synchronous phenomenon can be corrected by moving the delayed microphone signal n samples backwards. The value for n can be calculated in the following manner

$$n = \frac{df_s}{v} \tag{9}$$

where d is the distance between the guitar and the microphone, v is the velocity of sound in air ($\sim$ 344 m/s) and f_s is the sampling frequency. On the other hand, based on the experimentation done during this work, this synchronization correction does not radically affect the audible results of a target response.

Figure 31 represents body model target responses obtained with the deconvolution method for a flat-top (steel stringed) acoustic guitar. Figures 31 (a) and (b) show time responses of the deconvolution result, so that (b) represents a windowed version of (a). The used window fades in and out the response with a half-Hanning function that is 100 samples long, and total window length is 5000 samples which corresponds to $\approx$ 104 ms when $f_s = 48$ kHz. The time responses show the noisy quality of the filter, even for negative (non-causal) time values. The noisiness results from the noise produced by the measuring system (microphones, amplifiers, etc.) and because the pickup signal $x(n)$ in Eq. (8) has little energy at some frequencies. In effect, when the denominator coefficient in Eq. (8) is very small compared to the numerator the division result becomes large. Magnitude responses of Fig. 31 (b) are shown in Fig. 31 (c) and (d) on a logarithmic and linear frequency scale, respectively.

Figure 32 represents a time-frequency illustration of the response in Fig. 31 (b). Figure 32 shows clearly the noisy quality of the equalization target response. Also, the sustaining character of some of the strong high-frequency resonances is easily notable. These high-frequency components are not present in impulse responses measured by exciting the bridge with an impact hammer and recording the acoustic response, as in Figs. 13 and 14. Therefore, one can understand that the strong high-frequency components are not body resonances. The undersaddle pickup cannot sense the straight

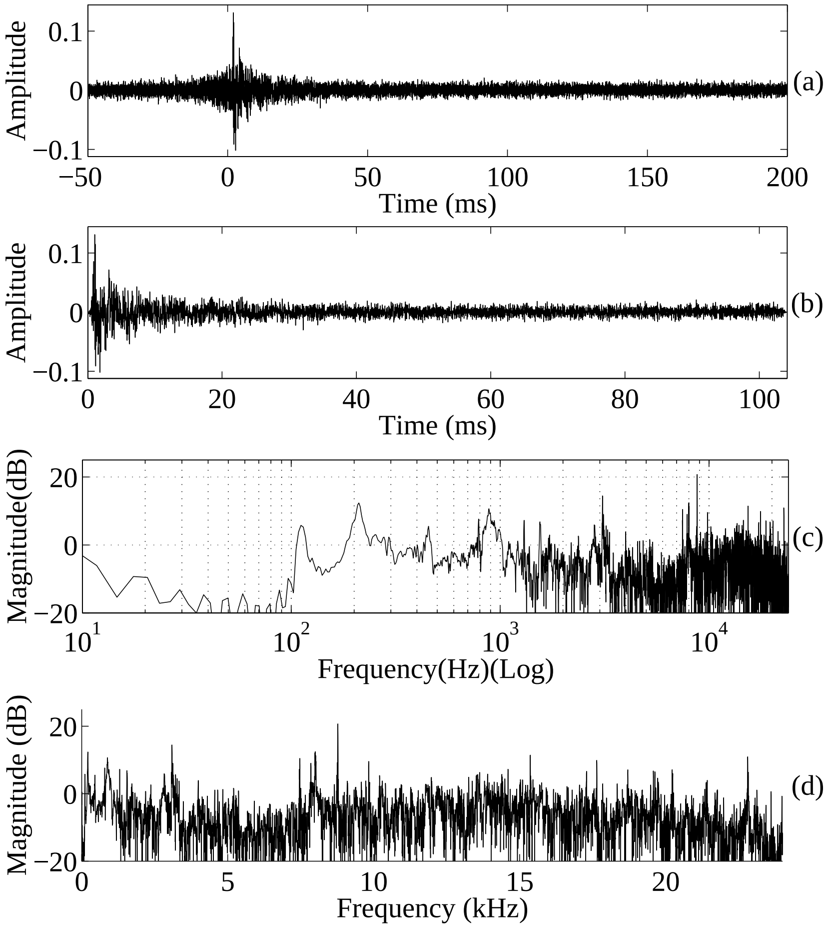

Figure 31: Body-model filter responses used for steel-string acoustic guitar obtained with deconvolution method for a small-sized guitar when excited by playing. (a) shows 250 ms of deconvolution result, (b) windowed version of the result, length circa 104 ms, (c) magnitude response of (b) on logarithmic frequency scale, and (d) on linear scale.

string radiation and is also not able to capture some of the friction noises produced by the plucking procedure. Hence, these frequency components will be present in the deconvolution result. This fact and the denominator and numerator coefficient aspect discussed above, or the combination of these things, result in the strong high-frequency components.

The responses shown in Figure 31 and 32 are from a relatively small [6] sized flat-top guitar, manufactured by Landola (LR-855). Guitars with middle (D-805E) and large (J-855) sized bodies were also measured. Figures for the responses of these guitars can be found in Appendix 1 on p. 96.

[6]Small in this context does not mean a 1/4 or 1/2 sized guitar.

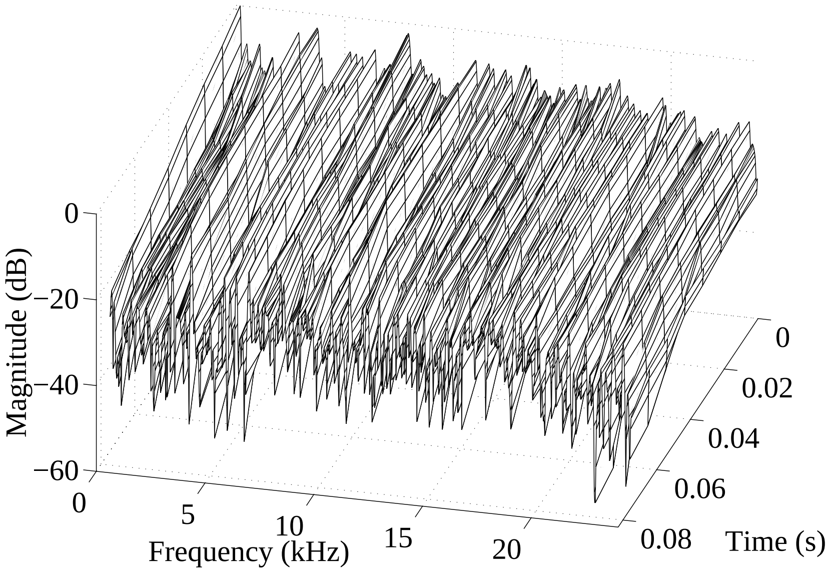

Figure 32: Time-frequency response, when hop-size is 1024 samples and with 50 % over-lap, of body-model target-response shown in Fig. 31 (b).

4.1.4 Measurement Setup for Electric Guitar

The measurement setup for obtaining a body model filter that makes an electric guitar sound more like an acoustic guitar is very similar to the one described for the acoustic guitar. The only actual difference is that the microphone used to capture the string vibration is a magnetic pickup used in electric guitars, rather than a undersaddle pickup. So in addition to the microphone placed one meter in front of the guitar, a magnetic pickup is placed to the sound hole of the guitar. The magnetic pickup used in the measurements was a single coil taken from a Stratocaster copy [7]. Acoustic flat-top guitars can be used in the measurements for this method, since they use steel strings. In contrast, classical guitars equipped with nylon strings cannot be used, since they do not induce a voltage in the coils of a magnetic pickup.

Figure 33 displays target responses obtained for an acoustic flat-top guitar equipped with a magnetic pickup. The figure arrangement is identical with figures for the acoustic guitar in Fig. 31: (a) and (b) show time responses for the deconvolution method, so that the latter one is a windowed version of the former, and (c) and (d) display magnitude responses of (b) on a logarithmic and linear frequency scale, respectively. First what strikes as a major difference, compared to responses in Fig. 31, is the high-frequency behaviour: the level of the high-frequency envelope is higher. This can be understood since the magnetic pickup acts as a low-pass filter. Hence, the equalization filter reverses and compensates for effect. Another difference is the even larger noise level which can be seen in the time domain responses. This is a cause of a worse overall SNR of the setup, when compared to the one used for the acoustic guitar.

[7] More specific information about the pickup was not available.

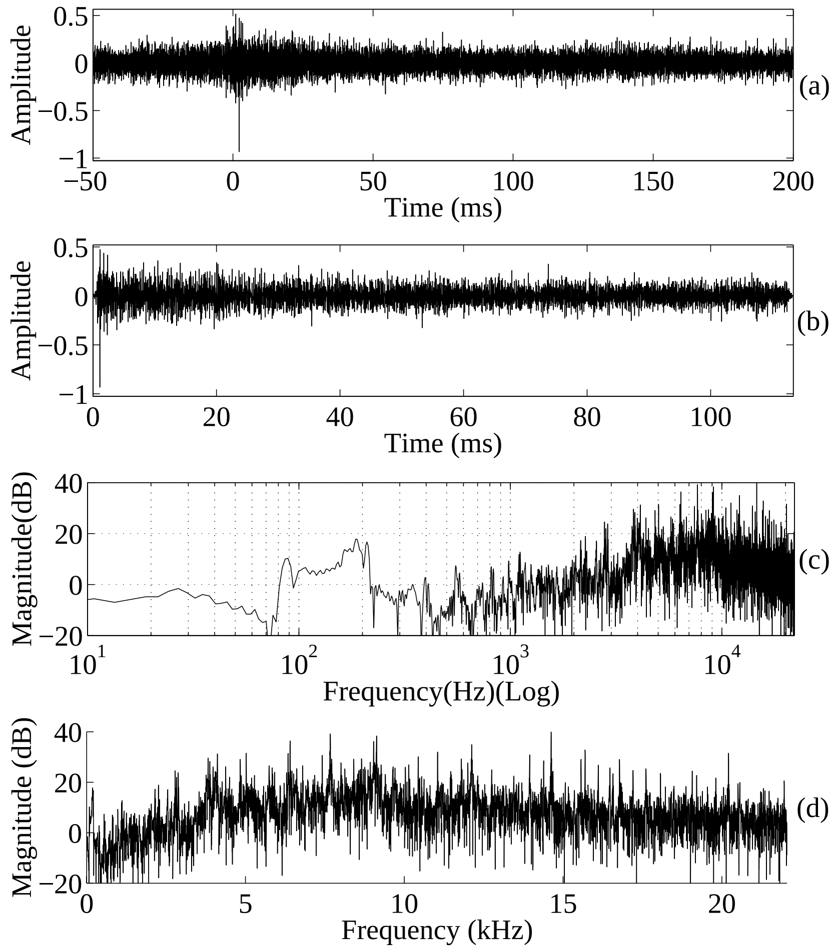

Figure 33: Body-model filter responses used for electric guitars obtained with deconvolution method when acoustic guitar was excited by playing. (a) shows 250 ms of deconvolution result, (b) windowed version of the result, length circa 104 ms, (c) magnitude response of (b) on logarithmic frequency scale, and (d) on linear scale.

This is especially notable at high frequencies where the low-pass filtering of the pickup deteriorates the SNR even further.

4.2 Modified Impulse Response Method

In this method the body of an instrument is excited with an impulse hammer and the air radiation is measured with a microphone. The microphone is placed one meter in front of the guitar as in the previous setups. The strings should be damped so that they are not excited during the actuation. The portion of the strings behind the nut should also be damped. The strings can also be removed, but this changes the tensions in the instrument and changes slightly the resonances of the body. However, these changes are very small compared to the sought effect.

In this work this method has been mainly used for electric guitar pickup signals. Therefore, after measuring a response of an instrument body the lowpass filtering of the magnetic has to be compensated. This explains why the name of the method is suggested to be called the modified impulse response method (MIR). The amount of applied compensation mimics the frequency envelope of the target response obtained with the deconvolution method in Fig. 33. Also, evaluation by informal listening was used. The compensation was done with a high-order finite impulse response (FIR) filter.

This method can also be used for acoustic guitar pickup signals. When used for an acoustic guitar the lowpass filtering effect of a magnetic pickup is not present. Therefore, the compensation for the lowpass effect is very small or unnecessary, for the reasons mentioned in Sec. 4.1.2.

If the instrument is very fragile the MIR method can be difficult to carry out without causing destructive consequences for the instrument. To avoid this impulse hammers with a changeable head or tip, e.g., PCB Piezotronics ICP model 086C1, can be used. However, when using a hammer with a soft tip the resulting response is strongly lowpass filtered and is reliable only up to maybe one or two kHz at the most.

Figure 34 shows responses obtained with the MIR method for a flat-top acoustic guitar. Subplots 34 (a) and (b) represent the measured impulse and magnitude responses. The length of the response is much longer than for the deconvolution method, since the body is excited more forcefully than in the case of playing. Therefore it could be again said that hitting the bridge is not the most natural way of exciting the guitar body or at least does not strictly speaking correspond accurately with the natural excitation process. In spite of this, the measurement gives us important and reliable knowledge of the behaviour of the resonant body. More importantly, for this work, we obtain a response that we can use to simulate the body of an acoustic guitar. The magnitude response in Fig. 34 (b) shows how the lowest resonances are much more well-defined, compared to the deconvolution method responses. The lowpass characteristic, however, is very strong and especially questions the reliability of the high-frequency portion. Subplots 34 (c) and (d) depict the impulse and magnitude responses after the lowpass behaviour of the pickup has been compensated. The compensation is rather approximate, but improves the situation significantly and gives the electric guitar an acoustic-like character.

4.3 Instrument Body Modeling with Reverb Algorithms

The hollow body of an acoustic guitar can be considered as a very small room. Therefore, to model a guitar body a reverb algorithm can be considered as a viable option. Reverb algorithms (Schroeder, 1962), (Moorer, 1979), (Jot and Chaigne, 1991), (Rocchesso and Smith, 1997) simulate reverberation of rooms, computationally in an efficient manner. From a time-domain viewpoint reverb algorithms are typically able to create a sufficient amount of reflections. Similarly in the frequency-domain, the number of resonances is usually sufficient. Also, the frequency dependent decay can be

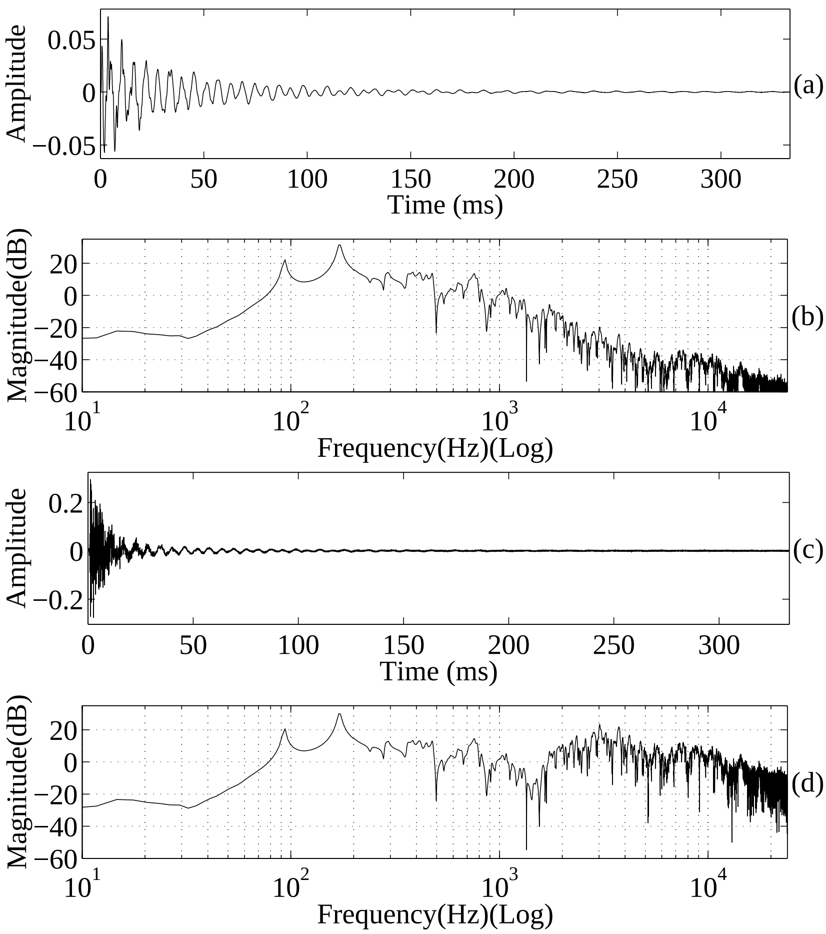

Figure 34: Body-model filter responses used for electric guitars obtained with the MIR method from a classical guitar. Figure shows (a) 330 ms of measured response, (b) magnitude response of it, (c) when response is compensated for lowpass characteristics, and (d) magnitude response of it.

controlled, except in the most primitive reverbs.

To support the assumption of using reverb algorithms to model instrument bodies, Mathews and Kohut (Mathews and Kohut, 1973) came to the conclusion that the exact frequency and gain of the resonators is not important when modeling a violin body. A violin body can be considered as a close 'relative' to a guitar body. Hence, the results of Mathews and Kohut can be interpretedly applied to this study. However, to model an instrument body accurately, reverb algorithms created so far are not controllable enough, since amplitudes and frequencies of single resonances cannot be adjusted accurately enough. This situation can be improved by modeling a few lowest and strongest resonances separately, which will be discussed in the next chapter.

Since the exact frequency and gain of the resonators is not important (Mathews and Kohut, 1973) a reverb algorithm can be used to model the soundbox of an acoustic guitar. In addition, the reverb algorithm multiple feedback networks (MFDN)[8] (Jot

[8]Often also referred simply as FDN.

and Chaigne, 1991), with short delay lines, can be used to produce resonances irregularly spaced over frequency (Rocchesso, 1993). In this study the RV reverb (Väänänen et al., 1997), a special case of the FDN, has been chosen for modeling the guitar body due to its efficiency and its computational requirements.

5 Realization and Experiments

A target response, simulating a guitar body response, can be realized with different filter structures. A filter approximating a target response should model the target response as well as possible. The most important quality of a filter approximating a target response is its sound quality and audible transparency compared to the target response. In addition to considering the audible quality of a body simulation filter its controllability, stability, accuracy, and computational load are important aspects, especially when considering an end user and a real-time implementation. These aspects bring new objectives to light and might reorder the importance of different goals. The filter structures and methods covered in the following have been successfully used in this work.

This chapter discusses different filter structures for realizing instrument body models and touches the subject of sound quality for each filter type at the same time. The results are discussed more thoroughly in Chapter 7, p. 76, which also presents the results of conducted perceptual assessments for the modulation of warped filters.

First, finite impulse response (FIR) filter design is discussed and then infinite impulse response (IIR) filter designed techniques. Two IIR filter design techniques have been used, namely, linear prediction (LP) and Prony's method. IIR filters are recursive structures and enable to model slowly decaying resonances efficiently. Then the discussion moves to frequency warped structures that enable emphasis of different frequency regions. After this the separation of isolated resonances and reverb modeling related issues are discussed. The end of this chapter addresses modifying of responses.

5.1 FIR and IIR Filters

The quality of a body simulation filter was informally assessed by first convolving a dry un-processed signal, $s_{\mathrm{dry}}(n)$, with an equalization filter, $h_{\mathrm{eq}}(n)$, simulating a guitar body, resulting with an equalized signal

$$s_{\mathrm{eq}}(n) = s_{\mathrm{dry}}(n) * h_{\mathrm{eq}}(n), \tag{10}$$

where $*$ represents discrete deconvolution. In the following the target response used to design different filters is the 5000 samples long response shown in Fig. 31 (b). The signal that was filtered, $s_{\mathrm{dry}}(n)$, was either an impulse or guitar playing obtained from an undersaddle pickup or a magnetic pickup. The signal $s_{\mathrm{eq}}(n)$ was compared either to the recorded air radiation, or when comparing filter structures, to the result with the target response. When comparing differences between pure impulses the differences are much more easier to detect than when comparing playing samples that have been convolved with the responses. This is because musical signals with more information enable the human auditory system to mask things in the time and the frequency domain much easier than with pure impulses.

5.1.1 FIR Filter Design

Approximating a target response with an FIR filter is the most straightforward option (Orfanidis, 1996)(p. 125):

$$H(z) = \sum_{n=0}^{N} w(n) h_{\mathrm{eq}}(n) z^{-n}, \tag{11}$$

where N is the order of the filter, n is the discrete time variable, $w(n)$ is the windowing function. Figure 35 shows FIR body filter responses of orders 5000, 4000, 2000, 1000, 500, and 300. The window fades in and out with a half-Hanning function of 100 samples. It what happens when the order of the filter decreases: at low-frequencies the bandwidth of the two lowest body modes is increased and finally disappears and in the same manner for resonances at high frequencies the mode density decreases.

Perceptually FIR equalizer lengths of 1000 to 5000 taps produce good results. Too long filters produce additional reverberation, since the response is so noisy. With filters shorter than 1000 taps the filtered signal starts to sound more and more dry and electric. When the length is below 500 taps both low and high frequencies degrade considerably: low frequencies decay too fast and high frequencies sound too sharp with a prominent attack.

It should be noted that there are differences in the target responses depending both on the size of the guitar body and the method the target response has been obtained. This makes general comparisons between different filter orders for different filter design techniques relative.

5.1.2 All-Pole Filter Design: Linear Prediction

Linear prediction (LP) is an IIR filter design technique often used in speech applications (Markel and Gray, 1976; Moorer, 1985). To obtain an all-pole model of order M, linear prediction is applied to a target response, resulting in the following transfer function

$$H_{\mathrm{LP}}(z) = \frac{b_0}{A(z)} = \frac{b_0}{1 + \sum_{k=0}^{M} a_k z^{-k}}, \tag{12}$$

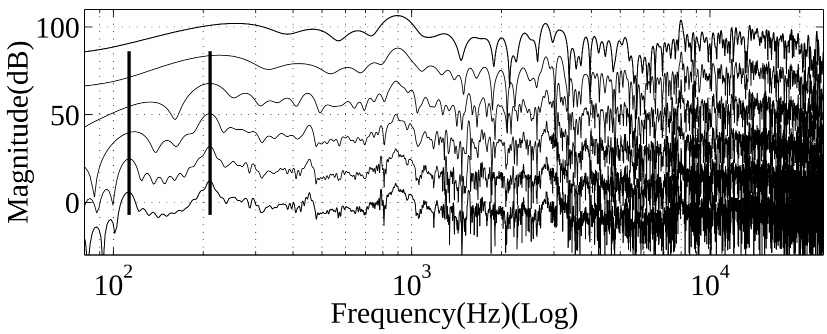

Figure 35: Magnitude responses of FIR body model filters of orders 5000, 4000, 2000, 1000, 500, and 300 stacked on each other, so that highest order at bottom and lowest order on top. Two lowest body resonances are indicated with vertical lines.

where a_k are denominator polynomial coefficients and b_0 is a gain term. Since the target response has a large number of resonances a high-order model is expected to be needed. Also, an LP model results in a minimum-phase response, i.e., temporal spreading is minimized and poles are inside the unit circle. The difference between a target and its minimum-phase response is again easier to notice when listening to pure impulse responses, rather than guitar playing filtered with the responses.

Figure 36 shows (a) the impulse of an LP-model of order 1000 and (b) the magnitude responses of the target response (bottom) as well as LP-models of order 1000 (middle) and 500 (top). The minimum-phase property appears as the missing build-up part and the impulse-like beginning in the time-domain. In the frequency domain the 1000 order LP-model works well at low-frequencies, but misses some of the details at high-frequencies. As for the 500 order LP-model the second body mode is modeled better than the first one, which can be understood since the second mode is stronger. The behaviour of high frequencies in the 500 order LP-model has also degraded, compared to the 1000 order model.

Informal listening was conducted with LP models of orders 1000, 500, and 300, so that dry guitar playing s_{dry} filtered with $H_{\mathrm{LP}}(z)$ was compared to the target response filtering result. None of them accomplished transparency with the target response, but the 1000 and 500 order filters were able to preserve some of the important characteristics. However, the minimum-phase property degrades temporal spreading of the simulation filter compared to a non-minimum-phase target response.

5.1.3 Pole-Zero Filter Design: Prony's Method

Prony's method is a pole-zero modeling technique that can realize non-minimum-phase responses. The transfer function of a pole-zero filter can be presented as (Orfanidis,

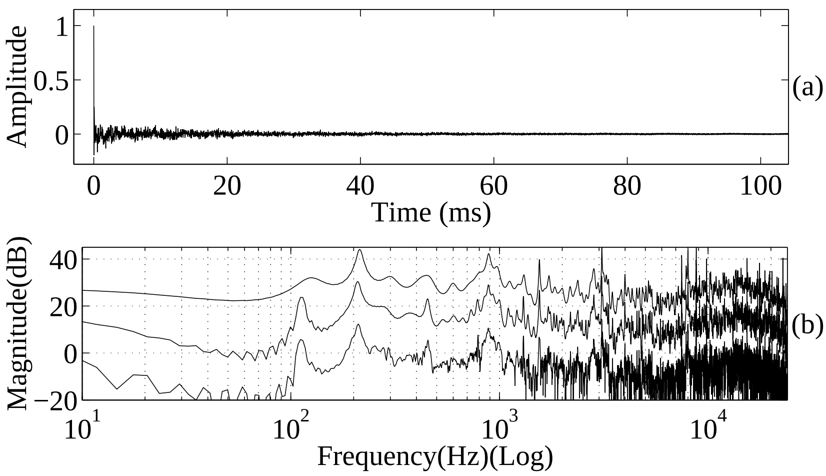

Figure 36: Body-model IIR-filters designed with linear prediction. (a) impulse response for a 1000 order filter and (b) magnitude responses for (top) 500 and (middle) 1000 order filters and (bottom) the target response.

1996)(p. 226):

$$H_{\mathrm{PZ}}(z) = \frac{B(z)}{A(z)} = \frac{\sum_{l=0}^{N} b_l z^{-l}}{1 + \sum_{k=0}^{M} a_k z^{-k}}. \tag{13}$$

where N and M define the orders of the numerator and denominator, respectively. The zeros, $B(z)$, are able to model the build-up of a response accurately as it replicates the beginning of a target response. Whereas the poles, roots of $A(z)$, model the slowly exponentially decaying resonances.

Figures 37 (a) and (b) show the impulse and the magnitude response, above its target response[9], for a Prony's model for orders $N = 250$ and $M = 250$. Now the build-up agrees better with the target response, but the tail dies out too fast. As for the frequency domain the resonances are modeled rather inaccurately and quite inconsistently: for example the lowest mode is ignored while the second body mode is modeled to some degree and resonances from 1 kHz to 10 kHz are modeled worse than with an LP-model. This is, however, rather natural since the order of the example LP-model was higher. Figure 37 (c) displays the magnitude response when the model orders are $N = 100$ and $M = 500$. Now the lowest mode is modeled better, but still the lowest body mode is not modeled accurately. A problem in comparing these to other filter design responses is that, as mentioned, the target response is shorter. In practice the use of a shorter target response decreases the amplitude and widens the resonance in the frequency domain. Hence, the filter design algorithm concentrates on the more prominent resonances and comparison is not that straightforward.

Perceptually the zeros slightly improve the attack part compared to all-pole filters, but the decaying tail, i.e., the low-frequencies lack some accuracy. The response in Fig. 37

[9]For these Prony's models the target response was the same as for the other cases but only a 2000 tap version was used, since MATLAB gave out off memory notifications for longer responses.

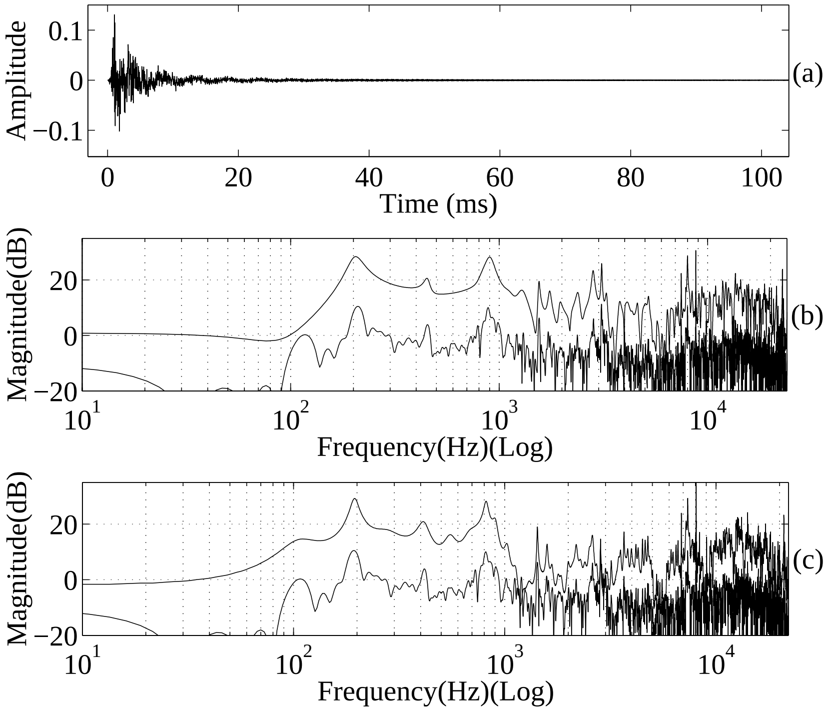

Figure 37: Instrument body model IIR filters designed with Prony's method. (a) impulse response and (b) magnitude response when N and $M = 250$, and (c) magnitude response when $N = 100$ and $M = 500$.

(c) gave a better result than the one in Fig. 37 (b). So increasing the order of $B(z)$ does not necessarily improve the audible response. Meaning that both low and high frequencies should be modeled, but only as far it is necessary.

On the whole, the design methods for IIR filters tested here were not convincing compared to simple truncation of an FIR response, since the order of the IIR filters was high compared to the obtained results. Frequency warping and frequency warped filters will be discussed next which changes the situation radically and makes the use of IIR filters a more plausible option.

5.2 Frequency-warped Filters

Frequency warped digital filters and their audio applications is a vast research field. Frequency-warped filters are shortly discussed in this section. For the reader to gain a deeper understanding of other warped applications the following citations are recommended (Steiglitz, 1980; Strube, 1980; Karjalainen et al., 1996; Karjalainen et al., 1997; Härmä et al., 2000).

The basic idea of warping is best illustrated using the filter structure in Figure 38 (a). When the unit delays of an ordinary digital filter are replaced with allpass sections, the resulting filter is called a warped filter. In this work first-order allpass sections were used. A filter can be designed on a warped frequency scale based on the bilinear conformal mapping

$$\tilde{z}^{-1} = D_1(z) = \frac{z^{-1} - \lambda}{1 - \lambda z^{-1}}, \tag{14}$$

where λ is a warping parameter and $D_1(z)$ is a warped delay element. The group delay of $D_1(z)$ is frequency-dependent so that positive values of λ yield increased resolution at low frequencies. Correspondingly, a negative value of λ produces a system with an improved frequency resolution at high frequencies. This is illustrated in Figure 39 that shows the warping by a first-order allpass section as a function of frequency. The value of the warping parameter should be $|\lambda| < 1$ to make the allpass filter stable.

Design and implementation of warped transversal (WFIR) structures is straightforward. However, the implementation of warped recursive filters (WIIR) is problematic because there are delay-free loops in the filter. Implementation techniques for warped recursive direct-form and lattice filters have been introduced in (Härmä, 2000). There are two approaches: (I) direct implementation of the filters using a specific two-step algorithm, or (II) elimination of the delay-free loops by modification of the filter structure. A modified structure is shown in Figure 38 (b). The new coefficients σ_k of the filter can be computed using an algorithm presented, e.g., in (Karjalainen et al., 1997).

As noted above, the value of the warping parameter λ controls the amount of warping that is desired. From the point of view of auditory perception a specific value of λ yields a good approximation of the Bark scale (Zwicker and Fastl, 1990) which is traditionally used as a psychoacoustical pitch scale. A formula to compute this value as a function of the sampling rate is given by Smith and Abel in (Smith and Abel, 1999). At the sampling rate of 48 kHz this yields $\lambda \approx 0.76$.

One more favorable property of warped filters is their inherent robustness in precision requirements, based on the use of allpass subsections. Particularly, when the density of poles and/or zeros in the z-domain—especially corresponding to low frequencies—is high, traditional filter structures such as direct form IIR filters become very problematic. Due to the bilinear warping (rotation) of poles and zeros in the z-domain, the pole and zero densities are relaxed considerably. Typically, direct form IIR filters higher in order than about 20–25 cannot be implemented even when using floating-point processors. Corresponding warped filters remain stable and realizable even with orders higher than 100 and fixed-point computation (Asavathiratham et al., 1999).

As can be noticed when comparing the warped IIR filter in Figure 38 (b) with traditional IIR filter structures of the same order, the warped structure is more complex and thus computationally more expensive. This is often compensated, however, since considerable reduction in filter order is possible, due to good match to human auditory frequency scale properties.

The design of warped filter models for a target response can be done in a straightforward

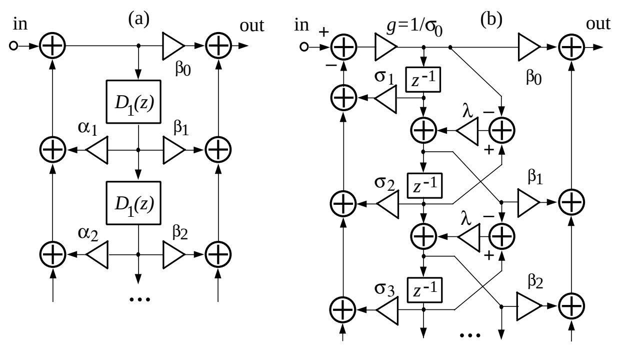

Figure 38: (a) The principle of warped IIR filter and (b) the implementation used in this study.

way as follows. The impulse response $h_{\mathrm{eq}}(n)$ to be implemented is first mapped to a warped time domain response $\tilde{h}(i)$ using the inverse mapping of Eq. (14) as described in (Karjalainen et al., 1997). This warped impulse response can then, in principle, be used in any filter design technique to yield an FIR or IIR structure, which has to be implemented as a corresponding warped structure. Furthermore, to create a warped model, the warping coefficient, λ, used in the warped structure is the same as the one used at the design stage of the filter. The warping toolbox for MATLAB (Härmä and Karjalainen, 2000) was used for the warped designs.

Warped Responses Figure 40 shows a time and a magnitude response of a warped body model, above its target response. The filter is designed using linear prediction with the order of 200, when $\lambda = 0.766$. At the design stage, the warped model is constructed of a minimum-phase response which represents the target response. This naturally ruins the build-up of the response, but without this the high-frequency components will get excessively long and modeling of them is impractical. The time response in Fig. 40 (a) reveals the minimum-phase property of the model. The magnitude response in Fig. 40 (b) shows distinctly how the modeling of the lowest resonances has been improved due to warping, when compared to LP and Prony's methods (responses shown in Figures 36 and 37, respectively). When λ is positive the frequency resolution gets sparse for high frequencies. Therefore, at high frequencies the warped model captures mostly the spectral envelope rather than specific resonances. The improved low-frequency modeling can be heard in filtered examples, but at the same time the lack of accurate high-frequency modeling is also notable. However, since the frequency resolution of the human auditory system can be approximated with warped structures it highly motivates the use of warped filters.

Cascading warped structures improves the frequency domain coverage of the modeling filter. First, modeling the target response with a Bark-scale approximation valued λ covers the low-frequencies. Then its residual response can be designed with a smaller λ (but $\leq$ -1) which models the high-frequencies. A more general approach is the usage of warped Kautz filters when combining filters to cover the whole frequency range as was done in (Penttinen et al., 2001b).

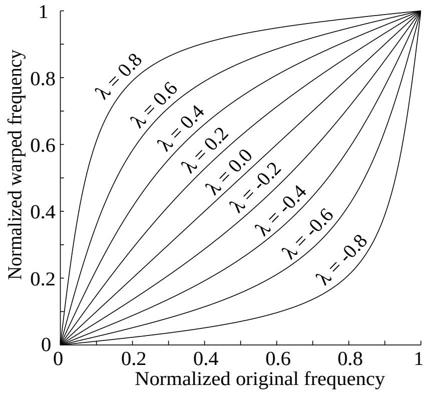

Figure 39: Frequency warping characteristics of the first-order allpass section for different values of the warping parameter λ. Frequencies are normalized to the Nyquist rate.

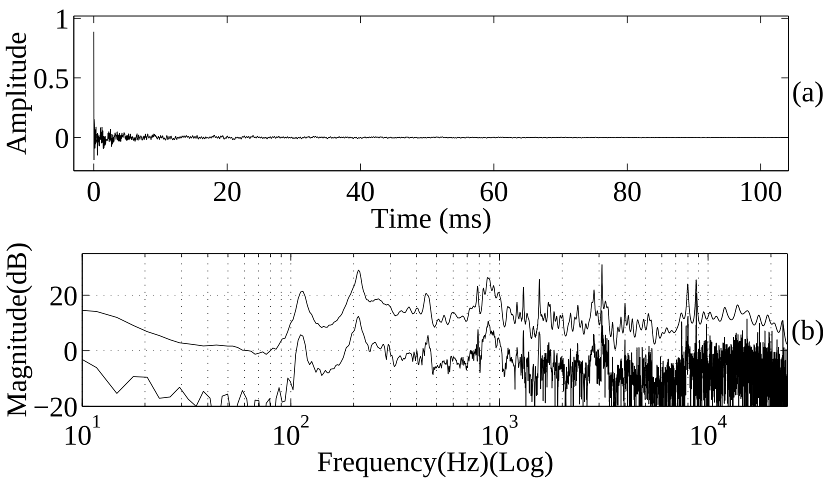

Figure 40: Body-model WIIR-filter designed with linear prediction. (a) impulse response and (b) magnitude response. Order of all-pole filter is 200, $\lambda = 0.766$ and $f_s = 48$ kHz.

5.3 Modulation of Warped Filters

In this section the warping parameter, λ, is set to be a free parameter which can be adjusted freely in real-time. This produces a non-uniform spectral modification where all resonances and anti-resonances are shifted non-uniformly up or down in the frequency domain.

The warped implementation structure will resolve in a low or high-boost filtering effect, which is dependent on the value of λ. This spectral tilt effect can be compensated by pre-filtering the input signal before the body model filter, with the following filter

$$H_c(z, \lambda_m) = \frac{\sqrt{1-{\lambda_m}^2}}{\sqrt{1-{\lambda_o}^2}} \frac{1-{\lambda_o}^2 z^{-1}}{1-{\lambda_m}^2 z^{-1}}, \tag{15}$$

where λ_o is the original warping parameter that was used to warp the minimum phase impulse response, and λ_m is the modified value of λ that will shift the resonances.

The value of the warping parameter, λ, can be altered in a pre-determined way or for example with a control pedal. This way an interesting audio effect is attained that can, at a principle level, be used at least in two ways: (I) One can maintain a particular λ value for a musically meaningful period (e.g., a riff or two, or a whole song) and then change it to another value. This way the impression of different sized guitars with the same instrument can be achieved. (II) By continuously adjusting the value of λ, a steadily changing timber will be observed. In this case identifying various sized soundboxes is almost impossible, but adds a pleasant and useful effect to the 'effect-toolbox' of guitarists. As mentioned before the spectral tilt effect caused by warping has to be reversed with Eq. 15. This in mind the body-modulation effect can be expressed in the z-domain as

$$\tilde{x}(z) = H_c(z, \lambda_m)\frac{1}{A(z)}x(z) \tag{16}$$

where $x(z)$ is the input signal, $H_c(z, \lambda_m)$ is the compensation filter, $A(z)$ is the body-model, and $\tilde{x}(z)$ is the output that has been transformed to something novel by means of warping. In real-time implementation, the mapping of the body-model coefficients (α_k to σ_k) can be performed for every sample or spread over a few samples. In most trivial warped FIR form the discussed technique is also analogous to the traditional digital phaser effect (Smith, 1982). The highly flexible usage of λ also perceptually resembles the phaser effect.

Figure 41 displays a set of magnitude responses with different λ values from 0.65 to 0.81 in steps of 0.01. The magnitude responses are stacked, one above the other, so that the λ value that corresponds to a spectrum is displayed on the y-axis. Figure 41 illustrates how the resonances shift when the value of λ is altered. The initial non-warped magnitude response is displayed in the middle of Fig. 41, with $\lambda = 0.73$. All the warped magnitude responses, i.e., $\lambda \neq 0.73$, are compensated with $H_c(z)$.

Figure 42 illustrates the highly flexible behavior of a body-modeling filter, when the warping coefficient, λ, is modified beyond a recognizable guitar body. Figure 42 shows magnitude responses of a body simulation filter as a function of the warping coefficient λ with steps of 0.01 from -0.756 to 0.756. Dark colors in Fig. 42 represent strong resonances of the body simulation filter.

5.3.1 Perceptual Assessments

Lansky and Steiglitz proposed that by warping an LP-model, used for synthesizing a digitized violin piece, different instruments of a violin family can be created (Lansky and Steiglitz, 1981). The LP models they used were derived from digitized violin playing. As the excitation for a single LP-model they used a roughly triangular waveform, with a

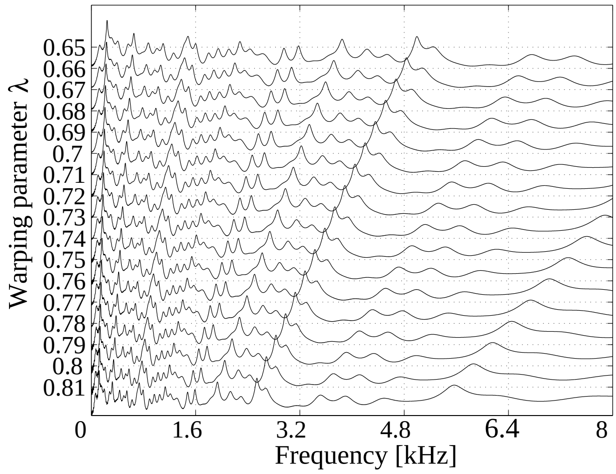

Figure 41: Magnitude responses of a body simulation filter with different warping parameter, λ, values.

certain frequency, and warped the LP-model in a way that corresponds to the tuning of the targeted instrument. In this work only the body simulation filter is altered and the original excitation signal is left untouched. The aim is to create an effect, to be applied to the signal from a guitar pickup, that will change the size of the perceived soundbox. As one part of this study, the objective was to verify the assumption of Lansky and Steiglitz in the case of a carefully designed acoustic guitar body model which is driven by a guitar pickup signal, through listening tests and spectral interpretation.

5.3.2 Description of Listening Test

In the conducted listening test the task of the testee was to size two perceived soundboxes as closely to one another as possible, by adjusting the warping parameter, λ.

A reference signal and a test signal were played, where the latter was to be matched with the previous one. The pair of signals were filtered with different body filters, derived from small and large sized guitars[10]. When the reference signal was processed with a large body-model the test signal was filtered with a small soundbox filter, and vice versa. In the test, the pair of signals was played in a continuous loop, with a short break between each sound sample. When the subject was satisfied with the matching of one pair, the testee proceeded to the next one. The warping parameter could be adjusted within steps of 0.001, from 0.53 to 0.93. The control of λ and the transition to the next sound sample pair was implemented through a graphical interface. The listening test was carried out by seven musically oriented subjects with normal hearing, in a standard listening room by using headphones.

[10]Target responses for the small and large sized body model filters are shown in Figs. 31, and 57, respectively.

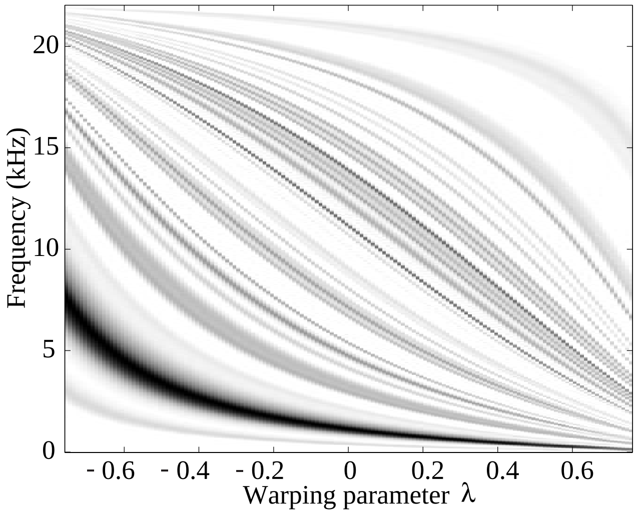

Figure 42: Magnitude responses of a body simulation filter as a function of the warping parameter λ. The tone values indicate the magnitude of the guitar body resonances. Darkest colors represent the strongest resonances.

Guitar playing, a short impulse train, and a burst of white noise were used as the excitation signals. Furthermore, there were two kinds of melodic guitar samples: one consisted of the instrument excited by picking and the other one was excited by strumming. All samples were filtered with WIIR filters of order 100. During the try out of the listening test it was noticed that altering λ shifted the resonances in a desired way. But as a side-effect, a distinct timbre emphasis was perceived, which altered with the adjustment of λ. This was considered subtly disturbing and viewed as a characteristic that might complicate the matching task. To diminish the timbre emphasis the resonances were broadened in the frequency-domain, i.e., shortened in the time-domain. This was done by smoothing the LP body-model coefficient sets (both large and small body filters) in the following way

$$\tilde{\alpha}_k = \alpha_k a^k; k = [0, 1, 2, ...m], \tag{17}$$

where k indicates the coefficient, and m is the order of the LP filter. The new $\tilde{\alpha}_k$ coefficients are mapped to $\tilde{\sigma}_k$ coefficients in the same manner as mentioned before (Karjalainen et al., 1997). The parameter a was set to be 0.98, which results in an exponential decay of a^k. The smoothened filters (both large and small) were applied to the guitar samples. Moreover, each sound example was normalized in respect to their energy, with the inverse of the squared sum of all samples contained in one sound example. The duration of the guitar playing samples was two seconds, and the white noise and the impulse train, with three impulses, lasted a second each. In overall, there were twelve sample pairs to be matched.

The results of the conducted listening tests are presented in the Results section in Chapter 7.

5.4 Modifying Responses

Next the discussion shifts to the subject of manipulating. The manipulation occurs off-line before the response is used for filtering. The most important characteristics of the modeling filter will be exactly or approximately preserved when implemented with alternative filter structures. The objective is to obtain more control over the resulting timbre than with, e.g, an inflexible FIR structure. The themes discussed here should be distinguished from the dynamic modifying done with the modulation of λ in the warped structures discussed in Sec. 5.3. First, the discussion concentrates on the manipulation of low-frequency resonances and then on what can be done for the high-frequency region.

5.4.1 Extraction and Replacement of Body Resonances

The two prominent low-frequency resonances are characteristic and typical for guitar body responses. They exist in every normal acoustic guitar and, as mentioned previously, their frequencies are size dependent. These resonances decay slowly compared to higher frequency resonances. Therefore, it is worthwhile to extract the resonances and synthesize them with recursive filters, e.g., second-order IIR filters (Karjalainen and Smith, 1996). This resolves in a model with n resonators $H_{\mathrm{reson_n}}(z)$ (here $n = 2$) and a filter for the remaining part $H_{\mathrm{main}}(z)$. After the slowly decaying modes have been removed the remaining $H_{\mathrm{main}}(z)$ response can be truncated. The truncation resolves in a computationally less expensive implementation, especially when it is implemented as an FIR filter. In addition, more control over the timbre of the response will be obtained, since parametric IIR filters are used for synthesizing the resonances.

A resonance can be removed and/or synthesized efficiently with a second-order IIR filter (Karjalainen and Smith, 1996). The transfer function of a second-order IIR filter is typically expressed as (Orfanidis, 1996) (p.583-601):

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}, \tag{18}$$

where b_n and a_n are coefficients that determine the response of the filter. The values of the polynomial coefficients b_n and a_n which affect the resulting zeros and poles are different depending on the implementation structure, i.e., if the resonator is implemented in cascade or parallel with the rest of the model. Figure 43 shows the block diagrams when the main part of the body model and discrete resonances are implemented (a) in cascade or series, i.e., as a product of partial transfer functions, and (b) in parallel, i.e., when implemented as a sum of partial transfer functions. With a cascade implementation the response should be unity everywhere else than in the vicinity of the resonance. In contrast, in a parallel structure the response should affect the remaining response as little as possible, meaning that the attenuation at other frequencies should be as large as possible or at least sufficient enough.

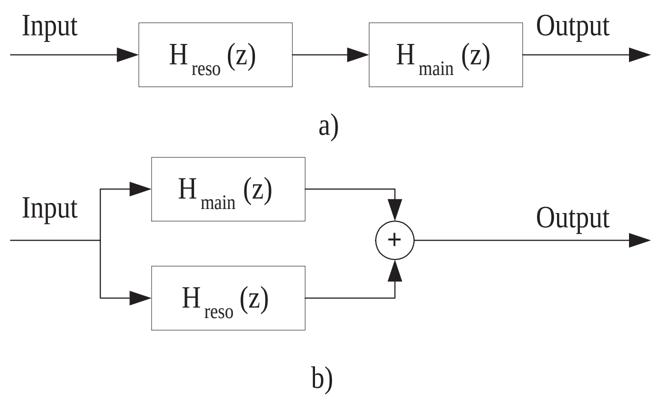

Figure 43: Block diagrams of separate resonator and simulation filter solutions: (a) cascade and (b) parallel.

A resonance is removed from a target response by notch filtering. A notch filter attenuates a narrow frequency-band by a certain amount. In this case the notch filter could be understood as a mirror-image, on the amplitude axis, of the resonance under interest. An estimation of the frequency, amplitude, and bandwidth for the notch filter is attained from the corresponding parameters of the resonance. These parameters can be obtained by using a pitch estimation (Rabiner, 1977) and a short time Fourier transform (STFT) based method discussed in, e.g., (Tolonen et al., 1998) and in more detail in (McAulay and Quatieri, 1986; Smith and Serra, 1987), where a recorded signal is decomposed into sinusoidal components. Here the method is used to define the bandwidth or decay time of a body mode. Fortunately amplitude modulation of resonances, which would make the analysis and synthesis of them more difficult, does not occur for the lowest body modes.

A parametric second-order equalizer can implement a notch filter and a resonance filter that are counterparts for each other. The general transfer function is given as (Orfanidis, 1996) (p.592-595):

$$H_{\text{paramreso}}(z) = \frac{(\frac{G_0+G\beta}{1+\beta}) - 2(\frac{G_0 \cos\omega_0}{1+\beta})z^{-1} + (\frac{G_0-G\beta}{1+\beta})z^{-2}}{1 - 2(\frac{\cos\omega_0}{1+\beta})z^{-1} + (\frac{1-\beta}{1+\beta})z^{-2}}, \tag{19}$$

where ω_0 is the center frequency in radians[11], and

$$\beta = \sqrt{\frac{G_{\text{B}}^2 - G_0^2}{G^2 - G_{\text{B}}^2}} \tan(\frac{\Delta\omega}{2}) \tag{20}$$

where $\Delta\omega$ is the bandwidth of the filter, G defines the gain (positive or negative) of the resonance, G_{B} is the gain at the bandwidth which is typically $\pm$ 3-dB, and G_0 is the so called reference gain where the response is flat. For a resonance filter $G_0^2 < G_{\text{B}}^2 < G^2$ and for a notch filter $G^2 < G_{\text{B}}^2 < G_0^2$. Figure 44 displays the magnitude responses

[11] Center frequency in Hertz: $f_0 = \omega_0 f_s / 2\pi$, where f_s is the sampling frequency.

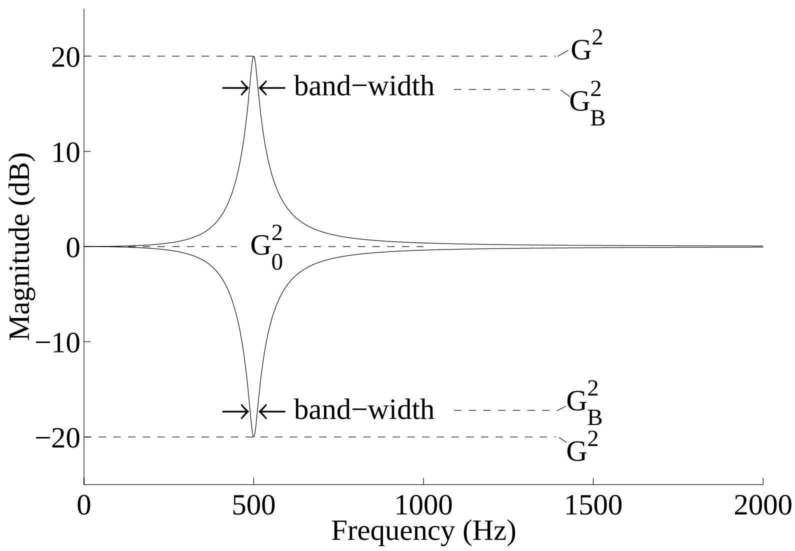

Figure 44: Parametric resonance filter magnitude responses for resonator and notch filter.

for a resonance and notch filter when for the resonator $G = 20$ dB, $G_{\mathrm{B}} = 17$ dB, $G_0 = 0$ dB, $f = 500$ Hz, and $\Delta f = 22.9$ Hz. For the notch filter the gains are reversed and G_0s are equal. See (Orfanidis, 1996) for a more thorough discussion.

In a cascaded structure the notch filter and resonance filter are complementary and give $H_{\mathrm{notch}} H_{\mathrm{reso}} = 1$. In other words a cascaded solution gives prefect reconstruction within the used numeric precision.

In a parallel case the body mode is removed with the same notch filter as for the cascade solution, but for synthesis a different filter has to be used. This is because, other frequencies than the resonance frequency have to be affected as little as possible. The transfer function of the used parallel resonator is (Smith and Angell, 1982):

$$H_{\mathrm{reso,par}}(z) = g_{\mathrm{par}} \frac{1 - z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}, \tag{21}$$

where $a_1 = -2\cos(\omega_0)$ and $a_2 = r^2$, $g_{\mathrm{par}} = 1$ - r, which controls the gain of the resonator with r being the pole radius and ω_0 being the center frequency in radians. The starting moment of the parallel filter should be set by introducing a delay to the resonator path, possibly using fractional delay filters (Laakso et al., 1996). The starting moment should be synchronized with the moment of maximum amplitude of the removed resonance. Defining the moment of maximum amplitude is not a big problem. However, there is typically a build-up before the maximum amplitude, and the build-up-time varies from a resonance to another. Figure 45 exemplifies the build-up of the second-lowest body resonance ($\sim$ 200 Hz) of a classical guitar. The figure displays the time response of the extracted resonance. In the case of a resonance with a long build-up-time the parallel synthesis structure can have a lot of artifacts due to phase mismatches. As the n parallel resonators are summed with the main filter there will be a phase mismatch compared to the target response. Therefore prefect reconstruction is difficult to obtain. However, when considering real-time implementation the parallel case has some advantages over the cascade one: (I) The number of needed instructions is smaller since the second numerator term $b_1 = 0$, i.e., one addition and multiplication can be saved. However, it is not self-evident that this property can always be harnessed to it full extent, when implemented on a DSP-processor. This is because of DSP processor architecture features and their parallel instructions and memory fetches capabilities. (II) When changing the frequency or gain of the resonance, the parallel implementation is more efficient since it needs a smaller number of coefficients to be changed and also needs less memory than a cascade implementation.

Figure 46 shows low-frequency magnitude responses for different implementational schemes to synthesize removed resonances. Fig. 46 (a) shows the target response which is the acoustic response of a classical guitar measured with an impulse hammer and the response with a dashed line after the two lowest resonances (100 and 200 Hz) have been removed. In the remaining panes from Fig. 46 (b) to (d) target response is displayed with a dashed line. Fig. 46 (b) depicts the target response and the result for a cascaded scheme when two lowest resonances have been implemented separately. No differences between the magnitude responses can be seen, since the notch and resonance filters are complementary with each other. In Fig. 46 (c) only the first resonance has been synthesized with a parallel filter. Some very slight differences can be noted at frequencies below 100 Hz and at the notch at circa 400 Hz. Fig. 46 (d) displays the result when both low-frequency resonances have been synthesized with two parallel filters. Now the differences are more notable: again at very low-frequencies the responses differ from each other and for example resonances at circa 240 and 300 Hz have been cancelled. These differences arise from the phase mismatches mentioned earlier. The build-up of the second removed resonance is quite slow and it is difficult or even impossible correctly to set the length of the delay associated with the second parallel resonator. In addition, as the number of parallel resonators increases the phase problems accumulate and affect the behaviour of the response in a vague non-explicit way. However, as long as the number of parallel resonators is small (e.g. 1 to 2) perceptual transparency can be obtained. This can be explained through the behaviour of the human auditory system, since humans are not that sensitive in noticing a few notches in a complex spectrum or small enough phase mismatches. However, further studies should be conducted if one would want to express something more definite. Moreover, the phase mismatch problems with parallel resonators are case-specific, since the characteristics of target responses can differ a lot.

5.4.2 Manipulating High-Frequencies

Next the experiments done on the high-frequency content of body model filters will be described. The discussion will concentrate on the attenuation of strong high-frequency components and then briefly describe the replacement of high frequencies with white noise.

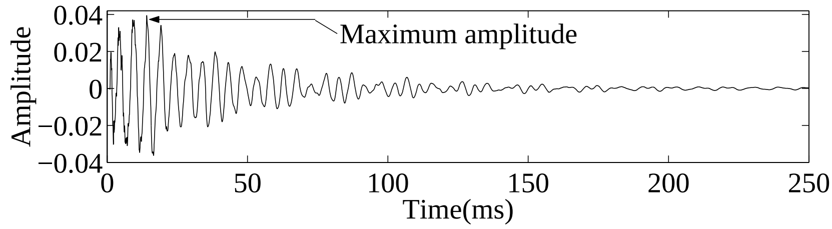

Figure 45: Time response of extracted resonance, illustrating build-up that can cause problems in parallel implementations.

Attenuating Strong High-Frequency Components The body models created with the deconvolution method described in Section 4 have strong high-frequency components: These frequency components are not actual body resonances as described in Sec. 4.1.3. A validation for this claim is that the body radiation responses obtained by hitting the bridge with an impulse hammer (e.g., the MIR method) do not contain these kinds of strong high-frequency components.

Figure 47 shows an example where strong high-frequency components above 2 kHz have been attenuated by 5 to 15 dB, depending on their initial amplitude. Fig. 47 (a) shows the original magnitude response of a steel-strung acoustic guitar body model filter obtained with the deconvolution method. Fig. 47 (b) displays the magnitude response after the high-frequency components have been attenuated. The overall level at high-frequencies has also been attenuated somewhat (circa 3dB to 5 dB). This is because the notch filters used to attenuate the resonances are not perfect, in the sense that the magnitude response is not immediately 0 dB just outside the notch as can be seen in Fig. 44. The audible difference before and after the attenuation of high-frequency peaks is notable. The high-frequencies are attenuated which gives the filtering result a 'darker' timbre. This could be compensated by equalizing the high frequencies to resemble closer the magnitude envelope of the original response. However, even if the high-frequency components have been attenuated the difference between a dry non-equalized bridge pickup signal compared to either filtered result (before or after component attenuation) is larger when these body model filters are compared.

In retrospect these kinds of responses should have possibly been used in the listening tests conducted for the altering of the perceived size of a guitar body by changing the warping parameter λ (Sec. 5.3). This is because some of the strong mid- to high-frequency components distracted and confused the perception of the size. This occurred especially when the perceived size of the body model was increased, i.e., the lowest body modes were shifted to lower frequencies. At the same time the components, higher in frequency, also moved lower which again at certain point start to dominate over the lowest modes. This is why the perception of the size, according to some comments from the testees, could some times be rather vague.

Replacing High Frequencies with White Noise The high-frequency portion of an acoustic guitar body air radiation response is comprised of dense amount of reso-

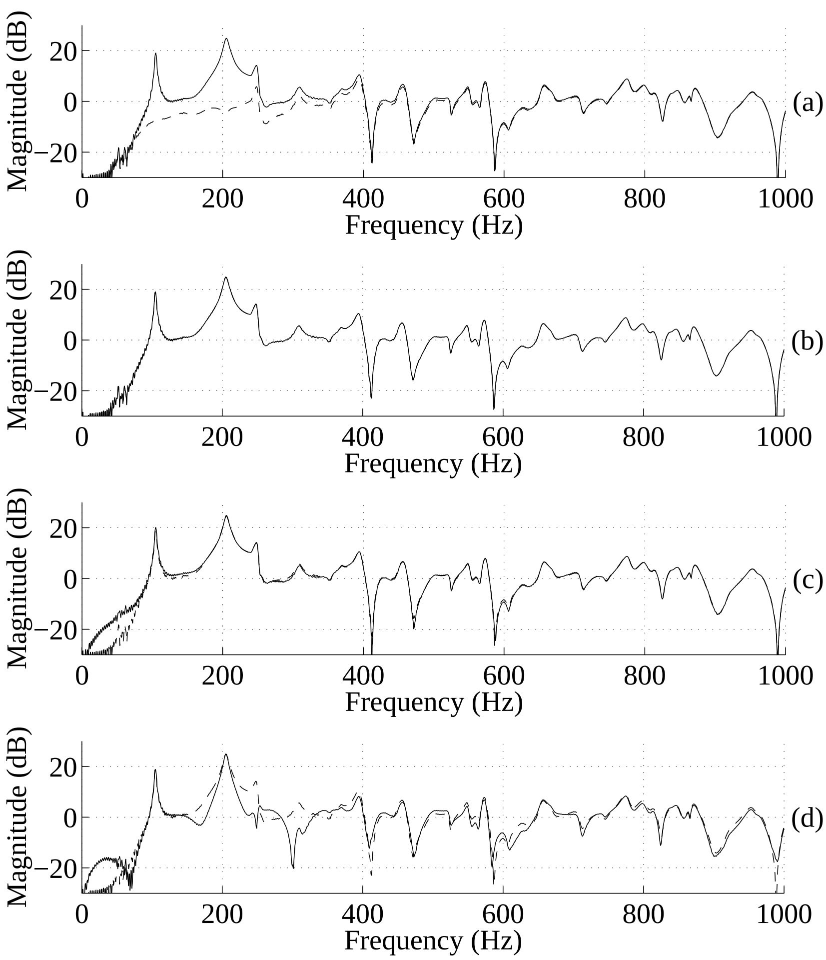

Figure 46: Magnitude responses when body modes replaced with resonators after notch filtering target response. (a) (solid line) Target response and (dashed line) two lowest resonances removed. (b) Two lowest resonances synthesized with two cascaded filters. (c) Lowest resonance synthesized with one parallel filter. (d) Two lowest resonances synthesized with two parallel filters. In panes from (b) to (d) target response is displayed with dashed line.

nances and resembles exponentially decaying noise. Therefore it was also experimented with replacing the high frequencies with exponentially decaying Gaussian white noise. When frequencies above 2 kHz are replaced with white noise the audible differences are quite small, especially when comparing with a response where the strong high-frequency components have been attenuated. Even with a cross-over frequency as low as 500 Hz the most important characteristics are preserved.

5.5 Reverberation Algorithms

Digital filters that efficiently model the reverberant behaviour of a room have been studied and developed since the 1960's (Schroeder, 1962). The Schroeder's (1962)

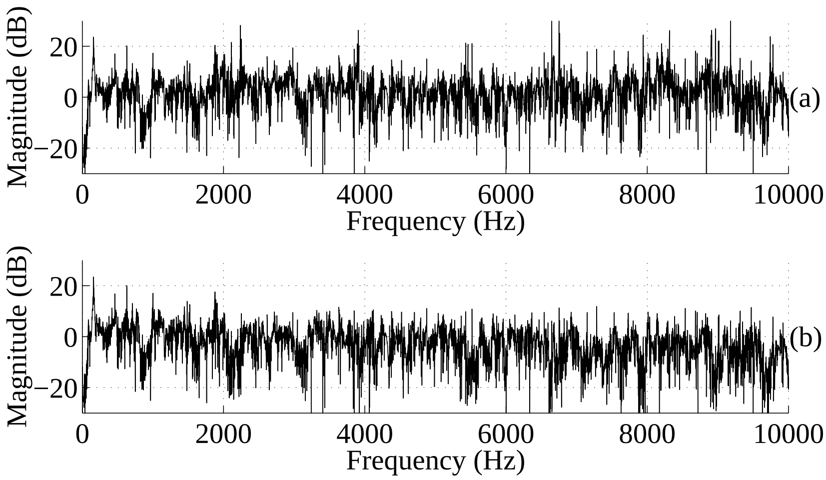

Figure 47: Magnitude responses (a) before and (b) after high-frequency components above 2 kHz have been attenuated.

reverb algorithm is quite clumsy in today's standards[12], it, e.g., lacks control over the frequency dependent decay. Moorer (Moorer, 1979) improved this situation by adding a low-pass filter to each feedback loop of the comb filters of Schroeder's reverb and increased the number of comb filters. Multiple feedback delay networks (MFDN) (Jot and Chaigne, 1991) are networks where a number of (N) delay lines are fed back through a (NxN) feedback matrix. The frequency dependent decay can be controlled in MFDNs with the lowpass filters embedded to the delay lines. MFDNs are capable of producing a denser amount of reflections than the Schroeder's or the Moorer's reverb.

The reverbs mentioned above are based on recursive digital filters and mainly model the late reverberation of a room. For discussion on different kinds of reverbs see, e.g., (Gardner, 1995) which discusses an efficient convolution method without an input-output delay, (Schoenle et al., 1993) proposes a multi-rate scheme for implementing room impulse responses, (Smith and Rocchesso, 1994) treat the issue on the connections between feedback delay networks and waveguide network reverberators, (Dattorro, 1997), and (Gardner, 1998) are also good sources for information on reverbs.

In this study a simplified version of the MFDN is used (Väänänen et al., 1997), where the feedback matrix is replaced with a single feedback coefficient. Figure 48 displays the block diagram of this reverb algorithm. The H_n blocks represent the loss filters and the A_n blocks represent the comb allpass filters. The single feedback coefficient makes the algorithm more efficient compared to an MFDN.

5.6 Basic Body Modeling with Reverb Algorithms

Normally one of the objectives of a reverb algorithm is to produce a non-colored response. In contrast, for instrument body models the objective is almost the opposite:

[12] Even if the Schroeder's reverb is a marvelous invention.

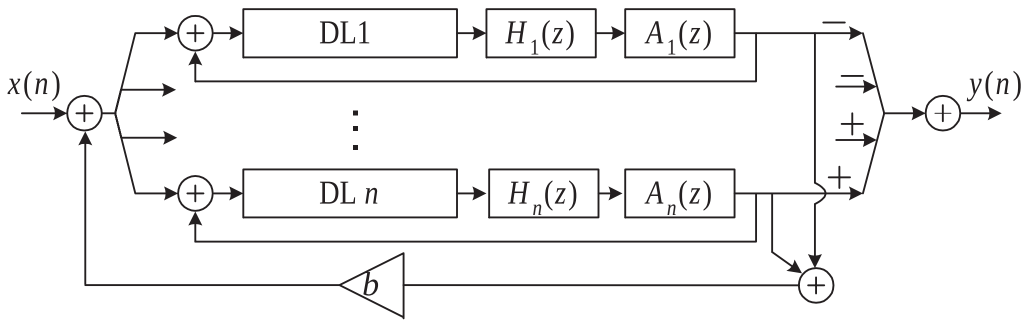

Figure 48: Efficient reverb.

low frequencies have sparse resonances with strong amplitudes and high frequencies are very reverberant like. Colored responses with reverb algorithms can be achieved by using short delay lines (Rocchesso, 1993).

The parameters for a reverb algorithm approximating the behaviour of a guitar body can be chosen as discussed in (Jot and Chaigne, 1991). The lengths of the delay lines can be chosen on the basis of the physical dimensions of the instrument. Since the lengths of the delay lines are short, the response is colored in the frequency domain and in the time domain the multiples of the delay line lengths are superimposed. To efficiently avoid superimposing, the lengths of the delay lines should be primes (Gardner, 1998). Moreover, very small changes in the delay line lengths can change the response considerably. As we remember, e.g., from Fig. 14 the high frequencies of a guitar body response decay much faster than the strong low-frequency body modes. Therefore, an appropriate frequency dependent decay behaviour should be considered. The parameter values for the loss/lowpass filters connected to the delay lines in series should be estimated properly. For example, the one-pole filter design method proposed by Bank in (Bank, 2000) can be used. Note that it should be taken care of that the whole filter, i.e., reverb algorithm is stable. This can be a problem, since the frequency dependent decay changes a lot from low to high frequencies. Summing it all up: the design of a reverb algorithm to approximate a guitar body response is a case with rather special parameter choices.

Figure 49 shows the time and frequency responses of a guitar body model obtained with a reverb algorithm. The algorithm is the one discussed in (Väänänen et al., 1997) with the following parameters: delay line lengths in samples: 199, 181, 227, and 257; for the lowpass filters the ratio between the reverb time at Nyquist and zero frequency is 0.5; the values for the first-order allpass coefficient in the comb-allpass sections are all 0.3 and their delay line lengths are 16, 14, 18, and 21, respectively. Figure 49 (a) displays how there are some multiples of the delay line lengths that are superimposed. Also, there is a delay before the first output sample, which is as long as the length of the shortest delay line. Figure 49 (b) illustrates how the resonances are sparse at low frequencies, and form sort of resonance clusters. These clusters are caused by the comb-allpass filters and their occurrence density can be controlled with the allpass section coefficients. Fig. 49 (b) also shows how at high frequencies the resonances are dense, i.e., all in all the whole response is colored in the same manner as a guitar body

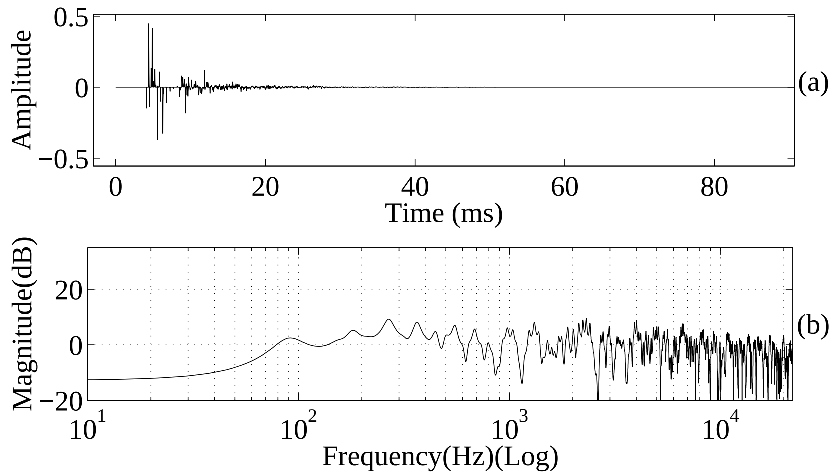

Figure 49: Guitar body model with reverb algorithm for acoustic guitar.

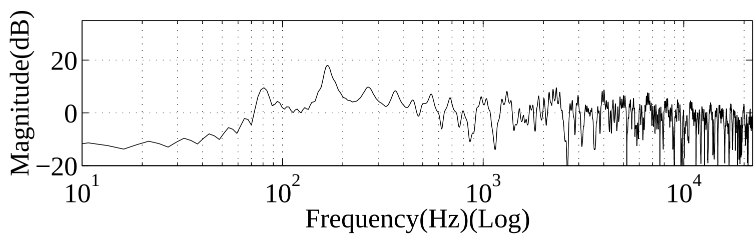

Figure 50: Body model with reverb algorithm and two resonators in cascade.

response. When listening to an acoustic guitar pickup signal filtered with a properly tuned reverb unit, the results are positive. Positive in the sense that even if accurate modeling of a target response is not possible the timbre is improved towards a more natural tone.

5.6.1 Improvements

In available reverb algorithms the amplitude, frequency, and decay rates of individual resonances cannot be controlled. Therefore, for accurate modeling the lowest body modes should be modeled separately. Here the lowest modes have been modeled with cascaded IIR resonators (see Eq. 19). For accurate low-frequency region modeling any appropriate scheme can be used. It is possible to implement them with parallel resonators or with more elaborate Kautz filters as in (Penttinen et al., 2001a) or with a multi-rate scheme as in (Bank et al., 2002). In the two latter schemes the modeling is split into two frequency regions. Also, special care has to be taken to avoid overlapping of the two frequency regions, with appropriate low and highpass filters.

In addition to separate resonators, low-order magnitude envelope filters improve the accuracy of the modeling result. For the design scheme, a low-order LP-model can be an appropriate choice. Also, a parametric equalizer can be a solution for this task. This way important envelope characteristics can be preserved without considerably exhausting computational resources.

Figure 50 displays the magnitude response of a guitar body model produced with a reverb algorithm and two resonators in cascade (compare with Fig. 49 (b)). The resonator parameters have been tuned to match with values obtained from a measured acoustic guitar response. Hence, the low frequency behaviour is more accurate than only with a reverb.

6 Further Questions and Problems

Further issues related to guitar body modeling are covered next: First discussing the behaviour of the low-frequency response and acoustic feedback. Then covering themes related to further improve the illusion of making an electric guitar sound more like an acoustic one[13]. Finally discussing the amplitude and decay of acoustic guitar body modes and three-band parametric equalization.

6.1 Low-Frequency Response

The radiation energy of a guitar body at low frequencies is very small. As a consequence, a guitar body model filter (DECON and MIR) attenuates the very low-frequencies ($<$ 80 Hz) that are present in the pickup signal. To change this situation a target response can be reformed by pre-filtering. Pre-filtering should be done so that the low frequency region stays approximately flat (0 dB) between very low-frequencies (40-50 Hz) and the lowest body mode (80-200 Hz). Hence, each body model filter has to be pre-processed separately. Pre-filtering can be performed with any suitable filtering method that does not corrupt the response otherwise. Furthermore, to improve controllability in the low-frequency region a controllable high-pass filter can be cascaded with the body model. This way the very low-frequencies can either be flat or they can be attenuated to simulate a more realistic situation. However, a rich timbre with a proper amount of low frequencies is often preferred. This can be understood by the fact that people using a bridge pickup are accustomed to have the low frequencies present, even if they are not present in the radiated sound of a real acoustic guitar.

6.2 Acoustic Feedback

Acoustic feedback is a problematic phenomenon that usually occurs in live performance situations. Acoustic feedback occurs when the amplified signal from a loudspeaker re-enters the sound system through a microphone. Hence it is fed-back to the system and amplified by an uncontrollable amount. Acoustic feedback is typically observed as a very loud ringing resonance. The ringing frequency is dependent on the resonances of the microphone, amplifier, and the acoustic environment. In the case of an acoustic guitar the feedback frequency is typically one of the lowest body modes.

[13] The differences between an electric and an acoustic guitar have been discussed in Chapter 2 and summarized in Section 2.5.

Acoustic feedback can be cancelled or the situation can at least be improved with (I) a high/low-pass filter or (II) with notch filters. For an acoustic guitar a parametrized high-pass filter improves the feedback sensitivity of the system since it attenuates the frequencies of the lowest body modes.

When using notch filters to improve feedback sensitivity they should be parametrized with variable center frequency (f), gain/attenuation (G), and bandwidth (Q). Another possibility is to have an automatic notch filter that detects the frequency of the feedback automatically and sets the parameter values (f, G, Q) of the notch filter thereafter. This can be implemented with adaptive filters with appropriate complexity (Kuo and Morgan, 1999). An adaptive filter increases the computational complexity of a system and can endanger real-time implementation. However, e.g., a least mean square (LMS) algorithm has only a few multiplications and summations and can be considered as a viable option.

6.3 Simulation of Beats in Harmonics

When simulating the body of an acoustic guitar for electric guitar signals the attack part of a pluck sounds relatively acoustic-like. However, in long notes the end part of a slow decay sounds somewhat steady when compared to a corresponding acoustic guitar note. This is due to the difference in beating of harmonics: In an acoustic guitar note the harmonics tend to have more beating than a corresponding electric guitar note. The beating of harmonics and envelope fluctuation is caused by a difference in the effective length of a string and nonlinear mode coupling, respectively, as discussed in Section 2.1.

For simulating beats in harmonics a modulated filter bank can be used as shown in Figure 51. Each channel is modulated with a proper signal. Psychoacoustically the filter bank does not need more than one filter per critical band. The modulation signal is typically around 1 Hz, the frequencies and phases of the modulating signals in different bands should not be the same, otherwise the effect is reduced to a tremolo effect. (Kahlin and Ternstrom, 1999) used a similar kind of structure for simulating ensemble sounds or the chorus effect.

The amount of amplitude modulation of harmonics is note-dependent, i.e., the amount depends on which string a note is plucked and on the amount of plucking force, among other things. Hence, to be more accurate in the simulation of beating harmonics the system should identify the plucking position and the fundamental frequency of a note. With this information the parameters of the modulation signals could be adapted to adequate values, e.g., with a help of a database search of acoustic guitar note characteristics. Note that this methodology is thought to be applied for single notes, so that a pluck triggers synchronized modulation. When applying this method to polyphonic signals each string should have a separate pickup.

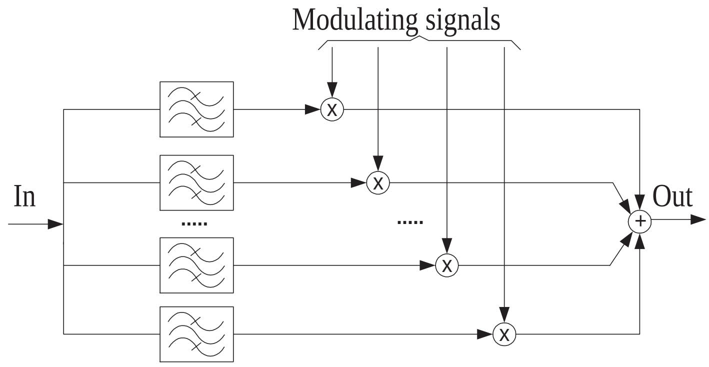

Figure 51: A block diagram for modeling beating harmonics with a filterbank modulator. This simulating scheme is used for electric guitar signals to model the amplitude modulation of harmonics present in acoustic guitar notes.

6.4 Control over Decay and Sustain

Another difference between an acoustic and electric guitar is the decay time of played notes. The body of an acoustic guitar consumes the vibrational energy of the strings, hence the electric guitar has a slower decay and longer sustain than an acoustic guitar. To achieve a faster decay when simulating an acoustic guitar sound for an electric guitar the strings may be damped mechanically or the pickup signal can be processed by DSP means.

The mechanical damping can for example be realized with a soft damper that touches the strings, near the bridge. A skilful player may damp the strings lightly with the palm of the right hand.

The signal processing way to make notes decay faster is to use a dynamic expander for the magnetic pickup signal. An arbitrary dynamic expander, e.g., (Orfanidis, 1996) (p. 384-388), using feedforward or feedback techniques can be used in cascade with a body simulation filter as long as it yields an appropriate envelope and does not distort the output in an undesirable way.

The dynamic expansion processing is useful and works for monophonic playing, but in the case of a polyphonic signal the task is problematic. If a chord is played by exciting the notes simultaneously the DSP approach should work relatively well. The problem is when a new note is plucked while another is playing. In such a case the new note (signal) will again amplify/expand the total gain up before scaling it down. As a result an illogical envelope for the output signal will be produced, as far as considering the note that was already decaying. A solution to this would be using a multichannel pickup that has a separate output for each string. Another possibility would be to use advanced signal segregation techniques to separate notes from each other, but this field is still under development and is not possible in real-time.

6.5 Simulation of Sympathetic Vibrations

The amount of sympathetic coupling and vibrations of strings that have not been plucked is stronger in acoustic guitars than in electric ones. This is due to the more rigid string termination of an electric guitar and the absence of the hollow body.

There is no simple way to simulate sympathetic coupling between strings for an electric guitar. One option could be to produce electromechanical couplings between strings, but this is not a true DSP solution. Another more DSP oriented solution would be to synthetically produce sympathetic vibrations, meaning that sympathetic string models like in Figure 29 would be excited with the pickup signal. A problem in this case is that the string models cannot easily or automatically be damped according to the fingering of the electric guitar.

6.6 Amplitude of Body Modes

The body model filtering schemes discussed in this work are in a sense always the same, i.e., the input signal is constantly filtered with a static high-order filter. However, the body modes of an acoustic guitar do not behave as statically as the modeling filters. Figure 52 shows how the two lowest body modes (100 and 200 Hz) of a classical guitar decay clearly faster than the fundamental of a e_1 note (above 300 Hz) played on the b-string. Investigations during this work have also shown that the amplitude of low frequency body modes are relative to the amplitude of the fundamental of the string and are dependent on the plucking position: the closer the plucking position is to the bridge the larger the the amplitudes of the body modes are when compared to the string vibration amplitudes. More research is needed for determining the relative amplitudes of the body modes when different strings are excited with different fret combinations.

To model this behaviour the amplitudes of the resonances in the body model filter should be changed. This way the amplitude of synthesized body modes change relatively, compared to the string vibration amplitude. To model this kind of a process the pluck position and moment should be known. The moment of a pluck can for example be determined by tracking the ratio of low and high frequencies in a dynamic manner. A method for estimating the plucking point on a guitar was proposed by (Traube and Smith, 2000), but this method does not run in real-time and works well only on open strings. The decay times and relative amplitudes of the modes should be known so that the synthesis of these phenomena could mapped as a function of the input signal.

6.7 Parametric Equalization

It is useful to be able to fine-tune the balance of the spectrum, especially when looking for a sound that satisfies the player. Therefore a three-band parametric equalizer

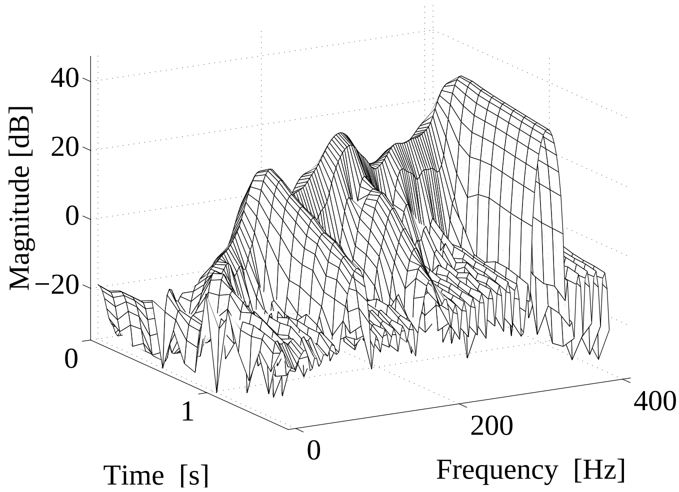

Figure 52: Exemplifying decay of lowest body modes with respect to decay of fundamental frequency string vibration.

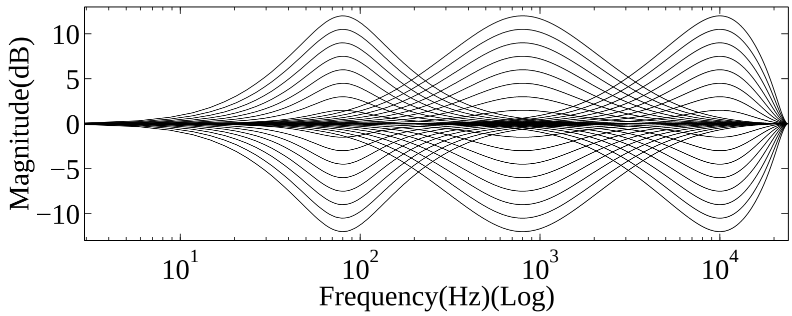

Figure 53: Parametric three-band equalization for bass, middle, and treble frequencies.

was also implemented for the real-time unit in cascade with the body filtering. The specifications for the center-frequencies and band-widths of the filters were given by Harri Saastamoinen[14]. The digital parametric equalizer magnitude responses are shown in Figure 53. These equalizers have been implemented with the second-order peak filter structure discussed in (Zölzer, 1997) (p.129-134). These equalizers enable to further specify the timbre of a sound, e.g., boost the high-frequencies even more if the acoustic guitar simulation for an electric guitar sounds too warm, caused by, e.g., a different low-pass filtering characteristics of a used magnetic pickup.

[14]Bass: f_c = 80 Hz, Δf = 85 Hz, Middle: f_c = 800 Hz, Δf = 1380 Hz, Treble: f_c = 12 kHz, Δf = 10 kHz.

7 Results

In this study the goal was to simulate the resonant body characteristics of a guitar, missing in a guitar pickup signal. The sound quality of an instrument and an equalization result is a matter of taste. In some cases the dry signal of an undersaddle pickup of an acoustic guitar, or the magnetic pickup signal of an electric one, may be preferred over an equalized signal. Also, a mix between a filtered and unfiltered signal might be used. In this work the results are viewed on the basis how a pure unprocessed pickup signal or an acoustically radiated signal sounds like, when compared to a signal filtered with a guitar body simulation filter. Also different filter structures and filters orders have been compared with each other and in respect to target responses. The joint goal was to make a pickup signal filtered with body model sound like the acoustic radiation of an acoustic guitar.

This chapter discusses the results obtained with the presented methods for enhancing the timbre of guitar pickup signals using digital filters. First modeling results will be discussed on a very general level. Then the focus will shift to results obtained for acoustic guitar pickup signals, so that results with different filter structures and design methods will be included, giving a more detailed treatment. After this results obtained for making the electric guitar sound more like an acoustic one will be discussed. In this part the discussion on different filter structures will be left out, since the results obtained for the acoustic guitar apply also for the electric guitar. Hence, needless repetition will be avoided. The results section ends with a discussion on the results obtained with modulated warped filters, i.e., how well a guitar body model can be made to sound like a smaller or larger guitar.

7.1 Basic Modeling Results

The techniques discussed in this work are able to simulate the important characteristics of a hollow acoustic guitar body. The main and typical characteristics of a guitar body response are the following. (I) Discrete resonances in the low frequency region, so that the two lowest ones decay slowly and are strong in amplitude. (II) At high frequencies the resonance density is higher than at low frequencies and the response is more reverberant like. In addition, the overall trend is that the high frequency components decay much faster than the low frequency body modes. These characteristics can be modeled and captured with all the methods discussed in this thesis, some methods succeeding better than others.

Proper listening tests were conducted for changing the perceived size of guitar body model filters. The discussion for the basic modeling results relies on informal listening test made by the author, Matti Karjalainen, and Vesa Välimäki. Some perceptual results are already and also discussed in Chapter 5 as the different filter structures are discussed.

7.2 Acoustic Guitar

One of the goals of this work was to improve the naturalness of a undersaddle pickup signal mounted to an acoustic guitar. In other words, the objective was to make an acoustic guitar pickup signal sound more like the timbre perceived as a listener stands beside a player in anechoic conditions. The signal of an undersaddle pickup does not contain all the characteristics of the guitar body. These characteristics can be simulated with digital filters. For improving the timbre of an acoustic guitar the deconvolution method (DECON) (discussed in Sec. 4.1) was mainly used. Filters were designed and applied for different sized steel stringed acoustic guitars and a classical guitar.

When comparing an unprocessed pickup signal to the same signal filtered with a body model and using normal playing as an excitation the difference is easy to perceive. This is the case both for flat-top (steel strings) and classical (nylon strings) guitars. When the guitar is played with a slide (a metal/glass tube in left hand) the difference is particularly notable.

— **FIR** — Straightforward FIR filters performed the heavy equalization task the best. FIR filters were compared to acoustic guitar radiation signals and unprocessed signals. Comparisons were also made between different filter orders. The lowest body modes are perceptually presented well and the high-frequencies likewise. However, the lowest body modes can sometimes be considered rather boomy, i.e., too emphasized. Hence, the adjustable resonators modeling the lowest body modes are a practical solution for enabling specific control over the low frequencies. When comparing the perceived attack of a filtered and unfiltered signal a difference can be noted. One explanation for this could be the masking effect the lowest body modes have, especially if their amplitude is high. This is further backed with the observation that when an FIR filter is shorter than 500 taps the low frequencies decay too fast and the attack and high frequencies are too prominent. Best results are obtained with 1000 to 5000 taps long FIR filters. If the filter (DECON) is longer than this the response sound reverberant, caused by the noisiness of the filter.

— **IIR** — Recursive IIR filters designed with Prony's method or linear prediction were compared to 2000 tap and 5000 tap FIR target responses, respectively. Linear prediction models result in a minimum-phase filter which degrades the temporal behaviour compared to the target response. Also the magnitude response is not matched to a satisfactory result (if one is being very critical), even with LP model orders as high as 1000. The problem of assuring stability arises for high-order LP model filters. Also the computational load is unreasonably high compared to the results. Hence, it is quite impractical to consider to use such high-ordered LP-model filters. The Prony's method produces a zero-pole model of the target response, i.e., it enables the modeling of the non-minimum phase build-up. However, the results are not as good as could be expected. Therefore, these IIR filter design methods, in their simplest form, are inferior compared to an FIR topology. On the other hand, both the Prony's method and the LP modeling are able to model some of the important characteristics of a guitar body response. This is why they should not be discarded completely when considering different filter designs and structures for modeling guitar or instrument body responses.

— **Warped Filters** — Warping a filter structure enables to model the frequency resolution of the auditory system. This gives the possibility to concentrate the modeling power to certain frequency areas and also decrease the order of a filter compared to a non-warped structure. Warping the recursive filter structures improves their performance considerably at relatively low filter orders. With a warping coefficient λ approximating the Bark scale the lowest body modes and low-frequency area are modeled adequately. For high frequencies the envelope of the target filter is preserved. This can be improved by cascading two warped filters: One with a positive λ value, modeling accurately the low frequencies, and another with a negative or zero valued λ. The latter filter in the cascade chain is designed from the residual of the first filter. Since the warped filters are designed from a minimum-phase response the temporal differences discussed earlier remain. All in all, warping a filter can significantly improve its useability to model guitar body responses.

— **Lowest Modes Separately** — When modeling low-frequency resonances separately with second-order resonators the residual impulse response can be truncated. This improves the computational load of the implementation, especially if the residual is modeled as an FIR filter, rather than modeling the whole response with a single FIR structure. Another important issue is that modeling the lowest body modes separately substantially improves the controllability of the resulting timbre. As for the implementational schemes, a cascaded resonator bank with the main filter can result in an equivalent response with the target response. A parallel implementation scheme cannot easily match the phase response, therefore artifacts are produced in the magnitude response. In this case, perceivable differences can often be avoided by designing the resonators carefully and using only a few of them. The controllability of the two lowest resonances enables a user to decrease (or increase) the low-frequency 'thump' or ringing characteristics of an acoustic guitar body model response. Also, the adjustability of the resonators makes it possible to diminish acoustic feedback problems.

— **Reverb** — The experiments with reverb algorithms show that they are a computationally efficient way to model a guitar body response. With short delay lines the colored characteristics is captured well, including the reverberant-like high frequency behaviour. Changes in the short delay lines affect considerably the resulting timbre. Modeling the low frequency resonances separately with cascaded resonators improves the accuracy of the modeling result. However, the low frequency modes should not be emphasized too much or else the response sounds too boomy. Very accurate modeling with reverbs is difficult, but practical results can be achieved.

7.3 Electric Guitar

Producing an acoustic guitar timbre for an electric guitar pickup signal is an interesting and practical challenge. To be able to use just one electric guitar and still reproduce both electric and acoustic guitar timbres is very motivating. The methods and filters used in this work are capable of doing just this. The modeling filters have two prominent tasks: synthesize the resonances of a guitar body and cancel the low-pass effect of a magnetic pickup.

— **Different Methods** — The modeling filters obtained with the deconvolution method (DECON) sound better than the ones obtained with the modified impulse response (MIR) method. High frequencies sound more natural and the lowest body modes are also present. In the case of the MIR filters the lowest body modes are very prominent, but the high frequencies lack clarity and brightness. On the other hand, the DECON filters are very noisy and the responses have to be very short, to avoid the response to sound reverberant. A short target response also decreases the decay time of the lowest body modes. This can be improved by modeling them separately with resonators and then truncating the FIR response. Both methods make the magnetic pickup signal of an electric guitar resemble the timbre of an acoustic guitar.

— **Choice of Magnetic Pickup** — The equalization results sound closest to the radiated sound with the middle pickup on a handmade Stratocaster copy. This is understandable, since the middle pickup is closest to the place it has been during the measurements. Theoretically, the magnetic pickup should be at the same place during the measurements and when applying the filtering to a signal from an electric guitar. Practically this might be very difficult, however it does not destroy the sought objective: Making the electric guitar sound more like an acoustic one. Even if the magnetic pickup is not at the exact same place and the pickup is not same one, the results are very positive.

— **Noise Problems** — High-frequency noise is a problem when simulating an acoustic tone for an electric guitar. This is a consequence of amplifying the small amplitude low-pass filtered portion of a pickup signal, which results in more noise. A trade-off can be done between the noisiness and high-frequency behaviour of the simulation filter.

The amplitude of very high-frequency partials of a vibrating string in a real guitar, are lower in amplitude and decay much faster than low-frequency partials. For this reason, the very high-frequency portion, e.g., above 15 kHz or even lower, can be lowpass filtered to decrease the noisiness of the simulation filter.

7.4 Listening Test Results with Modulated Warped Filters

Next the listening test results obtained for modulated warped filters will be discussed. The objective of the listening test was to find out if a larger sized body filter can be re-sized to a smaller one, and vice versa, by changing the warping parameter λ. Modulation of warped filters was discussed in Sec. 5.3 and the description of the conducted test can be found in Sec. 5.3.1 and 5.3.2. First, the main results of the conducted listening tests will be described. Then a more detailed look on the results and Figure 54 will be given. Finally, spectral interpretations of the results will be presented.

7.4.1 Listening Test Results

Let the frequencies of the lowest body modes be hypothesized as guidelines to which direction the warping should occur, in order to change the perceived size properly. Under these circumstances, when the observed size of the soundbox should be altered, a decrease in the warping coefficient, λ, should correspondingly resolve in an identifiably smaller sized instrument. In the same manner, an increment in λ should cause an enlargement in the apparent size of the guitar body. Figure 54 illustrates the listening test results, where the x-axis indicates each test case and the y-axis the values of λ. It shows that the direction of the adjusted warping parameter is in some degree consistent.

The median values for λ, from all the listening test results, are 0.713 and 0.742 for decrement and increment of the distinguished size of the guitar body, respectively. Hence, when comparing these values with the initial λ value, 0.73, the shift direction of the warping parameter is in line with the assumption of (Lansky and Steiglitz, 1981). The inter-quartile range, IQR, expresses the range of half of the data. The IQR values, from all data, for shrinking and enlargement of the observed body size are 0.032 and 0.049, respectively.

From informal feedback from the subjects it can be concluded that reducing the perceived guitar body-size was easier and the resulting timbre was more pleasing to the testee than in the case of enlarging. In addition, the perceived size is more identifiable in the reduction case than in the enlargement one. This is due to a grown uncertainty of the perceived size when the value of λ is increased. This can be interpreted as a consequence of the shift of the higher resonances to lower frequencies. Due to the shift, the lowest body modes are also shifted and emphasize even lower frequencies, and therefore strengthen the perception of a larger instrument. At the same time,

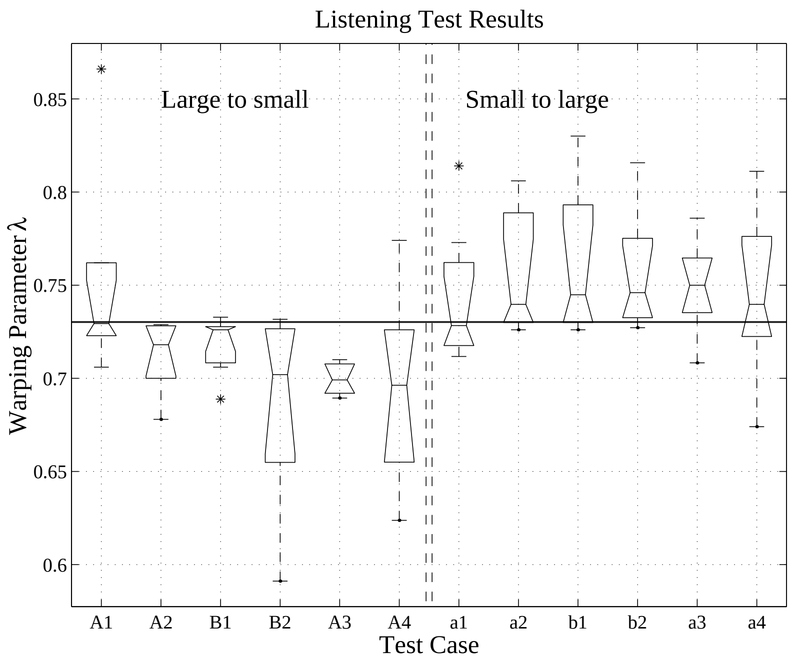

Figure 54: Listening test results as a box-plot.

the strong resonances at higher frequencies also get shifted downwards on a nonlinear frequency scale and apparently increase the feel of a distinguishably small sized guitar. It can be argued on the strength of the listening test results and the informal feedback that the lowest body modes are more significant when identifying the size of a guitar body model than higher modes. Therefore the ambiguity of the perceived size is also dependent on which frequency range the listener concentrates on or gives more weight to. In overall, best results were obtained when the purpose was to shrink the size of the body model.

The ambiguity of the changing of the perceived size of the body could be reduced by notch filtering some of the very strong resonances at frequencies, e.g., above 1 or 2 kHz, as discussed in Sec. 5.4.2. This could be justified since the strong resonances at high frequencies, in filters obtained with the deconvolution method, do not simulate body resonances as discussed in Sec. 4.1.3.

7.4.2 A More Detailed Look at the Listening Test Results

The vertical dashed lines in Figure 54 separate the matching cases, where a large soundbox filter was adjusted to sound like a small one, and vice versa. These cases are displayed at the right and left hand side, respectively. The original λ value, 0.73, is indicated with a thick horizontal line. The alphanumeric indicators on the x-axis correspond to different excitations and filters in the following way: Letters A and B indicate the used filter, so that A is a straightforward WIIR and B is an A filter smoothed with Eq. 54, when $a = 0.98$. The numbers reveal the excitation signal, so that 1 and 2 correspond to picked and strummed guitar playing, respectively, and 3

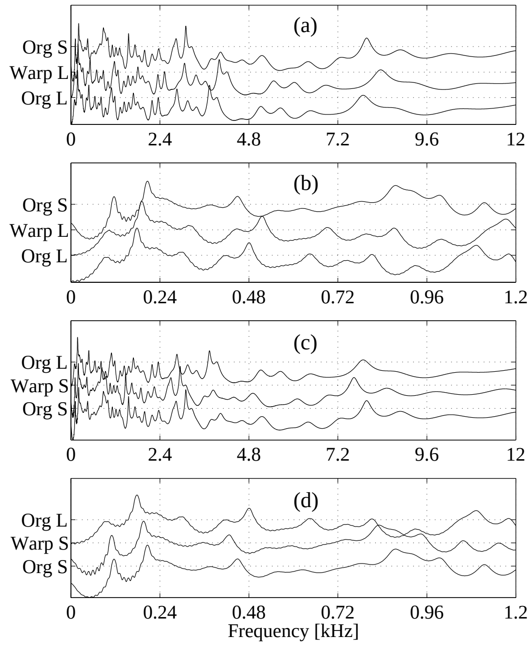

Figure 55: Magnitude responses before and after matching the perceived sizes of the guitar bodies. Reducing the observed size: panes (a) and (b), and enlarging: panes (c) and (d).

stands for the impulse train, while 4 indicates the white noise bursts. Furthermore, the cases where the large soundbox was matched with the small one are denoted with capital letters, and the small to large matching cases are displayed in lower case. In Figure 54, each box has lines at the upper (75%) and lower (25%) quartile values. Moreover, the median value is indicated between these values with a line and as a narrower part of each box. The whiskers are lines extending from each end of the box to show the extent of the rest of the data. Outliers, displayed with the symbol '*', are data with values beyond the ends of the whiskers.

The most systematic results were obtained when the excitation signal was an impulse train (see cases A3 and a3), even more so in case A3. All IQR values in the smoothed cases (B and b) are on the expected side of the initial λ value. Hence, the smoothing of the magnitude response can be considered as clarifying the perception of the size of the soundbox. Since test cases A1 and a1 were the first ones to be matched in the listening test it might be this that explains why these results behave poorly, rather than the lack of smoothing.

7.4.3 Spectral Interpretation of Results

Figure 55 shows how the resonance structures have matched when the median λ values, from the listening test results, are observed together with original λ values. Panes (a) and (b) in Fig. 55 represent the magnitude responses in the shrinking case while screens (c) and (d) display them in the enlargement case. The so called target response is the topmost magnitude response in all the panes. The initial body filter is depicted as the bottom response and the warped (median valued) in the middle. Moreover, Figs. 55 (b) and (d) zoom to lower frequencies.

By examining Figure 55 one can see that the large and small sized body filters differ from each other. The most notable similar features are the three lowest resonances (80-500 Hz). The higher modes behave more irregularly. By viewing Fig. 55 it can be noted: the two lowest resonances move to the assumed direction, whereas the next strong resonances match better before warping. Moreover, the body models differ from each other more than in the frequencies of their resonances: magnitudes, bandwidths, and the number of resonances vary. Therefore, it should be pointed out that by simply warping (change of λ) a body filter does not result exactly in another filter. An exact shift from a body model filter to another can be obtained by morphing a filter to another as discussed in (Penttinen et al., 2001a).

8 Discussion and Future Work

— **Subjective Opinions** — In more ways than one, the situation where and how an object, application, algorithm or instrument is used affects the expectations set on it. When building a guitar there is no one-right-method to achieve a desired result. The opinion of a player affects strongly what he or she considers to be adequate or good. Besides that, when considering the playing of an instrument, there are different instruments for different purposes. That is to say, an electric guitar suits better to some styles of music than an acoustic one. In the same manner, one body model filter is better suited to a situation or song than another.

— **Music vs. Science** — In contrast to the vagueness and opinion-emphasized world of music, different fields of sciences often give a strict and blunt solution to different problems, or at least try to. A theorem can be proven beyond dispute and it states a scientific truth. However, in many real-life situations the case may not be as straightforward as a line of thought has first suggested.

Considering the instrument body modeling themes in this light: The theoretically most correct concept does not self-evidently have to be the best one, when applied to a real-life situation. This means that even if a body modeling method has theoretically more flaws or ambiguous aspects than another, it does not necessarily mean that the audible results is also worse.

The more there are modeling methods, filters, and setups to obtain an instrument body model, the more possibilities, naturally. But for a musician too many options, can cause problems in the decision making process and if the timbres are very alike the musician would be doing unnecessary work. In parallel, this emphasizes the need for a model that is adjustable, so that natural transitions from a timbre to another are possible. From a DSP and real-time implementational point of view, this should also be as efficient as possible. These kinds of transitions are possible, e.g., by modulating or changing the value of the warping parameter λ of warped filters (Sec. 5.3).

The audible differences between filters of different order are not always as big as for example the magnitude responses might sometimes suggest. An auditory model could be used to give a more realistic perspective to the actual perceived differences between filters. Another option would be to conduct listening tests, which would also give an opportunity to get feedback on the naturalness of different body simulation filters.

The concept of a good sound or timbre versus a scientifically qualified sound/timbre is another interesting question. Typically, an acoustic guitar is not recorded in anechoic conditions at a one meter distance from the guitar, at the level of the sound hole. Rather, the recording occurs in a room, so that one microphone might be placed near the strings just beside the neck at the 12^{th} fret and possibly another one further away. Therefore, experiments with different measurement locations in respect to the body and strings would be interesting and informative.

— **Getting Excited** — One possibility for future work would be to investigate the effect of the excitation and create a model for the excitation process. The excitation process here means the interaction between the finger/nail/plectrum and the string. Some research on this interaction has been made, e.g., in (Cuzzucoli and Lombardo, 1999) a physical model was created for the process. In (Cook, 2002) the effect on the produced spectrum with different means of excitation are shortly discussed. The discussion shortly: A guitar pick or plectrum produces a brighter timbre than a finger. Cook (2002) also proposes two models for the excitation process. The up and down strokes of a string have different timbres, especially when played with fingers/nails. This is an example of an aspect, around this theme, yet uncovered and modeled. An excitation model with intuitive parameters would naturally be a challenging and worthwhile goal.

An excitation model could improve the result of enhancing a guitar pickup signal timbre by introducing characteristics associated with the plucking procedure. Typically, most of the high frequency plucking procedure noises are not captured by an undersaddle pickup on an acoustic guitar or even less by a magnetic pickup on an electric guitar. Also, in a real acoustic guitar the high frequency plucking noises are mainly radiated through direct radiation and not via the body. Therefore, a model for the plucking procedure would improve the tone of the attack or buildup part. This kind of a model would be used in addition to a guitar body model and would further enhance the naturalness characteristics.

A definite challenge in using an excitation model would be to determine the instances when to trigger the model. This is because it is difficult to obtain exact information from a pickup signal how and when a string has been excited. The guitar is a polyphonic instrument, therefore it is difficult to determine how a particular note from, e.g., chord is played, i.e., is the note excited by plucking with a finger or plectrum and with an up or down stroke.

— **Feel of an Acoustic Instrument** — When simulating an acoustic sound for an electric guitar the output from a loudspeaker changes. However, it is another matter to give the player of an electric guitar the feel of an acoustic guitar. One option could be to use the EMFi-film on the back of the electric guitar that would reproduce some of the vibrations produced while playing. This is highly hypothetical and maybe in someone's view almost irrelevant.

A theme close to this would be to simulate the signal path from the acoustic guitar to the ears of a player. This could be simulated by using head related transfer functions (HRTFs) (Huopaniemi, 1999) or by using instrument body models producing binaural responses. Binaural responses would be measured by having two microphones at both ears of the player, rather than one microphone in front of the guitar as in this work. This kind of signal processing would be naturally more reasonable for an electric guitar than an acoustic one since an electric guitar is a poor acoustic radiator. The use of headphones would be a natural choice for making the binaural sounds audible.

— **Anechoic vs. Reverberant Conditions** — The directional radiation properties of an acoustic guitar also affect the impression of modeling result quality. An acoustic guitar is typically listened to in a closed space, with inherent reverberation. Therefore the preconception a listener has, differs from the anechoic conditions, where the measurement in this work is set to. Naturalness to the modeling done in this work can be increased by adding reverberation to the filtered signal. However, to accurately model the path from a guitar to a listener in a room, the directional properties of a guitar should also be taken into account. Another issue is how much added reverberation affects the ability to distinguish between a computationally efficient instrument body model and a high-order target response.

— **Already Available** — Equipment modeling the properties of an acoustic guitar body response are commercially available, such as the BOSS AC-2 Acoustic Simulator, the BOSS AD-5 Acoustic Instrument Processor, the Zoom 504 Acoustic, and the Yamaha AG-Stomp. A number of multi-effect processors for guitarists also contain similar type of modeling, e.g., Line6's POD I/II, and Roland's BOSS GT-6. Some amplifiers also include acoustic guitar modeling, e.g., Marshall's AVT-150/275 and Line6's Vetta. An interesting study would be to compare the results obtained in this work with the modeling results (responses) of some of these products and give educated guesses for the modeling schemes used in them.

— **Only for Guitars and Measurement Microphones ?** — The use of the discussed body modeling methods is not restricted only to different types of guitars. The methods can be applied for other instruments with resonating parts that amplify and color the sound of the particular instrument. For example, a violin equipped with an EMF-film pickup can be measured in the same manner as a guitar and a body modeling filter can be obtained based on these measurements. However, some case-specific arrangements are expected, since the construction and excitation methods vary from an instrument to another.

Also, in the deconvolution method the characteristics of the microphone receiving the acoustic radiation could be modeled. By capturing the acoustic radiation with a microphone used in studios for recording acoustic guitars the microphone behaviour would also be present in the approximated transfer function. For this to be possible the microphone should behave linearly. If, however, the characteristics of a nonlinear microphone would be modeled, the nonlinear modeling has to be done separately, apart from the body modeling and its measurements.

— **At the End of the Day** — Regardless of the possible problems and inaccuracies involved in instrument body modeling, the techniques and methods proposed in this work give useful possibilities to enhance the timbre of guitar pickup signals, both for acoustic and electric guitars.

9 Conclusions

In this work we have studied and developed different acoustic guitar body modeling methods and techniques. The objective for equalizing or filtering the pickup signal of an acoustic guitar was to bring the timbre closer to the sound heard by a listener standing beside the player in anechoic conditions. All proposed and discussed methods and filter structures enable to do this, some better than the others. Also, the electric guitar can be filtered to sound more like an acoustic guitar. In addition, the perceived size of a body model filter can be changed, i.e., a small bodied model can be made to sound like a large one, and vice versa. The discussed parametrization of the lowest body modes also improves the control of the final timbre and enhances the implementational efficiency.

The process of making the string vibrations of an acoustic guitar audible is the following. First, the string vibrations of the guitar are transferred via the bridge to the hollow body. Then the body of the guitar amplifies the string vibrations and colors the response. This colored response is then radiated to the surroundings. Since the strings themselves have a poor radiation efficiency, the body of the instrument has a crucial part in making it audible and giving it character. Undersaddle pickups used in acoustic guitars are able to capture the transversal vibrations of the strings, but cannot efficiently capture the influence of the body. This explains the need for enhancing the timbre of guitar pickup signals. Implementing the enhancement with digital filters is possible since the transfer function from the undersaddle pickup to air-radiation can be approximated as being linear and time invariant.

To enhance the acoustic-like timbre of a guitar pickup signal, the transfer function from the pickup signal to air-radiation has to be modeled. In this work the approximation of the transfer function has been obtained in two ways. (I) By measuring two signals at the same time: the undersaddle pickup signal and the air-radiated sound. Then calculating the frequency domain deconvolution between these signals. (II) By measuring the air-radiation response of the guitar body when the bridge is excited with an impulse hammer. For an electric guitar the treatment is slightly different, since the lowpass characteristics of the magnetic pickup has to be reversed. In the measurement setup for the first method a magnetic pickup is used instead of an undersaddle pickup. In the second method the lowpass effect is compensated by high-frequency boosting the measured impulse response. These measurement setups enable to obtain a target response which approximates the transfer function from a guitar pickup signal to air radiated sound at one point.

The target responses obtained with the two methods discussed in this work can be implemented with different filter topologies. The best results were obtained with straightforward FIR filters of orders from 1000 to 5000. Recursive filter structures can theoretically improve the implementational load, compared to an FIR filter. The Prony's method and linear prediction models were used to design the recursive filters from the target responses. Strictly speaking, however, these design methods did not completely preserve the perceptual quality even if the computational cost could be reduced.

Frequency warping a recursive filter structure in a perceptual fashion improves the obtained audible response. This is because the lowest body modes are modeled better than with a corresponding regular filter. The quality at high-frequencies can be improved by using a cascaded filter designed from the residual of the first stage. For the cascaded filter the warping parameter λ should concentrate the modeling power to high frequencies.

By modeling separately a number of discrete strong and slowly decaying low-frequency body modes, the implementational efficiency and controllability of a guitar body model can be improved. To gain accurate reconstruction compared to a target response, a cascaded scheme should be used, rather than a parallel one. This is because in a parallel scheme phase-matching is very difficult, and as artifacts, dips in the magnitude response are easily created. However, this is not often so crucial that the parallel scheme should totally be ignored.

Acoustic guitar responses can also be modeled with reverb algorithms. Reverbs prove to be a computationally efficient manner to do this kind of modeling. However, accurate modeling of a target response solely with reverb algorithms is impossible. This is due to the fact that in reverbs the amplitudes, frequencies, and decay rates of resonances cannot be tuned independently. This situation can be improved by modeling strong low-frequency resonances separately with other filters.

When implementing a body model with a warped structure the λ coefficient can be set to be a free time-varying parameter. By changing λ, the frequencies of the resonances of the response will change. Hence, the lowest size-dependent body modes can be shifted in the frequency domain higher or lower. Perceptual studies done during this work prove that the perceived size of a body model can be changed by conservatively altering the value of λ in a stepwise manner. When altering λ in a more relaxed manner and faster the audible response results in an interesting sound effect, that resembles the phaser-effect. So by setting the λ free, virtual out-of-this-world instrument body models can be created.

Differences between body model filters can most easily be noticed when listening to the pure impulse response. In other words, differences are not as easily noticeable when guitar playing is filtered with different guitar body models. This brings the subject closer to real-life timbre issues, and the fact that playing is all that matters.

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A Appendix

Figures 56 and 57 depict guitar body model responses obtained with the DECON method for guitars with middle (D-805E) and large (J-855) sized bodies, respectively. In Figs 56 and 57 the pane arrangement from (a) to (d) is identical with Figs 31 and 33 on pages 44 and 46, respectively: In Figs 56 and 57 panes (a) and (b) display the time responses, so that (b) is the windowed version of (a), and (c) and (d) show the magnitude response of (b) on a logarithmic and linear frequency scale, respectively.

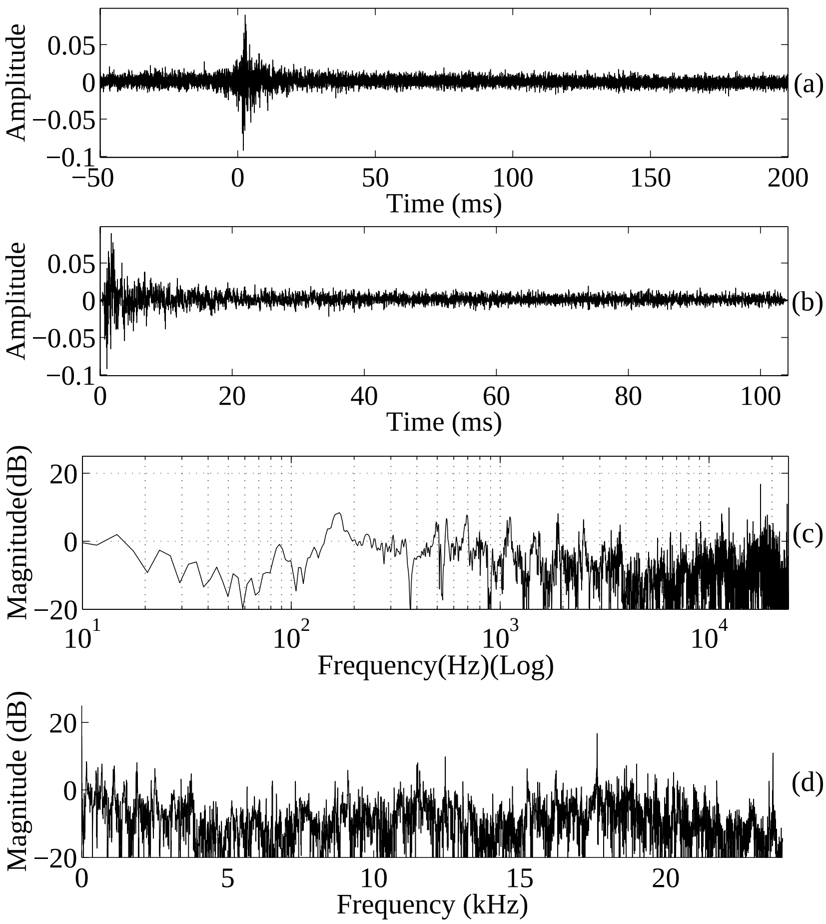

Figure 56: Body-model filter responses obtained with deconvolution method for middle-size bodied guitar when excited by playing. (a) shows 250 ms of deconvolution result, (b) windowed version of the result, length circa 104 ms, (c) magnitude response of (b) on logarithmic frequency scale, and (d) on linear scale.

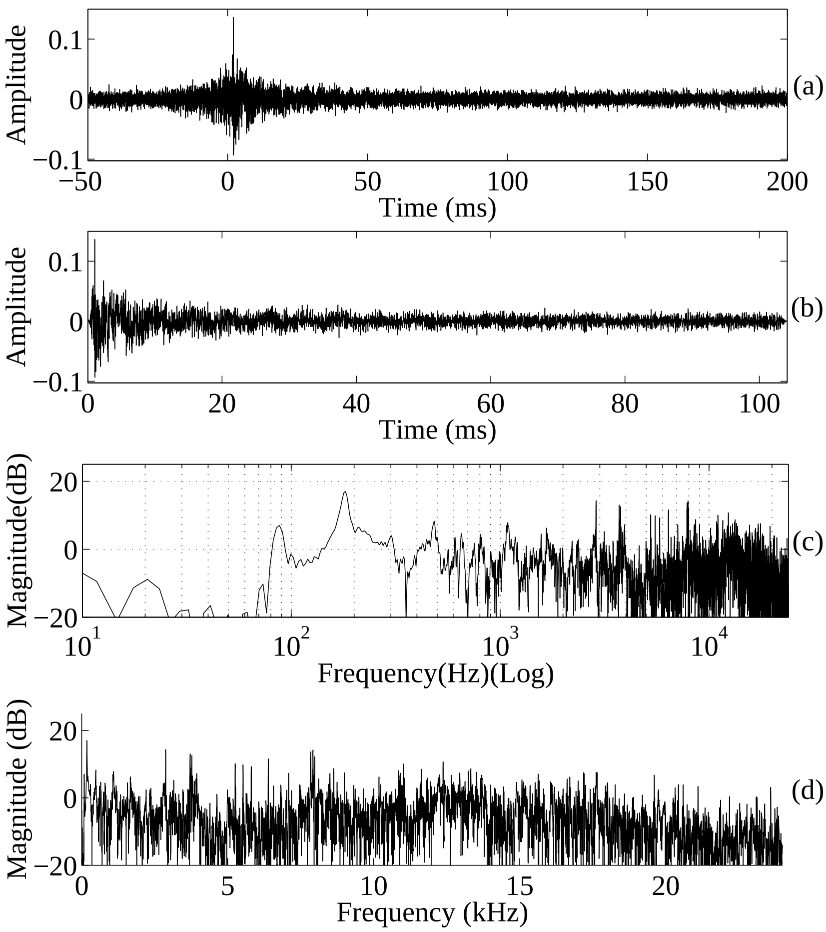

Figure 57: Body-model filter responses obtained with deconvolution method for large-size bodied guitar when excited by playing. (a) shows 250 ms of deconvolution result, (b) windowed version of the result, length circa 104 ms, (c) magnitude response of (b) on logarithmic frequency scale, and (d) on linear scale.