Morphing Instrument Body Models

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MORPHING INSTRUMENT BODY MODELS

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ABSTRACT

In this study we present morphing methods for musical instrument body models using DSP techniques. These methods are able to transform a given body model gradually into another one in a controlled way, and they guarantee stability of the body models at each intermediate step. This enables to morph from a certain sized body model to a larger or smaller one. It is also possible to extrapolate beyond original models, thus creating new interesting (out of this world) instrument bodies. The opportunity to create a time-varying body, i.e., a model that changes in size over time, results in an interesting audio effect. This paper exhibits morphing mainly via guitar body examples, but naturally morphing can also be extended to other instruments with reverberant resonators as their bodies. Morphing from a guitar body model to a violin body model is viewed as an example. Implementation and perceptual issues of the signal processing methods are discussed. For related sound demonstrations, see www.acoustics.hut.fi/demo/dafx2001-bodymorph/.

1. INTRODUCTION

The concept of morphing means a smooth transform from an object to another, so that the in-between stages have identifiable characteristics of both objects. Morphing is not a novel idea and the concept has been used both for images and audio. When applied for images, a morph process consists of smoothly changing images that move from one image to another. In audio the concept and the objective is the same: a perceived sound object should uniformly transform to another sound. To achieve a convincing morph in audio, spectrograms [1] and sinusoidal coding [2], [3] are often used as the morph domain where interpolation between the extremes is done. In audio, also (simple) parametric equalizers and shelving filters have been morphed [4].

This study can be considered as a continuation of work done on modulating instrument body models in [5], where frequency-warped IIR body models were modulated through one driving parameter, the warping coefficient λ. This enabled to alter the perceived size of a guitar body model. Complete matching in morphing between two body models cannot be achieved with that particular technique, whereas the techniques studied in this work make complete matching viable both at the beginning and the end of morphing trajectory. In addition, the shift between two different sized models is more natural. To improve the change (or shift) from a very large-size body model to a very small one it is useful to add an intermediate size model between the two extremes. Furthermore, an interesting morph is to move from one instrument class to another, e.g., from a guitar to a violin.

This study introduces techniques to morph between instrument body models, thus proposing an interesting and useful audio effect that can be implemented to run in real-time.

2. INSTRUMENT BODY MODELS

The role of body and soundboard resonators in musical instruments is twofold: (a) transmission of string vibration to the radiated sound field in order to make the instrument louder and (b) coloration of sound through spectral filtering and temporal spreading (reverberation). In a typical body of a string instrument the box exhibits mechanical resonances and the interior of the box create air resonances, radiating through a sound hole. The generation of the lowest resonances is relatively straightforward: a modified Helmoholtz resonance (around 100 Hz) in a classical guitar and the lowest mode of the top plate (around 200 Hz).

Physical modeling of the body behavior is rather complicated. The Finite Element Method [6] can be useful at low frequencies, while at higher frequencies the behavior can only be approximated in a more statistical sense. One further complication is the radiation pattern which varies for different vibration modes. Simulations based on detailed physical models are out of question in real-time sound synthesis and audio effects.

More practical modeling for audio frequency processing can be achieved when the instrument body is simulated by a digital filter. It is assumed that the propagation and air radiation of sound from strings through the body is linear and time-invariant (LTI) [7], [8], and [9], which in most cases is a valid assumption. Figure 1 depicts (a) the impulse response and (b) the magnitude response measured from a classical guitar. Even better overall picture is obtained from the time-frequency representation of Fig. 2.

Due to the LTI property, instrument bodies can be simulated efficiently by digital filters, and with modern processors they can be computed in real time. Different solutions to DSP-based body modeling have been proposed, e.g., in [9], [10], including FIR and IIR filters, frequency-warped filters, filters composed of separate resonators, waveguide models, artificial reverberation algorithms, etc.

A common feature for filter-based modeling techniques is that their parameters can be varied in time in order to dynamically change the body properties, thus making them potentially useful in morphing type of sound synthesis and audio effects processing.

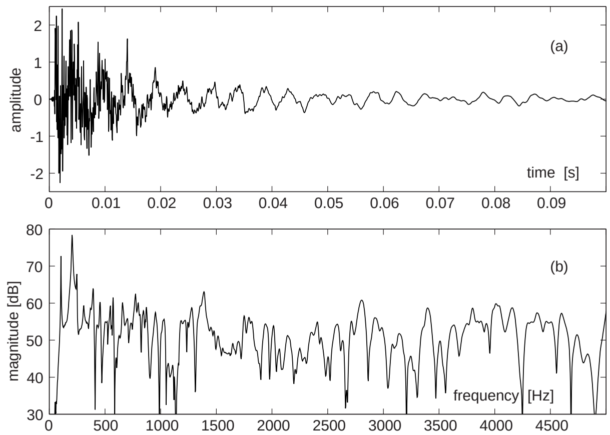

Figure 1: Measured response of an acoustic guitar body: (a) impulse response and (b) magnitude response.

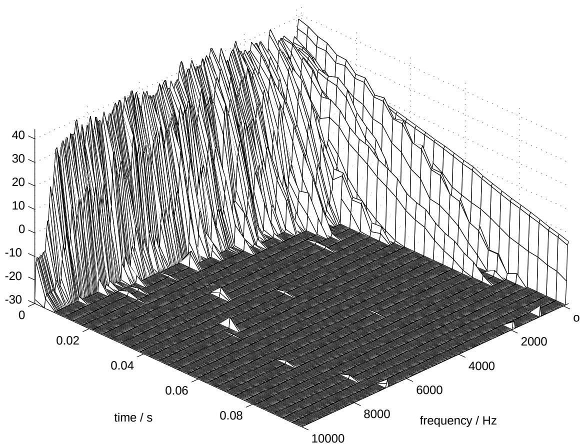

Figure 2: Time-frequency representation for the guitar body response of Fig. 1.

2.1. Measurement and estimation of body models

For filter-based modeling there is need for measurement and estimation of real instrument bodies. A traditional technique is to tap the bridge by an impulse hammer and to measure the radiated sound response for example in front of the sound hole [9].

In [10] we have proposed another technique, particularly for bridge pickups used for amplification of the acoustic guitar. There the guitar is played with a spectrally rich set of sounds when the bridge pickup output and radiated sound are recorded simultaneously. By deconvolution of the acoustic and the pickup signals (division of Fourier transforms) an impulse response is estimated for the body transfer.

After measuring a body impulse response, such as in Fig. 1, the response can be modeled by any of the filter approaches described in literature. As far as the model allows parametric interpolation through stable and meaningful states, it can be used for morphing purposes. In this study body responses have been estimated with warped IIR (WIIR) filters [11], [12], [13], designed with linear prediction (LP) methods.

3. MORPHING PRINCIPLES

Morphing of a musical instrument model can take many forms. Here we consider the case of string instrument bodies[1]. First we discuss general principles of body morphing, and in the next section a set of realization cases are presented.

3.1. Morphing with single filter model

The easiest way of filter-based model morphing is to design a digital filter that makes a good fit to a measured or estimated body in the beginning, and to another body, at the end of a morphing interval, then interpolating in between meaningfully and in a stable way. It is desirable that the modal resonance behavior of the bodies is reflected in the modeling, thus IIR type of filters are a natural choice. Since interpolating IIR filter denominator coefficients directly does not guarantee stability, other methods must be used, and they will be discussed in the next section.

The frequency resolution needed to represent single modes is an important design question. The lowest modes in the guitar, up to 2–4 kHz, may individually have an effect on perceived sound quality [14]. Therefore for high-quality modeling at least up to 1–2 kHz the filter should be able to follow both the spectral envelope and the temporal decay properties of the body response to be modeled. At higher frequencies we may model either (a) the spectral envelope only (i.e., tone correction) with low filter pole density, or (b) match the temporal decay properties also (i.e., reverberation), which requires high pole density. Warped linear prediction is a natural choice for balancing between low- and high-frequency resolutions [9].

Morphing between an initial and a final state is inherently an interpolation problem. It is possible, however, to extrapolate parameters also beyond the initial and final states, possibly with out-of-this-world effects, as far as the filter remains stable and the response is meaningful. We will describe one such case below.

3.2. Morphing with multi-part models

Based on the different behavior of low- and high-frequency parts of instrument bodies it is attractive to process them with separate models. In [14] we have studied body models consisting of a filter (such as warped IIR filter), for low-to-mid frequencies and using reverb algorithms to model the high-frequency end. In morphing applications this means that the two submodels are interpolated separately and crossover filters are applied to restrict the frequency range of each subsystem. Such partitioning could be extended to more than two bands (e.g., multi-rate filters), although the complexity grows.

3.3. Discrete Resonances

One more useful strategy for body filter models is to control some modes individually. Typically this means the lowest two resonances in the case of the classical guitar. These modes are eliminated from a measured/estimated model response and the rest of the response is modeled as a single higher-order filter. During synthesis the individual modes are interpolated separately and then the two submodels are combined. This can be based on a parallel filter

[1]Morphing could also cover string models and plucking models, although changing their properties, particularly the pitch, is more like regular musical expression than real morphing.

formulation, but cascaded second-order sections are a very natural solution [15].

4. MORPHING REALIZATIONS

When morphing between two filters by interpolating the filter coefficients the crux of the matter is to ensure stability at each morphing stage. The lack of assured stability at each stage is the reason why pole-zero filters described by polynomial coefficients (a_i) cannot directly be used. In some cases morphing (interpolation) with these kinds of filters might be realizable, but for example large changes in the z-domain might cause instable intermediate filters and hence produce an incomplete and distorted morph. The solution to the matter is to represent the filter coefficients in a domain where stability at each morphing stage is assured when the starting and end point filters are stable.

In this study the filter coefficients have been represented as reflection coefficients (RC) or as log area ratio (LAR) coefficients [16]. In other words, morphing between two filters has been carried out by interpolating reflection coefficients or log area ratio coefficients. In addition to interpolation, LAR coefficients enable extrapolation of filters. In this case we can go beyond morphing and shift into previously unexisting body model dimensions. Besides RC and LAR coefficients, line spectral frequencies (LSF) are a possible domain to represent all-pole filters that would assure stable morphs [16]. However, the use of line spectral frequency representation for high-order body models are ill-behaved, since significant numerical precision errors may occur.

4.1. Interpolating Reflection Coefficients

Representing coefficients of a zero-pole filter with reflection coefficients [16] enables us to morph from one filter to another by interpolating directly between filter coefficients. When an all-pole body filter of order p is presented by its filter coefficients, a_i, the reflection coefficients, a_j, can be calculated as follows

$$k_i = a_i^{(i)} \tag{1}$$

$$a_j^{(i-1)} = \frac{a_j^{(i)} - a_j^{(i)} a_{i-j}^{(i)}}{1 - k_i^2}, 1 \leq j \leq i-1, \tag{2}$$

where the index i takes values p, $p-1$,...1. Initially, $a_j^{(p)} = a_j, 1 \leq j \leq p$. Stability of the resulting filter is assured when $a_j \in$ [-1,1]. In the case of a zero-pole filter the nominator and the denominator polynomial coefficients are treated separately with Equations 1, and 2. Figure 3 illustrates magnitude responses when a small guitar body model is morphed to a larger one by interpolating reflection coefficients on a linear 12 step grid. The y-axis indicates the stage of the morph so that 0 and 1 correspond to the small and large body models, respectively. The body filters are warped IIR filters of order 100. Figure 3 shows how resonances are being morphed from one to another by subtly changing their magnitude and/or frequency characteristics.

4.2. Extrapolating Log Area Ratio Coefficients

As with the reflection coefficients, LAR coefficients allow us to interpolate directly between filter coefficients. The LAR coefficients are derived from RCs:

$$g_i = \log \frac{1 + k_i}{1 - k_i}. \tag{3}$$

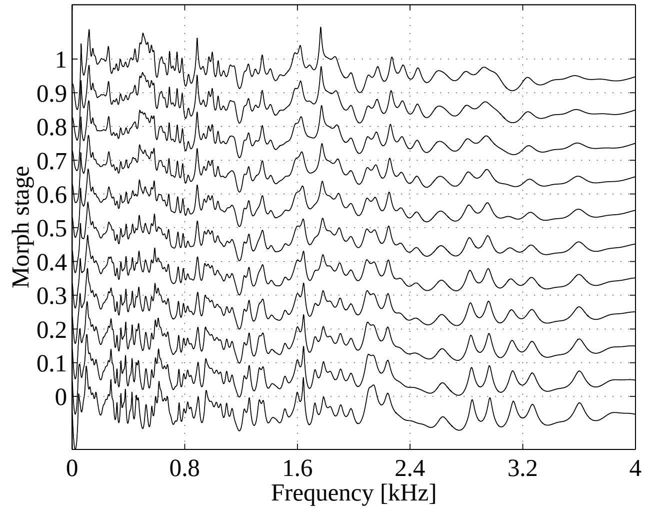

Figure 3: *Interpolation with reflection coefficients between two body models. Small to large (0 to 1).*

Whereas, RCs are bound $\in$ [-1,1] for stability reasons, LARs $\in$ [$-\infty$ to $+\infty$] with an assurance of a stable filter. This makes extrapolation of body models possible. Figure 4 depicts magnitude responses when LAR coefficients have been used to extrapolate beyond a small and a large guitar body filter. The original small and large sized body models are situated at zero and one on the y-axis, respectively. Whereas the changes in Fig. 3 might be more subtle, the changes beyond the starting point filters in Fig. 4, i.e. y-axis < 0 or > 1, are easily noticeable: drastic changes in resonance and anti-resonance magnitudes and frequencies.

5. MORPHING BETWEEN INSTRUMENT CLASSES

Instruments that have a resonant body or a soundboard are inevitably effected by their presence, through its amplification and coloration properties. The characteristics the body produces, changes from an instrument body to another. The changes in the responses are present in different sized bodies of the same instrument. Moreover, changes in response characteristics are even more definite when bodies of two different instrument classes are compared. Here we consider instrument body morphing from a classical guitar to a violin.

5.1. Case Study: Guitar to Violin

The bodies of a classical guitar and a violin differ in their physical size and shape. The difference in their size cause significant changes in the behaviour of their low-frequency response. Since the guitar body is larger in volume, longer wavelengths originate when the air inside the body and the top-plate start to vibrate. This again results in lower frequencies. As a consequence the two lowest body resonances of the guitar are situated clearly at lower frequencies than in the violin (e.g. guitar: 104 and 205 Hz, and violin: 270 and 490 Hz). In addition, the lowest body modes of the guitar decay much slower than the ones in the violin body. Both responses have a reverberant characteristic at high frequencies. In contrast to the low-frequency behaviour, the frequency-dependent decay at high frequencies can be considered to be much alike and somewhat the same. First, we look at morphing from a guitar body

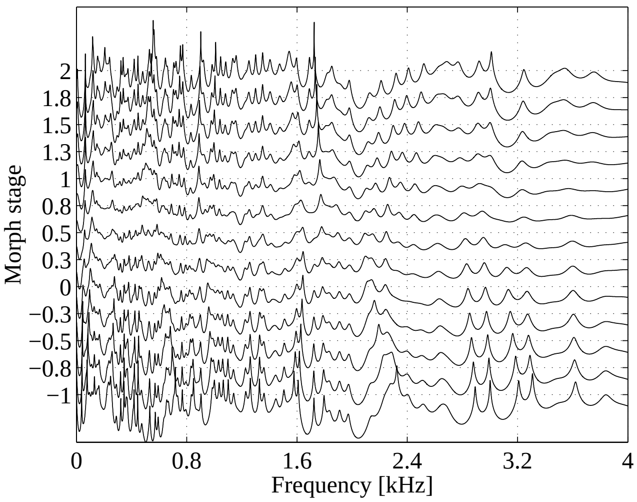

Figure 4: Extrapolation with LAR coefficients beyond existing body models. The original small and large sized body models are situated at zero and one (y-axis), respectively.

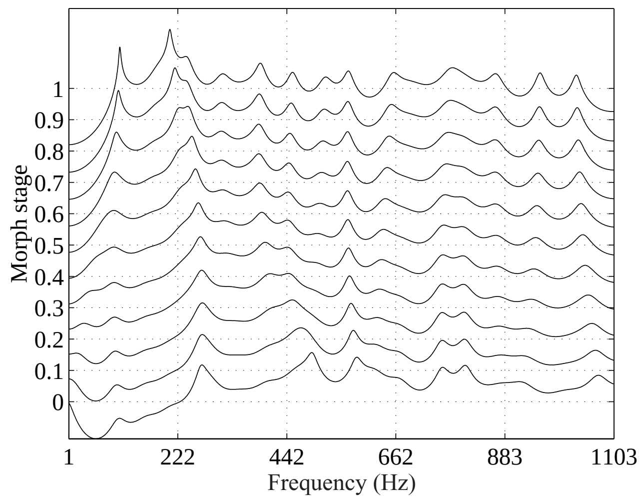

Figure 5: Morphing between a guitar and a violin body response. Figure shows eight stages between the bodies, so that 1 corresponds to the guitar and 0 to the violin.

to a violin body through a morph of two complete body models. Then, we consider a morph where the two lowest body modes are morphed separately from the rest of the body models. The single resonators are implemented as second-order IIR filters and the models representing the rest of the response are 100th-order WIIR filters. The morphs discussed in this Section are linear interpolation of RCs.

A morph between two complete body models is shown in Fig. 5, which zooms into low frequencies and displays eight morph stages between the guitar (1) and the violin (0) bodies. As mentioned before, the lowest modes in the guitar body are at considerably lower frequencies than in the violin body. Figure 5 shows how the lowest guitar body mode becomes weaker in magnitude, at each morph stage, and finally disappears. At the same time the second guitar body resonance changes in frequency and magnitude and becomes the first body mode of the violin.

A case, where the two lowest body modes are morphed separately from the rest of the body morph, is displayed in Fig. 6. The figure displays intermediate stages between the guitar (1) and violin (0) and zooms into low frequencies. The two lowest body modes have been extracted from the body responses with second-order notch filters. Each notch filter has two zero-pole pairs and a flat magnitude spectrum at other frequencies than the intended resonance. The inverse of the notch filter resynthesizes the extracted body mode exactly and each resonance filter is placed in cascade with the complete body model. The complete body model is created after the body modes have been extracted. See [17] for a more detailed discussion on extraction and resynthesis of instrument body modes. When comparing Figures 5 and 6 one can notice how in the latter figure the low body modes shift in frequency and change in magnitude and Q-values, while in Fig. 5 the changes occur more in the magnitude domain. Furthermore, the starting point responses are naturally the same, but the intermediate stages differ clearly. The audibility of the differences between these kinds of morphs depends on the number of independently morphed resonances, their frequency, magnitude and Q-value differences and finally also on the speed of morphing from a model to another. The lowest guitar body resonances ring longer and are located at lower frequencies. This can be seen from Fig. 7, where the morph of the lowest body mode is depicted by a zoomed view of the z-plane. Poles (x) and zeros (o) move to higher frequencies and move slightly away from the unit circle.

6. DISCUSSION

Now that some of the principle and realization issues related to instrument body morphing have been covered, it is natural to discuss aspects involved in use of the morphing concepts. Instrument body models composed of two or more parts enable the parts to be morphed independently. For example, when two cascaded WIIR body model filters have notably different morphing parameter, λ, values, their modeling ability is concentrated on different frequency regions, e.g., low- and high-frequencies. In this situation morphing or extrapolating them in a different manner, e.g., unequal morphing frequency or amplitude, gives more freedom to control the audio effect. Input signal dependent morphing is a possibility to create a controlled time-varying effect with some expression capabilities. On the basis of the level of the input signal, or amount of high frequency energy, the stage of a morph or extrapolation could shift to a chosen extreme with a desired rate and function. Morphing is a relatively general concept and it could be possible to apply the DSP ideas discussed in this paper to room responses and reverberation modeling, too.

7. CONCLUSIONS

In this work morphing between instrument body models was studied. In addition, extrapolation beyond existing body models is also found to be realizable. These interpolation and extrapolation techniques provide yet another musical audio effect. Audio examples related to this topic will are available at www.acoustics.hut.fi /demo/dafx2001-bodymorph/

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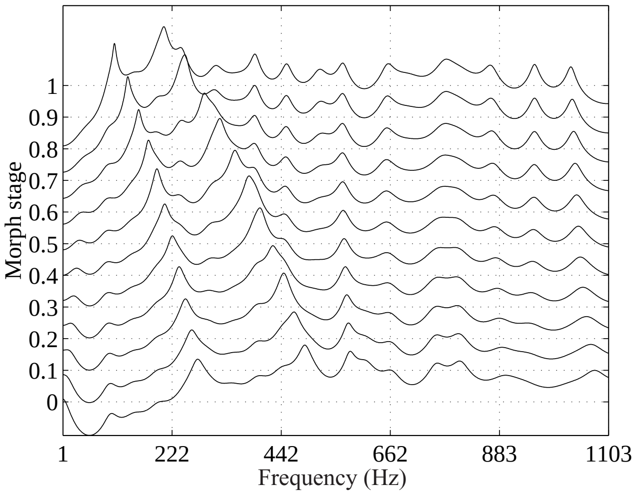

Figure 6: Morphing between a guitar (1) and a violin (0) body response, so that two low-frequency resonances are morphed separately.

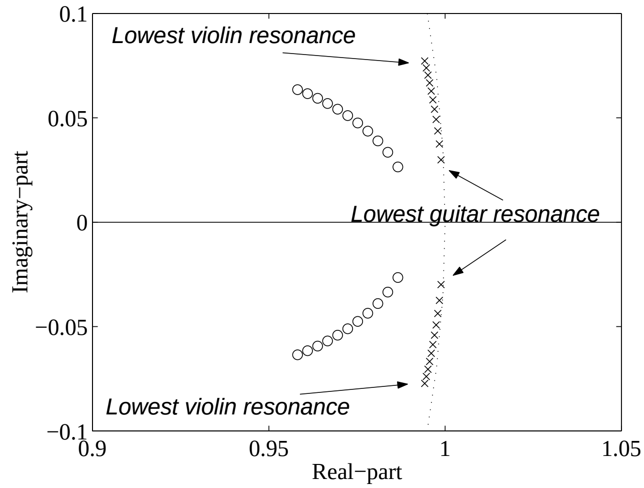

Figure 7: Discrete resonance morph between the lowest resonance of a guitar and a violin body. The figure zooms into low frequencies and displays the movements of the zeros (o) and poles (x).

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