EXCITATION SIGNAL EXTRACTION FOR GUITAR TONES

Nelson Lee
Stanford University
CCRMA

Zhiyao Duan
Tsinghua University
Dept. of Automation

Julius O. Smith III
Stanford University
CCRMA

ABSTRACT

This paper is concerned with extracting excitation signals from recorded plucked-string sounds from an acoustic guitar. Three pre-existing methods are reviewed: the matrix pencil inverse-filtering (MPIF) method by Laroche and Meillier (1994), the sinusoids plus noise inverse-filtering (SPNIF) method by Välimäki and Tolonen (1998), and the magnitude spectrum smoothing (MSS) method by Laurenti and De Poli (2000). The proposed method in this paper is based on removal of spectral peaks, followed by statistical interpolation to reconstruct the excitation spectrum in frequency intervals occluded by partial overtones. We call this the statistical spectral interpolation (SSI) method. The four methods are compared on synthetic data and data recorded from an acoustic guitar. Results show that the method presented in this paper outperforms previous methods in removing tonal components in the resulting excitation signal while maintaining a noise-burst like quality.

1. INTRODUCTION

Sound synthesis methods known as "subtractive synthesis" methods [1] are based on a *source-filter decomposition* (SFD), in which a sound is factored into an excitation signal that drives a filter. Electronic SFDs of speech signals date back to the *Vocoder*, developed at Bell Laboratories in the 1920s [2], and mechanical analogs go back even farther [2]. Another well known SFD is linear-predictive coding [1], also applied extensively to speech signals. In musical sound synthesis, SFDs are often used to synthesize percussion, piano, and plucked strings, to name a few. Calibration of digital waveguide models of stringed instruments [3] may also be formulated as the problem of estimating a source signal given the filter (a digital-waveguide string model) and the desired output signal. For acoustic stringed instruments, the commuted synthesis method may be calibrated in this way [3]. This paper is concerned with commuted-synthesis excitation estimation from recorded guitar sounds.

1.1. Related Work

In current digital waveguide guitar models [3], excitation signals may be theoretically-calculated plucks, filtered white noise, the first period of a recorded plucked-string tone, or inverse-filtered signals involving a recorded tone [4].

When recovering the excitation from recorded tones, comb-filters are often used to notch-out harmonic components of the signal [4, 5]. As the authors have noted, notching out all the harmonics leaves an excitation estimate that cannot excite the fundamental frequency and its harmonics in the string. It is therefore necessary to "fill in" the notches by some means. Since the excitation is a transient event, its spectrum is generally smooth, so the notches may be brought up to a level comparable to that of the surrounding spectral energy.

The two methods, matrix pencil inverse-filtering (MPIF) [5] and sinusoids plus noise inverse-filtering (SPNIF) [4], that use inverse-filtering for excitation extraction realize the problems of notches in the resulting residual, and address this concern in different ways, as summarized below.

1.2. Matrix Pencil Inverse Filter (MPIF)

In the MPIF method, developed primarily for piano, a single-source to multiple-resonators model is proposed for percussive sounds. This model assumes that adjacent notes of a percussive instrument are generated by a common excitation signal that excites a bank of resonant filters. Furthermore, this assumption is used to eliminate prominent notches by using a least-squares formulation for extracting a single excitation from multiple tones. The impulse response of the resonant filters of each note is made of exponentially damped sinusoids with the frequencies and damping factors corresponding to those of the harmonics in the recorded tone. Therefore, to model a recorded guitar tone from one string, two sinusoids are fitted for each harmonic, each sinusoid representing one of two orthogonal planes of motion for a vibrating string. The matrix pencil algorithm then computes the parameters of the sinusoids [6]. Since matrix inversion is needed, the signal has to be down-sampled to reduce matrix dimensions. Therefore, individual harmonics are fed to the matrix pencil algorithm instead of the whole signal to avoid aliasing when downsampling.

Furthermore, the MPIF method has to assume that tones used for extraction are played in a similar manner, to satisfy the assumption that the notes share a common excitation. One can see how this assumption would hold true for the piano, as the musician playing has no control over how the hammer hits the strings other than the striking velocity. However, in the case of the guitar, plucking is controlled much more freely. Furthermore, each pluck of the string is different. Reproducing the exact same pluck on a guitar by hand is extremely difficult, especially when a plectrum is used. Studies of guitar plucks and the exact reproduction of them have been explored and are complex and elaborate procedures. For those interested, we refer readers to [7] where the study of the physics of excitations, including the pluck of the classical guitar, is presented.

1.3. Sinusoids Plus Noise Inverse Filter (SPNIF)

The SPNIF method is based on prior work in modeling of the guitar that uses an extension of the Karplus-Strong (KS) Algorithm [3]. The method [4, 8] first decomposes the original signal into its deterministic and residual signals using the sinusoids plus noise modeling [9] approach. Inverse-filtering with respect to a string model (a digital waveguide model) is performed on both the residual and deterministic signals. The excitation signal is then acquired by adding the inverse-filtered residual signal and a windowed inverse-filtered deterministic signal, to compensate for the notches introduced from inverse-filtering.

In this method, the deterministic signal is generated using the sinusoidal function with the amplitude, frequency and phase estimated from the original signal. The residual signal is then acquired by subtracting the deterministic signal from the original signal in the time domain. It is well known (and we have confirmed) that subtraction-based methods are very sensitive to estimations of the fundamental frequency and phase. In fact, in use with real data, we found it extremely difficult to remove the deterministic signal entirely from the original.

1.4. Magnitude Spectrum Smoothing (MSS)

In more recent work, to avoid notches introduced by inverse-filtering, others have harnessed other approaches to spectral peak removal. The magnitude spectrum smoothing (MSS) method [10] performs spectral modification over a sliding FFT window. Within each frame of processing, the amplitude envelope in the frequency domain is twice-smoothed. Each amplitude point is adjusted to equal the average of its own and its neighbors' magnitudes. A second smoothing is then applied where the inverse of the amplitudes of the spectrum are passed through a median filter. The inverse amplitudes are then re-inverted to obtain amplitude values with peaks removed and all incidental-zero-amplitude terms removed for the residual signal. Since the method was presented for just estimating the amplitude envelope of the residual signal, changes were made to the algorithm to address actual excitation extraction. In our adaptation of this algorithm, we found it necessary to use a larger FFT size and an order of magnitude greater median filter to achieve comparable results.

1.5. Statistical Spectral Interpolation (SSI)

Similar to [10], the statistical spectral interpolation (SSI) method uses a sliding FFT window that only modifies the amplitudes, leaving the phase untouched, of the data in the frequency-domain. However, our method only changes amplitudes of points where partials occur. Amplitude values are changed to satisfy a Gaussian distribution, with mean and standard deviation equal to those of the amplitudes in the areas surrounding the peak, leaving points between partials untouched. As a result, the algorithm introduced in this paper replaces artificial nulls in the residual signal more locally and minimally using a statistically natural interpolation of the surrounding excitation spectrum.

We compare the performance of the four methods using both synthetic and real data recorded from an acoustic guitar. In extracting the synthetic excitations, the SSI method came closest to recovering the original waveform while suppressing tonal components. With regards to real data, the SSI method removed all tonal-components, whereas the MPIF and SPNIF methods had audible residual harmonics in the presence of natural fundamental frequency skew due to nonlinearity. The MSS method yielded an excitation with comparable harmonic removal, but lacked certain noise characteristics present in the original tone.

2. THE STATISTICAL SPECTRAL INTERPOLATION METHOD

A guitar pluck should be tone-independent and will typically be seen as a short noise burst. Therefore, the magnitudes of the original tone at the harmonic frequencies are greater than what they should be from only the attack.

From a high-level viewpoint, the SSI method only modifies the magnitudes of the STFT of the guitar tone without affecting phase information. The method introduced therefore collects statistics on the magnitudes of frequencies surrounding harmonic peaks and uses these statistics to generate non-deterministic gain-changes for the magnitudes at these peaks, without modifying the phase. From experience with previous methods, modifying phase will inevitably introduce artifacts. Our goal is to minimally-alter the original tone.

2.1. The Short-time Fourier Transform (STFT)

We use the STFT for analyzing and modifying the original recorded tone. The STFT can be seen as a sliding window that takes at each sample-window a Fast-Fourier Transform of the windowed signal. The transform of that windowed portion is then modified, and the inverse-Fast-Fourier Transform is then taken and saved in a buffer. The window is then slid according to how much overlap is wanted. The parameters for the STFT are the type of window used, the length of the window and the number of samples the window slides by. In our evaluations, we found a Hamming window of length 2^{12} samples with 0.9 overlap (hop size of 410 samples) to be satisfactory. Though 2^{12} samples with a sampling-rate of $44,100$ Hz is long (≈ 100 ms), we remedied pre-echo-distortion artifacts by starting the algorithm during the onset of the recorded tone. From our experiments, we have not found post-echo-distortion artifacts to be an issue.

Actual processing occurs at each window of the STFT.

2.2. Frame-level Processing

We consider each FFT window taken to be a frame for processing. Within each frame, we attenuate the harmonic peaks of the recorded tone. Harmonic peaks are found using the Quadratically Interpolated FFT (QIFFT) method [11].

Assume that the fundamental frequency of the recorded tone is at f_1 in Hz. We specify a bandwidth W_p in Hz, indicating the width of the peak. We specify a bandwidth W_n in Hz indicating the width of the interval that will be used for statistics collecting with respect to the fundamental frequency f_1. In using the SSI method, we had $W_p = 0.3 \cdot f_1$ and $W_n = 0.75 \cdot f_1$. We have found that these values ensure that the points used for statistics collecting do not reach into the next harmonic peak but are large enough to obtain a reasonable mean and standard deviation.

2.3. Harmonic-level processing

For each harmonic i with frequency f_i, the following is defined and used for processing.

We define the set of indices, Γ, whose frequency values satisfy the following:

$$\forall \gamma \in \Gamma, W_p \leq |\nu_\gamma - f_i| \leq W_n \tag{1}$$

where ν_k corresponds to the frequency in Hz of the kth FFT bin. The values in Γ correspond to indices within the current frame whose frequencies lie within the specified band W_n but outside the band W_p centered around f_i. See the circled points in Figure 1.

The mean and standard deviation of the magnitude of values in FFT bins in Γ are computed as follows:

$$\mu = \frac{1}{|\Gamma|} \sum_{i \in \Gamma} |X_i| \tag{2}$$

$$\sigma = \sqrt{\sum_{i \in \Gamma} (|X_i| - \mu)^2} \tag{3}$$

We now define the set of indices, Δ, whose frequency values satisfy the following:

$$\forall \delta \in \Delta, |\nu_\delta - f_i| \leq W_p \tag{4}$$

The values in Δ correspond to indices within the current frame whose frequencies lie within the specified band W_p centered around f_i. The magnitudes at these frequencies will be changed to remove the peak. See the starred points in Figure 1.

Thus, for all bins with indices in Δ, we modify their magnitude values to remove the observed peaks. This occurs as follows:

For each $\delta \in \Delta$, we generate a value $\rho \sim \mathcal{N}(\mu, \sigma)$.

$$X_\delta := \frac{\rho}{|X_\delta|} X_\delta. \tag{5}$$

Figure 1 shows the points the algorithm uses for statistics collecting and the points with gains altered.

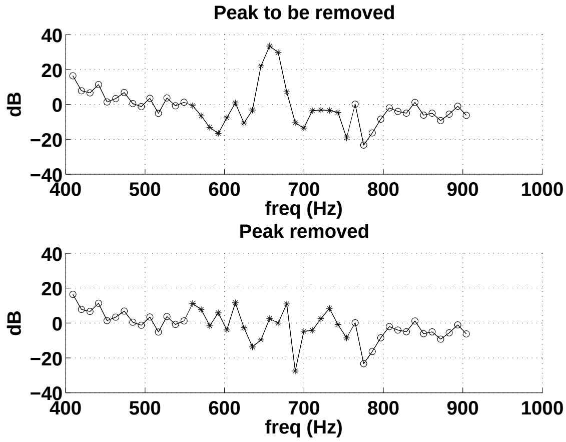


Figure 1. *Removing the peak at 660Hz: Starred points are FFT values to be changed. Circle dots are FFT values to be used for statistics collecting. Top plot shows pre-peak-removal. Bottom plot shows post-peak-removal*

3. EVALUATION

3.1. Synthetic Data

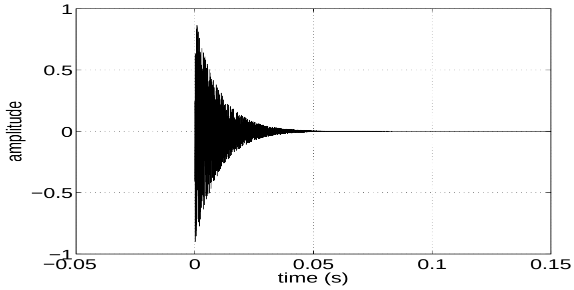
To compare the various methods in the literature, we synthesized guitar tones using a variant of the Karplus-Strong Algorithm [3]. The model consists of an all-pass-interpolating single delay-line loop. The delay-line had a linear-time-varying length to give a common effect, as is commonly observed in real recorded guitar tones, of the initial frequency being higher than that of the steady-state frequency as caused by the stretching of the string from the initial attack. The steady-state string frequency was set to 330 Hz, corresponding to the high 'e' of a guitar. The initial frequency immediately following the excitation was set to 333 Hz, a differential of 3 Hz. We have found in our measurements of recorded acoustic guitar tones that there is an initial pitch-shift that is subtle but present. In most cases, the pitch rises 5Hz above the steady-state frequency. The loop filter used was a simple one-zero filter:

$$H(z) = g\frac{1}{2}(1 + z^{-1}) \tag{6}$$

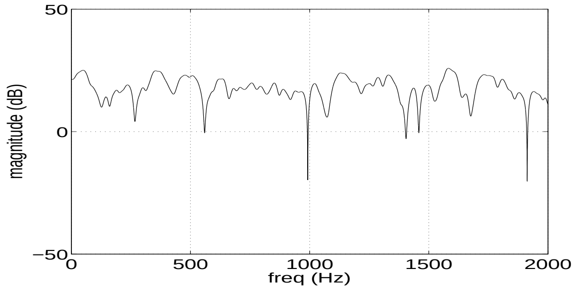
where g was set to 0.999. We used two different excitation signals to excite the model: an exponentially decaying white noise burst and an impulse. The excitation signals are shown in Figures 2(a) and 3(a), respectively.

We apply the four methods: the MPIF method, the SPNIF method, the MSS method and the SSI method to the data and compare how closely each method came in recovering the original excitation signals.

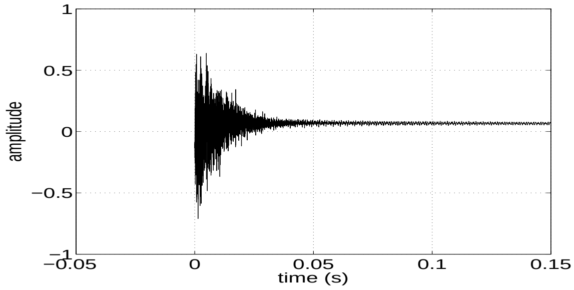
The results for the MPIF method are shown in Figures 2(c), 2(d), 3(c) and 3(d). In all cases, the time-domain signals visually resemble the original excitation signals used to generate the synthetic tones. However, as shown in the figures, harmonic components close to 660 Hz, 1 kHz and near 1320 Hz were not removed from the synthesized tone. This can be likely attributed to the frequency-



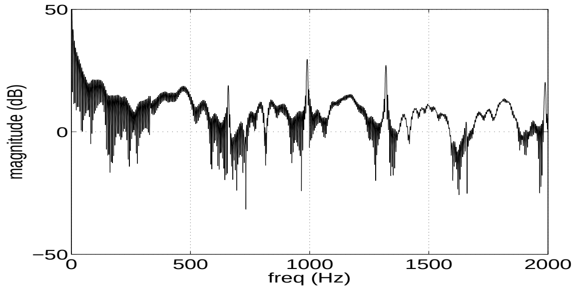
(a) Synthetic excitation signal. Time-domain amplitude plot.



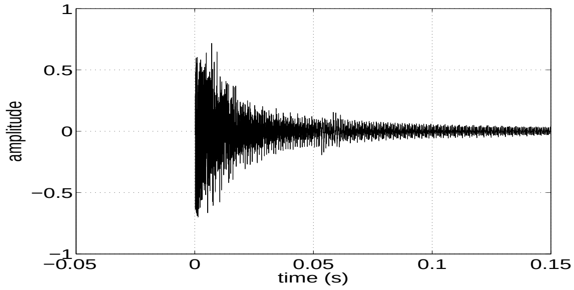
(b) Synthetic excitation signal. FFT magnitude plot.



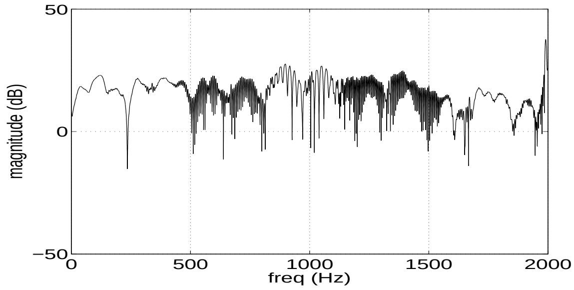
(c) MPIF extracted excitation. Time-domain amplitude plot.



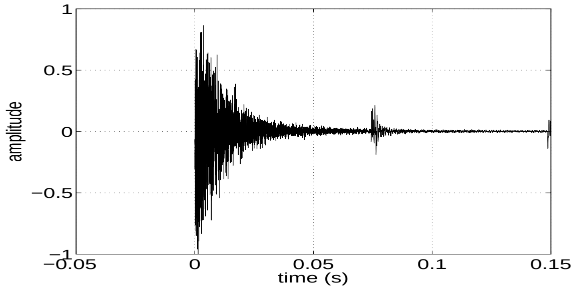
(d) MPIF extracted excitation. FFT magnitude plot.



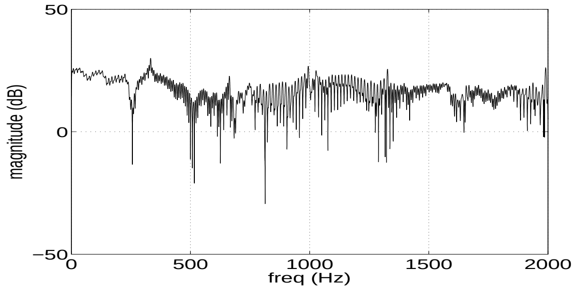
(e) SPNIF extracted excitation. Time-domain amplitude plot.



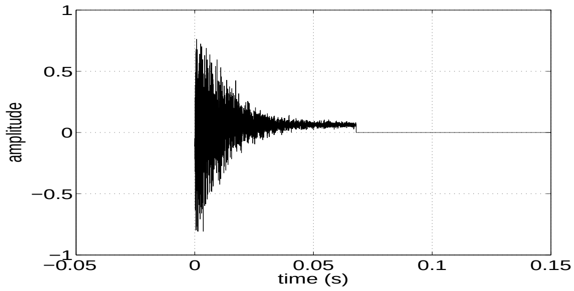
(f) SPNIF extracted excitation. FFT magnitude plot.



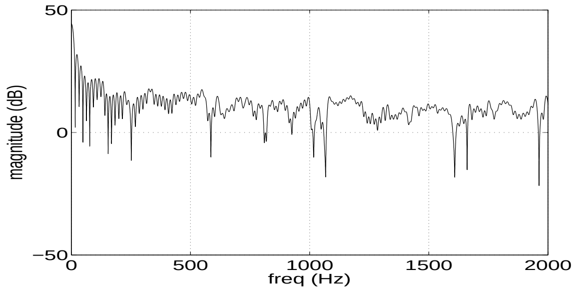
(g) MSS extracted excitation. Time-domain amplitude plot.



(h) MSS extracted excitation. FFT magnitude plot.

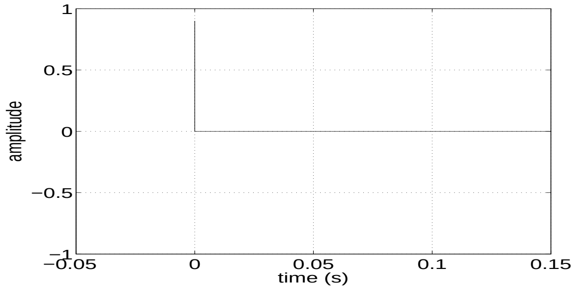


(i) SSI extracted excitation. Time-domain amplitude plot.

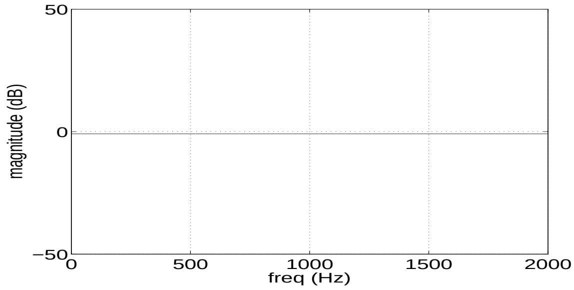


(j) SSI extracted excitation. FFT magnitude plot.

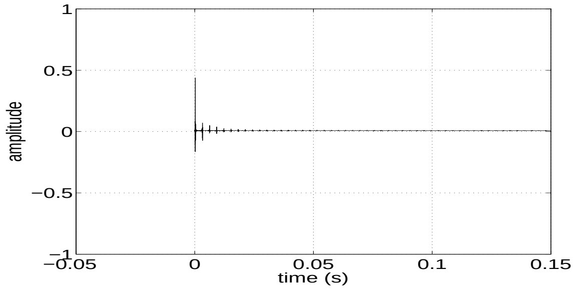
Figure 2. Results of excitation extraction using all methods are shown. The original excitation signal is an exponentially decaying noise-burst as shown in 2(a).



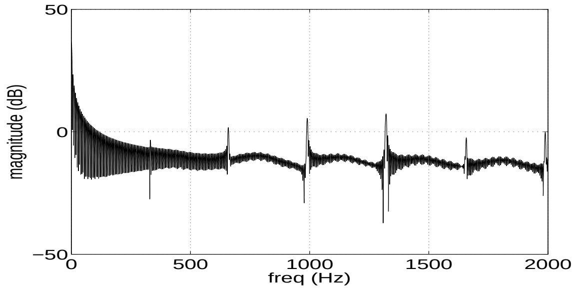
(a) Synthetic excitation signal. Time-domain amplitude plot.



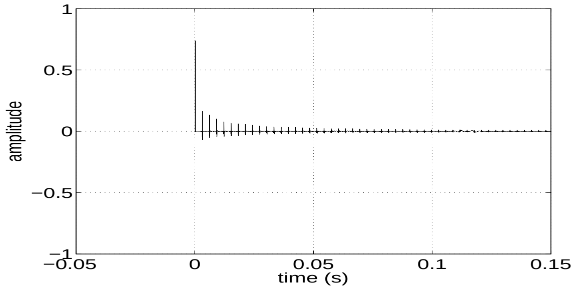
(b) Synthetic excitation signal. FFT magnitude plot.



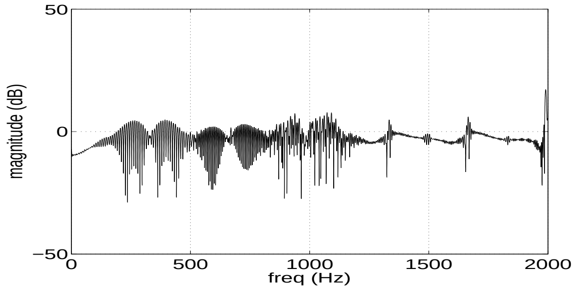
(c) MPIF extracted excitation. Time-domain amplitude plot.



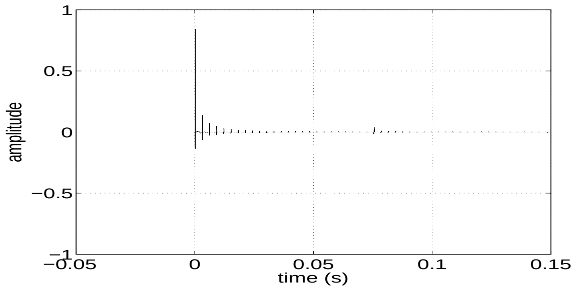
(d) MPIF extracted excitation. FFT magnitude plot.



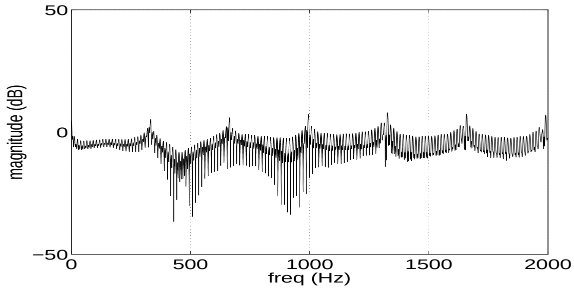
(e) SPNIF extracted excitation. Time-domain amplitude plot.



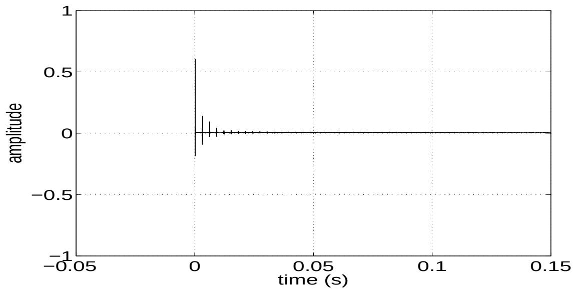
(f) SPNIF extracted excitation. FFT magnitude plot.



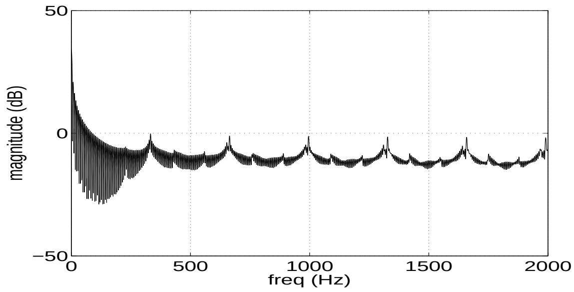
(g) MSS extracted excitation. Time-domain amplitude plot.



(h) MSS extracted excitation. FFT magnitude plot.



(i) SSI extracted excitation. Time-domain amplitude plot.



(j) SSI extracted excitation. FFT magnitude plot.

Figure 3. Results of excitation extraction using all methods are shown. The original excitation signal is an impulse as shown in 3(a).

skew in the original tones, which caused inaccurate estimation of parameters of the sinusoids of the model: the frequencies and damping factors. Furthermore, it can be seen from the frequency domain plot that artifacts reside in the excitation.

The results for the SPNIF method are shown in Figures 2(e), 2(f), 3(e) and 3(f). In the time-domain, the decay of the noise-burst is slower than that of the original tone. Since the SPNIF method uses inverse-filtering with a calibrated *string* model filter, the resulting excitation signals for both synthetic examples have significant audible artifacts as visible in the frequency-domain plots. Although harmonic components are not as visible as they are in the MPIF excitation signals, they are audible for both synthetic examples.

The results for the MSS method are shown in Figures 2(g), 2(h), 3(g) and 3(h). As the time-domain signal in Figure 2(g) shows, the excitation most closely resembles the original exponentially decaying noise burst thus far. However as Figure 2(h) shows, the method was unable to remove the fundamental. Therefore, in listening to the signal, there are audible artifacts related to the fundamental frequency. Furthermore, in the exponentially decaying noise case, much of the initial noise-burst is lost. For the synthetic impulse excitation, ringing of the fundamental is also present as harmonic peaks are not entirely suppressed as shown in Figure 3(h).

The results for the SSI method are shown in 2(i), 2(j), 3(i) and 3(j). In comparing its performance between the two synthetic examples, there are only harmonic components remaining in the near de-generate case where the excitation was an impulse. However, as Figure 3(i) shows, there are peaks at harmonic frequencies but at less than five dBs from the noise-floor, whereas in the MPIF method, they rise well over 10dBs above the noise-floor and in some cases, well over 20dBs above the noise-floor. Also, as the frequency-domain plots show, there are not many artifacts introduced into the spectrum.

3.2. Real Data

Using a tone recorded from an acoustic guitar, shown in Figure 4(a), using a condenser microphone, we applied all four algorithms for excitation extraction. The results are shown in Figure 4. In comparing the methods, zeros preceding the recorded tone were removed to prevent pre-echo-distortion artifacts.

The MPIF method removes harmonic peaks shown in Figure 4(a). However, there is a consistent notch in the resulting signal where the peaks previously existed. Audibly, the resulting excitation satisfies the criterion of not having harmonic components and resembling a noise-burst.

The SPNIF method was able to remove much of the harmonic content of the recorded guitar tone, but upon careful inspection of Figure 4(f), the peak at 1650 Hz is not entirely suppressed. In fact, harmonics above 1650 Hz are barely suppressed leaving tonal components in the excitation. The excitation signal extracted has audible ringing beyond that of the initial attack.

The MSS method is able to remove harmonic components better than the two previous methods. Furthermore, There is minimal ringing after the initial attack and much of the initial noise-burst is preserved. As the spectrum shows in Figure 4(h), there are no noticeable notches in the resulting signal's FFT magnitude plot. The extracted excitation has a noise-like quality to it, as expected from a guitar pluck, but maintains ringing during the initial onset.

The SSI method introduced in this paper removes all peaks shown in Figure 4(a). Furthermore, there is no noticeable artifact related to the fundamental frequency both visibly in the frequency-domain nor audibly. The noise characteristics of the extracted excitation are also preserved and match those of the original tone.

The results of the excitation extraction using all four methods for both synthetic and real data can be found at http://ccrma.stanford.edu/realsimple/icmc07_results/.

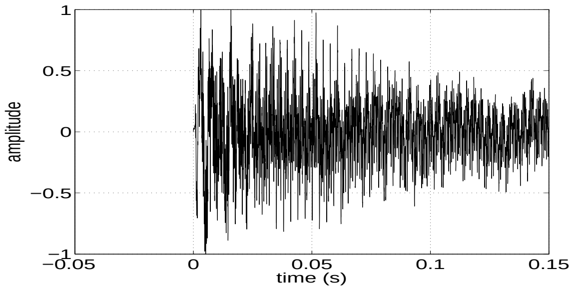
4. ANALYSIS

The methods reviewed in this paper were created with specific applications in mind. The MPIF, SPNIF and SSI methods were all created with the specific intent of obtaining a high-fidelity excitation signal for instruments for synthesis purposes (specifically, piano for MPIF, and guitar for SPNIF and SSI). The MSS method's main purpose, on the other hand, is to estimate and fit a closed-form-function of the residual spectral magnitude of a harmonic signal below its harmonic peaks. Therefore, in using this method for actually obtaining the residual signal in the time-domain, the method had to be adapted: increasing the size of the FFT buffer used and increasing the order of the median filter described in their algorithm.

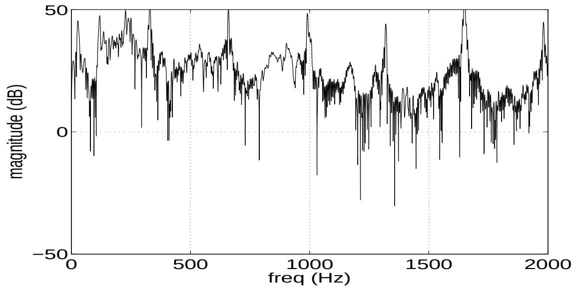
Comparing the efficacy of the MPIF, SPNIF, MSS and SSI methods, each has its own strengths and flaws. In using the matrix pencil algorithm, the amplitude, frequency and phase of each harmonic peak is computed. Using such a model made inverse filtering extremely effective when the fundamental frequency is constant. Furthermore, since a harmonic peak can be modeled by an arbitrary number of sinusoids, the MPIF method can directly model beating due to coupling effects, and we found it to be more impervious to fundamental-frequency skew, as was present in the synthetic data. However, a significant drawback to using the matrix pencil method is that it is relatively computationally expensive.

The SPNIF method approaches excitation extraction from a KS guitar synthesis perspective. Since the SPNIF method assumes its synthesis model for excitation extraction, if the original tone to be processed differs from its model, the extracted excitation will be left with artifacts and tonal components. In both the synthetic and real data, since pitch-shifting was present, their method left artifacts and ringing after estimation and removal of tonal components.

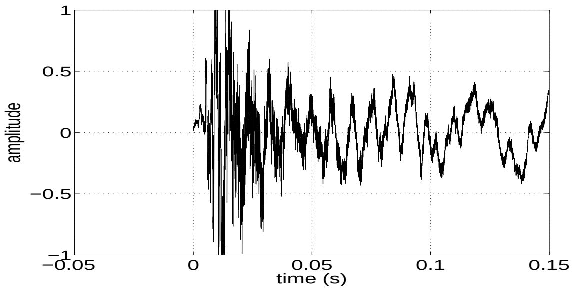
The MSS method presents an alternative approach to inverse-filtering. Its use of a sliding FFT window allows it to perform careful harmonic peak-tracking while process-



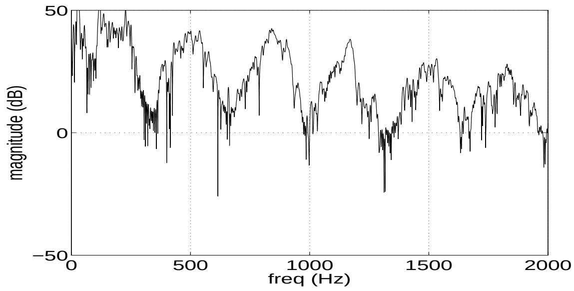
(a) Real recorded guitar tone. Time-domain amplitude plot.



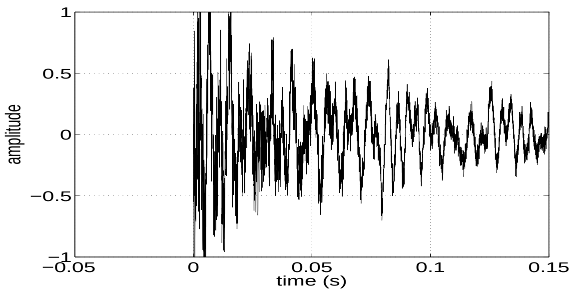
(b) Real recorded guitar tone. FFT magnitude plot.



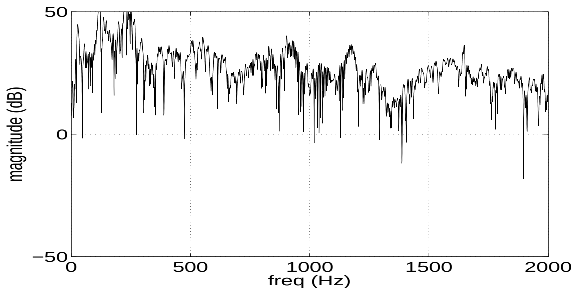
(c) MPIF extracted excitation. Time-domain amplitude plot.



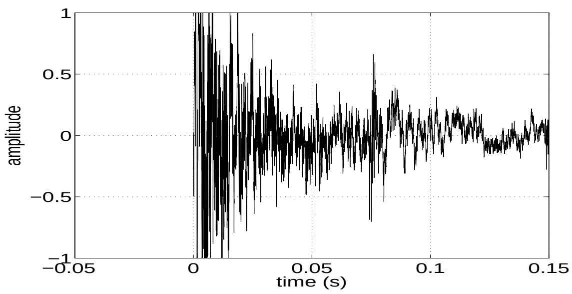
(d) MPIF extracted excitation. FFT magnitude plot.



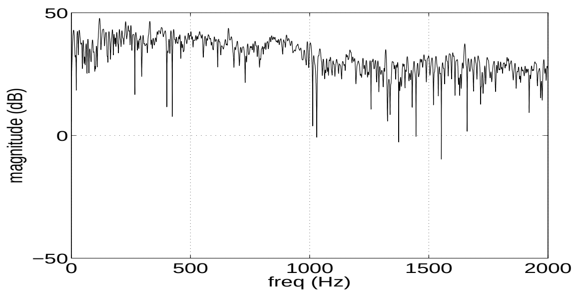
(e) SPNIF extracted excitation. Time-domain amplitude plot.



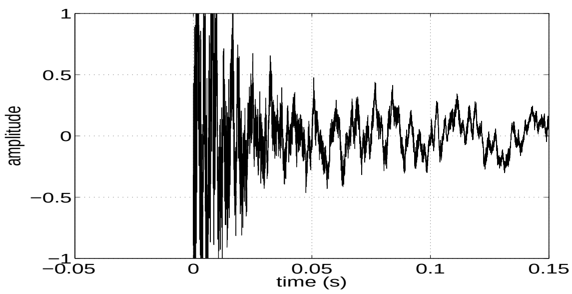
(f) SPNIF extracted excitation. FFT magnitude plot.



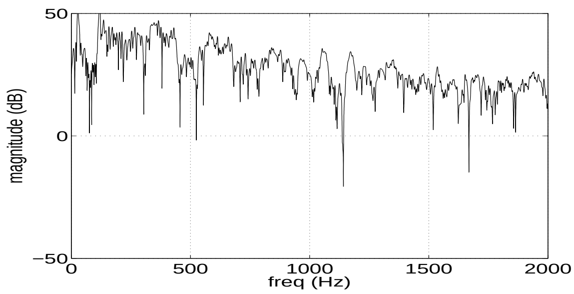
(g) MSS extracted excitation. Time-domain amplitude plot.



(h) MSS extracted excitation. FFT magnitude plot.



(i) SSI extracted excitation. Time-domain amplitude plot.



(j) SSI extracted excitation. FFT magnitude plot.

Figure 4. Results of excitation extraction using all methods on a recorded guitar tone are shown.

ing and changing the magnitudes of the harmonic components without affecting the signal's phase. The method, however, when adapted for use for excitation extraction became computationally heavy. Running the median filter the size of the FFT times per each FFT window taken made using this algorithm time-inefficient.

The SSI method presented in this paper is able to extract excitation signals without harmonic components in them. Furthermore, the excitation signals maintain a noise-burst-like quality. Since only the original tone's amplitudes are modified, and only the frequencies at which overtones occur are modified, most of the resulting excitation signal is true to the original tone, in that since the original tone is not inverse-filtered, and the phase of the original tone is entirely untouched and only a subset of the spectral magnitudes are changed. Furthermore, since no inverse-filtering is involved with the SSI method, there are several added benefits. As the other methods have shown, fitting the right filter for peak removal is intricate and computationally heavy. Furthermore, inverse-filtering is sensitive to harmonic signals with changing peaks. Therefore, the SSI method is more robust and invariant to peak changes as shown in Section 3. However, the MPIF and SPNIF method offer benefits the SSI method lacks. A deterministic/tonal component is not estimated nor produced in the SSI method, and similarly with the MSS method.

Overall, The SSI method presented in this paper produces a better excitation signal as defined previously, in that artifacts with respect to the fundamental are less apparent, the resulting excitation signal is toneless and the resulting excitation resembles a noise-burst. Furthermore, the MPIF method is computationally more intensive, in that it needs multiple notes to extract a notch-less excitation, whereas with the SSI method, a single-recorded note can produce a satisfactory excitation signal. The SPNIF method is computationally lighter than the MPIF method, but it produced somewhat inferior excitation signals compared to the MPIF and SSI methods in our tests. The MSS method compared comparably to the SSI method, but failed in maintaining all the noise characteristics of the original signal and in removing all harmonic peaks in the original tone. Furthmore, the MSS method suffers from conflicting objectives: lowering of the height of the spectral peaks with averaging versus maintaining the original spectrum of the signal without averaging.

5. CONCLUSIONS

In this paper, the proposed Statistical Spectral Interpolation (SSI) method was compared with three previous methods from the literature on the problem of extracting excitation signals from recorded acoustic guitar tones. Results for the four methods were measured on both synthetic data and real data. The four methods compete comparably, but the SSI method gave an excitation signal that was more "toneless" and sonically similar to the excitation noise burst used in the original. Moreover, among the four, SSI was found to be more robust and simple to use.

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