CONTROL OF INTERHARMONIC BEATING IN POLYPHONIC MUSIC

Robert C. Maher

Digital Audio Signal Processing Laboratory
Department of Electrical Engineering
University of Nebraska-Lincoln
209N WSEC
Lincoln, NE 68588-0511
(402) 472-3771
eerdrcm@engvms.unl.edu

Abstract

Amplitude beating between closely spaced frequency components is a well-known effect in musical acoustics, psychoacoustics, and other fields [1-3]. Depending on the musical context and the personal preference of the listener, the presence of amplitude beating can either be an undesirable artifact of the limited frequency resolution of the human hearing apparatus, or a pleasant quality that adds timbral variety to an ensemble performance. In either case it would be useful to be able to control the extent of interharmonic amplitude beating in some convenient manner.

The increased use of digital computing systems in music synthesis and post-production opens up many new avenues for innovative digital signal processing. This paper extends the repertoire of digital audio signal processing methods to include direct control over amplitude beating in complex audio signals due to interaction among spectral components of simultaneous musical voices. Applications of this technique include 1) discriminability improvement for weak or easily masked musical voices in complex sonic textures, and 2) alteration of the consonance/dissonance relationship of musical intervals and chords to retain the advantages of equal tempered tuning (for example, modulation between keys) while reducing the effects of out of tune partials.

Overview

As previously reported [4], one means to reduce amplitude beating during additive mixing operations is to perform a time variant spectral analysis on the signals to be mixed, identify the presence of closely spaced frequency components, and selectively attenuate those components which will give rise to amplitude beats. A convenient formulation for this procedure was found to be the sinewave model of McAulay and Quatieri [5]. This approach can be described as *exclusion filtering*, where one of the signals to be mixed is used to design a time varying comb-like filter to exclude competing spectral energy from the other signals.

The amplitude beating among closely spaced partials can also be *increased* to improve the detectability of a relatively weak musical voice in the presence of a complex background ensemble. The increased beating is accomplished by using time variant sinusoidal analysis to identify spectral *collisions* among the competing musical voices and then to increase the amplitude of the beating components. This technique is particularly useful when the weak voice has spectral energy in a confined range which overlaps the background material, e.g., a solo clarinet with string ensemble accompaniment. While simply boosting the level of the weak voice can improve its detectability, the combination of increased level *and* enhancement of interharmonic beating can increase the perceived separation between the weak signal and its competition. In other words, the presence of the weak voice is *cued* by its effect upon the other voices in the ensemble.

Discussion

Methods to adjust interharmonic beating can be applied to pre-recorded musical material as a post-processing step, or as pre-calculated spectral constraints for use in software synthesis algorithms.

The post-processing methods include the use of exclusion filters during mixing (as described above), and the identification of interharmonic beating in polyphonic recordings using short-time spectral analysis. We have developed several automatic methods to detect interharmonic beating in complex signals and further work is in progress to attempt to separate the inherent amplitude fluctuations in most musical voices (due to vibrato, tremolo, or natural variability) from the amplitude beats caused by interharmonic collisions.

Pre-processing methods can be applied to software synthesis algorithms if control over the spectral content of the synthesized signals is sufficient to allow individual amplitude adjustments of each partial. This requirement can be achieved easily in an additive synthesis situation, but the incorporation of special notch filters into an arbitrary synthesis algorithm can provide the same level of control--perhaps at the expense of a rather unwieldy design topology.

The results of these interharmonic beating adjustment methods have been quite good under informal evaluation. Suppression and enhancement of the beating components can be controlled in both rate and extent, but the processing has been found occasionally to introduce audible artifacts at transitions from one *steady state* signal segment to another, such as when a note transition occurs in one voice while other voices are sustained. This minor difficulty is due to the one-pass nature of the current processing implementation: a second pass or backtrack strategy could automatically identify and repair the offending transitions by observing the changes in the spectral distribution of the signal.

Conclusion

Interharmonic beating caused by the closely spaced frequency components has been the subject of considerable debate among musicians and composers and the subject of research into alternative scale tuning systems. The DSP methods described here are able to suppress or enhance the presence of beats by identification of spectral collisions using a sinusoidal analysis framework. This approach has been applied to additive mixing, post-processing, and signal synthesis situations with notable success. Additional work is required, however, to improve the latency of the process during rapidly changing musical passages.

References

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