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Phase Mitigation Through Filter Design

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ABSTRACT

In both acoustic and digital systems, delays and the resulting phase interference are an innate feature of sound recording; traditionally, phase-interference mitigation is applied through temporal offset to attempt time coherence between multiple signal paths. Filter design presents an alternative solution to phase issues, wherein predictive modeling allows for a filter to apply corrective magnitude response. Such application of filter design presents its own set of problems and could further be explored in creative, rather than remedial, settings.

1 Problem

In the scenario depicted by Fig. 1, a sound originating at source S_1 would arrive at transducer X_1 before arriving at transducer X_2. Similarly, sounds originating at source S_2 arrive at transducer X_2 before arriving at transducer X_1. Consequently, the addition of the signals captured by transducers X_1 and X_2 in processing and subsequent playback results in phase interference and cancellations along frequency bands related to the amount of distance between the sources and transducers.

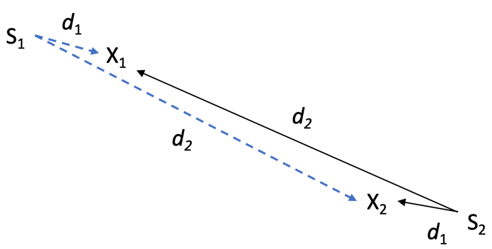

Fig. 1, depicting sources S_1 and S_2, transducers X_1 and X_2, and direct paths d_1 and d_2.

2 Other Factors

Fig. 1 represents a theoretical environment in which d_1 and d_2 represent respectively equal distances between sound sources and their intended and incidental transducers, and in which there are no additional acoustic reflections. The delay (Δ_t) in arrival times of sounds from source S_1 to transducers X_1 and X_2 in such a theoretical acoustic system may be derived from the distances and speed of sound (c) as shown in Equation 1.

$$\Delta_t = \frac{(d_2 - d_1)}{c} \tag{1}$$

The exact nature of consequent phase interactions depends on a number of factors, including (but not limited to) the frequency response of transducers, frequency content of the sound itself (and the effects on frequency content of any reflective room surfaces), distance between transducers, and magnitude at which each signal is captured and later reproduced in the end playback system (in filter design, this could be called the gain coefficient).

Fig. 2 visualizes the results of adding signals from transducers X_1 and X_2 together, with respect to sounds from sources S_1 and S_2.

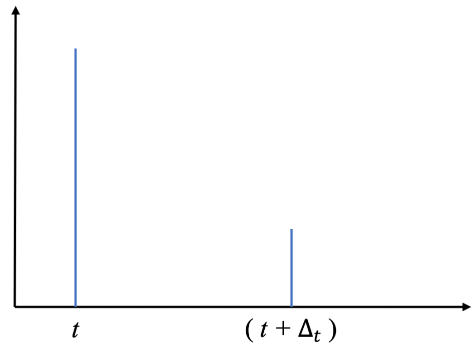

Fig. 2, showing relative arrival times (x-axis) of signals from S_1 and S_2.

Even with $d_1 = d_1$ and $d_2 = d_2$ in the theoretical system from Fig. 1, signals arrive at transducers at different times; signals reach the closer transducer at time t after traveling distance d_1, and they arrive at the farther transducer at time $(t + \Delta_t)$ after travelling d_2. In an acoustic system, signals would lose energy after traveling a further distance; this has been approximated for illustrative purposes along the Y-axis, which would be analogous to the magnitude of a captured signal.

3 Existing Workflows

Traditional phase mitigation typically means delaying one signal backwards (or forwards in a recorded DAW setting) by a number of milliseconds or samples derived from the distance between transducers and the speed of sound; while this method of delay compensation is rooted in the fundamental acoustics of sound travel and capture, its implementation is limited in circumstances in which multiple sounds in a space are being captured by multiple transducers in various relative positions. In application, this might mean that correcting the phase of the snare drum means further damaging the phase of the piano.

In music recording, louder and more improvisatory musical styles may require simultaneous capture of multiple musicians in the same space, and even with acoustic buffers complete isolation is often unattainable- most certainly in ‘live’ recording. This means that transducer ‘bleeding’ and the consequent phase interference are almost intrinsic artifacts of such recording.

Automatic bleed-reduction and noise-reduction filters also present a useful means for mitigating phase interference between multiple transducers; however, such applications can be signal-dependent and require constant analysis to adequately mitigate sound bleed of a real signal (e.g. one that changes frequency regularly, such as a musical instrument). Such filters could be useful in attempts to isolate a particular channel for the purpose of applying further processing to the desired contents of the channel.

Sound capture and reproduction takes many forms, and mitigation of phasing can also take multiple forms; an alternative to traditional temporal offset is presented here, in which phase is treated using filter design instead of time offset. Filter design is not intended or postulated to supplant temporal offset or commercial bleed-reduction/noise-reduction options, but rather such a phase-mitigating filter, when added to the engineer’s ‘toolkit,’ can make for new solutions to old phasing problems, or at least give the engineer a new set of tools for creative use.

4 Mitigation through Filter Design

Time delay between two identical transducers results in a close relative to inverse comb filtering, in which a fundamental frequency with wavelength

$$\lambda = \frac{(d_2 - d_1)}{2} \tag{2}$$

and the odd harmonic multiples of that fundamental tend to be reduced in magnitude response; and, the even-numbered harmonic multiples of that fundamental frequency reinforce and increase in magnitude. Since the response of such a system is predictable (in a sense) by modeling the time delay and relative loudness levels of each signal, a filter may be constructed to implement a corrective set of increases and decreases in frequency content. Such a filter presents the benefit of avoiding the introduction of new phase problems when one channel is offset in a DAW (such as the instance in which fixing the snare channel further disrupts the piano channel), and it may help engineers avoid making decisions requiring that one signal be prioritized over another in shifting tracks around for time coherence.

Since the distance between transducers (or time between signals in a digital environment) does not necessarily depend on a signal, a filter derived from the distance between transducers has potential for relative stability, when compared against signal-dependent filters designed to change response based on the frequency content of an incoming signal.

References

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