Colin Fahy
3D Audio
Dr. Agnieszka Roginska
2016-04-16

Crosstalk Cancellation though Recursive Inversion, Delay and Attenuation Based on Speaker and Listener Location

Introduction

This implementation of crosstalk cancellation is based on the Recursive Ambiophonic Crosstalk Elimination (RACE) (Glasgal, 2007) with added optimisation for specific speaker and listener setups. Given a listener's head radius, their distance from the speakers, and how far apart the speakers are, the distance sound travels from one speaker to each ear can be computed. These two distances can be used to compute the interaural time difference (ITD) and estimate the interaural level difference (ILD). Given these values, more accurate time delay and attenuation can be applied using the RACE algorithm.

Implementation

The RACE algorithm aims to cancel the signal traveling from one speaker to the contralateral ear by sending an inverted version of the signal from the other speaker. This signal must be attenuated and delayed in time to account for head-shadow and the extra distance traveled. However, this cancellation signal will itself cause crosstalk back to the other ear. For this reason, the same process is applied recursively on each cancellation signal produced. Thankfully, due to the attenuation built into the process, the crosstalk will eventually become negligible and the recursion can stop.

The original RACE algorithm used delay values of 60-100μs and attenuation of 2-3dB, and these values were adjustable to suit the listener. However, these values can be better estimated by taking into account the specific speaker and listener positions. Given the difference in distance traveled from one speaker to the ipsilateral and contralateral ears, an accurate ITD value can be computed and used as the time delay.

The ILD can be estimated according to the inverse distance law and a low-pass filter controlled by the azimuth angle to represent the head-shadow effect.

Distance Calculation

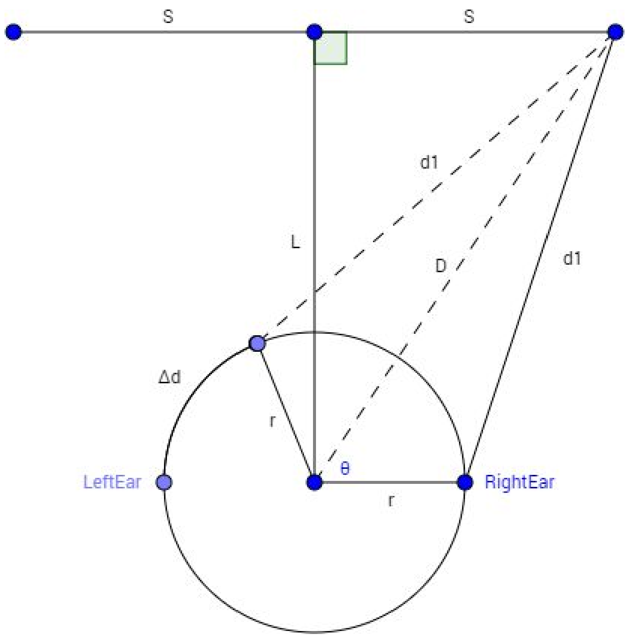

Fig 1. The geometry used to compute Δd, the difference in distance traveled from one speaker to the ipsilateral and contralateral ears.
S is the distance from one speaker to the midpoint, L is the distance from the midpoint to the center of the listener's head, r is the radius of the listener's head.

The difference in distance traveled from the speaker to each ear can be computed according to the geometry in Fig 1. The system assumes the listener is situated directly along the symmetry plane and is facing the midpoint of the speakers. Given the values for S, half the distance between the speakers, L, the distance from the listener to the midpoint of the speakers, and r, the radius of the listener's head, the values for d1 and d2 (the distance to the ipsilateral and contralateral ear respectively) can be computed.

$$d1 = \sqrt{L^2 + (S - r)^2}$$

$$\theta = cos^{-1}(S / \sqrt{L^2 + S^2})$$

$$\Delta d = r * (\pi - 2\theta)$$

$$d2 = \Delta d + d1$$

The azimuth angle from the center of the head to the speaker can also be computed as $tan^{-1}(S / L)$, this will be used for estimating the coefficients to the head-shadow filter.

ITD and ILD Calculation

The ITD is simply the difference in distance traveled, Δd, divided by the speed of sound, c, which is approximated to 343 m/s. This time delay is applied to the signal by multiplying by the sample rate and prepending with that many zeros. Since the delay is a very small amount, it is worthwhile to use fractional delay rather than round to the nearest integer sample. Fractional delay is implemented using linear interpolation.

The attenuation factor can be estimated according to the inverse distance law by dividing the distance d1 by d2. The head-shadow effect can be estimated as a low-pass filter relative to the azimuth angle as has been implemented by Udo Zolzer (Zolzer, 2011). Together, this filter along with the attenuation factor make up a rough estimation of the frequency dependant ILD.

Recursive Cancellation

For each channel of the binaural input signal, the signal is inverted, delayed by the ITD, attenuated and filtered by the estimated ILD to create a cancel signal. The same processing can then be done using this newly created cancel signal as the input signal. This process is repeated recursively until the cancel signal is -70dB with respect to the original signal. All of these cancel signals are summed together, alternating channels, and added to the the left and right channels of the original signal.

Conclusion and Future Work

After preliminary testing, the system performs very well with speakers that are fairly close together (approximately 12 inches apart). Not only does it widen the soundstage, but it can even induce the sensation that the sound is coming from behind the listener, this is particularly exhibited in the files knocking_from_behind.wav and virtualhaircut.wav.

This method is still quite restricting in that the listener must remain at a stationary location and distance between the speakers. The geometry calculations could be extended to compute the distance to each ear of a listener at any location with only the addition of either the angle from the speaker midpoint, or the distance from the symmetry plane. Along with a head-tracking system, this would allow for real-time optimisation of the ITD and ILD values to provide better crosstalk cancellation for a single moving listener.

References

Cecchi, S., Primavera, A., Virgulti, M., Bettarelli, F., Li, J., & Piazza, F. (2014, July). An efficient implementation of acoustic crosstalk cancellation for 3D audio rendering. In Signal and Information Processing (ChinaSIP), 2014 IEEE China Summit & International Conference on (pp. 212-216). IEEE.

Choueiri, E. Y. (2008). Optimal crosstalk cancellation for binaural audio with two loudspeakers. Princeton University, 28.

Glasgal, R. (2007, October). 360 localization via 4. x race processing. In Audio Engineering Society Convention 123. Audio Engineering Society.

Lee, K. S., & Lee, S. P. (2010). A real-time audio system for adjusting the sweet spot to the listener's position. Consumer Electronics, IEEE Transactions on, 56(2), 835-843.

Zölzer, U., & Arfib, D. (Eds.). (2011). DAFX: digital audio effects (Vol. 1). (pp. 155-157) Wiley.