

# A Novel Crossfeed Audio Plugin For Improved Headphone Listening Experience

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This paper describes the theory and implementation of a novel audio plugin designed to enhance the headphone listening experience. Listening through headphones lacks the natural crosstalk between the listener's ears that occurs when using a standard loudspeaker setup. The interaural transfer function at an azimuth angle of  $30^\circ$  and an elevation angle of  $0^\circ$  is approximated using a combination of a second-order minimum phase filter and a delay line. This approximated interaural transfer function is implemented in the crossfeed path of the plugin. As a result, the spatial perception is more realistic compared to existing solutions. Additionally, a novel equalizer allows for a seamless transition between full mono compatibility and minimal sound color changes for typical stereophonic signals.

The plugin is available for free on the author's website. It is a JSFX plugin and operates natively in the digital audio workstation Reaper. Furthermore, with the assistance of the YSFX bridge plugin released by Joep Vanlier, it can run on nearly every audio plugin host. YSFX is also available for free as a VST3, AU, or CLAP plugin for Windows, Mac OS, and Linux.

## 1. Theory

The European Broadcasting Union recommends placing the loudspeakers of a stereo system at azimuth angles  $a$  of  $-30^\circ$  for the left speaker and  $+30^\circ$  for the right speaker, directly in front of the listener. The recommended elevation angle  $e$  is  $0^\circ$ , and the ideal distance  $r$  from the listener is between 2 to 4 meters [1].

The sound from the two speakers reaches both ears, but with different head-related transfer functions (HRTFs). If we disregard any room reflections and assume symmetrical heads, we can derive:

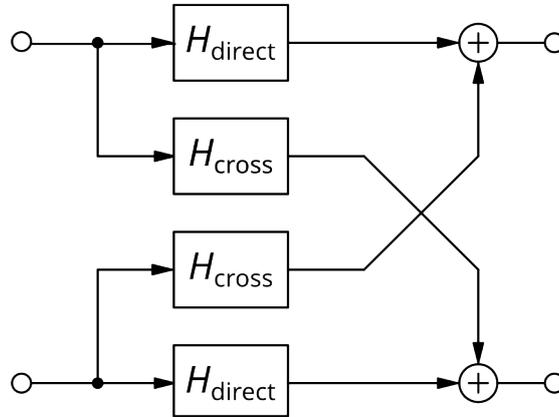
$$H_{\text{direct}}(f) = \text{HRTF}_{\text{left ear}}(a_{\text{left}}, e, r, f) = \text{HRTF}_{\text{right ear}}(a_{\text{right}}, e, r, f) \quad (1)$$

$$H_{\text{cross}}(f) = \text{HRTF}_{\text{left ear}}(a_{\text{right}}, e, r, f) = \text{HRTF}_{\text{right ear}}(a_{\text{left}}, e, r, f) \quad (2)$$

$$H_{\text{inter}}(f) = \frac{H_{\text{cross}}(f)}{H_{\text{direct}}(f)} \quad (3)$$

The interaural transfer function,  $H_{\text{inter}}$ , is essential for the spatial perception, while  $H_{\text{direct}}$  can be viewed as a component of an equalizer that influences the sound's color. If we aim to emulate the HRTFs in a crossfeed circuit for headphones, as demonstrated in **Fig. 1**, we need to implement  $H_{\text{inter}}$  as realistically as possible. However,  $H_{\text{direct}}$  should be selected in such a way that mono compatibility is primarily maintained. This means that a monophonic signal should not be

significantly altered by the crossfeed circuit. The rationale behind this approach is that we can assume good headphones are already equalized so that monophonic signals sound natural.



**Fig.1:** Crossfeed circuit for headphones

The transfer functions for monophonic and side signals are:

$$H_{\text{mono}}(f) = H_{\text{direct}}(f) + H_{\text{cross}}(f) \quad (4)$$

$$H_{\text{side}}(f) = H_{\text{direct}}(f) - H_{\text{cross}}(f) \quad (5)$$

We are now redefining:

$$H_{\text{direct}}(f) = \frac{1}{1 + k \cdot H_{\text{inter}}(f)} \quad (6)$$

$$H_{\text{cross}}(f) = \frac{H_{\text{inter}}(f)}{1 + k \cdot H_{\text{inter}}(f)} \quad (7)$$

With the redefinition of  $H_{\text{direct}}$  and  $H_{\text{cross}}$ , we retain the original  $H_{\text{inter}}$ , allowing us to smoothly transition between no change in the direct signal and full mono compatibility. A blending parameter  $k$  of zero yields a flat  $H_{\text{direct}}$ , while a value of one produces a flat  $H_{\text{mono}}$ .

We may also define a transfer function  $H_{\text{ind}}$  based on independent left and right signals. We suppose that the amplitude spectra of the left and right input signals are the same, but the phase spectra differ at random. Two uncorrelated pink noise signals serve as an example of such a signal pair.

$$H_{\text{ind}}(f) = \sqrt{|H_{\text{direct}}(f)|^2 + |H_{\text{cross}}(f)|^2} \quad (8)$$

Since stereo recordings contain a certain amount of independent left and right audio, the blending parameter  $k$  should be set so that both  $H_{\text{mono}}$  and  $H_{\text{ind}}$  are reasonably flat.

## 2. Some milestones in the history of crossfeed circuits

Benjamin B. Bauer proposed the first crossfeed circuit for headphones. He was granted a US patent [2] for it in 1963. Bauer attempted to approximate the HRTFs for azimuth angles of +/- 45°. He

used minimum phase filters in his method, resulting in unrealistic interaural delay times at high frequencies. The magnitude of the interaural transfer function was too low for frequencies exceeding 4 kHz. The circuit was completely mono compatible.

Sigfried Linkwitz [3] designed a substantially simpler circuit in 1971. For the crossfeed path, a first order lowpass filter with a low frequency gain of -5 dB and a pole frequency of 700 Hz was employed, whereas the direct path used a first order lowshelf filter with a low frequency gain of -2 dB. His circuit had the same issues as Bauer's circuit, but the magnitude of the interaural transfer function was too small, even at low frequencies. The circuit was not mono compatible.

Around 2005, Boris Mikhaylow released the VST plugin BS2B [4]. He used the same circuit as Linkwitz, but adjusted the parameters. He used lowpass gains of -6.75, -8.0, and -10.92 dB in his three settings (1, 2, and 3). The lowpass filter had pole frequencies of 700, 700, and 650 Hz. The low frequency gains of the low-shelf filters were -2.25, -2.0, and -1.42 dB. The low shelf filters had pole frequencies of 1038.8, 975.0, and 893.6 Hz. None of his settings came close to the original Bauer circuit because he used Linkwitz's reduced circuit.

The first setting was an original Boris Mikhaylow creation. He claimed that this setting would most closely resemble a speaker setup with azimuth angles of +/- 30° and a distance of 3 meters. The second setting was inspired by Chu Moy [5], while the third was influenced by Jan Meier [6]. All three configurations were less realistic than the original Linkwitz circuit because the magnitude of the interaural transfer function was reduced at all frequencies.

Some of RME's audio interfaces provide a crossfeed function. Page 19 of the ADI-2 DAC FS user manual [7] describes this features. The RME presets 4, 3, and 2 correspond to the original BS2B options 1, 2, and 3. The presets 1 and 5 are own modifications.

The original BS2B VST plugin is not supported by 64-bit Windows. However, the company Resonic issued the free 64-bit VST3 plugin BS2BR [8], which included the original three options as well as an additional Resonic preset. Please keep in mind that this plugin cuts signals over 0 dBFS and has a reduced total gain.

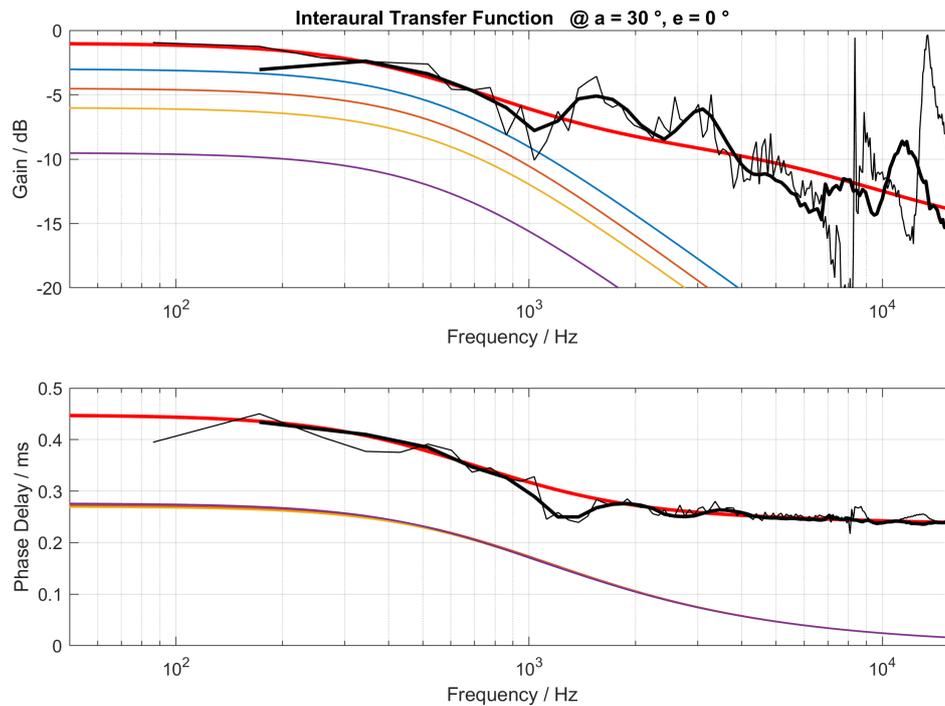
A universal crossfeed plugin was proposed by Francis F. Li in 2015 [9]. It was the first idea to implement  $H_{\text{inter}}$  as a series connection of a delay line and a minimum phase filter. Unfortunately, he replaced a realistic  $H_{\text{direct}}$  with a unity gain block and utilized a realistic  $H_{\text{cross}}$  as  $H_{\text{inter}}$ . As a result of this simplification, magnitude of the effective  $H_{\text{inter}}$  was too low at low frequencies and too high at high frequencies. This is unfortunate because he already utilized high-order filters, which might easily produce highly realistic results.

### **3. Approximation of the interaural transfer function**

The author analyzed publicly available HRTF measurements of an artificial head [10] and 48 subjects [11] at azimuth angles of +/- 30° and an elevation angles of 0°. The source distance  $r$  was 1.4 m for the artificial head and 1.5 m for the 48 individuals. Both distances are large enough to assume far field conditions, so the HRTFs are also applicable at higher distances. The author calculated two  $H_{\text{inter}}$  for two distinct pinnae sizes of the artificial head, as well as all 96  $H_{\text{inter}}$  of the

48 participants. For both situations, the mean gain and phase delay were determined. **Fig. 2** displays both results (thick black lines for the 48 subjects and thin black lines for the artificial head).

In **Fig. 2**, a thick red lines represents the author's approximation using a series connection of a fractional delay line and a second order minimum phase filter. The remaining colored thin lines represent Linkwitz and Mikhaylov's estimations.



**Fig.2:** Measured interaural transfer functions and some approximations.

The author's gain approximation is remarkably accurate at low frequencies and deviates only a few dB at middle and high frequencies. The author's phase delay approximation is fairly good at all frequencies except for a minor hump around 1.25 kHz. Of course, higher order filters could perform better, but the variations between individual interaural transfer functions are already on the same scale as the approximation deviations. Thus, the author does not expect a significant improvement in the spatial perception by employing higher order filters until the individual HRTFs of the actual listener are known.

The gain approximation of Linkwitz is better than Mikhaylov's, but it is still too low at low frequencies and much too low at high frequencies. The Linkwitz and Mikhaylov phase delay approximations are nearly equivalent, but significantly too low for all frequencies.

## 4. Implementation of the plugin

A second order minimum phase filter connected in series with a fractional delay line approximates the interaural transfer function.

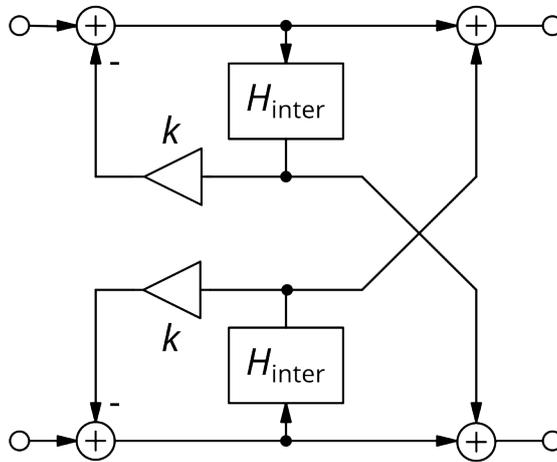
The second order minimum phase filter is based on an analog prototype of a universal tone stack. The filter was initially defined by the author in [12]:

$$H_{\text{TST}}(f) = \frac{B + M \cdot \frac{j \cdot f}{Q \cdot f_0} - T \cdot \frac{f^2}{f_0^2}}{1 + \frac{j \cdot f}{Q \cdot f_0} - \frac{f^2}{f_0^2}} \quad (9)$$

The parameters  $B$ ,  $M$ , and  $T$  control the tone stack's bass, middle, and treble levels, respectively. The parameter  $f_0$  represents the mid band's center frequency, and the reciprocal of  $Q$  determines its relative bandwidth. All parameter values should be non-negative real numbers. Thus, this filter is also the most general second-order minimum phase filter. The approximation in **Fig. 2** was accomplished using the following values:  $B = 0.8915$ ,  $M = 0.3448$ ,  $T = 0.1585$ ,  $Q = 0.25$ , and  $f_0 = 1.8$  kHz.

The analog prototype filter is converted to a direct form 2 digital filter using the bilinear transformation with  $f_0$  and  $Q$  prewarping, as described in [12].

The fractional delay line with a delay time  $t_d$  is implemented as a series connection of a delay line with an integer number of sample intervals  $T$  and a first-order digital allpass filter for the fractional part of the delay time. The delay time of the allpass filter ranges from  $0.618034 \cdot T$  to  $1.618034 \cdot T$ . This minimizes the allpass filter's time constant as well as phase deviations from an ideal delay line. The approximation in **Fig. 2** was attained with a delay time of 235  $\mu\text{s}$ .



**Fig. 3:** Modified crossfeed circuit

**Fig. 3** depicts the modified crossfeed circuit, which employs the approximated transfer function  $H_{\text{inter}}$ . This is an accurate and efficient implementation of the redefined transfer functions  $H_{\text{direct}}$  and  $H_{\text{cross}}$  from equations (6) and (7). This circuit is feasible as long as  $t_d$  is bigger than  $1.619034 \cdot T$ . The trick to avoid delay free loops is to separate  $H_{\text{inter}}$ 's fractional delay line into two parts: A delay of one sample interval  $T$  and a fractional delay line with a delay time of  $t_d - T$ .

The pseudo code below is an accurate implementation of the redesigned crossfeed circuit from **Fig. 3**. It must run for each new pair of left and right samples and overwrites them with the processed values.

```
direct_left = left -  $k$  * cross_left
direct_right = right -  $k$  * cross_right

left = direct_left + cross_right
right = direct_right + cross_left

cross_left = tone_stack( $B$ ,  $M$ ,  $T$ ,  $f_0$ ,  $Q$ , direct_left)
cross_left = delay( $t_d - T$ , cross_left)

cross_right = tone_stack( $B$ ,  $M$ ,  $T$ ,  $f_0$ ,  $Q$ , direct_right)
cross_right = delay( $t_d - T$ , cross_right)
```

The plugin's user can set a "Mono Compatibility" parameter between zero and 100%. This corresponds to values of  $k$  ranging from 0 to 1.

The plugin is a JSFX plugin created in the Jesusonic scripting language. It runs natively in the DAW Reaper. It is compatible with almost every audio plugin host thanks to Joep Vanlier's bridge plugin YSFX. YSFX is available for free on [13] as a VST3, AU, or Clap plugin for Windows, Mac OS, and Linux.

The author's plugin is available for free download from [14]. The source code is open and free to use, as long as the author's work is properly attributed. However, any commercial use of the source code and core ideas from this work requires the author's express consent. The author would gladly grant permission to any serious audio company without any expectation of license fees.



**Fig.4:** The GUI of the author's crossfeed plugin.

## 5. Derived transfer functions and listening tests

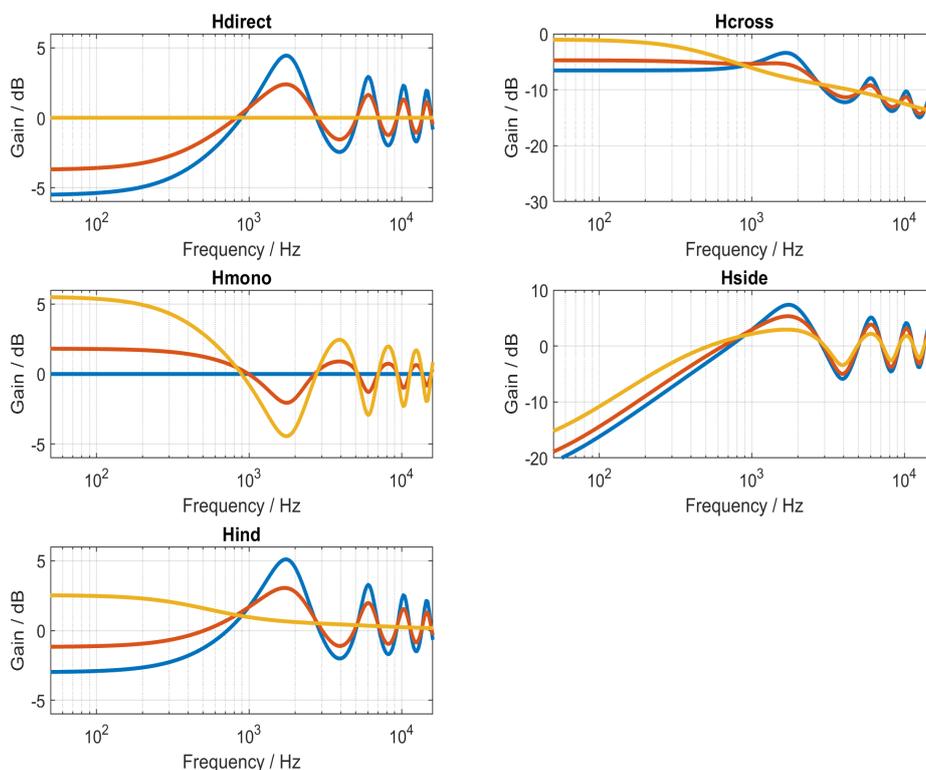
The gain plots of some derived transfer functions of the authors plugin are shown in **Fig. 5**. The color code for the mono compatibility setting is: 0 %, 60 % and 100 %.

The price of a flat  $H_{\text{mono}}$  is obvious: the other transfer functions are not flat. This is not a fault with this plugin; it is simply the result of implementing realistic interaural transfer functions. The signals

received by both ears influence the perceived sound color in a complicated way. Thus, none of the derived transfer functions represent the perceived sound color of stereophonic signals. Only for monophonic signals, there is no doubt that a flat  $H_{\text{mono}}$  will not change the perceived sound color when the plugin is enabled.

When listening to common stereophonic recordings, the author suggests using a mono compatibility value of approximately 60%. The perceived sound color change that may occur when the crossfeed plugin is engaged is minimal in this typical application.

When the crossfeed plugin is enabled in any of its settings, the spacious impression improves uniformly and significantly. Especially hard panned signals sound much more natural. In general, the signals are no longer situated merely between the two ears, but rather on a 60-degree wide arc in front of the head center. Unfortunately, the perceived radius of this arc is in the same order of magnitude as the radius of a common head, rather than several meters as desired.



**Fig. 5:** Derived transfer functions of the authors plugin.

**Fig. 6** displays the gain plots of the transfer functions for the Linkwitz and Mikhaylov circuits. The color of the Linkwitz circuit is blue.

Mikhaylov's settings improve the spatial perception slightly, but not as significantly as the author's plugin. The first setting has the most impact. The Linkwitz circuit was not available for hearing tests.

The author's website [15] has a web player for an audio file including signal samples that have been unprocessed (O), processed by the BS2BR plugin in the first Mikhaylov setting (B), and processed by the author's plugin with 60% mono compatibility (K). The BS2BR plugin's gain loss has been corrected, and it is operating below its clipping threshold. For each signal sample, you will hear the following sequence: O-B-K.

The first signal is pink noise that pans continually from left to right. The second signal is composed of two independent pink noise signals: a mid-signal and a side-signal with a 6 dB lower strength. This signal approximates the characteristics of a normal stereo recording. The final three signals are brief snippets from popular stereo records.

Enjoy your own listening test!

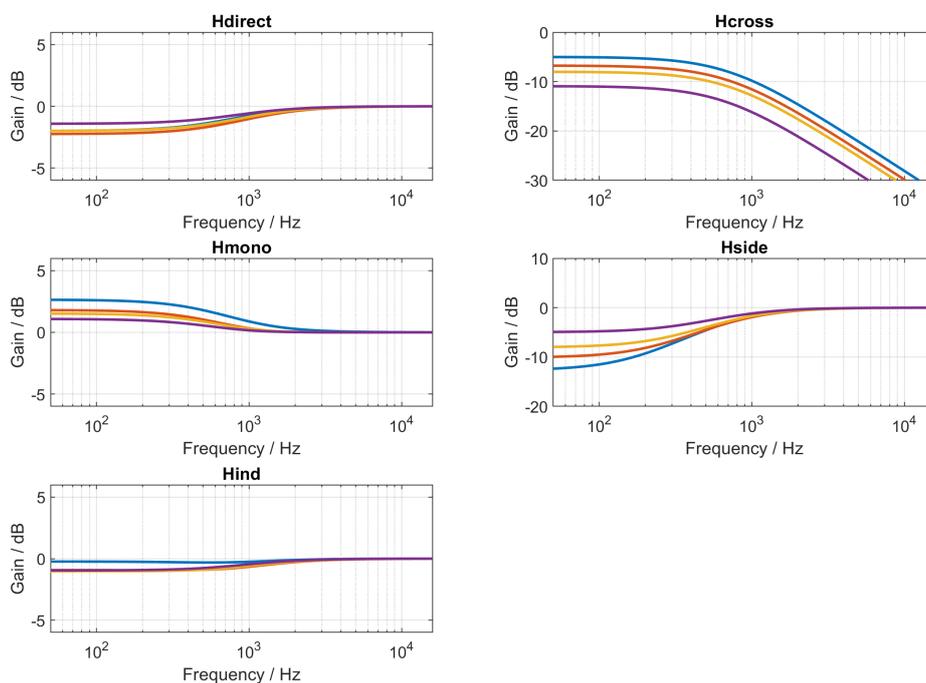


Fig.6: Transfer functions of the Linkwitz and Mikhaylov crossfeeders.

## 5. Summary

An innovative crossfeed audio plugin has been created to improve the listening experience with headphones. This solution offers a more realistic spatial perception than existing options. When a novel equalizer is applied to stereophonic signals, it produces minor sound color changes. There are no sound color changes for monophonic signals when the equalization is set to full mono compatibility. The free and open source JSFX plugin is compatible with practically any audio plugin host thanks to Joep Vanlier's free bridge plugin YSFX.

The author's method produces a stereo image close to an 60 degrees wide arc in front of the listener's head center, with a perceived distance of about 10 cm. Individualized HRTFs, changed by head tracking, are most likely required for more realistic distance perception. The tactile reaction to high low frequency sound pressure levels is not implemented because it would necessitate the

employment of skakers. The acoustics of the listening space are not mimicked either. This could be accomplished utilizing an ambient simulation, such as a convolution reverb with impulse responses recorded with an artificial head in high-quality listening rooms or mastering studios.

## 6. Links

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- [2] Benjamin. B. Bauer: Patent US 30888997, 1963  
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- [5] Mu Choy: “An Acoustic Simulator For Headphone Amplifiers”  
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- [13] Joep Vanlier: Download link for theYSFX plugin  
<https://github.com/JoepVanlier/ysfx/releases>
- [14] Helmut Keller: Download link for the authors crossfeed plugin  
<https://www.helmutkelleraudio.de/downloads/811aabeeac55bbaad0980342a67f65c4>
- [15] Helmut Keller: The authors website  
<https://www.helmutkelleraudio.de/>