

# Blind Cancellation of Stereo-Base Widening\*

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Cascading of stereo-base widening systems causes a deterioration of the listening performance. A solution to this problem is to cancel the widening effect before the signal is widened again. The main effect of stereo-base widening systems is to the effect on the phase difference. A model for the distribution of the phase difference of normal mixed stereo recordings is presented. Stereo recordings processed by a stereo-base widening system do not fit this model. An application of this property is illustrated with an example. This example can estimate the amount of widening of a particular widening system in an iterative blind way. At the same time it recovers the original audio signal.

## 0 INTRODUCTION

In today's world it is hard to imagine sound reproduction without the use of stereophonic techniques. There are numerous methods that attempt to record the stereo image in the best possible way in order to make a lifelike reproduction of the sound image. However, in the case where the distance between the listener and the loudspeakers is much greater than the loudspeaker separation, it appears as if the sound were coming from only one direction. This means that the stereo image collapses and the sound appears to be monophonic.

In order to prevent this collapse, methods to widen the stereo-base virtually have been developed. The idea of improving stereo dates back to the 1930s. Blumlein [1] pioneered in this field; then Schroeder [2] discussed artificial stereophony, and Bauer [3] published about stereophonic broadening. Atal and Schroeder [4] outlined a method for generating a phantom source. This idea was further elaborated by Cooper and Bauck [5]. These methods form the basis of a vast amount of literature that followed. For instance, Kitzen and Boers [6] state that widening of the stereo base can be achieved by adding the left signal, slightly attenuated, phase-inverted, and delayed, to the right channel and vice versa. Taking this approach a step further, a small delay can be added to the phase-inverted signal before mixing it into the original. This adds a slight reverberation-like effect to the widening effect. This principle was further elaborated by Aarts [7], which resulted in a system that gives a natural widening effect with the help of two digital or analog filters.

The research done in this area has led to the present situation. Nowadays stereo-base widening systems are part of a large number of multimedia systems such as televisions, boom boxes, and mobile phones. However, a problem arises when these systems are cascaded. The cascading of stereo-base widening systems causes a deterioration of the listening performance. Cascading occurs when, for example, a TV network broadcasts a widened audio signal and the receiving television widens the stereo base of this signal further. The most straightforward solution to overcome this cascading problem would be to detect whether stereo-base widening is applied and, if so, not to widen the signal again. A better solution is to recover the original signal by the television set itself before it applies its own stereo-base widening system. In this way the stereo widening is always performed with the same settings.

In this paper such a solution is discussed for the system proposed by Aarts [7]. We call this the stereo widening (SW) system. In Section 1 we consider stereo-base widening systems and determine the effects of such systems on the audio signal. Then we develop a model that can discriminate normal stereo audio from audio processed by stereo-base widening systems. Section 2 provides an example that can estimate the amount of stereo widening applied by the stereo widening system in a blind way. In Section 3 experimental results of this method are shown. Finally we present our conclusions in Section 4.

## 1 FORMULATION OF THE PROBLEM

### 1.1 Model of Widening System

Fig. 1 shows a general model of a stereo-base widening system [7]. The system consists of two pairs of filters ( $H_1$

Manuscript received 2006 October 29; revised 2007 March 8.

and  $H_2$ ) and the parameters  $\alpha$  and  $g$ . The filters  $H_1$  and  $H_2$  are in general derived from head-related transfer functions (HRTFs), but as HRTFs are very tied to the person and the position in space, these filters are smoothed in a particular way. A caricature of the filters  $H$  is that  $H_1 = 1$  and  $H_2 = -0.5$ , which was the basis for early widening systems. While the widening system described in [7] appears specific, we believe that many systems—including the caricature one—are similar to the one presented in [7], and therefore we consider it hereafter as more general. This is not always the case. However, in those cases the method presented might work less well.

The parameter  $g$  is a constant gain factor, which compensates for the signal weakening due to the filtering, and  $\alpha$  is a variable between 0 and 1. The latter denotes the amount of widening. If  $\alpha$  is 0, the output signals are equal to the input signals. If  $\alpha$  is equal to 1, then the output signals only consist of the filter outputs. In this case the stereo-base widening is maximal. For values of  $\alpha$  between 0 and 1 the output signals are a mixture of the input signals and the filter outputs and consequently influence the impact of the filters.

In order to present this model in algebraic form, we describe the system in the Fourier domain. For notational convenience the frequency indices are omitted. Suppose that

$$A = X_1 - X_2 \tag{1}$$

$$B = X_1 + X_2 \tag{2}$$

Then

$$\begin{bmatrix} Y_1 \\ Y_2 \end{bmatrix} = \begin{bmatrix} H_{11} & H_{12} \\ H_{21} & H_{22} \end{bmatrix} \begin{bmatrix} A \\ B \end{bmatrix} \tag{3}$$

where

$$H_{11} = -H_{21} = \frac{1}{2}[1 - \alpha + \alpha g(H_1 - H_2)] \tag{4}$$

$$H_{12} = H_{22} = \frac{1}{2}[1 - \alpha + \alpha g(H_1 + H_2)]. \tag{5}$$

Using the algebraic representation of Eq. (3), we will consider the effects of this system on the input signals.

### 1.2 Effects of Stereo-Base Widening

In order to evaluate the effects of the system, these are discussed for a single frequency. Consider the situation where  $g$  is constant,  $\alpha$  is equal to 1, and

$$H_1 = A_{H_1} e^{j\varphi_{H_1}} \tag{6}$$

$$H_2 = A_{H_2} e^{j\varphi_{H_2}}. \tag{7}$$

We assume that  $X_1$  and  $X_2$  for this frequency are characterized by the amplitude and phase parameters  $A_{X_1}$ ,  $\varphi_{X_1}$  and  $A_{X_2}$ ,  $\varphi_{X_2}$ , respectively. One of the effects of the widening system structure is the effect on the phase difference  $\varphi_{X_1} - \varphi_{X_2}$ . The relation between the input phase difference  $\varphi_{X_1} - \varphi_{X_2}$  and the output phase difference  $\varphi_{Y_1} - \varphi_{Y_2}$  is evaluated by changing one of the filter variables and keeping the others constant. The set of filter variables consists of  $A_{H_1}$ ,  $\varphi_{H_1}$ ,  $A_{H_2}$ , and  $\varphi_{H_2}$ . The effects on the phase difference are calculated by applying the harmonic addition theorem. This theorem states that it is always possible to write a sum of sinusoidal functions,

$$f(\theta) = a \sin(\theta + \varphi_a) + b \sin(\theta + \varphi_b) \tag{8}$$

as a single sinusoid of the form

$$f(\theta) = c \sin(\theta + \varphi_c). \tag{9}$$

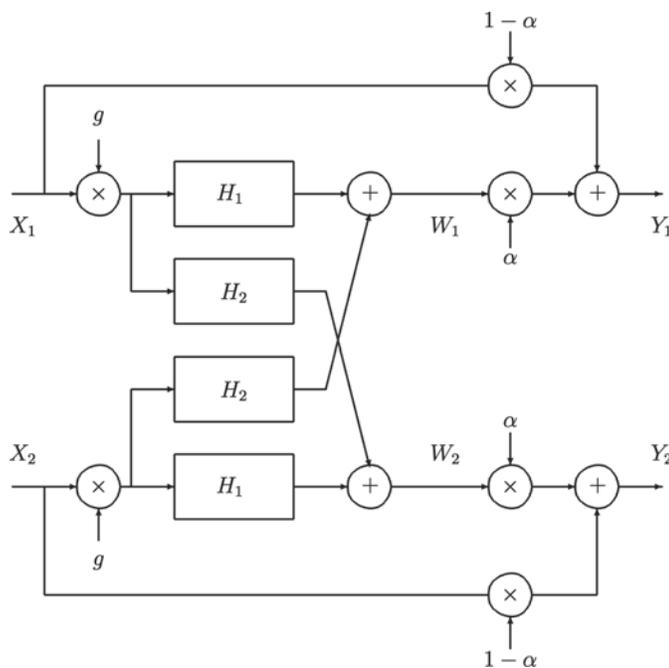


Fig. 1. General stereo-base widening system structure.

For the general stereo widening structure this results in

$$\varphi_{Y_1} = \tan^{-1} \left[ \frac{A_{X_1} A_{H_1} \sin(\varphi_{X_1} + \varphi_{H_1}) + \dots + A_{X_2} A_{H_2} \sin(\varphi_{X_2} + \varphi_{H_2})}{A_{X_1} A_{H_1} \cos(\varphi_{X_1} + \varphi_{H_1}) + \dots + A_{X_2} A_{H_2} \cos(\varphi_{X_2} + \varphi_{H_2})} \right] \quad (10)$$

$$\varphi_{Y_2} = \tan^{-1} \left[ \frac{A_{X_2} A_{H_1} \sin(\varphi_{X_2} + \varphi_{H_1}) + \dots + A_{X_1} A_{H_2} \sin(\varphi_{X_1} + \varphi_{H_2})}{A_{X_2} A_{H_1} \cos(\varphi_{X_2} + \varphi_{H_1}) + \dots + A_{X_1} A_{H_2} \cos(\varphi_{X_1} + \varphi_{H_2})} \right] \quad (11)$$

For reasons of simplification, the parameters  $A_{X_1}$  and  $A_{X_2}$  are assumed to be equal. The results of this are plotted in

Fig. 2. Table 1 shows the values of the filter variables used in this figure.

Eqs. (10) and (11) and Fig. 2(a) and (b) show that if the relative difference between  $A_{H_1}$  and  $A_{H_2}$  becomes smaller, the curve gets sharper. The results in Fig. 2(b) are equal to the results in Fig. 2(a), but with the phase difference  $\varphi_{X_1} - \varphi_{X_2}$

Table 1. Values for Fig. 2.

Plot	$A_{H_1}$	$A_{H_2}$	$\varphi_{H_1}$	$\varphi_{H_2}$
(a)	*	1	0	$\pi$
(b)	1	*	0	$\pi$
(c)	0.9	1	*	0
(d)	0.9	1	0	*

\*—Coefficient evaluated.

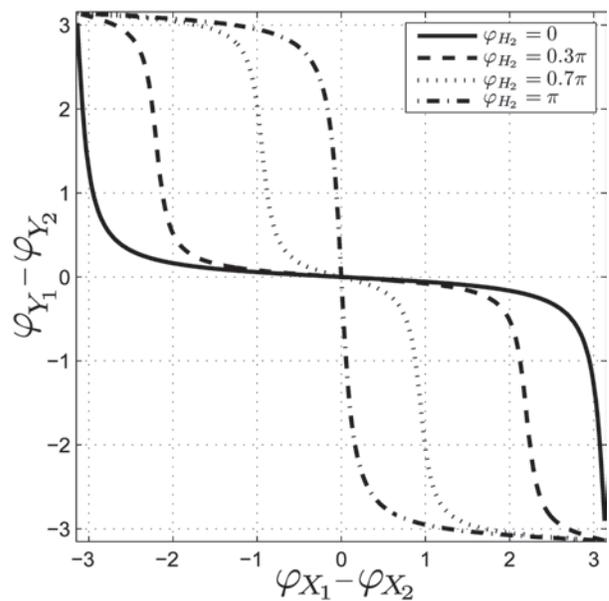
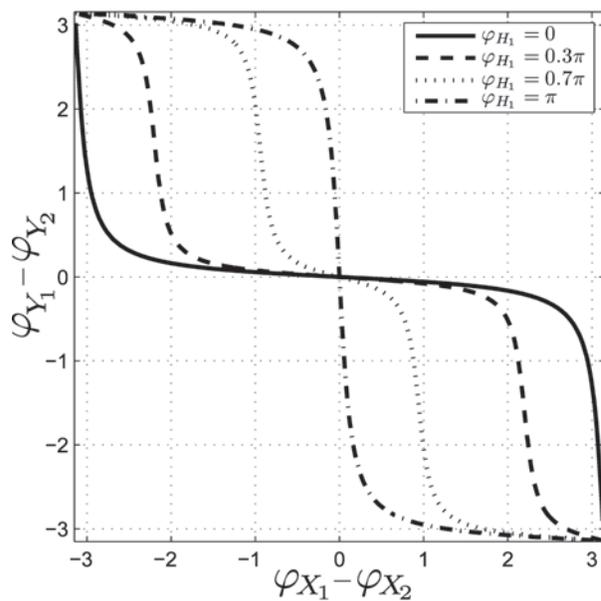
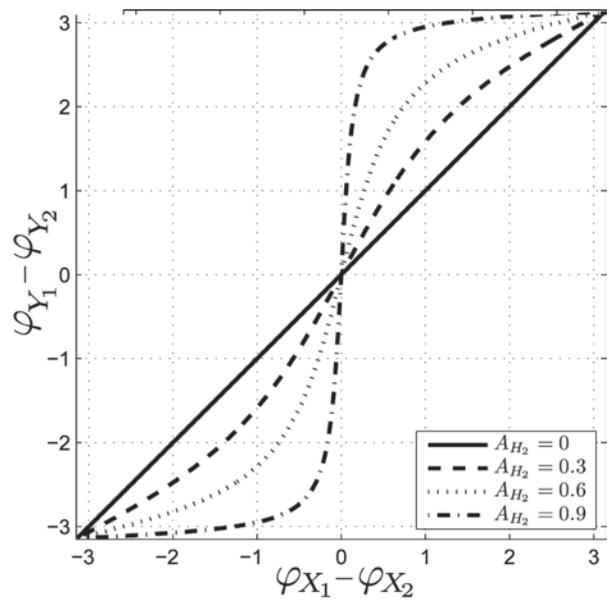
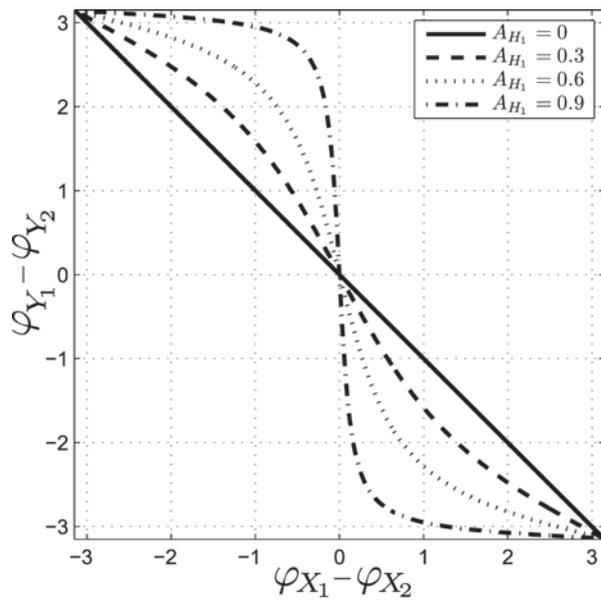


Fig. 2. Effects of filter coefficients on phase difference.

inverted. So it follows that if  $A_{H_1} < A_{H_2}$ , the output phase difference changes its sign. Furthermore, Fig. 2(c) and (d) indicates that the shape of the curve depends on the difference between  $\varphi_{H_1}$  and  $\varphi_{H_2}$ . As can be seen from Fig. 2(c) and (d), which are equal, the sign of the difference between  $\varphi_{H_1}$  and  $\varphi_{H_2}$  does not matter. The absolute value of this difference causes amplification or reduction of the phase difference of the input signals  $\varphi_{X_1} - \varphi_{X_2}$ .

### 1.3 Inversion of Widening System

To recover the original signals  $X_1$  and  $X_2$ , an inverse system is required. The inverse of the system shown in Eq. (3) is

$$\begin{bmatrix} A \\ B \end{bmatrix} = \frac{1}{2H_{11}H_{12}} \begin{bmatrix} H_{12} & -H_{12} \\ H_{11} & H_{11} \end{bmatrix} \begin{bmatrix} Y_1 \\ Y_2 \end{bmatrix}. \quad (12)$$

In the case of the stereo widening system we will refer to this inverse system as the inverse stereo widening (ISW) system.

From the preceding equations it follows that

$$A = \frac{1}{2}H_{11}^{-1}(Y_1 - Y_2) \quad (13)$$

$$B = \frac{1}{2}H_{12}^{-1}(Y_1 + Y_2). \quad (14)$$

An exact inversion is only possible if  $H_{11}$  and  $H_{12}$  are minimum phase. This is true in the specific case of the stereo widening system. Therefore the latter can be perfectly inverted if all the parameters and filters are known.

## 2 BLIND SOLUTION

### 2.1 Blind Identification

If the structure of the widening system is known, but one or more of the parameters or filter variables is not, the recovery of the original audio signal becomes a multichannel blind identification problem [8]. However, in this case this is not the separation of a set of signals from a set of mixed signals. It comes down to the conversion of mixed signals into another mixture. Because of this, we cannot use the blind signal separation techniques discussed by Lambert [8] and Abrard and Deville [9].

In order to be able to identify the unknown variables, an attribute of audio must exist which can be used to discriminate general audio from audio that is processed by the stereo-base widening system.

To this end we need to know more about stereo audio signals. There are two ways in which the human brain can extract the direction of sound from a stereo audio signal

[10], [11]. The first is the effect of intensity cues. If, for example, the level of the signal coming from the right loudspeaker is higher than that from the left loudspeaker, the phantom image shifts to the right.

The second way is the phase difference effect. Spatial cues are perceived at lower frequencies by the relative time of arrival of the signal at both ears. As a wavefront from the left loudspeaker sweeps past the head of the listener, the sound reaches the left ear shortly before the more distant right ear. So if the phase from the right channel lags behind that from the left channel, the signal appears to be coming from the left. If the phase difference gets bigger, this effect becomes more pronounced.

In Section 1.2 it was shown that the general stereo-base widening system structure affects the phase difference. Now we want to know whether this can act as a discrimination cue. Therefore we are interested in the behavior of the phase difference of nonwidened stereo audio. In order to evaluate this, we consider the distribution of the phase difference of a stereo audio signal. This phase difference is calculated via the method shown in Fig. 3, by which the phase difference per block of samples ( $\Delta\varphi_i$ ) is calculated.

Due to the fact that, in general, a single audio channel is nonstationary, the distribution of the phase evaluated over time of such a channel is uniformly divided between  $-\pi$  and  $\pi$  for all frequencies. So if sufficient blocks are processed, the distribution of the values of  $\varphi_{X_1}$  and  $\varphi_{X_2}$  from Fig. 3 can be seen as  $n/2$  uniformly distributed bins, where  $n$  denotes the number of samples per block and  $i$  the block number. For each frequency bin the distribution of the phase difference over time corresponds to the distribution of the difference of two correlated uniform distributions.

To evaluate the correlation per frequency approximately 20 CDs of several music styles were processed. This shows that, roughly speaking, this distribution of the phase difference is similar for all frequencies. However, the amount of this correlation can differ per fragment. A model for normal audio signals with a similar distribution for all frequencies can be created by

$$\tilde{X}_1 = \beta N_1 + (1 - \beta)N_2 \quad (15)$$

$$\tilde{X}_2 = (1 - \beta)N_1 + \beta N_2 \quad (16)$$

where  $\frac{1}{2} \leq \beta \leq 1$  and  $N_1$  and  $N_2$  are independent white Gaussian noise sources. Of course not all audio obeys this model, but experiments show that it gives a good insight.

In order to demonstrate the similarity between the model described by Eqs. (15) and (16) and audio, histo-

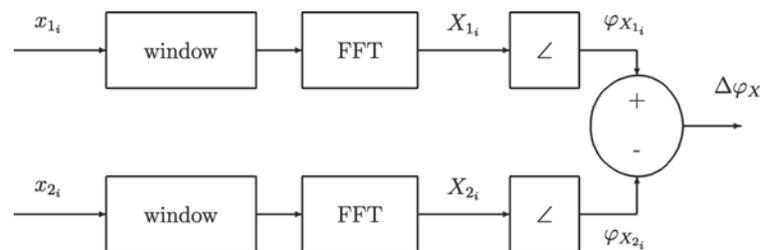


Fig. 3. Phase difference calculation, where  $\Delta\varphi_{X_i} \in \mathbb{R}^{n/2}$ .  $n$ —number of samples per block;  $i$ —block number.

grams of the phase difference as a function of the frequency are presented in Fig. 4(a) and (b). The distributions are created by evaluating 1000 blocks, each block consisting of 512 samples. Each block is windowed by the Hanning window. The histograms are normalized to a maximum value of 1. The phase difference is divided into 100 bins between  $-2\pi$  and  $\pi$ . Fig. 4(a) shows the distribution of the phase difference of a pair of noise sources generated via Eqs. (15) and (16) with  $\beta = 0.8$ . Fig. 4(b) shows the measured distribution of the phase difference for a fragment of classical music. It is clear that these plots resemble each other.

However, it should be noted that the value of the parameter  $\beta$  differs per audio fragment. The parameter  $\beta$  represents the amount of correlation between the left and right signals. If the left and right signals are equal ( $\beta = 1/2$ ), the phase difference will always be zero. If the left and right signals are not correlated ( $\beta = 1$ ), the distribution of the phase difference in time will be equal to the convolution of two uniform distributions [12], making this a triangular function. For the trajectory of  $\beta = 1/2$  to  $\beta = 1$  the shape of distribution changes gradually from the single peak to the triangular function. It is possible to imagine parameter  $\beta$  varying within a music fragment. If, for example, a trumpet is added to the right signal only, the

signal is less correlated, and consequently the value of  $\beta$  is increased.

Now that we have a model for audio, we need to know whether this model can discriminate widened audio from nonwidened audio. This is investigated for the stereo widening system with a value of  $g = 3.5$ . The magnitude and phase responses for the filters  $H_1$  and  $H_2$  are shown in Fig. 5. These plots show that the relative magnitude difference is small. The phase difference, however, changes from  $\pi$  for lower frequencies to 0 for higher frequencies. If the value of  $\alpha = 1$ , the effect of the system on the distribution of the phase difference in time can be predicted by the results presented in Fig. 2. From this figure we expect that, for lower frequencies, the phase difference will be pushed toward either plus or minus  $\pi$  and, for higher frequencies, it will be pushed toward zero.

To verify this prediction, the normalized distribution of the phase difference in time is presented in Fig. 4(c) and (d) for the same signals as in Fig. 4(a) and (b) but now processed by the stereo widening system with  $\alpha = 1$ .

Although simplified by the assumption of equal  $A_{X_1}$  and  $A_{X_2}$ , it turns out that the results presented in Fig. 2 are quite good estimators of the effects on the phase difference. Moreover, it follows that the effect on the audio model in Eqs. (15) and (16) is the same as the effect on real music.

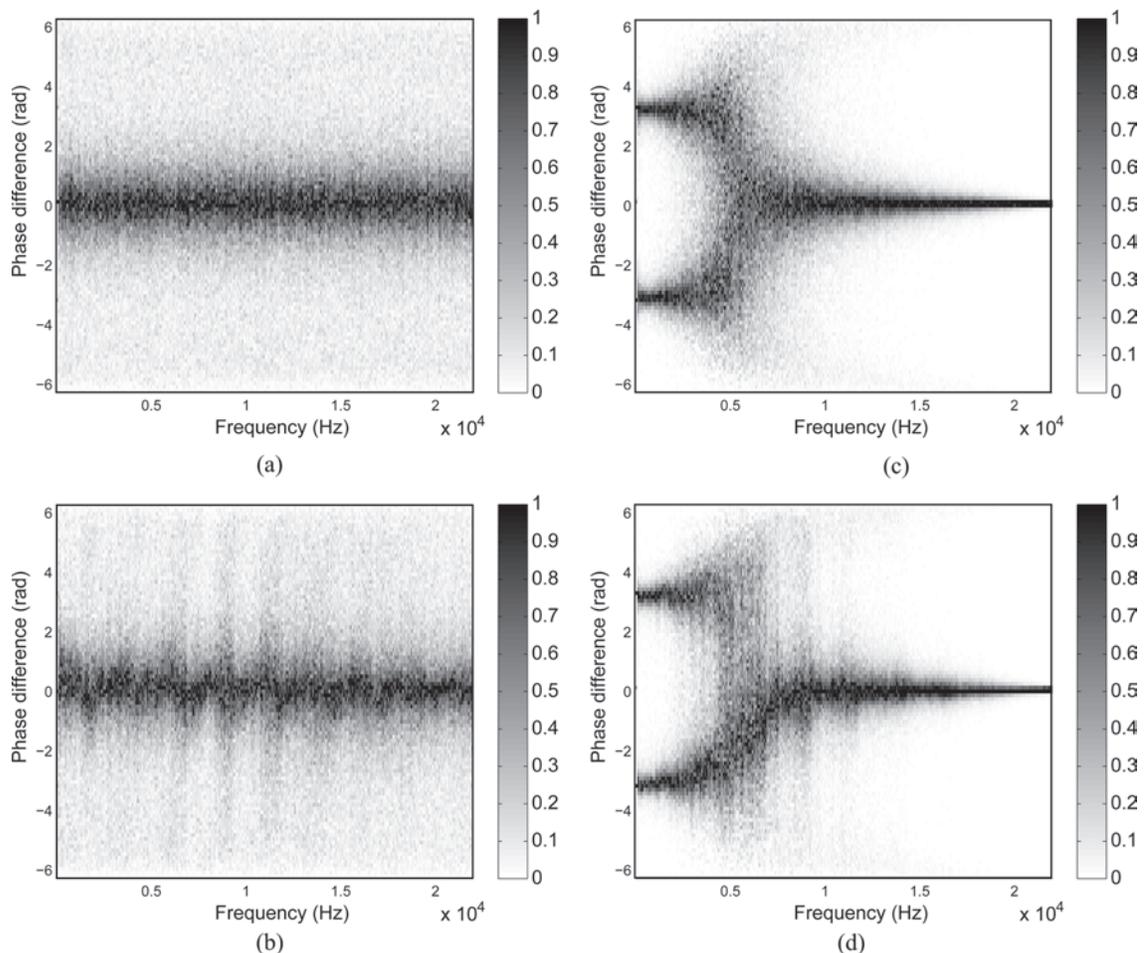


Fig. 4. Normalized histograms of phase difference in time for all frequencies. (a) Pair of noise sources with  $\beta = 0.8$ . (b) Sample of “Ah! del padre in periglio” from “Mozart—Don Giovanni” by Sir Neville Marriner. (c), (d) Similar to (a), (b), but signals are processed by the stereo widening system with  $\alpha = 1$ .

Subsequently audio processed with a stereo-base widening system does not fit the model of the phase difference of mixed stereo audio, so the model can indeed discriminate widened audio from nonwidened audio.

### 2.2 Iterative Blind Solution

In the previous section we presented a model that fits normal mixed stereo recording, but does not fit recordings processed by a stereo-base widening system. This model can be used as the basis for blind system identification. As an example of blind identification, a method to estimate the value of the parameter  $\alpha$  in an audio signal processed by the stereo widening system will be presented. In Fig. 6 this estimation is illustrated by a block diagram. The known signals  $Y_1$  and  $Y_2$  are created by processing the unknown stereo audio channels  $X_1$  and  $X_2$  by the stereo widening system with an arbitrary unknown value of  $\alpha$ . The example aims to estimate the value of  $\alpha$  in order to get an estimate of the original signals  $X_1$  and  $X_2$ .

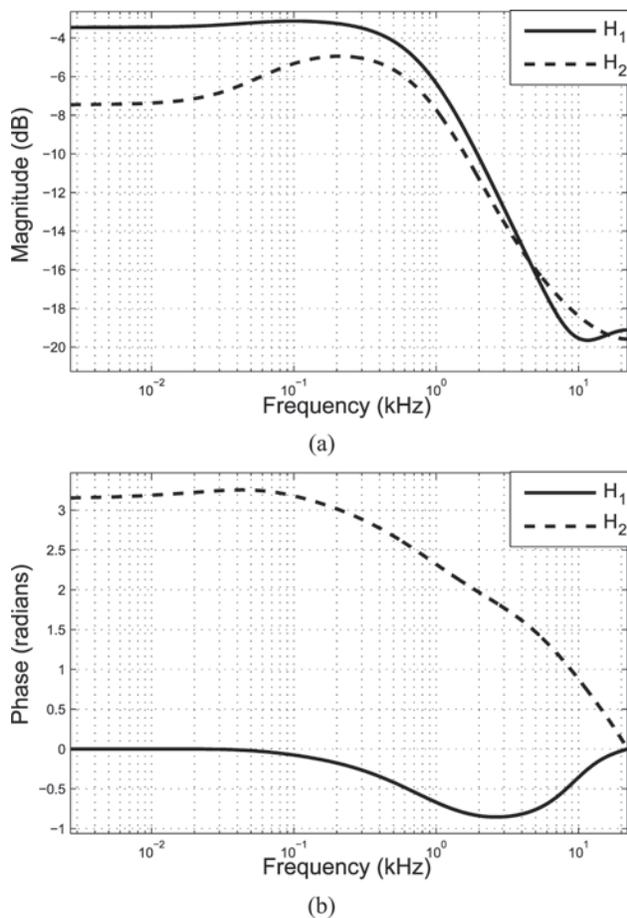


Fig. 5. Filters  $H_1$  and  $H_2$  of stereo widening system for 44.1 kHz. (a) Magnitude response. (b) Phase response.

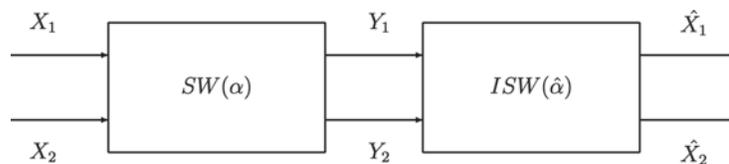


Fig. 6. Block diagram of blind identification problem.

The parameter  $\beta$  (the amount of correlation) in the model is equal for all frequencies. So the distribution of the phase difference over time is also equal for all frequencies. This means that the first-order polynomial fit of the variance of the phase difference as a function of frequency is a horizontal line.

From the plots presented in Fig. 4 it can be seen that this is not the case for signals filtered with the stereo widening system. In this case the variance will be high for low frequencies and low for high frequencies. The first-order approximations of the variance of a music fragment are plotted in Fig. 7. It can be seen that the angle of the slope

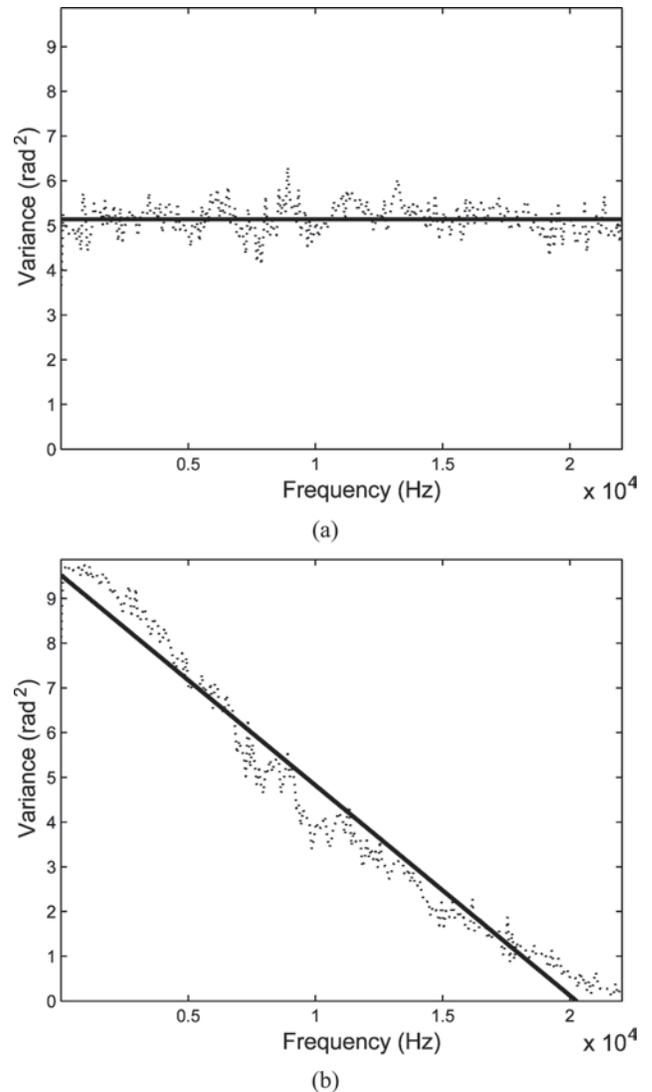


Fig. 7. “Ah! del padre in periglio” from “Mozart—Don Giovanni” by Sir Neville Marriner. First-order approximations of variance as a function of frequency of this music fragment processed with the stereo widening system (a)  $\alpha = 0$ . (b)  $\alpha = 1$ .

of the first-order approximation of the variance as a function of the frequency is 0 for nonwidened audio. If  $\alpha$  is larger than 0, this angle is negative. This property is the basis of the following method to estimate the value of the parameter  $\alpha$ .

A method that estimates  $\alpha$  and recovers the original audio signals in an iterative way is presented in Fig. 8. Here the incoming audio signal is processed by the inverted stereo widening system with a certain value of  $\hat{\alpha}$ . The estimates of the original signals are created by  $\hat{X}_1 = \frac{1}{2}(\hat{A} + \hat{B})$  and  $\hat{X}_2 = -\frac{1}{2}(\hat{A} - \hat{B})$ , where  $\hat{A}$  and  $\hat{B}$  denote the estimates of  $A$  and  $B$ , respectively. Then the phase differences of a number of blocks are put into a buffer. Now the variance of these phase differences is calculated for all frequencies. Next a first-order approximation of the variance as a function of the frequency is made. The slope depends on the amounts of the correlation  $\beta$  and the widening  $\alpha$ .

If the slope of this first-order approximation is less than 0, the value of  $\hat{\alpha}$ , that is, the estimate of  $\alpha$ , will be increased. If it is higher than 0, the value of  $\alpha$  will be decreased. This is carried out using the following update rule:

$$\hat{\alpha} := \hat{\alpha} - \mu\theta \quad (17)$$

where  $\theta$  is the slope. This way we do not need to estimate the value of  $\beta$ .

The variable  $\mu$  denotes the adaption step size. The maximum value of the slope is approximately  $\pi^2/22050 \approx 4.5 \times 10^{-04}$  in the case of a 44.1-kHz sample rate. A typical value of  $\mu$  will be 200 to 1000, depending on the desired tracking speed. In this way the value of  $\alpha$  will be estimated in real time while, simultaneously, the estimates of the original signals will be recovered. The value of  $\hat{\alpha}$  can also be used as a discriminator to detect whether stereo-base widening is applied.

### 3 EXPERIMENTAL RESULTS

In this section experimental results of the method to estimate the amount of widening in the stereo widening system will be presented. The amount of widening in this system is represented by the parameter  $\alpha$ . In order to estimate the value of  $\alpha$ , the method presented in Fig. 8 is used. Fig. 9 shows the results for a fragment of classical

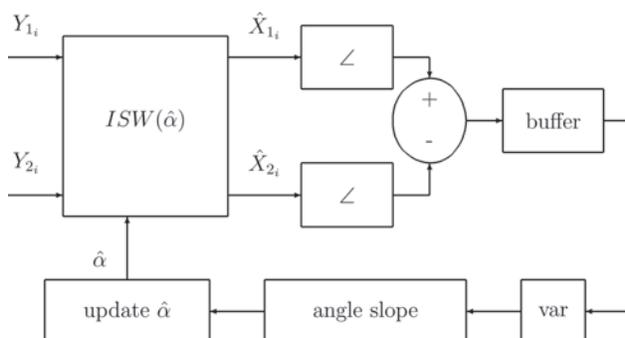


Fig. 8. Iterative system for blind identification.

music. For this plot the block size is 512 samples, and the number of blocks in the buffer is 50. The value used for  $\mu$  is 800.

For this audio signal the estimated value of the parameter  $\alpha$  is a good estimate of the true value, considering that the value is estimated in a blind way. If we listen to the recovered audio, we do indeed notice that the stereo image is changed from wide to normal. The audible effect of changing  $\alpha$  is negligible for the small variations that occur after about 5 seconds. Only for big variations, such as in the first 5 seconds, does one hear that the audio wideness is changing.

In order to see how the system works for different kinds of music,  $\hat{\alpha}$ , the mean values of the  $\alpha$  estimates for a number of music fragments, are plotted in Fig. 10. From this plot it can be seen that the detection error is smaller for higher values of  $\alpha$ . This is caused by the fact that, for higher values of  $\alpha$ , the effect on the slope of a small change in value is greater than for lower values.

Overall it appears that the angle of the first-order approximation of the variance as a function of frequency is a good parameter for estimating the value of  $\alpha$ . However, it must be said that not all music fits the model equally well. From Fig. 10 it can be seen that, for classical music in particular, the blind identification error is small. On the other hand, for house music the identification error can be relatively large. This is probably because in house music the aim is often not to create a “natural” stereo image, but to create an artistically interesting effect.

If we listen to the recovered signals, these signals are indeed very similar to the original signals. This is mainly due to the fact that the effect of the stereo widening system on audio is only noticeable from a value of  $\alpha$  that is larger than 0.3. However, Fig. 10 presents two “extreme” peaks. In track 10 the peak is downward and in track 11 it is upward. In the case of track 11, this means that the recovered audio signal is “overcompensated”.

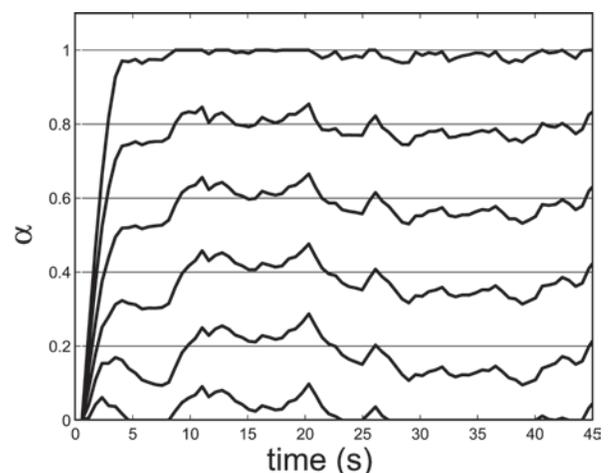


Fig. 9. Estimate of  $\alpha$  for “Ah! del padre in periglio” from “Mozart—Don Giovanni” by Sir Neville Marriner. Top to bottom—lines represent estimates for audio signal preprocessed using stereo widening system with  $\alpha = 1, 0.8, 0.6, 0.4, 0.2$ , and 0, respectively.

sated,” that is, more correlated, thus more monophonic, than the original track. In the case of track 10, the identification error results in a recovered audio signal that sound slightly widened compared to the original track. Nevertheless, when listened to without the original as reference, the recovered audio signals cannot be classified as widened both visually (by a similar plot as Fig. 4) and acoustically. This means it sounds like natural nonwidened stereo.

The example shows that the model of Eqs. (15) and (16) can be used as a basis for stereo-base widening detection or the cancellation of stereo-base widening in a blind way. However, to this end more research on the various widening systems on the market is needed. In the blind cancellation case the amplitude distribution of audio needs to be investigated too. This can lead to methods for reconstructing the original audio signals by estimating the filters in a blind way.

#### 4 CONCLUSIONS

We have studied the effects of a stereo-base widening system on the interchannel phase difference distribution of an audio signal. This showed that stereo-base widening systems have a large effect on the phase difference. In order to allow blind system identification, we investigated the phase difference of stereo audio signals. We showed that, with respect to the distribution of the phase difference, normal stereo signals can be modeled by two correlated white noise sources. It appeared that audio signals processed with the stereo widening system did not fit this model.

Next we presented an example of blind identification based on this property. This example detects the amount of widening of the stereo widening system in an iterative way. At the same time it recovers an estimate of the original audio signals. We tested this method using a number of music fragments. The experimental results were good, especially in the case of classical music.

In general the proposed model can be used as the basis of a stereo-base widening detector or canceler. However, further research in the various stereo-widening systems on the market and methods to reconstruct the original audio signal is needed.

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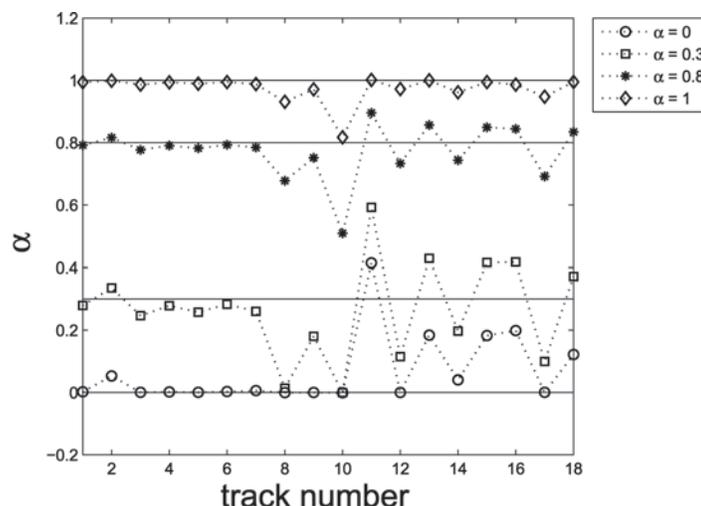


Fig. 10. Mean values of estimated  $\alpha$  of audio tracks processed using stereo widening system with  $\alpha = 0, 0.3, 0.8, 1$ , respectively. Tracks 1–6: “Schubert—Piano Sonatas D.784, D.850” by Alfred Brendel, CD tracks 1–6. Tracks 7–12: “1, 2, 3, Fiesta! (3 hot Latin house mixes)” by various artists, CD tracks 2–7. Tracks 13–18: “don’t give me names” by the Guano Apes, CD tracks 1–6.

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