

Surround Recording Based on a Coincident Pair of Microphones

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ABSTRACT

Recently, we proposed an adaptive beamformer with the target to simulate a highly directive microphone, as opposed to the conventional signal-to-noise ratio optimization target. It is shown that by applying this beamformer in different ways to a coincident cardioid or tailed stereo cardioid signal pair, audio channels suitable as front left, front right, surround left, and surround right audio channels can be generated. It is also described how to generate a center channel from the same microphone signals. While the proposed surround sound generation is not as flexible as linear B-Format decoding in terms of determining the direction of the response for each channel, the directional responses of the audio channels overlap less resulting in higher channel separation.

1. INTRODUCTION

Many different techniques and philosophies are applied for multi-channel surround recording and mixing [1, 2]. While often techniques from stereo recording and mixing are adapted for more channels, in order to fully explore the possibilities of surround sound, new techniques have been and are being developed [3].

This paper treats a specific topic related to surround sound: Recording of surround sound with small sound capturing setups, such as are used for recording surround sound for broadcasting or for home video. To fulfill the requirement for a small sound capturing setup, coincident microphones are used.

The most prominent coincident microphone system suitable for multi-channel surround sound capture is the Soundfield Microphone [4, 5]. Its four nearly coincident cardioid or sub-cardioid microphone elements' signals are combined and filtered, resulting in B-Format. B-Format contains four channels with one omni-directional signal (W) and signals of dipoles oriented towards x , y , and z directions (X , Y , and Z).

With appropriate processing [6, 7], not only horizontal multi-channel surround sound signals can be generated from the Soundfield Microphone's B-Format signals, but also signals for three dimensional sound systems with elevated loudspeakers.

The proposed surround sound capturing is based on two coincident cardioid or tailed cardioid signals (e.g. XY

stereo microphone). While this configuration is less flexible than expensive B-Format based systems, as will be discussed, its advantages are higher channel separation, stereo backwards compatibility, and low cost.

While B-Format enables total flexibility in terms of defining a first order directional response for each surround sound signal channel, the limitation to first order and the corresponding low directivity compromise the surround sound quality due to low channel separation (perceived as compromised localization and spaciousness). The proposed technique does not provide total flexibility in terms of directional response steering, but generates signal channels with a higher corresponding directivity resulting in higher channel separation than lin-

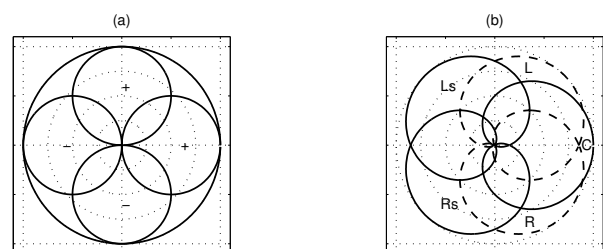


Fig. 1: (a) Directional responses of B-Format W , X , and Y signals. (b) An example of the directional responses of surround audio channels generated from a linear combination of the signals with responses as in (a).

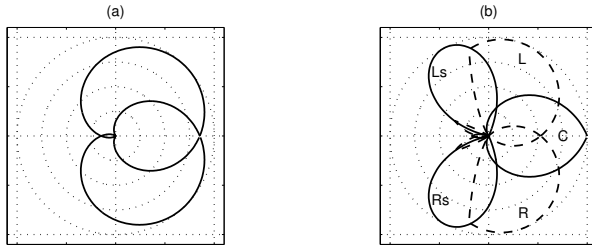


Fig. 2: (a) Directional responses of a coincident cardioid pair. (b) An example of the directional responses of surround audio channels generated from the signals with responses as in (a).

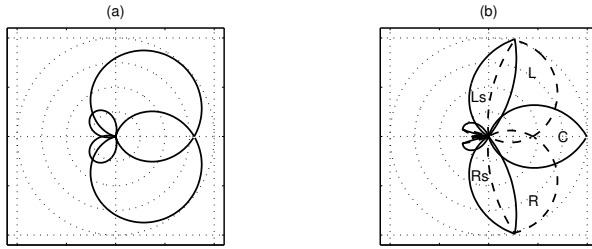


Fig. 3: (a) Directional responses of a coincident supercardioid pair. (b) An example of the directional responses of surround audio channels generated from the signals with responses as in (a).

ear B-Format decoding.

Figure 1(a) shows the directional responses of B-Format W , X , and Y signals, corresponding to an omni and two dipole responses. Examples of directional responses for generating a surround signal from B-Format are shown in Figure 1(b). In this case, a cardioid response was chosen for generating each of the five main audio channels, indicated as L (left front), R (right front), C (center), Ls (left surround), and Rs (right surround).

Figure 2(a) shows the directional responses of a pair of coincident cardioid signals suitable as inputs for the proposed processing. Figure 2(b) shows an example of the achieved effective directional responses of the generated multi-channel surround channels.

Another example of the achieved directional responses is illustrated in Figure 3, where the super-cardioid input microphone responses are shown in Panel (a) and the surround responses in Panel (b), respectively.

The previous examples imply that while rear center

sound can not be captured with unity gain by the proposed technique, the directional responses are more directive than B-Format based (linear) responses.

The proposed approach can be viewed as a simulation of different microphone signals for the different multi-channel surround audio channels with specific directional and diffuse responses. Different approaches for generating multi-channel surround from stereo microphone signals are [8, 9, 10]. These approaches are parametric, i.e. they are based on direction of arrival and direct/diffuse sound power estimates and use a spatial cues based renderer to generate the surround audio channels. The proposed technique is in this sense not parametric and generates each channel of the surround audio signal with an independent “highly directive microphone simulation” process, related to an adaptive beamformer [11] principle [12, 13, 14].

The paper is organized as follows. Section 2 reviews and generalizes a recently proposed technique [15] for modifying the directional responses of a coincident pair of microphones and explains how to generate L , R , Ls , and Rs audio channels. Section 3 describes how to generate a center channel from the same coincident pair stereo signal. Conclusions are in Section 4.

2. GENERATING FRONT AND SURROUND SIDE CHANNELS

The left and right coincident microphone signals are denoted $x_1(n)$ and $x_2(n)$, where n is the discrete time index. The signals are processed using a time-frequency representation based on a short-time Fourier transform (STFT). The STFT spectra of $x_1(n)$ and $x_2(n)$ are denoted $X_1(k, i)$ and $X_2(k, i)$, respectively, where k is the (sub-sampled) time index and i is the frequency index.

The technique described in [15] and the generalization presented here assume that microphone signal $X_1(k, i)$ can be written as

$$X_1(k, i) = a(k, i)X_2(k, i) + N_2(k, i), \quad (1)$$

where $X_2(k, i)$ and $N_2(k, i)$ are un-correlated and $a(k, i)$ is a time and frequency dependent gain factor relating to the cross-talk between the microphone signals $X_1(k, i)$ and $X_2(k, i)$. A modified left microphone signal $X_1(k, i)$ with a different directional and diffuse response is generated with

$$Y_1(k, i) = c(k, i) (X_1(k, i) - \min\{a(k, i), q\}X_2(k, i)), \quad (2)$$

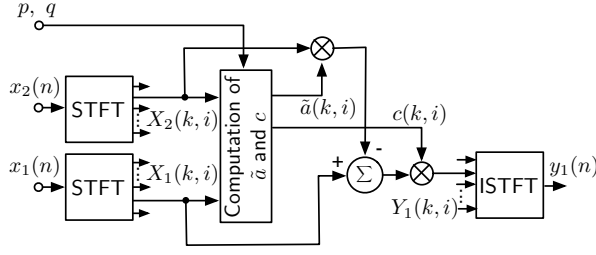


Fig. 4: Schematic diagram of the processing for generating a modified side channel. Processing for one subband signal is shown.

with $a(k, i)$ estimated as prediction gain

$$a(k, i) = \text{Re} \left\{ \frac{E\{X_1(k, i)X_2(k, i)^*\}}{E\{X_1(k, i)X_1(k, i)^*\}} \right\}, \quad (3)$$

where $\text{Re}\{\cdot\}$ is the real part of a complex number, $E\{\cdot\}$ is a short-time averaging function, and $*$ is the complex conjugate operator. As will be seen later, the limit q in (2) determines the directivity and direction of the directional response of $Y_1(k, i)$. The post-scaling factor $c(k, i)$ controls output gain and diffuse response of the output signal $Y_1(k, i)$.

If super-cardioid or other tailed cardioid microphones are used, the gain factor $a(k, i)$ (3) will not always be positive due to the negative rear-lobes of the directional responses of these microphones. This motivates to not only apply an upper but also a lower limit to the gain factor $a(k, i)$. Thus, the output signal is generated as

$$Y_1(k, i) = c(k, i) (X_1(k, i) - \tilde{a}X_2(k, i)), \quad (4)$$

where

$$\tilde{a}(k, i) = \max\{\min\{a(k, i), q\}, p\}, \quad (5)$$

with the limits $q \geq 0$ and $p \leq 0$.

Figure 4 summarizes the processing carried out to generate a modified side channel. The two given directional microphone signals, $x_1(n)$ and $x_2(n)$, are converted to their corresponding STFT representations. Further processing is shown for one subband signal. The parameters $\tilde{a}(k, i)$ and $c(k, i)$ are estimated and the subband signals of the output signal, $Y_1(k, i)$, are computed. The subbands of the output signal are converted back to the time domain using an inverse STFT (ISTFT), resulting in the time domain output signal $y_1(n)$.

The next two sub-sections describe how q , p , and $c(k, i)$ relate to the directional response and diffuse response of $Y_1(k, i)$, respectively.

2.1. Directional Response

For sound arriving from only one direction, the signals measured by two coincident microphones, pointing towards directions ϕ_0 and $-\phi_0$, can be written as

$$\begin{aligned} X_1(k, i) &= (v + w \cos(\phi - \phi_0))S(k, i) \\ X_2(k, i) &= (v + w \cos(\phi + \phi_0))S(k, i), \end{aligned} \quad (6)$$

where $S(k, i)$ is the short time spectrum of the sound and ϕ is the direction from which the sound is arriving. The constants v and w specify the shape of the input microphones' directional response. Table 1 shows the corresponding weights for cardioid, super-cardioid¹, and hyper-cardioid microphones. The corresponding directional responses are

$$\begin{aligned} d_{X_1}(\phi) &= v + w \cos(\phi - \phi_0) \\ d_{X_2}(\phi) &= v + w \cos(\phi + \phi_0). \end{aligned} \quad (7)$$

Table 1: Weights v and w and the corresponding microphone directional responses.

v	w	Name
1/2	1/2	Cardioid
1/3	2/3	Super-Cardioid
1/4	3/4	Hyper-Cardioid

Figure 2(a) shows the directional responses of $X_1(k, i)$ and $X_2(k, i)$ for $\phi_0 = 45^\circ$ and cardioid microphones. A super-cardioid coincident pair of responses with $\phi_0 = 45^\circ$ is shown in Figure 3(a). For simplicity, we are in the remaining part of the paper always showing examples based on such cardioid and super-cardioid coincident pair input signals. But note that the proposed processing is in principle applicable not only to first order signals but also to other signals with associated directional responses.

The directional response of the signal $Y_1(k, i)$ (4) without post-scaling ($c(k, i) = 1$) is

$$\begin{aligned} d(\phi) &= d_{X_1}(\phi) - \tilde{a}(k, i)d_{X_2}(\phi) \\ &= v(1 - \tilde{a}(k, i)) + \\ &\quad w(\cos(\phi - \phi_0) - \tilde{a}(k, i)\cos(\phi + \phi_0)). \end{aligned} \quad (8)$$

¹ Super-cardioid is sometimes also defined with $v = \frac{1}{2}(\sqrt{3} - 1)$ and $w = \frac{1}{2}(3 - \sqrt{3})$.

For direct sound, $X_1(k, i)$ and $X_2(k, i)$ are coherent. From this it follows that $Y_1(k, i)$ (4) is zero, except when the gain factor (3) is limited, i.e. $a(k, i) > q$ or $a(k, i) < p$ in (5).² Given this insight, the directional response can be specified by selecting for each direction one of the of the following cases:

- If $d_{X_1}(\phi) \geq 0$ and $d_{X_2}(\phi) \geq 0$, then the directional response is:

$$d(\phi) = \max\{v(1 - q) + w(\cos(\phi - \phi_0) - q\cos(\phi + \phi_0)), 0\}. \quad (9)$$

- If $d_{X_1}(\phi) \geq 0$ and $d_{X_2}(\phi) < 0$, then the directional response is:

$$d(\phi) = \max\{v(1 + p) + w(\cos(\phi - \phi_0) + p\cos(\phi + \phi_0)), 0\}. \quad (10)$$

- If $d_{X_1}(\phi) < 0$ and $d_{X_2}(\phi) \geq 0$, then the directional response is:

$$d(\phi) = \min\{v(1 + p) + w(\cos(\phi - \phi_0) + p\cos(\phi + \phi_0)), 0\}. \quad (11)$$

- If $d_{X_1}(\phi) < 0$ and $d_{X_2}(\phi) < 0$, then the directional response is:

$$d(\phi) = \min\{v(1 - q) + w(\cos(\phi - \phi_0) - q\cos(\phi + \phi_0)), 0\}. \quad (12)$$

In the previously discussed case, i.e. for sound arriving only from one direction, the post-scaling factor $c(k, i)$ is chosen such that the maximum gain of the resulting response is equal to one, i.e.

$$c_1 = \frac{1}{\max_{\phi} d(\phi)}. \quad (13)$$

²The short-time averaging operation, i.e. $E\{\cdot\}$, uses an adaptive averaging time interval of length between 15 and 80 ms and is assumed to adapt instantly. Thus, in the discussion it is assumed that if $X_1(k, i)$ and $X_2(k, i)$ are coherent, then $a(k, i)X_2(k, i)$ is the same as $X_1(k, i)$. In practice, effective instant adaptation is confirmed in the sense that conventional microphone measurement systems can measure the directional responses of the generated signals, e.g. similarly as implied by the measurements in [14]. Stability of the spatial image of the generated surround signals also implies that from a perceptual point of view the adaptation is fast enough.

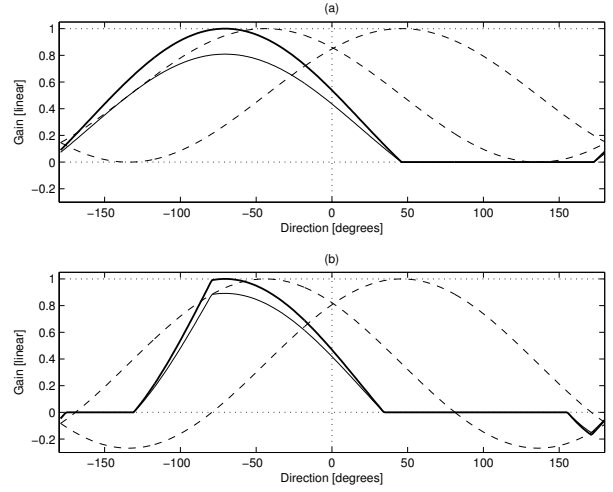


Fig. 5: The two panels show example directional responses for the cases of (a) cardioid and (b) supercardioid coincident pair microphones with $\phi_0 = 45^\circ$. The directional response parameters are $q = 0.5$ and $p = -4$. Both panels show the directional responses of $X_1(k, i)$ and $X_2(k, i)$ (dashed) and the directional response of $Y_1(k, i)$ before post-scaling (solid, thin) and after post-scaling (solid, bold).

Panel (a) and (b) of Figure 5 illustrate example directional responses of $Y_1(k, i)$ for a cardioid and supercardioid coincident pair with $\phi_0 = 45^\circ$ as input signals, respectively. The directional response parameters for both cases are $q = 0.5$ and $p = -4$ (the lower negative limit p is not necessary for the case of using cardioids since then $a(k, i) \geq 0$). The responses of $X_1(k, i)$ and $X_2(k, i)$ are shown as dashed lines. The resulting responses without post-scaling ($c(k, i) = 1$) are indicated by solid thin lines. Note that the maximum of these responses is smaller than one. The responses after post-scaling with c_1 (13) are shown as bold solid lines in the figure.

To illustrate the effect of the gain factor limit q on the directional responses when using a cardioid coincident pair, Figure 6 illustrates the corresponding responses for different values of q . The case $q = 0$ is equivalent to the two input signal directional responses. Small limits result in channels suitable for front left and right and larger limits are used for generating the surround audio channels. The signals with directional responses pointing towards the right (R, Rs) are generated similarly as those

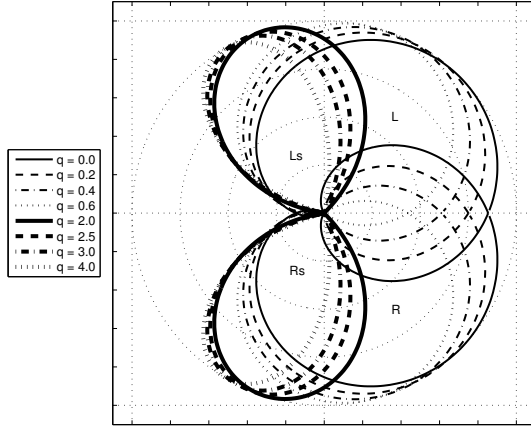


Fig. 6: The original microphone directional responses ($q = 0$, thin solid) and directional responses for different values of q are shown for cardioid input signals.

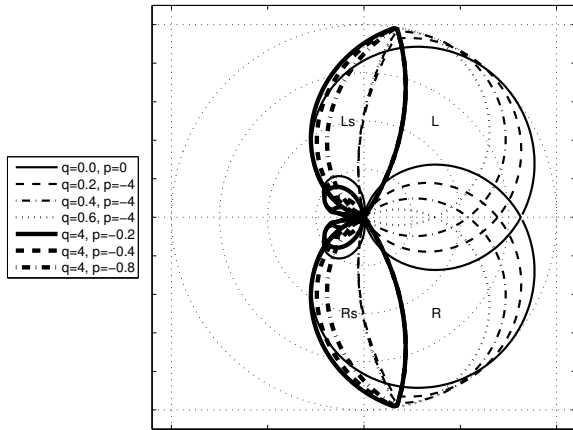


Fig. 7: The original directional microphone responses ($q = 0$ and $p = 0$, thin solid) and directional responses for different values of q and p are shown for super-cardioid input signals.

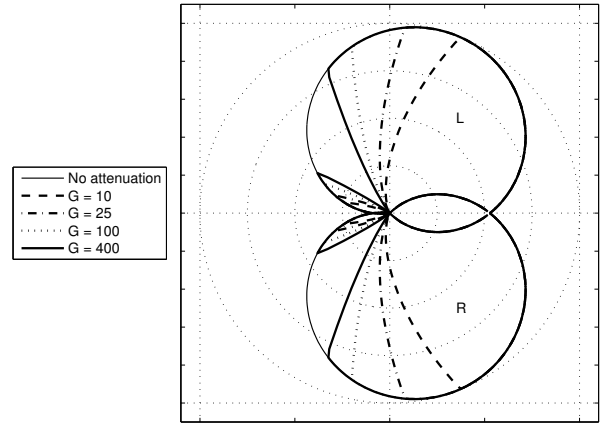


Fig. 8: Front left and right channel directional responses with different degrees of rear attenuation.

pointing towards the left ($Y_1(k, i)$) by exchanging $X_1(k, i)$ and $X_2(k, i)$ in the equations. The Ls and Rs responses in Figure 2(b) are other examples of achieved directional responses, using $q = 3$.

Similar data is shown for the case of super-cardioid microphones in Figure 7. The input signal directional responses are similar to the case when using $q = 0$ and $p = 0$. Front channels use a small q and a p with larger magnitude. To generate surround audio channels, the magnitude of q is chosen larger than the magnitude of p . Similar example L, R, Ls, and Rs responses are shown in Figure 3(b), using $q = 4$ and $p = -0.5$ for L and R, and $q = 0.5$ and $p = -4$ for Ls and Rs.

In the cardioid microphone case, shown in Figure 6, the front (L, R) and surround (Ls, Rs) directional responses are more overlapping than desired. This can be prevented by attenuating the response towards the rear, i.e. as the level difference increases. The front left channel $Y_1(k, i)$ (4) is multiplied with a gain factor

$$g(k, i) = \min \left\{ G \frac{E\{X_2(k, i)X_2(k, i)^*\}}{E\{X_1(k, i)X_1(k, i)^*\}}, 1 \right\}, \quad (14)$$

where the positive constant G relates inversely to how quickly the directional response is attenuated towards the rear. When generating the front R channel $Y_2(k, i)$, $g(k, i)$ is computed by exchanging $X_1(k, i)$ and $X_2(k, i)$ in (14).

Figure 8 shows example L and R channels with $q = 0.5$ and rear attenuation (14) with different values for G . The

L and R channels shown in Figure 2(b) use $q = 0.5$ and $G = 20$.

2.2. Post-scaling factor

As mentioned before, the post-scaling factor $c(k, i)$ in (4) is equal to c_1 (13) when there is only sound from one direction, to mimic a directional response with unity look-direction gain. The other extreme case is when there is only diffuse sound. In this case, one may want to use a different post-scaling factor $c(k, i) = c_2$ in order to control the diffuse response independently from the directional response. When there is a mix of direct and diffuse sound, a post-scaling factor between c_1 and c_2 is used, i.e.

$$c(k, i) = (1 - \alpha(k, i))c_1 + \alpha(k, i)c_2, \quad (15)$$

where $\alpha(k, i)$ is a diffuseness indicator, e.g. [13]

$$\alpha = \frac{\frac{1 - \max\{\Phi_{12}, \Phi_{\text{diff}}\}}{1 - \Phi_{\text{diff}}} \min\{E\{X_1 X_1^*\}, E\{X_2 X_2^*\}\}}{\frac{1}{2}E\{X_1 X_1^*\} + \frac{1}{2}E\{X_2 X_2^*\}}, \quad (16)$$

with $\Phi_{12}(k, i)$ and Φ_{diff} being the estimated and diffuse field cross-correlation coefficient,³ respectively. For brevity of notation the time and frequency indices have been ignored in (16).

When rear-attenuation is used (14) for generating the front L and R channels, the post-scaling including rear-attenuation is

$$c(k, i) = (1 - \alpha(k, i))g(k, i)c_1 + \alpha(k, i)c_2. \quad (17)$$

3. GENERATING A CENTER CHANNEL

The technique that is used to generate the center channel is similar to [16] with the addition that the diffuse response can be determined. It is assumed that the right microphone input signal can be written as

$$X_2(k, i) = a_1(k, i)X_1(k, i) + N_1(k, i), \quad (18)$$

where $X_1(k, i)$ is the left microphone input signal, $a_1(k, i)$ is a scale factor, and $N_1(k, i)$ is a signal un-correlated with $X_1(k, i)$. The scale factor is estimated as

$$a_1(k, i) = \text{Re} \left\{ \frac{E\{X_1(k, i)X_2(k, i)^*\}}{E\{X_1(k, i)X_1(k, i)^*\}} \right\}. \quad (19)$$

³ Φ_{diff} is a frequency-independent constant for coincident microphones. If the microphones are not coincident, frequency dependence has to be considered.

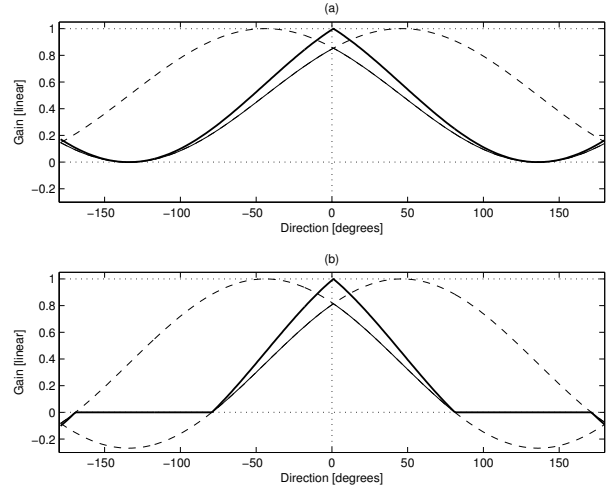


Fig. 9: The two panels show example center channel directional responses for the cases of (a) cardioid and (b) super-cardioid coincident pair microphones with $\phi_0 = 45^\circ$. Each panel shows the directional responses of $X_1(k, i)$ and $X_2(k, i)$ (dashed), the directional response of $C_1(k, i)$ and $C_2(k, i)$ (solid, thin), and the directional response of $C(k, i)$ (solid, bold).

The center channel contribution from the left microphone signal is

$$C_1(k, i) = \left((1 - \alpha(k, i)) \max\{\min\{a_1(k, i), 1\}, 0\} + \alpha(k, i)d \right) X_1(k, i), \quad (20)$$

where $\alpha(k, i)$ (16) is the diffuseness indicator and d is a constant relating to the diffuse response of $C_1(k, i)$, i.e. determining how much diffuse sound $C_1(k, i)$ contains.

Similarly, the center channel contribution from the right microphone signal is

$$C_2(k, i) = \left((1 - \alpha(k, i)) \max\{\min\{a_2(k, i), 1\}, 0\} + \alpha(k, i)d \right) X_2(k, i), \quad (21)$$

with

$$a_2(k, i) = \text{Re} \left\{ \frac{E\{X_1(k, i)X_2(k, i)^*\}}{E\{X_2(k, i)X_2(k, i)^*\}} \right\}. \quad (22)$$

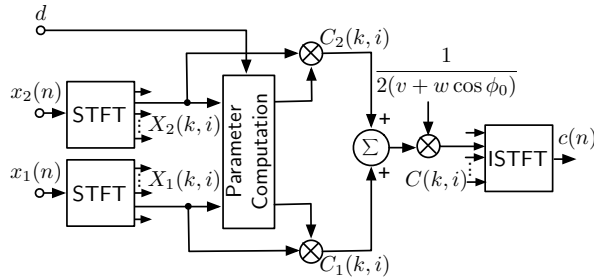


Fig. 10: Schematic diagram of the processing for generating a center channel. Processing for one subband signal is shown.

The center channel is computed by adding $C_1(k, i)$ and $C_2(k, i)$ and scaling the resulting signal such that its maximum response is unity

$$C(k, i) = \frac{C_1(k, i) + C_2(k, i)}{2(v + w \cos \phi_0)}. \quad (23)$$

3.1. Directional response

First we consider the directional response of $C_1(k, i)$ in (20). For direct sound, when $X_1(k, i)$ and $X_2(k, i)$ are coherent, $\alpha(k, i)$ is zero and the scaling of $X_1(k, i)$ in (20) with $\max\{\min\{a_1(k, i), 1\}, 0\}$ results in a $C_1(k, i)$ with a power (and thus directional response) equal to the weaker signal of $X_1(k, i)$ and $X_2(k, i)$. Further, the zero lower limiting of the factor $\max\{\min\{a_1(k, i), 1\}, 0\}$ in (20) results in a zero response whenever $a_1(k, i)$ is negative. Based on the previous reasoning, the directional response of $C_1(k, i)$ is

$$d_{C_1}(\phi) = \min\{d_{X_1}(\phi), d_{X_2}(\phi)\} \max\{d_{X_1}(\phi)d_{X_2}(\phi), 0\}. \quad (24)$$

Examples of cardioid and super-cardioid directional responses of $X_1(k, i)$ and $X_2(k, i)$ are illustrated as dashed lines in Panels (a) and (b) in Figure 9. The minimum of these, i.e. the directional responses of $C_1(k, i)$, are illustrated as thin solid lines.

Due to the left-right symmetry of the directional responses of $X_1(k, i)$, $X_2(k, i)$, and $C_1(k, i)$, $C_2(k, i)$ has the same directional response as $C_1(k, i)$. Example directional responses of $C(k, i)$ (23) are shown as bold solid lines in Panels (a) and (b) of Figure 9. The same responses in polar coordinates are the center responses in Panels (b) of Figures 2 and 3.

Figure 10 summarizes the processing carried out for generating the center channel. The two given directional microphone signals, $x_1(n)$ and $x_2(n)$, are converted to their corresponding STFT representations. Further processing is shown for one subband signal. The parameters are estimated and applied to the input subband signals. The resulting output subband signals are converted back to the time domain using an ISTFT, resulting in the time domain output signal $c(n)$.

4. CONCLUSIONS

A scheme has been proposed to generate multi-channel surround audio signals from a coincident pair of cardioid or tailed cardioid signals. A time-frequency adaptive process combines the left and right microphone signals to generate the multi-channel surround audio channels. The processing can be viewed as adaptive beamformers, not with signal-to-noise ratio optimization targets as usual, but with targets to simulate highly directive microphones with different directions, directional responses, and diffuse responses suitable for surround audio channels.

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