Hindawi Publishing Corporation
EURASIP Journal on Advances in Signal Processing
Volume 2008, Article ID 732895, 14 pages
doi:10.1155/2008/732895

Research Article

Binaural Rendering in MPEG Surround

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Received 29 June 2007; Revised 12 November 2007; Accepted 21 December 2007

Recommended by Antonio Ortega

This paper describes novel methods for evoking a multichannel audio experience over stereo headphones. In contrast to the conventional convolution-based approach where, for example, five input channels are filtered using ten head-related transfer functions, the current approach is based on a parametric representation of the multichannel signal, along with either a parametric representation of the head-related transfer functions or a reduced set of head-related transfer functions. An audio scene with multiple virtual sound sources is represented by a mono or a stereo downmix signal of all sound source signals, accompanied by certain statistical (spatial) properties. These statistical properties of the sound sources are either combined with statistical properties of head-related transfer functions to estimate "binaural parameters" that represent the perceptually relevant aspects of the auditory scene or used to create a limited set of combined head-related transfer functions that can be applied directly on the downmix signal. Subsequently, a binaural rendering stage reinstates the statistical properties of the sound sources by applying the estimated binaural parameters or the reduced set of combined head-related transfer functions directly on the downmix. If combined with parametric multichannel audio coders such as MPEG Surround, the proposed methods are advantageous over conventional methods in terms of perceived quality and computational complexity.

1. INTRODUCTION

The synthesis of virtual auditory scenes has been an ongoing research topic for many years [1–5]. The aim of so-called binaural rendering systems is to evoke the illusion of one or more sound sources positioned around the listener using stereo headphones. The positions of the sound sources can preferably be modified in terms of the perceived azimuth, elevation, and distance. More advanced systems also include room acoustic models to simulate the acoustical properties such as reflecting walls within the virtual space.

Binaural rendering has benefits in the field of research, simulation, and entertainment [6]. Especially in the field of entertainment, the virtual auditory scene should sound very compelling and "real." In order to achieve such a realistic percept, several aspects have to be taken into account, such as the change in sound source positions with respect to head movement [7], room acoustic properties such as early reflections and late reverberation [8], and using system personalization to match the anthropometric properties of the individual user [9–11]. Because of the complex nature of current state-of-the-art systems, several concessions are required for feasible implementations (cf. [12]), especially if the number of sound sources that has to be rendered simultaneously is large.

Recent trends in consumer audio show a shift from stereo to multichannel audio content as well as a shift from immobile to mobile devices. These developments cause additional constraints on transmission and rendering systems. Firstly, the number of audio channels that has to be transmitted increases significantly (e.g., from two to six). The corresponding increase in transmission bandwidth for conventional, discrete-channel audio coders is often undesirable and sometimes even unavailable. Secondly, consumers often use headphones for audio rendering on a mobile device. To experience the benefit of multichannel audio, a dedicated binaural rendering system is required. This can be quite a challenge given the limited processing power and battery life of mobile devices.

In this paper, two novel binaural rendering processes will be described, which exploit recent advances in parametric multichannel audio compression. Both methods operate on a parametric representation of a multichannel original signal and a corresponding downmix signal, as is defined by the recently finalized MPEG Surround standard [13] for multichannel audio compression. An outline of the basic principle of MPEG Surround is given in Section 2. The first method, referred to as "parametric approach," is based on the analysis and synthesis of perceptually relevant attributes "binaural parameters" of a virtual auditory scene. This method is especially suitable for low-complexity simulation of anechoic situations (possibly extended with parametric methods for room acoustics simulation). The analysis and synthesis of binaural parameters is outlined in Sections 3.2 and 3.3, and the integration of this method into MPEG Surround is described in Section 5. The second method is based on convolution-based synthesis that can be applied directly on the downmix signal, without the need of independent channel signals (as in conventional methods). This method, which is referred to as a "morphed-filter" approach, will be outlined in Section 4. It is especially suitable to simulate echoic virtual environments and/or if the parametric approximation of binaural parameters is not sufficiently accurate. Finally, the two different methods are evaluated in the context of MPEG Surround by means of listening tests.

2. MPEG SURROUND

MPEG Surround [13–17] is a novel parametric method for efficient transmission of multichannel audio. In this audio coding format, a multichannel audio signal is represented as a downmix signal (typically mono or stereo) and a set of "spatial parameters" that, among other aspects, describe the statistical relations of the original multichannel signals in terms of (relative) signal powers and correlation coefficients. The processing flow of MPEG Surround is visualized in Figure 1. An MPEG Surround encoder (left panel of Figure 1) generates a mono or stereo downmix from a multichannel input signal and accompanying spatial parameters. These parameters are extracted for individual time/frequency tiles of the input signals. The bandwidth of each tile is approximately equal to one critical band, and the duration is in the order of tens of milliseconds. The downmix can be encoded using existing compression methods (legacy coders). A multiplexer combines the resulting downmix bit stream with the parameter bit stream to form an output bit stream. The decoder, shown in the right panel of Figure 1, performs the inverse process to generate the multichannel output signals. The coding efficiency provided by the parametric approach to represent spatial attributes is quite significant; a parameter bit rate of about 6 to 12 kbps (in addition to the bit rate required for the mono or stereo coder) suffices to achieve high-quality multichannel audio [16–18].

The MPEG Surround coder operates in a hybrid quadrature mirror filter (QMF) bank domain [19] to enable independent processing of individual time/frequency tiles. The spatial parameter extraction process (at the encoder side) and the spatial synthesis process (at the decoder side) are all performed in this filterbank domain. The spatial encoding process is provided by so-called two-to-One (TTO) and three-to-two (TTT) encoding blocks, as outlined in Figure 2. The first type, which is essentially similar to a "parametric stereo" coder [19–24] encodes a stereo signal by means of a mono signal, a channel level difference (CLD) and an interchannel cross-correlation (ICC) parameter. The second type (TTT block) represents three input signals (typically, a left, right and center signal) as a stereo downmix accompanied by two channel prediction coefficients (CPCs) that enable decoder-side prediction of a third signal from the two downmix channels. A possible prediction loss may be compensated for by transmission of an additional ICC parameter (see [14, 16, 17, 25] for more details).

Several TTO and TTT encoding blocks (E_i) can be connected to create a certain tree configuration. Two examples of such tree configurations are shown in Figure 2. The left panel of Figure 2 shows a combination of 5 TTO encoding blocks to represent a 6-channel input (l_f, r_f, c, l_s, r_s, and LFE for the left front, right front, center, left surround, right surround, and low frequency effects channel, resp.) as a mono signal x accompanied by spatial parameters (P_i). A tree configuration for stereo output, involving 3 TTO encoding blocks and one TTT encoding block, is shown in the right panel, resulting in a stereo downmix pair x_l, x_r.

3. BINAURAL PARAMETER ANALYSIS AND SYNTHESIS

3.1. Background

There is evidence that spatial parameters such as employed in MPEG Surround and related spatial coding approaches (see [14, 20, 26, 27]) can also be employed to describe so-called head-related transfer functions (HRTFs) that are used for binaural synthesis. Sound-source localization in the horizontal plane is facilitated by interaural time differences (ITDs) and interaural level differences (ILDs) [5, 28, 29], caused by the relative path lengths and acoustic shadow effect of the head. The properties of sound propagation also result in an intricate frequency dependence of these cues. Sound source elevation is predominantly facilitated by elevation-dependent spectral peaks and notches that are superimposed on the original sound source spectrum [11]. The perceived distance of a sound source is based on the overall signal level, the ratio of direct and reverberant sound, and spectral cues [1, 2, 30, 31].

All acoustical cues that determine the perceived position of the sound source are captured by a pair of HRTFs. The corresponding time-domain impulse responses are denoted HRIRs (head-related impulse responses). If individualized HRTFs are used to simulate a virtual sound source, subjects are not able to discriminate between real and virtual sound sources [28, 32, 33]. This result indicates that HRTFs indeed supply sufficient information for adequate binaural rendering. However, several investigations have shown that HRTFs may comprise pronounced properties in the signal domain that seem perceptually *irrelevant*. For example, it has been shown that for low frequencies, ITDs dominate sound source localization, while at high frequencies, ILDs and spectral cues (peaks and troughs resulting from reflections

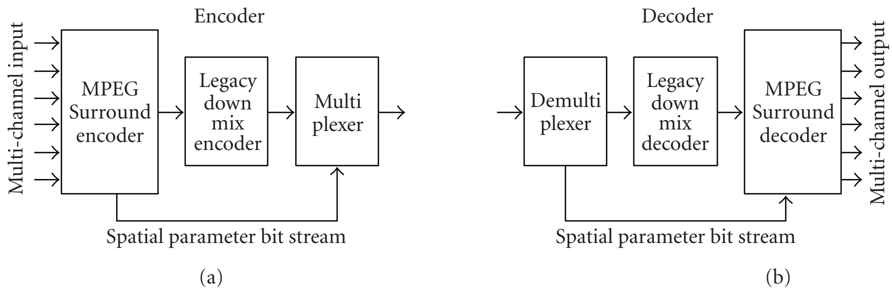

FIGURE 1: Concept of MPEG Surround. A multichannel audio signal is represented as a downmix signal and accompanying spatial parameters (MPEG Surround encoder). The downmix can be encoded using an existing (legacy) compression method. The decoder separates the spatial parameters from the core coder bitstream (demultiplexer), decodes the downmix, and reconstructs multichannel audio by reinstating the spatial properties (MPEG Surround decoder).

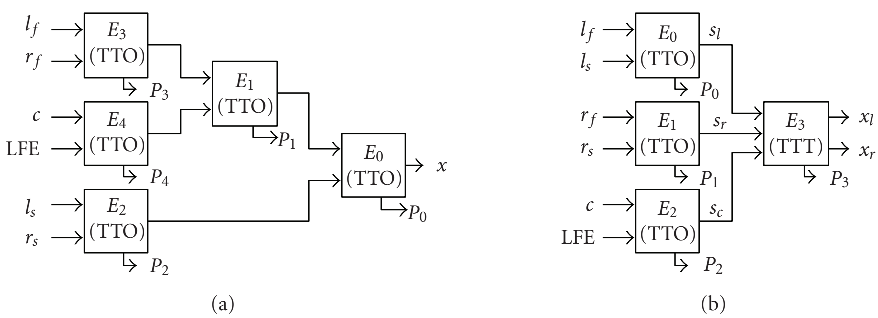

FIGURE 2: Two encoder tree configurations for 6-channel input and a mono downmix (left panel) or a stereo downmix (right panel). Each block (E_i) represents a TTO or TTT encoding block and generates a set of parameters (P_i).

of shoulders and the pinnae) are more important [34]. Other researchers have successfully demonstrated that the frequency-dependent ITD can be replaced by a constant, position-dependent ITD without perceptual consequences [14, 32, 35, 36]. A related finding is that the interaural time difference can be replaced by a constant interaural phase difference (IPD) within various frequency bands. The resulting piecewise constant-phase curve does not result in audible differences provided that the frequency bands are not broader than critical bands [14].

There is also considerable evidence that certain details of the HRTF magnitude spectra are irrelevant [37–39]. Specifically, it seems that *constant* spectral cues within critical bands (or frequency bands that follow the ERB scale [40]) are a sufficient requirement for high-quality binaural rendering [14, 38].

Given the commonalities between the parametric approach for audio compression and a parametric approach to describe HRTFs, these can be efficiently combined in a single binaural rendering application. In such a combined approach, the so-called "binaural parameters" are estimated representing simultaneous playback of all audio channels over a virtual standard loudspeaker setup [41]. The interrelations between the virtual loudspeaker signals are given by spatial parameters, while the relations between a virtual loudspeaker and the resulting ear-drum signals are described by HRTF parameters. The binaural parameter estimation process is outlined in the next section.

3.2. *Binaural parameter analysis*

In conventional binaural rendering systems, a sound source i with associated discrete-sampled time-domain signal z_i is rendered at a certain position by convolving the signal with a pair of head-related impulse responses $h_{L,i}$, $h_{R,i}$, for the left and right ears, respectively, to result in binaural signals $y_{L,i}$, $y_{R,i}$:

$$y_{m,i} = z_i * h_{m,i}, \tag{1}$$

with $m \in \{L, R\}$. This process is visualized in the left panel of Figure 3.

Expressed in a (complex-valued) subband domain with time-index k and frequency band index b, the power of signal $y_{m,i}(k,b)$ within a certain analysis frame $k = 0, \ldots, K-1$ is given by

$$\sigma^2_{y_{m,i}}(b) = \frac{1}{K}\sum_k y_{m,i}(k,b)\,y^*_{m,i}(k,b), \tag{2}$$

with $(*)$ the complex conjugation operator. If the HRTF magnitude spectra are locally stationary (i.e., constant within the frequency band b), this can be simplified to

$$\sigma^2_{y_{m,i}}(b) = \sigma^2_{h_{m,i}}(b)\,\sigma^2_{z_i}(b), \tag{3}$$

with $\sigma^2_{h_{m,i}}(b)$ the power within parameter band b of HRIR $h_{m,i}$ and $\sigma^2_{z_i}(b)$ the power of the source signal z_i in parameter band b within the current analysis frame.

Thus given the local stationarity constraint, the power in a certain parameter band b at the level of the ear drums follows from a simple multiplication of the power of the sound source and the power of the HRTF in corresponding parameter bands. In other words, statistical properties of binaural signals can be deducted from statistical properties of the source signal and from the HRTFs. This parameter-based approach is visualized in the right panel of Figure 3. Similar derivations lead to estimates of the interaural-phase difference (IPD) between the signals $y_{L,i}$ and $y_{R,i}$:

$$\text{IPD}(b) = \angle\left(\sum_k y_{L,i}(k,b)y^*_{R,i}(k,b)\right). \tag{4}$$

Under the assumption of local stationarity of interaural HRTF phase spectra, the IPD can be derived directly from the HRTF spectra themselves, without involvement of the sound source signal:

$$\text{IPD}(b) = \phi_i(b), \tag{5}$$

with $\phi_i(b)$ the average interaural-phase difference of the HRTF pair corresponding to position i and parameter band b:

$$\phi_i(b) = \angle\left(\sum_k h_{L,i}(k,b)h^*_{R,i}(k,b)\right). \tag{6}$$

The equations above assume local stationarity of HRTF magnitude and interaural phase difference spectra to estimate the resulting binaural parameters. This stationarity constraint has been shown to result in correct sound-source localization properties [14]. However, strong deviations from stationarity within analysis bands result in a decrease in the interaural coherence (IC) for certain frequency bands, since the relation between the two HRTF spectra within the band of interest cannot be accurately described by a single phase and level difference. Such decrease in the IC is perceived as a change in the spatial "compactness" [2]. To capture this property, the IC is estimated for each parameter band b. In our context, the coherence is defined as the absolute value of the average normalized cross-spectrum:

$$\text{IC}(b) = \frac{\left|\sum_k y_{L,i}(k,b)y^*_{R,i}(k,b)\right|}{K\sigma_{y_{L,i}}(b)\sigma_{y_{R,i}}(b)}. \tag{7}$$

The IC parameter has a dependency on the source signal z_i. The expected value is given by

$$\text{IC}(b) = \rho_i(b), \tag{8}$$

with

$$\rho_i(b) = \frac{\left|\sum_k h_{L,i}(k,b)h^*_{R,i}(k,b)\right|}{K\sigma_{h_{L,i}}(b)\sigma_{h_{R,i}}(b)}. \tag{9}$$

In summary, under the local stationarity constraint, the binaural parameters σ_{y_L}, σ_{y_R}, IPD, and IC resulting from a single sound source can be estimated from the sound-source parameters σ_{z_i} and the HRTF parameters $\sigma_{h_{L,i}}$, $\sigma_{h_{R,i}}$, ϕ_i, and ρ_i.

For multiple simultaneous sound sources, conventional systems convolve each individual source signal i with an HRTF pair corresponding to the desired position, followed by summation:

$$y_m = \sum_i z_i * h_{m,i}. \tag{10}$$

The binaural parameters σ_{y_L}, σ_{y_R}, IPD, and IC between signals y_L, y_R resulting from the ensemble of simultaneous sound sources z_i can be estimated in a very similar way as described above, based on the sound source parameters σ_{z_i} and their mutual normalized correlation coefficients c_{i_1,i_2} on the one hand, and the HRTF parameters $\sigma_{h_{L,i}}$, $\sigma_{h_{R,i}}$, ϕ_i, and ρ_i on the other hand:

$$\sigma^2_{y_m} = \sum_i \left(\sigma^2_{h_{m,i}}\sigma^2_{z_i}\right) + \sum_{i_1}\sum_{i_2 \neq i_1}\sqrt{r_{m,i_1i_2}}c_{i_1,i_2}\cos\left(\frac{\phi_{i_1}-\phi_{i_2}}{2}\right), \tag{11}$$

with

$$r_{m,i_1i_2} = \sigma^2_{h_{m,i_1}}\sigma^2_{h_{m,i_2}}\sigma^2_{z_{i_1}}\sigma^2_{z_{i_2}}\rho_{i_1}\rho_{i_2}. \tag{12}$$

In a similar way, the IPD and IC are given by

$$\text{IPD} = \angle(\chi), \quad \text{IC} = \frac{|\chi|}{\sigma_{y_L}\sigma_{y_R}}, \tag{13}$$

with

$$\chi = \sum_i\left(e^{j\phi_i}\rho_i\sigma^2_{z_i}\sigma_{h_{L,i}}\sigma_{h_{R,i}}\right) + \sum_{i_1}\sum_{i_2\neq i_1}\left(e^{(j\phi_{i_1}+j\phi_{i_2})/2}c_{i_1,i_2}\sqrt{q_{i_1i_2}}\right), \tag{14}$$

with

$$q_{i_1i_2} = \sigma^2_{h_{L,i_1}}\sigma^2_{h_{R,i_2}}\sigma^2_{z_{i_1}}\sigma^2_{z_{i_2}}\rho_{i_1}\rho_{i_2}. \tag{15}$$

In the equations above, the subband index (b) is omitted for clarity. The reader is referred to [14] for a more detailed derivation of σ_{y_L}, σ_{y_R}, IPD, and IC.

3.3. Binaural parameter synthesis

3.3.1. Synthesis from mono downmix

In the case of an MPEG-Surround encoded signal with a mono downmix, the synthesis process comprises reinstating the binaural parameters on the mono downmix signal x of the object signals. Assuming incoherent source signals z_i, the downmix is given by

$$x = \sum_i z_i. \tag{16}$$

In the case of (partially) correlated source signals (i.e., the pairwise correlation coefficient c_{i_1,i_2} is nonzero for certain signal pairs), the downmix is preferably scaled in each frequency band and for each frame independently to ensure energy preservation (cf. [14, 16]). As a result, the power σ^2_x in

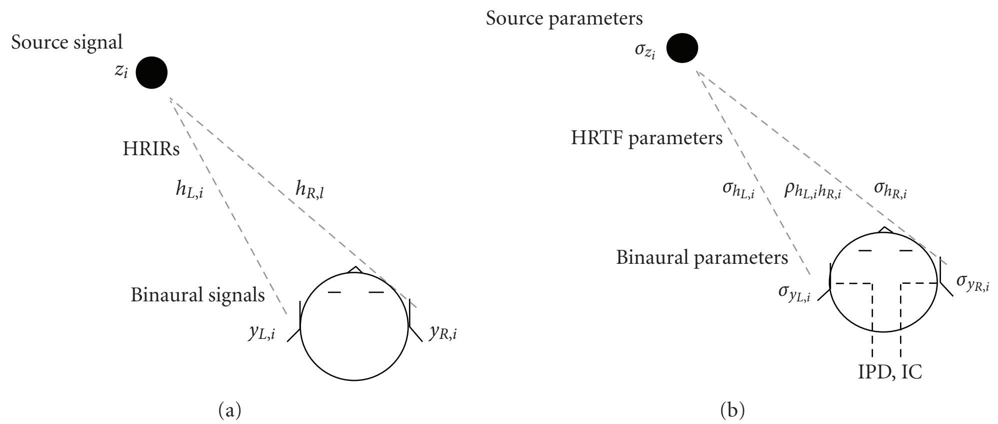

Figure 3: Synthesis of a virtual sound source by means of HRIR convolution (left panel) and by means of parametric representations (right panel).

each parameter band b of a downmix signal frame k is then given by

$$\sigma_x^2 = \sum_i \sigma_{z_i}^2. \tag{17}$$

The required binaural parameters are derived from HRTF parameters ($\sigma_{h_{L,i}}$, $\sigma_{h_{R,i}}$, ϕ_i, ρ_i) and signal parameters (σ_{z_i}, c_{i_1,i_2}) as described in Section 3.2. The signal parameters σ_{z_i} and c_{i_1,i_2} are assumed to be available as side information accompanying the down-mix x. In the case of MPEG Surround, the statistical properties of the input signals are described as pairwise level differences (CLDs) and correlations (ICCs) in a tree structure (cf. Figure 2, left panel), which need to be converted to relations between the original input channels. The $\mathrm{CLD}_i(b)$ is defined as the power ratio of the two input signals (q_1, q_2) in parameter band b of the encoding block TTO_i:

$$\mathrm{CLD}_i(b) = \frac{\sigma_{q_1}^2(b)}{\sigma_{q_2}^2(b)}. \tag{18}$$

Given the tree structure shown in the left panel of Figure 2, the powers of the input signals z_{l_f}, z_{l_s}, z_{r_f}, z_{r_s}, z_c are derived from the CLDs by combining the individual energy ratios of each TTO element:

$$\begin{aligned}
\sigma_{z_{l_f}}^2(b) &= \left(\frac{\mathrm{CLD}_0(b)}{1+\mathrm{CLD}_0(b)}\right)\left(\frac{\mathrm{CLD}_1(b)}{1+\mathrm{CLD}_1(b)}\right)\left(\frac{\mathrm{CLD}_3(b)}{1+\mathrm{CLD}_3(b)}\right),\\
\sigma_{z_{r_f}}^2(b) &= \left(\frac{\mathrm{CLD}_0(b)}{1+\mathrm{CLD}_0(b)}\right)\left(\frac{\mathrm{CLD}_1(b)}{1+\mathrm{CLD}_1(b)}\right)\left(\frac{1}{1+\mathrm{CLD}_3(b)}\right),\\
\sigma_{z_c}^2(b) &= \left(\frac{\mathrm{CLD}_0(b)}{1+\mathrm{CLD}_0(b)}\right)\left(\frac{1}{1+\mathrm{CLD}_1(b)}\right),\\
\sigma_{z_{l_s}}^2(b) &= \left(\frac{1}{1+\mathrm{CLD}_0(b)}\right)\left(\frac{\mathrm{CLD}_2(b)}{1+\mathrm{CLD}_2(b)}\right),\\
\sigma_{z_{r_s}}^2(b) &= \left(\frac{1}{1+\mathrm{CLD}_0(b)}\right)\left(\frac{1}{1+\mathrm{CLD}_2(b)}\right).
\end{aligned} \tag{19}$$

In the equations above, the LFE signal is assumed to be merged with the center speaker as one single signal, and hence the parameters of OTT_4 are absent in the equations above.

The $\mathrm{ICC}_i(b)$ is defined as the normalized cross-correlation coefficient of the two input signals of TTO_i. As can be observed from Figure 2, four ICC parameters (i.e., excluding TTO_4) are available to represent 10 unique pairwise correlation coefficients c_{i_1,i_2} of 5 input channels. This ill-defined problem is solved by a heuristic rule that all pairwise correlations are set to zero, except for

$$c_{l_f,r_f} = \mathrm{ICC}_3, \quad c_{l_s,r_s} = \mathrm{ICC}_2. \tag{20}$$

The reconstructed binaural signals $\hat{y}_L$, $\hat{y}_R$ can be obtained using a matrix operation $\mathbf{M}(b)$ that is derived for each parameter band (b):

$$\begin{bmatrix} \hat{y}_L(k,b) \\ \hat{y}_R(k,b) \end{bmatrix} = \mathbf{M}(b) \begin{bmatrix} x(k,b) \\ D(x(k,b)) \end{bmatrix}, \tag{21}$$

with $D(\cdot)$ a so-called "decorrelator" which generates a signal that has virtually the same temporal and spectral envelopes as its input but is independent from its input. This method of binaural synthesis is identical to the parameter synthesis method applied in "parametric stereo" decoders [20]. The matrix coefficients ensure that for each frame, the two binaural output signals $\hat{y}_L$, $\hat{y}_R$ have the desired levels, IPD and IC relations. A suitable solution for the synthesis matrix $\mathbf{M}(b)$ is given by (see [20] for details)

$$\mathbf{M}(b) = \begin{bmatrix} \lambda_L(b)\cos(\alpha(b)+\beta(b)) & \lambda_L(b)\sin(\alpha(b)+\beta(b)) \\ \lambda_R(b)\cos(-\alpha(b)+\beta(b)) & \lambda_R(b)\sin(-\alpha(b)+\beta(b)) \end{bmatrix}, \tag{22}$$

with $\lambda_L(b)$, $\lambda_R(b)$ two scale factors that determine the (complex) gain between the downmix signal and the left and right binaural output signals, respectively:

$$\lambda_L(b) = \frac{\sigma_{y_L(b)}}{\sigma_x(b)} e^{+j\mathrm{IPD}(b)/2}, \quad \lambda_R(b) = \frac{\sigma_{y_R(b)}}{\sigma_x(b)} e^{-j\mathrm{IPD}(b)/2}. \tag{23}$$

The angle $\alpha(b)$ determines the coherence between $\hat{y}_L$, $\hat{y}_R$ according to

$$\alpha(b) = \frac{1}{2}\arccos\left(\mathrm{IC}(b)\right), \tag{24}$$

while the angle $\beta(b)$ minimizes the decorrelator output signal:

$$\beta(b) = \tan\left(\frac{\sigma_{y_R}(b) - \sigma_{y_L}(b)}{\sigma_{y_R}(b) + \sigma_{y_L}(b)}\arctan\left(\alpha(b)\right)\right). \tag{25}$$

3.3.2. Extension to stereo downmixes

In the previous sections, binaural parameters were analyzed and reinstated from a mono downmix signal x. For several applications, however, it is beneficial to provide means to extend the downmix channel configuration to stereo. An example of a relevant application scenario is the synthesis of a virtual multichannel "home cinema setup" using a stereo downmix signal pair x_L, x_R accompanied by spatial parameters. This process will be discussed in the context of the MPEG Surround tree structure shown in the right panel of Figure 2. In the 3 TTO encoding blocks, input signals are pairwise combined to result in three intermediate signals s_L, s_R, and s_C. These intermediate signals are then combined into a stereo downmix pair x_L, x_R by a TTT encoding block according to

$$\begin{bmatrix} x_L \\ x_R \end{bmatrix} = \begin{bmatrix} 1 & 0 & \frac{1}{2}\sqrt{2} \\ 0 & 1 & \frac{1}{2}\sqrt{2} \end{bmatrix}\begin{bmatrix} s_L \\ s_R \\ s_C \end{bmatrix}. \tag{26}$$

The extracted CPC parameters enable reconstruction of the intermediate signals $\hat{s}_L$, $\hat{s}_R$, and $\hat{s}_C$ at the MPEG Surround decoder side (using a corresponding decoder block indicated by TTT^{-1}) according to

$$\begin{bmatrix} \hat{s}_L(k,b) \\ \hat{s}_R(k,b) \\ \hat{s}_C(k,b) \end{bmatrix} = \mathbf{M}_{\mathrm{TTT}}^{-1}(b)\begin{bmatrix} x_L(k,b) \\ x_R(k,b) \end{bmatrix}, \tag{27}$$

with an upmix matrix $\mathbf{M}_{\mathrm{TTT}}^{-1}(b)$ for each parameter band depending on the CPC parameters (see [16] for more details).

For each of the three reconstructed intermediate signals $\hat{s}_L$, $\hat{s}_R$, and $\hat{s}_C$, an individual 2×2 upmix matrix $\mathbf{W}(b)$ is computed for those virtual sources that are present in that particular downmix signal. In other words, one matrix $\mathbf{W}_{s_L}(b)$ is estimated to reinstate the binaural parameters resulting from channels l_f and l_s, one matrix $\mathbf{W}_{s_R}(b)$ to reinstate binaural parameters resulting from r_f and r_s, and one matrix to reinstate the binaural parameters from channel c, assuming that the content of the LFE channel is also reproduced by the center channel (i.e., $\mathrm{CLD}_2 = \infty$). The required channel powers σ_z are derived from the MPEG Surround OTT parameters (right panel of Figure 2) according to

$$\begin{aligned} \sigma_{l_f}^2 &= \left(\frac{\mathrm{CLD}_0}{1+\mathrm{CLD}_0}\right), \\ \sigma_{l_s}^2 &= \left(\frac{1}{1+\mathrm{CLD}_0}\right), \\ \sigma_{r_f}^2 &= \left(\frac{\mathrm{CLD}_1}{1+\mathrm{CLD}_1}\right), \\ \sigma_{r_s}^2 &= \left(\frac{1}{1+\mathrm{CLD}_1}\right). \end{aligned} \tag{28}$$

Furthermore, the channel correlation coefficients are assumed to be zero (i.e., $c_{i_1,i_2} = 0$, for $i_1 \neq i_2$). The derivation of the matrix elements is equal to the method described in Section 3.3.1, with the exception that the coherence (IC) for each individual matrix is assumed to amount to +1. This assumption is based on the observation that the coherence of these matrices predominantly represents coherence in a front/back direction, which is assumed to be a less salient cue than coherence in a left/right direction. Given a coherence value of +1, no decorrelator signal is required in the synthesis and hence each individual matrix simplifies to

$$\mathbf{W}_s(b) = \begin{bmatrix} \lambda_{L,s}(b) & 0 \\ \lambda_{R,s}(b) & 0 \end{bmatrix}. \tag{29}$$

Subsequently, the individual outputs of each 2×2 matrix operating on one intermediate signal are simply summed to result in the binaural output pair $\hat{y}_L$, $\hat{y}_R$:

$$\begin{aligned} \begin{bmatrix} \hat{y}_L(k,b) \\ \hat{y}_R(k,b) \end{bmatrix} &= \mathbf{W}_{s_L}(b)\begin{bmatrix} \hat{s}_L(k,b) \\ 0 \end{bmatrix} \\ &\quad + \mathbf{W}_{s_R}(b)\begin{bmatrix} \hat{s}_R(k,b) \\ 0 \end{bmatrix} + \mathbf{W}_{s_C}(b)\begin{bmatrix} \hat{s}_C(k,b) \\ 0 \end{bmatrix}. \end{aligned} \tag{30}$$

Given the fact that the intermediate signals $\hat{s}_L$, $\hat{s}_R$, and $\hat{s}_C$ follow from the downmix pair x_L, x_R given a matrix operation $\mathbf{M}_{\mathrm{TTT}}^{-1}(b)$ according to (27), the complete binaural rendering process can be written as a single, 2×2 matrix operation $\mathbf{M}(b)$ for each parameter band b:

$$\begin{bmatrix} \hat{y}_L(k,b) \\ \hat{y}_R(k,b) \end{bmatrix} = \mathbf{M}(b)\begin{bmatrix} x_L(k,b) \\ x_R(k,b) \end{bmatrix}. \tag{31}$$

4. MORPHED-FILTER APPROACH

4.1. Introduction

The parametric approach outlined in the previous section employs a lossy representation of HRTFs (using only spectral envelopes, average-phase differences, and coherences). In the case of echoic impulse responses (so-called binaural room impulse responses (BRIRs), or binaural room transfer functions (BRTFs)), the parametric approach is not capable of accurate modeling of all relevant perceptual aspects. In this case, a less compact HRTF or BRTF representation can be obtained by extending the 2×2 processing matrix in the time domain (i.e., having multiple "taps"). This extension is only defined for a stereo downmix and will be outlined below.

The basic principle is to combine the original set of HRTFs or BRTFs into a limited set of four impulse responses that can be directly applied on the stereo downmix. This is feasible when a representation of the original multichannel signal is available, which relies on stereo downmix and a set of spatial parameters, as is the case for MPEG Surround. The proposed method is beneficial since it only operates on four filters as opposed to ten filters normally used for binaural rendering of a five channel signal, and furthermore, it enables the use of echoic impulse responses (BRIRs). A design goal of the method is to maintain a waveform match with the conventional reference binaural signal (32) in situations where the MPEG Surround multichannel signal obtains a waveform match with the original multichannel signal. For a mono downmix this only happens for single loudspeaker sources, but for a stereo downmix the MPEG Surround decoding system enables waveform reconstruction for many two-loudspeaker combinations. The term "morphed-filter" approach refers to a dynamic combination of the front/back contributions which can be thought of as the creation of a virtual loudspeaker that for each time-frequency tile replaces a front/back loudspeaker pair. The corresonding HRTF data is interpolated in phase and amplitude with weights depending on the parametric surround side information.

4.2. Subband filter representations

The signal modifications of MPEG surround are performed in the domain of a complex modulated filter bank which is not critically sampled; see [19]. Its particular design allows for a given time-domain filter to be implemented at high precision by filtering each subband signal in the time direction with a separate filter. The resulting overall SNR for the filter implementation is in the 50 dB range with the aliasing part of the error significantly smaller. Moreover, these subband domain filters can be derived directly from the given time-domain filter. The filter conversion is specified in [13] and the details of its derivation can be found in [42].

We will consider a single fixed subband of the QMF filterbank and omit any subband indexing for clarity. The frequency resolution of the spatial parameters is adapted to this filterbank in the sense that there is only one parameter per subband. The reference output of the filtering approach is the superposition of the conventional single source contributions originating from each loudspeaker position, as given by (1). For the binaural rendering purpose, it is assumed that the contribution from the LFE channel is incorporated in the center channel, hence only five channels are considered in the derivations. Inside an arbitrary but fixed subband, this amounts to the two by five processing:

$$y_m = \sum_{i=1}^{5} h_{m,i} * z_i, \quad m = L, R, \tag{32}$$

where the star denotes convolution in the time direction and the subband signals z_i are those of the original multichannel signal (l_f, l_s, r_f, r_s, c) in that order.

4.3. Combining the HRTF filters based on the spatial parameters

As outlined in Section 3.3.2, an MPEG Surround decoder operates on a downmix signal which is input to a TTT^{-1} module, that recreates a center channel, a right side channel, and a left side channel. These three channels are further processed by several OTT modules yielding the six output channels.

The guiding principle is to require a very high fidelity of the binaural signal for the cases where the MPEG Surround decoding process can approach a waveform match with the original multichannel signal. This holds for example in subbands where only one channel or a selected pair of channels is active. For the more complex cases, rules for combining of the MPEG Surround parameters with the subband filters are applied, which aim at reinstating the correct channel powers of the reference binaural signal (32) in each parameter band. The IPD and IC cues are only indirectly considered.

The spatial parameters for the TTT and OTT modules are used to derive a limited set of HRTFs that can be applied directly on the downmix signal in the QMF filterbank domain. More precisely, the combination of spatial parameters and the subband domain BRIR responses $h_{m,i}$ results in the following two-by-two matrix processing, where (x_1, x_2) is the subband representation of the transmitted downmix:

$$\hat{y}_m = \sum_{i=1}^{2} g_{m,i} * x_i. \tag{33}$$

The filter combination is performed in two steps, one for each layer of the corresponding tree-structured encoder as depicted in Figure 4. In the figure, five of the ten BRIR responses are morphed into two filters, based on the parameters obtained during the encoding process, as depicted in the right panel of Figure 2.

4.3.1. OTT-based front/back morphing

The object of the front/back morphing is to arrive at a modified binaural reference signal defined by the two- by three- processing,

$$\tilde{y}_m = \sum_{p=1}^{3} \tilde{h}_{m,p} * s_p, \tag{34}$$

where the signals s_i are intermediate combined signals (L, R, C) resulting from the TTO encoding process, see Section 3.3.2. The filters $h_{m,1}$ and $h_{m,2}$ from (32) are to be combined into $\tilde{h}_{m,1}$ based on the left-side TTO parameters, and the filters $h_{m,3}$ and $h_{m,4}$ are to be combined into $\tilde{h}_{m,2}$ based on the right-side TTO parameters. The modified binaural reference is intended to serve as a target for the

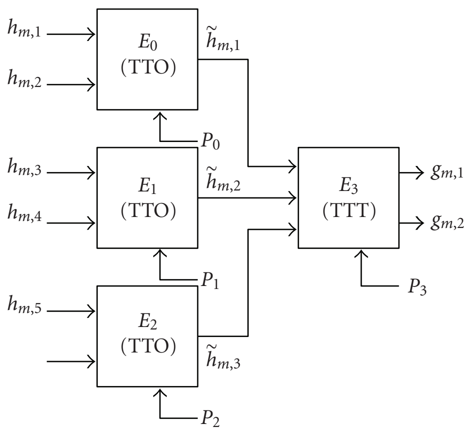

Figure 4: Tree structure overview of the morphing of five of the ten BRIR responses $h_{m,i}$. Note the similarity to the encoding process depicted in the right panel of Figure 2. Also note that the LFE channel is not taken into account in the HRTF filtering, and thus $\widetilde{h}_{m,3} = h_{m,5}$.

subsequent TTT combination. Without loss of generality, we will consider only the left side case and also omit the output channel index. From the CLD parameter of the TTO encoding block, one derives normalized weight parameters w_1 and w_2 such that $w_1^2 + w_2^2 = 1$, and w_1/w_2 equals the CLD in the linear domain. For instance, panning to the front corresponds to $w_1 = 1$ and $w_2 = 0$, while panning to the back results in $w_1 = 0$ and $w_2 = 1$. The morphing consists of forming a complex linear combination

$$\widetilde{h} = t_1 h_1 + t_2 h_2, \tag{35}$$

where the complex coefficients (t_1, t_2) depend on the weight parameters (w_1, w_2) and the filters (h_1, h_2). The contribution $\widetilde{h} * s_1$ should mimic the effect of the conventional approach of convolution followed by summation, that is, $h_1 * z_1 + h_2 * z_2$ according to the guiding principles mentioned above. More precisely, the extreme cases $(w_1, w_2) = (1, 0)$ and $(w_1, w_2) = (0, 1)$ should lead to the correct single source response, and the output energy should be preserved for all cases in between.

Let the complex inner product between subband signals be defined in the usual way,

$$\langle x, y \rangle = \sum_k x(k) y^*(k). \tag{36}$$

The energy of a subband signal is the square of the induced norm $\|x\|^2 = \langle x, x \rangle$. For subband signals x, y that have been filtered by HRTF related subband filters b, d, the following approximation will be assumed

$$\langle b * x, d * y \rangle \approx \langle b, d \rangle \langle x, y \rangle. \tag{37}$$

This approximation is justified by the fact that the time step of the applied time frequency transform is large in comparison to the main delay differences of the HRTF filters such that the energy of the subband domain filters is concentrated in a dominant single tap. (An alternative model situation where (37) holds for general filters is when the subband signals have only lag zero correlation.)

Applying the approximation (37) to align the energy of $\widetilde{h} * s_1$ with that of $h_1 * z_1 + h_2 * z_2$ leads to the requirement

$$\begin{aligned} &\left(|t_1|^2 \|h_1\|^2 + |t_2|^2 \|h_2\|^2 + 2\operatorname{Re}\{ t_1 t_2^* \langle h_1, h_2 \rangle \} \right) \|s_1\|^2 \\ &\quad = \|h_1\|^2 \|z_1\|^2 + \|h_2\|^2 \|z_2\|^2 + 2\operatorname{Re}\{ \langle h_1, h_2 \rangle \langle z_1, z_2 \rangle \}. \end{aligned} \tag{38}$$

From the MPEG Surround encoding process, it can be assumed that the combined signal s_1 carries the total energy of the front and back signals $\|s_1\|^2 = \|z_1\|^2 + \|z_2\|^2$. Hence the energy distribution derived from the weights (w_1, w_2) is given by $\|z_1\|^2 = w_1^2 \|s_1\|^2$ and $\|z_2\|^2 = w_2^2 \|s_1\|^2$. Note that taking into account the last term of the right hand side of (38) would require knowledge of the complex inner product $\langle z_1, z_2 \rangle$, but the phase of this product is not available from the real valued ICC parameter conveyed in MPEG Surround. Instead, this term is neglected, and the modified requirement reads, after removing the common factor $\|s_1\|^2$

$$\begin{aligned} &|t_1|^2 \|h_1\|^2 + |t_2|^2 \|h_2\|^2 + 2\operatorname{Re}\{ t_1 t_2^* \langle h_1, h_2 \rangle \} \\ &\quad = w_1^2 \|h_1\|^2 + w_2^2 \|h_2\|. \end{aligned} \tag{39}$$

A first solution consists of inserting the simple superposition coefficients $(t_1, t_2) = c(w_1, w_2)$ in (39) and subsequently deriving the necessary gain adjustment factor c. The first guiding principle is satisfied in the sense that a perfect output is achieved in the extreme cases $(w_1, w_2) = (1, 0)$ and $(w_1, w_2) = (0, 1)$. However, the resulting gain adjustment varies in an erratic and oscillatory manner as a function of frequency. In practical implementations it is necessary to limit the value of the gain c and a remaining spectral colorization of the signal cannot be avoided. Instead, phase factors are included as follows:

$$(t_1, t_2) = c\left(w_1 e^{-j w_2^2 \phi}, w_2 e^{j w_1^2 \phi} \right), \tag{40}$$

where ϕ is the phase angle of $\langle h_1, h_2 \rangle$, unwrapped over subbands. The role of this phase parameter in the morphing of filters is twofold. First, as it can easily be verified by insertion of (40) in (39), it makes the necessary gain compensation factor c stay between 1 and $1/\sqrt{2}$. Second, it realizes a delay compensation of the two filters prior to superposition which leads to a combined response which models a main delay time corresponding to a source position between the front and the back speakers. Athough this latter property was not explitly stated as a design goal, it leads to a desirable interpolation of binaural contributions.

4.3.2. TTT^{-1} *combination*

The object of the TTT^{-1} combination is to find the filters to be used in final two-by-two processing matrix (33) given the filters of the modified reference (34) defined by a two-by-three processing. The starting point consists of simply inserting the decoded combined channels $\hat{s}_p$ in place of the encoder channels s_p in (34). If the approximation $\hat{s}_p$ to s_p is good, this approach achieves the quality of the modified reference and thus it satisfies our first design principle, but in the general case the signals $\hat{s}_p$ carry linear dependencies due to the spatial upmix process. This fact does not prevent a high playback quality for multichannel loudspeaker listening. However, feeding a collection of binaural filters with such signals can give rise to unwanted spectral coloring. The second design principle of reinstating the correct binaural powers is solved here as in the front/back morphing by introducing gain compensation factors (γ_1, γ_2) for the left and right binaural output. Denoting the entries of the three by two upmix matrix in (27) by $M_{p,i}$, the resulting filters are

$$g_{m,i} = \gamma_m \sum_{p=1}^{3} M_{p,i} \widetilde{h}_{m,p}. \tag{41}$$

In order to derive appropriate values of the compensation gains γ_m, the first step is to model the combined encoding and decoding stages of the TTT, respectively, TTT^{-1} modules by

$$\hat{s}_p = \sum_{q=1}^{3} A_{p,q} s_q, \tag{42}$$

where the three by three matrix with entries $A_{p,q}$ is obtained as the product of the upmix matrix of (27) and the downmix matrix of (26). The resulting decoder output is given by

$$\hat{y}_m = \gamma_m \sum_{p,q=1}^{3} A_{p,q} \widetilde{h}_{m,p} * s_q. \tag{43}$$

The task is then to adjust γ_m such that the binaural output energy is equal to that of the modified reference $\|\hat{y}_m\|^2 = \|\widetilde{y}_m\|^2$. For this, in addition to the rule (37), we assume that the three combined channels s_q are uncorrelated. Indeed, this situation coincides to a large extent with the cases where the TTT^{-1} upmix leads to a significant prediction loss. A comparison of (43) and (34) reveals that the values of the compensation gains are a function of the relative energy distribution of s_p, for $p = 1, 2, 3$. By coincidence, under the assumption of uncorrrelated channels there is a one to one map from the CPC parameters to the energy distribution of the channels. Now it is clear that all the necessary information is present for deriving compensation gains as a function of the transmitted parameters and the HRTF responses in the subband domain. For the final formulas which incorporate further algebraic simplifications due to the CPC parameterization, the reader is referred to [13].

5. APPLICATION TO MPEG SURROUND

5.1. Binaural decoding mode

The parametric and morphed-filter approaches as described in Sections 3 and 4 can be integrated in an MPEG Surround decoder. The mode of operation is referred to as "binaural decoding mode" and its architecture is visualized in

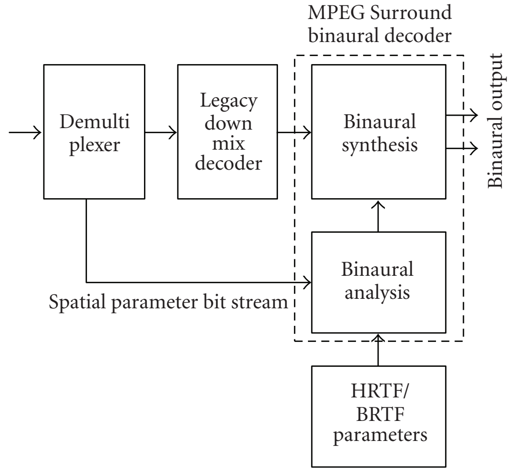

Figure 5: Overview of a binaural decoding mode for MPEG Surround.

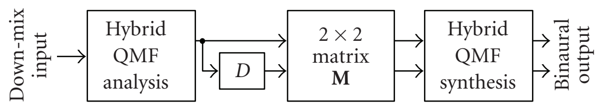

Figure 6: Overview of a binaural synthesis stage based on a mono downmix.

Figure 5. Instead of directly applying the transmitted spatial parameters to the output signals to generate multichannel output, the parameters are used in a binaural analysis stage to compute binaural parameters (using a parametric approach) or morphed filters (using the morphed-filter approach) that would result from the combined spatial decoding and binaural rendering process. The binaural output signals are subsequently generated by the binaural synthesis stage.

The binaural synthesis process is performed in a filterbank domain to enable independent processing of various time-frequency tiles. The synthesis stage for a mono downmix using a parametric approach is outlined in Figure 6. A hybrid QMF filter bank provides 71 down-sampled, nonlinearly spaced subbands that can be grouped in 28 parameter bands that approximate the bandwidth of critical bands. In case of a mono downmix, the hybrid-QMF-domain signal is processed by a decorrelator that consists of lattice all-pass filters to generate a signal that is statistically independent from its input [19, 21]. In case of a stereo downmix, the two downmix signals serve as input to the spatial synthesis stage (without decorrelator). Subsequently, a 2×2 matrix $\mathbf{M}$ is applied for each subband to generate two signals. The final binaural output is obtained by two hybrid QMF synthesis filter banks.

The 2×2 binaural synthesis matrix $\mathbf{M}$ is computed for each received spatial parameter set. In the case of a morphed-filter approach, the synthesis matrix has dimensions $2 \times 2 \times N$, with N the number of "taps" in the time direction. These matrices are defined for specific temporal positions that are signaled in the MPEG Surround bit stream. Typical MPEG Surround parameter update rates are in the order of 30 to

50 milliseconds, and the parameters are typically placed at or near positions where spatial attributes of the audio content show strong deviations over time.

For positions in-between parameter positions, the spatial properties of the incoming signals are not accurately defined and hence an interpolation scheme is required. Preferably, the interpolation scheme has a relatively low computational complexity such that the system could run on battery-powered devices such as mobile audio players. From informal tests it was observed that a piecewise linear approximation of the time-varying synthesis matrix variation (i.e., by linear interpolation of the synthesis matrix) did not have any negative effects on the resulting quality compared to more advanced interpolation schemes.

5.2. *Evaluation (parametric approach)*

5.2.1. *Procedure*

A listening test was pursued to evaluate the subjective quality of the proposed parametric binaural synthesis method. In this test, the quality of the MPEG Surround (MPS) binaural decoding mode ("MPS binaural") is compared to a reference condition. This reference condition comprised convolution of an original multichannel audio excerpt with HRIRs and subsequent downmix to stereo. As a control condition, the combination of MPEG Surround multichannel decoding followed by conventional HRIR convolution was employed (denoted "MPS + HRIR"). For all configurations in this test, anechoic KEMAR HRIRs [43] were used with a length of 128 samples at a sampling frequency of 44.1 kHz.

For both the binaural decoding mode and the control condition, the same MPEG Surround bit stream was employed. This bit stream was generated using a state-of-the-art MPEG Surround encoder using a mono downmix configuration. This mono downmix was subsequently encoded using a high-efficiency AAC (HE-AAC) encoder [44] at 44 kbps. The spatial parameters generated by the MPEG Surround decoder occupied approximately 4 kbps. This rather low bit rate of 48 kbps total was selected because it is foreseen that the binaural decoding mode is especially suitable for mobile applications that are under severe transmission bandwidth and complexity constraints.

Twelve listeners participated in this experiment. All listeners had significant experience in evaluating audio codecs and were specifically instructed to evaluate the overall quality, consisting of the spatial audio quality as well as any other noticeable artifacts. In a double-blind MUSHRA test [45], the listeners had to rate the perceived quality of several processed excerpts against the reference condition (i.e., uncoded items processed with HRIRs) excerpts on a 100-point scale with 5 labels, denoted as "bad," "poor," "fair," "good," and "excellent." A hidden reference and the low-pass filtered anchor (reference with a bandwidth limitation of 3.5 kHz) were also included in the test. The subjects could listen to each excerpt as often as they liked and could switch in real time between all versions of each excerpt. The experiment was controlled from a PC with an RME Digi 96/24 sound card using ADAT digital out. Digital-to-analog conversion was provided by an RME ADI-8 DS 8-channel D-to-A converter. Beyerdynamic DT990 headphones were used throughout the test. Subjects were seated in a sound-insulated listening room.

A total of 11 critical multichannel excerpts were used as listed in Table 1. The excerpts are the same as used in the MPEG Call for Proposals (CfP) on Spatial Audio Coding [46], and range from pathological signals (designed to be critical for the technology at hand) to movie sound and multichannel music productions. All input and output excerpts were sampled at 44.1 kHz.

Table 1: Test excerpts

Excerpt	Name	Category
1	BBC applause	Pathological/ambience
2	ARL applause	Pathological/ambience
3	Chostakovitch	Music
4	Fountain music	Pathological/ambience
5	Glock	Pathological
6	Indie2	Movie sound
7	Jackson1	Music
8	Pops	Music
9	Poulenc	Music
10	Rock concert	Music
11	Stomp	Music (with LFE)

5.2.2. *Results*

The results of the listening test are shown in Figure 7. The various excerpts are given along the abscissa, while the ordinate corresponds to the average MUSHRA score across listeners. Different symbols refer to different configurations. The error bars denote the 95% confidence intervals of the means.

The hidden reference (square symbols) has the highest scores. The results for the binaural decoding mode are denoted by the diamonds; the control condition using convolution is represented by the downward triangles. Although the scores for these methods vary between 45 and 85, the binaural decoding approach has scores that are higher than the conventional method for all excerpts. Finally, the low-pass anchor has the lowest scores of around 20.

The average scores for each method across subjects and excerpts are shown in Figure 8. The difference between the binaural decoding mode and the control condition amounts to 12 points in favor of the binaural decoder.

If the computational complexities of the binaural decoder and the conventional systems are compared, also interesting differences are observed. The number of operations (expressed in multiply-accumulates per second) amounts to 11.1 million for the binaural decoder and 47 million for the MPEG Surround multichannel decoder followed by convolution using fast Fourier transforms.

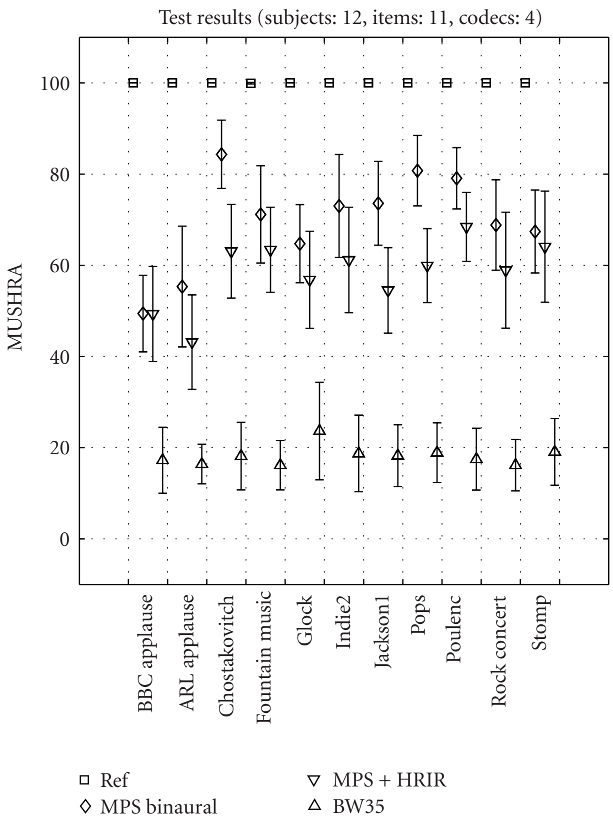

FIGURE 7: Subjective test results averaged across subjects for the parametric approach. Error bars denote the 95% confidence intervals of the means.

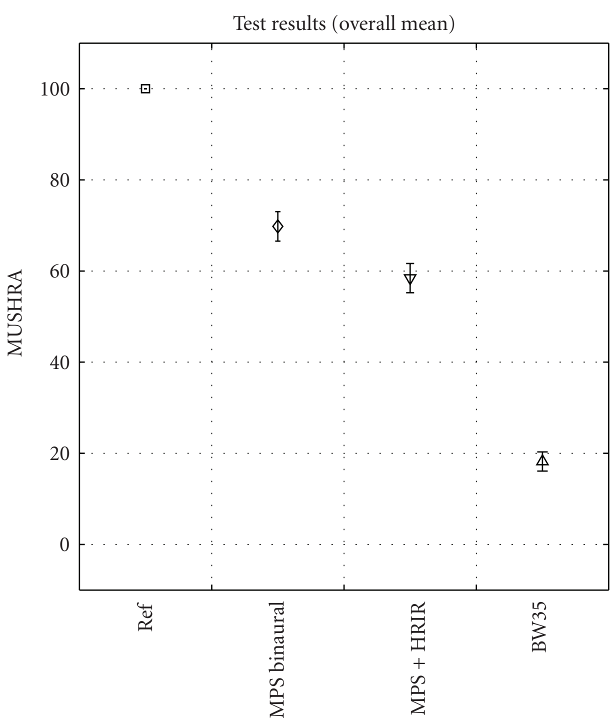

FIGURE 8: Overall mean scores (across subjects and excerpts) for the parametric approach.

5.2.3. Discussion

The results of the perceptual evaluation indicate that both of the binaural rendering methods (the parametric binaural decoding mode and the conventional HRIR convolution method) are distinguishable from the reference. This is most probably due to the low bit-rate (48 kbps total) that was employed to represent the multichannel signal in MPEG Surround format. For loudspeaker playback, the perceived quality of MPEG Surround operating at 48 kbps has been shown to amount 65 in other tests [15, 47]. In that respect, the quality for the test and control conditions isin line with earlier reports.

The parametric representation of MPEG Surround aims at perceptual reconstruction of multichannel audio. As such, at the bit rate that was under test, MPEG Surround does not deliver full waveform reconstruction of the multichannel output signals. Such waveform reconstruction requires the use of "residual coding" as supported by MPEG Surround. However, residual coding results in a significant increase in the bit rate which is undesirable or even unavailable in mobile applications. Given the low scores for MPEG Surround decoding followed by HRIR convolution, the multichannel signals resulting from the parametric representation seem unsuitable for further post processing using HRIRs. This is a property that is often observed for lossy audio coders. The binaural decoding mode, however, which does not rely on processing of decoded signals, outperforms the convolution-based method, both in terms of perceived quality and computational complexity. This clearly indicates the advantages of parameter-domain processing compared to the signal-domain approach.

5.3. Evaluation (morphed-filter approach)

5.3.1. Procedure

A second listening test was employed to assess the quality of the QMF-domain morphed-filter approach. The reference, control, test and anchor conditions were generated in the same way as described in Section 5.2.1, however with the following modifications to reflect a different application scenario, that is, that of an online music store. In this application scenario, multichannel audio is encoded using the MPEG Surround format employing a stereo downmix to ensure stereo backward compatibility. The down-mix was encoded using AAC at a bit of 160 kbps, a common bit rate for stereo content in online music stores, while the MPEG Surround parameter bit rate was set to 12 kbps. In the current test, echoic BRIRs were employed that were also used in the MPEG selection tests [48]. The test procedure and excerpts are identical to those employed in the previous test. A total of 12 subjects participated in this test.

5.3.2. Results

The results of the listening test for individual excerpts are shown in Figure 9. The various excerpts are given along the abscissa, while the ordinate corresponds to the average MUSHRA score across listeners. Different symbols refer to

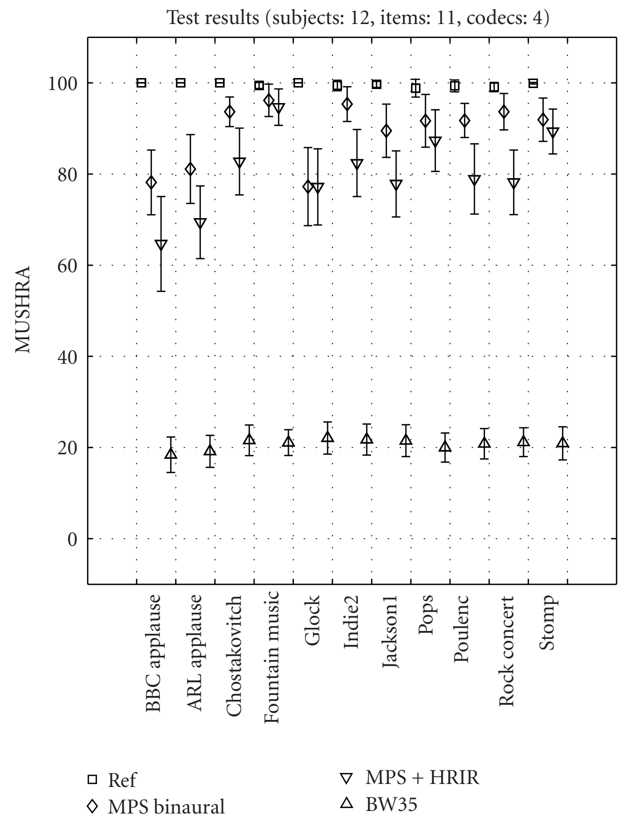

Figure 9: Subjective test results averaged across subjects for the morphed-filter approach. Error bars denote the 95% confidence intervals of the means.

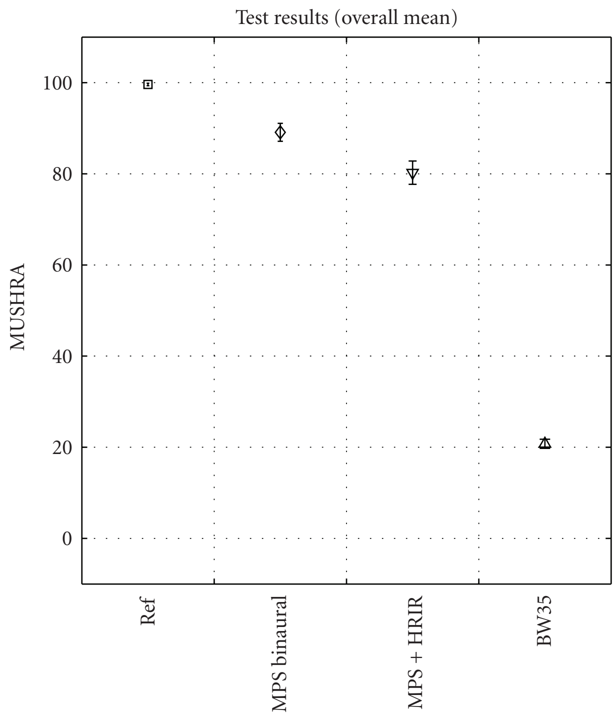

Figure 10: Overall mean scores (across excerpts and subjects) for the morphed-filter approach. Error bars denote the 95% confidence intervals of the means.

different configurations. The error bars denote the 95% confidence intervals of the means.

The trend observed in Figure 9 is very similar to the one observed for the parametric approach in Figure 7. The hidden reference (squares) has scores around 100. The MPEG Surround binaural decoding mode (diamonds) has scores between 77 and 95 and has in all cases a higher mean across subjects than the control condition (downward triangles).

The mean across subjects and excerpts for each configuration is shown in Figure 10. On average, the MPEG Surround binaural decoding mode scores about 90, which is roughly 10 MUSHRA points higher than the control condition.

The computational complexity of the morphed-filter approach in this case amounts to 41 million operations, compared to 47 million for the control condition (MPEG Surround multichannel output followed by BRIR convolution in the FFT domain).

5.3.3. *Discussion*

In analogy to the test results for the parametric approach, the QMF-domain filtering method achieves a higher quality than the control condition (i.e., multichannel decoding and subsequent HRTF or BRIR convolution). Hence, for both methods, it is beneficial to combine the spatial decoding and binaural rendering processes to achieve maximum perceptual quality.

The overall scores for the QMF-domain filtering approach are higher than those for the parametric method. This difference can be attributed to several factors.

(i) The employed binaural rendering method. The parametric approach employs a lossy HRTF representation, while the QMF-domain filtering method results in a more accurate representation of the original impulse response.

(ii) The spatial parameter bit rate. In the second test, the bit rate used for spatial parameters is higher than the bit rate employed in the first test, which results in a more accurate representation of the multichannel audio content.

(iii) The downmix configuration. In the second test, a stereo downmix was employed, while in the first test, one single audio channel was used as downmix signal. MPEG Surround will in most cases achieve a higher quality for stereo downmixes than for mono downmixes.

(iv) The bit rate of the core coder. In the first test, 44 kbps was used to encode the mono signal, while in the second test, 160 kbps was available for the stereo signal. Hence the perceptual quality of the transmitted downmix is higher for the second test than for the first test.

Although it is difficult to assess the effect of the individual factors on the resulting quality based on the current test results, it is expected that the downmix coder (and the associated channel configuration) has quite a large effect on the overall quality, a trend that can also be observed in loudspeaker reproduction test results published in [17, 18].

6. CONCLUSIONS

Two novel methods for binaural rendering based on parametric representations were outlined. In contrast to conventional, convolution-based methods, HRTFs or BRTFs are transformed to the parameter domain or filterbank domain and combined with parameters that describe the statistical properties of the various signals, which are radiated by virtual sources. From this combination, a 2×2 matrix operation (including the option to have taps in the time direction) is derived that converts a mono (using an additional decorrelator circuit) or stereo downmix to a binaurally rendered signal without the need of individual source signals as intermediate step.

The proposed method can be easily integrated with parametric multichannel audio coders (MPEG Surround) that rely on interchannel cues such as level differences and interchannel correlations. Results of a listening test revealed that the proposed method outperforms conventional, convolution-based methods in terms of perceived quality and computational complexity. These properties, combined with the unsurpassed compression efficiency of MPEG Surround, make the proposed method very suitable for mobile applications.

ACKNOWLEDGMENT

The authors would like to thank the anonymous reviewers and the associate editor for their thorough reading and valuable comments and suggestions for improving the manuscript.

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EURASIP Journal on Wireless Communications and Networking

Special Issue on
Fairness in Radio Resource Management for Wireless Networks

Call for Papers

Radio resource management (RRM) techniques (such as admission control, scheduling, sub-carrier allocation, channel assignment, power allocation, and rate control) are essential for maximizing the resource utilization and providing quality of service (QoS) in wireless networks.

In many cases, the performance metrics (e.g., overall throughput) can be optimized if opportunistic algorithms are employed. However, opportunistic RRM techniques always favor advantaged users who have good channel conditions and/or low interference levels. The problem becomes even worse when the wireless terminals have low mobility since the channel conditions become slowly varying (or even static), which might lead to long-term unfairness. The problem of fair resource allocation is more challenging in multihop wireless networks (e.g., mesh and multihop cellular networks).

The performance fairness can be introduced as one of the QoS requirements (e.g., as a condition on the minimum throughput per user). Fair RRM schemes might penalize advantaged users; hence, there should be a tradeoff between the overall system performance and the fairness requirements.

We are soliciting high-quality unpublished research papers addressing the problem of fairness of RRM techniques in wireless communication systems. Topics include (but are not limited to):

- Fairness of scheduling schemes in wireless networks
- Tradeoff between maximizing the overall throughput and achieving throughput fairness
- RRM fairness: problem definition and solution techniques
- Fairness performance in emerging wireless systems (WiMAX, ad hoc networks, mesh networks, etc.)
- Cross-layer RRM design with fairness
- Short-term and long-term fairness requirements
- Adaptive RRM to support fairness
- Fairness in cooperative wireless communications
- Issues and approaches for achieving fairness in multihop wireless networks
- RRM framework and QoS architecture
- Complexity and scalability issues
- Experimental and implementation results and issues
- Fairness in multiple-antenna transmission/reception systems
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Manuscript Due	July 1, 2008
First Round of Reviews	October 1, 2008
Publication Date	January 1, 2009

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EURASIP Journal on Image and Video Processing

Special Issue on Patches in Vision

Call for Papers

The smallest primitive employed for describing an image is the pixel. However, analyzing an image as an ensemble of patches (i.e., spatially adjacent pixels/descriptors which are treated collectively as a single primitive), rather than individual pixels/descriptors, has some inherent advantages (i.e., computation, generalization, context, etc.) for numerous image and video content extraction applications (e.g., matching, correspondence, tracking, rendering, etc.). Common descriptors in literature, other than pixels, have been contours, shape, flow, and so forth.

Recently, many inroads have been made into novel tasks in image and video content extraction through the employment of patch-based representations with machine learning and pattern recognition techniques. Some of these novel areas include (but are not limited to):

- Object recognition/detection/tracking
- Event recognition/detection
- Structure from motion/multiview

In this special issue, we are soliciting papers from the image/video processing, computer vision, and pattern recognition communities that expand and explore the boundaries of patch representations in image and video content extraction.

Relevant topics to the issue include (but are not limited to):

- Novel methods for identifying (e.g., SIFT, DoGs, Harris detector) and employing salient patches
- Techniques that explore criteria for deciding the size and shape of a patch based on image content and the application
- Approaches that explore the employment of multiple and/or heterogeneous patch sizes and shapes during the analysis of an image
- Applications that explore how important relative patch position is, and whether there are advantages in allowing those patches to move freely or in a constrained fashion
- Novel methods that explore and extend the concept of patches to video (e.g. space-time patches/volumes)
- Approaches that draw upon previous work in structural pattern recognition in order to improve current patch-based algorithms
- Novel applications that extend the concept of patch-based analysis to other, hitherto, nonconventional areas of image and video processing, computer vision, and pattern recognition
- Novel techniques for estimating dependencies between patches in the same image (e.g., 3D rotations) to improve matching/correspondence algorithmic performance

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Manuscript Due	May 1, 2008
First Round of Reviews	August 1, 2008
Publication Date	November 1, 2008

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Special Issue on Personalization of Mobile Multimedia Broadcasting

Call for Papers

In recent years, the widespread adoption of multimedia computing, the deployment of mobile and broadband networks, and the growing availability of cheap yet powerful mobile have converged to gradually increase the range and complexity of mobile multimedia content delivery services for devices such as PDAs and cell phones. Basic multimedia applications are already available for current generation devices, and more complex broadcasting services are under development or expected to be launched soon, among which mobile and interactive television (ITV). Among the many challenging issues opened by these developments is the problem of personalization of such services: adaptation of the content to the technical environment of the users (device and network type) and to their individual preferences, providing personalized assistance for selecting and locating interesting programes among an overwhelming number of proposed services.

This special issue is intended to foster state-of-the-art research contributions to all research areas either directly applying or contributing to solving the issues related to digital multimedia broadcasting personalization. Topics of interest include (but are not limited to):

- Mobile TV
- Mobile multimedia broadcasting personalization
- Interactive broadcasting services/interactive television
- Personalization and multimedia home platform (MHP)
- Multimedia content adaptation for personalization
- User behavior and usage modelling
- Standards for modelling and processing (MPEG-21, CC/PP, etc.)
- Personalization issues in DVB-H, DMB, MediaFLO, CMMB, MBMS, and other systems
- Mobile web initiative
- Personalized multimedia and location-based services
- Security and digital rights management
- Applications for personalized mobile multimedia broadcasting with cost-effective implementation

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Manuscript Due	March 1, 2008
First Round of Reviews	June 1, 2008
Publication Date	September 1, 2008

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International Journal of Digital Multimedia Broadcasting

Special Issue on
Iterative Decoding and Cross-Layering Techniques for Multimedia Broadcasting and Communications

Call for Papers

The explosive growth of multimedia applications over the Internet and the ever-increasing users' demands over commercial terrestrial digital multimedia broadcasting all over the world call for efficient physical and cross-layer techniques able to mitigate the potential problems limiting broadband services over wireless networks. In this scenario, mobile multimedia is expected to be one of the key services of future wireless mobile networks. Meanwhile, recent advances in digital communications have paved the way to a variety of standards aimed at providing multimedia services over terrestrial broadband networks. To cite but a few, DVB-H, T-DVB, T-DMB, wireless LANs, and wireless MANs are some of the most recent standards enabling such technology.

Iterative decoding techniques for both source, channel, and joint source-channel coding and decoding and cross-layering techniques have proven to be very effective for providing a viable means of achieving capacity-approaching performance at very reduced computational burden.

The main aim of this special issue is to highlight state-of-the-art techniques on the most recent research advances enabling digital multimedia services over broadband wireless networks, focused on physical and cross-layering solutions. Novel contributions, previously unpublished, that are not being submitted to any other journal, are sought.

Topics of interests include, but are not limited to, the following subject categories:

- Iterative decoding techniques for concatenated channel codes (turbo codes and serially concatenated codes)
- Iterative decoding techniques for joint source-channel decoding
- Novel capacity-approaching channel codes: coding strategies and efficient decoding algorithms
- Cross-layer modelling/analysis and optimization techniques
- Standardization activities on digital multimedia broadcasting protocols
- Space-time coding and decoding
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Manuscript Due	April 1, 2008
First Round of Reviews	July 1, 2008
Publication Date	November 1, 2008

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EURASIP Journal on Embedded Systems

Special Issue on Formal Techniques for Embedded Systems Design and Validation

Call for Papers

Computing platforms that are embedded within larger systems they control, called embedded systems, are inherently very complex as they are responsible for controlling and regulating multiple system functionalities. Often embedded systems are also safety-critical requiring high degree of reliability and fault tolerance. Examples include distributed microprocessors controlling the modern cars or aircrafts and airport baggage handling system that track and trace unsafe baggage. To address this growing need for safety and reliability, formal techniques are increasingly being adapted to suit embedded platforms. There has been widespread use of synchronous languages such as Esterel for the design of automotive and flight control software that requires stronger guarantees. Languages like Esterel not only provide nice features for high-level specification but also enable model checking-based verification due to their formal semantics. Other semi-formal notations are also being proposed as standards to specify industrial embedded systems using, for example, the newly developed IEC61499 standard for process control. This standard primarily focuses on component-oriented description of embedded control systems. The goal of this special issue is to bring together a set of high-quality research articles looking at different applications of formal or semiformal techniques in specification, verification, and synthesis of embedded systems.

Topics of interest are (but not limited to):

- Verification of system-level languages
- Verification of embedded processors
- Models of computation and verification
- Models of computation for heterogeneous embedded systems
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- Open system verification techniques such as module checking and applications of module checking
- Formal techniques for protocol matching and interface process generation
- Applications of DES control theory in open system verification
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Manuscript Due	August 1, 2008
First Round of Reviews	November 1, 2008
Publication Date	February 1, 2009

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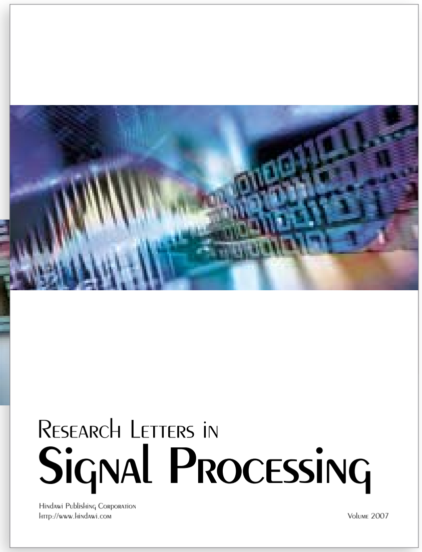

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ISSN: 1687-6911; e-ISSN: 1687-692X; doi:10.1155/RLSP

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