Computer simulation of Binaural, Stereo-Dipole, B-format and Ambiophonics impulse responses

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The paper addresses the problem of deriving synthetic impulse responses for auralization and spatialization systems, starting from the results of room-acoustics simulation programs. As it is well known, usually these numerical models are based on geometrical acoustics, and produce as output octave-band energetic impulse responses. These impulse responses have to be converted in wide-band waveform for being used as FIR filters, which can be convolved with dry signals, obtaining synthetic surround sound in various formats. The authors already developed in the past a suitable wide-band conversion algorithm for mono (omnidirectional receiver) impulse response and for binaural impulse responses.
In this paper, all the various possible “surround” formats are taken into account:

- Binaural, for headphone listening
- Stereo Dipole, for transaural listening on one or two pairs of closely-spaced loudspeakers
- B-format, for reproduction through a three-dimensional Ambisonics decoder and loudspeaker array (1st and 2nd order).
- Ambiophonics, which couples a frontal Stereo-Dipole for the direct sound, and an Ambisonics-like surround array for the subsequent reflections and reverberant tail

The paper explains the mathematics of the conversion from the energetic impulse response to the multichannel waveforms in each of the above formats.

ROOM ACOUSTICS SIMULATION

This paper describes the post processing of the results of a well known room acoustics simulation program [1].

As it is common to many others similar programs based on geometrical acoustics, the computation produces in each receiver an energetic impulse response, computed in several octave bands. In this case, in each receiver the impulse response has a constant time resolution (typically 1 to 10 ms) and 10 octave bands (31.5 to 16000 Hz). The value recorded in each cell of this data structure is the sum of the energy density carried by all the wavefronts arriving to the receiver during the corresponding time slot: the hypothesis is thus that these wavefronts do not interfere, as each of them is carrying a signal which is uncorrelated with the others. It is obvious that this hypothesis has some intrinsic inconsistency for low-order reflections and for low frequencies, whilst instead is reasonable for higher frequencies and in the late part of the reverberant tail.

Furthermore, for the lower order reflections, also the direction of provenience, the amplitude and the exact arrival time of each wavefront are saved in a separate file: this makes it possible to plot the paths of each energy arrival, and to derive realistic pressure impulse responses, in a variety of multi-channel formats corresponding to different types of stereophonic microphones.

In the following it is briefly described how to translate the energetic results of the simulation to the virtual signal, which would be recorded by different types of microphones placed at the receiver positions.

VIRTUAL MICROPHONES

The following types of stereophonic microphones can be virtually placed at the receiver position:

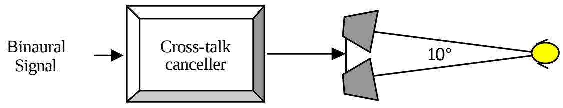
- omnidirectional pressure microphone: for comparison with measured impulse responses
- binaural 2-channels microphone: for auralization by means of headphones.
- 1st order spherical harmonics of pressure field (B-format Soundfield microphone, 4 channels): for auralization on an Ambisonics 1st order decoder.
- 2nd order spherical harmonics of pressure field (Furse-Malham microphone, 9 channels): for auralization on an Ambisonics 2nd order decoder.
- Single Stereo-Dipole “transaural” detector (binaural processed with cross-talk canceling filters): for auralization on a pair of closely-spaced loudspeakers
- Dual Stereo-Dipole “transaural” detector (4-channels, separate frontal and rear soundfields): for auralization on two pairs of closely-spaced loudspeakers
- Ambiophone (pinnaless dummy head for the first two channels, plus a 4-channels 1st order B-format for the surround): for auralization on an Ambiophonics playback system

COMPUTATION OF THE IMPULSE RESPONSES

For each type of microphone, the conversion from a 10-bands energy impulse response to a multichannel, wideband pressure impulse response is done employing the same approach: first the early reflections are processed, taking into account their known arrival direction and exact timing. For each energy arrival, a Dirac's delta function is generated at the exact arrival time, then it is convolved with the impulse response of an octave-band equalizer which imposes the proper 10-octaves spectrum, and finally it is convolved with the multichannel impulse response of the selected microphone, chosen depending on the direction of arrival.
Then the subsequent reverberant tail is added, based on the whole energetic impulse response data (which do not contain any directional information), assuming valid the hypothesis of a completely isotropic diffuse field, with arrival directions randomly chosen from all the angles. This is done by first generating an independent sample of white noise for each channel of the virtual microphone, and then applying to it a time-dependent octave-band equalization, given by the energy-time-frequency information stored as a result of the room acoustical computation: in more detail, the white noise is first octave-band filtered, then it is amplitude-modulated with the square root of the temporal hystogram of received energy in that frequency band; finally the processed signals of all the octaves are summed back together.
Regarding the impulse response of each directive microphone, it must be noted that for the omnidirectional and for the first and second order Ambisonics these are simply delta functions of proper amplitude and sign, obtained from the director cosines of the incoming ray (r_x, r_y, r_z). The following table contains the formulas which give these gains:

Channel	formula
W	0.707107
X	r_x
Y	r_y
Z	r_z
R	$1.5 \cdot r_z^2 - 0.5$
S	$2 \cdot r_z \cdot r_x$
T	$2 \cdot r_y \cdot r_z$
U	$r_x^2 - r_y^2$
V	$2 \cdot r_x \cdot r_y$

For the binaural and Stereo Dipole method, instead, the impulse responses employed come form the measurements made at MIT on the Kemar dummy head [3]. In the case of loudspeaker reproduction (Stereo Dipole, dual Stereo Dipole) they must be also convolved with the set of proper cross-talk canceling filters, which are derived by inversion with the Kirkeby/Farina approach the pair of binaural impulse responses correspondent to the position of the loudspeakers, as explained in detail in [4].



Reproduction over the Stereo-Dipole

For Ambiophonics, a similar data-base of theoretical impulse responses was computed analytically, considering an ideal rigid sphere of 180 mm diameter with pressure microphones exactly at the opposite sides. Only the direct sound and the first reflections coming from the frontal hemispace are sent to the sphere microphone, which drives the frontal Stereo Dipole (again by means of cross-talk canceling filters, derived by the impulse responses of the theoretical sphere). The subsequent lateral and rear reflections and the reverberant tail are instead sent to the B-format microphone, which drives the surround loudspeakers.
The working principle and some implementation details related to the Ambiophonics reproduction system are detailed in [5].

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