Equalization and fidelity of a sound environment simulator

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Introduction

The aim of a sound environment simulator is to reproduce all the acoustical elements of a real sound environment. The simulator is based on a mobile recording system and a reproduction system, which is generally installed in a controll room (cf. Figure 1). For example, binaural microphones may be used for the recording, and in a stereo dipole format for the sound reproduction.

Currently, there is no perfect method or technique to record and reproduce the sound field. For example, microphones can not perfectly capture the sound field coming from all directions and the loudspeakers' spectrum response is not perfectly flat. Hence, special treatments must be applied to the signals in order to achieve the best compromise.

This paper is divided in 3 sections. The first section deals with the practical approach in order to tune the fidelity of the sound environment simulator. The second part briefly reviews the equalization process. The final section discuss a procedure to validate the fidelity of the simulator.

Fidelity

The fidelity of a sound environment simulator may be seen through the ability to reproduce the 3 following acoustical clues:

- spatialisation / localisation,
- loudness,
- spectral content

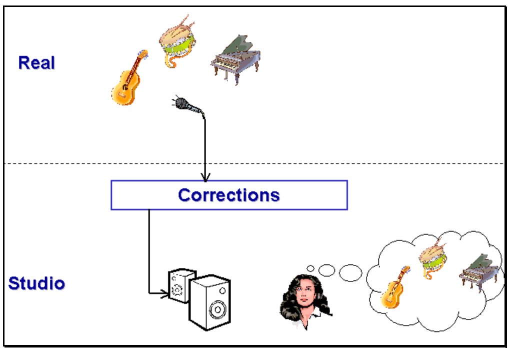

Figure 1: Sound Environment simulator schematic

In order to achieve the best fidelity, corrections need to be applied between the recorded signal and the loudspeakers (cf. Figure 1):

- The reproduction of the spatialisation and the localisation of sounds is linked to the method used by the simulator, some esamples of which are Binaural, Transaural, Stereo Dipole and Ambisonics. Some methods perform well with the localisation of sources. Others are more effective at immersing the listener in the sound field
- The loudness of reproduced sounds can be corrected by a gain factor in the simulator
- The reproduction of the spectral content of sounds can be corrected by filters which will decrease spectral coloration introduced by microphones and the loudspeaker. These filters could also reduce the reverberation effect of the listening room.

Equalization

In order to obtain the filters that need to be applied, two methods are available:

- The coupled equalization technique considers the microphones and loudspeakers as a whole system,
- The decoupled equalization technique considers the microphones and loudspeakers as separate systems.

Coupled equalization

The coupled equalization involves the measurement and inversion of the transfer functions H between the loudspeakers HP and the microphones M. For simplification, only loudspeakers and microphones are taken into account in eq. (1).

$$H(z) = HP(z) * M(z) \tag{1}$$

These measurements are conducted in the listening room with the microphones placed at the listening position (cf. placement on Figure 3), with for example pseudo-random sequences or sweep. When applied, the equalization H^{-1} assumes that the spectral coloration of the loudspeakers and microphones are indeed removed. These filters could also reduce the reverberation effect of the listening room.

This method has the advantage of simplicity, but it is valid only for the reproducing of sources eminating from the direction of the loudspeakers towards the listening position (ie. for transaural systems, fidelity is achieved for sources recorded in front of the listener. This is linked to the fact that the transfert function M has been measured with the loudspeakers placed in a frontal position). Moreover, the simulator is dependant on the recording system (ie, for binaural recording, the equalization is lost if you change the dummy head or the binaural microphones).

Decoupled equalization

The decoupled equalization involves distinct corrections to be applied: one for the recording system, and one for the diffusion system. Independence is achieved by the measurement of the transfer functions of each part with a reference sound field. The reference field normally comprising a free field or a diffuse field.

The free field involves measurements in an anechoic room, and the diffuse field needs a reverberant room. Normally, a diffuse field is preferable since it does not focus the fidelity on a particular source direction with regard to the free field equalization.

For a more in-depth explanation, see [1] and [2] for binaural/transaural cases.

This technique has the advantage of decoupling the recording system of the diffusion system. Knowledge of each system is not required; the only relevant information is the type of reference field used.

The drawbacks to this method is the relative difficulty in reproducing the reference field: an anechoic or reverberant room may not be easily accessible. Acceptable results have been experienced with rooms without any particular acoustical treatment for measuring the diffuse field response: the computation can be made on the impulse responses after the first reflexions have been eliminated, where the sound field can be considered as diffuse [1].

Validation of the equalization and the fidelity

In order to test and validate the recording and reproduction chain, comparative listening tests between the real sound field and the reproduced sound field in the studio are preferable. However, it is not generally possible to compare both, in case for example of vehicles sound studies.

Thus, the only way to achieve comparative listening tests is to create a "real" sound field in the studio, with a recorded signal reproduced with the help of loudspeakers (the simulator's for example), (cf. Figure 2):

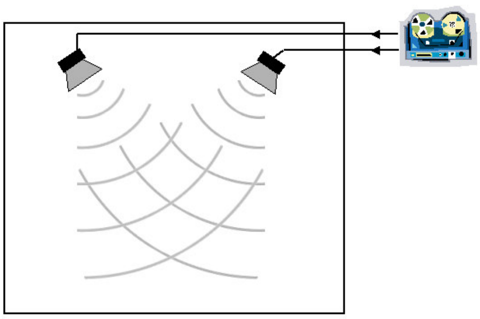

Figure 2: "Real" sound field created with any recorded signal diffused with the loudspeakers in the listening room (transaural example shown)

The general idea is to create a fidelity loop: the recording system is placed in the listening position and the whole system is then used (recording, corrections, reproducing). The reproduced sound field forms the "simulated" sound field, (cf. Figure 3).

With this technique, "real" and "simulated" sound fields can be reproduced and compared. The effect of the recording and reproduction chain can thus be heard.

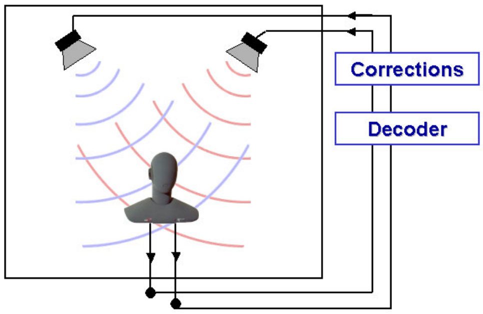

Figure 3: "simulated" sound field created from the "real" sound field (transaural example shown)

Moreover the gain correction can then be adjusted in order to achieve the same loudness of the "simulated" sound field compared to the real one.

It is also possible to subjectively tune the system: a real-time equalizer with ERB or Bark bands can be applied just before the loudspeaker outputs. A listener can thus adjust the system to achieve the best fidelity. If the system is correctly tuned, the difference between both sound fields should be minimal.

Conclusion

This paper presented a short review of the necessary steps to equalize and optimize the fidelity of a sound environment simulator. Spatialisation fidelity depends mainly on the technique used, while spectral and level fidelity depend on the equalization process. A subjective validation process has been presented. The aim of this fidelity loop is to validate the performance of the system by comparing in real-time, by a listener, the difference between the real and its the simulated one.

References

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[2] "Prospects for Transaural Recording", D.H. Cooper et J.L. Bauck. J. Audio Eng. Soc. Vol 37 No 1, 1989