PSYCHO-ACOUSTIC MASKING PHENOMENA -USE IN DIGITAL STORAGE AND TRANSMISSION OF SOUND

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Abstract

This article deals with psycho-acoustic masking effects and outlines their exploitation in the digital storage and transmission of sound.

Keywords:

Psychoacoustic masking, digital audio broadcasting, digital audio recording, source coding, DAB, MASCAM, MUSICAM

1. Introduction

Psycho-acoustic masking phenomena are the effects taking place in the human auditory system. These effects play a very important role in hearing and are differentiated into simultaneous and nonsimultaneous masking phenomena. Due to the simultaneous masking low level sounds become inaudible in the presence of the higher levels of sound at similar frequencies. The listener would not perceive the audio signals below the masking threshold (see Fig.1), where the masker is the 1 kHz pure tone in this case. It is not necessary for these inaudible sounds to be stored or transmitted. The masking threshold is determined as the sound pressure level of a test sound necessary to be barely audible in the presence of the masker.

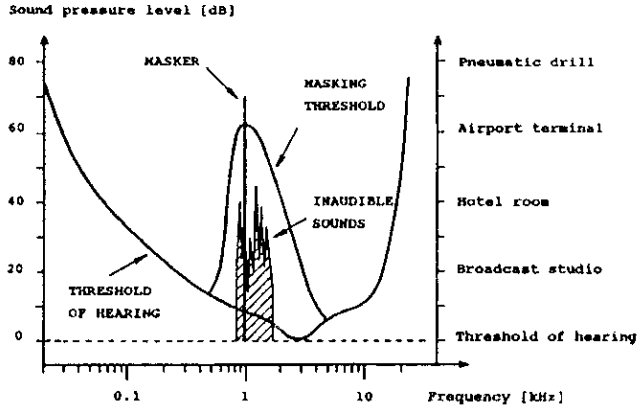

Fig.1
The simultaneous masking phenomenon

While the simultaneous masking phenomenon is defined in the presence of the masker, the nonsimultaneous masking phenomena influence in the periods just before and right after his occurence. The former nonsimultaneous masking phenomenon is called pre-masking, the latter is called post-masking. The post-masking results from the gradual release of the simultaneous masking effect. The masking is not immediately stopped with switching off the masker but after this switching off the masking lasts some time depending on the duration and sound pressure level of the masker. The dependence of the post-masking on the masker duration is shown in Fig.2. The delay time at which the test tone burst is presented after the end of the masker is plotted as the abscissa. The sound pressure level of this burst necessary for the audibility is the ordinate. The parameter is the duration of the masker. The sound pressure level of the masker is 60 dB. The masker is uniform masking noise in this case. The duration of the 2 kHz test tone burst is 5 ms. It is obvious that the post-masking period becomes longer with the masker

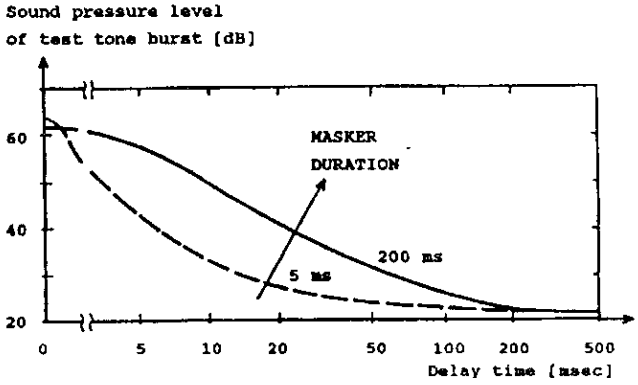

Fig.2
The postmasking phenomenon

duration. The duration of the pre-masking is negligible in comparison with the post-masking period, and therefore the pre-masking is not so important as the post-masking.

Most of the sounds cause the shift of the threshold of hearing in the human auditory system, however, this shift is different for different sounds, and, in addition, varies with time. The characteristics of the human auditory system result in the fact that the mutual masking of sound occurs in the limited sub-bands of the acoustic band (see Fig.1). For this purpose, the acoustic band is split into several sub-bands by means of the digital filtering. The digital processing of the samples of sound is presented in Fig.3. After filtering, the maximum value is determined in each sub-band each 8 ms. On the basis of these values the

masking threshold is calculated and depending on this threshold the data reduction is realized. As it has already been told, there are the inaudible sounds below the masking threshold which need not be stored and transmitted. It is, however, only one part of the data reduction. Additional savings can be achieved by decreasing the level resolution of the quantized samples.

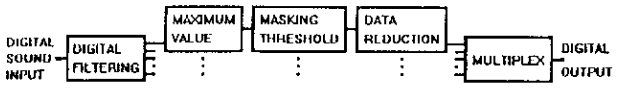

Fig.3
The digital sound processing before transmission or recording

This must be done in order that the quantizing noise, which grows with this decreasing of the resolution, should not exceed the masking threshold (see Fig.4). We must allocate this resolution dynamically depending on the instantaneous signal in each corresponding sub-band. Finally, we must multiplex the data of all the sub-bands.

After the transmission or recording this multiplex signal must be demultiplexed and decoded by the inverse digital processing, as shown in Fig.5.

A typical net bit-rate for the digital recording on the compact disc is 700 kbit/s for each channel. Here described

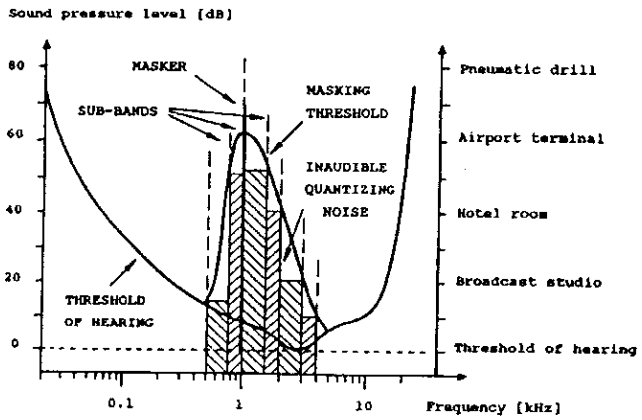

Fig.4
The quantizing noise

way of the digital sound processing makes the reduction to 100 kbit/s possible without any perceptible impairment. This savings are important not only for the digital recording of sound but, most of all, for the digital audio broadcasting. It is possible to extend recording time seven times and in the same proportion we can extend the number of broadcasting channels within an allocated band.

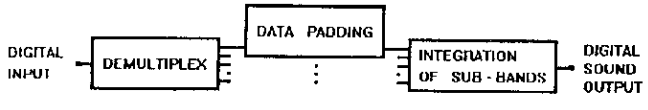

Fig.5
The inverse digital sound processing after transmission or recording

2. References

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