DIGITAL CODING SOUND SIGNALS AND COMPRESSION MASK

Mirsamiqov Mirislom Mirzoid o'g'li
Department of Improving Financial Literacy
and Retraining Banking Specialists
The Central Bank of the Republic of Uzbekistan

Abstract
In certain cases, one sound may be hidden by another sound. For example, talking next to railroad tracks can be completely impossible if a train passes by. This kind of effect is called disguise. It is said that a faint sound is masked if it becomes indistinguishable in the presence of a louder sound.

Аннотация
В некоторых случаях один звук может быть скрыт другим звуком. Например, говорить рядом с железнодорожными путями может быть совершенно невозможно, если поезд проходит мимо. Этот вид эффекта называется маскировкой. Говорят, что слабый звук маскируется, если он становится неразличимым при наличии более громкого звука.

Keyword
Digital coding , masking, compression , compression of the sound, the sound , the compression of audio data , frequency masking , Temporary masking decoder formats sounds .

Ключевое слово
Цифровое кодирование, маскирование, сжатие, сжатие звука, звука, сжатие аудиоданных, частотное маскирование, декодер временного маскирования форматов звуков.

Simultaneous masking - Any two sounds while listening at the same time affect the perception of the relative volume between them. A louder sound reduces the perception of a weaker one, up to the disappearance of its audibility. The closer the frequency of the masked sound to the frequency of the masking, the more it will hide. The masking effect is not the same when the masked sound is shifted lower or higher in frequency relative to the masking one. Low - frequency sound masks high - frequency. It is important to note that high - frequency sounds cannot mask low - frequency sounds .

Temporary masking - This phenomenon is similar to frequency masking, but here there is a masking in time. When the masking sound is stopped, the masked one continues to be inaudible for 94 some time. Under normal conditions, the effect of temporary masking lasts much less. The masking time depends on the frequency and amplitude of the signal and can reach 100 ms . In the case when the masking tone appears later than the masked one, the effect is called post - masking. When a masking tone appears earlier than a masked tone (such a case is also possible), the effect is called pre - masking. Post - stimulus fatigue - Often, after exposure to high - intensity loud sounds, a person's auditory sensitivity decreases sharply. Recovery to normal thresholds can last up to 16 hours.

This process is called a "temporary shift in the threshold of auditory sensitivity" or " post - stimulus fatigue." The threshold shift begins to appear when the sound pressure level is above 75 dB and accordingly increases with increasing signal level. Moreover, the highest influence on the shift of the sensitivity threshold is exerted by the high - frequency components of the signal. Lossy audio data compression is based on the imperfection of human hearing in the perception of sound information. The inability of a person in certain cases to distinguish between quiet sounds in the presence of louder ones, called the masking effect, was used in algorithms for reducing psychoacoustic redundancy.

The effects of auditory masking depend on the spectral and temporal characteristics of the masked and masking signals and can be divided into two main groups: Frequency (simultaneous) masking Temporary (non - simultaneous) masking. The masking effect in the frequency domain is due to the fact that in the presence of large sound amplitudes the human ear is insensitive to small amplitudes of close frequencies. That is, when two signals are simultaneously in a limited frequency domain, the weaker signal becomes inaudible against the background of the stronger one. Masking in the time domain characterizes the dynamic properties of hearing, showing a change in time of the relative threshold of audibility (the threshold of audibility of one signal in the presence of another), when the masked and masked signals do not sound simultaneously. In this case, it is necessary to distinguish between the phenomena of post - masking (changing the threshold of audibility after a high level signal) and pre - masking (changing the threshold of audibility before the arrival of the maximum level signal).

A weaker signal becomes inaudible 5 to 20 ms before the masking signal is turned on and becomes audible 5 0 to 200 ms after it is turned on. The best method of coding sound, taking into account the effect of masking, is strip coding. Its essence is as follows. A group of samples of the input audio signal, called a frame, is fed to a filter unit that divides the signal into frequency subbands. The output of each filter is the part of the input signal that falls into the passband of this filter. Further, in each band, using the psychoacoustic model, the spectral composition of the signal is analyzed and it is estimated which part of the signal should be transmitted without abbreviations, and which lies below the masking threshold and can be quantized to a smaller number of bits.

To reduce the maximum dynamic range, the maximum sample in the frame is determined and the scaling factor is calculated, which brings this sample to the upper quantization level. This operation is similar to companding in analog broadcasting. All other samples are multiplied by the same factor. A scaling factor is transmitted to the decoder along with the encoded data to correct the gain of the latter. After scaling, the masking threshold is estimated and the total number of bits is redistributed between all the bands. 95 Obviously, after eliminating the psychoacoustic redundancy of sound signals, their exact restoration during decoding is no longer possible. Methods of eliminating psychophysical redundancy can provide compression of digital audio data 10 to 12 times without significant loss in quality. Many other tricks can serve as a way to reduce the amount of audio information data.

Even a simple narrowing of the signal frequency band together with a decrease in the dynamic range can already be called compression of audio data. For example, the cellular audio compression standard uses both. In an effort to remove redundancy from sound, a codec with poor signal quality becomes selective for certain words, stubbornly swallowing them. Generally accepted data compression methods, such as RLE , statistical and dictionary methods, can be used to compress sound files without loss, but the result depends heavily on specific audio data. Some sounds will compress well with RLE , but poorly with statistical algorithms. Other sounds are more suited to statistical compression, but with a dictionary approach, on the contrary, expansion can occur. Here is a brief description of the effectiveness of these three methods in compressing audio files. RLE works well with sounds that contain a long series of repeating sound bites - samples . With 8 - bit sampling, this can happen quite often. Recall that the difference in electrical voltage between two 8 - bit samples is about 4 mV . A few seconds of homogeneous music, in which the sound wave will change by less than 4 mV , will generate a sequence of thousands of identical samples . With 16 - bit sampling , obviously, long repeats are less common, and therefore, the RLE algorithm will be less efficient.

Statistical methods assign variable - length codes to sound samples according to their frequency. With 8 - bit sampling , there are only 256 different samples , so a large sound file sampled sound can be distributed evenly. Such a file cannot be compressed well by the Huffman method. With 16 - bit sampling, more than 65,000 audio fragments are allowed. In this case, it is possible that some samples will occur more often and others less often. With strong asymmetry of probabilities, good results can be achieved using arithmetic coding.

Vocabulary based methods suggest that certain phrases will occur frequently throughout the entire file. This occurs in a text file in which individual words or their sequences are repeated multiple times. The sound, however, is an analog signal and the values of the specific generated samples are heavily dependent on the operation of the ADC. For example, with 8 - bit sampling , an 8 mV wave becomes a numerical sample equal to 2, but a wave close to it, say, at 7.6 mV or 8.5 mV may become a different number. For this reason, speech fragments containing matching phrases and sounding the same for us may slightly differ when they are digitized. Then they will fall into the dictionary in the form of different phrases, which will not give the expected compression. Thus, vocabulary methods are not very suitable for compressing sound.

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