Equalization for Noisy Listening Environments

Alec Wright
Aalto University

alec.wright@aalto.fi

Abstract

Listening to music and other media is often done in noisy environments. This is especially true with headphone listening in public spaces. Due to auditory masking caused by the background noise, there is a frequency dependent effect which either reduces the perceived volume, or renders totally inaudible, certain parts of the frequency spectrum. Simply increasing the volume might make all the frequencies audible, however as the masking effect is frequency dependant it will result in undesired changes to the perceived frequency spectrum. This paper describes the existing research into *intelligent equalization*, which are methods that seek to quantify, in real-time, the perceptual impact of background noise and apply compensatory equalization to mitigate the masking effect. The most recently published work in this topic presents a system which successfully adapts in real-time based on the background noise and media itself, whilst considering the isolation capability and frequency response of the headphones being used. The resulting system applies a significant level boost at the most heavily masked frequencies, whilst keeping the overall level boost relatively low.

1 Introduction

The topic of this seminar paper is equalization for listening in ambient noise. This is a topic which can apply to loudspeaker listening in a noisy environment (such as in a car), or to headphone listening. In both cases, the user is listening to the *target audio* (such as music or the radio), and the presence of background noise has a detrimental effect on how the target audio is perceived.

Headphones and loudspeakers are generally designed to achieve a certain target frequency response for playback. Headphone listening (and sometimes loudspeaker listening) is often done in the presence of significant background noise, such as on a busy street or on public transport. The frequency response perceived by the listener in these conditions can differ significantly from the target frequency response. Whilst headphones have existed for many decades, the issue of headphone listening in noisy environments has only been a subject of study in relatively recent years. This is most likely due to the mobile listening revolution experienced since the early 2000s, with the introduction of portable music players and smartphones. Whilst the problem of noisy listening environments can apply to both headphones and loudspeakers, this seminar paper focuses on headphone listening.

To achieve an effective equalization system for noisy listening environments, there are two distinct tasks which must be achieved. Firstly the effect of the ambient noise must be estimated. The effect of the ambient noise varies with frequency and depends on the frequency spectra of both the ambient noise and the target audio. This estimation is made using an *auditory masking model*. The second task is to apply equalization to the target audio, to compensate for the effect predicted by the auditory masking model.

Auditory masking and the prediction of auditory masking has a range of applications. In lossy audio coding (such as MP3 or AAC), it is used to determine which parts of the signal can be removed with minimal effect on listener perception [9, 2, 15, 26]. Masking is also used in the management of tinnitus, where a noise signal can mask the ringing tone experienced due to the tinnitus [27, 19]. It has also been used in automatic speech recognition [23] and in digital speech coding [20]. In [16] the masking effect of listening to music in noisy environments was simulated, allowing the effect of auditory masking to be heard in a controlled environment.

Various research has also been conducted to develop systems to apply real-time compensation to mitigate the effects of masking caused by background noise. Early work, published from the late 80s to the early 90s, focused on automotive applications [3, 11, 12], in which loudness compensation was applied based on the noise level in the car. In the early 2000s, adaptive equalization systems which apply EQ based on the target and background noise signals were proposed [25, 21]. In [18] the effects of background noise on mobile TV listening were investigated, with compression and noise-specific equalization being used to compensate. A more recent automatic EQ system for automobiles was presented in 2011 [10], which also considered the music genre. Finally, in 2013, a perceptually motivated adaptive real-time EQ system for headphone listening was presented [17], which considered the headphone frequency response and isolation effects.

The rest of this paper is structured as follows. Section 2 provides some background theory which is relevant to the topic of intelligent equalization. Section 3 describes how auditory masking can be predicted, based on analysis of the noise and target signals. Section 4 discusses various solutions which have been proposed for listening in noisy environments. Section 5 concludes the paper.

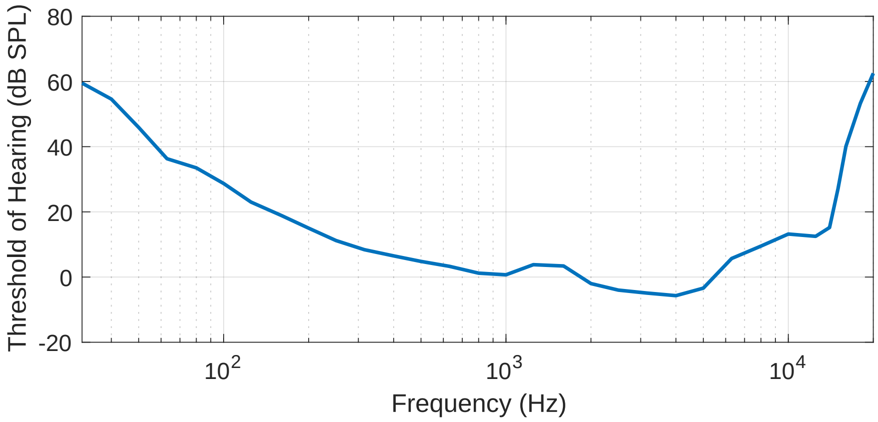

Figure 1: *Threshold of human hearing against frequency, recreated from [24]*

2 Background

It is well known that the presence of unwanted background noise is detrimental when listening to music or other media. The cause of this is *auditory masking*, which can result in reduced audibility at certain frequencies. To combat this an intelligent equalization system can be applied, which compensates for the disturbance caused by the background noise. To create the intelligent equalization algorithm it is first necessary to define what frequency response the equalizer is trying to achieve. In this section, to allow for a clearer definition of the problem, theory relevant to listening in noisy environments is described.

2.1 Auditory Masking

Auditory Masking is the main mechanism through which background noise affects the listening experience. It occurs when different sounds are heard concurrently, and it affects the threshold of hearing and the loudness perceived by the user. Generally there is a *masker* signal, which masks a *maskee* signal. In the context of this seminar paper, the masker is the background noise and the maskee is the media being played through the headphones.

The threshold of hearing for a given frequency is defined as the minimum level (in dB SPL) that a listener can perceive a pure tone of that frequency at, in the absence of any other sound. Figure 1 shows a plot of the threshold of human hearing against frequency, recreated from previously published research [24]. The key concept in auditory masking is that the threshold of hearing for a given frequency can be increased , when in the presence of another noise.

A sound with an increased threshold of hearing means that it has to be louder before a human can perceive it. This leads to the definition of two terms, the

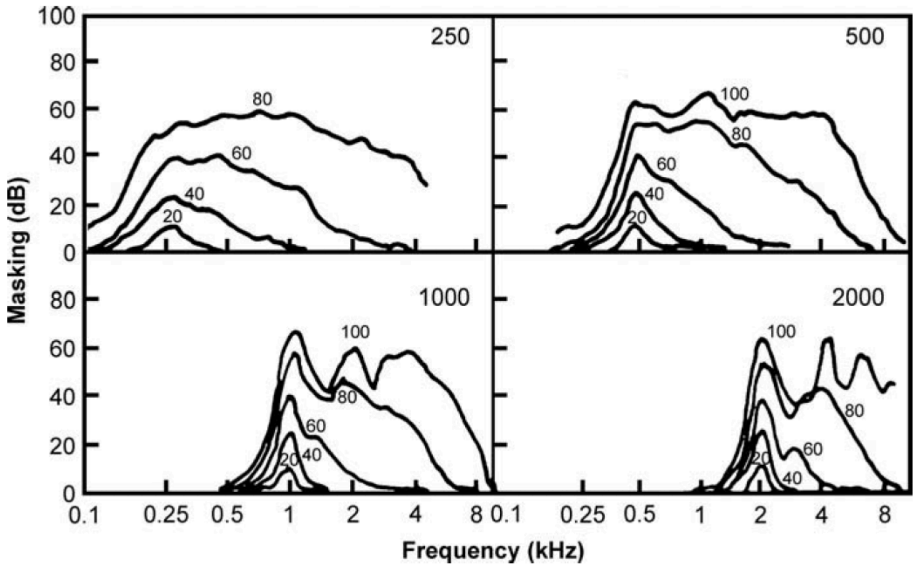

Figure 2: *Masking patterns produced by pure tone maskers of different frequencies (masker frequency indicated in top right of each plot), figure from [5]*

unmasked threshold, which is the threshold of hearing for a sound produced in a quiet environment, and the *masked threshold*, which is the threshold of hearing for a sound produced in the presence of a *masker* signal. Generally, the masking effect is stronger at frequencies closer to those found in the masker signal, although masking does occur at other frequencies also.

Masking also includes what is sometimes referred to as *partial masking* [5], this is where the perceived loudness of a sound is reduced, when in the presence of another sound. Listeners will generally increase the overall volume when the background noise level increases. However, as background noise generally doesn't have a flat spectrum, the frequency response perceived by the listener will be adjusted when listening in the presence of noise.

Auditory masking is quite complex and generally is measured through listening tests. It has been known for some time, for example, that for pure tones, low frequency tones are effective at masking high frequency tones, whilst high frequency tones are comparatively quite bad at masking low frequency tones. This can be seen in Figure 2, which shows the *masking patterns* of four pure tone masking signals at varying intensities. This phenomenon is often referred to as spreading, as the masking effect of a sound *spreads* to other frequencies. As can be seen in the figure, spreading occurs much more strongly in the upward direction, then in the downward direction.

Auditory masking tells us that when listening to headphones in the presence of ambient noise, simply increasing the gain of the signal is not the most effective way to adjust the headphone signal. Instead the signal level should be filtered to boost the gain at the frequencies which are masked most heavily by the ambient noise.

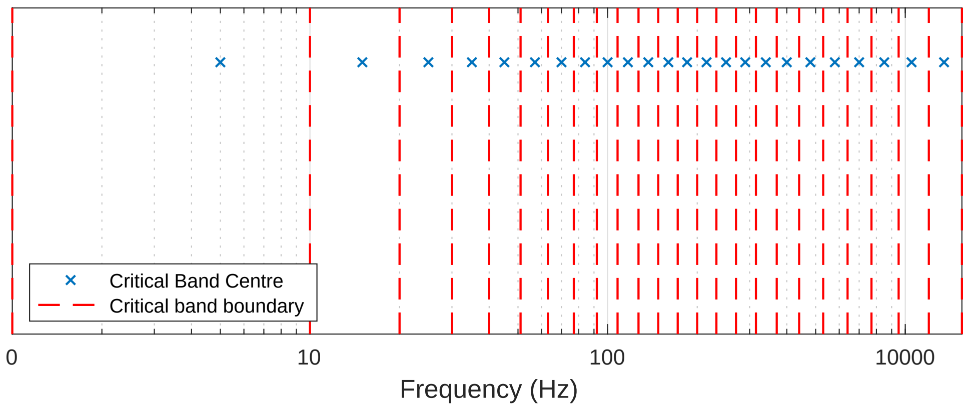

Figure 3: *Critical Bands of Human Hearing*

2.2 Critical Bands of Hearing

An important aspect of human hearing is that all frequencies are not perceived equally. Critical bands were first introduced by Fletcher in 1933 [4]. They are based on the idea that a masking noise only masks sounds which are within a certain distance of the frequencies found in the masking noise. For a masking tone, the critical bandwidth is the range of frequencies over which that masking tone will cause auditory masking. The critical bands have been determined through listening tests. At frequencies up to 500 Hz, the critical bandwidth has been found to have a constant bandwidth of about 100 Hz. Above this the critical bandwidth tends to be about 20 % of the centre frequency [29].

Whilst, as discussed in Section 2.1, masking is actually not limited entirely to the critical band of the masking frequency, the masking effect is strongest within its own critical band boundaries. As such, critical bands remain a useful idea when estimating masking effects, as they allow the masking effect to be divided into discrete bands. The spread between critical bands can then be estimated using a spreading model.

2.3 Psychoacoustical Scales

When optimizing sound reproduction systems, it makes sense to use a scale that doesn't apply equal weighting to each frequency. The Bark scale [28] is based on the critical bands of hearing and is commonly used as a more perceptually accurate representation of human hearing. The centre frequency of each critical band of human hearing corresponds to 1 unit in the Bark scale. In practice the Hz to Bark conversion can be carried out using the following approximation [29]:

$$b = 13\mathrm{arctan}\left(\frac{0.76f}{\mathrm{kHz}}\right) + 3.5\mathrm{arctan}\left(\frac{f}{7.5\mathrm{kHz}}\right)^2, \tag{1}$$

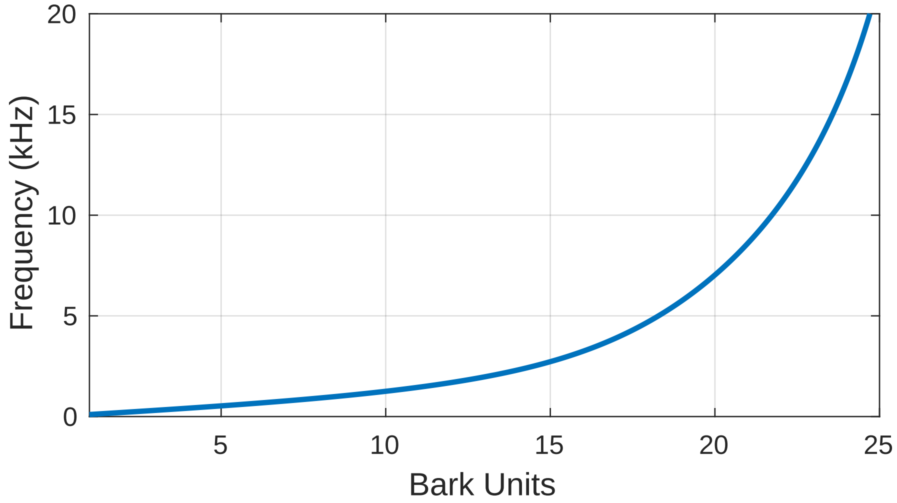

Figure 4: *Bark units against frequency*

where f is the frequency in Hz and b is the corresponding frequency in Bark units. Figure 4 shows the Bark scale plotted against frequency.

Another commonly used psychoacoustical scale is the Mel Scale [22]. This scale is also defined based on human perception, and is intended to correspond to the perceived pitch difference between tones.

3 Models for Auditory Masking

The first stage in developing an intelligent EQ system is to predict the extent of the masking by the ambient noise over the frequency spectrum of the target audio. As the music signal and the noise signal are both time variant and are not known in advance, this requires the use of a masking model that can predict the masking effect of an environment in real time.

Figure 5 depicts headphone listening in a noisy environment. Depending on the headphone design, the ambient noise present in the listening environment is attenuated somewhat. This property is known as the *headphone isolation*. What the listener hears is the *masker* signal, which is the headphone isolated ambient noise signal, in combination with the *maskee* signal, which is the signal being produced by the headphones.

The extent of the masking which occurs is influenced by the relative volume and frequency content of the masker and the maskee signals. There is some work focusing on the prediction of perceived loudness in the presence of a masking sound [6, 13], however these generally deal with fairly simple sounds. The main source used for the auditory masking model described in this section is [16, 17], which presents a complete model for estimating the masking effects of background noise on music.

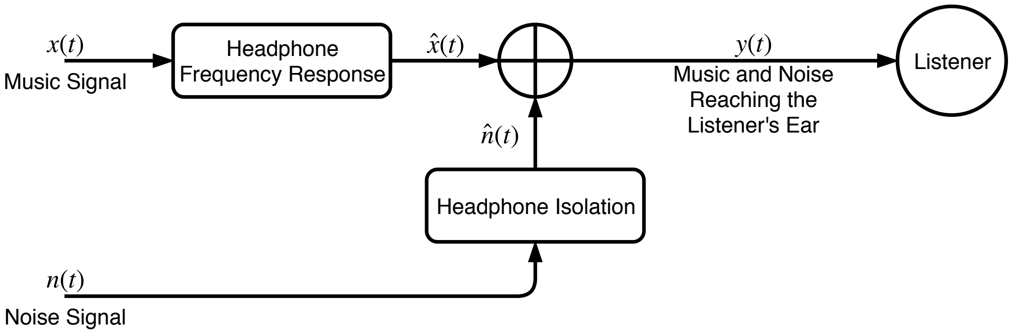

Figure 5: *Headphone listening in a noisy environment*

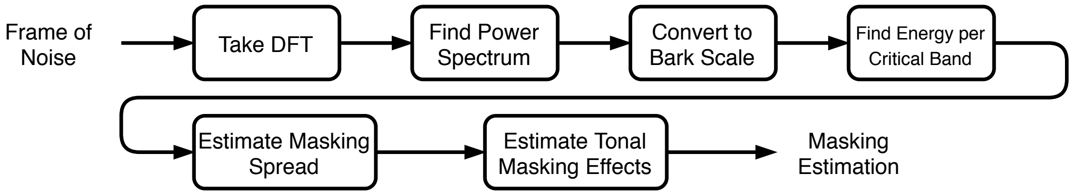

Figure 6: *Process proposed in [17] to estimate masking effects*

Generally the masking effect is estimated separately for each of the critical bands of human hearing (shown in Figure 3). For each critical band, the threshold of hearing in the presence of the background noise is estimated, as well as the partial masking effect. An block diagram of the process for estimating the masking effect is shown in Figure 6.

3.1 Finding Critical Band Power

The first four stages shown in Figure 6 are concerned with determining the power spectrum of noise and the power in each of the critical bands. The noise is analyzed on a frame by frame basis. The frame is first converted into the frequency domain using the Discrete Fourier Transform (DFT). For each DFT bin, the power is found by taking the squared magnitude. As the critical band bandwidths increase with frequency, the number of DFT bins in each critical band also increases with frequency. The power contained in each critical band is thus estimated by taking the average power of the DFT bins contained in that band. Instead of explicitly using the critical band boundaries, the authors of [17] use an approximation (shown in equation 4) to map the frequency from Hz onto the Bark scale. Figure 7 shows the power spectrum and power per critical band of a short segment of background noise recorded from a bar.

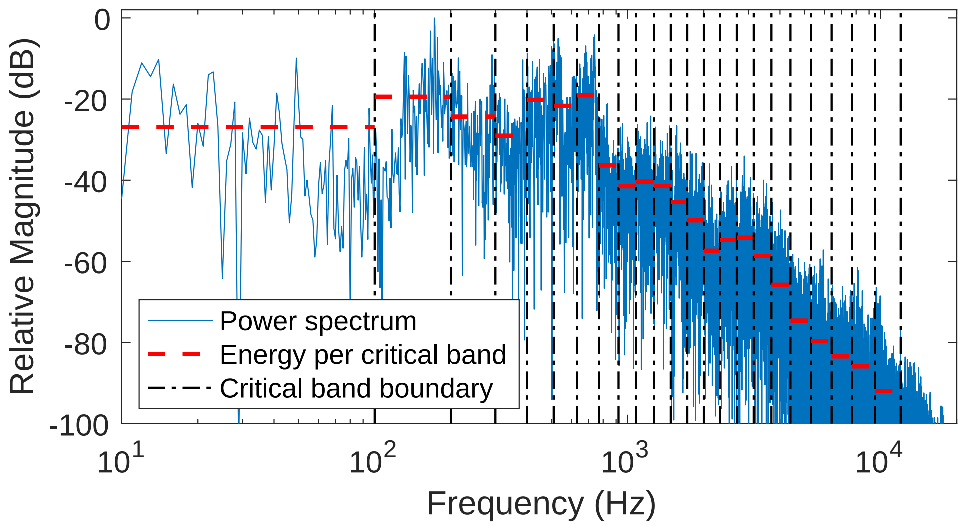

Figure 7: *Power Spectrum and energy per critical band for frame of recorded background noise*

3.2 Mask Spread

The next stage is to estimate the effects of spreading between the critical bands. Masking sounds cause masking over a range of frequencies. Generally, the masking is strongest close to the frequencies contained in the masker. Masking also occurs above and below the masker frequencies. Frequencies above the masker are generally masked more strongly than frequencies below the masker frequencies. To estimate the spread between critical bands, the following function can be used [1]:

$$10\log_{10}[B(\Delta v, L_M)] = [-27 + 0.37\max\{L_M - 40, 0\}\theta(\Delta v)]|\Delta v|, \tag{2}$$

where Δv is the Bark Scale difference between the masker and maskee frequencies, L_M is the SPL of the masker, $\theta(\Delta v)$ is a step function equal to zero for negative Δv and equal to one otherwise. The step function means that the estimated spread for frequencies above the masker is greater than for frequencies below the masker.

The overall masking effect can then be calculated for each critical band in turn. For each band the masking contribution from itself as well as from each of the other bands is estimated using Equation 2, then summed according to the following formula:

$$S_P = \left(\sum_{v=1}^{N_c} B_v^{\alpha} \right)^{1/\alpha}, \tag{3}$$

Where S_P represents the masking effect at the critical band P, N_c is the number of critical bands, v is the critical band number and α is a parameter used to control how the effect from each critical band is summed. An α value of 1 means that the combined effect of all the masks is simply their sum. As α increases the effect of the strongest masking frequency becomes increasingly significant, with it becoming the only component as α tends to infinity. If α is chosen to be lower than one, it results in the combination of two masks being greater than the sum of their individual intensities.

3.3 Tonality Effects

The final stage in the estimation of the noise masking threshold involves consideration of the tonal characteristics of the noise signal. Generally, broadband noise is a more effective mask than a tone. The masker is analyzed to determine its *spectral flatness* in dB, which is defined as [9]:

$$V_m = 10\log_{10}\frac{\prod_{k=0}^{N-1} P_m(k)^{\frac{1}{N}}}{\frac{1}{N}\sum_{k=0}^{N-1} P_m(k)}, \tag{4}$$

where $P_m(k)$ is the power spectrum at DFT bin k and N is the number of bins in the DFT. V_m is used to define the tonality factor of the noise,

$$\alpha_m = \min\big(V_m/V_{\max}, 1\big), \tag{5}$$

where V_{max} is chosen to -60 dB. The tonality factor is thus a number between 0 and 1 which indicates whether the background noise is purely tone-like ($\alpha_m = 1$) or spectrally flat noise ($\alpha_m = 0$).

Two different thresholds for hearing in noise have been proposed by Johnston [9], for a pure tone mask, masking noise, the threshold is estimated as $14.5 + v$ dB below the total masking effect in that critical band, S_P (given by Equation 3). For noise masking a tone, the hearing threshold is estimated to 5.5 dB below the total masking effect at that critical band.

These two different hearing thresholds and the tonality factor are then used to calculate the offset between the masking effect and the threshold of hearing:

$$U_m(v) = \alpha_m(14.5 + v) + (1 - \alpha_m)5.5. \tag{6}$$

For each critical band, the threshold of hearing in the presence of the noise (or the masked threshold of hearing) is then calculated to be:

$$R_m(v) = 10^{\log_{10}(S_P(v)) - \frac{U_m(v)}{10}}. \tag{7}$$

The final step is to compare R_m with the absolute threshold of hearing, if the absolute threshold of hearing is greater than R_m then it is used instead of the masked threshold of hearing for that critical band.

3.4 Partial Masking

The previous steps determine the masked threshold of hearing for the target in the presence of noise. This can inform you of the level required before that frequency band becomes audible, but doesn't describe the extent of the partial masking effects. In [17] the authors presented a model for partial masking, based on listening tests which they conducted. This consisted of playing some musical sounds to the participants. Each musical sound was played in the presence of noise and on its own. The participants would then adjust the volume of the musical sound without the presence of noise, until it matched the volume of the musical sound in the presence of noise. The amount of reduction in level required to make the perceived volume of the musical sound with and without the presence of noise equal, indicates how much partial masking that level of noise causes.

4 Solutions

In the previous section, a complete masking model which quantifies the effects of listening in a noisy environment was described. This section initially describes a couple of methods used to deal with noisy listening environments, which don't involve this masking model. The final part of this section describes intelligent equalization systems which analyze the noise and target audio in order to determine how to compensate for the ambient noise.

4.1 Volume adjustment/compression

Research on intelligent audio equalization in noisy environments began with work on automotive applications [3, 11, 12], with the aim of improving listening conditions within automobiles. The general approach used was to measure the noise level in the vehicle and apply volume adjustment and/or compression to compensate. This approach is limited in its usefulness, as it will cause an uneven frequency adjustment to the audio, due to the frequency dependant auditory masking effects discussed in Section 2.1.

4.2 Active Noise Control

Active Noise Control (ANC) is an effective solution used for listening in noisy environments. ANC uses the background noise to derive a second signal, which is intended to cancel the background noise when summed with it. ANC can produce effective noise cancellation, however it is limited in how much reduction in background noise it can achieve. As such, for noisy listening environments, masking still may occur even if ANC is applied.

4.3 Intelligent EQ

This section briefly describes some of the earlier published work related to intelligent equalization in noisy environments, before describing the most recently published work in greater detail.

4.3.1 Earlier Research

In the paper *Sound Equalization in a Noisy Environment* [25] the authors proposed an EQ for noisy environments. They note that depending on the listening environment both the level of the noise and its spectral content varies considerably, and that the inside of a car generally has a much more predictable ambient noise level/spectrum than an environment where a mobile phone user might be listening to audio. They propose a system to measure the ambient noise and apply compensatory EQ. The system does consider the masking caused at each critical band based on the noise and target signal, however the compensatory EQ is only applied if the target audio level is estimated to be below the masked threshold of hearing. As such this scheme may be effective in making sure that all parts of the signal are *audible*, but it does not account for partial masking effects.

In *Loudness and Auditory masking compensation for Mobile TV* [18], the authors focus on audio for mobile TV watching. They note that both the ambient noise present and the changing audio signal itself can contribute to a decreased listener experience. Through listening tests the authors conclude that environmental noise with a higher SPL has a stronger negative effect on the user experience. They trial some combinations of multiband equalization and compression to improve the listening experience, but don't propose a system which can adapt based on the environment conditions.

In *Car Audio Equalizer System Using Music Classification and Loudness Compensation* [10], the authors present an automatic EQ system for car stereos, which considers the music genre classification and uses loudness compensation targeted at certain frequencies. This system analyses the music in frames to classify its genre, using a variant of Mel-Frequency Cepstral Coefficients (an audio feature commonly used in speech recognition [7]). The system applies an equalizer based on the predicted genre of the music. The loudness of the noise is also measured and compared to target signal, and a loudness compensation is applied.

4.3.2 Most Recent Research

In *Perceptual Headphone Equalization for Mitigation of Ambient Noise* [17], an EQ scheme for headphone listening in ambient noise was proposed. A simplified block diagram of the system is shown in Figure 8.

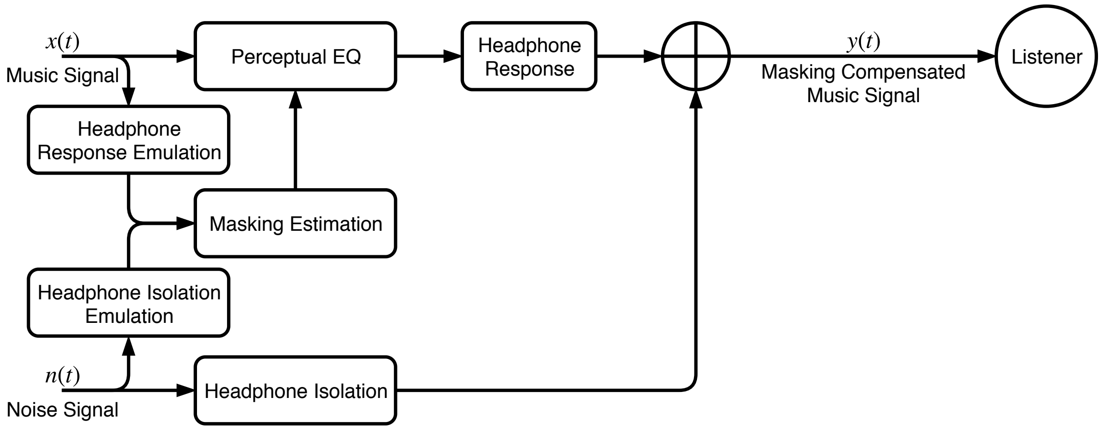

Figure 8: *Simplified block diagram of the perceptual headphone equalization system presented in [17]*

The masking is estimated using the system described in Section 3 [16]. For each frame of audio, the effects of the headphone frequency response on the music signal and the headphone isolation on the noise signal are first emulated. Then the music and noise signal are used to estimate the masking effect. The predicted masking effect is then used to determine a target gain to be applied to each critical band.

To avoid sudden changes in gain between audio frames, gain averaging was also applied. The number of previous frame gains to average over is a parameter that can be selected by the user. This presents an additional problem, when the user moves from a noisy to a quiet environment abruptly. To manage this a noise activity detector was also implemented, which detects when the mean energy in each critical band of the noise drops below a threshold value. If it falls below the threshold, the EQ and average gain for that critical band are reset. Similarly a music activity detector was included, to detect quiet parts of the music.

The compensatory gain was applied using a graphic EQ, based on filter designs proposed in [14, 8]. These were chosen as the gain interaction between adjacent frequency bands is relatively low. The resulting system was found to work well when limited to the first nine bands of critical hearing. This also corresponds to the region where a lot of ambient noise tends to be loudest, and also where headphone isolation tends to be weakest. The system allowed for an EQ that adapts to the environmental noise, counteracting the masking effect whilst limiting the overall level increase required.

5 Conclusions

This seminar paper has provided an overview of research into intelligent equalization for noisy listening environments. The main challenge presented by intelligent equalization is determining, in real-time, an appropriate equalization curve to apply to the target audio. Some background into relevant phenomena was provided, such as auditory masking and the critical bands of hearing. A complete scheme for estimating the masking effect of noise was also described. A review of research into automatic equalization was presented. The most up to date research reviewed in this paper implemented an auditory masking model to predict the masking effect in real-time, and then applied a compensatory EQ.

References

[1] Bosi, M., and Goldberg, R. E. *Introduction to digital audio coding and standards*, vol. 721. Springer Science & Business Media, 2012.

[2] Brandenburg, K., and Bosi, M. Overview of mpeg audio: Current and future standards for low bit-rate audio coding. *Journal of the Audio Engineering Society 45*, 1/2 (1997), 4–21.

[3] Clark, D., Blind, H., Dorfstatter, W., and Geddes, E. Compensation for road noise in automotive entertainment systems. Tech. rep., SAE Technical Paper, 1987.

[4] Fletcher, H., and Munson, W. A. Loudness, its definition, measurement and calculation. *Bell System Technical Journal 12*, 4 (1933), 377–430.

[5] Gelfand, S. A. *Hearing: An introduction to psychological and physiological acoustics*. CRC Press, 2017.

[6] Glasberg, B. R., and Moore, B. C. Development and evaluation of a model for predicting the audibility of time-varying sounds in the presence of background sounds. *Journal of the Audio Engineering Society 53*, 10 (2005), 906–918.

[7] Holmes, W. *Speech synthesis and recognition*. CRC press, 2001.

[8] Holters, M., and Zölzer, U. Graphic equalizer design using higher-order recursive filters. In *Proc. Int. Conf. Digital Audio Effects (DAFx-06)* (2006), pp. 37–40.

[9] Johnston, J. D. Transform coding of audio signals using perceptual noise criteria. *IEEE Journal on selected areas in communications 6*, 2 (1988), 314–323.

[10] Kim, H.-G., and Cho, J.-M. Car audio equalizer system using music classification and loudness compensation. In *ICTC 2011* (2011), IEEE, pp. 553–558.

[11] Kitzen, W. J., Kemna, J.-W., Druyvesteyn, W., Knibbeler, C. L., and van de Voort, A. T. Noise-dependent sound reproduction in a car: Application of a digital audio signal processor. *Journal of the Audio Engineering Society 36*, 1/2 (1988), 18–26.

[12] Miller, T. E., and Barish, J. Optimzing sound for listening in the presence of road noise. In *Audio Engineering Society Convention 95* (1993), Audio Engineering Society.

[13] Moore, B. C., Glasberg, B. R., and Baer, T. A model for the prediction of thresholds, loudness, and partial loudness. *Journal of the Audio Engineering Society 45*, 4 (1997), 224–240.

[14] Orfanidis, S. J. High-order digital parametric equalizer design. *Journal of the Audio Engineering Society 53*, 11 (2005), 1026–1046.

[15] Painter, T., and Spanias, A. Perceptual coding of digital audio. *Proceedings of the IEEE 88*, 4 (2000), 451–515.

[16] Rämö, J., Välimäki, V., Alanko, M., and Tikander, M. Perceptual frequency response simulator for music in noisy environments. In *Audio Engineering Society Conference: 45th International Conference: Applications of Time-Frequency Processing in Audio* (2012), Audio Engineering Society.

[17] Rämö, J., Välimäki, V., and Tikander, M. Perceptual headphone equalization for mitigation of ambient noise. In *2013 IEEE International Conference on Acoustics, Speech and Signal Processing* (2013), IEEE, pp. 724–728.

[18] Sack, M. C., Buchinger, S., Robitza, W., Hummelbrunner, P., Nezveda, M., and Hlavacs, H. Loudness and auditory masking compensation for mobile TV. In *2010 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB)* (2010), IEEE, pp. 1–6.

[19] Schleuning, A., and Johnson, R. Use of masking for tinnitus. *International Tinnitus Journal 3*, 1 (1997), 25–29.

[20] Schroeder, M. R., Atal, B. S., and Hall, J. Optimizing digital speech coders by exploiting masking properties of the human ear. *The Journal of the Acoustical Society of America 66*, 6 (1979), 1647–1652.

[21] Sergey, K., Budkin, A., and Goldin, A. A. Automatic volume and equalization control in mobile devices. In *Audio Engineering Society Convention 121* (2006), Audio Engineering Society.

[22] Stevens, S. S., Volkmann, J., and Newman, E. B. A scale for the measurement of the psychological magnitude pitch. *The Journal of the Acoustical Society of America 8*, 3 (1937), 185–190.

[23] Strope, B., and Alwan, A. A model of dynamic auditory perception and its application to robust word recognition. *IEEE transactions on Speech and Audio Processing 5*, 5 (1997), 451–464.

[24] Takeshima, H., Suzuki, Y., Kumagai, M., Sone, T., Fujimori, T., and Miura, H. Threshold of hearing for pure tone under free-field listening conditions. *Journal of the Acoustical Society of Japan (E) 15*, 3 (1994), 159–169.

[25] Tzur, M., and Goldin, A. A. Sound equalization in a noisy environment. *PREPRINTS-AUDIO ENGINEERING SOCIETY* (2001).

[26] Van De Par, S., Kohlrausch, A., Charestan, G., and Heusdens, R. A new psychoacoustical masking model for audio coding applications. In *2002 IEEE International Conference on Acoustics, Speech, and Signal Processing* (2002), vol. 2, IEEE, pp. II–1805.

[27] Vernon, J., and Schleuning, A. Tinnitus: A new management. *The Laryngoscope 88*, 3 (1978), 413–419.

[28] Zwicker, E. Subdivision of the audible frequency range into critical bands (frequenzgruppen). *The Journal of the Acoustical Society of America 33*, 2 (1961), 248–248.

[29] Zwicker, E., and Fastl, H. *Psychoacoustics: Facts and models*, vol. 22. Springer Science & Business Media, 2013.