Wearable Technology (2022) Volume 3 Issue 2, *pp. 99–111*.
doi: 10.54517/wt.v3i2.2140

Asia Pacific Academy of Science Pte. Ltd.

ORIGINAL RESEARCH ARTICLE

Binaural hearing and the use of hearing aids and cochlear implants

***Fernando Martín San Victoriano*[1*]*, Enrique A. López-Poveda*[1,2,3]**

*[*1] Instituto de Neurociencias de Castilla y León, Universidad de Salamanca, Salamanca 37008, Spain. E-mail: fermartin@usal.es*
[2] Instituto de Investigación Biomédica de Salamanca, Universidad de Salamanca, Salamanca 37008, Spain.
[3] Departamento de Cirugía, Facultad de Medicina, Universidad de Salamanca, Salamanca 37008, Spain.

ABSTRACT

Binaural hearing is essential to understand speech in noisy environments and to determine the location of sound sources. This article discusses various signal processing strategies aimed at improving these aspects of hearing in people using hearing AIDS and cochlear implants. Two binaural and bio-inspired strategies developed by the university of salamanca are described, the so-called 'moc strategy' and an algorithm for canceling contralateral sounds. It is shown that these strategies can significantly improve speech intelligibility in noise for some spatial configurations of sound sources, without impairing their location. Despite these and other promising advances in improving the effectiveness of hearing AIDS and cochlear implants, the hearing achieved by people with hearing loss is still far from equal to that of a person with normal hearing. This makes further research in this field necessary.

Keywords: binaural hearing; hearing aids; cochlear implants; intelligibility in noise; signal processing

1. Introduction

The simultaneous use of two ears is essential in human hearing. Our brain is able to analyze the differences between the sounds captured by each ear and extract from them clues that are used, for example, to improve speech intelligibility in noisy environments or to localize sound sources.

This ability is usually impaired in people with hearing loss and the use of hearing aids or hearing implants, far from restoring it, alters it to a greater extent[1]. To solve this problem, research is being conducted on sound processing strategies that can restore binaural cues in hearing aid users.

In this article, some of these strategies are described, emphasizing those developed by the Computational and Psychoacoustic Hearing laboratory of the University of Salamanca.

2. The human auditory system

The functioning of the healthy ear, described very succinctly, is as follows. Sound waves reach the pinna and travel through the external auditory canal until they vibrate the eardrum. This vibration is

ARTICLE INFO
Received: September 1, 2022 | Accepted: October 3, 2022 | Available online: October 20, 2022

CITATION
Victoriano FMS, López-Poveda EA. Binaural hearing and the use of hearing aids and cochlear implants. Wearable Technology 2022; 3(2): 99–111.

transmitted by the chain of ossicles of the middle ear to reach the oval window, a membrane that covers the entrance of the cochlea. The oval window, the beginning of the inner ear, is thus subjected to mechanical vibrations that generate a pressure difference between it and the round window, a membrane located at the other end of the cochlea. These pressure differences displace the organ of Corti and thus the cilia of the hair cells present in it. The displacement of the cilia causes an electrical depolarization of the hair cells which is ultimately responsible for the emission of action potentials by the neurons of the auditory nerve, thus transforming sound into neuronal 'firings'. These firings ascend from the nerve to the auditory cerebral cortex via what is known as the afferent pathway, passing through different neuronal nuclei in the brain.

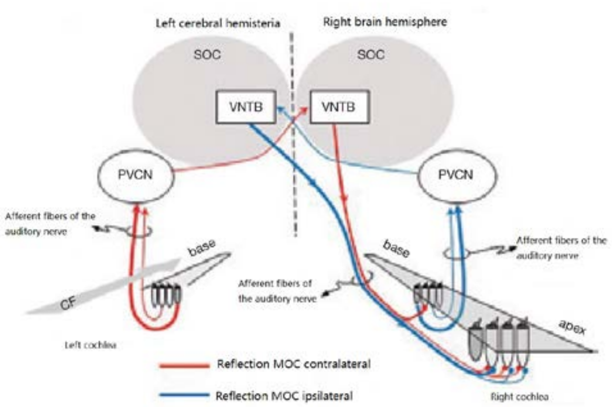

Figure 1. Activation pathways of efferent fibers responsible for the ipsilateral (blue) and contralateral (red) MOCRs in the right cochlea. Red illustrates how afferent (ascending) fibers from the left cochlea carry neural signals to the posteroventral cochlear nucleus (PVCN), which in turn transmits them to the ventral nucleus of the trapezoid body (VNTB) on the right side of the brain and from there to the right cochlea. Blue illustrates how afferent fibers from the right cochlea carry signals to the left-sided VNTB and from there to the right-sided cochlea. figure adapted from[2].

There is also, however, a descending pathway (or efferent pathway) that connects the auditory neuronal nuclei of the brain with the cochlea, allowing the auditory brain to control the mechanical vibrations of the organ of Corti dynamically during listening[2]. In other words, the auditory brain does not merely receive and interpret the acoustic stimuli it receives, but controls, to some extent, how these sounds are received.

An example of this efferent control is the medial olivocochlear reflex, or MOCR. There are two bundles of nerve fibers connecting the neurons of the ventral nuclei of the trapezoid body (VNTB) of both cerebral hemispheres with the outer hair cells of the same cochlea (**Figure 1**). The efferent fibers terminating in one cochlea can be activated by sounds picked up by the same ear (ipsilateral sounds) and/or by sounds picked up by the opposite ear (contralateral sounds). Since activation by sound is involuntary (reflex), the activation of olivocochlear fibers by ipsilateral and contralateral sounds is referred to as ipsilateral and contralateral MOCR, respectively.

Activation of the MOCR inhibits (reduces) the displacement of the organ of Corti, making a more intense sound stimulus necessary to elicit the same displacement of the organ of Corti. The fact that the sound received in one ear inhibits the displacement of the organ of Corti of the opposite ear could enhance the binaural cues generated by the head, in particular, the interaural difference in intensity.

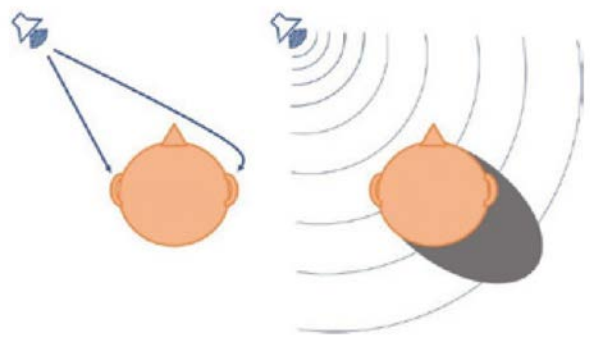

Figure 2. Representation of the paths that the sound wave travels to reach each ear (left) and the attenuation produced by the shadow effect of the head (right). The sound arrives earlier and more intensely to the left ear than to the right ear, generating interaural differences in time and intensity.

3. Binaural hearing

The fact that the head is located between the two ears causes it to function as an acoustic barrier for sound reception in the ear contralateral to the sound source (**Figure 2**). Thus, the sound produced by a sound source located outside the sagittal plane of a listener (the imaginary plane that divides the human body into two approximately symmetrical halves on the left and right) will reach each of his ears with different intensities and spaced over a time span. These differences are known as interaural differences.

The interaural time difference is the interval elapsed from the time the acoustic wave front is received by the ear closest to the sound source until it is received by the ear farthest away. This time difference also means that the phase with which the sound wave reaches each ear can be different. On the other hand, the interaural intensity difference is defined as the difference in the sound level of the same signal received in both ears. This intensity difference is caused by the acoustic 'shadow' that the head exerts especially for high-pitched sounds that diffract worse around the head.

The existence of these cues, as well as the brain's ability to process them binaurally, are essential for performing such everyday auditory tasks as, for example, locating sound sources in space or carrying on a conversation in noisy environments[3].

3.1. Spatial location of sound sources

Interaural differences in time and intensity are essential clues that our brain uses to know where the sources emitting the sounds we hear are located. For low-pitched sounds (frequencies < 1,500 Hz), the interaural time differences are large enough for the brain to take advantage of them. However, to determine the position of sources emitting high-pitched sounds (frequencies > 1,500 Hz), our brain primarily uses interaural differences in intensity, as these sounds are more effectively attenuated by the head shadow effect. It is worth mentioning that, although with limitations, humans can also determine the place of origin of sounds emitted from the sagittal plane. To do so, we detect variations in the sound wave caused by acoustic reflections in the ear folds[4].

3.2. Intelligibility in noisy environments

Hearing with two ears is advantageous when we want to understand what one person (which we will call 'signal') is saying to us while another person (which we will call 'noise') is talking at the same time. In these situations, the signal and the noise are usually located in front of us, but on different sides of our head. Let's say the signal is on the left, while the noise is on the right. In such situations, the head shadow effect causes the signal-to-noise ratio (SNR) to be up to 15 dB higher at the ear closest to the signal[5]. This helps to better understand the speaker of interest by paying attention (unconsciously) to the most favorable ear in each case, i.e., the ear that has the best SNR. The phenomenon is known as "best ear listening".

The brain, however, not only takes advantage of the information present in the more acoustically favorable ear. It also combines the sounds captured by both ears to further facilitate intelligibility. Two phenomena attest to this: the 'squelch effect' and 'binaural summation'. The squelch effect is the improvement in speech recognition when listening with two ears compared to listening only with the ear closest to the source emitting the speech of interest (the ear with the better SNR). Binaural summation is the improvement in speech recognition when both ears receive identical sounds compared to receiving the same sound in only one ear. The squelch effect can be interpreted as the brain using the ear closer to the noise source to pick up the noise and subtract it from the sound picked up by the better ear. Binaural summation suggests that the brain is able to take advantage of redundant information present in both ears[6].

4. Hearing aids

The alteration of any of the stages involved in the acoustic-neural transduction process can cause hearing loss, both by deteriorating the quality of the signal that each ear is able to transmit to the brain, and by altering the relationship between the signals transmitted by each ear, thus modifying the binaural cues available to the brain.

These alterations can have origins as diverse as middle ear stiffness (called conductive hearing loss), hair cell loss or damage (called sensorineural hearing loss), alterations in cochlear homeostasis (metabolic hearing loss), etc. The number of factors involved in hearing loss and the combinations that occur between them make this a complex problem for which there are different treatments depending on the type of loss.

Conductive losses can often be treated by medication or surgery. However, when the loss is sensorineural or metabolic (the most common types of age-related hearing loss), the usual treatment is the use of assistive listening devices: hearing aids or cochlear implants. **Figure 3** shows the usual prescription according to the degree of hearing loss: hearing aids for moderate hearing loss and cochlear implants for profound hearing loss.

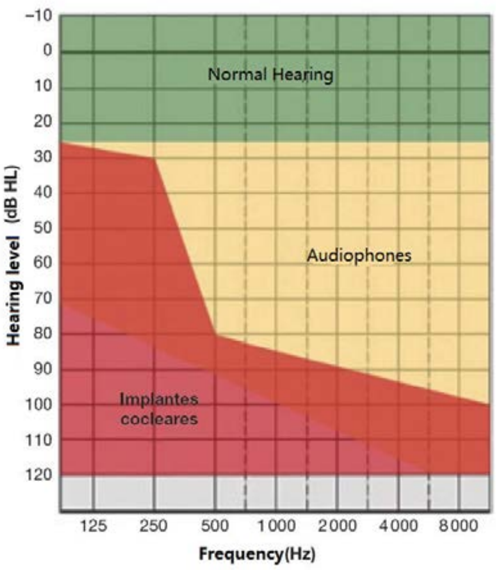

Figure 3. The audiogram represents the threshold sound level for detecting pure tones of different frequencies relative to that considered normal. The scale on the ordinate axis is inverted, so that the higher the hearing loss, the lower the thresholds are represented. In this figure the typical hearing aid prescription according to the audiogram has been overprinted.

4.1. Hearing aids

Hearing aids are the most commonly used hearing aids. They are electronic devices that amplify sound at the entrance of the ear canal. Their basic operation is as follows. A microphone picks up the sound reaching the user's ear, thus converting an acoustic signal into an analog electrical signal which is then digitized and transmitted to a digital signal processor. This processor decomposes the signal into frequency bands, and then modifies each of them according to the hearing loss of each user and the amplification strategies with which it has been configured. Finally, the processed signals are summed, and the resulting signal is transformed back into an analog electrical signal that is sent to a loudspeaker located in the ear canal.

Once the hearing aid parameters have been set, the hearing aid will amplify the sound picked up according to its intensity and frequency, amplifying sounds of lower intensity more and applying a different amplification to each frequency channel according to the user's hearing loss.

In everyday life, the sound level of the waves reaching the ears fluctuates continuously, so the amplification applied by a hearing aid varies constantly. This variation is not instantaneous, as this would lead to annoying distortions, but is applied progressively over time (dynamic gain), giving rise to different types of hearing aids according to their times of action, i.e., according to the speed with which the amplification is modified[7].

4.2. Cochlear implants

Cochlear implants began to be implanted about 60 years ago, managing to help people with profound hearing loss and even restoring the sense of hearing in completely deaf people. They work very differently from hearing aids. They do not amplify sounds, but transform them into electrical impulses that stimulate the user's auditory nerve. The cochlear implant is therefore a true artificial ear.

The first element of a cochlear implant is a microphone that picks up the sound reaching the pinna.

The captured sound is digitized and transmitted to a sound processing unit. This unit, depending on its configuration and the characteristics of the captured sound, transforms the corresponding digital signal and then sends it to a coil that is attached to the skull by a magnet. This coil communicates via radio waves with a receiver implanted in the skull, thus transmitting the information generated by the processor without any physical contact between the parts. The receiver, anchored to the bone and physically isolated from the outside of the skull, uses the data and energy received by the coil to generate appropriate electrical impulses, which are sent to the cochlea via an electrode array (**Figure 4**). The electrical impulses evoke action potentials in the user's auditory nerve and thus an auditory sensation.

The properties, placement and use of the electrode array are of great importance, as there are multiple factors that can influence the degree and quality of auditory rehabilitation, such as the number of electrodes used, the interaction between the electrodes, the correct frequency distribution to stimulate different cochlear regions, etc.

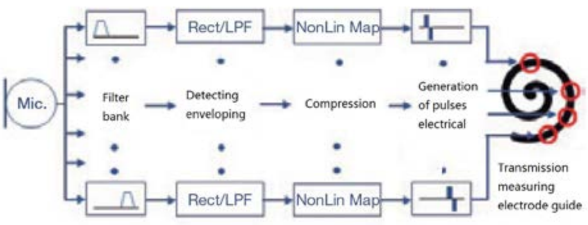

Figure 4. Diagram of the basic operation of a cochlear implant. The signal collected by the microphone is separated into frequency channels by a filter bank. The envelope of the output signal from each filter is extracted by using a wave rectifier (Rect) followed by a low pass filter (LPF). The envelope of each frequency channel is then compressed using a NonLin Map. This mapping serves to accommodate a wide dynamic range of amplitudes to a narrower range of electrical currents tolerated by the implant user. Finally, the compressed envelopes are sampled with electrical pulses that are transmitted to the corresponding electrode located in the user's cochlea. figure adapted from[8].

The function of the cochlear implant is, in short, to replace the functioning of the auditory receptor system; that is, the system upstream of the nerve neurons, which it stimulates directly, thus enabling the brain to receive an electrical signal to interpret the sound stimuli. The auditory perception will then depend on the pattern of electrical charges transmitted by the implant electrodes, the place within the cochlea where these charges are delivered and the frequency of electrical stimulation, that is, the number of pulses per second.

The use of hearing aids restores audibility of inaudible sounds. Cochlear implants, on the other hand, restore activity in the auditory nerve where it had disappeared. Both factors (audibility and neural activity) are essential for hearing. However, even the most modern hearing aids are far from perfect. One of the aspects to be improved is the fact that they often diminish their users' access to binaural cues, which is already impaired due to partial or total hearing loss[9].

5. Research avenues

Despite the great revolution brought about by the development and improvement of hearing aids and cochlear implants, auditory rehabilitation by means of prostheses is a complex process and there is still much room for improvement. The hearing sensation provided by these devices to their users is far from normal hearing. A great deal of research around the world focuses its efforts on trying to reduce these limitations by developing new technologies or refining existing ones. Some approaches are described below[10].

5.1. Amplification

Amplification is the foundation of hearing aid rehabilitation, because getting sounds to a sound level that is audible to the user is essential for, for example, understanding speech. Speech that cannot be heard cannot be understood.

Applying linear amplification, that is, increasing the sound level of soft and loud sounds equally, is not usually an appropriate solution, since hearing impaired people usually have a reduced auditory dynamic range. That is, the range of intensities from the time they can detect a sound until they find it annoying is lower than normal (**Figure 5**).

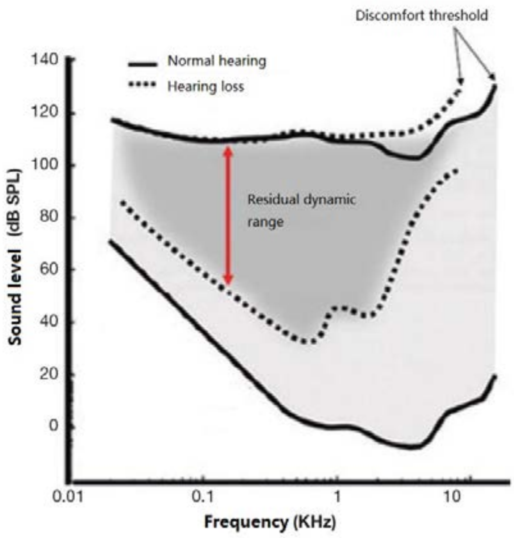

Figure 5. Comparison of the auditory dynamic range of people with hearing loss versus people with normal hearing. The lower curves show the hearing thresholds of hearing impaired (dotted line) and normal hearing people (solid line). The upper curves indicate the sound level at which sounds start to become annoying. The difference between the two levels (defined as the auditory dynamic range) is narrower in people with hearing loss than in those with normal hearing.

For this reason, most hearing aids amplify compressively above a certain sound threshold (compression threshold), i.e., they amplify low intensity sounds more and do so differently for different frequency channels (different gains for different frequency bands)[5]. Thus, WDRC (wide dynamic range compression) systems, typically implemented today in hearing aids, aim to restore the dynamic range in each frequency band, so their adjustment must be specific for each user[11].

Sometimes part of the hearing loss is caused by the death of a cochlear region, which occurs when the inner hair cells in an area of the organ of Corti are lost or rendered useless. For these cases there are frequency ‘reduction’ or ‘compression’ algorithms, since the dead regions are usually those that respond to high-frequency sounds. The strategy is to move the information that should be represented in the dead cochlear region to areas that would naturally have a higher sensitivity to lower frequency sounds[12].

In other cases, the inner hair cells have lost their functionality over too large a region to apply frequency reduction algorithms and it is more convenient to apply electroacoustic stimulation (EAS)[13]. These systems combine the operation of a hearing aid with that of a cochlear implant, applying electrical stimulation (via the implant) to represent the high-frequency components of a sound and acoustic amplification (via the hearing aid) for the low frequencies.

5.2. Listening modes

One of the options available in hearing aids and cochlear implants marketed today is the possibility of varying the hearing aid configuration. These have preset different listening modes designed for various everyday situations such as speech recognition in quiet or noisy environments, listening to music or talking on the phone, etc. For each of them appropriate values of relevant parameters such as gain, frequency response or compression are set. Depending on the device model these listening modes will be manually selected by the user or automatically activated according to the received signal.

Other more current versions of this strategy are not limited to a few listening modes, but optimize their configuration for each particular listening situation. Places frequently visited by these users are geotagged and associated with the configurations created for them, being automatically activated when returning to them[14].

5.3. Noise reduction strategies

One of the most frequent complaints from users of hearing aids and cochlear implants is the difficulty in following conversations in noisy environments, which, for example, makes it difficult for them to work in crowded places or to socialize in everyday settings such as a coffee shop.

Hearing impaired people have an impaired ability to recognize and understand speech when it is presented at the same time as other masking signals, whether these are other people speaking or other types of noise. They benefit less from elements such as the spatial separation between the signal and the mask or from the pauses present in the masking sounds than do normotensive individuals. In addition, and notably, binaural information is distorted when sound is processed through compressors or other sound processing strategies that work independently in each of the two ears.

A solution to this problem may lie in one of the fundamental parts of how hearing aids work: Processing strategies. These strategies determine how each sound processor works to decide how the received sound will be amplified (in the case of hearing aids) or how the sound is encoded into a pattern of electrical impulses (in the case of cochlear implants). Numerous strategies exist for designing such processing to improve speech recognition. Some are designed to work with a single microphone and others are designed to work with multiple microphones.

Single-microphone solutions

Single-microphone noise reduction strategies rely on detecting and exploiting differences between the acoustic characteristics of the signal and the noise that the processors are able to use to amplify the signal of interest and attenuate the noise.

When the signal and mask spectra do not coincide, SNR can be improved by filtering the captured information at different frequencies. In the case of traffic noise (which is low frequency), for example, it would be sufficient to apply a high-pass filter so that, by reducing the intensity of the information at low frequencies, the noise would be attenuated, obtaining a better SNR. Unfortunately, there are few listening situations where the noise is easily separable from the signal by means of a simple filter.

Other strategies base their operation on the detection of modulation changes. Since speech is characterized by amplitude modulations of about 4 Hz, these processors continuously analyze the spectrum of the captured signal by frequency bands. If a band is found to be dominated by typical speech modulations, it will be amplified; otherwise, little or no amplification will be applied to that channel. These strategies are quite effective against stationary noise. However, often the masking sound is also speech, so the modulation spectrum of the signal and the noise will overlap to a large extent. This is the case, for example, in a coffee shop, where the noise that makes it difficult to understand the conversation with a speaker is actually other people talking simultaneously at nearby tables. In these cases little or no benefit on SNR will be achieved through modulation analysis[15].

Another single microphone solution, present in some current devices, consists of detecting the pauses in speech and using them to estimate which part of the captured signal is of interest, subtracting the rest. Signal (and noise) estimation is performed constantly to adapt the system to acoustic changes in the environment. More recent strategies include deep *learning* methods to refine these calculations[16].

Multi-microphone solutions

There are, on the other hand, processing strategies based on the availability of information captured by more than one microphone. Some of them start from the assumption that, as usual, the signal and noise sources will be in different spatial positions and seek to create directionality patterns that amplify the signal coming from a certain direction while attenuating the rest (**Figure 6**)[17].

Two microphones are sufficient to create some of these directionality patterns. In the most basic case, one microphone will be focused toward the back hemisphere of the head, while the other will be focused toward the front. Since we normally face the person with whom we are conversing and look towards them, the signal that will be delivered to the hearing aid user will be the amplification of the result of subtracting the 'back signal' from the 'front signal', thus improving the representation of everything that emits sound in front of the listener.

These strategies, known as beamformers, reach much higher degrees of complexity, using multiple microphones to create more complex and defined directionality patterns, even connecting the information obtained in the microphones of different ears[18], or including algorithms to adapt the direction of the pattern in changing environments according to various factors such as the properties of the signal captured at each location, the direction in which the listener directs the gaze, etc.

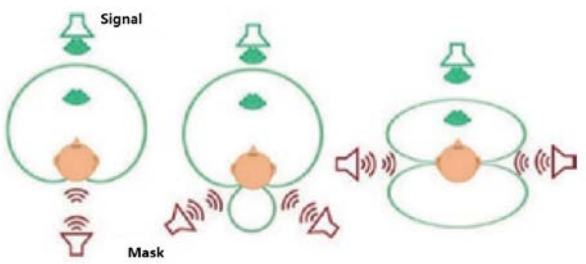

Figure 6. Some of the directionality patterns that beamformers can achieve and the spatial configurations of signal and noise for which they would be appropriate.

Binaural versus monaural solutions

Multi-microphone solutions can be applied in a single device, provided that the device has multiple microphones. In this way, speech intelligibility can be facilitated for people using a single hearing aid or cochlear implant. These are therefore monaural solutions.

The development of technology capable of exchanging information between devices wirelessly has led to the development of research avenues that seek to relate the functioning of the sound processors present in each ear. In these cases, the operation of each of the processors will depend on the information captured by all the microphones available in the hearing aids in both ears. These approaches, in which the processors in the two devices will no longer operate independently but in a linked or coupled manner, are referred to as binaural processing strategies.

In the case of hearing aids, the most basic version of this idea consists of linking the devices on both sides so that they apply the same gain (amplification); in fact, so that they apply the smaller of the gains that each of the two hearing aids would apply if they were operating independently[19]. Different studies show that this strategy, versus two hearing aids operating independently, increases speech recognition in fluctuating noise conditions. Benefits have also been found in sound localization and naturalness of the delivered sound.

But there are more applications to the possibility of coupling the operation of both processors or combining the signals captured in both ears. Two of them, devised at the Computational Hearing and Psychoacoustics Laboratory of the University of Salamanca, are described below: the MOC strategy and the contralateral cancellation algorithm.

6. The MOC strategy

The MOC strategy is one of the binaural processing strategies to improve the performance of hearing aids. Its operation is inspired by the effects of contralateral MOCR.

As described above, in the human ear, a particular area of the cochlea does not always behave in the same way. Its sensitivity varies constantly according to the activation state of the efferent fibers of the MOC bundle. This efferent bundle can be reflexively triggered by ipsilateral and contralateral sounds, acting on the outer hair cells and thus modulating the gain of the cochlear amplifier.

Activation of this reflex reduces the response of the auditory nerve to pure tones in silence, while decompressing (enhancing) its response to pure tones in the presence of background noise and thereby restoring the dynamic range of the auditory nerve fibers in noisy environments to their values in silence. That is, activation of the MOC efferents, by reducing the sensitivity of the auditory nerve, achieves approximation of the auditory ranges of the auditory nerve listening in noise and in silence (**Figure 7**)[20].

This mechanism could facilitate the neural encoding of loudness and the detection of intensity changes, as well as the neural encoding and intelligibility of speech in noisy environments[2]. The MOCR is activated at relatively low sound levels and the time required to fully activate it is 300 ms[21], so it is probably active during much of the daily listening of a person with healthy hearing.

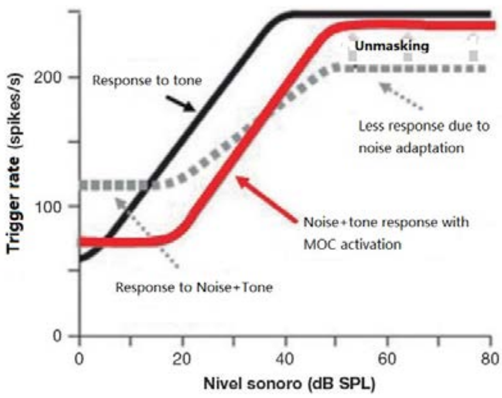

Figure 7. The different lines show the effect of MOC efferent activation on the firing rate of auditory neurons when the stimulus is a pure tone (black and continuous), when the stimulus is a pure tone immersed in noise and the MOCR is not active (gray and dashed), and when the stimulus is a pure tone in the presence of noise, but the MOCR is active (red and continuous). Note how the activation of the MOCR in the presence of noise approximates the dynamic range of the auditory neurons to the situation where the stimulus is only a pure tone. Figure adapted from Guinan[20].

People suffering from sensorineural hearing loss often suffer from a total or partial deficit of the outer hair cells involved in this mechanism, or from some kind of dysfunction of these cells, which prevents them from benefiting from the unmasking effects provided by the MOCR. These effects are not recovered with the use of current hearing aids, whose parameters, at most, are adapted to the signal captured in only one ear.

In an attempt to alleviate this deficit, the MOC strategy has been designed to reproduce the effects of the MOCR through the combined operation of the sound processors present in both ears. This strategy, intended for both hearing aids and cochlear implants, proposes to dynamically couple the compression of both ears by emulating the modulatory role that the efferent system has on the cochlear mechanics[22]. To this end, to the typical operation of hearing aids and implants, a contralateral control of compression is added that couples the operation of the two sound processors, generating dynamic control signals that are sent to the opposite processor to modulate its responses. Thus, in each ear a stimulation pattern is delivered (electrical in cochlear implants and acoustic in hearing aids), adjusted according to the sound detected by microphones present in both ears[23].

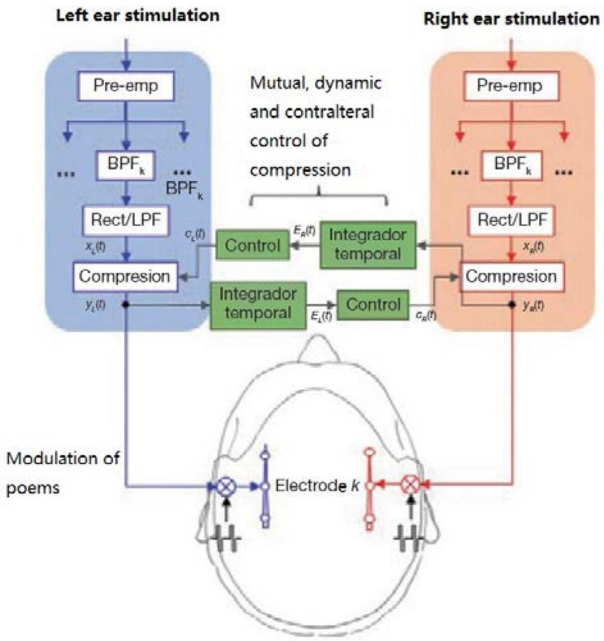

Figure 8. Diagram of the functioning of the MOC strategy in cochlear implants. In blue and orange, the typical sound processing scheme of a cochlear implant is illustrated. In green the contralateral control that each implant applies on the sound processor of the opposite ear. Figure adapted from Lopez-Poveda, et al.[23].

Figure 8 shows a schematic of the operation of the MOC strategy in cochlear implants. The compression of each frequency channel of the right processor depends on the time-weighted output level of each corresponding frequency channel in the left processor, and vice versa. In this way, the MOC strategy can decrease the compression applied to each channel, linearizing the processor response.

In addition, the information exchange between the processors in each ear required to carry out the strategy is relatively small, which is essential for the strategy to be implemented in commercial devices.

7. Experimental results

The MOC strategy has been extensively evaluated in cochlear implant users. It has been found to significantly improve speech intelligibility in noise[24] (**Figure 9**) without altering (or even slightly improving) sound localization[25], even when combined with off-the-shelf cochlear implants[26]. A version of the strategy for hearing aids is currently being developed and evaluated[27].

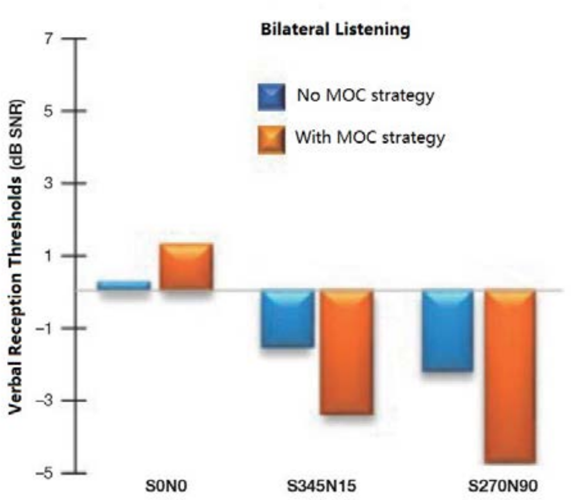

Figure 9. Speech reception thresholds (VRU) in noise listening with two cochlear implants when they are operated with the MOC strategy (orange) and when they are not (blue) in different spatial configurations of signal (S) and noise (N). In the notation SxNy, the numerical values of x and y indicate the azimuthal positions of the signal and noise, respectively. The URV is the SNR at which the listener recognizes 50% of the sentences presented to him. Therefore, the smaller URV values obtained with the MOC strategy indicate that with this strategy, the listener tolerates higher noise levels while maintaining the same intelligibility. Figure adapted from[28].

Hearing aids whose processors operate according to the MOC strategy manage to improve the signal-to-noise ratio with which the stimulus is delivered to the user, in addition to enhancing the interaural differences in intensity and improving the spectral contrast and amplitude modulations in each frequency channel. Experimental results show that this improves intelligibility in noise, restoring, to some extent, the spatial unmasking and localization of sound sources[28].

7. Contralateral cancellation

Another idea developed by the Computational Hearing and Psychoacoustics Laboratory of the University of Salamanca has as its starting point the fact that, in real listening situations, the sources that generate the signal and the noise are usually located on different sides of the speaker. As explained above, in these situations, there will often be a better ear: the one that is closer to the signal. The contralateral cancellation strategy proposes to increase the SNR in the better ear by attenuating sound sources located in the opposite hemisphere (**Figure 10**)[29].

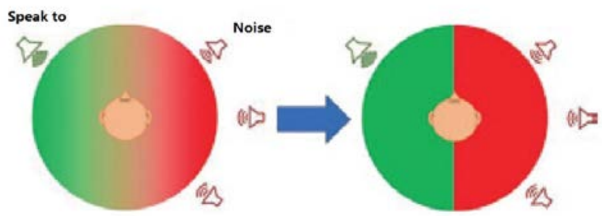

Figure 10. Schematization of the purpose of the contralateral field cancellation algorithm. The illustration on the left shows the situation prior to the application of the algorithm: signals with content of what is emitted from the left hemisphere (green) and from the right hemisphere (red) arrive at each ear. After applying the algorithm (right illustration) the signal arriving at each ear has less representation of what is emitted from the opposite hemisphere, since the contralateral signal has been attenuated.

To achieve this, the proposed algorithm subtracts from the sound picked up in each ear the sound picked up in the opposite ear weighted appropriately. The weighting is determined by parameters calculated from head-related transfer functions (HRTF), functions that establish the relationship between the location of a sound source and the spectrum with which the sound reaches each of the eardrums. Thus, thanks to the proper selection of these parameters it is possible to obtain different directionality patterns, which will make the algorithm more or less useful depending on the location of the noise sources.

The efficiency of the algorithm has been explored, both technically and experimentally. Experimental results have shown that the algorithm, operating independently on normotensive individuals without hearing aids, improves intelligibility in noise significantly at the expected locations according to the technical analysis (**Figure 11**) without deteriorating spatial localization[29]. These evaluations are currently being extended to the performance of the algorithm in combination with hearing aids in hearing impaired individuals.

Regarding the technical evaluation, interesting results have been obtained because of the similarity of the benefits provided by the algorithm with the unmasking achieved by listening binaurally when there is no hearing loss (**Figure 12**)[29]. This suggests that the brain could process the sounds it receives through both ears in a similar way to how the algorithm does and, therefore, that we could use this algorithm, combined with hearing aids or cochlear implants, to try to recover the unmasking benefits existing in healthy binaural hearing.

One of the strengths of this algorithm lies in its ability to improve SNR and, therefore, speech intelligibility in noisy environments, without the need for prior information about the type of noise, its spectrum, or the location of the source(s) generating it, beyond, of course, assuming that they are on the opposite side of the signal source and the ear closest to it. This property, together with the fact that it can be implemented immediately downstream of the prosthesis microphone, gives this algorithm greater applicability compared to other noise reduction strategies. In addition, the possibility of varying the weightings using HRTFs provides great versatility in deciding which parts of the acoustic field will be amplified and which will be attenuated.

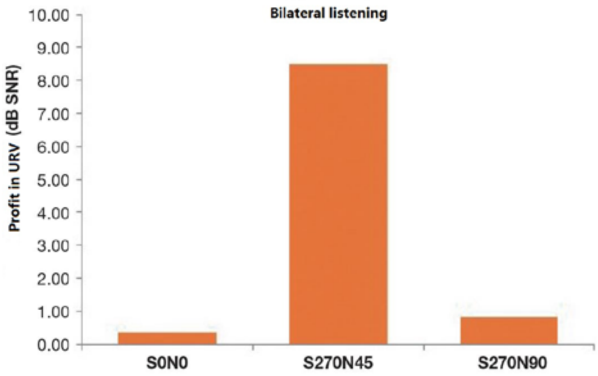

Figure 11. Average benefit in verbal reception thresholds (URV) in noise of normoyent people, in bilateral listening and for different spatial locations of signal and mask. In the notation SxNy, the numerical values of x and y indicate the azimuthal positions of signal and noise, respectively. The spatial positions chosen are representative of those where, according to simulations, the benefit should be zero (S0N0), large (S270N45), and medium/low (S270N90).

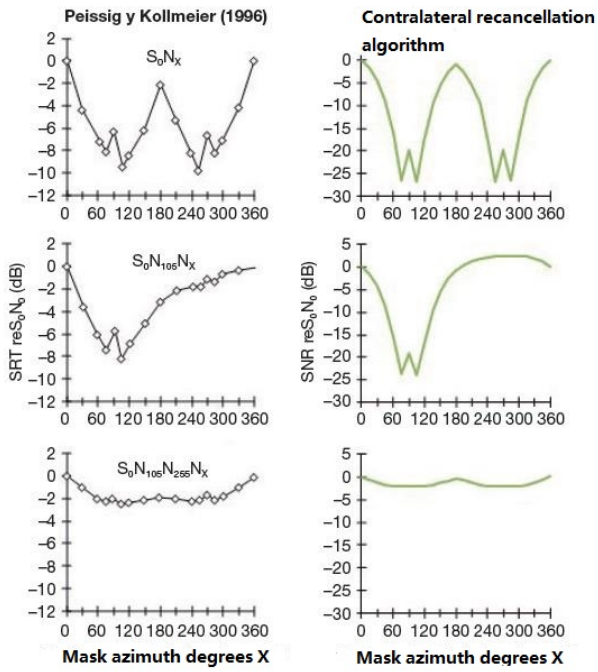

Figure 12. Comparison of Peissig and Kollmeier's experimental results (left) on the spatial unmasking characteristic of healthy hearing in binaural listening with the variation in SNR produced by the contralateral cancellation algorithm. The x and y values in the SxNy notation indicate the location of the signal and mask in the azimuthal plane. See Lopez-Poveda et al.[29] for details. Figure adapted from Lopez-Poveda et al.[29].

8. Conclusions

Humans, thanks to having two ears, have access to binaural cues that our brain uses to locate sound sources and understand speech in noisy environments. People with hearing loss often see the availability of these cues impaired and the use of hearing aids, far from restoring them, deteriorates them even more, accentuating the problem.

Much research is being done on hearing aids and cochlear implants to provide better binaural hearing. The approaches are diverse. In the Computational Hearing and Psychoacoustics Laboratory of the University of Salamanca, strategies inspired by the functioning of the healthy ear are being developed. The two strategies described in this paper (the MOC strategy and the contralateral cancellation algorithm) are promising, as they can improve speech intelligibility in noisy environments without significantly altering the localization of sound sources.

However, much remains to be done to achieve normal binaural hearing for people with hearing loss.

Conflict of interest

The authors declare no conflict of interest.

References

1. Diedesch A, Gallun F, Stecker GC. Effects of clinical hearing aid settings on sound localization cues. The Journal of the Acoustical Society of America 2017; 141: 3638–3638.
2. Lopez-Poveda EA. Olivocochlear efferents in animals and humans: From anatomy to clinical relevance. Frontiers in Neurology 2018; 9: 197.
3. Avan P, Giraudet F, Büki B. Importance of binaural hearing. Audiol Neurotol 2015; 20(1): 3–6.
4. Lopez-Poveda EA, Meddis R. A physical model of sound diffraction and reflections in the human concha. Journal of the Acoustical Society Amrica 1996; 100(5): 3248–3259.
5. Dubno JR, Ahlstrom JB, Horwitz AR. Binaural advantage for younger and older adults with normal hearing. Journal of Speech, Language and Hearing Research 2008; 51(2): 539–556.
6. Marrufo-Pérez MI, Araquistain-Serrat L, EustaquioMartín A, et al. On the importance of interaural noise coherence and the medial olivocochlear reflex for binaural unmasking in freefield listening. Hearing Research 2021; 405: 108246.
7. Kates JM. Principles of digital dynamicrange compression. Trends in Amplification 2005; 9(2): 45–76.
8. Wilson B, Finley C, Lawson D, et al. Better speech recognition with cochlear implants. Nature 1991; 352: 236–238.
9. Denk F, Ewert SD, Kollmeier B. On the limitations of sound localization with hearing devices. The Journal of the Acoustical Society of America 2019; 146: 1732–1744.
10. Kollmeier B, Kiessling J. Functionality of hearing aids: State-of-the-art and future model-based solutions. International Journal of Audiology 2018; 57(3): S3–S28.
11. McCreery RW, Venediktov RA, Coleman JJ, et al. An evidence-based systematic review of amplitude compression in hearing aids for school-age children with hearing loss. American Journal of Audiology 2012; 21: 269–294.
12. Kluk K, Moore BC. Dead regions in the cochlea and enhancement of frequency discrimination: Effects of audiogram slope, unilateral versus bilateral loss, and hearing-aid use. Hearing Research 2006; 222: 1–15.
13. Von Ilberg CA, Baumann U, Kiefer J, et al. Electric-acoustic stimulation of the auditory system: A review of the first decade. Audiology and Neurotology 2011; 16: 1–30.
14. Dittberner AB, Jakobsen GH, Vries A (inventors). A location learning hearing aid. European Patent. EP2884766A1. 2015 Jun 17.
15. McCreery RW, Venediktov RA, Coleman JJ, et al. An evidence-based systematic review of directional microphones and digital noise reduction hearing aids in school-age children with hearing loss. American Journal of Audiology 2012; 21: 295–312.
16. Wang D. Deep learning reinvents the hearing aid. 2017. IEEE Spectrum 2017; 54(3): 32–37.
17. Beck DL, Schum DJ. Directional hearing aids: Concepts and overview (2005). The Hear Journal 2006; 59: 40–47.
18. Kreikemeier S, Margolf-Hackl S, Raether J, et al. Comparison of different directional microphone technologies for hearing aid users with moderate to severe hearing loss. The Hearing Review 2013; 20: 44–45.
19. Wiggins IM, Seeber BU. Linking dynamic-range compression across the ears can improve speech intelligibility in spatially eparated noise. Journal of the Acoustical Society of America 2013; 133(2): 1004–1016.
20. Guinan Jr JJ. Olivocochlear efferents: Anatomy, physiology, function, and the measurement of efferent effects in humans. Ear and Hearing 2006; 27(6): 589–607.
21. Backus BC, Guinan Jr JJ. Time-course of the human medial olivocochlear reflex. Journal of the Acoustical Society of America 2006; 119(5): 2889–2904.
22. Lopez-Poveda EA (inventor). Sound enhancement for cochlear implants. EU patent. WO2015169649A1. 2015 Nov 12.
23. Lopez-Poveda EA, Eustaquio-Martin A, Stohl JS, et

al. A binaural cochlear implant sound coding strategy inspired by the contralateral medial olivocochlear reflex. Ear and Hearing 2016b; 37(3): e138.

24. Lopez-Poveda EA, Eustaquio-Martín A. Objective speech transmission improvements with a binaural cochlear implant sound-coding strategy inspired by the contralateral medial olivocochlear reflex. Journal of the Acoustical Society of America 2018; 143(4): 2217–2231.
25. Lopez-Poveda EA, Eustaquio-Martín A, Fumero MJ. Lateralization of virtual sound sources with a binaural cochlear-implant sound coding strategy inspired by the medial olivocochlear reflex. Hearing Research 2019; 409: 108–320.
26. Fumero MJ. Multifaceted evaluation of a binaural cochlear-implant sound processing strategy inspired by the medial olivocochlear reflex [PhD thesis]. Salamanca: Universidad de Salamanca; 2020.
27. Hernandez-Martin A. Un audífono binaural inspirado en el reflejo olivococlear medial contralateral (Spanish) [A binaural hearing aid inspired by the contralateral medial medial olivocochlear reflex] [PhD thesis]. Salamanca: Universidad de Salamanca; 2020.
28. Lopez-Poveda EA, Johannesen PT, Edwards B, et al. Predictors of hearing-aid outcomes. The Hearing Journal 2017; 70(7): 12–13.
29. Lopez-Poveda EA, Eustaquio-Martín A, San-Victoriano FM. Binaural (pre)processing for contralateral sound field attenuation and improved speechin-noise recognition. BioRxiv 2021; 1(22): 427–757.