University of Pécs

Faculty of Health Sciences

Doctoral School of Health Sciences

Head of the Doctoral School:

Prof. Dr. József Bódis MD, Ph.D., DSc.

Programme leader:

Prof. Dr. Gábor L. Kovács MD, Ph.D., DSc.

Supervisors:

Prof. Dr. József Pytel MD, Ph.D.

Dr. László Lujber MD, Ph.D.

Nonlinear distortion of the ear

Doctoral (Ph.D.) thesis booklet

Viktor Bagdán

Pécs, 2021

Introduction - Noise-induced hearing loss (NIHL)

Today, hearing loss caused by high Sound Pressure Levels (SPL) is a growing problem. Extreme loud noises and sounds can lead to noise damage. Noise is an unwanted sound and can cause hearing damage. In our modern society, millions of people in Europe are exposed to the harmful effects of noise. Workers in certain sectors, such as construction or the service and entertainment industries, are more exposed to noise, but lower sound pressure levels can also cause problems if they persist for a longer period of time. In workplaces, noise can have a negative impact on concentration because of its disturbing, stress-inducing effects, but it also increases the risk of accidents.

"Noise-induced hearing loss is a perceptual type of hearing loss that increases towards high frequencies, which can be shown to be caused by exposure to noise at work (sound trauma), and no other fatal disease has been implicated in the development of hearing loss. It may be acute or chronic." (Department of Health Professional Protocol).

The sensory cells in the inner ear are very sensitive. Unfortunately, once damaged, the hair cells cannot be replaced.

Noise-Induced Hearing Loss (NIHL) can be Temporary Threshold Shift (TTS) or Permanent Threshold Shift (PTS) and can affect one or both ears. In such cases, speech intelligibility is impaired in noisy environments or over a telephone line. NIHL may be immediately noticeable or may be proven later. By definition, the Absolute Threshold of Hearing (ATH) is the minimum sound pressure level associated with pure tone that the average human hearing can just detect, with normal hearing and without the presence of other sound. Thus, the threshold of hearing is the value associated with the sound that is first discriminated and heard by the organism (Durrant and Lovrinic, 1984). An increase in the hearing threshold in young people has been statistically proven (Doheny, 2010). The underlying causes are very often due to the excessive noise in our living environment, which is different from the one we originally created. Portable media players are becoming increasingly popular among younger generations, most of which are fitted with ear-canal type headphones (Révész, Gerlinger, 2011). These electronic devices use class D sound amplifier integrated circuits that operate at extremely high efficiency (>90%), thus requiring little battery power. They are capable of producing high sound pressure levels over long periods of time, and therefore carry a high risk of hearing damage. The equivalent sound pressure level for permanent hearing loss over 8 hours is 85 decibels, measured with an 'A-filter' (Widen et al., 2017; Lutman, 2000). These efficient portable media players can reach and even exceed this level (Fligor and Clarke, 2005). This threat can also arise at events and various types of sound reinforcement. Levels above 85dB(A) can be tolerated by the human ear for short periods, but not for 8 hours. Another problem is increased noise levels and noise pollution in our increasingly noisy and crowded environment. Increased noise levels are not just a problem in themselves. A certain signal-to-noise ratio is necessary for speech intelligibility or music enjoyment (Meyer, Dentel, and Meunier, 2013). Therefore, if the noise level increases, the signal level must also increase, which can also lead to noise annoyance. Hearing fatigue is the phenomenon where prolonged exposure to noise reduces ear sensitivity. This temporary increase in hearing threshold (hearing loss) can be confirmed by audiological measurements. This can be detected by the person concerned and after a period of rest the hearing threshold returns to its previous normal level. The rest period depends on the level of noise exposure and can be hours or even days. If the rest period is not sufficient and the ear is exposed to repeated noise exposure, a metabolic deficit, fatigue, occurs, leading to the destruction of hair cells in the inner ear (Kryter, 1994). For this reason, sound pressure levels should not be measured instantaneously but integrated over a longer period of time and the equivalent sound pressure level over 8 hours should be taken into account when assessing the extent of exposure. The destruction of hair cells in the inner ear is manifested as permanent hearing loss. Repeated exposure to noise exacerbates the symptoms. However, regardless of noise exposure, age-related hearing loss can be observed (Bielefeld, 2012), but ageing does not necessarily cause hearing loss. Hearing loss with ageing mainly affects higher frequencies, and the hair cells in the inner ear are affected. Calcification of the middle ear, on the other hand, causes hearing loss due to a lack of amplification of low frequencies.

Problem statement, objectives

There are several ways to prevent hearing loss caused by noise. In addition to keeping noise levels low, it is also very important to keep the level of the information-carrying sound that is meaningful to us low. This sound can be electronically amplified speech, sound transmitted by telecommunications equipment or even music. Electronic sound amplification equipment and various sound effect modules are therefore of particular importance, since they are almost always used to increase the volume. Based on the subjective opinion of our research team, we considered it possible to create a sound amplification mechanism that, due to its inherent special nonlinearity, would result in a stronger perceived volume than a purely linear characteristic low-distortion sound amplification device. This conclusion was reached from preliminary observations that different sound amplification devices produce different degrees of timbre and loudness perception, despite the identity of the electronically measured sound pressure level. This observation, empirically, is almost common knowledge among musicians, who prefer certain types of instruments (Hadi T. et al., 2015) or electronic sound amplification equipment over others. An example is the renaissance of electron tube sound amplification equipment. The total harmonic distortion + noise (TDH+Noise) values are higher, the frequency response of the amplifier is much worse, yet the tone and the loudness of the sound are better than that of semiconductor amplifiers. With our research, we wanted to model and measure this effect, to prove or disprove the idea. None of the solutions we have identified to prevent hearing damage work on the principle of the difference between the instruments and sound amplification equipment outlined above. Although they reduce hearing damage, they interfere with the quality of the musical material, modifying its dynamic range or frequency response. We have not found any solution that achieves a perceived increase in volume by targeted manipulation of the harmonic range. According to Okabe and Nakatoh's 2018 study, the extent of hearing damage caused by headphone music listening can be prevented by applying different degrees of volume control at different stages of the musical material (introduction, chorus, solo, ending) by mapping the musical structure. Another solution is to apply non-linear amplification with dynamic compression. This is also widely used in hearing aids. Dynamic compression can be beneficial if you want to increase the volume in certain frequency bands, for example because hearing loss can be demonstrated in certain frequency bands. A 2018 study by Zou, Hao and Panahi points out the shortcomings of these solutions, as multi-channel dynamic compression introduces distortion into the system and increases processing complexity in digital systems. To reduce this, a compensating filter is designed to reduce the distortion.

Summary of theses

I. Non-linear amplification causes a loudness increase;

II. Is it possible to create a method or device that can maintain the perceived loudness level at a higher value without changing the measured sound pressure level;

III. It is feasible to increase the perceived loudness without changing the fundamental harmonic;

IV. the increase in perceived loudness does not cause audible distortion;

V. The increase in perceived loudness is not dynamic compression;

VI. The new method or device created can be implemented in digital devices;

VII. The method can prevent hearing loss caused by noise.

Preliminary research

Investigation of the perception of distorted sounds

Several studies have been carried out to verify the effects of the harmonic range and the corresponding distortion on loudness. In the first study, we demonstrated that distorted sounds have a higher loudness perception compared to undistorted sounds. To verify this, we played a melody on electric guitar and recorded it digitally. After normalisation (bringing the sound to 0 dB), a second recording was made in which a special distortion was applied to the same melody, using a guitar distortion device specially made for electric guitars. The distorted recording was also normalised using software, so that the distorted and undistorted recordings were identical in terms of maximum amplitude. When the two recordings were played back, the distorted recording clearly produced a higher perception of loudness (Bagdán, 2013, Science In Practice, Schweinfurt). However, when significant distortion is applied to the distorted sound material, there is a clear loss of information, the melodic character is preserved, but the distortion reduces the speech intelligibility or the recognisability of the instrumental character.

Experiments with special distortions

Despite the considerable distortion and the loss of information that this entails, in very many cases it is the emergence of specific sound images, the creation of a new sound image or timbre that has caused the popularity or the pleasant colouration of a particular type of distortion that is acceptable to human hearing. An example of this is a sound recording created in 1960 in Nashville, Tennessee, during an unexpected event. During the recording of Grady Martin's guitar playing, the 6-channel mixing console malfunctioned, presumably due to a primary coil of the coupling transformer breaking (Thanki and DeGennaro, 2019). The recording was nevertheless released and went on to great popularity (number one on the national Billboard charts in 1961). This special distortion, called fuzztone, was later deliberately created, and the Gibson company patented a process using three Germanium transistors (US3213181A, 1965). The patent description reveals that one of the purposes of the distortion is to allow a sound source to imitate the feel of another instrument, with special settings. In the case of a stringed instrument input, for example, it can produce the sensation of a trumpet, trombone or tuba, with the appropriate potentiometer settings, according to the patent description. So we are talking about a significant amount of distortion and a significant intrusion into the harmonic range. Once the circuit had been built, it was possible to test it and analyse the harmonic content it generated. The next step was to test the circuit in the harmonic domain, using a circuit simulation program (TINA, Toolkit for Interactive Network Analysis). In the first two steps, a musical "a" tone was given as input, corresponding to a fundamental frequency of 440 Hz. Since the sound was coming from a guitar, the input signal already contained overtones in addition to the fundamental frequency. The distorted output signal contained even and odd-numbered harmonics, with a slight excess towards the even-numbered harmonics, based on the measurement points. We then investigated the effects of a more complex input, a triad. The input signal was a major triad chord (A major chord), consisting of a major third and a pure fifth in addition to the root. As in the previous experiment, the frequency of the fundamental is 440 Hz, the major third 550 Hz and the pure fifth 660 Hz. The frequency of the pure fifth has a ratio of 3/2 to the fundamental, and the frequency of the pure acoustic major third has a ratio of 5/4, i.e. the frequency ratio of the three tones is 4/5/6. The 4/5/6 frequency ratio in the example is pleasing to the human ear, and this ratio is probably also true for the overtones. The dominance of an even harmonic is more pleasing than the odd harmonic because of its octave character. In this case, both types appear, and the corresponding proportions are assumed to be perceived as acceptable and natural to the human ear. The estimation of the appropriate ratio values necessitated an analysis of the harmonic range of different types of sound amplification equipment.

Analysis of the harmonic range of amplifiers

In today's world, there are different options when it comes to amplifying sound. In the early days of electronically powered sound amplifiers, only vacuum tube, or electron tube, technology was available, but now there are several types of semiconductor technology. The different arrangements have their advantages and disadvantages and, due to environmental concerns, the focus is now on low power consumption and efficient operation. The aspect of energy-efficient operation becomes particularly important when one considers that practically every household contains a variety of sound amplification equipment and consumer electronics. The previously mentioned class D amplifiers solve this problem, but they have other undesirable characteristics that make them unlikely or rarely used in high quality systems. To map the appropriate distortion characteristics, I have chosen sound amplification equipment that has been used and sought after by musicians worldwide for decades. The wiring diagrams and assembly instructions for these well-established pieces of equipment were available, so it was possible to construct the missing amplification equipment. The aim of the study was to identify the differences in the distortion characteristics of these sound amplification devices, grouped according to their principle of operation, and thus to provide a possible guideline for defining the distortion characteristics that are acceptable or particularly pleasing to the human ear. The ultimate goal is to create a device that produces a higher perceived loudness while keeping the measured sound pressure level low. As a result, the energy used is also lower, since the electronically measured power is also kept at a lower level. In addition to the sound amplification equipment, the test required a special calibrated microphone, a measuring instrument capable of performing a real-time Fourier transform, a signal generator, and a calibrated room where these tests could be carried out. Nine sound amplification devices operating on different principles were tested. Of these, five use semiconductor technology and four use electron tube technology. The measurements were carried out in a special measuring booth, where the amplifiers and the measuring microphone were located in a calibrated measuring room, with the tester acoustically separated from the room. This separation was also justified by the need to measure up to the power limit of each amplifier, which would have resulted in excessive sound pressure levels, thus risking the subject's hearing. In the acoustically isolated room, a calibrated hand-held meter, connected by a cable, capable of, among other things, performing a real-time Fourier transform, was used. A calibrated condenser measurement microphone was connected to the instrument by cable. The input signal in each case was a sinusoidal waveform of three different frequencies. A low, a mid, and a higher frequency. A signal generator was connected by cable to the input of the amplifiers and provided the input signal in each case. The output signal came from the calibrated microphone. The condenser microphone was located one metre away from the loudspeaker and mounted on a microphone stand. The output microphone signal was connected to the handheld FFT instrument via a special calibrated cable, but in parallel the amplifier speaker signal was electronically coupled to an oscilloscope for peak-to-peak measurement of the output signal voltage. Measurements were started at 10 mW power and increased in steps up to the power limit indicated on the data sheet of each amplifier. Each measurement point was recorded and the results were plotted as a graph with the output power on the horizontal axis and decibel levels on the vertical axis. The displayed curve series are the second, third, fourth and fifth harmonics of the respective input signal, indicated in different colours. For all nine amplifiers, the ensembles of characteristics were constructed from the data measured at three different input frequencies (40 Hz, 1 kHz and 4 kHz). So 27 characteristics similar to the previous one were created. The triode (cathode ray tube) amplifiers amplify the second harmonic better than the third, as measured. In the results of the cathode ray tube amplifiers, the second harmonic is nearly straight (logarithmic on the X axis) and sits above the third harmonic line up to the power limit, where the distortion increases sharply. Thus, for the cathode ray tube group, the k2 line is nearly straight across the entire spectrum from the lowest measured power to the power limit, but k3 has a breakpoint, and when the amplifier power reaches it, the k3 distortion starts to increase. For semiconductor bipolar transistor amplifiers, however, this k3 breakpoint is absent. From all the data processed, the k2/k3 ratio is an interesting value that can provide information about the quality of the equipment and its corresponding distortion characteristics. Experience has shown that electron tube amplifiers with low distortion and a balanced sound image are popular among musicians and those interested in quality music listening. The harmonic distortion character of these amplifiers has been measured to be different from that of semiconductor equipment.

Patent: Device for modelling human ear distortion and method for sound signal processing - Our own sound signal processing method

The device and process developed by our research group has been granted patent protection (Bagdán, Czimerman, Máthé, Pytel, 2014). The following figure shows the functional block diagram of the procedure.

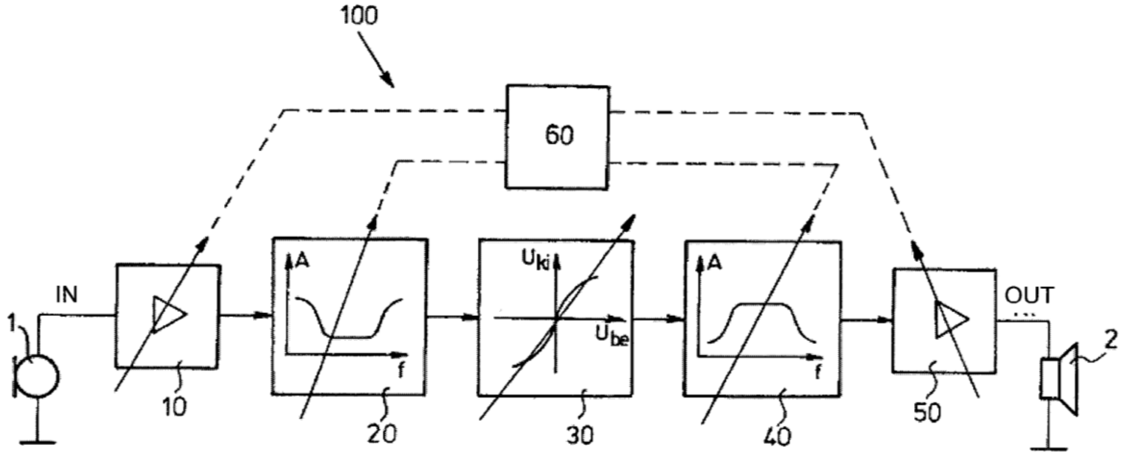

Figure 1: *Block diagram of the patented process*

The patented method is an effective imitation of the distortion and non-linear behaviour of the ear. Any nonlinear distortion causes an increase in loudness, but the modified sound becomes lifelike when we approximate the distortion of human ear as closely as possible. The middle of the frequency spectrum of a human voice or song has the lowest threshold of hearing and is also where the human ear is most sensitive to distortion. This range is modelled with the linear distortion module "20". The non-linear distortion module "30" is used to enrich the sound with even and odd harmonics of the right proportion. Finally, the inverse linear distortion module "40" is used to restore the original tone. The "10" and "50" modules are used to set the optimal signal level.

Prototype of the device for modelling human ear distortion

The prototype was built according to the principles in the patent. The device used for our research is a device that leaves the frequency spectrum of the incoming sound between 20Hz and 50kHz unchanged, but modifies the overtones. The invention is a device for processing the sound signal that provides a lifelike sound experience even at lower sound pressure levels (Bagdán, 2014). Our research concluded that by adjusting or 'mimicking' the harmonic range similar to the distortion of the human ear over the entire audible frequency spectrum, loudness perception can be increased without compromising sound quality. If a device for processing a sound signal could, as a combination of the various electronic units used in it, artificially create such a harmonic range, its use in sound amplification equipment would avoid the use of unnecessarily high and harmful sound pressure levels. The Proof Of Concept Prototype of our method was designed to allow subjective comparative loudness tests. The prototype is a hybrid circuit that includes both electron tubes and operational amplifiers. A switch allows you to choose whether you want to hear a modified or unmodified sound in the harmonics. In the modified branch the input and output have the same amplitude, and in the unmodified branch the gain can be varied by a potentiometer. This makes it possible to adjust the potentiometer to the level corresponding to the volume perceived during the subjective test by toggling the switch during the headphone test. After the test has been carried out, the potentiometer position can be used to infer the increase in loudness perception and this can be expressed in decibels.

Verification of concept - Subjective comparative loudness perception test

High-resolution, compression- and compression-free music with low distortion was used for the test. The test was conducted in a silent chamber, in which the test administrator and the volunteer were seated, and the prototype was placed. The test was conducted using headphones, but to control the acoustic sound pressure level, the headphone signal was also fed to active speakers in parallel to control the sound pressure level. The sound pressure level was measured outside the chamber by a third person. The measuring microphone measured the sound pressure level produced by the active loudspeakers. By measuring the equivalent sound pressure level (logarithmically averaged level), we ensured that the sound pressure levels of the unmodified and modified sounds were the same. Using this method, all positions of the potentiometer on the front panel of the prototype were scaled in decibel levels. The decibel level increment associated with the perceived loudness increment was determined by the decibel level of the distortion-free channel perceived to be of equal loudness. The decibel level of each potentiometer position associated with the gain of the distortion-free channel was verified using a certified sound pressure level meter. Data were collected individually using a questionnaire and the responses were arithmetically averaged to give the result in decibels. Questions were also asked about whether the participant had a diagnosed hearing loss or had been in a noisy environment before the test. The device was tested by 66 participants and students, all of whom successfully completed the questionnaire. The test and the questionnaire were designed according to the criteria of objective audiometry (Pytel, Audiology, 1996). Mean age of test takers: 33.1 years, standard deviation: 12.1, median: 32 years, minimum: 14 years, maximum: 68 years. 36% of those who completed the test were female, 64% male. Participation in the study was voluntary and was conditional on being free of hearing complaints.

Results and conclusions

On average, an increase of 2.73 dB can be measured electronically (standard deviation: 2.19, minimum: -1.5dB, maximum: 6.5dB, median: 2.5dB), based on the 66 completed questionnaires used with the triode prototype. (Bagdan, 2020.) This represents an increase of 87.5% in terms of power change. The power increment can be calculated as follows. If the input power is taken as 1 W, then:

$P_0 = 1\ \text{W},$ ....(1)

$L_p = 2{,}73\ \text{dB},$ ....(2)

$P_1 = P_0 * 10^{\frac{L_p}{10}}\ \text{W} = 1\ \text{W} * 10^{0{,}273} = 1{,}875\ \text{W},$ ....(3)

$\frac{P_1}{P_0} = \frac{1{,}875\ \text{W}}{1\ \text{W}} = 1{,}875.$ ....(4)

This represents a power increment of 87.5%. (An increment of 3dB is defined as exactly 100% power increment, i.e. double the power.) This means that, on average, the perceived loudness is that much higher when the overtones are appropriately modified, compared to unmodified, without any change in the measured sound pressure level. The measured increase of 2.73 decibels is above the just noticeable difference in amplitude (JND). In 1977, Jesteadt et al. defined amplitude selectivity, i.e. the just noticeable difference, as 0.5 decibels measured at a sound pressure level of 80 decibels. Based on this, our measurements demonstrated the operability of our patented method.

A new approach: harmonic enrichment with a multiplier circuit

A more recent approach is to create the desired overtones directly using special multiplier circuits. This means that the amplitudes of the harmonics can be controlled individually or in combination. Previous tests with tube/valve amplifiers have shown that the second, third and fourth harmonics have a significant effect on the sound image, while the higher harmonics have less effect. The following equations show the mathematical production of harmonics (Gardánfalvi, 2016):

$$f(t) = \sin(\omega t), \quad (5)$$

$$f^2(t) = \frac{1}{2} - \frac{\cos(2\omega t)}{2}, \quad (6)$$

$$f^3(t) = \frac{3\sin(\omega t)}{4} - \frac{\sin(3\omega t)}{4}, \quad (7)$$

$$f^4(t) = -\frac{\cos(2\omega t)}{2} + \frac{\cos(4\omega t)}{8} + \frac{3}{8}, \quad (8)$$

$$f^5(t) = \frac{10\sin(\omega t)}{16} - \frac{5\sin(3\omega t)}{16} + \frac{\sin(5\omega t)}{16}, \quad (9)$$

$$f^6(t) = -\frac{15\cos(2\omega t)}{32} + \frac{6\cos(4\omega t)}{32} - \frac{\cos(6\omega t)}{32} + \frac{10}{32}. \quad (10)$$

The multiplier function was implemented using an Analog Devices AD633 integrated circuit, while the amplification was done using a Texas Instruments NE5534 operational amplifier. The harmonic enrichment arrangement thus allows for artificial intervention in the harmonic range, and thus gives the possibility to create the desired sense of loudness. The circuit is still under further testing.

Ultrasonic noise and its health implications

Most of our studies so far have been in the overtone range. The frequencies of these sounds are often in the lower ultrasonic range and higher than 20 kHz. We can talk about useful ultrasound sounds, which also carry information, or about noises whose negative health effects have now been demonstrated. According to Wegel's research in 1932, the audible frequency range is between 16 Hz and 24 kHz. However, Rosen and Stuart's 2011 results suggest a generally accepted range of 20 Hz to 20 kHz. The A-weighting curve commonly used in noise measurements, however, takes into account higher frequency sounds and noises with a continuously decreasing weighting, thus somewhat following the non-linearity of the human ear on the frequency scale. In our study, Dieroff and Ertel also described the detection of 120 kHz sine wave sound. There are two common theories about the detection of ultrasonic sounds. The first is that the inner hair cells on the basement membrane of the cochlea detect the sounds (Nishimura, 2003), the second is that the brain is vibrated by this high frequency sound and modulates it down to an audible frequency spectrum which can then be detected by the cochlea (Lenhardt, 2003). In their study, they divided music below 22 kHz (LFC) and above 22 kHz (HFC) into parts, rich in high frequency components. HFC alone was not detectable, but when LFC and HFC were present together, they found a statistical increase in alpha electroencephalography (alpha-EEG) signals compared to measurements of LFC-only material. Positron emission tomography (PET) measurements of brain function showed that when both LFC and HFC were present, brain activation and regional cerebral blood flow (rCBF) were detected in the left thalamus. As early as 1954, Deatherage and colleagues showed that speech sound modulated by ultrasound carrier frequency can be used to detect intelligible speech with a high degree of accuracy, especially in noisy backgrounds. Lenhardt and his research team published their results in Science in 1991, showing that a signal at 108 kHz was still detectable. Currently, several countries have legislation limiting the permissible level of ultrasonic noise. As technical progress has led to the introduction of ultrasound pollution in many areas, some countries have introduced restrictions. Ultrasound is used in underwater positioning, in industry (from 20 kHz) or in medical diagnostics (up to 10 MHz). It also includes a variety of home devices, including, but not limited to, burglar alarms, dog whistles, bird and rodent alarms, humidifiers, inhalers, or car-mounted game alarms. As early as the 1940s-1950s, the harmful effects of ultrasound in industrial areas were reported, with symptoms including hearing damage, thermal effects, subjective symptoms and functional loss. (Davis, 1948.) In 1986, Grzesik and Pluta reported NIHL at 13-17 kHz in workers who had been working in ultrasonic washing or welding for many years. A study by Smagowska and her research team published in 2013 describes several negative health effects that call for attention to the harmful effects of ultrasonic range noise and for a review of the legal limits. Such effects include noise from dental drills (25 - 42 kHz), which caused mild hearing loss at 3 kHz, and adverse effects from other ultrasound sources such as excessive fatigue, nausea, agitation, headaches, discomfort, irritation, nervous irritability, memory problems and difficulties in concentration and learning. More severe effects have been detected at higher sound pressure levels, at 21 kHz, 110 decibels, after 3 hours of exposure per day for 10-15 days, causing functional changes in the cardiovascular and central nervous systems (Il'nitskaia, 1973). Nowadays, various DC-DC converters, which operate with high-frequency switching signals with very high efficiency, are widely used, especially in portable devices. The GTEM cell at PTE-MIK Faculty and the attached high-frequency spectrum analyzer have been used to verify the existence of these interference signals (Kvasznicza, 2021). Thus, the above studies highlight the fact that noise or harmonics in the ultrasonic range can have a significant impact on human cognitive functions and thus can cause negative (or positive) effects.

Discussion

In our research, we worked to develop a method to potentially reduce hearing loss caused by noise. All members of our research team have an interest in music and in any electronic equipment that amplifies the sound source appropriately. This common interest has formed the members into a research team and has resulted in a joint patent on the subject. The different characteristics of various electronic devices, electron tubes, semiconductor devices and decades of professional experience allowed us to formulate hypotheses which were mostly verified by various measurements and research. Our aim was also to identify the currently unclear variables responsible for the perception of loudness in human hearing. It was important for us not to start off on the usual path, not to create yet another device that applies dynamic compression or 'interferes' with the frequency response of the broadcast material or speech. In our preliminary research, we have checked whether the loudness perception of distorted sounds is higher than that of unmodified sounds, and in a long study we have found that sound amplification devices operating under different amplification principles have different distortion characteristics. This data was also used as input for our patent. Our patent-protected process demonstrates a device or sound modification method that can increase the sense of loudness without interfering with the fundamental harmonics of the signal or altering the sound pressure level. Our measurements have demonstrated the feasibility of the procedure and several possible implementation methods have been described. Based on our literature research in the field of ultrasonic noise, it can be concluded that harmonics and noise are significant even when their frequency is much higher than the generally accepted maximum detectable frequency (20 kHz). Further research is needed to digitise the process and adapt it to digital portable media players and hearing aids. As the widespread use of portable media players poses a great risk to the younger generation, compatibility with them is a priority.

Evaluation of theses

I. Non-linear amplification causes a loudness increase;

The first thesis has been proven, the results of experiments with different distortions have shown that nonlinear amplification, distortion, leads to an increase in perceived loudness.

II. It is possible to create a method or device that can maintain the perceived loudness level at a higher value without changing the measured sound pressure level;

Our research team has created a new patent-protected method and device that can raise the perceived loudness level without raising the measured sound pressure level.

III. Increasing the perceived volume can be achieved without changing the fundamental;

The new method created by our research team displays the fundamental harmonic unchanged at the output.

IV. Increasing the perceived volume does not cause audible distortion;

The new device developed by our team has not caused audible distortion in the tests.

V. The perceived loudness increase is not dynamic compression;

In this thesis, several techniques have been described that manipulate the original sound material by dynamic compression or by different amplification according to the division of the musical material, thus modifying its information content, its musical enjoyability, and in some cases causing information loss. The novelty of the method we have developed lies precisely in the fact that we have not used methods that have been known and widely used for a long time, but have intervened in the harmonic domain. The thesis has been proven.

VI. The new method and tool created can be implemented in digital devices;

The verification of this thesis has not been done in this thesis, since the results obtained show that the procedure is operational in analogue form, but the digitalization has not yet been done. The sampling and quantization applied in the digitization process significantly reduces the presence of overtones and thus the efficiency of the procedure. Further tests and developments are needed to verify this thesis.

VII. The method can help prevent hearing loss caused by noise.

Further tests are needed to prove this thesis. The results obtained suggest that the increment is significant, but that digital devices are the main cause of noise-induced hearing loss among young people. A large number of tests of the digital version are needed to verify this thesis.

Acknowledgements

I would like to thank my supervisors, Prof. Dr. József Pytel and Dr. László Lujber, who have been a constant inspiration during my scientific work and have advised me in publishing my articles and performing the measurements. I am also grateful to my fellow researchers, Kálmán Máthé and László Czimerman, without whom this research would not have been possible. Kálmán Máthé was the inventor of our patent and the creator of the prototype. Without him, this research would not have been possible. I am also grateful to Prof. Dr. József Bódis and Prof. Dr. Endre Sulyok of the Faculty of Health Sciences of the University of Pécs for supporting my research and initiating my scientific work. I also owe my thanks to Viktoria Prémusz and Petra Szabó, who have given me a lot of useful advice over the years and have always been helpful in solving a problem. I also owe thanks to the Faculty of Engineering and Informatics of the University of Pécs, where I was able to continue my research as an assistant lecturer and researcher. Thank you also for your support in publishing our scientific communication. I am also grateful to Prof. Dr. József Ásványi, without whom our latest publication would not have been possible. I thank my wife, Ágnes Juhász, and my two young sons, Mihály Bagdán and Márton Bagdán, who have lived through these years with me, for their encouragement, help and patience.

But first and foremost, I thank my Lord and my God, who has kept me through all this and who has brought me to the possibility of obtaining a Ph.D. degree, as He willed.

MTMT Statistics, MTMT ID: 10031947

IF value:	**6.718**
Journal article	6
Book excerpt	2
Book	1
Other conference press releases	11
Intellectual Property Rights	1
Other	5
Total	26

Publications on which the thesis is based

Z, Kvasznicza ; I, Gyurcsek ; Gy, Elmer ; **V, Bagdán** ; I, Horváth, *Mathability of EMC Emission Testing for Mission Crucial Devices in GTEM Waveguide*, ACTA POLYTECHNICA HUNGARICA , 15 p. (2021), **IF: 1,219**

Bagdán, V. ; Máthé, K. ; Czimerman, L. ; Pytel, J., *Non-linear distortion against hearing loss*, TEHNICKI VJESNIK-TECHNICAL GAZETTE 27 : 1 pp. 53-57. , 5 p. (2020), https://doi.org/10.17559/TV-20180412145630, **IF: 0,67**

Bagdán, Viktor, *Halljuk-e az elektromágneses zavarjeleket? (2020)*, Pollack Expo 2020.02.27-28, Villamosipari és informatikai szakmai előadások

Kopcsányi, Gábor ; Vincze, Olga ; **Bagdán, Viktor** ; Pytel, József, *Retrospective analysis of tympanoplasty in children with cleft palate: A 24-year experience. II. Cholesteatomatous cases*, INTERNATIONAL JOURNAL OF PEDIATRIC OTORHINOLARYNGOLOGY 79 : 5 pp. 698-706. , 9 p. (2015), doi:10.1016/j.ijporl.2015.02.020, **IF: 1,125**

Bagdán, V ; Máthé, K. ; Czimerman, L. ; Pytel, J., *Pszicho-akusztikai eljárás halláskárosodás megelőzésére / Psycho-acoustic method for preventing hearing-loss*, In: Kósa, Balázs; Springó, Zsolt (szerk.) III. Interdiszciplináris Doktorandusz Konferencia 2014 : Abstract

Bagdán, Viktor, *Harmonic Distortion Character of Amplifier Types*
In: 32nd International Conference Science in Practice (SiP 2014), Eszék, Horvátország (2014) Paper: Section 8/1.

Bagdán, Viktor ; Máthé, Kálmán ; Czimerman, László ; Pytel, József dr., *Pszicho-akusztikai eljárás halláskárosodás megelőzésére*, FÜL-ORR-GÉGEGYÓGYÁSZAT 2014 : 60 (2) pp. 44-46. , 3 p. (2014)

Bagdán, Viktor ; Máthé, Kálmán ; Czimerman, László ; Pytel, József dr., *Pszichoakusztikai eljárás halláskárosodás megelőzésére*, In: A Magyar Fül-, Orr-, Gége és Fej-, Nyaksebész Orvosok Egyesülete Audiológiai Szekciójának 51. Vándorgyűlése

Bagdán, Viktor, *Halláskárosodást megelőző elektronikai eszköz: TV interjú (2014)*, Kutatói portré, Pécsi Tudományegyetem, Tudománykommunikáció a "Z generációnak", http://www.zgeneracio.hu/

Máthé, Kálmán ; Prof., Dr. Pytel József ; Czimerman, László ; **Bagdán, Viktor**, *Emberi fül torzítását modellező eszköz, valamint eljárás hangjel feldolgozására*, 110798-13773E/SZT , Benyújtás éve (szabadalom): 2012 , Benyújtás száma: 110798-13773E/SZT

Bagdán, V. ; Máthé, K. ; Czimerman, L. ; Pytel, J., *Elektronikus eszköz halláskárosodás megelőzésére: Electronic Device for Preventing Hearing-Loss*, In: II. Interdisciplinary Doctoral Conference, Pécs, Magyarország (2013) pp. 138-141. , 4 p.

Bagdán, Viktor ; Máthé, Kálmán ; Czimerman, László ; Pytel, József, *Egy zajos kísérlet – Megelőzhetjük-e a halláskárosodást?*, EGÉSZSÉG-AKADÉMIA 4 : 3 pp. 195-198. , 4 p. (2013)

Bagdán, Viktor ; Máthé, Kálmán ; Czimerman, László, *Innovációs díjat kapott a Természetes Hangtorzítás elnevezésű Pollackos fejlesztésű találmány.: TV interjú (2013)*, Pécsi Tudományegyetem, Universitas Televízió (UnivTv), Megjelenés: 2013. október 30., http://www.univtv.pte.hu/hirek/692,

Bagdán, Viktor ; Máthé, Kálmán ; Czimerman, László ; Pytel, József, *A Noisy Experiment: Can we prevent hearing-loss?*, In: PTE ETK Egészségtudományi Doktori Iskola III. Tudományos Fóruma (2013) Paper: 15:45-15:55

Bagdán, Viktor ; Máthé, Kálmán ; Czimerman, László ; Pytel, József, *Természetes Hang Torzítás: Natural Sound Distortion (NaSDi)*
In: „Középpontban az INNOVÁCIÓ”, (2013) Paper: 2. 9:00-9:15

Bagdán, Viktor, *Megelőzhetjük-e a halláskárosodást?: Előadás nem csak fesztivál-függőknek*, In: Kutatók Éjszakája 2013 (2013) Paper: E78, 20:00-21:00

Máthé, Kálmán ; Czimerman, László ; **Bagdán, Viktor** ; Pytel, József, *NaSDi: Natural Sound Distortion, Természetes Hang Torzítás (2013)*
Rádió interjú a Kossuth Rádióban, Adás: 2013. június 25. 16:00-16:30,

Viktor, Bagdán ; Kálmán, Máthé ; László, Czimerman ; József, Pytel, *A Noisy Experiment: Can we prevent hearing-loss?*, In: SIP 2013 : 31st International Conference Science in Practice, (2013) Paper: sip_2a2

Viktor, Bagdán ; Kálmán, Máthé ; László, Czimerman ; József, Pytel, *Electronic Device for Preventing Hearing-Loss: Poster and Prototype stand showing / prototípus és poszter stand*
In: Dobay, Kata (szerk.) PÉCSI TUDOMÁNYEGYETEM INNOVÁCIÓS NAP 2013 (2013) pp. 9:00-15:00.

Viktor, Bagdán, *Electronic Device for Preventing Hearing-Loss: Innovation Forum, University of Debrecen*, In: Dr. Bene, Tamás (szerk.) A Debreceni Egyetem Napja : Innovációs Fórum, (2013) Paper: 1.

Viktor, Bagdán ; Kálmán, Máthé ; László, Czimerman ; József, Pytel, *Electronic Device for Preventing Hearing-Loss*, In: Zoltán, Kvasznicza PhD (szerk.) Science in Practice 2012 : Scientific Electrotechnical Conference, Pécs, Magyarország : Pécsi Tudományegyetem, Pollack Mihály Műszaki és Informatikai Kar, (2012) Paper: A202-O2

Viktor, Bagdán ; Kálmán, Máthé ; László, Czimerman ; József, Pytel MD., *Electronic Device for Preventing Hearing-Loss*, In: János, Szentágothai Scholastic Honorary Society (szerk.) Abstracts of the János Szentágothai Memorial Conference and Student Competition
Pécs, Magyarország : János Szentágothai Scholastic Honorary Society, Faculty of Sciences, University of Pécs, (2012) pp. 25-25. , 1 p.

Viktor, Bagdán ; Kálmán, Máthé ; László, Czimerman ; József, Pytel, *Electronic device for preventing hearing-loss*, In: Szabó, István (szerk.) 1st International Doctoral Workshop on Natural Sciences, University of Pécs, Pécs, Magyarország : University of Pécs, (2012) pp. 15-16. Paper: O-02 , 2 p.