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ABSTRACT

Costruction of an inearmicrofon-concept to explore the quality of voice by positioning a pressure reciever in the human, outer auditory canal.

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The thesis will explore whether or not the usage of a custom pressure receiver within the outer auditory canal can deliver the same signal quality compared to Lavalier microphones measured by the signal to noise distance. Lavalier microphones and its disadvantages are identified by the typical use cases of the system, namely television and theater productions. Among other things, the frequency response which is produced by the positioning is significant for the thesis.

Due to the positioning of the microphone in the external auditory canal it is assumed that due to the loss of the sound energy at high frequencies the positioning in the ear is not meaningful and language cannot be transmitted in high quality. The motivation of this work is that only lavalier microphones have been used for television productions. The idea of positioning the pressure receiver in the ear arises from the fact that people can hear themselves. The eardrum works like a pressure receiver therefor the thesis explores whether a pressure receiver in the ear can achieve the same quality as a lavalier microphone or not.

Further problems that the custom pressure receiver need to solve, appear when using artificial human heads. An artificial human head is reproduced at the artificial head stereophony. The aim is to mimic the natural temporal modulations of the sound, the distance between the ears and the texture of the auricle. Frequency-dependent directional hearing by the nature of the auricle is also achieved. Since the experiments are held under controlled conditions, the volume of possible background noises in the television studios need to be discussed. As they add additional noise and interference to the signal.

Noise within television studio can be caused by the audience but also by the electrical installations. It is reported in the literature that the resting volume of a television studio is about 20_dB of noise. While the audience is usually instructed when to make noise or be quiet, a maximum entertainment level noise of 60_dB and a minimum level of 35_dB (whisper). These possible interfering signals are included in the measurements. The 20_dB silence of the television studio were not included in the investigation, since the test environment is considered critically and depending on the results of the still signals, noise can be attributed to the television studio.

Furthermore, the hearing is capable of amplifying signals via the auditory ossicles and the eardrum. It is also known that the own language can be transmitted through bones. The eardrum is connected to the bones by the stratum fibrosum. Also the snail in the ear is embedded in the skull over the oval window and connected with it. Anatomically, the best preconditions are that the sound is directed to the bone of the external hearing. It becomes clear that it is physiologically possible to transfer sound to the path of the airborne sound and the sound diffraction around the Kof but also to the human body. The physiological conditions of the natural human physiology make this possible.

Problems of the custom pressure receiver can be blood rush, joint sounds, breathing or movement by the rotation of the head. As the internal and external ear is a strongly perfused organ, there may be noise transmission by the blood flow. Additional to the intrusion of the system, blood noise in the eardrum can raise the noise, this had to be investigated. Breathing can also provide for a pressure difference in the inner ear via the connection of the inner ear with the nasal sinuses and thus for a tympanic movement, which can be transmitted to the system. This is also the case. The system must be designed for an industrial maturity in such a way that the rotation of the head by a good strain relief of the cable does not set the system in motion. This problem could not be investigated in this work because the cable of the prototype was not yet flexible enough. Swallowing is also a possible noise. This could not be included in the work because swallowing cannot be done continuously (without a break). Breaks distort the mean value. A further investigation is needed to determine how much hiccups affect the system.

The position of a transducer in the outer auditory canal as well as the human auditory canal itself need to be briefly elaborated. The phenomena which have a decisive connection with the system are explained in more detail. Acoustic effects of human physiology on the acoustic signal, such as frequency modulation, temporal modulation or body sound transmission are listed here. On the basis of the research of the biological conditions, the investigation and the idea of the construction should be justified. In addition to research, expert interviews were conducted. The expert interviews serve as a source of knowledge in this work. As the area investigated in this work is still relatively unexplored, the decision was made to use the many years of experience of the expert experts on the subject in order to prevent sources of error.

A method adapted to the study had to be used for the thesis. The quality of the signals recorded in the course of the measurements had to be estimated by means of their energetic distance to signals defined as interference signals. The methodology provides for the signals to be divided into frequency ranges (depths, centers and highs) and to compare the distance of the useful signal to the interference signals in each frequency range. The determination of the signal-to-noise distance is carried out in parallel for both systems. Both systems use the identical capsule from Sennheiser.

When recording, both systems had to be recorded simultaneously on the same recording device (Sounddevice 664), so that the same signal can be recorded on both systems. This was important for the comparability of both systems. The final, qualitative assessment and evaluation takes place by means of a comparison of the two systems with respect to their frequency-dependent distance from the interference signals.

The recordings for the measurements were carried out in a reflexion room at the Technical University of Berlin. In order to increase the validity and reliability, direct sound measurements had to be recorded by means of diffuse sound without influence or modulation. A speaker, a reference microphone, an interface, a recording device and a PC were used for the measurements. In the measurements, the human speech is considered at normal speech volume (60_dB). For this the speaker had to speak consistently throughout the tests, to ensure that both in ear and lavalier microphone recordings were reliable.

A corridor of a maximum of + - 6_dB had been defined to avoid deviating from the specified language level. To enable the speaker to be controlled, the reference microphone was placed one meter away from the loudspeaker and the speaker. The reference microphone should be able to control the volume of the speaker by metering on the DAW. To find out how much dBFS is generated in the DAW at a certain level, the following procedure was used: The assistant increased the volume of the loudspeaker up to the level meter, which stood at the same level as the sound source as well as the reference microphone. The examiner said stop as the desired level of 60 dB was reached. On the DAW, it was then possible to see how large the resulting level in dBFS was. The gain of the reference microphone was set to Unity at the interface.

Below, the steps from the initial sketch to the completed microphone are described. For the construction of the in-ear microphone, it was necessary to develop a device which allows to integrate the microphone into the human auditory canal. During the construction of the device, care has also been taken to achieve the best possible decoupling of the system with respect to possible vibrations / impact sound on the basis of the material selection. The steps during the development phase and the design were well discussed with the German hearing aid manufacturer Audia Akustika. In this way a smooth and fast production took place. Communication errors were prevented through efficient communication by email as well as telephone conference calls. After completion of the prototype, which was delivered by Audia Akustika. The assembly of the microphone took place at the Little Media Studios in Berlin under the supervision of the sound engineer Sebastian Metzner. Before that, Sennheiser, who supplied the microphone capsules and the lavalier microphones, discussed the process of interconnecting the capsules. The completion of the in-ear microphone took two days. After completion, the microphone was checked for its functionality. The detailed description of the construction, the microphone can be found in the attached production logbook in the appendix of the thesis.

The measurements were carried out as planned in a measuring laboratory in the Technical University of Berlin. The continuity of the procedure was observed throughout the measurements. Care was taken to set the measuring instruments correctly. For example, use the highest possible recording quality of the Sounddevice 664 for the measurements or have the same amplification of both systems in order not to raise the noise artificially in one of the two systems. Care was taken to ensure that the level was always checked at a distance of one meter. The systems were taken parallel as planned so that the same signal is present on both systems. The systems would have to achieve the same values for the same signal for the same quality as the signal to interference signal.

As mentioned earlier the plan was to use an analyzer (Voxengo plug-in) to measure the amplitudes of the respective frequencies over a time of 10 seconds, in 500_ms. The values were measured every 500_ms using the analyzer and then inserted into an Excel sheet. Here all the amplitudes which were present in dBFS were first converted into dBu and then into voltage. From the voltages, the table calculated a rms value for each frequency range, each signal. The rms values were then used to determine a distance between the two signals.

Later during the evaluation, reference was made to the difference between the signal and the interference noise distance of the two signals. The deep frequency range, which is not relevant for the transmission of human speech, was explained briefly. The two systems itself differed very little. In the middle range, it was found that the in-ear receiver is slightly worse than the Lavalier microphone. This could have been caused by the noise of the blood and surrounding noise. Therefore a useful signal with lower energy and a disturbance signal with higher energy were present.

The high frequencies showed worst results for the in-ear receiver compared to the Lavalier microphone. This was caused by the diffraction of the sound around the test person's head. Hence information gathered through prior research was confirmed here. The signal from the in-ear receiver is, however, with the exception of a noise (loud breathing) according to the expert Sebastian Metzner still usable. The microphone is suitable for the transmission of one's own human voice. Using equalizers and noise gates, which is also the case with the Lavalier microphone. An acceptable result can be achieved in the future. The hypothesis could not be confirmed. An in-ear receiver cannot achieve the same quality, measured at the signal-to-interference distance. However, this does not mean that the system cannot be used in industry. The company Soundman (artificial head manufacturer) is enthusiastic about the idea and would like to further develop the system, influencing different market segments such as hearing aids. In conclusion, although the hypothesis is invalid, a foundation was built for further research, which deals with the transmission and recording of one's own voice with the use of natural auditory physiology.