Field Study News

SoundFlow:
Seamless adaptation to every soundscape

Overview

For several years Phonak has been the only company worldwide to use a "multi-base" automatic system. This is an automatic system that not only activates individual parameters such as directional beamforming or noise cancellation in a hearing program, but also independently switches between defined hearing programs. This successful and reliable principle has been further improved with the new automatic system called SoundFlow, which is now employed by products based on the CORE audio-processing platform. Depending on the classification of the listening environment, SoundFlow mixes the individual hearing programs for the best hearing comfort and optimized intelligibility. This adjustment is made continuously, in real-time resulting in a unique blended program for each situation. 15 test persons with normal hearing were enlisted to assess speech comprehension and the presence of audible program switching in SoundFlow. SoundFlow shows considerable improvements with regard to both the audibility of program switching and speech comprehension compared to the previously multi-base automatic system "AutoPilot."

Introduction

SoundFlow is a new automatic system first introduced by Phonak with the new products Exélia & Naída, based on the CORE audioprocessor. With the previous automatic system, AutoPilot, the hearing systems analysed the noise environment and subsequently selected one of four defined hearing programs: "Calm Situation", "Speech in Noise", "Comfort in Noise" or "Music". The ability of the system to select the best option for each situation from these programs offers a major advantage compared to automatic systems with only one base program (Büchler 2001). However, in complex environments with a mixture of elements, such as speech in noise and music, the automatic system is forced to compromise and select the most appropriate program. In addition, the somewhat abrupt transitions between the individual hearing programs are sometimes noticeable to the hearing system user. SoundFlow was developed to eliminate these disadvantages and to optimize use and comfort for the end-user. Depending on the situational analysis, different parameters are continuously adjusted within SoundFlow which blends the various base programs to create a unique program specifically adapted to the current situation.

Test objective

The goal of the investigation was to determine whether the audibility of program switching is reduced by SoundFlow's continuously adapting method. In addition, the study wanted to determine whether speech comprehension can be subjectively improved by mixing the base programs, thereby achieving a more exact adjustment to the hearing environment.

Study design

First, the audibility of the program switching using the SoundFlow algorithm would be compared to the AutoPilot algorithm. For this purpose, a sound sample was created in which differently defined listening situations alternated, such as between a quiet speech situation and a pure noise situation, etc. In order to exclude differences in the source sound materials the sound track was electro-acoustically fed into hearing systems in which the different automatic systems were active. The output sound tracks were subsequently recorded and digitally stored for later presentation via headphones. Normal hearing test persons listened to these sound samples and were requested to note the number of audible program transitions. In the second part of the test, participants assessed speech comprehension in a complex noise situation, meaning a situation which could not be clearly classified as one of the four basic programs. An example for this is a conversation in a noisy environment in which music is also in the background. For both automatic systems, short sound tracks were recorded and subsequently presented in a double-blind, paired comparison.

Test persons and hearing systems

Fifteen people with normal hearing participated in the tests. A criterion for exclusion for test participation was defined as a hearing threshold of greater than 20 dB at any frequency between 125 Hz and 8 kHz. Ten of the test participants already had experience in listening to hearing systems, while

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five people were not familiar with the sound of a hearing system prior to the tests. The hearing systems used included four SaviaArt 111 dAZ (digital AudioZoom) hearing systems which were modified for the tests such that the selected sound tracks could be supplied digitally. For two of the hearing systems, the AutoPilot algorithm was replaced by a prototype of the SoundFlow algorithm.

Results

Analysis of the automatic switching between the programs shows that for AutoPilot the number of noticeable transitions was between 11 and 51, whereas for SoundFlow 0 to 46 switches were identified. As shown in Figure 1, all participants noticed considerably fewer program switches, or even none at all with SoundFlow (red) compared to AutoPilot (blue; Figure 1). It is noteworthy that whenever there was a sound situation which was classified distinctly as only one of the four defined programs there was no difference between AutoPilot and SoundFlow. This is to be expected, as for such situations, the SoundFlow blend is essentially the same as the individual AutoPilot program selected for that situation. Different SoundFlow behavior related to program blending becomes more meaningful in complex and/or mixed situations.

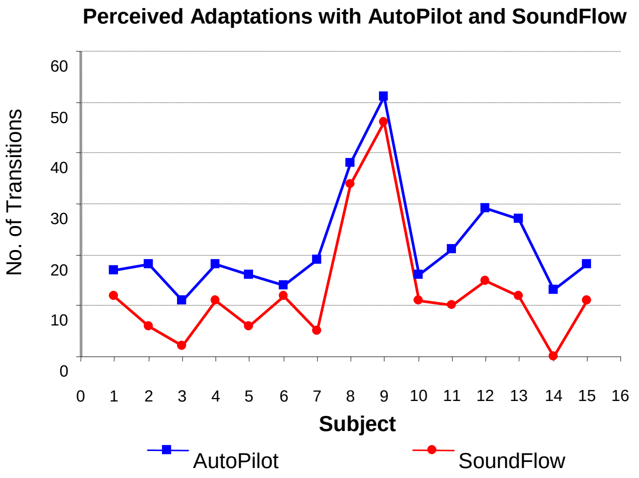

Figure 1: Means of the program transitions noticed by each subject tested. (blue) = AutoPilot; (red) = SoundFlow

In the second part, test participants heard short alternating sound tracks from the same noise situations which were recorded with the AutoPilot or the SoundFlow system. 44% of the sound samples were rated as clearer in speech intelligibility with SoundFlow, whereas 21% of the sound samples were rated to be clearer with AutoPilot. However, 35% of all sound tracks were rated similarly in terms of speech intelligibility. As noted above, this is to be expected for listening situations only requiring one hearing program to be active (e.g. "Speech in Noise"). For more complex, mixed situations, SoundFlow blends different hearing programs in order to adapt more specifically to the sound environment, (see Figure 2). It appears that both multi-base automatic systems adapt adequately to different sound environments, but in different ways.

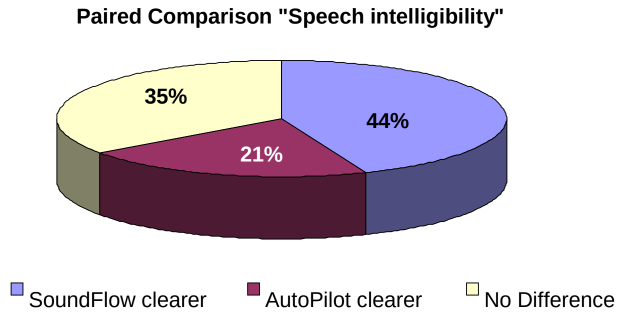

Figure 2: Subjective assessment of speech comprehension according to sound samples.

Conclusion

Single-base systems activate features only within one program. Fine tuning to improve performance in one situation can actually decrease performance in other environments. The multi-base approach as used in Exélia and Naída overcomes this limitation allowing independent fine tuning to enhance differing settings. In complex situations where multiple hearing programs are blended, SoundFlow differs from AutoPilot. In situations where only one hearing program is active, SoundFlow and AutoPilot might not be distinguished as the SoundFlow blend and the AutoPilot program are essentially the same at that moment. SoundFlow accurately analyzes the ongoing sound environment and facilitates smooth mixing of individual hearing programs in real-time. Therefore, SoundFlow always offers good audibility and hearing comfort even in complex or quickly changing situations. The blending transition of the SoundFlow algorithm reduces the number of noticeable program switches considerably and thus ensures even greater comfort and a higher spontaneous user acceptance. In some cases, regular volume changes of the test sample may also have been mistaken as program switching. Nonetheless, the ongoing blend of hearing programs with adaptive integration of various parameters results in a subjective improvement of speech intelligibility. Thus, it appears that the already proven and accepted multi-base approach is now even more effective.

References

Büchler, M. (2001), "How good are automatic program selection features? A look at the usefulness and acceptance of an automatic program selection mode," Hear. Rev. 9/2001, 50-54, 84 Also in Hörakustik 10/2001, 63-71.

Büchler, M (2002) "Algorithms for Sound Classification in Hearing Instruments"; ETH Zürich, Dissertation No. 14498

For further information, please contact:
Kai.Hessefort@phonak.com