An introduction to The Polaris™ platform

The new Polaris platform is the most intelligent platform ever developed by Oticon.

Much of the hardware has been updated. These updates make it possible to run the newest audiology features and to meet the constantly growing demands for connectivity to improve the lives of hearing impaired people.

This paper will introduce you to the Polaris platform in Oticon More 1. Specifically:

• The platform, how it is built, and the hardware updates

• The processing flow with all the different audiology features

• The new MoreSound Amplifier™ – the high-resolution flexible amplification system

• The new setting in MoreSound Optimizer™

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Introduction

The Polaris platform is the foundation for the Oticon More™ hearing aids. The platform consists of hardware, firmware (which for instance enables the communication from software to hardware), and software. The HW platform is a very important part of an advanced hearing aid. It sets the limits on what is possible in terms of sound quality, connectivity, signal processing and battery use time. The ability to tailor and optimize Oticon's HW platform to specific user needs is the reason behind in-house development of integrated circuits.

This tech paper will cover the platform, the complete processing flow, the new MoreSound Amplifier™, and a brief introduction to MoreSound Optimizer™.The other brand new feature, MoreSound Intelligence™, is covered in its own tech paper (Brændgaard, 2020a).

The Polaris platform

The Polaris platform consists of a digital signal processing (DSP) chip for audiology processing, a front-end (FE) chip that samples the microphone signals, handles power management, telecoil and near-field-magnetic-induction (NFMI) radio, and a Radio-Frequency (RF) chip that contains a 2.4 GHz radio supporting leading edge Bluetooth® Low Energy technology and various proprietary standards. These two radio technologies (2.4 GHz and NFMI) together form our dual radio system, TwinLink. TwinLink ensures simultaneous binaural connectivity (NFMI) and connectivity to external devices e.g., a smart phone and firmware updates (2.4 GHz). A solid-state (non-volatile) memory chip is also part of the Polaris platform.

The integrated circuits are very compact and consist of several million transistors. The DSP chip has been upgraded and now uses 28nm technology for the transistors. That represents a doubling of the number of transistors compared to the Velox S™ platform on a chip that is half the size of the DSP chip in Velox S.

Additionally, the computational power and the working memory (RAM) have been doubled compared to Velox S. This means the platform can make twice as many calculations and run twice as many applications all without adding to the processing time.

The solid-state (non-volatile) memory, used for storing code, has been increased eight-fold. This extra capacity allows for more features now and for future improvements via firmware updates.

One of the new developments made possible by the Polaris platform is an embedded Deep Neural Network (DNN). For more information on the DNN please see Brændgaard, 2020a.

The overall processing works in 64 channels. The signal processing, which makes the most important difference for the hearing aid, has been updated and now works in 24 frequency channels. That is 50% more channels than in Velox S. The extra channels are added into the frequency range from approximately 1500 Hz and up (figure 1).

The extra channels double the precision in a frequency range that includes the 1.5-5 kHz frequency channels. These frequency channels are the most important for speech sounds.
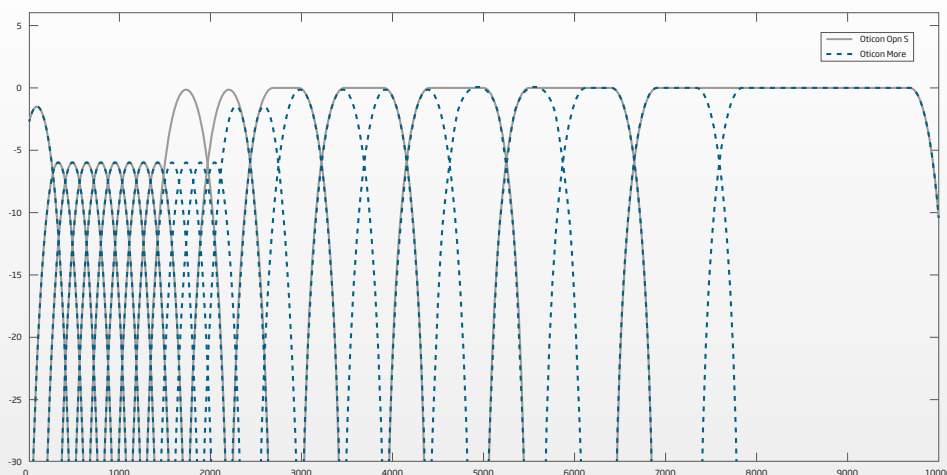

Figure 1: 24 processing channels on the Polaris platform (dotted line) compared to 16 processing channels on the Velox S platform (solid line). The number of channels is doubled from app. 1500 Hz and upwards.

Processing

To get a better understanding of what happens to the sound when it goes through the hearing aid, let's take a look at the processing flow shown in figure 2.

The audio signal is captured either by the two microphones, received from the telecoil, or streamed through a wireless connection to the instrument.

First, the input adaptive gain control in Clear Dynamics adjusts gain to prepare the signal for the A/D-converter, and re-applies any gain reductions after the A/D-converter to ensure that a larger dynamic range is available for further processing. Thus, the full sound input range up to 113 dB SPL is delivered to the DSP chip for processing without any initial, unnecessary compression.

Inputs from all types of input sources enter into individual filter banks with 64 channels that are used for processing. The filter banks are in linear phase and with the same group delay. This ensures that interaural time differences (ITDs) from all 4 microphones in a binaural fitting are intact.

The three filter banks convert the time domain signal into 64 channels, equally spaced with a bandwidth of 156.25 Hz, enabling more advanced mathematical processing.

The input calibration in 64 channels allows for precise compensation of head acoustics and preserves sound quality. At the same time it calibrates microphone sources and AUX sources (e.g., telecoil) separately.

From here the signal processing continues in 24 channels. Unlike the 64 channels used in other parts of the processing, these 24 channels approximate the auditory filters in the cochlea: they are narrower in the low frequencies than in the high frequencies. The fitting handles in the fitting software (Oticon Genie 2) follow the same 24 frequency channels.

First, the signal is processed individually on the two microphones by MoreSound Intelligence™ (MSI), which include Spatial Clarity Processing and Neural Clarity Processing (incl. the Deep Neural Network). MSI makes meaningful sounds stand out from the background while preserving access to all sound sources and directions containing distinct sound information. For more information on MSI please see Brændgaard, 2020a.

Wind Noise Management is also enabled here, if needed, before the two microphone signals are combined.

Next, the two microphone signals are combined and MoreSound Amplifier™ (MSA) together with VAC+ and Soft Speech Booster estimate and apply gain. MSA is a high-resolution, flexible amplification system that adapts in both speed and frequency resolution
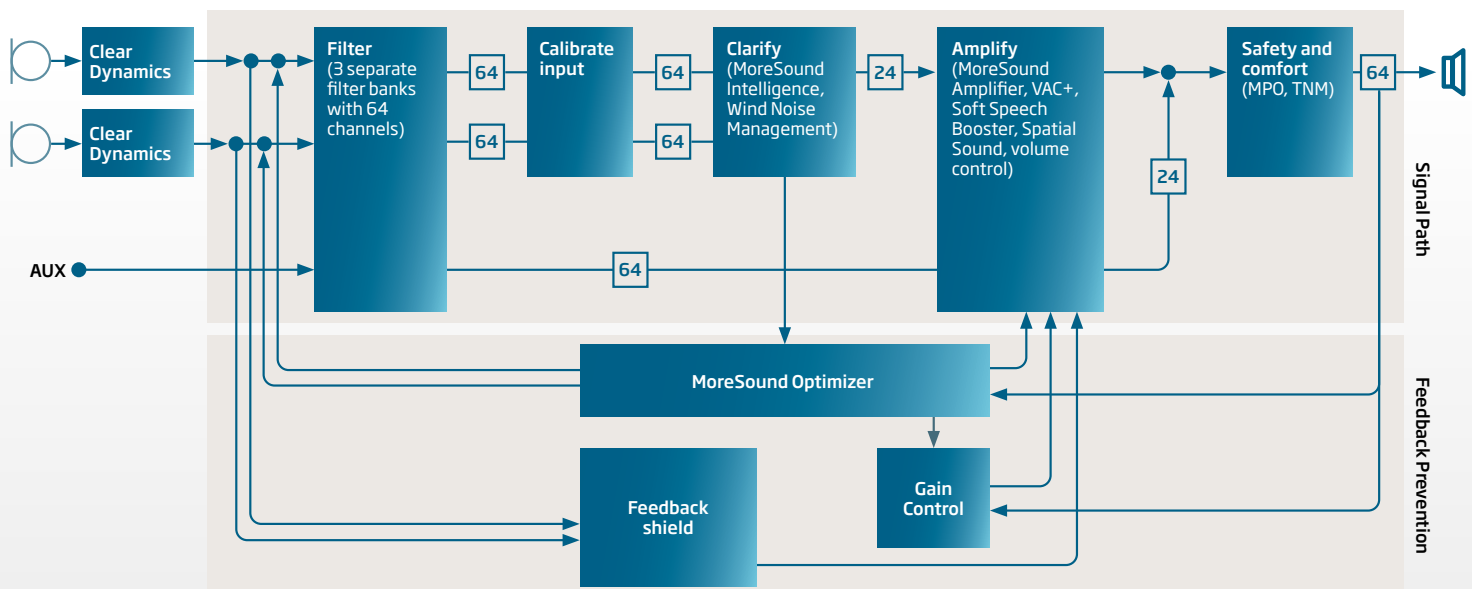

Figure 2: The processing flow from input to output on the Polaris platform. See the text for information on the different steps.

depending on the characteristics of the sound scene (see more detailed description later in this paper). Finally, Spatial Sound™ ensures that differences in level between the two instruments on the head are preserved for a better binaural balance.

The last steps before the processing is complete are the Maximum Power Output (MPO) and Transient Noise Management (TNM). These modules act as "backstoppers": they ensure that the output of the instrument is safe and comfortable and that it does not exceed MPO and that loud, sudden transients have been sufficiently attenuated for comfort. Thus, the full signal is processed and limitations to output are based only on audiometric concerns (hearing loss, uncomfortable loudness levels etc.).

Finally, a filter bank summation occurs. The signal from the 64 channels are combined and transformed back to the time domain and delivered to the speaker.

The feedback prevention system consists of MoreSound Optimizer™ (MSO) and Feedback shield. These two systems work together to prevent audible feedback – see more details later in this paper.

MoreSound Amplifier

MoreSound Amplifier (MSA) is a high-resolution flexible amplification system. It seamlessly adapts its resolution and speed to the nature of the prevailing sound scene. MoreSound Amplifier makes the full sound scene audible while maintaining the fine contrast and balance between sounds with a sixfold increase in frequency resolution and an adaptive speed pilot.

MSA takes over the role from Speech Guard™ LX. Where Speech Guard LX runs in one path with four channels, MSA runs in two paths, one with four channels very similar to Speech Guard LX, and another path with 24 channels (figure 3). The incoming sound signal is always processed in both paths simultaneously.

Firstly, the 24 channels are combined into the four channels used in the four-channel path. The sound is processed within the 12 dB linear window in each of the four channels as known from Speech Guard LX. This means that attack and release times will vary depending on the variations in the incoming signal. As long as the average signal stays the same the window remains stable and processing will be linear. Any sounds outside the window cause the window to shift quickly up or down to ensure sounds are kept within the comfortable range and remain audible. If the average sound level changes the window will slowly move to the new average.

After this processing, the four channels are split back into the 24 channels for further processing with the other path in MSA.

The other path in MSA constantly runs in 24 channels. Here again the signal is processed with adaptive attack and release times.

When both paths reach the 'Compare and Prioritize' step as can be seen in figure 3, the signals from the two processing paths are compared. Depending on which type of sound dominates in the processed signal, one of the two paths will be given priority in one or more frequency channels. The end result can be a mix of sounds processed by the two paths.
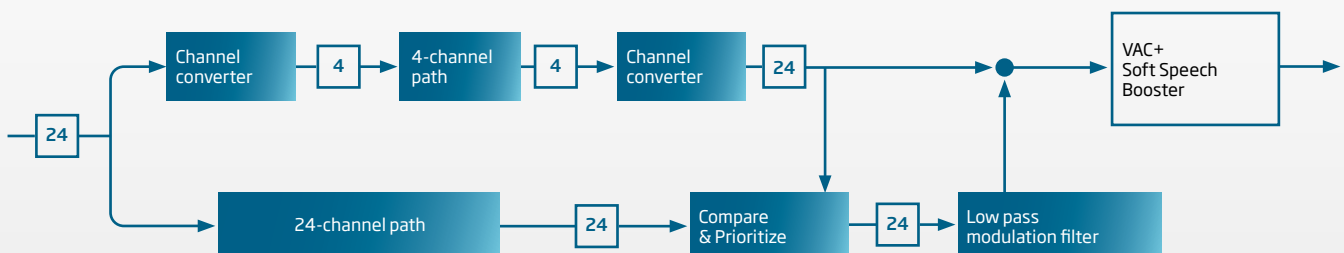

Figure 3: Processing flow of MoreSound Amplifier. See the text for explanation of the different steps.

The four-channel path is good at processing fast modulating signals such as speech signals which change rapidly in both amplitude, frequency and time where high precision in time is important. The 24-channel path is, on the other hand, good at processing stationary, slow modulating sounds like steady, narrow band noise which does not change much in either amplitude or frequency, where high precision in frequency is important.

So, in the comparison, if a fast modulating sound is dominant in the frequency channel, the four-channel path takes priority. On the other hand, if a slow modulating noise is dominant in the frequency channel the 24-channel path takes priority. A steady, narrow band noise will, for instance, be prioritized from the 24-channel path, but it will not necessarily dominate the entire frequency range of one of the bands in the four-channel path. This is shown graphically in figure 4. Technically, this prioritization is created by a low pass modulation filter.

The mix of the 2 paths is instantaneous and always on ensuring the right mix of the different sounds are passed on to the next step in the processing scheme: the gain map.

The benefit of the two-path system is seen very clearly in figure 5. The input sounds are a broadband speech signal and pure tones at 1, 2, and 5 kHz at 65 dB SPL. The magenta curve is measured using Speech Guard LX, and the blue curve is measured using MoreSound Amplifier.

In between the interfering pure tones we can see that speech gets more amplification adding to audibility and thereby to speech understanding. At the same time, we can see that amplification of the pure tones is lower which makes them more comfortable and gives a better target match.

MoreSound Optimizer

MoreSound Optimizer (MSO) and Feedback shield work together to counteract acoustic feedback from the receiver to the microphone to prevent audible feedback and gain reductions. The phase inversion and frequency shift from Feedback shield is always on, and MSO is constantly monitoring the feedback loop and will activate when needed. As soon as the open loop gain exceeds 0 dB MSO introduces a spectro temporal modulation signal to break the feedback path. This prevents audible feedback in the hearing aid, which is of great annoyance for both the user and people close by, and at the same time decreases audibility.
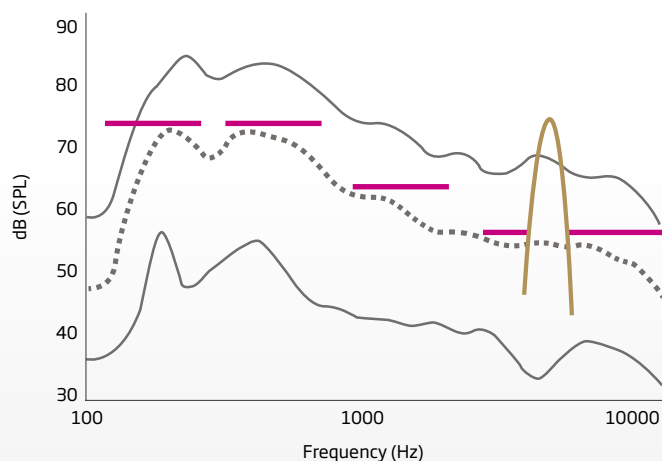

Figure 4: An unaided speech spectrum showing an example of the four level estimators in the four channel path in magenta. In golden the interfering steady, narrow band noise comes in and interrupts the highest of the four channels, but only in the narrow frequency range needed (Unaided speech spectrum adapted from Holube et al., 2010.).
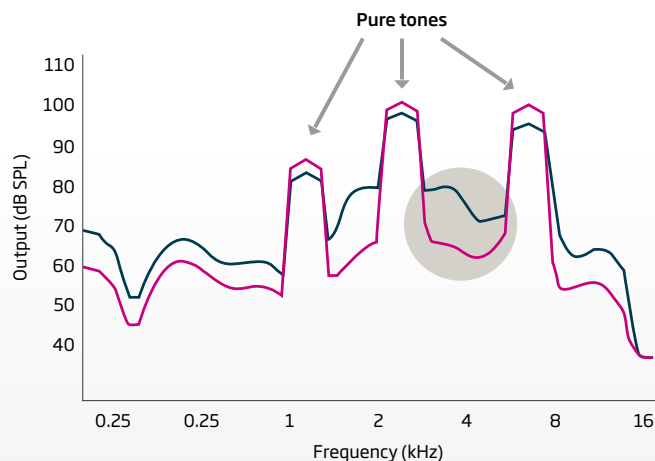

Figure 5: The input sounds are a broadband speech signal and pure tones at 1, 2, and 5 kHz at 65 dB SPL. The following common settings were used for the hearing aid configuration: Audiogram: 50 dB HL at 500 Hz and 70 dB HL at 2 kHz; Fitting Rationale: NAL-NL2; Noise reduction: off; Directionality: omni. The magenta curve is measured using Speech Guard LX, and the blue curve is measured using MoreSound Amplifier. In between the interfering pure tones MoreSound Amplifier provides more amplification for the speech signal than Speech Guard LX. This translates to more audibility of speech sounds in the presence of a narrow band noise.

In situations where the open loop gain exceeds +6 dB Feedback shield will activate the gain reduction and stabilize the situation.

MoreSound Optimizer uses the same technology as OpenSound Optimizer (Callaway, 2018), but to make the solution more flexible another setting is made available in the fitting software (Oticon Genie 2).

This additional setting will move the detection and activation point for the spectro temporal modulation, frequency shift, and phase inversion from approximately 940 Hz to approximately 1500 Hz.

The default setting with activation around 940 Hz is still recommended, but the new setting can be used for people who are very sensitive to the alterations of the signal, for instance when playing music.

Conclusion

The Polaris platform is the most intelligent platform ever developed by Oticon. Polaris is required to run the two new features, MoreSound Intelligence and MoreSound Amplifier. At the same time it also provides processing power for the updated MoreSound Optimizer, all the other effective features known from Oticon Opn S, and not least the increasing demand for wireless technology.

References

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