SONIC Spotlight

Radian Speech Processing

Intelligent and integrated systems on the Sonic Extend technology platform deliver a new approach to natural sound, speech understanding in noise and listening comfort in digital hearing aids.

Digital signal processing (DSP) is at the core of many audio-engineered products, from smartphones to portable music players, and hearing aids are no exception. In today's hearing aid industry, cutting-edge advancements that impact sound quality frequently occur. At Sonic, we constantly look for new ways to apply technology and design principles to create beneficial hearing solutions. That time is here again.

Introduction

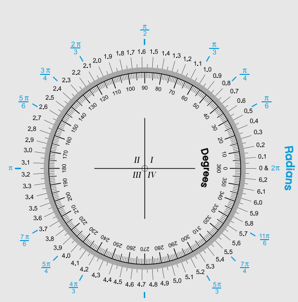
The **Extend** technology platform from Sonic provides new noise management and compression technologies – streamlined for improved sound quality, speech intelligibility and listening comfort. This paper describes the new compression system; ***a separate paper examines the new noise management system***. Collectively, they comprise our new core DSP – designed to actively **find, filter and fine-tune** sound with precision – in order to provide an optimal hearing experience for the listener. Read on to discover how **Radian Speech Processing** does its part to fine-tune sound, to help increase user satisfaction with their hearing aids.

Background

Hearing aids must constantly process sound to fit a listener's dynamic range of hearing. The use of wide dynamic range compression (WDRC) – the application of more gain to soft sounds, and less gain to louder sounds over a wide dynamic range – is commonplace in hearing aids. What's not common is how manufac-turers implement a compression strategy into a hearing aid. Thanks to the expanded capabilities of the Extend platform, new Radian Speech Processing (RSP) redefines our approach to compression with three main goals:

1. to amplify speech from the optimum angle in the listening environment accurately;
2. to improve the output SNR by applying less gain to noise in the presence of speech;
3. to preserve loudness levels for speech in the presence of persistent noise.

The following sections explain how it works, respective to the goals above.



The term **'Radian'** might sound familiar to you – it is a unit of angular measure that we often learn about in mathematics. Radians can represent a full 360° of a sphere or any angle in between (Fig. 1). Like radians, both meaningful sounds and distracting noise can arise from any direction, at any time. Radian-*technologies* are poised to perform rapidly, responding to speech and noise where and when they emerge in the listening landscape—for a better sound experience.

Figure 1: Radian example

Level estimation overview

A compression system performs many functions. Hearing care professionals know quite well that a selected fitting rationale will apply the desired compression characteristics for soft, medium and loud inputs for a specific hearing loss. The system must also amplify the signal while continuously adjusting the gain – the compressor does this automatically with a variable-gain amplifier. But before the system can perform either of these two straightforward tasks, it must first accurately measure the input levels of the signals it receives from the microphone. It does this with a level estimator.

A level estimator, therefore, belongs at the starting point of the compression system and is unique to every hearing aid family's DSP assembly. Hard-coded parameters determine the temporal behavior (i.e., speed) and the frequency contrast (i.e., clarity) that ultimately contribute to the hearing aid's overall speech intelligibility and sound quality (Salorio-Corbetto et al., 2020). In terms of speed, a fast-acting estimator uses a time frame with a short duration, providing short-term signal analysis to capture rapid changes in the listening environment. Alternatively, a slow-acting estimator uses a longer time frame, providing a longer-term signal analysis to characterize the overall level of the environment. For context, fast-acting systems may have release times less than 200 milliseconds whereas slow-acting systems may have release times greater than 200 milliseconds (Souza, 2002). In terms of frequency resolution, estimators can measure the input level either continuously or in segments (Scheller, 2002). The following text explains the two types in more detail.

Segmented estimation

Segmented estimation – A very common way to estimate the signal level in a hearing aid is by a process that first segments or splits the frequency range of the system into channels (Holube et al., 2016). Channels are sequential, overlapping frequency bands with fixed bandwidths. Hearing care professionals recognize this design as a multichannel WDRC system. This method works by splitting the input signal into separate frequency bands, determining a discrete level within each frequency band, and applying compression in each band based on its level (Plyler et al., 2013). Typically, a greater number of channels will provide better flexibility to shape the gain as desired. Following amplification, the channels recombine to construct an output signal. The temporal behavior of multichannel compression is of importance, as it influences the clarity of the processed speech sounds. For example, short-term signal analysis with fast processing times can reduce the frequency resolution for speech, because the pattern of gain across frequency changes rapidly with time (Moore, 2008), and can cause distortions like spectral smearing, overshoots and noise artifacts (Souza et al., 2012). Long-term signal analysis with slower processing times can resolve these issues (Hansen, 2002), and in addition, can improve listening comfort (Gatehouse et al., 2006; Moore, 2008). Measuring the band signal levels with slow time constants allows the system to apply gain more consistently with less chance of overshoot as the overall sound level is measured.

Continuous estimation

Continuous estimation – An equally established method for level estimation is possible via a continuous measurement process. This unique design avoids splitting the input signal into bands or channels. Instead, it measures the level of the intact wideband signal to apply compressive amplification. With this procedure, a digital filter applies a smooth gain curve to the measured sound pressure level that it receives from the estimator, according to the compression characteristics selected for the hearing loss; it then interpolates gain curves at additional input levels, providing variable compression ratios across frequency (Schaub, 2010). Adjustments to the compression ratio at each audiometric frequency can shape the gain as desired. The temporal behavior with this method also impacts the amplification of speech: slow-acting compression can potentially under-amplify soft inputs or over-amplify loud inputs, while fast-acting compression supports increased audibility of soft inputs and comfort of loud inputs.

Radian Speech Processing: Fine-tune

Radian Speech Processing puts forth a unique non-linear compression strategy to capitalize on the benefits of each level estimation type described above. While both types are customary in the hearing industry, RSP runs two estimation systems – continuous *and* segmented – together at the same time in the DSP assembly. This approach fine-tunes the amplification of sound in a new, efficient way. For context, the use of two estimating systems simultaneously is not a new concept – Sonic has already established this method with the previous platform. In it, a continuous estimation system tracks rapid, short-term changes in the listening environment to identify speech inputs *(Speech Variable Processing)*. An *optional* second continuous estimation system tracks long-term changes in the listening environment to identify noise inputs and treat them differently than speech *(SmartCompress)*. Evolving from this concept, RSP advances its new strategy, with *Speech Estimator* and *Segmented Estimator* technology. **Here's how it works.**

Speech Estimator

Tracking the cleaned signal input from Radian Noise Management, RSP's Speech Estimator uses continuous estimation with relatively fast time constants to rapidly measure short-term changes in the listening environment. It identifies speech inputs and estimates their level, to capture the fine structure of speech received from all angles – whether the input arrives from the front, sides, or behind. Fast application of gain with phonemic compression on the wideband signal provides audibility of soft phonemes and prevents their under-amplification.

The compressor continuously amplifies the variable speech input at the appropriate level for the hearing loss. It treats changes in level in a non-linear fashion so the system can quickly and accurately respond to soft, medium or loud inputs as they occur (Figure 2). As such, it prevents over-amplification of loud speech elements, so all sounds fit comfortably within the dynamic range of the individual. These actions allow RSP to fulfill its first goal related to speech intelligibility: *to accurately amplify speech from the optimum angle in the listening environment.*

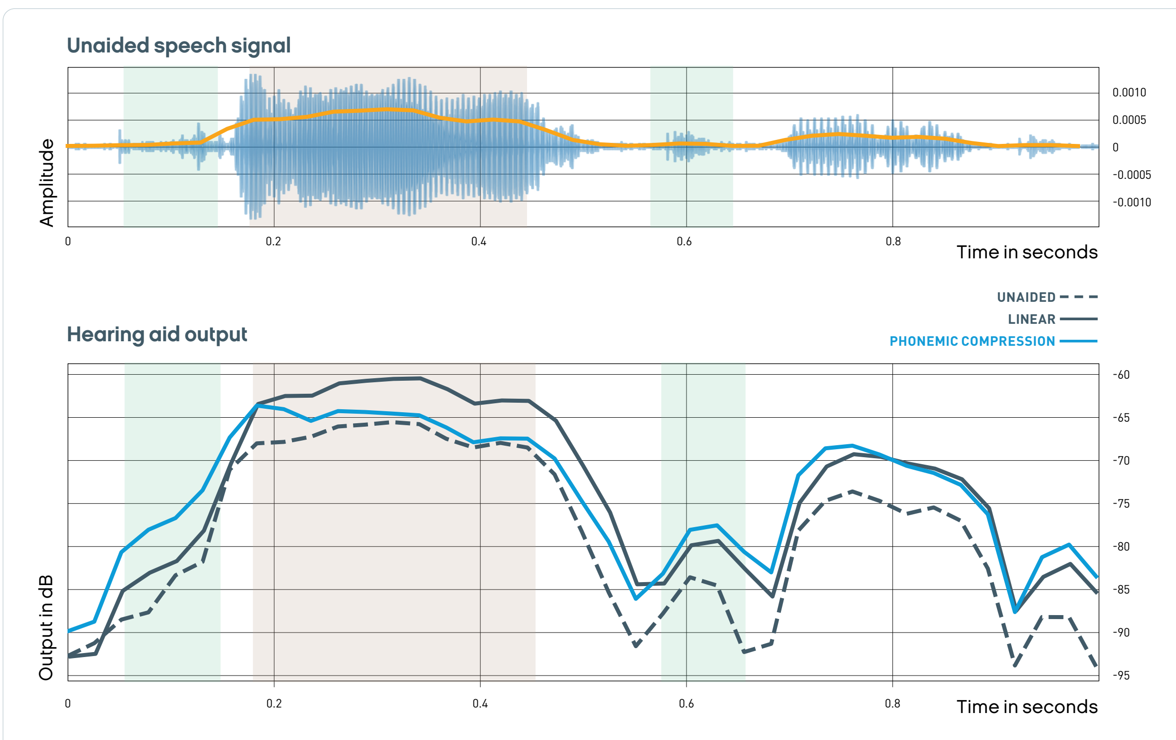


Figure 2: Clean speech signal (ISTS) @ 65 dB SPL (top graph) aligned with the measured effects of phonemic compression with RSP (bottom graph, blue line) and linear amplification (dark grey line). A soft phoneme and speech pause are in the light green areas and loud phonemes are in the light brown area.

Segmented Estimator

Accurately amplifying speech is a complex process indeed. However, the process becomes more complex when noise enters the environment. By nature, common everyday noise that fluctuates over time fills the spaces, or 'dips' in between syllables and pauses of a continuous speech signal, making it extremely difficult for some listeners to understand conversations in these situations (Lorenzi et al., 2006). Adding to the complexity, the mix of speech and noise can often occur at the same level, causing greater difficulty for a hearing system to amplify only the desired signals. Knowing the short-term *and* long-term SNR estimate of the listening environment helps the compressor differentiate between the two (Figure 3).

RSP relies on its Segmented Estimator which splits the input signal into 24 channels and uses a long observation window to measure these ongoing changes in the listening environment. Since gain requirements for noise are different than for speech, the Segmented Estimator instructs the multichannel compressor to limit the amount of amplification on the long-term signal level. At the same time, the Speech Estimator instructs the variable gain amplifier to apply the prescribed amount of amplification (according to the fitting rationale) on the short-term signal level. This twofold approach aims to improve the SNR coming out of the hearing aid (output SNR), since the noise – often found between speech pauses – receives less amplification than speech. And, it permits RSP to reach its second goal related to listening comfort: *to improve the output SNR by applying less gain to noise in the presence of speech* (Figure 4).

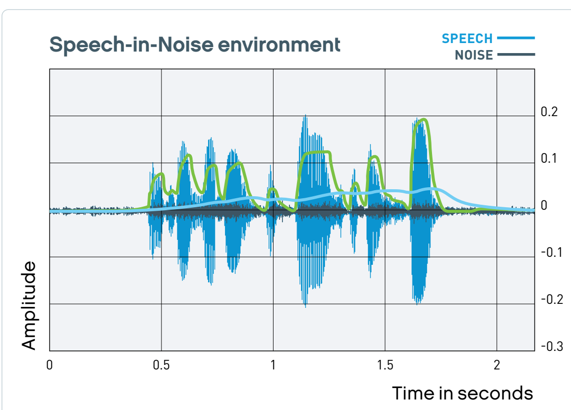


Figure 3: RSP detects the short-term (green) and long-term (light blue) SNR of a speech-in-noise environment.

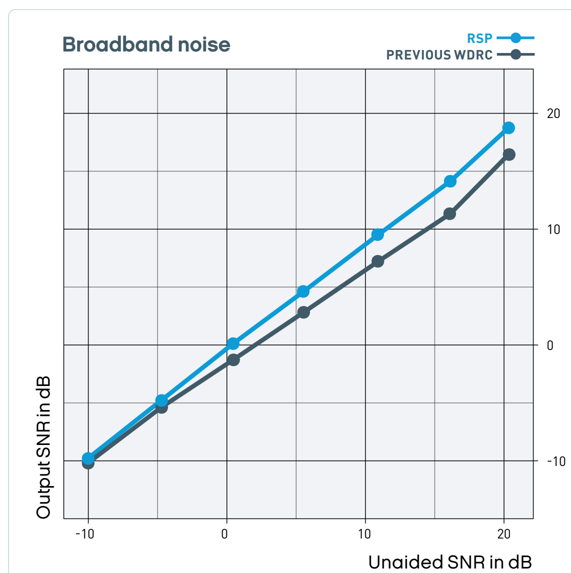


Figure 4: Hearing aid output SNR as a function of the unaided SNR and broadband noise (ISTS). RSP (blue) provides an improvement ranging from 0.3 up to 3.0 dB SNR with the broadband noise over previous technology (dark grey).

The design of the Segmented Estimator gives RSP a further advantage over previous technology in an additional listening scenario. We know background noise can come in a wide array of levels (from soft to loud), frequencies (from low to high) and durations (from brief to prolonged). Compared to common background noise fluctuations that are less intense, shorter, and generalized over a greater number of frequencies (*e.g., restaurant noise*), loud-level narrowband noise that persists for a long duration (*e.g., band saw, dental drill, alarm signals*) is especially hard for a hearing aid to manage. Persistent noise like this can skew the short- and long-term level estimations and negatively affect the amplification of speech. The Segmented Estimator's multichannel structure enables more precise suppression of persistent noise, thanks to its narrowband design. Meanwhile, speech is managed independently in parallel. This satisfies the third goal of the system: *to preserve loudness levels for speech in the presence of persistent noise* (Figure 5).

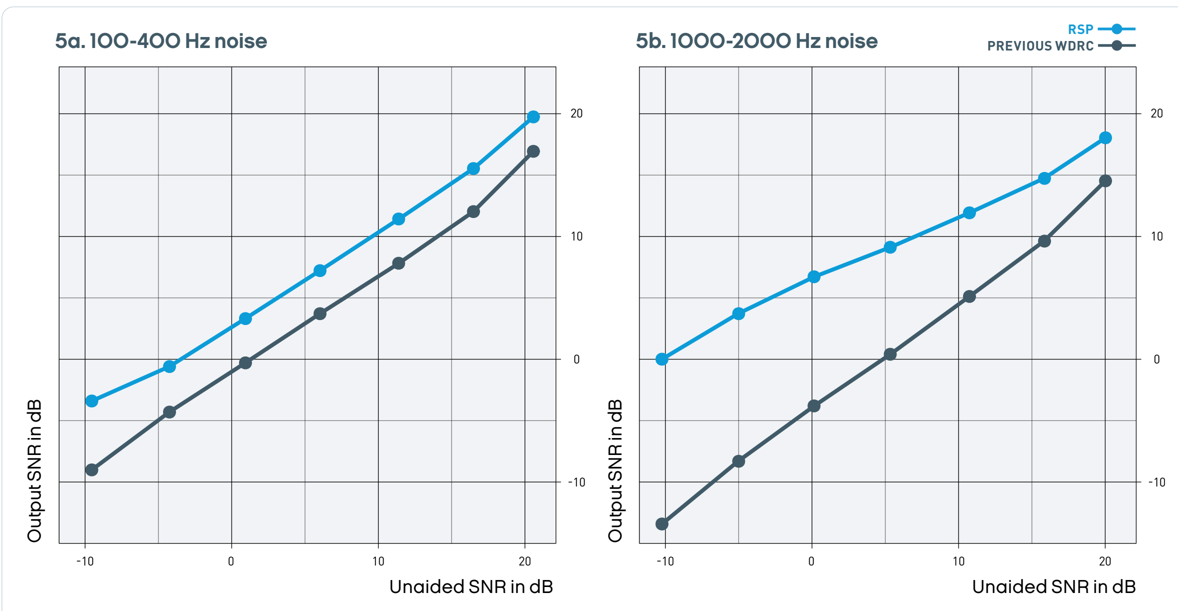


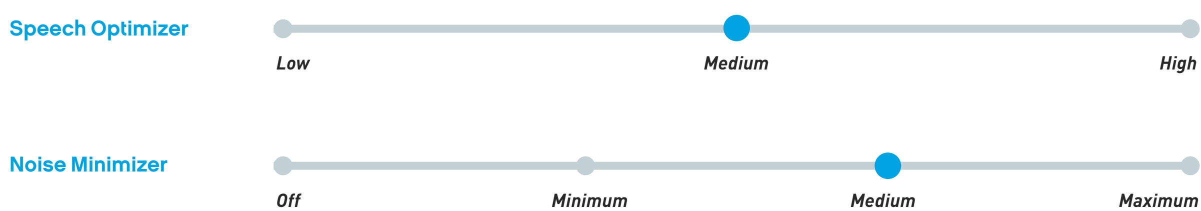
Figure 5: Hearing aid output SNR as a function of the unaided SNR and two different narrowband noise types (low, medium frequency). RSP (blue) provides an improvement ranging from 3.3 up to 5.2 dB SNR for the low frequency noise (5a) and from 3.7 up to 13.0 dB SNR for the medium frequency noise (5b) over previous technology (dark grey).

Software controls

Radian Speech Processing offers two fine-tuning controls to optimize speech and minimize noise, which help to customize performance according to user needs and preferences:

» *Speech Optimizer – increases the contrast of speech in a speech-in-noise listening environment when the noise reduction system is active. This control boosts speech between 1.5 - 5 kHz by a selected amount ranging from 1, 3 or 5 dB. Select **Low, Medium or High** to progressively increase the level of additional gain applied to speech signals in noisy environments.*

» *Noise Minimizer – controls the gain of noise-only input signals. To achieve optimal comfort in settings without speech, select from **Off, Minimum, Medium or Maximum** to gradually limit the amplification of noise by 1, 2 or 3 dB in non-speech environments.*



Summary

This paper describes Radian Speech Processing, part of the advanced hearing aid technologies in the Extend platform designed to improve speech intelligibility, listening comfort and sound quality for listeners with hearing impairment. RSP is a non-linear compression system which receives cleaned input from ***Radian Noise Management***. It then uses: 1) continuous estimation to amplify speech clearly and naturally without distortion, and 2) segmented estimation in 24 bands to limit the gain applied to general environmental noise to improve the output SNR, as well as preserve speech loudness levels in the presence of persistent noise.

United by the goal of enhancing speech while reducing distracting noise, the combined efforts of these systems form the all-new core DSP sound strategy from Sonic. Streamlined together, Radian-*technologies* actively find, filter and fine-tune sound with precision to advance the hearing experience for the listener.

For a demonstration or to learn more, please contact your local Sonic provider.

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