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Hybrid Sound Processing™: wide dynamic range compression with improved time and frequency resolution

Christophe Lesimple,
Research Audiologist

Wide dynamic range compression (WDRC) is one of the core functionalities embedded in a hearing aid. It is designed to compensate for the reduced residual dynamic range of hearing-impaired people by applying amplification as a function of the input level. The level estimator is one of the key elements as it drives the compression. In other words, the estimated level is used to decide how much amplification will be applied in the time and frequency domains (Figure 1). While the amount of amplification is computed according to the selected fitting rationale, different strategies can be applied to design and implement the level estimator which is sensitive to dynamic signals like speech.

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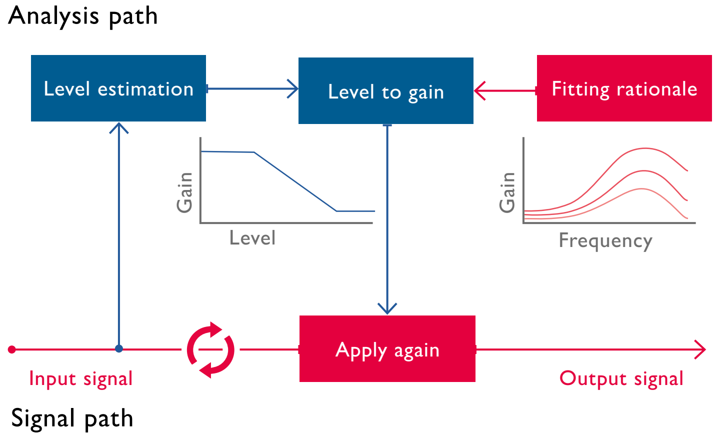

Figure 1: Block diagram for WDRC implemented in a hearing aid. The incoming signal is distributed into an analysis and a signal path. The level is estimated in the analysis path and the appropriate gain is retrieved from a level-to-gain function. The fitting rationale will calculate gain values for frequency and input level based on the audiogram and other individual characteristics (age, gender, language, etc.). The gain is applied to the synchronized input signal within the signal path.

This White Paper will first summarize the challenges met during the design of the level estimator, then describe the effect of different compression strategies for listening situations such as speech in quiet and speech in noise. It also includes a review of how human factors like cognitive skills interact with WDRC. Finally, it introduces Bernafon's Hybrid Sound Processing™ which is designed to provide optimal amplification for any listening situation.

The applied amplification directly depends on the level estimator's output.

Level estimation for dynamic range compression

Different parameters must be defined when compression is implemented in a hearing aid and it is important to understand how they will affect the way amplification is applied to dynamic signals. The analysis of the incoming signal is one of these important aspects, as the applied amplification directly depends on the level estimator's output (Figure 1). This estimation is challenging for signals like speech because the cues that are needed to make speech understandable vary over time and frequency. The time and frequency resolution of the estimator must therefore be adapted to provide the most important information for speech understanding and to remove details in the signal that are not needed to understand what is said.

Variations in the time domain are estimated from the envelope of the input signal. The envelope can be obtained with a low pass filter that will remove the faster variations from the signal. Different types of information can be retrieved by changing the filter's cut-off frequency. Rosen (1992) proposed the following classification of temporal features and the type of information that they contain for a speech signal:

- Envelope with the slowest variations between 2 and 50 Hz provides information about intensity, rhythm, and the phoneme's manner;
- Periodicity with variations between 50 and 500 Hz provides information about stress, intonation, and voicing;
- Fine structure with the fastest variations above 500 Hz provides information about the phoneme's place and voice quality.

An estimator with good time resolution is needed to retrieve segmental variations of the speech signal, which roughly follow the phoneme production rate.

The effect of time resolution can also be visualized on a short speech segment (Figure 2). An estimator with good time resolution is needed to retrieve segmental variations of the speech signal, which roughly follow the phoneme production rate. A slower estimator with lower time resolution will provide supra-segmental cues like prosodic variations. Segmental cues help the listener to understand what is said while prosodic cues will indicate how it is said.

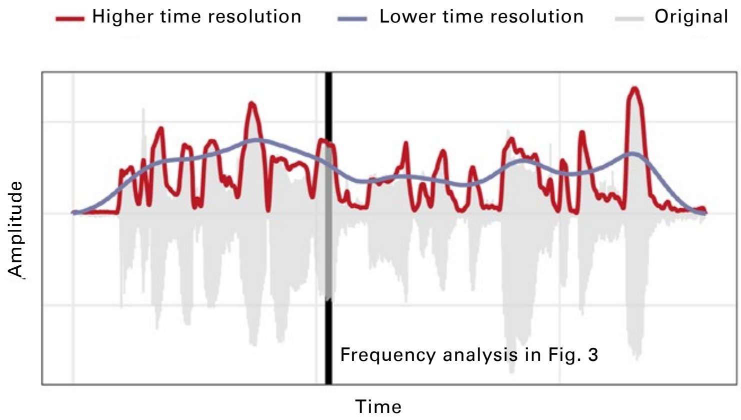

Figure 2: Envelope estimation of a speech signal (in gray) with a high time resolution in red (50 Hz low-passed signal) and a low time resolution in blue (2 Hz low-passed signal). Higher time resolution with fast estimation follows the production of each phoneme while the slower estimator provides information about long-term variations like intonation, rhythm, and stress.

The time resolution of the level estimator for dynamic range compression in hearing aids can be measured by the attack and release times (ANSI, 2014). The attack and release times indirectly reflect how the estimator is adapting to a sudden level change. They are defined by the time needed to stabilize the gain when the level of the input signal is increased for the attack time or when the level of the input signal is decreased for the release time. Attack and release times are used to evaluate the temporal behavior of the level estimator, i.e. so-called "fast" compression for a high resolution and "slow" compression for a low resolution in the time domain. Linear amplification is a special case where the estimated level has no influence on the programmed gain which remains constant over time.

Information like the vowel's formants or the phoneme production's place are in the frequency domain.

Until now, the analysis in the time domain was described on a broadband signal. However, information like the vowel's formants or the phoneme production's place are in the frequency domain. The information in the frequency domain can be provided by a Discrete Fourier Transform (DFT) on the speech signal (Figure 3). One argument of the DFT is the window's length which has a direct influence of the frequency resolution; if you take a wider window, you collect more information over time, and consequently obtain a better frequency resolution.

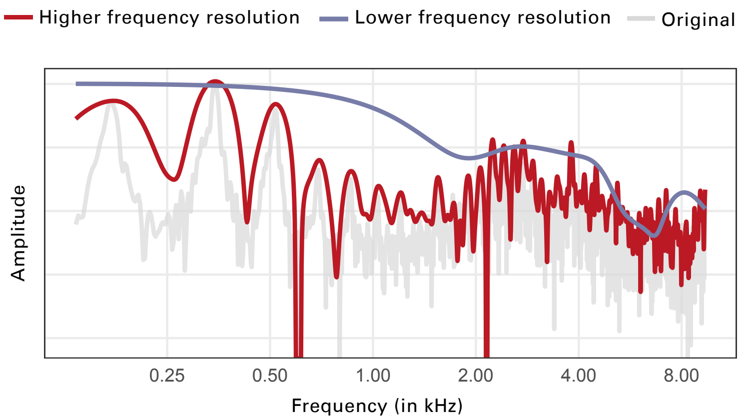

Figure 3: Analysis in the frequency domain of a vowel with different frequency resolutions. The original analysis is done on 4,096 samples which cover the entire steady portion of the vowel (in gray). A lower frequency resolution is given by a DFT over 32 samples (in blue) and a higher frequency resolution with a DFT over 512 samples (in red). For this analysis, the frequency resolution is linear over the entire bandwidth which produces a lot of ripples in the higher frequencies. This effect can be reduced with non-linear resolution in the higher frequencies.

A certain resolution in the frequency domain is necessary to find the location of the formants in Figure 3. The DFT, based on 512 samples, seems to be sufficient to extract the frequencies of the first three formants. With more samples, no additional information is provided and only noise will be added to the analysis. If an analysis requires more information in the frequency domain, then it must be made over a longer period of time. This implies that there is a direct relation between the time and the frequency resolution. The estimator must be slower, analyzing the signal over a higher number of samples, for better frequency resolution or a broadband estimator is needed for better time resolution. The frequency resolution in a hearing aid is not linear over the entire bandwidth and it is represented by the number of compression channels. The width of the channels usually increases with higher frequencies to avoid unnecessary information like that obtained with the DFT with linear frequency resolution.

The trade-off between the resolution in the time and frequency domains is similar to the photography of moving objects where the focus can be made either on time-based information or on the location of steady objects for a better spatial resolution (Figure 4). While the settings can be chosen before the event and some post-production treatment is possible for photography, the analysis of the incoming signal in hearing aids must be done in real time with pre-defined parameters. The parameters of the estimator must be carefully selected as their effect depends on the type of input signal and the expected benefit, i.e. provide audibility or comfort.

The analysis of the incoming signal in hearing aids must be done in real time with pre-defined parameters.

Figure 4: Trade-off between higher time resolution (top left) or higher place resolution (bottom left) in the analysis of a time-varying event. Combining both types of information (right) gives simultaneous details about the fast-varying event and the steady environment. The fast-moving owl would represent the fluctuations of a speech signal and the fixed background the frequency characteristics of noise for the analogy of a speech-in-noise condition.

There is a consensus about the advantage of fast attack times to protect hearing-impaired listeners from loud sudden sounds.

The optimal resolution in the time and frequency domains for a hearing aid has been intensively investigated in different listening situations, with different outcome measures, and different listener characteristics (Moore, 2008; Davies-Venn et al., 2009; Naylor & Johannesson, 2009; Alexander & Masterson, 2015; Kowalewski et al., 2018; Kuk et al., 2019; Salorio-Corbetto et al., 2020). While the results might change regarding the methodology and testing conditions, it seems that there is a consensus about the advantage of fast attack times to protect hearing-impaired listeners from loud sudden sounds that might be overamplified over too long a period of time if the gain reduction is not quickly applied. The appropriate release time and the optimal frequency resolution is still under discussion as it depends on the expected benefit, the listening situation, and the residual auditory capacities of the listener.

WDRC for speech in quiet

For a clean speech signal, WDRC must be able to restore the audibility of soft phonemes and keep louder phonemes at a comfortable level. Ideally, the estimator should be able to estimate each phoneme independently so that the applied gain would be phoneme specific (Dillon, 2012, p.181). It is assumed that the improved audibility of the softest phonemes should contribute to a better understanding of soft speech by the listener. The only strategy to achieve this requirement is to have a level estimator that is fast enough, with a sufficient resolution in the time domain, to estimate the level of each phoneme independently. Davies-Venn et al. (2009) and Kowalewski et al. (2018) have shown that fast compression improves consonant recognition over slow compression for soft speech. The advantage of level estimation with greater time resolution is important, especially when a soft target word follows an intense sound, for example, when someone's cough is followed by soft speech (Kuk et al., 2019). Improved audibility of soft phonemes has, however, some consequences on the temporal envelope of speech (Figure 5); compression with a fast estimator tends to reduce the contrast between the louder and softer phonemes (Jenstad & Souza, 2005; Moore, 2008).

The advantage of level estimation with greater time resolution is important, especially when a soft target word follows an intense sound.

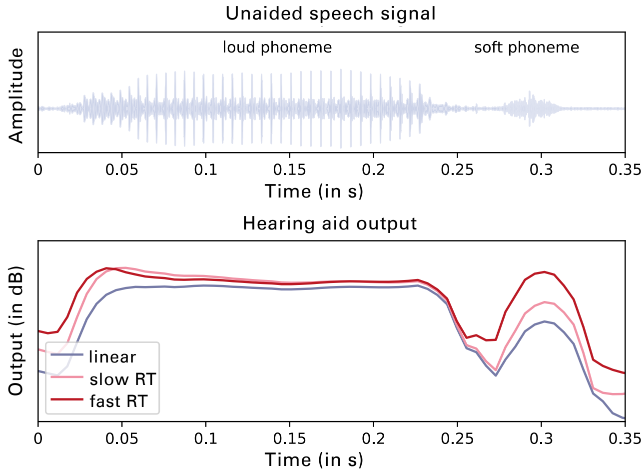

Figure 5: Effect of time constant on the amplification of a loud phoneme /a/ preceding a soft phoneme /t/. The unaided signal is represented by its waveform (top). The envelope of the hearing aid's output is shown for dynamic range compression with slow (pink) and fast (red) release times and linear amplification (blue). Effective compression given by fast compression improves the audibility of the soft phoneme but reduces the contrast between both phonemes.

Longer released time constants are usually used with multi-channel level estimators with an increased frequency resolution.

The temporal envelope can be preserved by using an estimator with longer time release values which will behave more like a linear amplifier (Moore, 2008). Longer released time constants are usually used with multi-channel level estimators with an increased frequency resolution (Alexander & Masterson, 2015). However, this strategy produces distortions in the frequency domain by reducing spectral contrast when the number of channels is increased. This spectral smearing can impair the identification of vowels where information is static and distributed in the frequency domain (Bor et al., 2008). This loss of information is critical for listeners with a hearing loss who have broadened auditory filters, i.e. with poor spectral resolution (Souza et al., 2012).

For speech in quiet, WDRC must restore the audibility of soft speech as a first intention. However, at the same time, it must try to reduce distortions in the time and frequency domains that might affect suprathreshold phoneme identification (Holube et al., 2016). This is important for hearing-impaired listeners using spectral information for static signals like vowels and temporal cues for dynamic information (Souza et al., 2015a and Souza et al., 2018). The requirements and effects of dynamic compression change when noise is added to the signal.

WDRC for speech in noise

It is quite challenging to compare different compression systems in speech-in-noise conditions because the result directly depends on the presented signal-to-noise ratio (SNR), the noise type (spectral shape and modulation), and the test conditions. WDRC reduces the level differences between two signals, such as a speech source and interfering noise, at the output of the hearing aid, when the signals are presented at different levels at the hearing aid input. It means that for listening situations with a positive input SNR, dynamic range compression will reduce the output SNR (Figure 6, top row). This SNR reduction is even larger when the effective compression is increased by a higher compression ratio, or better time or frequency resolution (Naylor & Johannesson, 2009; Alexander & Masterson, 2015; Kowalewski et al., 2020), i.e. at positive SNRs. Improved time resolution will increase the amplification of noise during the speech pauses and improved frequency resolution will increase the amplification in channels without speech. This effect of compression is especially audible during speech pauses when the level estimator approaches the noise floor and increases the amplification of soft background noise.

At negative input SNRs, the effect of compression depends on the noise modulation and reverberation (Naylor & Johannesson, 2009; Rhebergen et al., 2009; Reinhard et al., 2017). For example, if the interfering signal has the same modulation as speech, then the effective WDRC system will improve the output SNR (Figure 6, bottom row). More effective compression given by an estimator with a fast release time will be able to track the low-level signal in the dips of the modulated interference and provide more audibility of the target signal (Moore, 2008). However, if the interfering signal is steady over time (e.g. a steady speech shaped noise, and much louder than the target signal), then it will drive the level estimator so that the output SNR will be close to the input SNR.

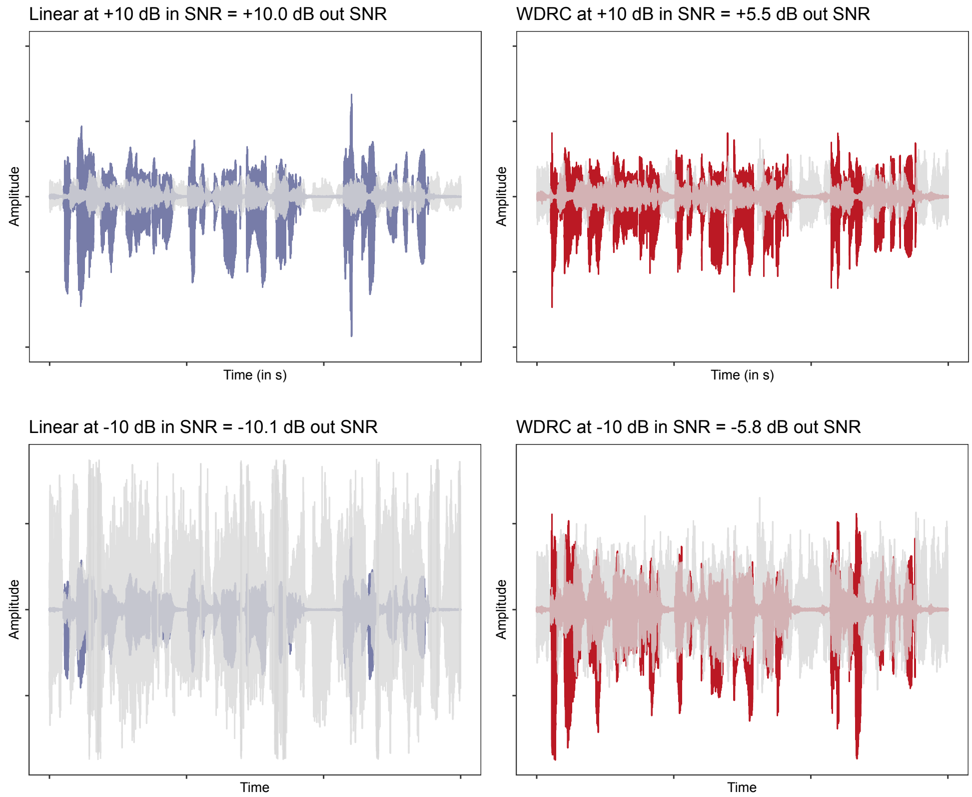

Figure 6: Comparison of the hearing aid output with WDRC (right) and linear amplification (left) with two speech-in-noise conditions. The speech signal is presented at 65 dB SPL (in red or blue) with a steady speech shaped noise at 55 dB SPL (top, in gray) or a babble noise at 75 dB SPL (bottom, in gray). WDRC increases amplification of noise during the speech pauses at positive input SNRs (top rows) but also increases the speech signal presented with a babble noise at negative input SNRs (bottom).

WDRC using estimators with a lower compression ratio, slower time release, or improved frequency resolution provide more comfort for speech-in-noise listening situations.

Steady noise sources might not have the same long-term spectrum that speech in daily life situations has. When the noise source has a motor (car, vacuum cleaner, coffee machine etc.), then the frequency content depends on the engine's rotational speed and all the acoustical couplers that will enhance or attenuate specific frequencies. In this condition, the dynamic compression should have a good frequency resolution to isolate loud noise components in the frequency domain and apply less amplification to the interfering signal. WDRC using estimators with a lower compression ratio, slower time release, or improved frequency resolution provide more comfort for speech-in-noise listening situations (Souza, 2002; Moore, 2008).

The complex interactions between compression architecture (frequency and time resolution), listening environment (noise type and input SNR), and end-user requirement (audibility or comfort) make the interpretation of an evaluation of WDRC systems very difficult. Results from listening tests with hearing-impaired listeners were not able to clearly indicate which processing strategies provide an overall benefit (Moore, 2008; Alexander & Rallapalli, 2017; Salorio-Corbetto et al., 2020).

Additionally, the listener's individual characteristics such as working memory might also explain some variability of test results when comparing different compression approaches.

Benefit of effective compression influenced by human factors

Working memory capacity is another individual factor that might interact with speech intelligibility,

The role of working memory (WM) in speech understanding is as follows: if the language input cannot be matched unambiguously to a phonological representation, then semantic inferences or irrelevant information inhibition might be engaged: WM will focus on the available speech cues to make the best guess regarding the context to decode the degraded language input. This kind of information processing is known as WM which is activated in complex tasks or when speech cues are degraded due to hearing loss or distortions from the hearing aid's signal processing. It is expected that the role of WM is reduced when speech is audible and not distorted (Rönnberg et al., 2013; Souza et al., 2015b).

WM capacity is another individual factor that might interact with speech intelligibility, i.e. listeners with low WM are more challenged by reverberation (Reinhart & Souza, 2016) and with additional background noise (Ohlenforst et al., 2016). While individual WM is not strictly related to the degree of hearing loss or age, it seems to explain some variance when comparing slow and fast-acting dynamic compression. Results suggest that listeners with low WM scores might prefer less effective compression while listeners with higher WM scores might benefit from more effective compression (Arehart, 2013; Ohlenforst et al., 2016; Souza et al., 2019). The hypothesis is that effective compression will provide more audibility especially of the soft phonemes with faster release time constants. The additional amount of information might overload cognitive processes especially when the WM is already challenged by reverberation and background noise. In these challenging situations, listener's with lower WM might rely more on slow variations of the speech envelope as provided by compression with a slow release time (Moore, 2008).

These findings should be used to drive the development of WDRC, e.g. fast-acting compression for listeners with a good WM and slow-acting compression for listeners with poor WM. However, this approach always ends in a compromise, as fast-acting compression might lead to poorer sound quality or slow-acting compression might reduce the audibility of soft signals. Additionally, WM is difficult to measure in a clinical setting and because the compression strategy in the hearing aid cannot be selected from the fitting software, other strategies for the compression architecture must be developed. The idea is not to choose a definitive strategy, but to provide simultaneously for any listener the advantages of fast and slow WDRC in a new approach (Rallapalli & Alexander, 2019; Kuk et al., 2019; Kowalewski et al., 2020).

Hybrid Sound Processing™: one solution optimized for any situation

Hybrid Sound Processing™ was developed to provide the advantage of two compression strategies.

Hybrid Sound Processing™ was developed to provide the advantage of two compression strategies as a function of the input signal in a single solution:

- in the time domain, a phonemic compressor with a fast, broadband level estimator will ensure audibility for any modulated signal like speech;
- in the frequency domain, signals with lower modulation like noise are analyzed with 24 slow level estimators to provide better sound quality.

Analysis in the time and frequency domains are made in parallel (Figure 7) based on the modulation of the incoming signal's envelope. The modulation of the signal's envelope is highly correlated to the amount of information available in the time domain. Soft and loud portions of fast varying signals like different phonemes or sounds in nature (e.g. a bird's vocalization) are estimated independently. This level estimation with a high resolution in the time domain will ensure audibility of soft sounds and comfort for sudden loud sounds. Additionally, the broadband estimator will avoid spectral smearing during the amplification of vowels. Steady or slow varying signals over time with information placed in the frequency domain are estimated with 24 estimators and a lower resolution in the time domain. This approach preserves the temporal envelope at the output of a hearing aid when the level of a signal changes slowly.

The decision unit will merge information from the time and frequency domains. The estimated levels in the time domain are distributed in 24 channels to match the output dimension of the level estimator in the frequency domain and the level to gain functions. If the envelope of the incoming signal starts to vary at phonemic speed, then the estimated levels are adjusted with information provided by the estimator with a greater resolution in the time domain.

While level estimation is optimized for clean speech signals with a fast broadband estimator and for noise signals with slow multi-channel estimators, the situation is more complex for speech-in-noise situations.

With Hybrid Sound Processing™, the noise floor is estimated with a lower time resolution to keep its output as low as possible.

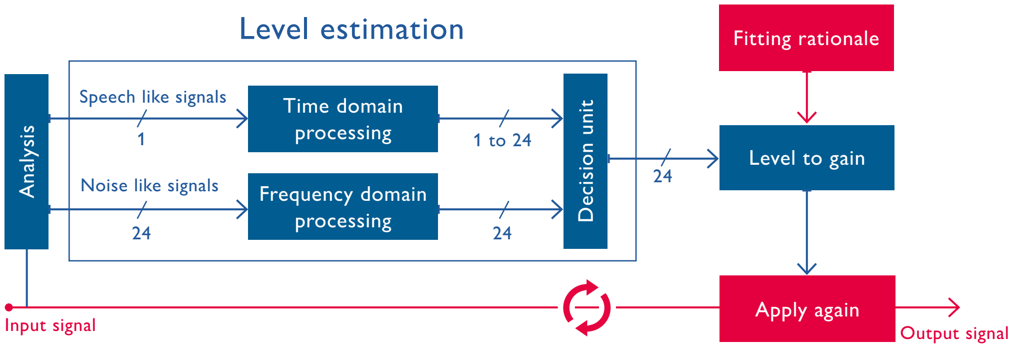

Figure 7: Block diagram of Hybrid Sound Processing™, 24 slow-acting estimators in the frequency domain ensure the accurate amplification of steady sounds while a fast-acting estimator in the time domain provide the necessary audibility for the softest portions of speech-like signals. The information is combined in the decision unit which provides the final estimated level. This level is delivered to the gain unit with the programmed gain from the fitting rationale and the user's specific adjustments.

At positive input SNRs and with an analysis in the time domain, traditional WDRC will increase the noise floor that is still present during the natural pauses of speech. This "speech pause" effect degrades the output SNR and might be perceived as a pumping hearing aid (Dillon, 2012, p. 182). With Hybrid Sound Processing™, the noise floor is estimated with a lower time resolution to keep its output as low as possible while the speech level will be estimated with a higher time resolution to ensure audibility of softer phonemes. This effect is measurable and shown in Figure 8 (figure on the left). For speech presented with a steady noise at +10 dB SNR, Hybrid Sound Processing™ improves the output SNR by 3.3 dB over another WDRC system without compromising the audibility of soft speech; only the noise during the speech pauses is reduced.

Hybrid Sound Processing™ reduces the level of noise and increases the level of speech, especially in the frequency range where less noise is present.

For speech in loud narrowband noise, level estimation with good frequency resolution might be able to isolate the noise source and reduce its amplification. On one hand, amplification of the speech signal in the remaining frequency regions might not be sufficient for the softer phonemes. While, on the other hand, a fast and broadband level estimator will only be driven by the louder steady noise which will reduce the amplification over the entire bandwidth. Hybrid Sound Processing™ will combine the information in the frequency domain, i.e. where the noise is located, while applying phonemic amplification to the speech signal unaffected by the noise. This effect is also measurable and shown in Figure 8 (figure on the right). For this measure, the noise is located between 100 and 500 Hz and presented 10 dB louder than the running speech signal. At the output of the hearing aid, Hybrid Sound Processing™ reduces the level of noise and increases the level of speech, especially in the frequency range where less noise is present. It improves the output SNR by 4.8 dB SNR and will provide more audibility of the speech signal.

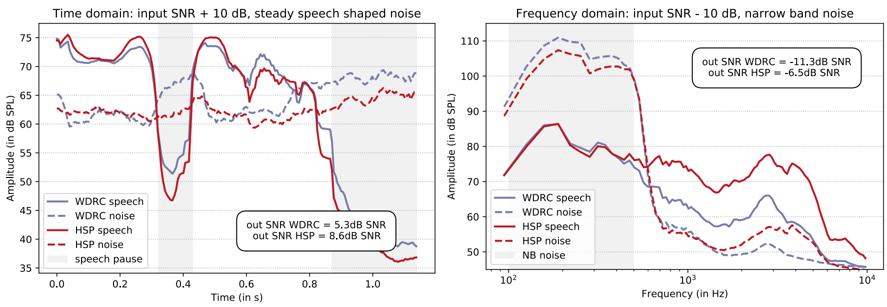

Figure 8: Comparison of the behavior of fast-acting WDRC (blue) and Hybrid Sound Processing™ (red) for speech in steady noise at +10 dB input SNR (left) and for speech in narrowband noise at -10 dB input SNR (right). Noise location is highlighted with the shaded area. Speech and noise signals are extracted with the inversion technique (Hagermann & Olofsson, 2004).

The differences between processing as shown in Figure 8 are used to illustrate the behavior of different dynamic range compressions. Speech and noise, measured simultaneously at the output of the hearing aid, are separated using the inversion technique proposed by Hagermann & Olofsson (2004). This objective measurement allows repetition of the test with different noises at different SNRs to get a holistic understanding of potential differences between the tested processing strategies (Naylor & Johannesson, 2009; Rhebergen et al., 2017; Miller et al., 2017; Lesimple & Sans, 2018).

The principle of the inversion technique is to make two successive recordings with the selected signals that you want to separate. While these signals are presented simultaneously, the level of each one can be adjusted to cover a defined range of input SNRs. The first recording has both original signals and the second one, the speech signal inverted. In a post-processing phase, the recordings are combined to isolate each signal of interest. The estimated level of the extracted speech and the extracted noise can be finally used to estimate the hearing aid output SNR.

The International Speech Test Signal (ISTS) (Holube et al., 2010) was presented at 65 dB SPL to engage the WDRC for soft and loud phonemes. Three types of noises were selected to represent different listening situations: a speech-shaped noise (SSN) with the same long-term average spectrum as the speech signal, a low-frequency narrowband noise between 100 and 500 Hz (LF NBN), and a mid-frequency narrowband noise between 1 and 2 kHz (MF NBN). The input SNR was defined to cover a test range from -10 to +20 dB SNR in 5 dB steps. Both signals were presented from the front in a sound isolated test box to the test hearing aid mounted on an ear simulator (IEC 711).

The hearing aids were fitted to provide a 10 dB flat insertion gain between 250 and 4,000 Hz for a 65 dB input speech signal. The compression ratio was set to 2:1 over the entire bandwidth and the effective gain for a clean speech signal was verified with a test box measurement using the ISTS signal. Further adaptive features (e.g. noise reduction, directionality, feedback canceller) were disabled during the recordings. Output SNRs were then computed for the Hybrid Sound Processing™ and a fast-acting WDRC system (Figure 9).

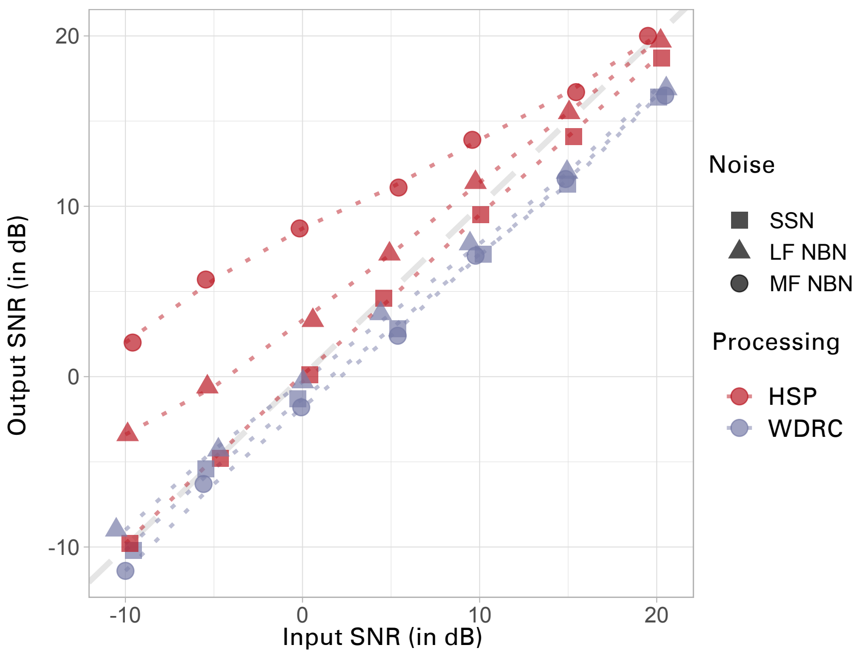

Figure 9: Results of output SNR measures with Hybrid Sound Processing™ (red) and a fast-acting WDRC (blue) with different noise types (see shape legend) as a function of input SNR.

Hybrid Sound Processing™ will provide audibility of soft phonemes and avoid spectral smearing for a cleaner speech signal, improve the output SNR for any steady noise, and preserve the contrast between the speech envelope and the noise.

The output SNR measure shows the expected SNR loss with dynamic compression especially at the positive input SNRs. This SNR loss can be observed with all the noise signals used for these measurements and the fast-acting WDRC. Hybrid Sound Processing™ reduces this loss for the SSN and the LF NBN conditions and even improves the output SNR for the MF NBN over the entire input SNR range. There are no differences for the SSN condition at negative input SNRs as the estimator will be driven by the steady noise signal. However, the effect of Hybrid Sound Processing™ is clear at -10 dB input SNR with a narrowband noise: + 5.2 dB SNR for the LF NBN condition and + 13.0 dB SNR for the MF NBN.

The interpretation of these results must be nuanced as the effect of the WDRC instrument in real fittings might be reduced by the amount of direct sound that will enter the ear canal naturally with open fit acoustics, the effect of noise reduction or directionality, and by individual fitting settings like compression ratios and frequency specific gain (Lesimple & Sans, 2018). However, Hybrid Sound Processing™ will provide audibility of soft phonemes and avoid spectral smearing for a cleaner speech signal, improve the output SNR for any steady noise, and preserve the contrast between the speech envelope and the noise. This solution was designed to get the best out of a fast-acting broadband level estimation system combined with a slow-acting multichannel compression system in many listening situations for the hearing aid user.

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World Headquarters

Switzerland
Bernafon AG
Morgenstrasse 131
3018 Bern
Phone +41 31 998 15 15
Fax +41 31 998 15 90